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Chapter 1

Using Logic

This chapter summarizes Logic's general operating functions. But first, a quick word about this manual.

1.1 Conventions of this Manual

Menu Functions

Menu functions are written in this font: **Function**.

If the text is describing functions which can be reached via hierarchical menus, the different menu levels are described as follows: **Menu > Menu Entry > Function**.

Key Commands

When a function can be operated by a key command of the same name, you will see this symbol at the side of the text. If names differ, or if a function is only available as a key command, its name will be printed like this: *Key Command*.

All of Logic's key commands can also be accessed via MIDI commands (for more on this, see the section *Key Commands and Remote Control via MIDI* on page 1-44).

Options and Parameters

The options you can set from the Preferences or Song Settings, and the parameters in dialog boxes are printed like this: *Parameters*.

Different parameter values are printed like this: *Parameter value*.

Detailed Explanations

Hints, tips, examples, or information mainly of interest to the more technically-minded are printed like this.



Sometimes, detailed explanations will be given for things which you don't necessarily need to know about to understand how to use Logic. These are printed in this smaller font.

1.2 Dialog Language and Look

Interface Language

Logic supports multiple languages from the menu **Options > Settings > Display Preferences**. Switching to another language requires that you re-launch Logic.

The preset “Default Language” automatically selects the language of your operating system (or, to be more precise, the language of your keyboard layout), if Logic supports that language; otherwise English will be used as the default.

Logic's Look

You can, if you prefer, continue to work with the familiar look & feel of Version 3.x. Go to **Options > Settings > Display Preferences** to switch looks.

1.3 The Mouse

Basic Functions

Clicking

Place the mouse pointer on the object (button, input field, etc.) and press the mouse button once.

Double-clicking

The same as clicking on an object, but you press the mouse button twice, in quick succession. You can set a comfortable interval for this in the Finder (>Mouse< control panel).

Grabbing or Clicking and Holding

The same as clicking on an object, but you keep the mouse Button held down.

Moving or Dragging

Grab the object and move the mouse (keeping the mouse button held down) to the desired position.

Mouse Input

Checkboxes

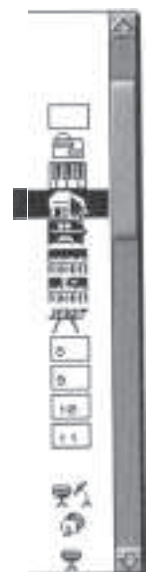
Checkboxes are square boxes which become “checked” when you click them to activate an option (or function). Click them again to remove the “check”, and deactivate the option.

Pull-down Menus

Pull-down menus open when you grab certain input fields or buttons. You select a command by moving the mouse onto the desired item. If you want to select an item which is outside the visible section,

move the mouse over the top or bottom edge of the menu; the further you move it, the faster you will scroll through the menu.

hold down the  key as you do this... You can then let go of the mouse button and use the scroll bar on the right of the pull-down menu to scroll through it. When the mouse reaches the entry you want, release the  key.



Mouse as Slider

You can set practically all the numerical parameters, (even note values or names) by grabbing the parameter value, and moving the mouse up or down. If the parameter is made up of several separate numbers (e.g. song position), you can adjust each number individually.

Using the Mouse for In/Decrementing

All the parameter values which can be set using the mouse as a slider (and even some of the flip menu parameters) may also be increased or decreased in single units by clicking on the top or bottom half of the value, while holding down the  key.

Numerical Input

Double-clicking on a numerical parameter value opens an input field. The previous value appears pre-selected, (i.e. highlighted), to allow it to be overwritten by a new entry. You can also use the mouse to make a partial selection in an input field, so that only the highlighted part is overwritten (for more on this, see the section *Numerical Value Input* on page 10-7). As long as the input field is open, all the keys may be used only for inputting data, and may not be used for key commands (the exceptions are the main menu functions).

...by arithmetic

You can enter numbers at any time by typing in an arithmetical operation, e.g. “+2” or “-5”, which then simply changes the current value by that amount.

...as ASCII-symbols

You can also input numbers as ASCII symbols: just put a ‘ or ! in front of it, and the ASCII code will be input as a number. For example:

“! gives 33

“a gives 97

This function is particularly useful for entering text in SysEx strings.

...Cancel Numerical Input

Direct numerical input can be canceled by entering no text at all and .

Text Input

You input text names in the same way as numbers, but you only have to click a name field once to allow input.

Numbered Names

As you might expect, you can give any number of selected objects the same name. If the name ends in a number, the number will automatically be incremented by 1 for each object. This allows you to name all the sequences on one track, or all the faders in the Environment quickly.



Disabling automatic numbering

To turn off automatic numbering, place a space after the number at the end of the name. All the selected objects will then end in the same number.

Tools and the Toolbox

Logic allows recorded data to be handled graphically. This means that you don't have to carry out operations by inputting instructions in command lines, but rather by manipulating graphic "objects".

When editing objects graphically, you always have two tools available at the mouse pointer position: one is already active, the other is activated by holding down the  key.

You change the currently active tool by clicking on the toolbox. The mouse pointer then adopts the shape of the tool you click on, so that you can tell what its function is by looking at the symbol: the Eraser is used for deleting, the scissors for cutting, and the glue tool for merging objects. To assign a tool to the  key, click on the desired tool in the Toolbox, with the  key depressed.

Below is a diagram of an artificial toolbox, containing all Logic's tools. Since you don't need all the tools in all the windows, the toolboxes in the various windows will contain an appropriate selection of the tools displayed here.



upper row

pointer	hand
pencil	Eraser
scissors	glue tool
text marker	crosshair
layout pointer	MIDI Thru tool
size tool	voice splitter
Magnifying glass	Camera
Solo	Quantize
Mute	Velocity

lower row

Effective Range of the Tools

Tools are effective only in the working area of the window they were selected from (you can define individual tools for each opened window).

A tool (e.g. the scissors) basically affects the objects you click on. If this object was selected together with other objects, all

selected objects will be affected by the tool. (E.g. the scissors would cut all selected objects at the same song position.)

Hiding/Showing the Toolbox

The Toolbox can be hidden in the Arrange and Score Editor windows to save space, which can be very helpful if you only have a small monitor. The function is **View > Hide/Show Toolbox**.

Selecting Tools

You select the tool you want to use by clicking on it in the toolbox (or clicking while holding down the  key for the alternate tool).

If a toolbox is opened at the mouse position, you can also use one of the number keys to select a tool at the corresponding position. The tools are always numbered from left to right and top to bottom. Press the Show Tool's key command again to switch to the pointer, and close the box.

Moving to the Next Tool

Use *Set next Tool* and *Set previous Tool* to switch to the neighboring tool in the top window.

Opening the Toolbox at the Mouse Position

Use *Show Tools* (default: ) to open a toolbox at the mouse position. This window will disappear as soon as

- you select a tool by mouse click,
- you select a tool by pressing a number key,
- you click anywhere outside the tool window or
- press any key.

Info Line

When operating many of the tools an info line appears at the top edge of the window for as long as the mouse button is held down. The info line replaces the local menus in the window. This info line provides useful feedback about the type of operation you are performing.

During operations involving arrange objects, the line will look something like this:

Divide	3	1	1	1	SequenceName	3	4	0	0	0

From left to right, the readouts are: operation, mouse (or arrange object) position, arrange object name, track number, and length of the arrange object.

During operations involving events, the line looks something like this:

Length Change	1	1	1	1	NOTE	1	C1	70	1	0	0	0

From left to right, the readouts are: operation, mouse (or event) position, event type, event MIDI channel, first data byte (i.e. the pitch for notes), second data byte (i.e. velocity for notes), and for notes: length of the note.

The Tools

Pointer

The pointer is the default tool. The mouse also takes on this shape outside the working area, when you are choosing from a menu or inputting a value. Within the working area the pointer is used for selecting (by clicking on objects), moving (by grabbing and dragging), copying (by holding down the **Alt** key and dragging), and editing lengths (by grabbing the bottom right corner and dragging). Grabbing and dragging anywhere on the background opens a “rubber band” (see page 1-24).



Pencil

The pencil is used to add new objects. You can also select, drag, and alter the length of objects.



Eraser

The Eraser deletes clicked objects. When you click on a selected object, all of the currently selected objects are deleted (as if you had used the  key).



Text Tool

The text tool is used to name arrange objects, or add text to a musical score.



Scissors

The scissors are used to split arrange objects, e.g. before copying or moving individual sections (see page 3-24).



Glue Tool

The glue tool is the opposite of the scissors: all selected objects are merged into a single object, which is given the name and track position of the first of the objects on the time axis (more information can be found on page 3-27).



Solo Tool

Grabbing with the solo tool allows you to listen to only selected objects during playback. Moving the mouse vertically also outputs any events the cursor touches, even when the sequencer is stopped (please refer also to the section *Soloing Sequences or Regions* on page 3-28).



Mute Tool

Clicking on an object with the mute tool stops it from playing, and places a dot in front of its name, to indicate that it is muted. You can unmute it by clicking it again with the tool. If multiple objects are selected, the setting of the object you've clicked on

applies to all selected objects (see also section *Muting Sequences or Regions* on page 3-28).

Magnifying Glass

The magnifying glass allows you to zoom in on a “rubber-banded” section, right up to full window size. You revert to normal size by clicking on the background with the tool (for more on this, see the section *Zooming Selectively* on page 1-15). You can also access this function from other tools by holding down the  key.



Finger

The Finger is used in the Matrix Editor to alter note lengths, and, in the marker list window to jump to the markers, and position the locators to the marker position simultaneously.



Crosshair

The Crosshair is used in the Hyper Editor to input a linear series of MIDI events.



MIDI Thru tool

Clicking on an instrument in the Environment with the MIDI Thru tool, assigns that instrument to the selected track in the Arrange window, thereby making it the active MIDI Thru instrument.



Layout tool

The Layout tool is used for graphically moving objects in the Score Editor to optimize the display (e.g. bars to lines: “Local Formatting”), without altering the timing of actual MIDI events.



Size Tool

The Size tool is used to adjust the size of graphic elements in the score; for this reason, it looks similar to the bottom right corner of a Mac window, which is also used for resizing.

Voice Separation Tool

You can separate polyphonic voices onto different staves in the Score Editor, by drawing a dividing line with the voice separation tool, provided you are using a polyphonic score style.



Camera

You use this tool in the Score Editor to select and export sections of the display as graphics files.



Quantise Tool

You use the Q tool in the note editors to quantize notes to the most recently set value.



Velocity Tool

In the note (i.e. Matrix and Score) editors), you can use the V tool to change the velocity of notes.

1.4 Window Functions

The basic functions of the Logic windows are the same as those in other Macintosh application programs. However, the display options in Logic's windows are far more extensive.

In Logic, you can open different combinations of windows (even several of the same type) and adjust each one individually. All open windows in a song are constantly updated. This means that the windows update to follow the song position, and any alterations that you make in one window immediately update the display of all the other windows you are working with. It is also easy to save different window arrangements (called "screen sets"), and recall them with the push of a button.

Working with Windows

Opening Windows

All Logic windows can be opened from the **Windows** main menu. After each menu item you are given the relevant key command (e.g. *Open Arrange*  ) , which allows you to open the window without using the mouse. You can open as many of the same type of window as you like.

Every time you use the Windows Menu have a quick look at the key commands-that way you will soon memorize them . You may also define your own personal key commands.

Closing Windows



You close windows ( ) by clicking on the close symbol in the top left of the windows. If you hold down the Command key as you click, all the windows of the active song will be closed. Hitting the  key again, closes all the windows of all the songs currently loaded.

Moving up One Level in the Display Hierarchy



Clicking the black box in the line below takes you to the next highest display level. For more on this, read the section *Changing Display Levels* on page 3-57.

Setting Window Size



You adjust window size by pulling the lower right-hand corner of the window, as with any window in the Finder.

Maximizing window size



The windows maximize button, in the upper right corner of the window, toggles the window between its

maximum size, and the size it was before you clicked on the button (**Window > Zoom Window** or ).

Selecting the Working Area

Scroll Bars

The scroll bars are situated at the right and bottom edges of a window, if you can only see a portion of the total working area in either the vertical or horizontal dimension.



You can move the visible section by clicking the arrows, or grabbing and dragging the scroll slider. There are two points of note:

The size of the scroll slider in relation to the size of the entire scroll bar corresponds to the size of the visible section in relation to the overall size of the window, the visible section changes as you move the scroll slider.

X/Y Element



The X/Y element is situated at the bottom left corner of the window. By grabbing and dragging it, you can move the horizontal and vertical window section, as if you were dragging both scroll bars simultaneously.

Page Scrolling

Use the key commands *Page Up*, *-Down*, *-Left* and *-Right* to scroll one page up, down, left or right, as if you had clicked in the grey region above / below the vertical scroll bar, or to the left / right of the horizontal scroll bar. The key commands *Page Top*, *-Bottom*, *-Left most* and *-Right most* take the visible section of the working area to the top, bottom, left or right, just as if you had grabbed one of the scroll sliders, and moved it to one of its extreme positions.



Scrolling to selected Objects

In the Arrange, Event List, Hyper and Matrix Editors, **View > Scroll to Selection** allows you to move the visible window section to the first set of selected events. This function is available as a key command from the “Various sequence editors” area. The key command works in the currently-active window.



Zooming

The telescope-shaped zoom symbols at the top right of the window are used to zoom in and out of the working area display. Clicking the left, smaller side of the telescope symbol reduces the size of the objects in the display, allowing you to see more objects in the same space (zoom out) and clicking the right, larger side enlarges the objects in the display (zoom in).



The telescope with the downwards-pointing arrow is for vertical zooming (Zoom *Vertical In/Out*, default assignment: /, while the one with the sideways-pointing arrow handles horizontal enlargement (Zoom *Horizontal In/Out*, default assignment /). In some windows, only one telescope is available, and this then handles both horizontal and vertical zooming at once.



While “zooming”, the top left selected object is kept in the visible region, whenever possible.

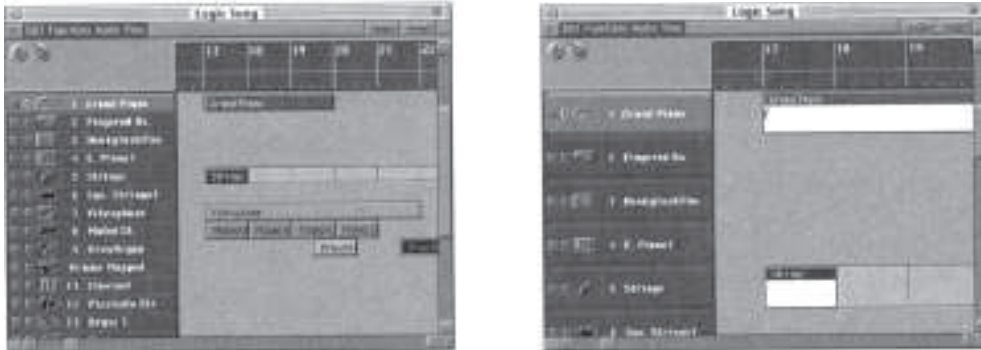
*You can zoom tracks individually in the window. See the section *Zooming Individual Tracks* on page 3-9.*

Storing and Recalling Zoom Settings

You can store three different zoom settings for each window using the key commands *Save as Zoom 1-3*. Use *Recall Zoom 1-3* to call them back up. These commands always apply only to the top window.



Zooming Selectively



Zooming in on One Section of the Screen

To enlarge a section of the screen to the size of the whole window, use the magnifying glass tool to drag a “rubber band” over the section you want. You can do this more than once.

Reverting to the Previous Zoom Setting

Click on the background with the magnifying glass. This will return the zoom to the original setting, by backtracking through the previous steps.

You can call up the magnifying glass functions with any other tool (apart from the pencil) by holding down the **ctrl** key. The pointer will display the previous tool, until you press the mouse button. As long as the button is pressed, it will display the magnifying glass.

Window Elements

Adjusting the Size of the Window Elements



If you move the mouse over the top left corner of the arrange area the mouse pointer turns into a crosshair. By dragging it, you can now adjust the size of the bar ruler, arrange area, track list, and transport buttons. You can also adjust the window elements in the Score, Hyper, and Matrix Editors using the same method.

Concealing/Revealing the Transport Functions

Use **View > Show Transport** to display or remove the transport panel buttons from the top left corner of the Hyper, Matrix or Arrange windows. The number of buttons and indicators depends on the amount of space available (see the section *Adding the Size of the Window Elements* on page 1-15).

Concealing/Revealing the Parameters

The function **View > Show Parameters** (*Hide/Show Parameters*) allows you to display or remove the entire area to the left of the Arrange, Environment, and other Editor windows, which contains the sequence parameters, instrument parameters, and the toolbox. Hiding these parameters gives you more space for the editor window itself.



In many windows, you can hide and/or reveal further screen elements, such as the toolbox. These display options are always available from the View menu.

Menus

Because of Logic's great range of functions, most of them are not found in the main menus, but appear as local menus. These are always in the menu bars of the Logic windows where they are required.

Hierarchical Menus

Logic's menus (whether main or local) are accessed in the same way as in other application programs: grab the menu title, move the mouse over the desired item, and release the mouse button. In hierarchical menus, there is a right-pointing arrow after the item. If you highlight this item, a sub-menu drops down to the right. To select an item from this sub-menu, move the mouse to the right, into the sub-menu, and then vertically over the desired item. Releasing the mouse button will activate the selected function.

Permanent Drop-down Menus

If you give any of the local menus a short click, they will stay open even after you have released the mouse button. If you give them a long click, they will behave as described above.

 This also applies to the main menu controlled by the operating system in MacOS 8.

Menu Options with “...” in their Title

Three periods, like this: ..., next to the menu function title indicate that the entry does not activate a function immediately, but instead opens a dialog box.

Dialog Boxes

In Dialog boxes, you can hit the thick-bordered button by pressing the  key, or by clicking with the mouse.

Window Types

There are two different types of window in Logic: normal windows and float windows. The contents of all the windows belonging to the current song are always updated, no matter what the type of window.

Normal Windows

You can open as many normal windows as you want, including several of the same type. Even though the contents of all the windows is constantly updated, only one of the windows ever has the status of being the “top”, or “active” window. This is the window which is in the foreground when several normal windows are overlapping.

Top or Active Window

This window can be recognized by its fully-shaded title bar. The main distinguishing characteristic of this window is that key commands only affect this window, and not any of the

others. **Windows > Next Window** () brings the next window forward, if it is fully covered up by others.

Background Windows

Here you can not only observe changes, but make almost any kind of change, without having to bring the window into the foreground, simply by planting a long click on it. Background windows are characterized by a white title bar (they can be positioned next to the top window, not just tiled underneath it). You bring a window to the foreground by clicking on it, clicking on the title bar, or calling up one of the local menu functions.

Float Windows

Float windows are so named because they always “float” in the foreground, even above the top normal window (if there are too many open float windows, they will inevitably cover each other up—just click on one to bring it to the front). Float windows are recognizable by their narrower title bar, which contains no name. Mouse operations can be carried out in the same way as in normal windows.

The most often common example of a float window is the Transport window. However, you can open any other window as a float window by holding down *Alt* when you open the **Windows** menu.

Relationships between Windows

The two buttons at the top left of a window (shown here), determine its relationship to the song position (Catch), or to other windows (Link, Show Contents, or Contents Catch).

Catch

The Catch function means that the visible section of a window follows the song position as the song plays.



If the button with the walking man on it is lit (i.e. if it is blue), the window's display follows the song position as the song plays. If the button is not lit, the display does not update, even when the song position line moves past the right edge of the visible portion of the window(*Catch Clock Posion*).



Automatic Catch Disabling

If you move the visible section manually, Catch is automatically switched off, so that the new section you have chosen doesn't then disappear, the display is always updated to the song position line.

Autocatch

The function *Enabe Catch when Sequencer starts* (**Options > Settings > Global Preferences**) always enables Catch mode whenever you press play or pause.

Link, Show Contents, and Contents Catch

You can define these display options to control how information is displayed when working with related editor windows.

Link

When the button with the chain link icon is activated (i.e. when it is violet), this window always displays the same contents as the top window. The display is adjusted whenever the selection in the top window is altered.



Here's an example: imagine the top window is an editor. In Link mode, the other editor windows can display the same data in another form (though please remember: you cannot have any event display as the background window of an Arrange window while in Link mode).

Here's another example, using the Environment window; The top window is the Arrange window. In Link mode, the Environment window will display the instrument that corresponds to the selected track in the Arrange window. As you switch tracks

in the Arrange, the Environment will update to reflect the selection.

Contents Link

Double-clicking on the link button (yellow) activates Contents Catch mode. This means that the window always shows the contents of the object selected in the top window. The display is therefore always one level below that of the top window.



Here's an example: if the top window is an Arrange window, in Contents Link mode the editor windows can show the events of a selected sequence. Selecting a different sequence in the Arrange window will cause the display of the linked editor to switch to that sequence, as well.

You could also use Contents Link mode in an Arrange window, to display the contents of the folders in another Arrange window.

Contents Catch

By simultaneously switching on Catch and Contents Link, you activate Contents Catch mode. Initially, this is equivalent to Contents Link mode, but when the song position reaches the next object on the same track, the contents of *this* object are then displayed.



You could use this mode in an Arrange window, much as you would in Contents Link mode. The editor windows would then show the events of whichever sequence is currently being played on a track.

Screensets

Normally you will lay out your windows on the screen in a way that suits your way of working. This layout with various windows, with all their different parameters (display, zoom, position of each window) is called a screenset, and can be

stored. You can then swap between different screensets, much as you might swap between different computer monitors.

Storing Screensets

Screensets are numbered from 1-99 using only the numbers 1-9. You can see the number of the current screenset in the main menu next to the word **Windows**.

You don't have to save screensets with an explicit command. It happens automatically, as soon as you switch to another screenset. Thus, without any effort, your current working view is always stored in the current screenset.

Switching Screensets

Just input the number of the desired screenset (1-9). For two-digit screensets hold down the  key while inputting the first digit.

Protecting Screensets

Use the key command *Lock/unlock current Screenset* or **Windows > Screen Sets > lock Screenset** to protect the current screenset from being altered. A "." then appears in front of the screenset number. Repeat the key command or menu command to unlock the screenset.



The New Song command deactivates all Screenset locks.

Copying Screensets

To copy the current screenset to a destination screenset, hold down Shift when you switch screensets. Two digit screensets are copied with    +  ...  + .

Logic offers a dialog box if you copy Screensets.

You can also use menu functions to copy Screensets via clipboard: Switch to the Screenset you would like to copy and select **Windows > Screen Sets > Copy Screenset**, then switch to

the Screenset you wish to copy to, and select **Windows > Screen Sets > Paste Screenset**.

You can also copy individual Screensets between different songs using this method.

To import all Screensets from another song it's so much faster to use "Import Settings".

Reverting to a Stored Screenset

The command *"Revert to current screenset"* resets your screen to the way it was when you called up the current screenset.

Sequencer-controlled Switching

You can automatically switch screensets using meta event # 49-just add it to a sequence in the Event Editor.

Set the song position to the point where you'd like the screenset to change.

Hold down the  key and click in the Event Editor on the button shown. The inserted meta event has the default value 50 (*Song Select*).

Alter the number in the **NUM** column from 50 to 49. This changes the name to *Screenset*.

Input the desired screenset number in the data by the column (VAL).

You can stop the switching by muting the sequence that contains the Meta 49 event.



Other Functions

The **Windows > Tile Windows** function, tiles all the open windows on the screen and makes them all the same size. The **Windows > Stack Windows** function stacks all the open windows on top of one another, so that at least a part of the title

bar is visible for each window, no matter which is in the foreground window at the time. The function **Windows > Tile Windows horizontally** tiles all open windows on the screen, using the available horizontal space for each individual window (if possible).



1.5 Selection Techniques

Whenever you want to carry out a function on one or more objects, you have to select the object(s) first. This applies to arrange or environment objects, and individual events alike. Selected objects are either displayed in inverse color, or will flash (in the Score editor).

The selection status of an object applies to all windows. An object selected in one window will also be selected in all other windows that display that object. Changing the top window doesn't affect the selection (as long as you don't click on the background, which deselects everything. Be sure to click on the window's title bar).

Selecting Individual Objects

Individual objects may be selected by clicking on them, and deselected by clicking in the background, or by selecting another object.

You can also use the key commands *Select next/previous Object* *Select first/last* selects the first/last object in the current display level.



Selecting Alphabetically

The  key selects the next alphabetical object. In the Arrange or Environment windows, pressing any letter key selects the first object whose name begins with this letter (as in the Mac's Finder), providing there is no key command assigned to this key.

Selecting Several Objects

To select several non-contiguous objects, hold down S as you click them. As subsequent objects are selected, the previous selections are retained. This also works with horizontal or rubber band selection.

Use the key commands *Toggle next/previous Object (Event)* to select the following/previous object (or event) as well.

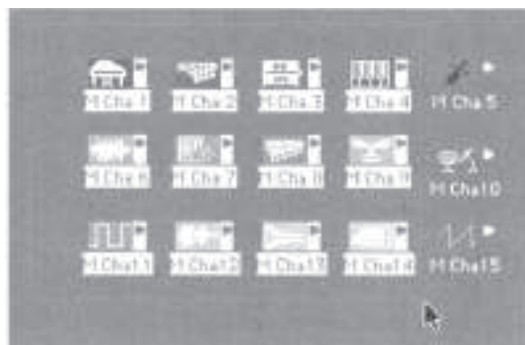
Horizontal Selection

To select all objects on a track, click on the track name in the track list. In the same way, you can select all events with a certain event definition in the Hyper Editor, by clicking on the event definition name, or all notes of a certain pitch in the Matrix Editor by clicking the relevant key on the screen keyboard.

In Cycle mode, the above selects only the events within the Cycle zone.

Rubber-Banding

To select consecutive objects, click on the background and drag a “rubber band” over them.



All objects touched, or enclosed by the rubber band will be selected.

Toggling the Selection Status

When you make any selection (including by rubber band or horizontal selection), holding down the  key at the same time will reverse the selection status of the objects.

You can reverse the selection status of all objects using **Edit > Select > Toggle Selection**. For example; if you want to select all objects except for a few, first select these few and then use **Toggle Selection**.



Selecting Following Objects

To select all objects after the current one (or, if no object is currently selected, to select all objects after the song position), select **Edit > Select > Select all following**.



Selecting Objects within the Locators (Vertical selection)

Edit > Select > Select inside Locators selects all objects lying wholly or partly within the locators.



Deselecting Objects outside the Locators

Edit > Select > Deselect Outside locators deselects all objects outside the locators: objects within them remain unchanged.



In the Score window, this command works on whole sequences, as well as events.

This command is particularly useful after using the Select Equal or Select Similar commands, to restrict the commands' effects solely to the area between the locators.

Deselecting Objects outside the current Track

The ***Deselect All Objects Except Recording Track*** key command deselects any objects not on the currently selected track.



This command is especially useful after other special selection commands, ensuring that the commands' effects are restricted to the recording track only.

Selecting Empty Objects

Edit > Select > Select empty Objects selects all empty objects.



Selecting Overlapping Objects

Edit > Select > Select overlapped Objects selects all overlapping objects.

Selecting Similar or Identical Objects

If you have selected an object, you can use the function **Edit > Select > Select Similar Objects** to select all similar objects, and the function **Edit > Select > Select Equal Objects** to select all equal objects.

The table shows what counts as similar or equal:

Object	Similar	Identical
Controller Event	Controller# equal, any data byte	Controller# and data byte (control value) equal
Note Event	Note equal, any octave	Note and octave equal
Environment Object	Same object type (e.g. fader)	-Same fader type (e.g. text)

If you want to delete all similar objects, select one of these objects and select *Delete similar Objects*.

Conversely, if you want to keep all similar objects and delete all the other objects in this sequence, select *Delete but keep similar Objects*.



Selecting All Objects

To select all objects, select **Edit > Select All** or press .



Deselecting All Objects

You can deselect all objects by clicking on the background, or using the key command *Deselect All*.

1.6 Edit Operations

The local edit menus in Logic's various windows all take the same form. The first item is Undo. Below Undo are the typical clipboard functions, and at the bottom of each are the main selection commands.

Undo

Undo allows you to reverse the previous edit. In the Global page of the Preferences, you can disable the warning message that normally appears when you activate Undo, by checking the *Disable safety alert for Undo* box.



The default key command for Undo is  .

The Clipboard

The clipboard is an invisible area of memory, into which you cut or copy selected objects, so that you can paste them to a different position.

The clipboard spans all songs, which means you can use it to exchange objects between songs.

Cut

All selected objects are removed from their current position, and placed on the clipboard. The previous contents of the clipboard are overwritten in the process (key command Command X).



Copy

A copy of all selected objects is placed on the clipboard. The selected objects are left in place. Here too, the previous contents of the clipboard are overwritten (key command  .



Paste

All objects from the clipboard are copied into the top window. The clipboard is not erased in the process (key command  ).



The contents of the clipboard are added at the current song position (if they are events or arrange objects). The song position is incremented by the length of the pasted objects.

In the Arrange window, the contents of the clipboard are pasted to the selected track. If events are pasted in the Arrange window, either a new sequence is created for them, or the events are added to a selected sequence.

Any previously existing objects are unchanged.

In the Environment window, the objects are pasted into the layer currently being displayed, at their original position.

Clear

Any selected objects are erased. Clear has no effect on the clipboard, and is the same as pressing the Backspace key.



Delete and Select next object

This key command erases the objects currently selected in the Arrange and Editor windows, and selects the next object.



Paste at Original Position

This function works in a similar way to Paste, but the objects on the clipboard are always pasted to the position they were cut from, regardless of the current song position (which is where the objects would be placed with the ordinary Paste function).



Paste Replace

This function only works in the Arrange and Editor windows, and again, resembles Paste, except that with Paste Replace, all existing events are replaced by the objects being pasted.

When using this function, all the arrange objects or events that lie within the time period occupied by the objects on the clipboard are erased.

The effect of this function can be carried out by using two of Logic's other functions in combination, one after the other: first a normal Paste and then (in the Arrange window) Replace Overlapped Objects (or in the Editors) Erase Unselected Events within Selection. This gives you the option of retaining some of the old objects by Manually selecting them, before carrying out the second function

1.7 General Functions of the Editors

Opening Editors

A  button double-click on a sequence opens the Event editor. There is an option on the page **Options > Settings > Global**, which lets you select the editor that will be opened when you double-click on a sequence.

The options are: the Score Editor, the Event Editor, the Matrix Editor, and the Hyper Editor.

Control Output via MIDI

Switching on the MIDI Out button causes MIDI events to be output when they are added, selected, or edited. This allows you to audibly monitor every editing stage, whether you are scrolling through the Event List (automatic selection) or transposing a note.



Automatic Scroll Functions

...scrolling to the Song position

The button with the walking man on it activates the Catch function, which means that the window view will always show the current song position.



...scrolling to the selected event

The key command *Scroll to Selection* allows you to automatically move the window, so that the first of the currently-selected events is visible.

Scroll in Play

The “View” menus of all windows that display time horizontally (Arrange, Score, Matrix, Hyper), offer the option **View > Scroll in Play**. If the “Catch” function of the particular window is also activated, then the SPL will stay in the middle of the window, while the background scrolls smoothly from right to left.

Please keep in mind that Scroll in Play requires a powerful graphic card, and a fast computer to operate smoothly.

Contents Link

Clicking on the button with the chain-link symbol on it activates the link function (the button will turn violet), and double-clicking it activates the “show contents” function (the button will turn yellow, with an arrow). In “show contents” mode, the window always displays the contents of an object selected in the top window; in link mode the window shows the same objects as the window where you are making the selections.



For a full description of the catch and link functions see page 1-18.

In Editor windows, contents link is usually active. For example, you can then select some sequences in an arrange window, and



when you switch to a screenset with an opened editor window, you will see the contents of the selected sequence(s).



Alternative Play Function

In all windows with a horizontal time axis (Arrange, Matrix, Score, Hyper, Sample) you can use the global command *Play left window corner* to play from the beginning of the visible area.



Selection Commands & Editing Functions

Selecting Events with same MIDI channel

If you've selected an event, you may select all the other events on the same MIDI channel by using the command **Edit > Select > select equal channels**.



Imagine you're editing a sequence that contains volume and pan controller information for 16 MIDI channels. To select all the events on channels 1 and 3, you simply grab one event which is on channel 1 and one which is on channel 3, use the "Select Equal Channels" function, and all the other events on these two channels will be selected.

"Event channel +1" or "Event Channel -1" alter the channel number of the selected event by one. The commands are covered in the section "Arrange and various Sequence Editors".



Select Equal Subpositions

You can select all events with a certain relative position, such as all snare drums on the off-beat.

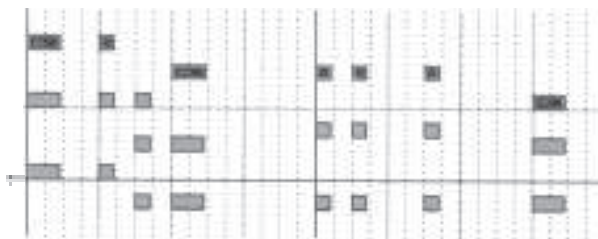
Select an event at the desired relative position, and then **Edit > Select > Select Equal Subpositions**. All events, in all bars within the sequence, with the same relative position in a bar, are selected.



You can use this function for up to 10 relative positions simultaneously.

Splitting Chords

In the “Functions” sub-menu in the Editor windows, there is a new function called “Note Events”. This allows you to manipulate selected notes in a manner influenced by the notes surrounding or overlapping them; musically speaking, you can think of this function as adding division lines within chord sequences. These functions can be very useful if you want to set up a polyphonic display within the Score window.



Selecting the Soprano voice

The command **Functions > Note Events > Select top Line** selects the highest notes in the selected chords of a sequence. All previous selections are ignored.



Selecting the Bass voice

The command **Functions > Note Events > Select Bottom Line** selects the lowest notes in the selected chords in a sequence. All previous selections are ignored.



Select *Top Line* and *Select Bottom Line* are purely selection commands, which can be used in conjunction with any of the editing commands, like Cut; for example, to move a voice into another sequence, or to alter just the velocity of all the notes in a sequence.

Splitting voices across MIDI channels

The command **Functions > Note Events > Lines to Channels** assigns MIDI channel numbers (in ascending order) to the

single notes in the selected sequence. The highest note in each chord will be assigned as MIDI channel 1, the next note down in each chord channel 2, and so on. Only selected events are affected, so you should use the **Select All** command first, or a related command, such as **Select within Locators**.

The audible result will only change if you play the sequence back through an “All Cha” instrument.

Using different MIDI channels, you can assign the individual notes in polyphonic score styles, or split each note off into its own sequence, using the Arrange window function **Functions > Split/Demix > Demix by Event Channel**.

Setting Locators by Objects

The **Functions > Objects > Set Locators by Objects** command allows you to set the locators in any of the the Editors (just as in the Arrange window), so that they just encompass the currently-selected events. The key command can be found in the Key Commands window under *Global Commands*.



Edit Functions

For a description of the undo and clipboard functions (particularly adding events at the Arrange level), refer to page 1-27.

Transform Functions

The process of calling up preset (or your own) parameter sets in the Transform window is described in the section *Calling up Presets and your own Sets* on page 14-13. You can get to these directly in the Editors with the menu command **Functions > Transform . . .**

Altering Note Lengths

When changing the lengths of multiple objects, you can make all lengths equal by holding  .

Removing overlaps

When you select **Functions > Note Overlap Correction, any** overlapping notes are shortened, so as to remove the overlaps.



The affected notes must be selected first. There are three different variations of this:

Note Overlap Correction (selected/any)

this function removes overlaps for all selected notes, regardless of whether the following overlapped notes are selected or not.

Note Overlap Correction (selected/selected)

this function removes overlaps for all selected notes, but only if the following overlapped notes are selected.

Note Overlap Correction for repeated Notes

this function removes overlaps for all selected notes, but only if the following overlapped notes have the same note number (pitch).

If the overlapping notes appear to be part of a chord, you will be presented with the following options:

Keep	the simultaneously-sounding notes will be shortened together.
Delete	the simultaneously-sounding notes will be removed, leaving a monophonic line behind.
Shorten	the simultaneously-sounding notes will be shortened separately so that again, only a monophonic line remains.

Legato

With **Functions > Note Events > Note Force legato**, you can lengthen all selected notes, so that each note sustains all the way up to the start of the next one, in a “legato” manner.

There are two possible ways of doing this:

Note Force legato (selected/any)

this function forces legato for all selected notes, regardless of whether the following note is selected or not.

Note Force legato (selected/selected)

this function forces legato for all selected notes, but only if the following note is selected.

The end of the sequence is treated as a non-selected note:

(selected/any)lengthens the last note up to the end of the sequence:

(selected/selected) does not lengthen the last note.

If Logic finds a chord, a dialog box will appear saying

Chord found! Delete Chord Notes? Keep / Delete.

Delete erases any incorrect notes in a poorly played monophonic line.

Keep evens the start points and durations of all the notes in chords, which were played intentionally.

Converting Sustain Pedal Events to Note Lengths

The command **Functions > Note events > Sustain Pedals to Note Length** analyzes the sustain pedal events (controller #64) on selected notes, and lengthens the actual notes, so that they sustain until the controller #64 “off” message is reached.. The pedal events are erased by this function altogether.



A point of note when using this function in the Score window: in Page View, all selected sequences are edited. Even if the result sounds the same, the score may look different because the notes will have different durations. Removing the controller #64 messages will delete the corresponding pedal marks from the Score.

Repeating or Copying Events

To repeat selected events from one to several times (with the ability to control the grid that the first repeated event will snap to), you can use the function **Functions > Object > Repeat Objects**, as described on page 3-21.



By selecting **Functions > Copy MIDI Events**, you can move or copy all events between the locator positions, to a different position (default: song position). This function also offers several other options, which are described on page 3-77.

Deleting Events

The basic techniques are the same as for deleting sequences:

the delete key deletes all selected events,

the Eraser also deletes all selected events, and any events that you click with it (whether previously selected or not).

Deleting Similar Events

For a description of how to delete similar events (e.g. events with the same controller number), or all events except similar ones, see page 1-26.

Deleting Doubled Events

Doubled events at the same bar position may be deleted by selecting **Functions > Erase MIDI Events > Duplicates**.

Doubled events may have different second data bytes (velocity, aftertouch or controller values); Logic looks only at the event type and position, when determining whether two or more events are doubled.

Deleting Events by Reference to the Locator Positions



Selecting **Functions > Erase MIDI Events > ...** also allows you to delete all events between (... > **Inside Locators**) or outside (> ... **Outside Locators**) the locator positions.

Deleting Events Outside the Sequence

When copying events or shortening sequences, events can sometimes end up outside the limits of a sequence. These technically still belong to that same sequence, although they won't sound. You can delete these events by selecting **Functions > Erase MIDI Events > Outside Object Borders**.

Deleting Unselected Events Within the Selection

If you have selected certain events within a range, and you want to delete all the unselected events, select **Functions > Erase MIDI Events > Unselected within Selection**.

Locking Events to SMPTE Times

Refer to the section *Fixing Objects to Frames* on page 16-19.

Moving Events

You can move one or more events in the editors by using the following key commands. In the Arrange window the same commands may be used to nudge entire sequences by the corresponding grid amount:

Nudge Event Position by SMPTE frame +1
Nudge Event Position by SMPTE frame -1
Nudge Event Position by SMPTE Bits +1
Nudge Event Position by SMPTE Bits -1
Nudge Event Position by Tick +1
Nudge Event Position by Tick -1
Nudge Event Position by Division +1
Nudge Event Position by Division -1
Nudge Event Position by Beat +1
Nudge Event Position by Beat -1
Nudge Event Position by Bar +1
Nudge Event Position by Bar -1



The selected events (or sequences in the Arrange window) are shifted by one step right (+1) or left (-1), by whatever unit is referred to in the particular command name.

Step-time Recording

step-time recording allows you to enter notes with the keyboard, but not at any defined tempo, as in real-time recording. After each note is inserted, the sequencer “steps” ahead by an increment which is determined by the division setting in the Transport window. Every note or chord you enter automatically receives the note value of the display format, or a multiple of it. MIDI Step-time input is activated in the Score, Event List and Matrix Editors by pressing the MIDI In button.



Here's how to do it. Select an existing sequence, or create one in the Arrange window with the Pencil tool. Open one of the note editors (Score, Event List or Matrix).

Switch on MIDI In.

Play and hold a note or chord. You may even play the notes for the chord one after another, if you like. The important thing is that at least one note remain pressed from the beginning.

Release the last note. This moves the song position by one step (the format value on the Transport), and you may enter a new note.

Rests are entered with the sustain pedal. Each time you press this, you step through the song by the format value, without entering a new note.

You may enter longer notes or chords by pressing the sustain pedal while holding down the note(s).

You may change the format value on the Transport at any time, even while you are holding down notes.

Step-time input is also possible from the score display. However, the quantize value *Default* should be avoided, as the note display will then change every time the step size is altered.

If you are step-time recording in the Score Editor, you can also define the note value by clicking notes in the part box (but only if step-time recording is switched on.)

Keyboard Control

When employing step-time input, there are a few special, non-editable key commands that can only be used if the keys in question have not been redefined for another purpose (see the section *Key Commands Window* on page 1-45).



acts as the sustain pedal.

The space key  is the default key for Record Toggle. If you wish to use  to move along in steps, you will first have to delete this key assignment; see the section *Deleting Assignments* on page 1-48).



jumps to the next bar division position; so in 4/4 time, to the next quarter note.



jumps to the next bar.



moves back a step and erases the event there.



sets the division value to 1/4.



sets the division value to 1/8.



sets the division value to 1/16



sets the division value to 1/32.



sets the division value to 1/64.



the current division value is set to the next highest triplet value: for example from 1/3 to 1/24.



the current division value is set to the next lowest triplet value: for example from $\frac{1}{16}$ to $\frac{1}{12}$.

The key commands will continue adding these values, as long as the MIDI controller key is held down.

Goto Selection

The key command *Goto beginning/end Selection* moves the SPL to the position of the first/last selected event in the top window.



Editing via MIDI input

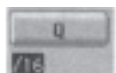
By double-clicking on the MIDI In button, you switch on “Edit by MIDI In Mode”. The values of the MIDI data being input are then used to edit the “Pitch” and “Velocity” of the currently-selected note. The note length remains unchanged. In contrast to MIDI step-time input, no new data is created; what exists already is merely changed. The key commands *Select Next / Previous Event* allow you to move one note further forward or back.



Event Quantization

The Event Editor has a separate quantization function, which can be applied to all selected events. Unlike quantization for arrange objects, the event quantization applies to all events, not just notes. It irreversibly alters their positions (only notes can be returned to their original recorded positions).

Quantising Events



Select the events that you want to quantize. Then open the pull-down Quantization menu by grabbing the “Q” button shown here.

This is identical to the matching pull-down menu for the playback parameters, and contains its own quantization grid (for details see the section *Quantization* on page 3-41). As soon as you select an item from the menu, all the selected events are quantized.

You can on/y reverse note quantization. All other events are permanently shifted.

To apply the same quantization grid to another series of selected events (even in other Editor windows), select **Functions > Quantize again**, or click quickly on the “Q” button again.



Note Quantization

Normally, all notes in a sequence are quantized according to the Quantization parameter in the sequence parameter box and the extended sequence parameter box (explained in greater detail in the section *Quantization* on page 3-41).

To quantize single notes in either of the Note Editors (the Matrix or Score Editors), you can use the Q tool, as well as event quantize.

If you click on a single note (or a selected group) with the Quantize tool, and hold down the mouse button, the quantize menu should open and you can select the quantization you want.

If you click quickly on notes, the last quantize value will be used again, just as with the Quantize Again command.

If you click-hold on the background with the *Q* tool, you get the usual rubber band for selection of several objects at once.

When working in the Score window, the display quantize setting will have an effect on how this works.

Unquantizing Notes

Note events can be returned to their original record positions, or moved manually, by selecting the setting *Qua off* (3840), or by clicking on the Q button while holding down the **Alt** key.

You can also unquantize all the currently selected notes by clicking on them with the Q tool while holding down the  key.

You achieve the same result by selecting **Functions > De-Quantize**.



Display Functions

The section *Window Functions* on page 1- 11 describes the basic window functions, including how to lay out the window elements to make more space for the event display in the graphic editors, and how to operate the zoom functions.

Many of the display options in the editor windows correspond to those in the Arrange window:

By selecting **View > Parameters**, you can hide or show the parameter fields to enlarge the window's working area. The key command is *Hide/Show Parameters*.

In the Matrix and Hyper Editors, you can conceal or reveal the transport panel in the top left corner by selecting **View > Transport**.

In all editor windows except for the Event List, you can replace the bar ruler with a SMPTE time display by selecting **View > SMPTE Time Ruler** (see page 16-20).



Changing Display Levels

Normally, the Editor windows are at the lowest display level, which shows individual events. However, clicking the small black box button, at the upper left corner of the editor window, you can move up one level in the folder/sequence hierarchy. For example, if you are currently looking at the events in a sequence inside a folder, the display will switch to a view of the sequences in the parent folder.



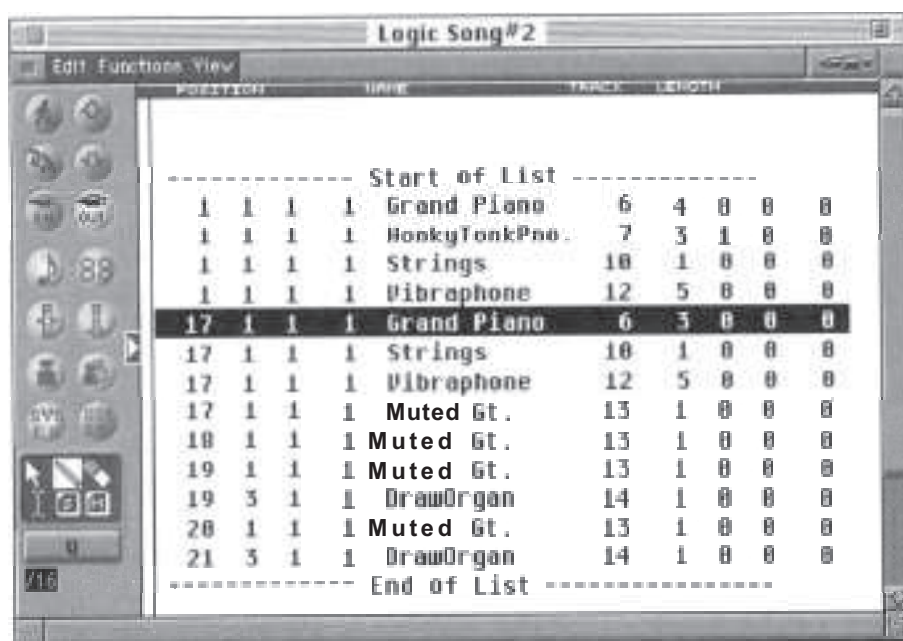
Matrix and Hyper Editors

In the Matrix and Hyper Editors, this means that you will then see an Arrange window. In this case, when you change to the lowest level, the relevant editor reappears. In this Arrange level, you will see the local menus of the Matrix or Score Editors, which also contain all the functions of the Arrange window. Double-clicking on a sequence opens the usual Hyper or Matrix display of the contents of the sequence. This means

You can quickly switch to another sequence, and edit its contents.

Event List

Clicking the small black box button in the Event List moves you up one level in the hierarchy, just like the other editors. In the Event List, however, the form of the display remains the same but instead of a list of the individual events you now see a list of the sequences along with their position, name, track number and length. The sequence that you have just left will be selected.



POSITION	NAME	TRACK	LENGTH					
----- Start of List -----								
1	1	1	1	Grand Piano	6	4	0	0
1	1	1	1	HonkyTonkPno.	7	3	1	0
1	1	1	1	Strings	10	1	0	0
1	1	1	1	Vibraphone	12	5	0	0
17	1	1	1	Grand Piano	6	3	0	0
17	1	1	1	Strings	10	1	0	0
17	1	1	1	Vibraphone	12	5	0	0
17	1	1	1	Muted Gt.	13	1	0	0
18	1	1	1	Muted Gt.	13	1	0	0
19	1	1	1	Muted Gt.	13	1	0	0
19	3	1	1	DrawOrgan	14	1	0	0
20	1	1	1	Muted Gt.	13	1	0	0
21	3	1	1	DrawOrgan	14	1	0	0
----- End of List -----								

Here too, double-clicking on a sequence (or using the “Go into Folder/Sequence” key command) takes you back to the lowest display level, showing the contents of the sequence.

Score Editor

In the Score Editor, clicking on the black box button, or double-clicking on the background takes you to the higher display level. Unlike the other editors, in the Score Editor individual events can also be edited in higher display levels. Double-clicking on a staff (at an empty point), takes you back to a lower display level.

Don't worry if all this sounds a bit confusing. In fact, the whole process of changing levels is much easier to grasp if you simply try it yourself. Read first, then experiment with the various editors in one of the Tutorial songs.

1.8 Key Commands and Remote Control via MIDI

You can activate all Logic's functions with key commands or MIDI messages. The Key Commands window is where you assign key commands to the keys, or to MIDI messages.

Whenever this manual mentions a key command, this refers to a command which can be called up by either a keystroke or a MIDI message. This allows you to completely customize Logic to suit your own working style.

If any function described in this manual is also available via a key command with the same name you will see this symbol.



Your personal key assignments are stored (together with the settings of the Preferences pages) in a file called "Logic Preferences" in your Macintosh's System folder (in the *Preferences* folder, to be precise). You should:

- make a backup of this file at another location on your hard disk;

- make a floppy disk backup of it, in case you need to use a Logic system on another computer.

When you install updates to your version of Logic, your personal key commands will remain unaltered.



Even the fixed global Key Commands of older versions are now user definable. Please keep in mind that some of these commands like Undo, Save, Quit, New, Open, Cut, Copy, Paste... follow "common conventions" and probably shouldn't be altered.



The user defined Key Commands are displayed in the corresponding menu entries



Logic Audio Platinum 4.0 has a completely new set of default key commands, which incorporates the key assignments of many experienced Logic users around the globe. A complete description follows in a separate file. The installation of version 4.0 with a pre-existing installation will of course keep your old assignments untouched.

Special Keys

Some keys have special functions:

The modifier keys can only be used in conjunction with other keys.

The backspace key () has the fixed function “delete selected objects”. It can only be assigned a function in conjunction with the modifier keys , , Alt, and .

The key combinations assigned to the options in the main menu bar cannot be reassigned. The relevant keys are displayed after the main menu items.

The  and  keys increase or decrease any selected parameter value in single units. However, they can be assigned different functions which override this function.

The keys , , , , , , plus some more keys, have invisible predefined functions for step recording, which can also be overwritten if you choose to redefine them.

Key Commands Window

The Key Commands window can be opened by selecting **Options > Settings > Key Commands**.





All the available key commands are listed on the right side,
They are grouped according to the following categories:

Global commands

Functions affecting all windows (various windows)

Functions affecting all Editor windows (various sequence
Editors)

Functions for the Arrange window

Functions for the Environment window

Functions for the Score Editor (Score window)

Functions for the Event List Editor (Event window)

A  in front of the description of the function indicates that the function is only available as a key command, or MIDI command. To the left you can see the currently-assigned key, and even further to the left, the assigned MIDI message. A dash here indicates that no assignment has been made.

Assigning a Function to a Key

Click the **Learn Key** button.

Select the desired function with the mouse.

Press the desired key, if necessary together with the desired
modifier key(s) (, , **Alt** and ).

If you want to make another assignment, repeat steps 2 and 1.
Deactivate Learn Key!



Chapter 2

Transport Functions

2.1 The Transport Window

The Transport window is used to control and display Logic's recording and playback functions.

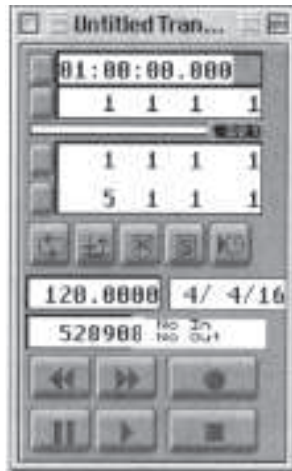


Being a float window, it is always in the foreground, and can never be covered by other windows. As an alternative to the horizontal Transport window shown above, you can open the vertical Transport window shown below. This is not a float window, and can be covered by other windows.

Opening the Transport Window

To open the normal (horizontal) Transport window, select **Open Transport** from the **Windows** main menu, or press  . To open the vertical Transport window, hold down  when selecting the same menu item.





Logic Audio Platinum allows an unlimited number of simultaneously opened Transport windows. (Additional windows can be used as giant displays for the Bar or SMPTE position.)

Closing the Transport Window

Unusually, the top bar of the horizontal Transport window is located on the left, with the close box at the top.



The Transport Bar in Other Windows

You can also configure a fixed Transport window in the Arrange, Score, and Matrix windows. When you select **View > Show Transport**, the transport bar functions appear in the top left corner. The number of visible buttons and displays depends on the size of the area available, which you can adjust by grabbing and dragging the bottom right corner as shown in the diagram.



In this chapter, the horizontal transport will be used for examples, rather than the vertical one.

Altering the Display

To move the Transport window around the screen either grab it by the left edge, anywhere in the gray area, or near the song name (right side, bottom half).

The Transport window pull-down menu opens when you click on the arrow at the bottom right (as shown here). This is where you configure the way the Transport window looks.



Smaller/Larger

You can adjust the size of the Transport bar to nine different size settings-in steps, or using **Size**.

Legend

This display option conceals/reveals a description of all the window elements, and is very helpful if you are still getting to know the program.



Position Slider

This hides/shows a bar along the bottom edge of the window, whose size represents the current amount of the song shown on screen, relative to the entire song length (read the section Song End on page 2-11). You can also grab the bar and move it, to quickly take you to a different song position.



SMPTE View Offset

This allows you to display a different SMPTE time for the start of your song, regardless of the true SMPTE start time being

read from external time code. There's more about this in the section *SMPTE Vim Offset* on page 16-4.

Giant SMPTE Display/Giant Bar Display

This converts the Transport window into a large display of the current SMPTE time, or current bar position.



You can switch back to the normal display, using the “downwards arrow menu”, or open a second Transport window.

Sections in the Transport Window

The following options let you conceal different parts of the Transport window independently. This in turn, allows you to control the size of the Transport window.

Control Switches Display (Play, Stop, Rewind...)

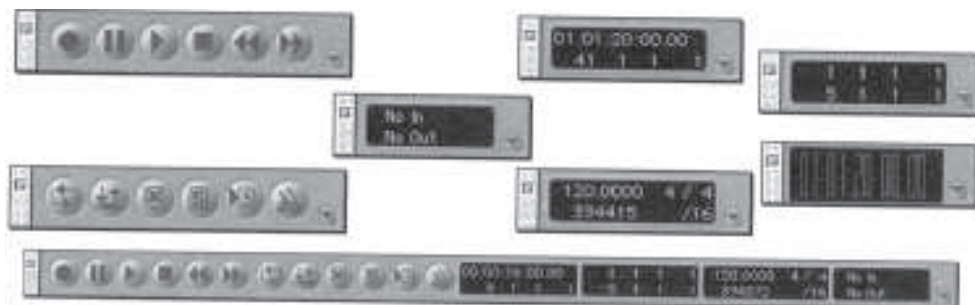
Mode Switches Display (Cycle, Autodrop...)

Position Display

Locator Display

Tempo/Signature Display

MIDI Indicator Display



If many parts are used together, the individual elements are displayed horizontally, next to one another. This allows you to configure a very narrow Transport window.

The **smaller**, **larger** commands are used as you would expect. When you switch to:

Giant SMPTE Display or Giant Bar Display

the whole window is used for the giant display of either the SMPTE time, or the bar display. This gives you freedom in sizing and shaping these displays.

All Elements Horizontal

Achieves the same effect as if you had selected all the display sections, one after the other: a very narrow Transport bar.

All Elements Normal

This restores the normal configuration of the Transport.

Parameter Fields and Displays

All the displays in the Transport panel (apart from the song name), can be used for inputting data. You can either input the numbers with the keyboard, double-clicking on the display field, or adjust the individual numbers, using the mouse as a slider.

Position Display (2 & 3)

The current song position is shown in two formats:

Above: SMPTE time (2)

Hours: Minutes: Seconds: Frames / Subframes.

For more on setting the song Start time when using external SMPTE sync, please read the section *SMPTE Offset* on page 16-4. This section also shows you how to enter a Display Offset, independent of the start time.

Below: bar position (3)

Bar-Beat-Division-Ticks.

A beat corresponds to the denominator in the time signature, a division is a freely-definable part of a beat (see section *Time*

Signature and Divisions (6) on page 28). A tick is the smallest possible bar subdivision or system quantization—just $1/3480$ note.

Using options from the **options > Settings > Display Preferences...** page, you can change the Bar Position display considerably:

Have dots as dividers

Have the tick counter operate from 0, not 1

Leave out the division value

The various options will give you the following results:

Clock Format 1 1 1 1

Clock Format 1. 1. 1. 1

Clock Format 1 1 1 0

Clock Format 1. 1. 1. 0

Clock Format 1 1 1

Clock Format 1. 1. 1

Clock Format 1 1 0

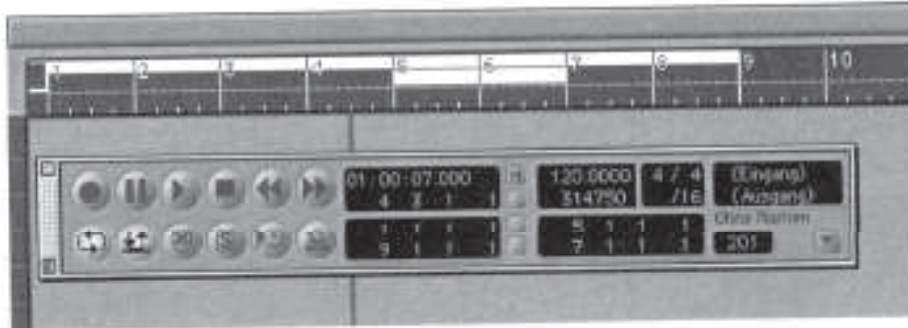
Clock Format 1. 1. 0

Buttons (area

There is a column of four small gray buttons, between the locator display area, and the tempo display area. The bottom two of these are aligned just to the right of the two locator points. The upper of these will take the SPL to the left locator, when pressed. The lower button will take the SPL to the right locator.

The topmost of these four buttons opens a Marker list window. This can prove useful, if you are just using markers as nanable means of storing locator positions. For more on this, read the section *Marker List Windows* on page 3-68.

Locators (9 & 10)



You define two sets of locator points, one for the Cycle zone, and the other for the Autodrop zone.

The locators 9 on the left define the cycle region, which is a passage that will be constantly repeated (take a look at the section *Cycle Mode* on page 2-18).

The locators 10 on the right define the autodrop range (take a look at the section *Autodrop* on page 2-14). They are only displayed when the cycle and autodrop functions are both switched on.

Any mention of the left or right locators is a reference to the left-hand cycle locators. The top one is the left locator and the bottom one is the right locator.

Switches

On the right of the cycle locators (area 4) are two placement buttons. When you click on either of the buttons, the song position jumps to the relevant locator (*Goto Left / Right Locator*).

On the other hand, clicking on them while holding down the  key places the selected locator at the current song position (*Set Left / Right Locator by rounded Song Position*).

Finally, clicking on the locators while holding down the   keys prevents any rounding to the nearest bar (*Set Left / Right Locator by Song Position*).



Key Commands for Locators

Global key Commands *Move Locators forward by Cycle Length* and *Move Locators backwards by Cycle Length* move the cycled passage by it's own length.



The command *Swap Left and Right Locator* exchanges both cycle locator points.



The key commands *Set Locators by Marker & Enable Cycle*, *Set Locators by previous Marker & Enable Cycle* and *Set Locators by next Marker and Enable Cycle* explain themselves.



Tempo (area 5)

The tempo is given in quarter notes per minute, or beats per minute (bpm). In Logic it ranges from 0.5 to 9999 bpm, and is adjustable to a precision of 4 decimal places.

Programming Tempo Changes

Please refer to the section Tempo on page 15-1.

Free Memory (area 5)

Below the tempo you are shown the free memory, in events . You can increase the amount of free memory by reconfiguring the memory. To do so, double-click on the free memory display. A dialog asking “Reconfigure Memory?” will appear. Confirm with . This reorganization is also carried out automatically after saving or loading a song. A good use of this feature is to free memory after closing a song, if two or more songs have been open at the same time. select **Functions > Trash > Empty Trash** before you reconfigure the memory.

Time Signature and Divisions (5)

This is where you can see, and set the time signature of the song. The format is:

Bar numerator/ Bar denominator/ Division

The division defines the third format value in all position displays (i.e. in the Event Window), and forms the grid for various length and placement operations. The division is normally set to 1/16 notes, but has a value range of 1/4 to 1/96 note. If the note value of the division is equal to or greater than the bar denominator, the third value of the position display is automatically removed.

The function Select next *higher / lower format* can be used to switch to the next highest or lowest division.



Programming Time Signature Changes

If you alter the bar numerator or denominator in the Transport window, a time-signature change is created at the start of the bar where the current song position is. This is shown in the Bar Ruler, to the left of the bar number. Of course, a time change does not affect the absolute positions of any events that are already there.

You can also add time signature changes directly in the Score Editor. For more about this, see the section *Time Signatures and Time Signature Changes* on page 13-5.

Editing Time Signature Changes

Move the Song Position Line to the first bar where you would like to change the time signature. Set the new time signature from the Transport bar.

You can also edit the time signature in the Score Editor, by double-clicking on it.

Erasing Time Signatures

Simply change the time signature back to the value of the previous time signature.

You can also erase time signatures in the Score Editor by selecting them, and hitting the **Backspace** key. To erase all time signatures, select **Edit > Select similar Objects** before doing this.



Copying Time Signatures between Songs

You can copy *all* time signature changes of a song to the clipboard using the **Functions > Signatures/Keys > Copy All to Clipboard** command in the Arrange Window. After changing to another song, the **Functions > Signatures/Keys > Paste All from Clipboard** command can be used to paste the copied meter changes into the other song.

This procedure also copies all key signature changes and all global score symbols (repeat signs etc.) to the other song.

Since the copied time and key signature changes are mixed with those of the destination song when they are pasted, this procedure should be mainly used to copy time and key signature changes to a song which does not yet contain any time and key signature changes, or global score symbols.

It is not possible to copy time and key signature changes within the same song,

Moving Time Signature Changes within a song

To move the time signature changes of a certain part of the song, set the *locator positions* as desired, and use the **Functions > Cut/Insert line** commands in the Arrange Window, *without* selecting any sequences or folders:

Snip: Cut Time and Move by locators will delete all meter changes between the locators (if there are any), and move the subsequent changes correspondingly to the left.

Insert Time and Move by locators will do the opposite: A section without any meter changes will be inserted between the locators; previously existing meter changes are moved to the right by the length of the cycle area, which is defined by the locator positions.

When you use this function, the following dialog will appear on the screen: Not all objects are selected. *Do you also want to move tempo changes and global score symbols?* Select “Yes” unless you want to cancel (“Abort”) the procedure.

This procedure not only moves meter changes, but also all (graphical) key signature changes and global symbols (e.g. repeat signs) in the score.

MIDI Monitor and Panic Function (7)

The top line shows the last MIDI message received, and the bottom line the last MIDI message transmitted. The monitor is mainly used to check the MIDI connections.

Click on the MIDI monitor to silence any hanging notes. If this doesn't work, double-clicking on it should do the trick (*Full Panic* take a look at the section *For Hanging Notes-Panic Function* on page 3-83).



Song Title

The title of the active song is shown below the MIDI monitor.

Song End (11)

Below the song title, on the right, you are given the position of the song's end. As soon as the sequencer reaches this position, it stops automatically, except when recording. In this case the song end is automatically moved to the end of the recording.

For internal system reasons, the maximum length of a Logic song is 8550 quarter notes, or about 2138 bars in 4/4 time.

A song can therefore last a maximum of around 70 minutes at a tempo of 120bpm. At 95bpm, the maximum length is over an hour and a half.

If you need to increase this length, e.g. for film synchronization, just halve the tempo. You can achieve the same result by using 4/8 time instead of 4/4 time, and treating quarter notes as eighth notes.

A 4/8 song at a tempo of 60bpm (equivalent to 4/4 at a tempo of 120) has a maximum length of more than 4273 bars, or over 2 hours and 22 minutes.

Keys and Buttons

All the functions of the Transport panel are also available as pre-set, but user-defineable keyboard commands, even if the Transport window is not open.

Transport (1)

The basic functions of these keys are the same as on tape machines, or cassette recorders, and should be familiar to everyone. Here are a few special features.

Record

Recording normally starts after the count-in, at the start of the current bar. In cycle mode, it starts at the left locator. You can also choose to have one beat constantly repeated, until a MIDI message is received, at which point recording begins (more information can be found on page 2-22).



Pause

Pauses recording or playback, until you press pause or play again. During paused recording, you can add individual events which will still be recorded.



Play

Starts playback at the current position, or in cycle mode from the left locator.



Play from beginning always starts at the beginning of the song, *Play from previous bar* starts at the beginning of the previous bar, *Play from Selection* starts from the first selected object, and *Play from Left Locator* and *Play from Right Locator* always start at the left or right locator, regardless of the cycle mode.

In addition to the above play commands, in any window with a horizontal time axis (Arrange, Matrix...) you can use the global command "*Play from left window orner*" to play from the beginning of the visible area.



Stop

Ends recording or playback-the sequencer stops. If the sequencer is already stopped, pressing stop moves the song position to the song start, or in cycle mode to the left locator.



The Key Command *Stop and Go To Left Locator* stops playback, and moves the SPL to the Left Locator position.



The Key Command *Stop & Goto Last Play Position* stops playback, and moves the SPL to the last position that was reached by a positioning command, or direct positioning of the SPL, using the mouse or bar ruler.



The default for this command is the space bar .

Play or Stop

This key command toggles the status of play or stop, depending on the status when the command is used.



This command is particularly helpful if you are using a Powerbook or Notebook without a separate numerical keypad.

Rewind/Forward

If the sequencer is stopped, these work as normal. If the sequencer is running, you can monitor as you go, (cueing/scrubbing), i.e. the MIDI events are output at a faster rate, (even when rewinding.). By dragging your mouse to the left or right, you can increase or decrease the speed, even reverse direction (keep the mouse button held down).



Both normal (*Rewind/Forward*) and fast (*Fast Rewind / Fast Forward*) winding are available from the keyboard.

Mode

The mode buttons do not immediately trigger an action, but switch operating states. The relevant button will illuminate to indicate that the mode is activated.

Cycle

Switches on cycle mode (more information can be found on page 2-18).



Autodrop

Switches on autodrop mode (more information can be found on page 2-25).



Replace

Switches on replace mode (more information can be found on page 2-24).



Solo and Solo Lock

In solo mode, only the selected objects are played. The data output from all other objects is muted. This is known as: "soloing the objects". You can of course, change the solo mode of objects by changing what you have selected (if necessary, refer to the section *Selection Techniques* on page 1-23). Default: .



If you want to carry out specific functions on individual objects, regardless of the soloed sequences (or folders), you have to be able to select these objects without affecting the solo status.

This is what the solo lock function is for. After soloing the desired objects, double-click on the solo button, whose colors will then be inverted (as shown). You can now alter the selection without affecting the solo status of the objects.



The key command *Reselect Solo-locked Objects* selects all currently solo-locked objects. This command is very useful if you want to add or remove some objects from your solo-locked selection:



Activate the *Reselect Solo Locked Objects* Command;

Use Shift click to add or remove objects from the selection;

Reactivate "Solo-Lock".

Sync

This button allows you to synchronize Logic from an external source. If Logic is running by itself, or acting as a synchronization source (i.e. master), this button should not be activated. When you first boot Logic, Manual Sync mode is automatically switched off. Default: .



Grabbing the sync button opens a pull-down menu, where you can define the following:

- the type of external synchronization (more information can be found on page 16-1),

- whether MMC commands should be transmitted by Logic via the Transport buttons, so that remote control of MIDI capable tape machines is possible (more information can be found on page 16-1),

- direct access to the tempo editors (more information can be found on page 15-1).

Metronome

This button (default: ) is used to turn the metronome on and off. Logic keeps a separate record of its setting for recording and playback. Grabbing the button opens a pull-down menu where



- you have direct access to the **Recording Options** of the **Song Settings** (described on page 17-2),

- you can open an Environment window with a visible and selected **MIDI Metronome Click**.

2.2 Bar Ruler

There is a bar ruler at the top of all the horizontal time-based windows (i.e. the Arrange, Matrix, Hyper, and Score).



This ruler is used to display and set the;

song position,
start and end of the song or folder,
cycle and autodrop locators, and
markers (more information can be found on page 3-62).

Bar Ruler Color

The Bar Ruler changes color depending the transport mode:

Red	Record
Yellow	Solo
Blue	External Synchronization

Display



Depending on the zoom settings (page 1-14), the bars are shown at the top edge in units of 1, 4, 8 or

16. Changes in time signature are also shown here.

In the bottom third, there is a vertical line for each bar. The shorter lines represent one beat, but are not always visible (depending on the zoom setting).

For more on converting to SMPTE times, please read the section **Bar Ruler-Time Ruler** on page 16-18.

Start and End Markers



The song start is normally at position 1.1.1.1. You can move the song start to an earlier position for playing upbeats or program change commands, by grabbing and dragging it with the mouse. The position display in the top left of the window will tell you where it is. The song end (default: bar 201) can be set using the same method, or with the numerical display (see page 2-11).

In folders, the start and end markers refer to the folder rather than the song. The length of a folder can therefore also be adjusted from within the folder itself.



Song Position Line

The song position line (SPL) is a vertical line which indicates the current song position in all horizontal, time-based windows. You can grab the line with the mouse, and drag it to the desired position (but only if there is no object at the mouse position when you drag it). Selecting *Wide Song Position Line* in the **Display** page of the **Preferences**, switches to a thicker SPL.

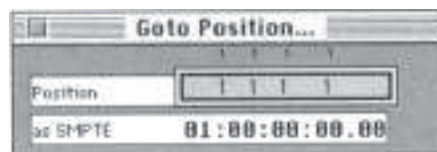
Direct Placement...

Since it is fairly difficult to grab the SPL in the window itself, you can also directly position it using the bottom third of the bar ruler. Just click here to make the SPL jump to the point you've clicked at.

...while starting or stopping

Double-clicking on the bottom third of the bar ruler repositions the SPL, and also toggles between playback (or record) and stop.

...numerically



Selecting *Goto Position* calls up the dialog box shown above, to allow you to input the song position numerically. The last division used (bar position or SMPTE time) is automatically selected, with the last input value as a default. Since the numbers are registered from the left, it is enough just to enter the bar number.



...in the score

In the Score Window you can set the SPL directly by clicking into a staff at the desired position while pressing Alt. This function is only available when the sequencer is stopped. The selected position is displayed in the Info Line while the mouse key is pressed.

This method is especially handy when working in page view mode, since there is no bar ruler in this case.

Placement at a Marker

If you have labelled a passage with a marker, clicking anywhere on the marker while holding down the  key, positions the SPL at the start of the marker. If the sequencer is stopped, double-clicking it begins playback at the start of the marker.

Scrubbing

Start playback, and grab the bottom third of the bar ruler. By moving the mouse to the left or right, you can now scrub through the song, moving the SPL to hear all the “cued” events. As soon as you stop moving the mouse, playback resumes as normal. You can also do this using the *Scrub Rewind* and *Scrub Forward* key commands.



The *Scrub by MIDI value (-2-)* command is designed to allow scrubbing via external MIDI control, but you can use any MIDI controller number. Data bytes over 64 scrub forward and those under 64 scrub backward.

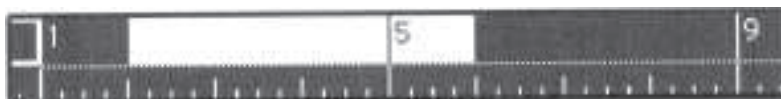
2.3 Cycle Mode

In cycle mode, a selected passage of your current song is constantly repeated. This is useful for;

composing part of a song,

practicing a recording,
recording individual tracks consecutively
editing events.

The cycle region is shown as a white stripe in the top part of the bar ruler.



Switching on Cycle Mode

There are four ways of switching cycle mode on and off:

clicking on the cycle button,
using the Cycle key command,
clicking on the top part of the bar ruler, and
inputting graphically in the bar ruler.



How Logic Behaves in Cycle Mode

The Song Position Line jumps from the end of the cycle region to the beginning;

When this happens, playback from the Environment objects which are generating notes is interrupted;

The Play command starts playback from the beginning of the cycle region;

To start playback from another position, hit Stop twice, or Stop and then Play;

At the cycle jump point, you can use the Chase Events function (more on this later) **Options > Settings > Chase Events > Chase on Cycle Jump**).

You can determine the way recording works in cycle mode, by using the various options on the **Options > Settings > Recording Options** page (for more on this, take a look at the section *Recoding in Cycle Mode* on page 2-25).

Defining the Cycle Region

Graphically Using the Bar Ruler

Click-hold in the top third of the bar ruler, and define the desired region, by dragging the mouse from left to right (dragging from right to left generates a Skip Cycle; see below).

Grab the bar in the middle to reposition it.

Grab the bottom corner to move the start or end points of the cycle (you can even do this while the sequencer is still running.). If you set the start and end points to the same position, cycle mode is switched off.

You can reset the nearest edge of the beam by clicking on it while holding down the  key, even if the beam is outside the visible range, or if cycle mode is switched off.

Drag a marker into the top part of the bar ruler (for more details, read the section Markers on page 3-62).

When you set the size of a cycle graphically in the bar ruler, your sizes are quantized to the nearest bar. The locator positions can only be changed division by division at high zoom resolutions, or if you hold down the  key as you drag. If you hold down  and , at a really high zoom resolution, you can drag and change the size of the region in ticks.

To set locator positions that do not lie on ~~whole~~ bar whole regardless of the zoom resolution enter your locator positions numerically in the Transport window.

Numerical Entry

The positions of the left and right locators (start and end points of the cycle) are shown in the Transport window, where they can also be altered.



There are also two keyboard shortcuts available (Set *Left/Right Locator*) to allow you to directly enter either of the points in the dialog box, shown above (default = last input).



By Objects

You can use the key commands *Set Locators by Object(s)* (default: **[B]**) and *Set Locators & Play* to set the locators at the start and end of selected objects; the latter command also starts playback.



Set rounded Locators by Object(s) rounds the position of the locators to the nearest bar.



The commands *Set rounded Locators & Play* and *Set rounded Locators & Record* achieves the same, while simultaneously starting playback or recording.



Set rounded Locators & Cycle Play and *Set rounded Locators & Cycle Record*, again, achieves the same, while simultaneously switching to cycle mode.



By Markers

With the key commands *Set Locators to Marker and enable Cycle* you can set the cycle to the current marker. You can also select the next or previous marker with the commands *Set Locators to next/previous Marker and enable Cycle* (in relation to the current song position).



These commands are ideally suited for a spontaneous repetition of Passages in live performances. Define markers for all parts of the song that you might want to repeat in this way during your performance.

Skip Cycle

When you are playing the song, you can skip a passage, which is useful for trying out the musical effect of various transitions.

Setting up Skip Cycle

Drag out the skip cycle region from right to left in the bar ruler.

If there is already a (normal) cycle region just move the left locator to the right of the right locator.



The skip region is shown as a very thin strip at the top edge of the bar ruler.



With the key command **Swap left and right Locators** you can change a cycle into a skip cycle.

Features of Skip Cycle Mode

When the SPL reaches the right locator, it skips to the left locator (i.e. the “right” and “left” locators swap positions).

You can also use it when editing, to leave out those parts of the song which you don’t want to be affected by the edit.

2.4 Recording

Selecting a Track

First you have to select the track you want to record on, for example by clicking on it.

You can record on several MIDI tracks simultaneously. MIDI tracks therefore have record enable buttons, too, which light red if enabled. These are enabled automatically by selecting a track. When you want to record additional tracks, click their record enable buttons directly. In any case, only one track can



be selected at a time. You can read more about this feature in the section *Multi Track MIDI Recording* on page 3-10.

During the recording, the incoming events are stored in a sequence (with audio: in a region), on the selected track.

Audio tracks have to be record enabled, and a recording path (a hard disk drive) has to be defined for the audio files to be recorded. Please refer to everything that is audio-specific in the section *Audio Recording* on page 4-13.

Changing MIDI Tracks

You can change the record track, without having to stop recording -just select a new track, for example using the *Select previous/ next track* command (default keys: ).



Count-in

After you press the record button, the recording begins with a count-in. This is defined under **Options > Settings > Recording Options**. The choices are:



- No count-in** The recording begins without a count-in.
- Wait for note** Logic repeats the first 1/4 note (or note value of the bar denominator), until a MIDI note is received. The recording then begins.
- 1-6 Bar count-in** You may select a 1 to 6 bar count-in (default: 1 bar).
- 2/4-9/4 count in** There is a count-in of 2 to 9 quarter-notes. (These options are particularly handy for count-ins during time signature changes).

...Click only during count-in

If the **Options > Settings > Recording Options > Click only during Count In** option is active, the click will be switched off after the count-in-so-called “Drummer mode”.

This is often useful if the section of ~~the~~ song just before the part you're recording lacks the sort of rhythmic information necessary to play new parts in time, but there's plenty of rhythmic reference, once the section you're recording to gets going.

Record Options

Record Toggle switches between playback and record mode.
Record Repeat repeats the recording from the previous drop-in point.



Record into selected sequences

Normally a new sequence is created during every recording. In the **Recording Options** of the **Song Settings** you can activate *Merge New Recording With selected Sequences* so that any new data is incorporated *into* an existing sequence, when this sequence is selected.



Retrospective Recording

If you played a great solo while playback was running, but forgot to record—don't worry. Simply use the key command *Capture Last Take as Recording*. All events you played after the last change of the Song Position Line will belong to the recorded sequence—just as if you had recorded right from the start.



All events played live after the last time the song position was changed are included in the recording. If you move the song position, the next-received event deletes the “Recording Play Mode Buffer”.

Replace Mode

To activate replace mode, press the replace button. In replace mode, any newly recorded data is always stored in a new sequence. In addition, any existing sequences on the destination track are cut at the punch in/out points of the recording, and any data between these points is deleted.



Destructive MIDI Recording

If you select *Merge New Recording With Selected Sequences (r)* and switch on replace mode (the recording head symbol in the Transport window), the new events you record will replace the ones in existing sequences.

The Merge/Replace combination can itself be coupled with the Autodrop, and/or Cycle functions.

Recording in Cycle Mode

All settings for recording in cycle mode can be made in the **Recording Options** of the **Song Settings** (see page 17-2. You can either use several cycles to record a single sequence (*Merge only New Sequences in Cycle Record* checked). or you can create a new sequence for every cycle (...unchecked). A new track can be automatically created for each of these sequences (*Auto Create Tracks in Cycle Record*). The sequences you create can also be automatically muted (*Auto Mute in Cycle Record*). This mode is very well suited to recording several consecutive versions of a solo, and then picking the best one.

Cycle and Replace

During a cycle recording in replace mode, existing sequences are deleted during the first cycle, from the punch-in point to either a punch-out point or up to the end of the cycle. When the second cycle begins, recording continues, but no more sequences are deleted. If you want to replace the end of an existing sequence, you don't have to stop recording before the second cycle begins: the start of the existing sequence remains intact.

Autodrop

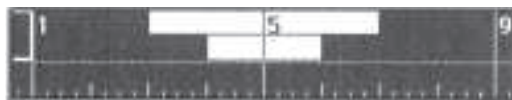
Autodrop means automatically going into and out of record at previously defined positions. Autodrop mode is most commonly used to re-record a badly played section of an other-

wise well played recording. The advantage is; you can concentrate on the playing.

If cycle is not active, the left and right locators serve as drop-in, and drop-out points. Autodrop is activated by clicking the auto drop button.



Defining the Autodrop Region



If both cycle and autodrop are active, there is an independent pair of locators

available for the autodrop. In this case, there will be two stripes in the bar ruler, the top one representing the cycle region, and the bottom one the autodrop range. The autodrop stripe can be graphically altered in exactly the same way as the cycle stripe. If the bar ruler display is very narrow, holding down the  key as you alter it will ensure that all your actions apply to the autodrop stripe. To activate autodrop graphically in the bar ruler, drag down the stripe while holding down the **Alt** key.

For graphic operations, the grid scale can be reduced to division resolution by holding down , and to tick resolution by holding down the Shift + Control keys.

Setting Autodrop Numerically

The positions of the autodrop locators are displayed numerically, to the right of the cycle locators on the Transport window. Their position can be altered by changing the numbers from here.

Recording in Autodrop Mode

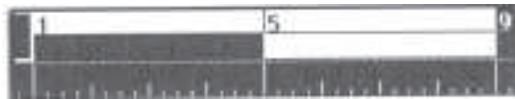
To carry out an autodrop recording, put the sequencer into record anywhere before the drop-in point. Any events which you play before the drop-in or after the drop-out are channelled through the sequencer as usual, but are not recorded.

If the song position line is behind the right locator when you start recording, recording automatically begins at the drop-in point, after the count-in.

If autodrop is enabled during recording, the record button will flash as long as the song position is outside the drop locators. This indicates that a recording takes place.



Combining Cycle and Autodrop



If you want to improve a difficult part of a certain passage, you can use a combination of cycle and autodrop. Cycle mode lets you practice as many times as you like before the “final take”. Each time, only the autodrop range is recorded. You can use the previous section to “get into the groove”.

Recording with Skip Cycle

If skip cycle is switched on the cycle region is left out during the recording.

2.5 Chase Events

Chase Events is a function which searches all the sequences playing at a jump-in point. The function examines what all these sequences are doing *before* the jump-in point, to find out which events would be affecting playback at the jump-in point, if it had been reached by playing through the song, rather than by just jumping there.

This is a complex point to grasp, so here's another explanation:

If you start playback of a song in the middle, by jumping straight to that point (via Cycle mode looping or by direct placement), you might expect there to be problems with note play-

back. If an important note started playing just before the play-start point, you would expect Logic to miss it, and the note would not sound. Notes, however, are not the only potential problem. If there were a maximum pitch-bend message just before the play-start point, playback would leave this out too. Fortunately, Logic is smarter than your average MIDI sequencer, to misquote Yogi Bear...

Open the **Chase Events** page in the **Song Settings**, to set up the chase events function.



The function searches all the sequences playing at the play-start point, looking before the play-start for a selection of the following:

- any notes still due to be playing at the play-start point;
- any notes still due to be playing at play-start, because of a held-down sustain pedal (*Chase sustained Notes*);
- program changes;
- pitch-bend information;
- continuous controllers 0-15;
- continuous “switch” controllers 64-71;
- all other controllers (*all other Controls*);
- monophonic (channel) aftertouch;
- polyphonic aftertouch;
- SysEx data (the last SysEx message before the play-start point is transmitted)

Special Note:

There is a potential problem with chasing notes that are being used to trigger a drum loop in a sampler. Unless you are lucky enough to start the sequence precisely at the beginning of the sample loop, the sample will be triggered at the wrong time and therefore, they will be played out of sync with the rest of the sequence, (at least until the next trigger note). The problem is that most samplers can only play their samples from the beginning, and cannot synchronize them to the beat by starting in the middle.

Well, here's how you solve that one:

Chase Events

Activate the No Seq Trp parameter of your drum loop instrument, in the instrument parameter box, and switch off Chase *Notes in No Seq Trp Instruments* in the **Chase Events** page of the **Song Settings**. The result is that whenever the song jumps to a new position, your drum loops will not play until they reach the next “trigger note”.

The No Seq Trp parameter actually prevents transposition by the sequence playback parameters, which is also not desirable for drum sounds or loops.

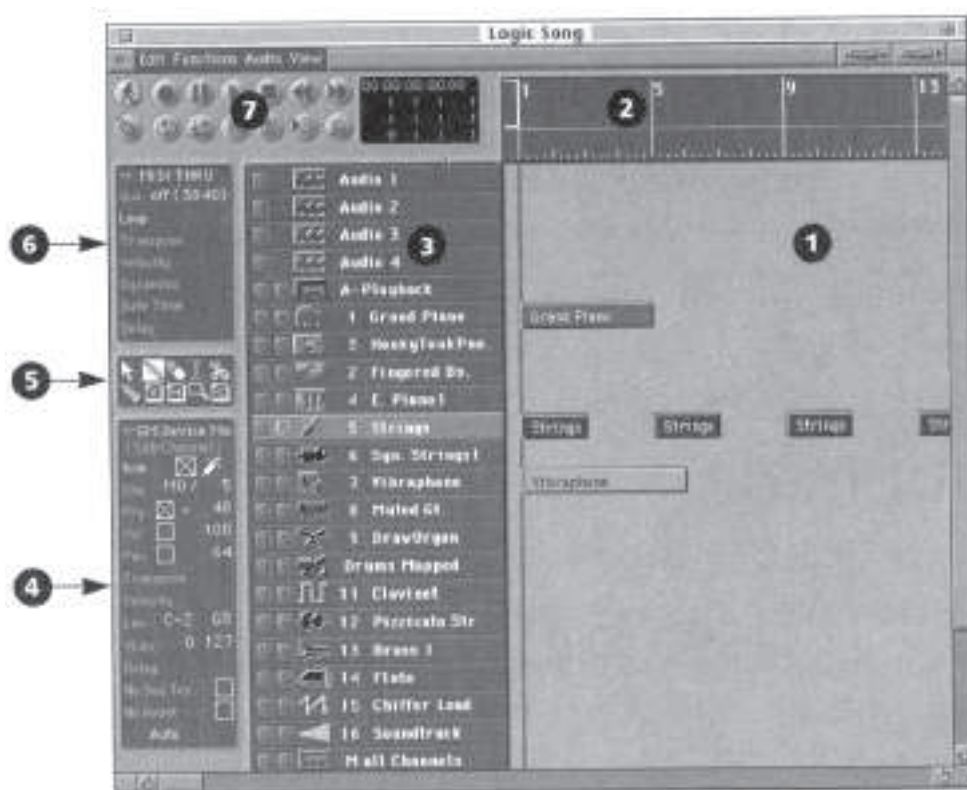
Chapter 2

Transport Functions

Arrange Window

3.1 Overview

The Arrange window is the heart of Logic. Study it closely, because it is the view of the program that you will see most often when you're working with Logic.



It appears in the first screenset whenever you start Logic.

The arrange area **1** is where all MIDI information is recorded, on horizontal “tracks”. Individual MIDI recordings are called “sequences”, audio recordings are called “regions” and are displayed as beams as well. Above the arrange area is the bar ruler, which displays position information **2**.

To the left of the arrange area is the track list **3**. This is where you determine which “instrument” should play the MIDI or

audio information on each track. You can make various settings for the instruments in the instrument parameter box ❹ in the lower left corner.

Because the sequences are arranged graphically, you can also use specialized mouse tools from the tool box ❺ to help you perform different operations.

The sequence parameter box ❻ is where you set the playback parameters for the individual sequences, such as transposition and “quantization”.

At the upper left corner is the transport panel ❼ which is functionally almost identical to the Transport window.

Opening the Arrange window

The Arrange Window can be opened by selecting **Windows > Open Arrange** in the main menu, or by using the keys  **A**.

In addition to the this command, you can define any other key command for *Open Arrange Window*.



3.2 MIDI Tracks

This chapter merely deals with MIDI Tracks, although the use of audio tracks is the same wherever possible. Each individual track is stretched horizontally across the arrange area, and the tracks are stacked vertically. A track is where the notes and other MIDI events are recorded.



In the track list each track is numbered from top to bottom. During playback a small “level indicator” appears over the track number. This indicates the veloc-

ity of the recorded notes, and becomes red for a maximum velocity value. The small “c” stands for Controller command.

To the right of this is where you assign an instrument to the track, which is represented by an icon and a name.

The instrument determines which sound generator or audio instrument plays the MIDI events or audio data on the track. For more information, refer to the section Instruments on page 3-12.

Strictly speaking, this doesn't have to be an instrument, because you can assign any environment object to a track. The track data could therefore be sent to a fader object, or directly to a MIDI port. However, since you will normally assign instruments in the track list, we will use the term “instrument” instead of “track data destination object”.

A MIDI track will be automatically record enabled automatically when you select it. The record enable switch glows red to indicate this. During recording, a sequence is created on this track containing the recorded MIDI events or audio data.



Selecting a Track

You select a track by clicking its name or icon in the track list. This also selects all objects on the track (if the Cycle function is switched on, it only selects the objects within the defined cycle region).

Use the Select *Next/Previous Track* function to select the track above/below it in the track list (key command:  / ).



Changing tracks while keeping selection

If you click on a new track while holding down the **Shift** key, the track will be selected without changing the existing selection of sequences and folders.

Sorting Tracks



To change the position of a track in the track list, grab it near the track number, and drag it up or down. The cursor will

become a hand, to indicate that the track can be repositioned vertically.

Selecting Objects of the same color

If you have selected a sequence of a certain color, you can use the **Edit > Select equal colored objects** command to select all other objects of that same color. One use for this would be to compile mute and solo groups based on color.

Creating Tracks

To create a track, use the **Functions > Track > Create** command. The new track is created at the same position as the track currently selected, and all the tracks below it are moved down accordingly.

To create a track at the bottom of the track list, double-click in the track list below the lowest track (*Append Track to Track List*).



Creating a Track with the Next Instrument

The **Functions > Track > Create with next Instrument** function, creates a new track under the selected track, and assigns it the next instrument from the instrument selection list. Under most circumstances, this would be the next MIDI channel in the same sound module.



You can also hold down the **Alt** key when you create a new track by double-clicking under the track list. Or, you can use the "*Append Track with next Instrument to Track List*" command.

Copying a Track

If you hold down  when you sort a track (vertically move it), this creates a copy of the track with the same instrument at the destination position.

All selected sequences on the old track are also moved to the new track. This is an easy way of carrying out a few useful functions:

If you hold down  before grabbing the track number, no extra sequences on the track will be selected.

If no sequences were selected in the first place, an empty track with the same track instrument is created at the destination position.

Otherwise, all selected sequences are moved to the new track.

If you grab the track first and then *Alt*-press sequences on the track will be selected-as is normal when you grab a track:

If the cycle function is switched off, all sequences are selected and moved to the new track.

Otherwise, all sequences in the cycle range are moved from the old track to the new track. The musical result is not any different, because the new track is played by the same track instrument.

Moving Objects onto a Track

The function **Functions > Object > Move Selected Objects to Track** moves all selected sequences from different tracks onto the selected track. The time position of all the sequences is retained.



This is useful for displaying sequences in a score stave, or gathering together regions of different vocal takes.

Special Methods for Creating Tracks . . .

...for overlapping objects

The function **Functions > Track > Create for Overlapped Objects** creates new tracks for wholly or partly overlapping Objects. The objects are distributed onto these tracks so that no more overlaps occur.



...for selected objects

Use the function **Functions > Track > Create for Selected Objects** to create new tracks for all selected objects. The objects Usen appear on these new tracks. Only one new track is

created for any selected objects which were previously on the same track.

Deleting Tracks

To delete the selected track, use the **Functions > Track > Delete** function. If there are any objects on the track, a warning message appears first.



If there is no object selected in the arrange area, you can perform the same function by pressing the ***Backspace*** key.

A third method is to grab the track as if to reposition it (see above), and remove it from the track list by dragging it to the left.

Deleting Unused Tracks

Use the **Functions > Track > Delete Unused** function to delete any tracks that don't contain any sequencers or folders.



Naming Tracks

To name a track independently of its assigned instrument, select **Functions > Track > Create Trackname**.



You can also change the name of an existing track by double-clicking on the instrument name in the track list, while holding down the  key, and typing in the text input field that appears.



The track name is then displayed in place of the instrument name in the track list, and will be used as the default name for recorded sequences.

If there is enough room vertically (achieved by zooming in vertically), both the track and instrument name will be displayed.

MIDI Tracks

You can see the instrument name in the instrument parameter box, or check it by clicking on the track name and holding down the mouse button.

Name while creating a track

When you create a track (by double-clicking below the bottom track in the track list) you can hold down  at the same time to open the input box for the track name.

Deleting Track Names

To delete a track name, select **Functions > Track > Delete Trackname**.



You can also double-click on the track name while holding down . This opens the text input field for the track name. The name can be deleted using **Backspace**. The instrument name will then appear in the track list again.

Muting Tracks



Both MIDI and audio tracks have mute buttons to the left side of the track list,

between the track number and the icon. Muting a track stops it from playing. To show or hide the blue mute buttons, select **View > Mute Switch**.

Muting when the mute buttons are hidden



If the mute buttons are hidden-to save space-you can still mute a track to the left of the track number.

If you move the mouse to the left edge of the track list by the track number, the mouse pointer turns into the mute tool ("M" for "Mute"), clicking the mouse now mutes the track (or if the track is already muted, it cancels the mute).

You can also use the key command *Mute Track* to mute a track.



Muted tracks are indicated by a •

Muting All Tracks

If you hold down  while clicking in the left side of the track column, all tracks in the currently-selected display level (or folder) are muted together. If they were already muted, they will be unmuted.

There is a key command *Mute All Tracks of Folder*.

Mute Instrument

If you mute a track while holding down  , all the tracks in the current song with the same track instrument (including those in all the folders) are muted.

There is a key command *Mute All Tracks with Same Instrument*.



Selecting an Instrument

By clicking on an instrument's name (or icon) in the track list, and holding down the mouse button, you open a flip menu. This is where you select and assign instruments to a track.

Changing an Instrument assignment globally

If you hold down the **Command** key while selecting an instrument, the previously selected instrument will be replaced in every track in the current song by the new instrument (even in tracks in folders).

Selecting track instruments by “Drag & drop”

You can select any of the objects in the Environment window to be the current track instrument, by simply dragging one out of the Environment window, and dropping it onto the track list.

If you move a Multi-instrument into the track list, the selected sub-channel is set as the track instrument. If no sub-channel is selected, all initialized (i.e. those not crossed out) sub-channels are set for the destination track, and the tracks below that. If there are no tracks below the destination track, new tracks will be created.

In all cases, the Environment objects involved are completely unaltered.

Special Types of Instrument

In addition to the normal instruments which are described in the section Instruments on page 3-12. There are two other possible track settings for which there is no corresponding Environment object.

No Output

Tracks assigned as “No Output” send no data. This can be useful in situations where you wish to store data (such as Sysex) that you don’t wish to send.

Folder

This setting is used when you want the track to play a folder (more information can be found on page 3-56). Normal sequences will not play on a track set to the Folder instrument.

Zooming Individual Tracks

Tracks in the Arrange window can be zoomed individually. Also, all MIDI or all Audio tracks can be zoomed as a group. In this way you can, for instance, zoom on an individual track to edit it’s sequences and/or regions with HyperDraw, or zoom on all audio regions to get a more detailed view of the waveform display.

Zooming on a Track

Move the mouse in the track list to the bottom left corner of a track. The mouse pointer will change to an index finger. Now you can zoom in or out on this track-the zoom factor, compared to the normal settings in the Arrange window, will be shown in the Info Line while zooming.

Zooming all MIDI/Audio Tracks

When zooming all MIDI or all audio tracks, keep the -key pressed.

This changes all tracks of the same type.

Restoring Individual Zoom Settings

When zooming on a track, hold the -key down.

Key Commands for Zoom Settings

Using the Key Command “Individual Track Zoom In/Out”, you can adjust the display of the selected track.



Using the Key Command “Individual Track Zoom Reset”, the selected track will be reset to the global (default) zoom factor.



Using the Keyboard Command “Individual Track Zoom Reset for All Tracks” all tracks will be reset to the global (default) zoom factor.



3.3 Multi Track MIDI Recording

You can record multiple MIDI tracks simultaneously. For this, MIDI tracks have a Record Enable button ("R") to activate the recording function. This is automatically activated by selecting a MIDI track. To record additional tracks, just click on their record-enable buttons.



In stop-or pure “MIDI Thru”-mode, you can layer the sounds of your instruments easily-or create split or velocity switched sounds, by setting the appropriate instrument parameters.

This feature is new with Logic Audio Platinum version 4.0

As before, only one track can be selected, because the instrument parameters of only one track can be displayed. You can change the selected track easily from

among the record-enabled tracks. Selecting a track which has not been record enabled will cause the record function for all other tracks to be turned off.

Layer and Split Mode

For simultaneous recording of multiple MIDI tracks, there are two modes of operation; Layer or Split.

The Layer Mode is used to play (and record) multiple tracks, without having to configure them beforehand in the Environment. In this way, layered sounds using different instruments can be quickly created.

The Split Mode is used to record multiple MIDI sources to different tracks simultaneously. In this way, a number of different MIDI instrumentalists, or different instruments, can be recorded onto separate tracks.

Selecting Layer/Split

To switch from the Layer mode (default) to the Split mode, select the option: **Song Settings > Recording Options > Auto Demix by Channel if Multitrack Recording.**

Layer Recording

Incoming MIDI events are sent to all MIDI tracks that have been Record enabled. You can listen to all the selected instruments before and during the recording. After recording, only the selected track will display a sequence. On the other tracks, aliases from the sequence on the selected track will be recorded.

The advantage of this is that any edits applied subsequently to the sequence will affect all the aliases, and all Layer tracks will remain identical.

Split Recordings

Incoming MIDI events will be distributed to the various Record enabled tracks, according to their transmitted MIDI channel numbers. For this, the channel of the incoming event

must correspond to a Record-enabled track's channel. If no track is found with the corresponding channel number, the event will be routed to, and recorded onto the selected track.

Be sure that the different MIDI controllers transmit on different channels. If this is not possible, simply use different inputs on your MIDI interface, and change the MIDI channel of one input, using a transformer between the physical input and sequencer input.

Switching MIDI Thru off

The record button makes very easy to switch off the MIDI Thru function: simply deactivate the Record button of the selected track.



Keep in mind that you can go to Options > Settings > MIDI Options... to select an instrument for which the MIDI-Thru function is permanently disabled.

3.4 Instruments

In order to know where specific MIDI messages are to be sent, Logic needs to have some idea of how the MIDI devices in your setup are connected. This is portrayed graphically in the Environment window, which shows the sequencer together with many other objects, which we will refer to as “instruments”. Each instrument is therefore an Environment object, representing a physical MIDI device. You can think of the instrument object as being a virtual instrument, or a representation of a real instrument.

To find out how to give Logic all the necessary information to configure the environment, refer to the Chapter 5: *The Environment*.

Making an Instrument Visible

If you double-click on the instrument name in the track list, this directly opens the Environment window, with that object selected.

Creating a New Instrument



It is best to create instruments directly in the Environment, but you can also do so from the Arrange window, by using the function **Functions > Track > Create New Instrument**. The selected track will then contain a new instrument, whose parameters can be adjusted in the instrument parameter box. The new instrument will also be present in the Environment window. All instruments in the Arrange window exist in the Environment, no matter where they were created originally.

The Instrument Parameters



The instrument parameter box is located at the bottom left corner of the Arrange window.

The instrument parameters belong to the instrument, not to the track per se, so if you alter the parameters here, it will affect all tracks playing that instrument.

The instrument parameter box in the Arrange window is identical to the corresponding parameter box for the same instrument in the Environment.

Opening/Closing Instrument Parameter Box

Click the small triangle at the top left to open or close the instrument parameter box.

Name

The top line shows the instrument name, which can be edited by clicking on it.

Track Object Type

The track object type is given in brackets, and cannot be altered. Normally this will be: (Instrument) or, if you're using Multi-instruments: (*Sub Channel*).

Selecting an Icon

Grabbing the icon opens a flip menu, where you can assign an icon to represent the instrument. If you hold down the  key while you do this, the flip menu will remain open when you let go of the mouse button.

If you have closed the instrument parameter box, or it is hidden from view, you can select an icon directly from the track column by clicking on the Icon to be changed while holding down the   keys,

Color of the Instrument Icon

The color of the selected track's instrument icon can be altered, using the usual color palette with: **View > Object Colors...**, as long as no Arrange object (sequence) is selected.



The color of the instrument icon serves as the default color for all sequences recorded on this track.

Hiding an Instrument



The small box to the left of the icon determines whether the instrument appears in the instrument selection flip menu of the track list. You will generally always have this checked for instrument objects.

This option is primarily used to shorten the instrument flip menu, by hiding other Environment objects, such as faders or MIDI ports.

MIDI Channel, Port (MIDI Output) and Connection

The parameter “Port” defines the MIDI Output to which the data for the instrument is sent. Your real sound module is connected to this output jack. The MIDI channel is set with the *Cha* parameter. This defines the channel on which the instrument outputs data, so that your “real” instrument can receive the data.

Which connection, in other words, which computer interface connector (modem-, printer- or USB) your MIDI Interface is connected to has to be set under **Options > Settings > MIDI Interface communication**. If you touch the left side of the value beneath the Port parameter, you can select the connection (the computer interface). If you grab it to the right side, you can select the 3 x 64 MIDI Outputs directly, each of which has 16 MIDI channels. You can read more about this in the section *Connection (Modem Port, Printer Port, USB)* on page 17-12.

The purpose of a hidden direct connection is to enable you to address the individual MIDI ports directly from the Arrange window without having to switch to the Environment. For most situations, this is the best way to assign instruments to ports, as it eliminates the need for cabling.

Keep in mind that if you directly assign a port, and also connect the instrument to a MIDI out object with a cable, all MIDI data sent via that instrument will be doubled. If you attempt to do this, Logic displays a dialog box asking whether to Keep or Remove the direct connection. You will generally select Remove in this case.

Setting the MIDI Channel: Multi Instruments

In the instrument parameter box of the Arrange window, you can also alter the MIDI channel of a part instrument ("sub-channel" or "part" of a multi instrument). This does not reassign the channel for the current part, but rather selects another existing sub-channel from the current Multi Instrument. (In versions prior to 3.0 an alert message displayed: "Channel is protected".)

You can't actually change the receive channel of a part in the multitimbral synthesizer. This is only possible in a very few models (and anyway is not particularly useful when using a multitrack sequencer). The "Cha" parameter is really an alternative way of changing the track instrument, so that the track can be played by a different part in the multitimbral synthesizer.

If the MIDI channel is set to "All", you can edit the parameters of the whole multi instrument, e.g. the MIDI port.

Adjusting the Sound of a Track



The *Prg*, *Vol* and *Pan* parameters transmit program changes, volume controllers (#7) and pan controllers (#10) respectively.

If there is no check in the respective box, the default value on the MIDI device itself is used. The corresponding value is only transmitted if you place a check in the box, by clicking it. If there is already a check in the box, any value alterations you make are transmitted immediately.

A program change may be selected on the right, with a flip menu; volume and panning are set using the mouse as a slider. If the Autolink function to SoundSurfer or SoundDiver is active, or when dealing with multi-instrument sub-channels, sounds can be selected from a flip menu by name. If you hold down the  key, any value can be increased or decreased by clicking on it above or below the current value.

To the left of the program number, there is an extra parameter which is used for Bank Select. If your sound source can receive Bank Select messages (MIDI controller #0 or #32-check your synth manual for format details), you can switch between different banks, each containing a maximum of 128 sounds.



As many devices cannot receive these messages, you can deactivate this parameter by making the setting shown here.

For more on the other instrument parameters, take a look at the descriptions in the section *Standard Instrument*.

Recording Program Changes, Volume or Pan Controllers

Any of the types of events that can be transmitted by checking the selections in the instrument parameter box can also be stored when in record mode. For example, in record-pause mode, you can store program changes at specific positions in the following way:

Remove the "x" in the box next to Prg (or Vol, or Pan);

Click on Pause, then Record;
Move the Song Position line to the place you want;
Select the sound (program) you want (or the volume/pan setting you want);
Click on the box next to Prg (or Vol, or Pan). Each selected event will be sent and recorded.
Click on Stop to exit Record mode.

In this context, please remember Logic's sophisticated Mixer Automation features as described in the mixer chapter.

The values of the Parameters *Prg*, *Vol* and *Pan* in the instrument parameter box will not be updated every time such control change events are played back. These parameters are initial values, when you load a song, or use the command **Options > Send To MIDI > Used Instruments MIDI Settings** - before you begin to record mixer automation data.

Open a Sound Editor for a Given Instrument

If you have installed SoundDiver and Autolink is activated you can select **Functions > Track > Open SoundDiver for Instrument** to open the sound editor for the given instrument.



3.5 Sequences



Sequences are containers for the MIDI events within them. Their purpose is to make things clearer and easier to deal with. They also correspond to the musical convention of treating a phrase or a riff as a single unit. It is often better to apply many operations (e.g. quantization) to these units, rather than to individual notes.

Sequences within a track can overlap one another, either partly or completely, but for clarity's sake this should generally be avoided.

Remember that all the following operations with a plural in the heading (sequences), apply just to selected sequences.

Creating a Sequence

Normally a sequence is created automatically when you record on the selected track. It begins at the start of the bar in which the first event was recorded, and stops at the end of the bar in which the last event was recorded.

Sequences can also be created by directly inserting events from the clipboard into the Arrange window (see the section *Inserting Events* on page 3-30).

Capture Recording

You can also create a sequence using the key command *Capture Last Take as Recording*. Please refer to the section *Retrospective Recording* on page 2-24.



Creating an Empty Sequence

Click with the Pencil at the position in the Arrange area where you want the empty sequence to begin. You can now enter the events manually in one of the Editors.

Deleting Sequences

You can delete all the selected sequences by clicking them with the eraser, or by pressing the *Backspace* key. You can also delete any non-selected sequence by clicking it with the eraser.

Deleting Muted Sequences

The function **Edit > Select Muted Objects**, as the name suggests, selects all muted sequences, folders, and-in Logic Audio- Regions. Pressing the  key will then delete these objects.

Retrieving Deleted Sequences

Of course if you accidentally delete a sequence you can restore it by selecting **Edit > Undo** ( ) immediately afterwards.

Even after several editing operations, it is still not too late to retrieve a sequence, because all deleted sequences actually end up in the trash. The function **Functions > Trash > Open Trash** opens the otherwise invisible trash folder. You can then transfer the desired sequences to the clipboard (Copy or Cut), and add them back to the song (Paste). An even neater method is simply to drag the sequences from the trash window into the Arrange window!

The function **Empty Trash** irreversibly deletes the contents of the trash. If there are a lot of objects in it the song will take up less space when it is saved if you choose this option.



The option ***Empty Trash after Saving a Song*** on the Global page of the Preferences (when checked), deletes the contents of the trash after a song is saved. This means the next time you load the song, you still have access to the deleted sequences from the previous session.

Moving Sequences

Sequences may be moved by grabbing them and dragging them with the mouse pointer. You can move sequences both along the timeline and from one track to another. You can also move them between two Arrange windows, even if they belong to different songs.

...Between different Songs

If you move a sequence from one song into the Arrange window of another, the sequence is automatically copied over, in much the same way the MacOS copies files automatically with drag and drop, if they are located on different drives.

...Limiting Direction

You can limit the movement or copying of objects in the Arrange window to either the horizontal or vertical axis- just as in the Score and Matrix editors. To do this, just activate the option **Options > Settings > Global Preferences > Limit dragging to one direction in Arrange**.

While you're moving objects, you can toggle the direction limitation on and off, by pressing the  key.

Tying Sequences or Regions by Length Change

With **Functions > Object > Tie objects by length change**, the selected objects become longer, so that they end exactly at the beginning point of the next object on the track.



Removing Gaps between Sequences or Regions

By selecting **Functions > Object > Tie Objects by Position Change**, you can remove any empty space between selected sequences. When you do this, the first sequence remains unchanged, and all subsequent sequences are moved to the left. This is particularly handy for seamlessly joining audio regions together in Logic.



With audio regions, this feature can be used to speed up spoken word recordings that have been processed with strip silence.

Tie Sequences within Locators

Functions > Object > Tie sequences within Locators makes all selected sequences within the Locators longer, so there are no gaps between them. The last object on each track within the selection won't be affected.



Moving Sequences to the Current Song Position

The key command *Pickup Clock Position* allows you to move the selected sequence (or the first of several selected sequences) to the current song position.



Editing the Start Position Numerically

You can edit the start point of a sequence (and its length) numerically. This can be done from an Event list window set to view objects at the same level as the Arrange window (see the section *List Structure on the Arrange Level* on page 10-10) or in a floating event window (see the section *Event Float Window* on page 10-18).

Copying Sequences

To copy sequences to another point in the song, or to other tracks, hold down the  key while moving the sequence.

Making Multiple Copies of Sequences



The **Functions** **Object > Repeat objects** menu allows you to repeat one or more selected sequences. The repetition always starts at the end of the sequence itself, or at the end of the last (along the time line) sequence.



Number of Copies

This is where you enter the number of copies (excluding the original).

Adjustment

This is where you determine whether you want a copy to begin exactly at the end of the original (or the previous copy) (setting: *None*), or whether you want the start-point to be “quantized”. In most cases the *Auto* setting will do what you want.

AS

This parameter determines whether the repeats are *copies* or *aliases* of the original (more information can be found on page 3-59).

*There is also a sequence parameter called "Loop", which repeats a sequence over and over up to the next object on the same track (see the section *Loop* on page 3-33). This may be preferable to copying sequences in many cases.*

There's an even more extensive copy function that is strictly for MIDI events, called "Copy MIDI Events". For more on this, check out the section *Copying Events Selectively* on page 3-77.

Altering the Length of Sequences . . .

Grab the sequence at the bottom right corner, with the pointer or pencil. You can now move the end of the sequence wherever you want. Even when you shorten sequences, the data in them is never deleted; playback just stops at the end of the sequence. You can get the events at the end of the sequence to play again, by simply lengthening the sequence.

The extended sequence parameter *Clip Length* determines whether the notes sounding when the sequence ends should be abruptly cut off (on), or whether they should be played to their normal end point regardless of where the sequence ends. Also see the section *Extended Sequence Parameter Box* on page 3-39).

...of Multiple Selection...

You can change the length of multiple objects in the same way. The length will be changed by the same absolute amount.

...to the same absolute length

If you want to make all selected sequences the same absolute length-even if they had different original lengths-simply hold   while changing the length (just as in the Matrix Editor or the Event List).

...Adjusting the Start

You can also adjust the length of the sequence by grabbing it at the bottom left corner. However, you can never move the left corner beyond the first event in the sequence, i.e. you can never “hide” events from the front. If you want to remove the start of a sequence you have to cut it, and mute or delete the start.

... Rounded to Whole Bars

The function **Functions > Object > Snap Objects** rounds the start to the nearest whole bar.



When you lengthen or shorten sequences, this never alters the position of the events contained within it. If you wish to do so, please do the following:

... by Time-Compressing or -Expanding

Hold down  while you are altering the length. The timing of the events in the sequence will be stretched or compressed in proportion to the amount by which you’ve altered its length.

Using this method, you can make a rhythm sequence play “half-time” by stretching it to twice the original length, or “double-time” by shortening the length to half of the original.

...to Adjust It to Fit Its Contents

The function **Functions > Object > Set Optimal Object Sizes** reduces or increases the length of an object, so that it is just large enough to contain the events (or other objects) within it. The object borders are rounded to the nearest bar.



If you hold down the **Alt** key when you call up this function, the borders are rounded to the nearest beat instead.



...to Adjust It to Fit Other Sequences

With **Functions > Object > Remove Overlaps** all selected sequences in a track are searched for overlaps. If an overlap is found, the earlier sequence is shortened by enough to remove the overlap.



With **Functions > Object > Tie Objects**, all selected sequences are lengthened so that they meet the following sequence on the same track.



...Using a Finer Grid

All these operations (moving/copying, lengthening/shortening) snap to the bar or beat grid (depending on the current resolution of the bar ruler, and the setting of the zoom function).

However, there are two ways of reducing the grid:

Display Format values as grid scale:  + operation.

No grid (ticks as grid scale):   + operation.

All you need to do is hold these keys during the particular operation.

Let's say you want to shorten a sequence in a 4/4 bar so that the "4" is played but not the "4" and of the last bar. Enlarge the screen display until you can see quarter notes in the bar ruler. Now grab the bottom right corner of the object and move the mouse to the left until the sequence is shortened by one quarter note. Then press (and hold)  and move the mouse carefully to the right until the sequence has become one division longer. While you are doing this, the right side of the Arrange window? title bar shows the track number, followed by the current length of the object in bars, beats, divisions and ticks. The three right-hand numbers should be #3 1 0".

Dividing Sequences...

Click on the Scissors in the Toolbox. Now click the desired sequence(s), holding down the mouse button. The left side of



the title bar of the Arrange window shows the current position of the mouse. When you release the mouse button, all selected objects are cut at the position shown in the Info *Line* when the button is released. The grid is based on the Display Format value. This value is displayed and is adjustable in the Transport window, just below the Time Signature indicator.

If any notes overlap other notes by more than a 1/16 note, the following dialog will appear on-screen:

Overlapping Notes found! Do you want to keep, to shorten or to split those? [KEEP|SHORTEN|SPLIT]

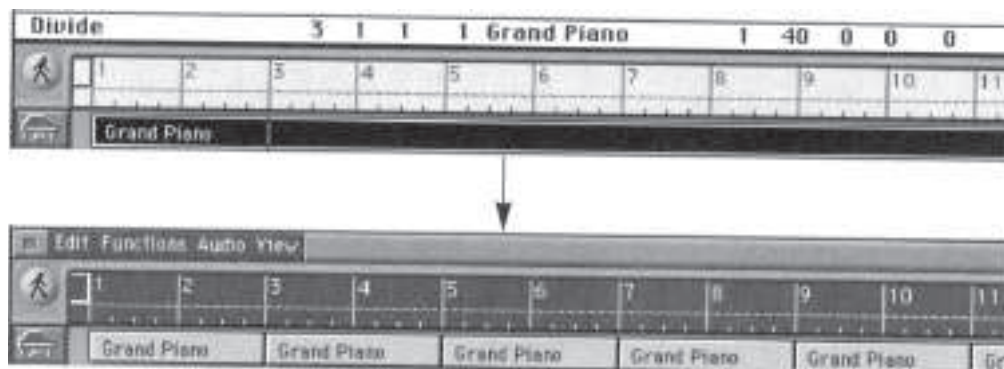
KEEP (default) leaves all notes unaltered. The sequence is cut as usual, but when you do this, you can end up with notes in the left-hand half that are much longer than the sequence containing them. Such notes will play normally, unless “Clip Length” is set to ON, which cuts off all sustaining notes at the end of a sequence.

SHORTEN truncates all overlapping notes, so that they end at the point where the sequence was divided.

SPLIT divides overlapping notes across the two sequences created when a cut is made; two notes are created, with the same pitch and velocity as the original, and with the same total length as the original note.

... Multiple Division with the Scissors

If you hold down the *Alt* key while cutting an object with the Scissors tool, the selected sequence will be cut into multiple pieces, whose lengths are determined by the length of the first segment.



For example: To divide a 16-bar sequence into 8 2-bar sequences, cut the sequence at the start of bar 3, while holding down the Command key.

... at the Locator Positions

The function **Functions > Split/Demix > Split Objects by locators** cuts all selected sequences which are located wholly or partly between the locators, at the left and right locator positions. All the objects created between the locator positions as a result of the cutting operation, are then selected.



...graphically

You can achieve the same effect by defining a cycle region with the mouse in the bar ruler, while holding down the  key. The difference between this method and the above, is that if no objects are selected, they are all cut into pieces.

Cut at the Song Position

With **Functions > Split/Demix > Split Objects by Song Position**, you can split all selected objects exactly at the SPL.



To cut at the nearest bar, use the key command *Split Objects by rounded Song Position*.



Merging Sequences

You can merge two or more selected sequences in a track into a single sequence, by clicking on one of them with the Glue tool.



The function of the Glue tool is the same as **Functions > Merge Objects**. This merges all selected sequences, even those on different tracks, into a single sequence, in which all the events from the individual objects retain their original time position. Be aware that the individual MIDI channels are replaced by the MIDI channel of the current track instrument. The new object adopts the name and track of the first (along the time line) of the merged objects.



See the section *Merge/Normalize and MIDI Channels* on page 3 - 38 for more on this.

The function **Functions > Merge > Objects per Tracks** has the same effect as Merge Objects, but if the selected objects are on several different tracks they are combined into one object per track.



Merge > All Objects, **Merge > Objects per Tracks** and the Glue tool handle the sequence parameters *Transpose*, *Velocity* and *Dynamics* intelligently. If these parameters are not identical in all selected sequences, the sequences are normalized before the merge (just as in older versions). However, if one of the sequence parameters is set identically in all selected sequences, for instance the Transpose value, that value is preserved, and the events remain unchanged.

Demixing Sequences

The function **Functions > Split/Demix > Demix by Event Channel** searches the selected sequence for events with different MIDI channels. A separate sequence containing all the matching events is created for every MIDI channel found. Each of these sequences is created on a track with an instrument that has the same MIDI channel. If no such tracks exist, the tracks



are created with the track instrument of the original instrument.

This function is useful for the import of Standard MIDI Files (Format 0) when all note events are located in a single sequence.

The function **Demix by Note Pitch** searches the selected sequence for notes with different note numbers. A different sequence of the same length is created for every note number found. The tracks created for these sequences will have the same instrument as the original sequence. This feature is especially useful for separating drum parts that have been recorded into Logic from a drum machine.

Muting Sequences or Regions

Often when you are arranging, you will want to test musical ideas by muting certain objects. This is what the Mute tool is for.



You can mute individual or selected objects by clicking on them with the tool. They are then displayed with a dot before the object name. Clicking a muted object reverses this state (unmuting).

You can perform the same function using the *Mute Folders/Sequences* key command.

Soloing Sequences or Regions

You can solo any selected sequences using the Solo button in the Transport window, or the key command of the same name. Double-clicking on the Solo button or using the key command *Solo Lock* “locks” the soloed status, so that changing the selection doesn’t affect what is being soloed. This is especially useful when you want to make changes to the Playback parameters of a sequence while listening to a specific solo group. Another click (or using the key command again) quits solo mode. For more on this, read the section *Solo and Solo Lock* on page 2-14.



When the Solo function is activated, the Bar Ruler will change from gray to yellow to make it easy to see that Solo is engaged.

The Solo tool enables you to solo individual or selected objects, by clicking them and holding down the mouse button. You can also scrub the object by moving the mouse horizontally. If Logic is in play mode, solo playback starts as soon as you stop scrubbing. Release the mouse button to end solo listening.



Naming a Sequence or Region

One sequence: click the sequence with the Text marker tool and a text input box appears. Enter the name and press ***Enter***, or click another object, or the background.



The same applies if you are entering the name in the Sequence Parameter box.

Multiple Naming...

Select a group of sequences (e.g. with the rubber band), and click one of them with the Text tool. A text input box will appear, where you can name the sequence. All sequences are given the same name.

...with incrementing Numbers

If you end the name of the sequence with a number, all selected sequences are given the same name, with increasing numbers. They are numbered according to the time position of the sequences. If you want to end all the sequences with the same number, just add a space after the number.

The above also applies if you enter the name in the top line in the Sequence Parameter box, instead of using the text tool.

Transferring Track Names to Sequences

A quick reminder: the track name is normally used as the default name for any sequences recorded on it. However, if you have copied sequences from another track and you want to give them the same name as the track:

The function **View > Tracknames to Objects** gives all selected objects the same name as the track they are on.

Special Functions

Inserting Events

MIDI events can be added directly to the selected track in the Arrange window, at the current song position, from one of the edit windows (or even from another song) from the clipboard.

This function allows you to insert MIDI events-e. g. from the Matrix Editor-directly into the Arrange window. Logic checks as it does this that you have selected a sequence, and then adds the data to that sequence. If you have not selected any sequences, Logic will create a new sequence on the selected track.

Deleting Events

...Outside the sequence borders

Functions > Erase MIDI Events > Outside Object Borders

allows you to erase all MIDI events beyond the start and end points of an object.

...Duplicate Events

The function **Functions > Erase MIDI Events > Duplicates** searches all selected sequences and all sequences in selected folders, for identical events with the same time position. Any events occurring twice or more at the same position are deleted (so that only one remains).



The search looks for notes, controllers, monophonic after touch and program change data.

Events of the same type, but with different channels are not treated as identical.

The “same time position” includes two notes which are output simultaneously because of the current quantization setting.

There are more functions for deleting events on the Arrange level, which are described in the section *Copying Events Selectively* on page 3-77.

For more on the other functions used for editing events on the Arrange level, take a look at the section *Arrange Window Techniques* on page 3-74.

3.6 Sequence Playback Parameters

The Sequence Parameter box is located to the left of the track list, above the tool box.

Displaying Sequence Parameters

When you select a sequence object, its parameters are automatically displayed in the Sequence Parameter box. If the Sequence Parameter box isn't visible, select **View > Hide/Show Parameters**).

None of these parameters alter the original data of the sequence, they only affect the playback.

Default Sequence Parameters

If no sequence is selected, the upper line of the sequence parameter box will read "*MIDI Thru*". Any live MIDI input (in stop mode as well as during recording or playback), will be played back with the instrument and settings chosen here. When you record a new sequence, the settings in the **> MIDI Thru >** para-

meter box are carried over into the new sequence's parameter box. The MIDI Thru parameter box can be viewed as an adjustable "default" parameter box.

Realtime processing

An important concept to understand is that incoming MIDI data is always relayed "through" the instrument assigned to the track that is currently selected—as long as you do not disengage the record enable button, or exclude the chosen MIDI channel from MIDI Thru. This data is also processed through the MIDI Thru parameter box, described above. The MIDI Thru parameters are always displayed and are adjustable when there are no sequences selected. Click on the window background to deselect all sequences.



Setting up the MIDI Thru parameters to match those of a sequence

Using the key command Set *MIDI Thru Parameters by Seq. 1 Folder*, you can set the MIDI Thru parameters to match those of a selected object (whether a sequence or folder). This action also selects the track belonging to that object.



Editing Several Sequences Simultaneously

If several sequences are selected, the number of selected sequences is displayed, instead of an object name. If you now alter any sequence parameter, this alteration affects all the selected sequences. If a parameter was set differently in the individual sequences, a >*< appears. You can still alter this parameter for all the selected sequences, and the value differential will be retained (relative alteration). If you want to set the same value for all the selected sequences, hold down  while you input the value (absolute alteration).

Opening and Closing

To the left of the name is a small triangle; clicking this toggles between showing or hiding the contents of the box, just as with folders in the Macintosh Finder (if you are viewing the

contents *By Name*). Closing the box leaves more room for the elements below.

The Sequence Parameters

Though these are called sequence parameters, they may also be made to affect folders, in which case they apply to all the sequences within them. The following paragraphs will often refer to objects, which in this case means arrange objects, i.e. sequences or folders.

If a sequence parameter is set to its default value, the display of the parameter value is hidden, to make the display clearer.

Name

The top line in the Sequence Parameter box simply shows the name of the selected sequence.

Qua

Covered in section *Quantization* on page 3-41.

Loop

Value range: On or Off. Normal value: Off.

When switched On, the object is repeated on its track until it encounters another object. A loop will also terminate at the end marker of a folder (if the sequence being looped is within a folder), or the song end marker. These repeats are displayed as grey beams (with no object name). The length of each loop is exactly the same as the length of the original object, so if you want to create polyrhythmic structures, try experimenting with the length of the original object.



One way of ending a loop early is to create an empty sequence with the Pencil. A better method is to place the object with the loop in a folder. You can then control the total number of loops, by simply altering the length of the folder.

The *Toggle Loop* key command may be used to switch the loop parameters for selected objects on and off.

Functions > Sequence Parameters > Turn loops to Real Copies transforms the loops into real copies of the original object, and simultaneously switches the *Loop* parameter for the resulting objects to off.

Functions > Sequence Parameter > Loops to Aliases changes sequence loops to aliases. This function is also available as a key command.

Transpose

Value range: +/- 96 semitones. Normal value: 0.

All note events contained in the sequence are transposed up or down by the selected amount during playback. Even complete folders can be instantly transposed in this way. If some individual sequences within the folder have already been transposed the relative differences between them are retained.

If you want to transpose by octaves, grab to the left of the parameter field between the word “Transpose” and the actual value. This opens a flip menu to allow direct input of octave transpositions.

To guard against drum notes, etc. being transposed, the Instrument Parameter box contains a parameter *No Instrument Transpose*. If you place a check in this box, the transpose parameter is ignored in all sequences played by this instrument.

Velocity

Value range: +/- 99. Normal value: 0.

All notes in the relevant object are offset by the selected value. Positive values add to the originally recorded velocity, and negative ones subtract from it, although naturally it is impossible to go outside the limits prescribed by the MIDI Standard (0-127). If you select a velocity offset that exceeds the maximum or minimum possible value for a particular note, that note will play at the extreme possible range. For example, a setting of + 20 will cause a note with a velocity of 120 to play at 127.

Dynamics

Value range: see diagram. Normal value: 700%.

This parameter also affects the velocity values of the notes, but instead of adding or subtracting a fixed amount, the differences between “soft” and “loud” notes (the dynamics) are increased or decreased. This works in a similar way to a compressor or expander. Values above 100% expand the dynamics, and so increase the difference between “loud” and “soft”, while values below 100% compress the dynamics, i.e., reduces the differences in velocity.

The FIX setting causes all notes to be transmitted with a velocity of 64. When used in conjunction with the velocity parameter (see above), it is therefore possible to set any fixed velocity value.

Gate Time

Value range: see diagram. Normal value: 700%.

The term “gate time” stems from the technology used in analog synthesizers, and refers to the time between pressing

and releasing a key. This parameter therefore affects the absolute note duration or length. This should not be confused with the musical note value, which normally refers to the amount of time until the next note. The practical effect is to make the notes in the object more staccato or legato. The parameter range is referenced to the original note lengths. “*Fix*” produces extreme staccato. Values below 100% shorten the note lengths. Values above 100% lengthen the notes. The “leg.” setting produces a completely legato effect for all the notes, no matter what their original lengths, eliminating all space between notes in the affected sequence. If this is used on a folder, all notes in all sequences in the folder will be affected.

Delay

Value range: -999 to 9999 ticks. Normal value: 0.

This parameter alters the time position of the selected objects. Positive values correspond to a delay (laid-back playing style or dragging), negative values cause a pre-delay (driving or rushing).

The units are ticks. A tick is the smallest time resolution in a sequencer, which in the case of Logic is 1/3840th note. On the right side of the delay value field you can use the mouse as slider to input the value.

If you click to the left, between the word “Delay” and the delay value, the flip menu shown here, opens where you may set the delay in terms of note value.

You can use the View menu to switch the display to milliseconds, using **Delay in ms**. This can also be used to calculate delay times for different note values at various tempos:

Set the desired note value by clicking to the left of the delay value.

Go to the View menu and switch to **Delay in ms**.

Set the desired tempo in the transport panel,



The delay parameter will now show the delay time.

The delay parameter is mainly used for musical purposes. It is also a way of fixing timing problems that may occur for a variety of reasons. Some examples of these are:

The attack phase of the sound is too slow. A good musician will automatically compensate for this by playing the notes slightly early. With very slow sounds, you may need a pre-delay of over 100 ms to even them out.

The sound generator is reacting too slowly to the incoming note-on messages. Older multitimbral sound generators often take tens of milliseconds before beginning to output a voice. This effect may be better compensated for by using the delay parameter in the instrument parameter box, since this will affect all sequences on all tracks sending to the slowly reacting device.

The delay in outputting the voice is not constant because it depends on the order of notes arriving serially at the sound generator. You should therefore try pre-delaying rhythmically important tracks by as little as one tick—it can work wonders.



Active MIDI Transmission (AMT), featured on the Unitor 8 and AMT 8 MIDI Interfaces, can solve this problem, because they can output MIDI events simultaneously on different MIDI Outputs. Always make sure to share the load of MIDI data among as many MIDI outputs as possible.

Fixing/Neutralizing Sequence Parameters

You can normalize the sequence parameter settings of all selected objects using the command **Functions > Sequence/Instrument Parameters > Normalize Sequence Parameters (Normalize)**. This means that all settings are actually written to the data, and the playback parameters resume their normal values. The audible result remains the same. The loop and extended sequence parameters are not affected. Using this function is effectively like saying “make these sequence/instrument parameter values permanent”. Under most circum-



stances, it is better not to do this, as it is more flexible to leave the original data untouched. This gives you unlimited opportunity to change your mind about sequence edits.

Merge/Normalize and MIDI Channels

Just like the Merge function or the Glue tool, the Normalize function is intelligent with respect to stored MIDI channel numbers. If all stored events have the same MIDI channel number, the channel will be changed into that of the instrument assigned to the current track. If the events are on different channels, Logic will ask whether you would like to convert the channels or not.

In addition, the following variations of Normalize are now available by selecting **Functions > Sequence | Instrument Parameter >**

... **Normalize w/o channel** leaves the stored channel number untouched;



... **Normalize w/o channel & delay** leaves the stored channel number *and* delay parameters untouched.



If the instrument playing back is has a channel setting of *All*, or you're dealing with a completely different type of Environment object (e.g. a Channel Splitter, for use with *A-Playback*), then the stored MIDI channel numbers will remain unaffected by the “usual” Normalize function.

A quick tip: if you're editing sequences that are appearing as notation on a polyphonic score style, we recommend using *Normalize without channel*, as the event channel is used in the Score Editor to assign notes to individual polyphonic voices.

Extended Sequence Parameter Box



The Extended Sequence Parameter box contains additional sequence parameters. These include the more comprehensive quantization parameters which are covered from page 3-41 onwards.

Opening the extended sequence parameter box

To open the float window containing the Extended Sequence Parameters, you either use the command **Options > Extended Sequence Parameters**, or double-click on the left half of the of the Sequence Parameter box (next to the parameter names).



Clip Length

Value range: On / Off. Neutral value: Off.

When switched On, this parameter causes any notes sustaining beyond the end of a sequence to stop playing at the end of the sequence. Remember however, Clip Length is a purely playback parameter, the data of these longer notes is not affected, and their display in the editor windows is unchanged.

The point of this function is to allow you to alter the length of a note directly on the arrange level, by adjusting the length of the sequence. Clip length has no effect on folders.

Score

Value range: On / Off. Neutral value: On.

When switched *Off*, this parameter removes the selected object(s) from the Score Edit window. This is useful if the sequence does not contain any notes, but just other data, such as controller data for fader automation.

Special Functions for Sequences

Adding Prg, Vol and Pan as Events

The command **Functions > Sequence/Instrument Parameter > Insert Instrument MIDI Settings as Events** allows you to insert the program, volume, and pan (*Prg, Vol, Pan*) settings as actual MIDI events from the Instrument Parameter box to one or more selected sequences. The events are placed a quarter note before the beginning of the relevant sequences. Only the activated (checked) settings in the Instrument Parameter box(es) are added as events. If there are already any events of the same kind, these are overwritten.



Adding Variable Program Changes

The function **Functions > Sequence/Instrument Parameter > Insert Variable Program Changes** in the same menu adds a “variable program change command” to one or more selected sequences. This is placed a quarter note before the beginning of the sequence(s) concerned.



0 4 1 1 Program 1 of Instrument

During playback, this event replaces the missing program number in the Prg-Ch message with the setting in the Instrument Parameter box. If you subsequently change the Prg setting in the Instrument Parameter box, the “variable” program changes in the sequences on that track will change accordingly. By clicking the checked box in the in Event List, you can turn this into a normal “fixed” program change event. You can also alter this event’s position or its program number data byte in the Event List.

3.7 Quantization

Quantization is the rhythmic correction of notes to a specific time grid. Any inaccurately played notes are moved to the nearest position on this grid.

For example, if the smallest notes in a passage are 1/16th notes, you should use 1/16th quantization to move all recorded notes to their ideal rhythmic value. This will only work if no note has been played more than 1/32nd from the ideal position, otherwise the note will be moved 1/16th later or earlier than the proper position.

How It Works

Quantization settings are made in either the Sequence Parameter box or the Extended Sequence Parameter box. They are non-destructive playback parameters, so they can be replaced by another setting at any time. This also means that by setting the **Qua** parameter to off (3840), you can always revert to the unquantized original recording. Unlike other playback parameters, quantization affects the way notes are displayed in the editor windows, thus allowing you to see from the position of the notes what effect the quantization parameters are having. The quantization affects only notes, not other types of events (e.g. controllers).

For every note event, two positions are stored internally: the *original position* that was played in, and the *playback position* (also shown in the Editor windows). For unquantized sequences, both positions are the same. Each time you quantize, a new playback position is calculated from the *original position*.

The *Fix Quantize* command (see the section *Fixing the Quantization* on page 3-46) overwrites the original position with the *playback position*. The same thing happens if you change a note in one of the Editors manually-although then you can't return to the original value.

Which Events Can Be Quantized

Sequence parameter quantization only works on notes, not on other types of events like controllers. Apart from notes, all

events have just one position parameter, which you can leave or change permanently by using Event Quantize, or dragging with the mouse.

To quantize events other than notes, read the section *Quantizing Events* on page 1-40.

Essentially, quantization works on the whole sequence. To quantize single notes only, use the Event quantization in one of the Editor windows (see the section *Note Quantization* on page 1-41). Keep in mind that this overwrites the original record position of the quantized note.

If you wanted to quantize various parts of a sequence differently, you could divide the sequence up, and use different quantization settings on each part of the sequence. You can then recombine the parts of the original sequences without this affecting the way the parts are played.

The quantization grid always begins at the start of a sequence. If the sequence does not start at the beginning of the bar, neither does the quantization grid.

Qua

The quantization grid is accessed from a flip menu, under the *Qua* parameter in the Sequence Parameter box.

You can also step through the available quantization resolutions from the flip menu, one at a time, using the key commands *Quantize: next value* and *Quantize: previous value*. This function sets the quantize for all selected sequences.



Quantization off

The setting *off* (3840) plays the notes with the finest possible time resolution: 1/3840 note, which is practically unquantized playback.

Normal quantization

The settings 1/1, 1/2, 1/4, 1/8, 1/16, 1/32, and 1/64 note quantize the sequence to the equivalent note value.

Triplet quantization

The settings 1/3, 1/6, 1/12, 1/24, 1/48, and 1/96 note quantize the sequence to triplet note values. A 1/6 note is equivalent to a quarter triplet, 1/12 note to an eighth triplet, 1/24 note to a sixteenth triplet and 1/48 note to a thirty-second triplet, etc...

Mixed quantization

The setting 8 & 12 corresponds to eighths and eighth triplets, 16 & 12 to sixteenths and eighth triplets and 16 & 24 to sixteenths and sixteenth triplets. Mixed quantization always applies both note values, and thus naturally requires greater precision when playing.

Odd quantization

The setting *9-Tuplet* means novetuplets (1 bar = 9 beats), *7-Tuplet* is septuplets (1 bar = 7 beats), *5-Tuplet/4* is quarterquintuplets (1 bar = 5 beats), and *5-Tuplet/8* is eighthquintuplets (1 bar = 10 beats).

Additional Quantization Parameters

The following quantization parameters are located in the Extended Sequence Parameter box, which can either be opened by double-clicking on the left half of the Sequence Parameter box, or with **Options > Extended Sequence Parameters...**



Q-Swing

Value range: 1% - 99%. Normal value: 50%.

This percentage value alters the position of every second point in the current quantization grid. Values over 50% delay the beats and values under 50% pre-delay them. The best practical

settings are between 50% and 75% which give strictly quantized sequences a swing feeling.

Q-Strength

Value range: 0% to 100%. Normal value: 100%.

This percentage value determines how far a note is shifted towards the nearest grid position. 100% produces a full quantization, 0% leaves the note in its recorded position.

Q-Range

Value range: 0 to 3840 ticks. Normal value: 0 ticks.

All notes whose distance (in ticks) from the nearest grid position is greater than the value here, are not quantized. A value of 0 means that every note is quantized.



If you enter negative values for *Q-Range*, you are using *Far only Quantize*. In this mode, only the notes *outside* the set region (in ticks) are moved to the ideal quantization grid positions; while notes closer in remain unquantized. This way, the most poorly played notes outside this region are moved to perfect timing positions on the quantization grid, or at least moved towards these positions, depending on the Q-Strength setting (see below).

Q-Flam

Value range: ± 3840 ticks. Normal value: 0 ticks.

Notes with the same time position (e.g. chords) are spread out by this parameter. Positive values produce an upwards arpeggio, negative values a downwards arpeggio. The first note of the arpeggio is unaltered.

Q-Veloc

Value range: -99% to +127% Normal value: 0%.

This percentage value determines how much the velocity values of the quantized notes are affected by the velocity

values in the template sequence (see the section Groove on page 3-46). At 0% the notes retain their original velocity, and at 100% they adopt the velocity values of the template. Negative values alter the velocity so that the deviation from the template becomes even greater.

Q-Length

Value range: -99% to +127%. Normal value: 0%.

This percentage value determines how strongly the note lengths of the quantized notes are affected by the equivalent note lengths of the template sequence (see the section Groove on page 3-46). 0% has no effect, while at 100%, the notes adopt the exact note lengths of the template. Negative values alter the note lengths so that the deviation from the template is increased.

Linear Quantize (“Q-Range”)

“Linear Quantize” is a quantize algorithm which is always used automatically whenever you set a positive value in the “Q Range” parameter.

The “Q Range” parameter is located in the Extended Sequence Parameter box, which you can open by selecting **Options > Extended Sequence Parameters**.

Operation

Linear Quantize is well-suited to “soft quantization” of recordings which already have the right groove but in some places are too hurried or “laid back”. Linear Quantize retains the original feeling, but positions the rhythmic “center” precisely in the groove.

To obtain the best results, use a low, even quantize value, such as: 1/4-note. Set the “Q Range” parameter to the maximum error in the recording, for example 96 ticks (= 1/40-note).

Don't forget desite the 1/4 quantization setting, shorter note values such as 1/32 or 1/16 triplets are retained. (as long as the Q-Range parameter is set to a Lower value.)

Linear Quantize is a very musical quantization strategy that requires a certain amount of technical musical prowess.

Fixing the Quantization

In the same way that the other sequence parameters can be normalized, the quantization settings can also be applied permanently to the stored data.

The command is **Functions > Sequence/Instrument Parameters > Fix Quantize**.



Of course, once this is done, it is impossible to revert to the original recording, unless you use *Undo* immediately after performing the operation.



Pre-quantization

The **Fix Quantize** function is especially suited to Pre-quantization. For example, if a recorded sequence cannot be immediately quantized to a 1/16 swing without producing a few incorrect notes, start by quantizing it to 1/16th notes and then call up Fix Quantize. You can now apply any swing quantization to this cleaned-up version of the sequence.

If you are recording a hi-hat pattern consisting of 1/16 notes and one single 1/32 note, first play the rhythm (quantized to 1/16 notes) without the 1/32 note and choose Fix Quantize. You can now raise the quantization to 1/32 notes and add the extra note without misquantizing any badly-played 1/16 notes.

3.8 Groove

In the early days of sequencers, many people thought that electronic music sounded mechanical or sterile. This impression

(which, at the time, might have been a fair observation) was based on the fact that sequencers could only output notes that were quantized. This is no longer the case. Logic's resolution is so high that at a tempo of 120 it can differentiate between two notes that are just 520 microseconds apart. This is equivalent to a single oscillation cycle of the highest note that you can play on a soprano saxophone.

So, if you think modern sequencers always sound mechanical, you probably haven't yet had the pleasure of working with Logic.

Of course, using the quantization functions described from page 3-41 onwards, it is still *possible* to generate sterile, mechanical timing, if that is what you actually want (although even straight quantization can be given a groove with the aid of the Q-Swing parameter).

Logic also allows you to use your own quantization grid (or Groove Template), making full use of the system's high timing resolution.

Your Own Quantization Grid (Groove Template)

Instead of quantizing the notes to one of the preset grids, you can create new ones based on the rhythms in other sequences. These are referred to as "groove templates". Groove templates make it possible to capture the small timing deviations which give a sequence its "feel", and apply them to other sequences.

Creating Your Own Quantization Template (Groove)

The function **Options > Groove Templates > Make Groove Template** transforms the exact timing of the notes in the selected sequence into a quantization template. It then appears in the flip menu of the **Qua** parameter (in the Sequence Parameter box), with the name of the sequence on which it is based.



It can be used on any other sequence, in exactly the same way as the rest of the items in the Quantize flip menu.

In general, two-bar model sequences work particularly well for this function, but you can use sequences of any length. Make sure that the model sequence actually contains a note at every desired quantization value.

Deleting a Quantization Template

Use Options > Groove Templates > Remove Groove Template from list to delete the selected model sequence from the list of possible quantization templates. The next time you open the flip menu, this quantization template will no longer be available.

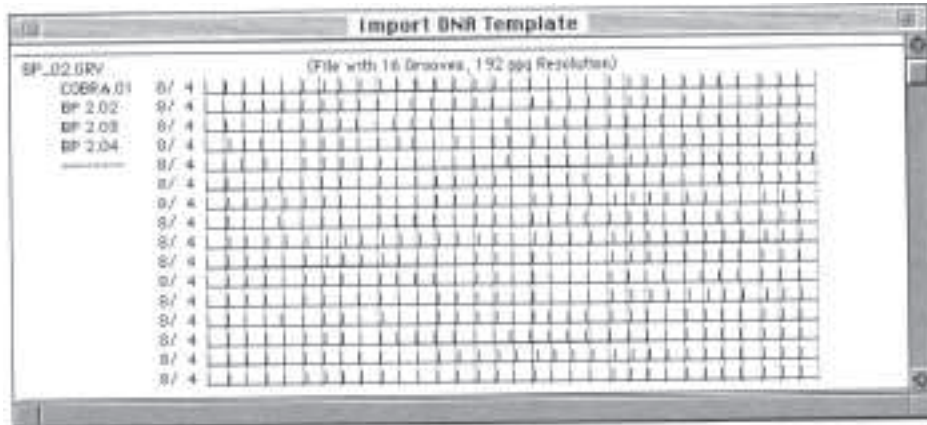


Importing Quantization Templates

You can also import 3rd-party quantization templates, such as those made by the Canadian firm WC Music Research (DNA Groove Templates). First, you must copy the templates into a folder called “Grooves” which must be on the same level as the Logic program itself. If necessary, create a folder there with that name yourself.

You can then use the function **Options > Groove Templates > Import DNA Groove Templates...** to import the templates. All the DNA Groove Templates in the Grooves folder are displayed graphically, along with their names. Just click a groove template to apply it immediately to the selected sequences. If you like the groove, you can transfer it into Logic’s list of quantization templates, by double-clicking on it.



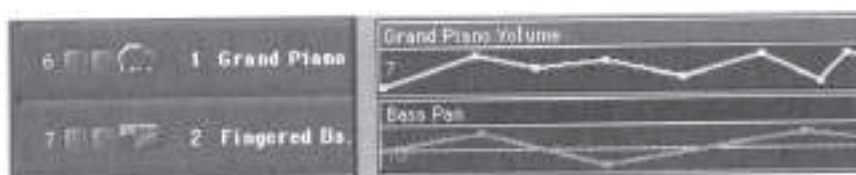


Logic then creates a folder in the Arrange area (if one is not already present) for these imported groove templates called “Grooves!” The groove templates are then stored as sequences within the folder.

By converting the DNA grooves into stored sequences, the quantization templates can now be used, even when the song is played on another computer whose hard disk does not have a Grooves folder with the corresponding DNA Groove Templates.

3.9 Hyper Draw in the Arrange Window

Hyper Draw is the easiest way to create and edit volume and pan changes (or almost any kind of MIDI data) in the Arrange window. The changes in MIDI data are made by graphically inserting points, which represent fixed controller values. The points are then interpolated automatically by Logic, which creates a series of events to smoothly connect the first MIDI event with the last. These automatically generated events are recalculated every time you edit the curve points.



Hyper Draw is activated on a per sequence basis. For example, you could open Hyper Draw windows to control volume in one sequence, and pan in another.

In sections where the MIDI data being edited does not change in value, the data curve appears as a thin, horizontal line.

You must set a large enough vertical zoom resolution in order to see the HyperDraw curves.

Despite some similarities, don't confuse the HyperDraw function with the Hyper Editor, which is described under section *The Hyper Editor* on page 11-1.

Activating Hyper Draw

To activate Hyper Draw for the currently-selected sequence(s), select **Options > Hyper Draw...** You can select different Hyper Draw modes for different sequences.

Event Type

You can select the following types of events directly from the Hyper Draw menu:

- Volume
- Panning
- Balance
- Modulation
- Breath
- Foot Control
- Portamento Time
- Expression
- Channel Pressure
- Pitch Bend
- Program Change

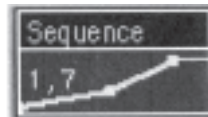
MIDI Channel

By **calling up View > HyperDraw > Channel . . .** you can select the MIDI channel on which events will be displayed and edited.

Display

When sequences are displayed in Hyper Draw mode, both the MIDI channel and the Controller being edited are displayed to the left, divided by a comma.

MIDI channel # 1,
Controller # 7 (Volume)



If the channel is set to auto (default) the first number (MIDI channel) will not be displayed. Then the controller events of all channels will be displayed. Inserted events get the channel of the track's instrument.

Autodefine

This Option sets the Hyper Draw parameters automatically, so that the first event in the sequence will be displayed.



Auto Define is not limited to switching the HyperDraw display to the first incoming controller, but it can also respond to other types of events as well. It responds in this order:

Controller
Aftertouch (Channel Pressure)
Pitch Bend
Program Change
Note Velocity

Sustain Pedal (Controller 64)

Controller 64 (sustain) is displayed in a special way in HyperDraw: there is no graph, merely the event's "on" and "off".

Defining other data types in Hyper Draw



If you **want** to use Hyper Draw to edit a controller which is not directly available in the Hyper Draw menu, call up **View > Hyper Draw > Other...** and enter the number of the controller and the MIDI channel in the resulting window. As soon as you select a new sequence, you will see the corresponding Hyper Draw parameters in this window.



This floating window always displays the Hyper Draw parameters for the selected sequence(s).

Deactivating Hyper Draw

To switch off Hyper Draw for selected sequence(s), select **Options > HyperDraw > Disable**.

Key Commands for Hyper Draw

The following key commands are available in the Arrange window, to speed up the use of Hyper Draw:



Hyper Draw: Disable

Hyper Draw: Modulation

Hyper Draw: Volume

Hyper Draw: Pan

Hyper Draw: Other..

Hyper Draw-Functions

Making Hyper Draw visible

If you can't see the blue Hyper Draw window, increase the vertical resolution (zoom) of the Arrange window, either with a key command, the telescope icons, or by holding down **alt** and using the magnifying glass tool.

You can zoom individual tracks by touching them to the left of the track number:

Quick Delete

You can delete all events displayed in Hyper Draw for a sequence by Shift-Alt-double-clicking in the blue HyperDraw area.

Adding new points

Just click on the blue background of a sequence with Hyper Draw active—you can use either the Pointer or the Pencil, and a point will appear. As long as you hold down the mouse, you can move any of these points. The position and value are displayed in the Info Line at the upper edge of the Arrange window.

No events will be inserted until you release the mouse button.

Deleting points

Click briefly on any point to delete it.

Moving an existing point

Just grab a point with the mouse and move it. If you hold down the mouse button while you do this, you can restore erased curve points by dragging the mouse over where they used to be.

Fine-Tuning Curve Points

As you move a point, hold down . You will then be limited to moving the point vertically, but be able to select all 127 possible steps.

Moving a Curve Point and All Subsequent Curve Points

Hold down *Alt* while moving a point. All following points will then move as well. You can then adjust the “amplitude” of what

is, to all intents and purposes, a controller “waveform”, by simply touching its upper or lower edge with the mouse.

Moving a Curve Vertically Only

With , the movement is limited to the vertical (value) direction. The time position will not change.

Using Colors with Hyper Draw

You can globally assign colors for different controller numbers and MIDI channels, which are then used as presets to display the Hyper Draw lines. However the background color for Hyper Draw (blue) is fixed.

Select a Hyper Draw sequence, open the color selection window by selecting **View > Object Colors...** and choose a color. This will globally assign the color to this controller and channel. Now whenever you select this controller on this channel in any song, the color assignment will be used.

If you want to change the color of a sequence, you must first deactivate the Hyper Draw function.

Making Practical Use of Hyper Draw

Hyper Draw is very simple to use. For example, select **View > HyperDraw > Volume**. A blue area appears beneath the sequence name, and by clicking there with the mouse, you can insert or delete the points. You can also move the points by holding down the mouse button. While you are editing an Info Line appears at the top of the Arrangewindow, indicating the bar position and controller value.

If you want to edit pan values, select **View > HyperDraw > Pan**. This means that only pan events will be generated and displayed. Only one type of event can be displayed at any one time.

If you've selected *Cha:All* in the current Instrument Parameter box, so that all 16 MIDI channels will play back, you will have to assign the MIDI channel you want to edit with **View > Hyper Draw > Channel...**

Editing Sequences-Move and Copy with HyperDraw engaged

You can still move and copy a sequence that has Hyper Draw active, by grabbing it by its top edge, above the blue Hyper Draw area.

When Hyper Draw is active, you can still change the start point and length of a sequence, but you have to grab above the blue Hyper Draw area in order to move the left or right edge.

HyperDraw and Cutting with the Scissors

When using the Scissors to cut audio regions or MIDI sequences, Logic is intelligent regarding the HyperDraw nodes: New nodes are created for all event types if necessary at the split points in both new segments just before resp. after the cut.



If Hyper Draw starts behaving oddly...

Hyper Draw supports Program Change events. These are not interpolated like Controller events to give smooth curves; instead, one program number remains set until the next point.

*Controller events will only be displayed if their MIDI channel matches the one set for Hyper Draw (set by calling up **View > HyperDraw > Channel...**).*

If you make alterations to a group of events, such as Volumes, in another editor (e.g. the Event or Hyper Editor), the Hyper Draw display will not update until you have selected or deselected the relevant sequence.

3.10 All about Folders

A folder is an arrange object that can contain other arrange objects, in the same way that a folder in the Finder can contain other folders or files.

One way to imagine a folder is as a song within a song. A folder can have as many tracks as you want, with sequences on them. The inside of a folder looks just the same as the arrange area and track list in a song.

Within a track of the song, a folder looks like a sequence, only with a dotted surface.



This could contain tracks like trumpet, saxophone and trombone which you arrange into a brass section, or 14 tracks of drum instruments, which you may want to treat as a single drum pattern object.

In the same way, your entire song, with all its tracks and sequences, could itself be a folder, appearing as a grey beam in another song. In this way, you could arrange several songs to make up a concert.

This is not all folders can do. You could use folders to represent the parts of a song (e.g. choruses and verses). And just as on your hard drive, you can place as many folders as you like within other folders, within yet more folders (say for the instrument groups within the different parts of the song, for example), with no limit to the number of levels you can create, if you wish.

Another possible usage might be to store different arrangements of a song in different folders, so you could switch between them rapidly... That's enough for you to be thinking about for the moment. We'll leave the rest up to your imagination...

The “Folder” as a Track Instrument

2  (Folder) A folder is normally placed on a track assigned to *folder* in the track list, instead of to an instrument.

This means that all the objects in the folder are played by the instruments set in the *folder's* track list. All the instruments within the folder will play back, just as they do on the top level of the Arrange window.

If you place a folder on a track that is set to a normal instrument, it's entire contents are played by this instrument. This usually only makes sense if the folder contains tracks for just this Instrument. However, this could be a quick way of listening to a string arrangement, if for example, some of the intended sound sources are unavailable.

Creating Folders

The function **Functions > Pack Folder** places all the selected objects into a folder. The folder is created on an existing folder track. If no track with a folder instrument is available, Logic creates one.



In a similar way, if no object is selected, Logic creates a track with a folder on it. This contains no objects-just tracks with all the instruments from the current level.

Changing Display Levels

Going into a Folder

To change to a lower level (go into a folder) of the Arrange window, double-click the folder itself.

Go *Into Folder or Sequence* does the same, but make sure the folder is selected before choosing this option. If a sequence is selected instead, an Event List is opened showing its contents.



Double-clicking on a track assigned as “folder” in the track list, opens the track mixer, as described in the section *Track Mixer* on page 6-4.

Exiting a Folder

To change to a higher display level of the Arrange window, double-click on the background.

The key command *Go Out of Folder or Sequence* does the same.



Moving Objects into Folders

...is as easy as drag-and-drop. Drag the object(s) from the original track onto the folder at the desired position and release the mouse. If the folder does not already contain a track with the same instrument as the original track, Logic creates one. When you look inside the folder, you will see the object you dropped, at the position you dropped it.

Unpacking Individual Objects

If you want to move individual objects from a folder to a higher level there are two ways of doing so:

Go to the display level where you want to add the object.

Now open a second Arrange window by pressing  (default). Go into the folder from which you want to remove the object and drag it from one Arrange window into the other.

Go to the display level from which you want to remove the object. Select the object and transfer it onto the clipboard by pressing  . Change to the display level where you want to add the object. Select the desired track, and set the song position line to the point you want, and then add the object by pressing Command-V.

Unpacking Folders

Use the command **Functions > Unpack Folder** to dismantle the selected folder. If you select **Unpack Folders (Create new tracks)** Logic will create new tracks on the same level where the folder was located (below the former folder track). If you select **Unpack Folders Use existing tracks** instead, the objects contained within it are placed on tracks set to the appropriate instruments.

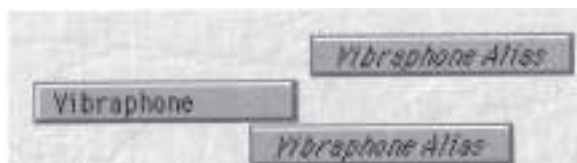


Folder Editing: Like Sequences or Regions

In principle, you can do anything with folders that you can do with sequences. Please reread all the operations described from page 3-18 onwards if you need a refresher for any of these. Anything described here for sequences also applies to folders.

3.11 Aliases

An alias in Logic is the same as an alias in the Finder. It looks like an object, but contains no actual data. It is just a reference to the data of the original object—a virtual “reflection” of the original. You can always recognize an alias, because its name is written in italics.



If the data in the original object is altered, this immediately affects all the aliases created from it.

In fact, this is the whole point of aliases. For example, if a riff or phrase (a sequence or folder) keeps recurring throughout a song, it makes sense to use aliases, rather than have full copies of the original eat up your storage space. Furthermore, when using aliases, if you feel something is not quite right, you only have to alter the original, and the correction will automatically

take effect throughout the whole song. If you just want to alter a detail at one point in the song, you can turn that particular alias into an independent object.

Creating an Alias

To create an alias, just hold down  when you are copying the object as usual (so hold down   +drag).

There is also the command **Alias > Make** in the **Structure** menu. The alias appears on the selected track, beginning at the current song position.



If several objects are selected, their relative time and track positions are retained. The selected track is the destination track for the first object along the time line.

When you use the function **Repeat Objects**, you have the option *as Alias* (see page 3-21).

You can use **Functions > Alias > Make but Copy Folder** to create aliases of the current selection. If a folder is selected, a “real” copy of the folder is created, and aliases of all the objects in the original folder are placed inside. The advantage is that within this “real” folder, you can (for example) set up mutes, and alter program, instrument or playback parameters if, you wish to.

Playback Parameters

Although an alias is a “slave” of the original, it has its own playback parameters. The exception is the original’s quantization parameters, which, because of their special status, always apply to aliases too.

Reassigning an Alias

The function **Functions > Alias Reassign** assigns a new original object to an existing alias. Both the alias and the intended new original object must be selected first.



Search Functions for an Original or Alias

Finding the Original of an Alias

If you have forgotten where the original is for a certain alias, select the alias, and select the function **Functions > Alias > Find Original**, which will select the original object for you. Another method is described in section *Contents Visible in the Object on* page 3-81.



Finding the Alias of an Original

Conversely, you can select the original of a particular object, and find any aliases you've made from it. Select **Functions > Alias > Select All Aliases of Object**. All existing aliases will then be selected for you.

“Orphan” Aliases

If you erase an object from which one or more aliases have been created, Logic will warn you with the following message:

One or more Aliases are made from objects to be cleared! Do you still want to clear these?

“Don't Clear” cancels the erase procedure, while *“Clear”* deletes the object. If you do this, it makes no sense to keep the aliases after you've deleted the original, but because Logic does not do this for you, so you could end up with “orphan” aliases—that is, aliases without an original. Although such objects can serve no useful purpose, Logic doesn't automatically delete them, as you might decide to assign them new originals later.

Selecting All Orphan Aliases

Functions > Alias > Select All Orphan Aliases selects all aliases whose original sequences no longer exist.

Deleting Orphan Aliases

Functions > Alias > Delete All Orphan Aliases deletes all the aliases that no longer have originals. You can use this function to tidy up after deleting lots of objects, as this will possibly leave a load of old, unnecessary aliases in your song file.

Turning the Alias into a Real Object

You can use **Functions > Alias Turn to Real Copy** to make a real object from an alias; its contents will be identical to those of the original object of the alias.

Editing the Events in an Alias

It is not possible to edit aliases. If you double-click on an alias, Logic assumes that you either want to edit the original, or turn the alias into a real object. A dialog box appears asking whether you want to create and edit a real copy, or whether you want to edit the original instead.

Click on the desired button;  opens the editor window for the original object.

Merging Two or More Objects

If you edit an original object by merging it with others, any existing aliases are automatically updated to reflect the changes.

3.12 Markers

As their name implies, Markers are mainly used to mark particular time positions or sections in a song. They usually appear as short texts and (optionally) colored sections in the bar rulers of all Logic windows, and (optionally) as longer texts (of any length), in the Marker Text window. Therefore markers can also be used as notepads to store text comments with a song file.

In the bar ruler only the *names* of the markers are shown (if there is enough space). The complete text of a marker can be viewed and edited in the *Marker Text window*.

The names of all markers, together with their bar position and length are also listed in the *Marker List window*, which, as its name suggests, is a specialized event list window for markers. Markers have a specific duration or length, regardless of the extent of the actual text. A length of 1 tick means that the marker extends to the beginning of the next marker, or to the end of the song or folder, if there is no other marker following it.

Apart from the text storing functions, markers can also be regarded as namable storage areas for locator positions.

Creating Markers...

...using the mouse

To create a marker with the mouse, position the mouse cursor in the lower third of the bar ruler at the desired bar position, hold down the -keys and click. A Marker Text window will open (see below).

If a marker already exists at that bar position (or up to a quarter note before or after it), no new marker will be created. The same applies for all other methods of creating markers (via menus or key commands, see below).

...using menu or key commands

To create a marker at the current song position (rounded to the nearest bar), select **Options > Marker > Create**, or use the key command *Create Marker*. This does not open a Marker Text window, thereby making it easier to quickly create several markers, one after another.



Options > Marker > Create without rounding (or the key command *Create Marker w/o rounding*) does almost the same, but the position of the marker is not rounded to the nearest bar.

...above objects

Selecting **Options > Marker > Create by objects** creates markers at the position(s) of all currently selected objects (sequences or folders). These markers take on the names, bar position and length of the objects they are derived from.



The menu and key commands are also available directly in the Marker List window.

...matching a cycled section

To create a marker which corresponds exactly to the length and position of a cycled section, drag the cycle bar down into the lower third of the bar ruler. For more on this, see also section *Adjusting a Marker to Fit a Cycle* on page 3-71.

Deleting Markers

To delete a marker, grab it with the mouse in the bar ruler, and drag it down, out of the bar ruler. Let go when the mouse turns into a hand pointing downwards.



To delete a marker at the current song position, you can also choose **Options > Marker > Delete** or the key command *Delete Marker*. In the Marker List window, a marker can be deleted by selecting it and pressing *Backspace*.



Marker Names

Newly created markers are automatically named “Marker ##” (unless they were derived from sequences or folders as described above). “##” stands for their order of appearance along the time line, which will be displayed in the bar ruler instead of those characters (resulting in “Marker 1”, “Marker 2”, etc.). The number always depends on the actual order of *all* markers in the song, including those markers whose name has been changed.

Markers which were created using the *Create by objects* function will automatically be assigned the name of the sequences or folders they were derived from.

The marker names appear in the bar ruler of all Logic windows, the Marker List window, and the title bar of the Marker Text window. Automatically assigned names can be changed, either directly in the bar ruler, or in the Marker Text window:

Editing Marker Names and Text

...directly in the bar ruler

Double-click on the marker while holding down the *Command* & *Control* keys. A text entry field will open, where you can edit the marker's name. The same effect is achieved with **Options > Marker > Quick Edit Marker**.

...in the Marker Text window

The name of a marker can also be edited directly in the Marker Text window: The first paragraph in that window (up to a length of 60 characters including spaces) will be used as the name for that marker. However, how much of that name is displayed in bar rulers and the Marker List window, depends on the available space there.

The Marker Text Window

To open the Marker Text window, either double-click on a marker in the bar ruler while holding the  -keys, or select **Options > Markers > Open Text** or **. . . > Open Text As Float**.



Reminder: A floating window will always stay “on top” of regular windows.

However, it is not possible to perform any text input or editing in floating marker windows.

Marker Text Input

In this window, you can enter as much text as you like. The first paragraph (up to 60 characters) will also be displayed in the bar ruler as the name of the marker, the rest will only be visible in the Marker Text window itself.



In this window, the marker text can be input and edited as usual. The local **Edit** menu contains the usual *Cut*, *Copy*, *Paste*, *Clear* and *Select All* commands.



The Copy & Paste commands can also be used to import or export text to and from other software applications (such as word processors), to print the text.

The text line breaks are always automatically formatted according to the current size of the window.

If you click on the button with the open book (deactivating it), the window will become a Marker List window. Another click on that switch (highlighting it again), will bring back the Marker Text mode.

Text Appearance

It is possible to assign a different font, size and style to any selected part of the text, using the **Font**, **Size** and **Face** menus. All fonts which are installed in your system folder can be used here.

Also, different colors can be assigned to the entire text, or selected parts, and to the background (for details see below).

Disabling Text Input in the Marker Text Window

If you deactivate the “A” switch in the left part of that window, the existing marker text is protected and cannot be altered until the switch is activated again. This is useful to prevent unwanted text editing, which could happen if you perform key



Markers

commands while the Marker Text window is open. To be able to enter or edit any text, this switch has to be activated (highlighted).

Any "normal" computer keystroke (with or without ) will be interpreted as text input in this mode (if that window is the active window), even if a key command is defined for that key.

Automatic Text Selection

If a "*" is used as first character in the text, the complete marker text will always be selected when the Marker Text window is opened. That way the existing text can always be overwritten immediately.

Colors...

In the Marker Text Window, **Face > Object Colors...opens a color palette.**

...for marker text

You can assign colors to all currently selected characters in the Marker Text window by clicking on the desired color in the palette.

...for the Marker Text window and bar ruler background

-click one of the colors in the palette will assign that color to the background of the marker text, and also to the marker background in the bar ruler. (If dark colors are used, the marker text will automatically be displayed inverted there).

Markers without a Name

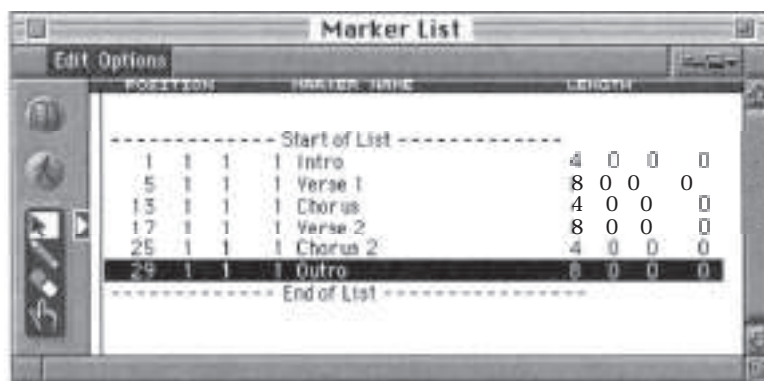
If you want the marker have no name, just type  instead of the first line, or   and **Return**.

Marker List Window

The Marker List window shows all existing markers in an event list. Here you can copy and delete markers, and edit their position and length. You can also select any marker for text editing.

Opening the Marker List window

The commands **Options > Marker > Open List** and . . . **Open List as Float** (also key commands) open the Marker List window.



Reselecting this command closes the Marker List window again.

The **Edit** menu contains the usual clipboard and selection commands. The *Select all following*, *Select inside Locators*, and *Toggle selection* are available as key commands.

Tools in the Marker List Window

The *Pointer* has the usual selection and editing functions. Selection: by clicking, clicking with the **Shift** key held down, and “rubber-banding” in the vicinity of the marker name. Editing: use the mouse as slider, direct value input by double-clicking. Double-clicking on the name converts the Marker list into a Marker Text window where the text of that particular marker is displayed.



The *Pen* duplicates a marker if you click on its name. You can then edit the position and length of the duplicated marker.



The *Eraser* deletes a marker by clicking on it.



The *Index Finger* sets the song position to the start of the marker, and sets the locator positions according to the marker's beginning and end. This tool can also be used for regular editing functions (like the pointer). The index finger is the default tool in the Marker List window when you open it.



Playing Markers

If you click with the index finger on a marker in the Marker List window, and keep the mouse button pressed, the sequencer will start playing at the beginning of that marker, and continue playing as long as you keep the mouse button pressed.

Buttons in Both Marker Windows

In Marker Text mode, you will see five buttons along the left side of the Marker window, in List mode only two of them:

The highlighted open book on the top switch indicates that you are in the Marker *Text* window. If this switch is not highlighted, you are looking at the Marker List window. With this switch, you can toggle between these two modes.

Clicking the walking man (*Catch*) ensures that you always see the text of the current marker displayed in the Marker Text window.

The two arrows (only available in Marker Text mode) enable you to jump to the previous or next marker, without having to leave the window. These are equivalent to the key commands *Goto previous Marker* and *Goto next Marker*.

The “A” Switch enables/disables text input in Marker Text mode (for details: see above).



Marker Position and Length

In the Marker List window, there are two display options for marker position and length:

Options > Position & Length in SMPTE units toggles the Marker position and length display between bar positions and SMPTE time positions.



Options > Length as absolute Position toggles the marker length display between actual (relative) length and (absolute) display of the end position of the marker (shown as bar position).

Moving Markers in the Bar Ruler

To change a marker's position directly in the bar ruler, just grab the marker in the bar ruler with the mouse while holding the -key, and drag it to the right or left. The smallest movements you can achieve this way are quarter-note steps, however, you can change the size of the steps by holding down additional keys:  to move in division steps (whatever is currently set in the Transport window), and   to move in ticks (however, this is still dependent on the horizontal zoom). Watch the Info Line for precise feedback on the movement:



Moving Markers in the Marker List Window

The marker position can also be changed in the Marker List window, using the mouse as a slider on the bar position display, or by double-clicking on a position value and entering the new value on the computer keyboard. The same principle applies to marker length.

Adjusting a Marker to Fit a Cycle

This works the same way as if you create a marker from a cycled section. If you drag a cycle onto an existing marker whose right or left end (or both) is inside the cycled section, the marker takes on the length and bar position of the cycle. To accomplish this, drag the cycle down until the cursor takes on the shape of the index finger (see opposite illustration).



Using Markers to Set Song Position and Locators

...in the bar ruler

If you click on a marker in the bar ruler while holding the -key, the current song position will be set to the start of this marker.

When you drag a marker up into the top third of the bar ruler, this will set a cycle with the same position and length as the marker (and thereby set the locator positions accordingly). If the sequencer is stopped when you do this, the song position line will be set to the beginning of the marker.



...with key commands

There are quite a few key commands which will set the song position line to the beginning of a particular marker, and adjust the locator positions to fit the marker's beginning and end. The first three commands in the following list will also activate the cycle mode:



Set Locators by Marker and Enable Cycle
Set Locators by Previous Marker and Enable Cycle
Set Locators by next Marker and Enable Cycle
Goto previous Marker
Goto next Marker
Goto Marker Number #
Goto Marker number 1-20 ('20 different commands)

The "Marker Numbers" in these commands refer to the serial order of *all* markers in the song.

...in the Marker windows

Clicking on one of the arrow buttons in the Marker Text window, or clicking on a marker with the Pointer tool in the Marker List window will also set the current song position to the beginning of the corresponding marker, and the locators to the beginning and end of that marker.

Locking Markers to SMPTE Positions

Options > SMPTE Position > Lock will lock (fix) the SMPTE position of markers. To indicate this status, a lock symbol is displayed at the beginning of the marker name in the list (the same as is used for locked events or sequences in the Event List window). These markers always keep their absolute time position: if the song tempo is changed, their bar positions will change to keep them at the same SMPTE positions.



Options > SMPTE Position > Unlock will reverse this status again.

Summary of Marker Actions Using the Mouse in the Bar Ruler

Without Modifier keys:

Grabbing a marker and pulling it upwards sets a cycle zone corresponding to the length of the marker.

Grabbing a cycle and pulling it down into the marker area on the bar ruler creates a new marker with corresponding posi-

tion and length, or, if the beginning of the cycled section is inside an existing marker, that marker's position and length will be adjusted to the cycle.

Grabbing the marker, dragging it below the bar ruler and letting go of it deletes the marker (Undo possible).

All other mouse actions without modifier keys in the marker area of the bar ruler have no effect on markers, but on the current song position instead, e.g. positioning (clicking), scrubbing (dragging), and Start/Stop (double-clicking).

With the  -keys pressed:

...creates a new marker and opens the Marker Text window.

With the -keys pressed:

A short click on an existing marker sets the SPL to the beginning of that marker.

Double-clicking on an existing marker starts the sequencer from the beginning of that marker.

Grabbing and pulling left/right moves the markers (also holding down  or   allows smaller increments for movement).

Grabbing and pulling upwards / downwards sets cycle lengths/erases markers (same as without modifier keys).

Double-clicking on a marker while holding the   keys

...opens a small text edit box for the marker name. However, if the marker already holds more than a few words of text, the Marker Text window will open instead.

Double-clicking while holding the    keys

creates a marker and opens the small name input field.

Double-clicking on a marker while holding the -keys

...opens the Marker Text window for that marker (without creating a new marker).

Marker Meta Events

Whenever Meta event #51 ("Goto Marker Number") is played, Logic jumps to the position of the marker whose number (in your song) matches the value (data byte) of that Meta event. You can generate this Meta event in one of two ways:

With an Environment fader object whose Out definition is set to "Meta", or

As a Goto marker Meta event (#51), which you insert into the event list by clicking on the "001 011" button while holding the -key.

Meta event Goto... instructions are ignored when external synchronization is in use.

For more on Meta events, read the section *Meta Events* on page 10-17. The section *Event Faders* on page 5-84 also tells you how faders can generate Meta events.

3.13 Arrange Window Techniques

If necessary you should reread the section *Señor Mouse* on page 1-2, the section *Selection Techniques* on page 1-23, and the section *Edit Operations* on page 1-27 in the Chapter 1: *Using Logic*. These sections and the techniques covered in the section *Sequences* on page 3-17 will teach you how to perform most arrange operations.

This section only describes functions which:

only have an effect on a particular part of the song, which you determine with the *Locator positions*, or cause a reaction between objects and *Locator positions*.

Adjusting Locators to Fit Selected Objects

selecting **Functions > Set Locators by Objects** sets the locators to that they correspond with one or more selected objects.



Set Rounded Locators by Object(s) rounds the locator positions to whole bars.



Adding or Removing Passages

The functions under **Functions > Cut/Insert Time >...** allow you to remove a passage from a complete arrangement. All bar changes, tempo changes and global score symbols are shifted accordingly. You can also insert a gap within your arrangement. As with cutting time, all tempo changes etc. are moved accordingly. You can also cut and paste a section of your song from one location to another.

You can change just a few selected objects as well. In this case, you have to tell Logic whether any bar changes, tempo changes and global score symbols should also be shifted. A dialog box appears, where you will be asked whether you also want to move Tempo Changes and Global Score Symbols.

Press  if you don't want to move the tempo changes and global score symbols too.

The above mentioned global editing is then done automatically, if *all objects* are selected. Otherwise, the functions are restricted to those tracks whose objects (at least those between the locator points) are selected.

Inserting a Gap across All Tracks

Functions > Cut/Insert Time > Insert Time and Move by locators



The selected objects are cut at the left locator. All objects to the right of the left locator are moved to the right by the cycle length. This creates a gap between the locators. Objects which began at the left locator now begin at the right locator.

Removing a Section

Functions > Cut/Insert Time > Snip: Cut Time and Move by locators. The selected objects are cut at the left and right locators. The objects created between the locators are deleted (i.e. copied to the clipboard and erased from the song, as with Cut) and all objects to the right of the right locator are moved to the left by the length of the cycle zone. All objects which began at the right locator now begin at the left locator, and the section between the locators is removed from the arrangement.



Inserting a Section

Functions > Cut/Insert Time > Splice: Insert snipped Part at Song Position. This function combines the existing functions “Insert Time and Move by Locators” and “Paste”: All selected objects at the song position are cut and a passage of a cycle length is inserted. The cut objects are then pasted back from the clipboard to the (unchanged) song position.



Practical Usages

“Snip”-removing an entire section of a song-e.g. a chorus.

“Splice”-inserting the part in another place in the song. To ensure all tracks are shifted, including tempo, time signature changes, and markers, please select everything beforehand (with  ).

Deleting Events Selectively

There are several ways of deleting events from selected objects, directly in the Arrange window. One method is to select the **Erase MIDI Events >...** command from the **Functions** menu.

Erase MIDI Events > Inside locators deletes all MIDI events from all selected objects between the locators.



Erase MIDI Events > Outside locators deletes all MIDI events from all selected objects outside the locators.



Copying Events Selectively



Functions > Copy MIDI Events... is used to copy (or move) MIDI events within selected sequences. You define the area that you want to move or copy with the left and right locators. The current song position marks the destination position where you want the data to be inserted.



Left Locator & Right Locator

These define the area containing the source data (also called the source region).

Destination

This allows you to define the destination position independently of the song position.

Number of Copies

Defines the number of copies.

Create New Sequence

When set to on, a new sequence is created at the destination position containing the copied (or moved) events.

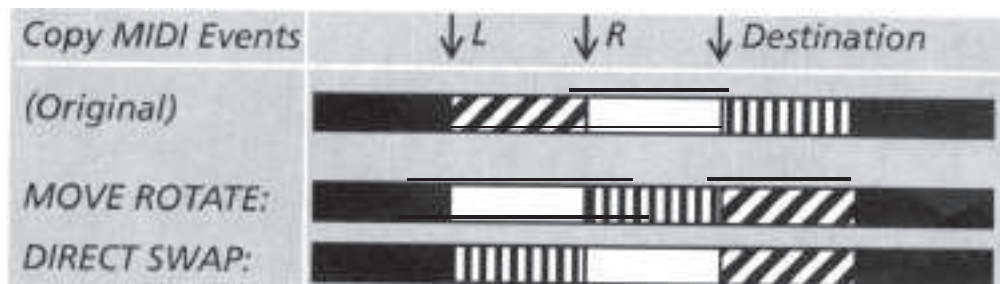
Mode

Defines the copy mode:

- Copy...* copies the selected section (the selected section remains in its original position).
- Move..* moves the selected section (the events are removed from their original position).
- Merge...* The source data is mixed with the data already present at the destination position.

- ...Replace:** The data already present at the destination position is deleted and replaced by the source data.
- ...Insert:** All data already present at the destination position is moved to the right by the length of the source region.
- ...Move Rotate** The source region is moved to the destination position. The data that was previously there and the data between the source region and the destination position are all moved towards the source. Thus the data is not altered or deleted, but rotated.

Select a sequence. Set the song position exactly to the start, the right locator exactly to the end of the sequence, and the left locator 1/4 note before the end of the sequence. Select **Functions > Copy MIDI Events...** and set the *Mode* to *Move Rotate*. Now, every time you press , all events are moved a 1/4 note backwards, and the events in the last quarter are moved to the front.



- Direct Swap:** The source region is exchanged with a region of the same length, starting at the target position.

3.14 Altering the Display

For information on the basic window functions, please refer to the section *Window Functions* on page 1-11.

If you want to know how to maximize the room available for the Arrange area, please refer to the section *Relationships between Windows* on page 1-18. The Arrange window also contains the following functions:

Display Options for the Track List

You can use the **View** menu to show or hide various parts of the track list:

Track Numbers/Level Meters

Selecting **View > Track Numbers/Level Meters** allows you to show or hide the track numbers and level meters.

Mute Switch

Every track has its own **Mute Switch**. If the mute-switches are hidden, you can mute any track to the left of the track number. A muted track is marked "*".

Record Audio Switch

Selecting **Record Audio Switch** allows you to show/hide the record audio switches (see section *“Arming” Tracks* on page 4-13).

Instrument Icon

Selecting **Instrument Icon** shows/hides the track instrument icons.

Shadow for Instrument Icon

Selecting **Shadow for instrument icon** shows/hides the shadows next to the instrument icons.

Instrument Name and Track Name

Hiding one of the names makes more horizontal space available. To distinguish them, the instrument name is always shown in bold type. One of the two names is always hidden.

Horizontal Layout



If both the instrument and track names are shown, the names are divided by a line. You can move this dividing line by grabbing the top of the marker with the mouse.

Instrument and Track Name stacked on each other



If you zoom in far enough vertically, you will be shown both names, one above the other: the instrument name (in bold) on top, then the track name below. This happens even if one of the names are hidden.

Sorting the Tracks

You can sort the tracks by grabbing the track number and moving it vertically. You can use the right edge of the name for sorting as well.

Here are the rest of the track list functions involving names:

Selecting a Track Instrument

You can assign an instrument to a track by click-holding the left half of the name or the icon.

Opening the Environment Window for an Instrument

You can open an Environment window with the relevant instrument highlighted, by double-clicking the instrument name.

Editing instrument Names

You can edit the instrument names by Alt-double-clicking directly in the track list. You can also use the Instrument Parameter box.

Editing Track Names

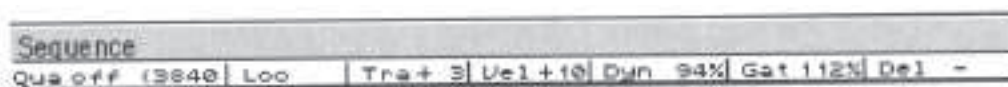
You can edit the track names by Command double-clicking directly into the track list-even if only the instrument name is shown there.

Object Display

Making Sequence Parameters Visible

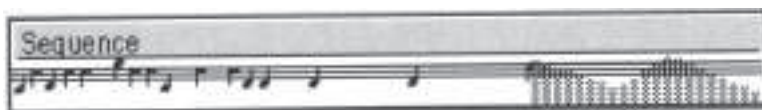


Enlarge the display using the vertical telescope. If the zoom setting is large enough, you will be able to see the sequence parameters displayed in the bottom half of all the objects. Here you can directly edit the sequence parameters with the mouse (by means of sliders and/or flip menus).



Contents Visible in the Object

If you vertically zoom the display further, you will be able to see the actual events contained within the objects.



Sequences show notes or controller events, folders show the objects they contain, and aliases show the name and position of their original.

The option **View > Object Content** allows you to see a display of the contents, at large enough zoom settings. If you uncheck this option, the contents will not be displayed, no matter what the zoom setting.



Object Colors

Newly recorded sequences initially adopt the color of the track instrument. However, after copying sequences between tracks, you could find the Arrange area will start to resemble a patchwork quilt. In this case, the function **> Instrument Colors To Objects** replaces the colors of all selected objects with the colors of the instruments playing the objects concerned.



For instructions on how to define normal object colors, please refer to the section *Arrange Window Technique* on page 3-74. The instrument colors may be set in the Environment.

The **Options > Settings > Display > Colored Background of unselected Sequences**. option allows you to have all sequences displayed in gray, with colored names; only selected sequences then adopt the chosen colors.

Altering the Background of the Arrange

Use the function **View > Show Grid** to switch on/off a track/bar number grid in the arrange area's background.

White Background toggles the background between grey/high resolution (depending on the setting of **Options > Settings... > Display > High Resolution Background**) and white.



3.15 Reset Functions

To switch off stuck notes, click on the MIDI monitor in the Transport window, or hit “STOP” twice in rapid succession. In both cases, reset messages are sent, as defined in **Options > Settings > Reset Messages**. There's more on this in the section *Reset Messages* on page 17-22.

For Hanging Notes-Panic Function

If the notes continue to sound, then your sound sources may not be able to respond to “All Notes Off” messages. If this happens, try the following:

Double-click on the MIDI Monitor in the Transport window (or use the key command *Send discrete Note Offs*). Separate Note Off messages will now be sent for every note, on all channels of every MIDI port. This should do the trick.

Unwanted Modulation-Controller Reset

Options > Send to MIDI > Reset Controllers transmits a control change message #121 with the value 0 (reset all controllers) on all MIDI channels and outputs used by defined instruments. This neutralizes all MIDI controllers like modulation or pitch wheels.



If Some Sounds Are Suddenly Too Quiet-Volume Reset

The function **Options > Send to MIDI > Maximum Volume** transmits a control change message #7 (main volume) with a value of 127 on all MIDI channels and outputs used by defined instruments. This sets all channels to their maximum volume, giving your sound sources the optimum signal/noise ratio.



If You Suddenly Hear the Wrong Sounds-Send Instrument Settings

The function **Options > Send to MIDI > Used Instrument MIDI Settings** sends all the Prg, Vol and Pan settings from all the instrument parameter boxes used by all the instruments in the current song. This could help to reset your sound sources for the current song, if your synths suddenly change patches mid-performance.



You can set this function to operate automatically after loading any song by turning on the Send *Used MIDI Instrument Settings*

after loading option, on the **Options > Settings > MIDI Options** page.

Using Audio in the Arrange Window

4.1 Regions in the Arrange Window

Creating Regions

Regions are similar to sequences as far as how they are used and edited. While sequences contain MIDI data, regions refer to audio files. When you record audio in real time, Logic not only creates an audio file on the hard disk, but also automatically creates regions, which represent the recorded audio file.

Regions can be seen both in the Arrange window (on tracks), and in the Audio window (the list of all audio files and regions used).

You can import existing audio files into a song by dragging them into the audio window. A region encompassing the entire file will be automatically created. You can then drag the region directly onto an audio track in the arrange window.



Immediately after importing audio files from other songs, it is recommended that you physically copy the files to the current song folder, using the Audio window function “Copy Files”. Answer “Change References?” with “Yes”, otherwise destructive edits of these audio files will also affect the original song.

To play back a region at a specified point in a song, you have to place the region in the Arrange window first (see the section *Adding a Region to the Arrangement* on page 7-19).

Drag regions only onto audio tracks. For more information about this, see the section *Creating Audio Tracks* on page 4-13.

Inserting Audio Files using the Pencil tool

Audio files can be inserted by shift-clicking on audio tracks with the Pencil tool at the desired position. A file selector appears, that lets you select the file you wish to insert. The length of new regions inserted in this way defaults to the complete file length, which can be freely edited later.

Dividing Regions

When you divide a region using the Scissors tool, you create two new regions. The newly created segments of the region will have the same name as the original, with a sequential number appended to the end of the name.

Scrubbing

You can scrub playback of a region to accurately select a division point, by using the Scissors tool and holding down the mouse button. This is only available with audio systems that support scrubbing. When using this feature, the Scissors tool works like a virtual playback head, which moves over the “virtual tape” (the region). If you release the mouse button, the region will be divided at the current position of the Scissors tool.

Resolution

When you are selecting a cut point with the Scissors tool, you can move it backwards and forwards in steps of one division. If you need a finer resolution, press   after you have selected the sequence with the Scissors. To obtain the maximum resolution (ticks) press , as well.

On Zero Crossings

If **Edit > Search Zero Crossings** is switched on in the Audio window, the division-point is moved to the nearest point where the waveform crosses the zero amplitude axis. When enabled, this also applies to all other methods of altering the start or end-point of a region in the Arrange window.



Remember that this can cause the precise start-point and length of an region to differ slightly from the selected value. In most cases, this will be inaudible

Erasing Regions

Regions can be erased either by selecting them and pressing or by clicking on them with the Eraser tool.

Erasing recordings

When you erase a region that you have just recorded (since opening the song), Logic will ask if you also want to erase the corresponding Audio File. This is a good way to avoid using up unnecessary space on the hard drive by storing bad takes and unwanted recordings.

If the recording was made before the song was loaded for the current session, this dialog box will not appear. This prevents you from accidentally deleting valuable recordings. Pressing will only remove the region from the Arrange area. If you want to delete the corresponding Audio Files from the hard disk, you can do so by selecting **Audio File > Delete File(s)** in the Audio window.

Copying Regions

Creating New Regions

Copies of regions are made exactly the same way as MIDI sequences-by dragging them while holding down the key. This automatically creates a new region in the Audio window. The new region will retain the same name as the old one, but with a sequential number added to the name.

This allows you to alter the start and end-points of the copied region independently of the original. It is comparable to a genuine copy of a MIDI sequence, which is created in the same way.

Using an Existing Region More than Once (Alias)

You can also create another version of the *same* region in the Arrange window—a cloned Region, if you will. To do this, hold down the  and  keys as you are copying. This is comparable to an Alias of a MIDI sequence, which is created in the same way.

Now whenever you adjust the start or end-points of any one of these regions, all the other regions which were cloned from the same region will be adjusted as well.

Making Cloned Regions Independent

To make several regions which are all cloned from the same region independent of each other again, select **Functions > Alias > Turn to real Copy** in the Arrange window. This converts all selected clones in the Arrange window into individual regions.

Converting several Regions into Audio Files

In the Arrange window, (not the main menu) you also have the option of saving all selected regions in the Arrange window as independent audio files: **Audio > Convert Regions to individual Audio Files**. The audio files are created in the same folder as your original file. The files are added to the Audio window as well, and the references of the regions in the Arrange window are changed to the newly created files.

Making Multiple Copies of Regions

You can make multiple copies of both audio regions and MIDI sequences with. **Functions > Object > Repeat Objects...** This function always creates clones of the same region (the regions are not independent). This function is described in more detail in the section *Making Multiple copies of Sequences* on page 3-19.



Another way to make regions repeat is to use the *Loop* parameter in the Region Parameter box (see the section *Loop* on page 4-9).

The command **Functions > Sequence Instrument Parameters > Turn Loops to Real Copies** will change loops of selected regions into audio regions. The arrange objects which are created, however, represent the same region. Thus any alterations in the length of the region or audio material effects all of the objects (as with cloned regions).



Moving Regions

You can grab regions in the Arrange window with the mouse, and move them around, just like MIDI sequences. Dragging will snap to increments of one beat (e.g. quarter notes-see below).

If you hold down  while moving sequences, you can move them in steps of one division (in sixteenths, for example).

If you hold down  and  while moving sequences, you can move them by single ticks (maximum resolution).

You can also use the Delay parameter in the Region parameter box to shift the playback position of regions (see the section *Delay* on page 4-9).

Fine-tuning

In practice, a resolution of “1 tick” will usually be fine enough.

To move a region by a finer resolution than ticks, i.e. by a resolution of 1 sample word, please use the Anchor in the Audio window or Sample Edit window.

Don’t forget that all cloned regions in the Arrange window will be affected as well.

Moving Regions in the Event List

You can also move regions by adjusting their positions in the Event List. Select the region you wish to edit, and select **Windows > Open Event List**, or press  . You can then move sequences by units of a single tick if you wish, by changing the value in the *Position* column.



Audio files in the Arrange window  can also be displayed alphanumerically in the Event List. Highly accurate position or length adjustments can be easily carried out, using the Event List.

...to SMPTE Positions

The position of regions can also be displayed in the Event List as SMPTE values. To do this, select **View > Position & length in SMPTE units** from the Event List menu.

If you're synchronizing music to picture, you can then type in the required SMPTE positions for particular events. A simpler and more convenient method in the Event List is the key command called Pickup Clock. This automatically places the Anchor of the selected region at the current song position (i.e. the point in the film you're locked up to when in Sync mode).



You can also display the end point of a region as a SMPTE time. To do this, select **View > Length as absolute Position** in the Event List.

Moving Sequences Back to Record Position

You can move any region selected in the Arrange window back to the time position at which it was originally recorded, by selecting **Functions > Region(s) to Original Record Position**.

This function only works if the corresponding audio file was recorded in the current song. Imported audio files which were brought in using Add Audio File... have no original record position.

This command is also available as a key command: *Set Region(s) to original Recording Position.*



Inserting at the record position

You can also copy (default: ) or cut () a region to the clipboard, and then use **Edit > Paste at Original Position** to insert it into the selected track, at the original record position.



Changing Start and End Points

You can shorten or lengthen any region by grabbing it by its lower right hand corner with the mouse and dragging it. You cannot make a region longer than its original audio file. This means that all the *other* regions derived from that region will be lengthened by the same amount (cloned regions-see page 4-4).

Of course, the same goes for any changes you make to the start point of a region. Such a change can be made by grabbing the lower left edge of the region with the mouse and dragging.

You can edit a region? start and endpoints much more accurately with the Sample Editor, which can be opened by double-clicking on the region. Simply slide the "S" and "E" markers to change the Start and

Endpoints of the region. While moving the start point, use the -key, to make sure that the Anchor and therefore the timing relative to the rest of the arrangement remains unchanged. Keep in mind that the Anchor must be inside the region.

If you wish to move the startpoint to the right, first move it to the approximate desired position while in the Arrange window. Then, open the sample editor, and use the -key to move the start point to the exact position.

Adjusting the Grid to Zero Crossings

If **Edit > Search Zero Crossings** is **switched** on in the Audio window, every time you alter the start or end point of a region in the Arrange window, the adjustment will snap to the nearest zero crossing of the waveform. The Anchor is not affected.

The disadvantage of this option is that after you have altered the start or end points they will never be exactly on the selected musical grid, but will always be a few ticks out. Normally this will not matter, because the Anchor reference point is not affected. However, if this causes problems in special situations, you can switch off the *Search Zero Crossings* option. The primary reason to have zero crossings enabled, is that it prevents clicks and pops at region start and end points.

Region Parameter Box

Name

Just as with MIDI sequences, the top line of the Sequence Parameter box displays, and allows you to edit, the name of the region. You can name several selected regions at once; the regions are then given the same name, with sequential numbers added to the end of the name. Logic inserts these numbers automatically, unless you deliberately leave a space at the end of the name.

When you change the name of a region, the corresponding region is also renamed in the Audio window.

Loop

As with MIDI sequences, you can use the loop parameter to set a region to loop automatically. The loop repeats until it reaches the next region on that track, the end of the song, or the end of the folder containing the region.

Don't forget that these loops are based on the sample grid of the region. This means that even if the song tempo matches precisely, after a while the loop repeats may start to go out of sync. You can avoid this by setting the end point of the region to an exact measure boundary. In these cases, it may be better to use the *Repeat Objects* function (section *Making Multiple Copies of Sequences* on page 3-21).

Delay

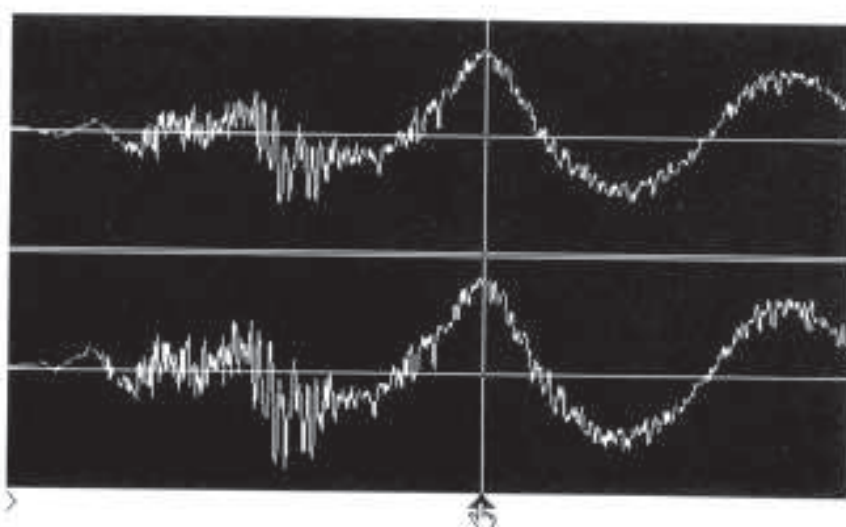
Just as with MIDI sequences, you can advance or delay the playback of regions, the smallest available units being ticks.

The Region Anchor

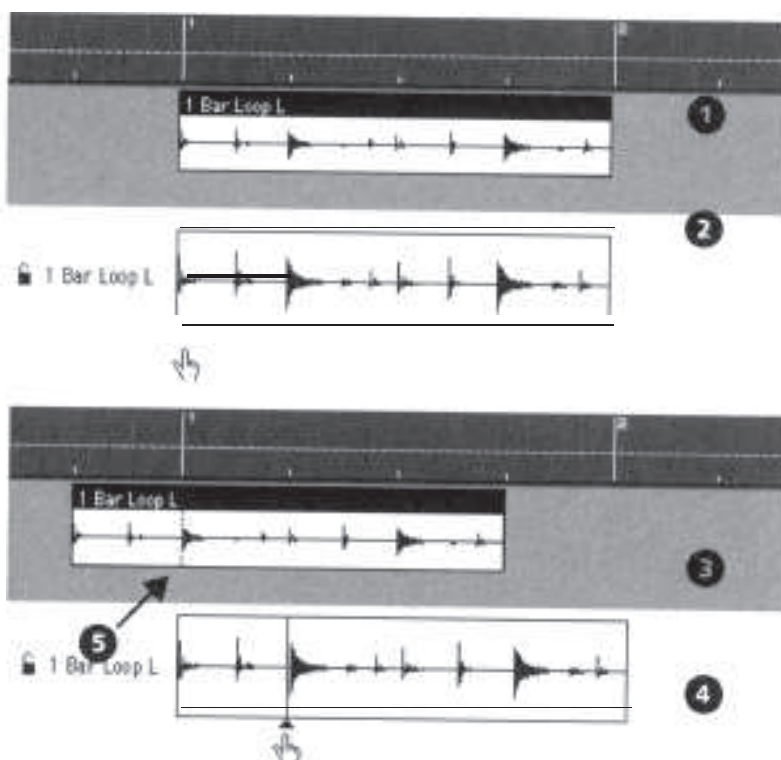
The Anchor is a region's temporal reference point. When you move a region, it's not the start point that is displayed in the Info Line (as with MIDI sequences)-it's the Anchor point.

For example, to guarantee perfect sync between a one-bar drum loop and your MIDI sequences, the Anchor must be assigned to a well-defined musical point. If the loop begins with a significant level peak (say a kick drum beat), set the Anchor to the point where the volume of that beat is at its precise peak.

Chapter 4
Using Audio in the Arrange Window



The following procedure can be useful in determining whether or not the Anchor is set to its optimum position: play a drum sequence in via MIDI that duplicates the main rhythmic accents of the drum loop. You should then be able to correct the Anchor position by ear. In order to check the anchor's position, you also can use the two key commands described in the section *Checking the Position of the Anchor*.



The changes made to the position of the Anchor in the Audio window (2 and 4) directly affect the region in the Arrange window (1 and 3). The position of the audio shifts relative to the sequencer's time axis, while the Anchor remains tied to the same bar value, and is marked by a dotted line 5.

MIDI Sequences and Regions Compared

The following table sums up the main differences between audio regions and MIDI sequences. There are some notes at the bottom of the table to clarify the most important points.

Characteristic:	MIDI sequences:	
	Regions:	
Composed of discrete data	Yes	No, because regions are just references to parts of audio files
Can be given names	Yes	Yes
Sequence parameter "Loop" available	Yes	Yes
Sequence parameter "Quantize" available	Yes	No, but the position of regions themselves can be quantized, using the Event List display on the Arrange level. You can use the Quantize Engine to quantize audio files destructively.
Sequence parameter "Transpose" available	Yes	No. You can use the Time Machine to transpose audio files destructively.
Sequence parameter "Velocity" available	Yes	No
Sequence parameter "Dynamics" available	Yes	No
Sequence parameter "Gate Time" available	Yes	No
Sequence parameter "Delay" available	Yes	Yes
Freely positionable	Yes	Yes

Chapter 4

Using Audio in the Arrange Window

Characteristic:	MIDI sequences:		Regions:
Left or Right Corner Edit	Yes	Yes	Yes, and this affects the audio region, but not the position of the audio waveforms relative to the time, axis of the sequencer
Can be cut with the Scissors tool	Yes	Yes	Yes, creates new regions
Can have alias objects' made from them	Yes	Yes	Yes, regions are already alias objects
Left corner may be dragged to conceal data at the beginning	No	Yes	
Have a variable musical reference point	No	Yes	Yes, a variable Anchor. This affects all the regions derived from one region and can change the position of the audio signals relative to the time axis of the sequencer
Can be turned off with the Mute function	Yes	Yes	
Can be grouped into folders	Yes	Yes	
Can be soloed	Yes	Yes	

Naturally, some of the parameters that can be applied to MIDI events have no effect on audio signals. Equivalent operations for most of these are available as destructive edit operations in the Sample Edit window.

Only regions have the flexible reference point provided by the Anchor.

The fundamental functions (such as the freely determinable position and length, the ability to create complex arrangements with the aid of folders, and the ability to name, mute, and solo) are generally available to all types of objects (MIDI sequences, regions, folders and Alias objects).

4.2 Audio Recording

For details on preparing to make an audio recording, please refer to the relevant sections in the Chapter 8 *Audio Driver* chapter.

Creating Audio Tracks

To create an audio track, you must first assign an audio object as a track instrument. Clicking on a track name in the Arrange window, while holding down the mouse button. This opens the Instrument list flip menu, where you may select an audio object. To help things along, audio objects “Audio 1” to “Audio X” are preset for the HDR hardware.

If you want to make adjustments to the audio objects, you can do it directly from the Arrange window. The Instrument Parameter box is located under the toolbox.

“Arming” Tracks

1. Method:

Click the switch with the "R" in the track list. To show/hide these switches, select **View > Record Audio Switch**.



If you have checked **View > Record Audio Switch** but the switch is still not visible, you should switch on the driver for the relevant audio hardware in the **Audio > Audio Hardware & Drivers** window.

2. Method:

Open the Environment Layer containing your audio objects, (for example from the main window by selecting **Audio > Audio Mixer**).

Click on the REC button of all the audio objects whose tracks you wish to record to.



Click the audio object once more (as shown above) to disarm the track again.

You can only make audio recordings on tracks that have been armed, no matter what track is highlighted in the Arrange window.

If you select a MIDI track, you'll record MIDI events.

If you select an audio track, you'll record on all the tracks that have been armed (record enabled).

Space is reserved on the hard disk for armed tracks, and is no longer available for Undo files. For this reason Logic will automatically disarm audio tracks if editing in the Sample Editor, if the hard disk is nearly full.

You can record simultaneously on one MIDI track and several audio tracks together, by clicking the MIDI track and the audio tracks you're going to record to, while holding down the  key.

If several Arrange window tracks are assigned to the same audio instrument (e.g. "Audio 1"), then the new audio file (or region) will be recorded to the selected track.

"Disarming" all tracks

If several tracks are record-enabled, you can instantly disable them all at once by clicking one of the flashing REC buttons, while holding down the *Alt* key.

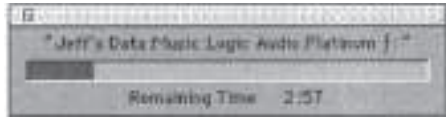
Recording Modes

Logic gives you many options for recording audio at various points in your song.

Standard Recording with Count-in

You can start recording at any point you like in a song, by setting the Song Position Line to the desired place. If the audio tracks you want to record to are correctly set, and the necessary input

signals are connected and properly adjusted, just hit the record switch on the Transport, or press .



A “count-in” will sound. Signals will be recorded during this time. Depending on the number of tracks required, Logic will pause

for a short interval between the time you press the record button, and the start of recording.

During recording, the Recording float window opens. This displays the remaining recording time.

It also shows you the destination folder of the current recording.

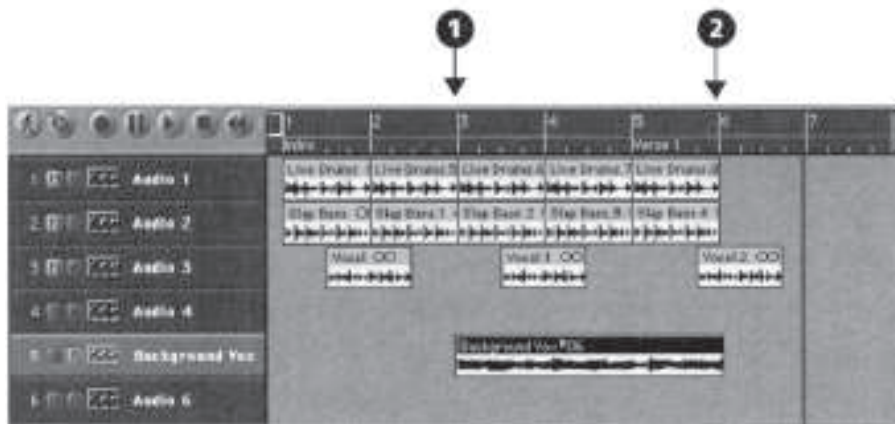
Logic will automatically create a region in the Arrange window after recording, and will create an overview of the audio file. This is a graphic file that is used to display the waveform in the Arrange window. You can listen back to the new recording in the Arrange window as the overview is created.

Punch-in Recording

You can actually engage recording mode in the midst of playback—“on the fly”. To do this, start playback and press   (preset key command for **Record Toggle**) at the point where you want to start recording. Audio recording will start immediately. Recording can be stopped at any time, by pressing STOP in the usual way, or by pressing **Shift *** again. If you use this second option, recording ceases, but the sequencer will continue to play.

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A recording made during playback, by hitting  twice (at the points shown by ① and ②).

Punch on the Fly

If you click-hold the Record button in the Transport window the Record menu opens, where you can switch on “Punch on the Fly”. This can also be switched on from the **Audio** menu.

Punch or drop recording is when you "punch in" to a previously recorded passage while the tape is running, in order to fix a mistake in an otherwise good recording, and then "punch out" of the recording again, once the mistake has been corrected.

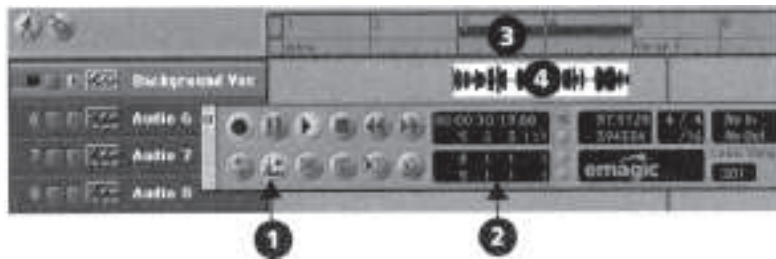
On hard disk recorders this function is technically difficult, due to the unavoidable delay in switching between reading and writing (creating a file etc....). In practice, recording takes place from the moment you start. Afterwards, Logic then adds the result of this automatic background recording to the exact place where you wanted the recording to occur.

A free track therefore has to be available for this purpose. A track is “free” if there are no regions (or only muted regions) on it within the area that playback is started and ended.

If you attempt a “Punch on the Fly” recording and Logic produces an error message, just temporarily mute some non-critical audio tracks. After the recording is complete, you can unmute them.

Pre-programmed Drop-Record

Logic's autodrop function can be used on audio tracks, just as it can with MIDI sequences. Autodrop is enabled by clicking on the autodrop switch on the Transport window. Here is a step by step breakdown of the procedure:



- ❶ The autodrop switch must be turned on.
- ❷ You can set the autodrop locators numerically in the Transport window.
- ❸ The autodrop recording zone is marked by a thick black bar in the middle third of the bar ruler.
- ❹ Recording only takes place within the autodrop zone you have set up, with the exception of a short “lead-in” just before recording is due to start (see below).

The autodrop zone start and end points can be set either from the Bar Ruler, or by using the locators in the Transport bar. Please note that if the Cycle function is switched on, you can set the Autodrop Locators in the window to the right of the Cycle Locators (under tempo).

Start the recording. Logic will begin recording about one bar before the drop-in locator. Afterwards, a region will be created that corresponds exactly to the length of the autodrop zone. This method allows the start of the region to be adjusted later, so that the lead-in becomes audible. This way, if the performance anticipates the autodrop zone slightly, the beginning can be easily recovered. Doing this does not change the position of the recording relative to the time axis.



A tip: if you wish, you can even define a small autodrop zone within a larger Cycle Zone (see below). This can be handy if a player needs to re-record a difficult passage in the middle of a song, and requires many attempts to get it right.

Record and Pause Mode

If Logic is in pause mode, you can start audio recording by clicking “Record Toggle”, “Pause” or “Play”. Recording then begins at the current song position.

Audio Cycle Recording

You can make audio recordings even when ‘Cycle’ is switched on. A new track is created for every cycle. All the tracks created are still played by the same audio object. This prevents the tracks from being played simultaneously.



If you stop recording just after the end of a full cycle, no region is created for the cycle which has just begun. However, the audio material is not lost, but is recorded after the last region in the audio file.

The whole recording (during all cycles) is stored as a single audio file. This audio file is split into regions, which are as long as the length of the cycle. The region for the previous cycle is automatically muted at the beginning of the next cycle.

Audio cycle recording also works when you are simultaneously recording two channels (e.g. stereo tracks in split stereo mode).

After recording, please note...

After you’ve made your first audio recording in a song, you should not make any further adjustments to the tempo of that song. Decide on a tempo, and any changes in tempo well before the first audio recording, and stick to it. Audio recordings have a fixed playback rate, and can only be matched to new tempos if you’re prepared to go through a great deal of hassle. The time compression/expansion algorithms currently available only permit you to match audio to new tempos when the tempo differences involved are relatively small; if you try and use

them to significantly change the tempo of your audio recording, its quality may be significantly compromised.

If you've made your audio recordings in real time, playing them in over the top of say, an existing MIDI arrangement, it is not particularly recommended that you move the Anchor of any of your regions. You can make slight adjustments to the timing of your regions using the "Delay" object parameter.

Auto Input Monitoring

Input monitoring allows you to determine what signal you listen to on record-enabled tracks. If Auto Input Monitoring is switched on, you only hear the input signal during the actual recording-before and afterwards you hear the previously recorded audio on the track, while the sequencer is running. This helps you judge drop-in and drop-out points for punch recording. If Auto Input Monitoring is switched off, you will always hear the input signal. With Auto Input Monitoring switched on, you will also hear the input source when the sequencer is stopped. This allows you to set levels easily.

There is no Monitoring at all if the parameter *Monitoring* is not checked under **Audio > Audio Hardware & Drivers** for the driver you use.

Since input monitoring always involves a little latency, in certain configurations it is recommended that you not use Logic Monitoring at all, to get the best monitor timing possible. Route the signal to be recorded directly to the headphones or monitoring amplifier. You do not risk unknowingly overdriving your A/D converters, since the input level meters will still work, and Logic Audio Platinum warns you in a dialog as soon as overload occurs. The latency depends on the hardware, and the driver settings. Please also refer to the section Monitoring on page 8-7.

Switching Auto Input Monitoring on/off

If you click-hold the Record button (Transport window) the Record menu opens. This is where you can toggle Auto Input monitoring. You can also use the key command *Toggle Auto Input Monitoring*, or enable it from the **Audio** menu.



The table below shows what you will hear at the output of record-enabled tracks:

	-----Playback-----	Record
	Logic stopped	Logic running
Audio Input Monitoring switched on	Input signal	Track signal
Auto Input Monitoring switched off	-----Input Signal-----	Input signal

Possible Problems with Input Monitoring

Typical problems that can arise with record-enabled tracks:

Singers cannot hear themselves while Logic is running. All they can hear is the old recording.

Switch off Auto Input Monitoring.

The track cannot be heard, even though Logic is not yet recording.

Switch on Auto Input Monitoring.

Depending on the situation, both options can be useful. Normally Auto Input Monitoring is switched on, and is only switched off in special situations.

Whenever you are doing punch recording (e.g. Punch on the Fly), you should switch on Auto Input Monitoring.

Stereo Recordings

For information on how to create stereo recordings, please read the appropriate sections in the chapter Chapter 6 *Mixers, Effects*

and *Audio Objects*, such as the section *Stereo Objects and Channel Strips* on page 6-73 and the section *Faders and Level Adjustment* on page 6-66.

Stereo recordings will normally be saved as Interleaved Stereo files. The names of regions made from Interleaved Stereo files appearing in the Arrange window will be appended with a stereo symbol consisting of two overlapping circles (left example below).



Split Stereo Region names will be appended with a stereo symbol consisting of two adjoining circles (right example above).

You can freely mix both formats, even on the same track. However, if your audio hardware supports the Interleaved Stereo format (e.g. Audiowerk8 AV/MME, ASIO, and others), we recommend that you use it.

While Split Stereo files use approximately 200% hard disk I/O performance compared to Mono files, Interleaved Stereo files need only 150%.

Under **Audio > Audio Preferences**, you can check *Record & Convert into Split Audio Files* (force Logic to create Split Stereo files), for example, to be compatible with ProTools systems.

4.3 Functions

Automatic Tempo Matching

Among all its other amazing features, Logic offers an automatic function for matching the length of a passage of free-form music with the length of an audio region. The length of the

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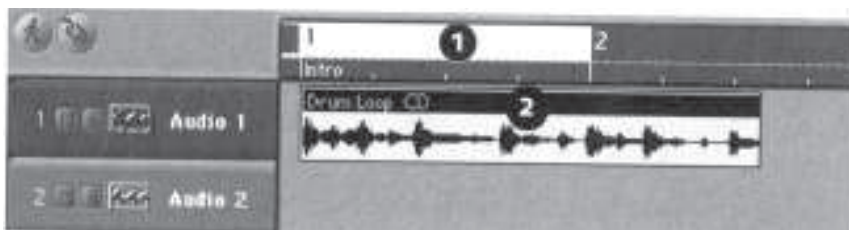
region remains constant here, but the sequencer tempo is varied automatically, so that the region and the musical passage end up exactly the same length.

To show how to make use of this function, we can take a one-bar drum loop as an example:

Imagine you've recorded a drum loop live, or from a sampling CD. Using the Audio window and the sample editor, you've adjusted the start and end points of the audio region so that the loop cycles perfectly, without any glitches.

Now you drag the drum loop into the Arrange window, where it appears as a region. Make sure that you place the beginning of the region at the start of a bar.

Construct a cycle in the Arrange window bar ruler whose length matches the intended musical length of the region. The drum loop is one bar long, so you should pick a cycle length of one bar, to match.

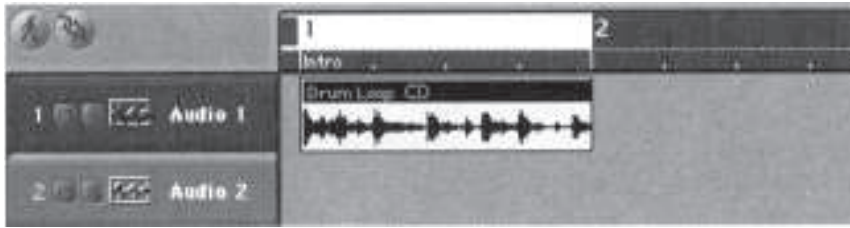


In the bar ruler you set the cycle ① to the desired musical length. The current length of the region is shown ② this will be different than the cycle until you use the "Adjust Tempo..." function...

select **Options > Tempo > Adjust Tempo using Object Length & Locators.**



The tempo is reset so that the region is now exactly one bar long, and fits the length of the cycle. This does not change the tempo of the audio, but rather makes the song tempo conform to the audio.



Audio and MIDI Scrubbing

You can scrub audio and MIDI in the Arrange window simultaneously; but audio scrubbing is currently only available with DAE hardware.

Please note that due to restrictions in DAE, scrubbing is only possible with one stereo, or two mono tracks. Future versions of DAE may not be subject to these restrictions.

If your Song contains regions running on more than two neighboring tracks (e.g. on 1-2-3 & 4), select just (for instance) the ones on tracks 2 and 4, so that you can scrub them.

Before scrubbing can be carried out, you have to enable **Options > Settings > MIDI Options > Scrubbing with Audio** in Arrange.

Let's get Scrubbing!

Hit Pause on the Transport window.

Grab the Song Position Line with the mouse and move it back and forth through the parts of the Song you want to hear played back. Note how audio and MIDI data are played back in perfect sync when scrubbing.

You can scrub at any zoom resolution-so set up a magnification that you're comfortable with, for the section of the song you're scrubbing through.

If your HDR hardware permits, scrubbing can also be activated when you are dividing regions with the scissors in the Arrange window.



Digital Mixdown

Using Logic, you can mix down audio data in the digital domain from within the Arrange window. This is done with the glue tool. This function is non-destructive, as Logic always creates a new file for the mixed-down material. This new audio file is stored on your hard drive.

To carry out a mixdown select the Glue tool from the Toolbox and use it to select the required regions for mixdown (if necessary using the shift key as well).



The mixdown process can be aborted with (command-period). All audio files will remain unchanged.

No Mixdown Situation

If there are several regions (mono or stereo) in a row on the Same track, which have been cut out of one region using the scissors tool, i.e. which originate in this order from the same audio file, no mixdown is carried out. Instead, a single region is simply created over the entire area. This gives the desired result, without using up any additional disk space.



Logic can recognize associated regions even if there are gaps between the regions. The determining factor is that the relative position of the regions in the Arrange window corresponds to the relative position of the regions in the audio file.

No mixdown is carried out, if you try to mix two regions whose tracks are panned to opposite sides in the Arrange window.



The two resulting mixdown files would be identical to the original audio files in the used areas of the regions. If you don't need the unused areas of the audio files, select "Optimize Files".

"Mixdown" of Regions in a Track

If several regions that overlap each other are selected from one track, the program does not mix them down. Instead, you are asked to create a new audio file, which is named after the first

region. The selected regions are then mixed together without my changes to volume, and without clip scanning.

If there are empty sections between two regions, these are added into the new audio file as silent passages.

Genuine Mixdown with Clipscan

If you combine audio data from two or more tracks, the current values for pan and volume found in the Environment for the individual tracks will define the pan and volume settings in the new audio files. If you combine both sides of a stereo audio file, first set the pan controls of the component mono sides to hard left and hard right, respectively.

You can carry out a mixdown while the sequencer is playing.

After the digital clipping scan (Clipscan) is over, and the mixdown is complete, Logic replaces the previously selected regions with one region which contains the new, mixed-down audio file in its entirety. If you wish, you can use the undo function to restore the original audio regions. If you do this, you will then be asked if you wish to keep the newly created mixed audio file, or delete it. If you decide to keep it, it will remain in the audio window, and can be further processed there.



During a mixdown the Clipscan function uses 32-bit resolution to ensure that the highest possible level is maintained, without clipping.

Audio Crossfades

The Digital Mixdown function (Glue tool, menu or key command) in the Arrange window supports audio crossfades of selected regions.



The parameters are defined in the *Audio Crossfades* dialog window, which can be opened either via a key command or by selecting: **Audio > Default Audio Crossfade Options**.



The key command for opening the dialog window can also be used to close it.

The parameters apply globally, and are stored in the Preferences file.

There are two parameters:

Time [ms]: This is the length of the entire crossfade. To switch off the crossfade, set this value to zero.

Curve: To obtain a linear crossfade, set this value to zero. Other values (positive or negative), produce various exponential fades. The fade-outs and fade-ins are always symmetrical, to avoid deviations in level.

The preset values are: *Time* = 20 ms, *Curve* = 0 (linear).

The graph shows the actual shape of the crossfade; the original algorithms are also used to calculate the curve display. Values over about 2.50 ms are scaled in the graph, to ensure that the entire crossfade is visible.

HyperDraw for Regions



HyperDraw™ can also be used directly on audio regions, and can be used for example, to draw changes in volume and panning.

Select **Options > HyperDraw > ...** or use the HyperDraw key commands. If you are not yet familiar with HyperDraw, please refer to the "HyperDraw" section in "The Arrange Window" chapter of the Logic manual.

If you want to use HyperDraw on stereo regions, you should always edit the left (top) region. Logic automatically uses this information for both sides.

When you use HyperDraw on audio regions, MIDI events are created, which are always organized together with the audio region. You can also edit these MIDI events in the Event List, or Hyper Editor.

Non-destructive Fades



Fade In, Fade Out and Crossfades can be done with the Fade tool (in the tool box), or with the related parameters. The parameters for *Fade In*, *Fade Out*/*X* times and their curves can be found in the region's parameter box. You can adjust these parameters in all selected regions simultaneously.



These fade parameters are non-destructive (will not change the original audio file), as opposed to the fade functions in the Sample Editor, or with the Glue tool.

Technical Discussion

A background fade file exists for each song, that contains all the fade areas. You will not normally need to access this file. This file is located in the same folder as the original region, and named after the song, with the extension "-f16m" for 16 bit recordings, or "f-24m" with 24 bit recordings".



This background file is created the moment you start playback, after setting the first crossfade for the song. If you are using many fades on multiple regions, this could take several seconds. During the creation of the fade file, you can see this window.

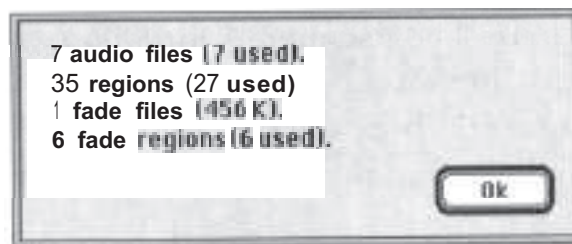
This method of creating fades is non-destructive, because you can change the fade parameters at any time, without changing

the original files. The background file will be updated and optimized automatically, whenever you save the song.

The fade file will also be updated after destructive changes applied to the original regions in the Sample Editor.

Information about Regions and their Fade file

With **File > Delete Fade File(s)** in the Audio window, you can delete the fade file. No information will be lost by doing this, because Logic automatically creates a new fade file the next time playback is started, based on the regions' fade parameters. With **Edit > Info**, Logic displays information about the number and sizes of fade files.



If the song uses audio files from a number of different folders or drives, Logic creates a corresponding number of fade files at these locations.

Enter Fades using Mouse and Fade Tool

As you'd perhaps expect, a fade *in* starts at the beginning, and a *fade out* occurs at the end of a region. The length of the fades can be adjusted in the Region Parameter box, but it's simpler to use the Fade tool to select the section inside the region where the fade is to take place.



 The fades are only visible if you are zoomed in closely enough to see the waveform in the region.

While using the Fade tool in conjunction with , you can edit existing fade times. In this instance the curve will not be affected.

You can delete a fade by using the Fade tool, in conjunction with .

Fade curve shapes (Curve)

You can adjust the curve shape by using the Fade tool in conjunction with **ctrl**.

You can also use the Curve parameter in the Region Parameter box.

In the example below, Curve has a positive value for both *fade in* and *fade out*.



In the example below, Curve has a negative value for *fade in* and a zero value for *fade out*.



Crossfades

If you select Fade X instead of *Fade Out*, the selected region will be crossfaded with the following region. The parameter Fade In and corresponding Curve are then irrelevant.

Fade **X**

Fade **Out**

Crossfading with the Fade tool



Use the Fade tool to mark the crossfade area of two adjacent regions. This works, even if the two sections are not, as in the example below, adjoining.

Chapter 5

The Environment

The Situation

This chapter will help you to understand the basic concept of the Environment, and make full use of its options.

Your setup probably consists of a number of different MIDI devices. The purpose of the Environment is to help you manage your setup flexibly and efficiently, down to the smallest detail (e.g. the pre-delay of a specific sound parameter in a synthesizer).

Logic's Environment gives you a number of objects for sophisticated MIDI processing. These include: arpeggiators, MIDI delays, transformers, note mappers, chord memorizers, and even Touch Tracks™ for playing entire sequences at the press of a single key. Of course, you need use only the parts that you really want, and forget the rest.

The Idea

The Environment refers to Logic's "virtual" environment inside your computer. It was developed to allow you complete control over your MIDI setup from within Logic.

Imagine that the Environment window gives you a virtual view of your MIDI studio, including port objects which represent the MIDI inputs and outputs of your MIDI interface, instruments which are virtual representations of each device in your MIDI setup, and various objects for modifying the MIDI signal flow.

In order for this to work properly, you have to connect the different objects in the Environment via virtual MIDI cables. This allows you to control the overall MIDI signal flow.

Chapter 5

The Environment

In the Environment, you can also create virtual faders which generate MIDI events when you move them. The movements of these fader objects can be remote-controlled by other MIDI events, and recorded by the sequencer. In addition, there are specialized objects which can split a MIDI signal into different channels, make pre-programmed alterations, or even re-route the signal path.

Layers

As you can probably imagine, the Environment can quickly fill up with a large number of objects. To keep things organized, you can assign the objects to different display levels, which are referred to as “layers”. Think of these layers as being different, partial views of the overall Environment. Naturally, you can easily connect objects between different layers.

The Concept of the Environment

Flexibility

The Environment allows you to control your MIDI studio and Logic’s MIDI signal flow. Virtual cables give you complete control over the MIDI signal path, from the input ports to the output ports. This gives you a high degree of flexibility when connecting Logic’s individual “effect modules”.

Adaptability

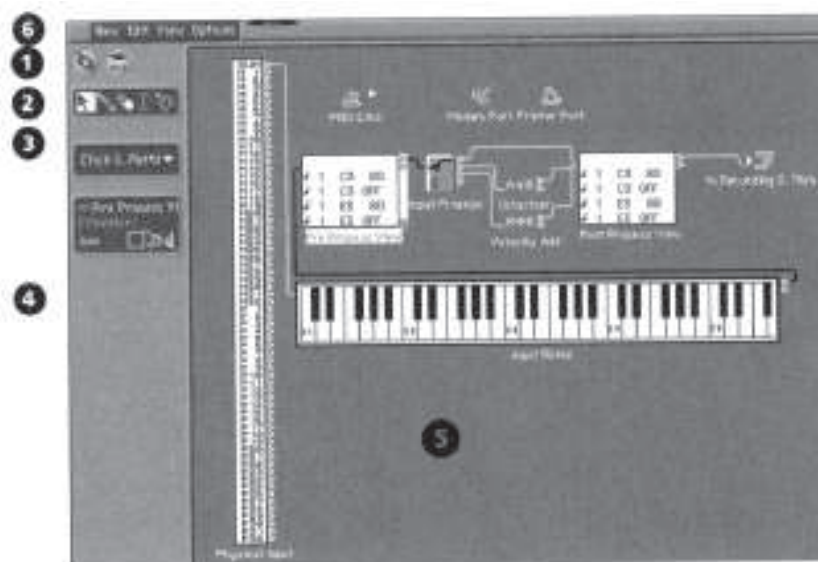
Logic addresses the MIDI ports of your interface directly via virtual instruments which you create in the Environment. These virtual instruments act like junctions: events can easily be addressed to them from the sequencer and then sent from them to a MIDI or internal (software) sound generator. Instruments include parameters which you can use to select sounds, transpositions, keyboard splits, volume, pan, MIDI delay, etc. Instruments can therefore also be thought of as virtual copies of individual sounds in your sound generator. In addition, Mapped instruments provide note assignment tables (drum maps) which can be used to adapt your preferred playing and

programming habits to the different key assignments of your sound sources. Finally, you can create virtual faders and knobs or controlling effect devices or automated mixing desks. The point is, the Environment allows you to adapt Logic to manage virtually all aspects of your MIDI setup.

5.1 Environment Overview

Opening the Environment Window

To open an Environment window, select **Windows > Open Environment** (default key command:   ). Double-clicking on an instrument in the track list of the Arrange window also opens an Environment window, to the layer containing that track instrument. When the Environment window opens, the track Instrument you double-clicked will be highlighted:



- ① Buttons
- ② Tool box
- ③ Layer box
- ④ Object parameter box
- ⑤ Layer contents (the default “Click & Ports” layer)
- ⑥ Local menus

Layers

Layers are “display levels” of the Environment window. They allow you to configure the Environment’s display so that you can view and work with specific groups of related objects (e.g. audio instruments), rather than having to deal with all Environment objects at the same time. The name of the current layer is displayed in the layer box.

The distribution of objects on different layers has no affect on their function—it’s just a system of organization.

There are some exceptions: objects in the “global objects” layer also appear on all other layers; and the “All Objects” layer shows all the Environment objects in a list. However, you can ignore both of these layers at first. They are not needed very often, even in very complex Environments.

Switching Layers

Click-holding the layer box opens the layer flip menu, which you use to switch layers.

Select **Options > Go to previous layer** to switch back to the last active layer. This allows you to toggle between two layers.

Remember you can assign key commands to this and many other Environment menu functions for faster navigation and greater ease of use.

Creating Layers

Selecting “**Create!**” from the layer flip menu creates a new, empty layer titled “(unnamed)” at the end of the list. You can add a new layer above the current position in the list by selecting **Options > Layer > Insert**.

Naming Layers

Double-clicking on the layer box opens the text field for entering a name for the current layer.

Deleting Layers

You can remove the current layer from the list by selecting **Options > Layer > Delete**. Because all the objects contained in the layer are deleted too, an alert box appears, to warn against deleting objects unwittingly. You must click “Delete” to complete the operation. “Cancel” or *Escape* cancels the operation.

Specialized Layers

The position and existence of the first two layers are protected:

All Objects

The top layer in the flip menu always displays all the objects in the Environment. The objects in this layer are normally shown in the form of a list. (You can change this in the Environment’s **View** menu, but doing so is not recommended.)

If you turn off the **File > Preferences > Display > Allow All Objects' Layer in Environment**, the “All Objects” layer will be removed from the layer flip menu.

The function **Options > Goto Layer of Object** allows you to switch to the layer of a selected object.



Global Objects

In this second layer from the top, you can place objects which you want to be visible in all layers (e.g. the output ports). These will subsequently appear at the same position, in all layers. However, you should place as few objects as possible on this layer, due to the on-screen clutter that global objects can create.

You may want to temporarily move some objects to the Global Objects layer while you are building multi-layer setups then move them to one of the other layers when you are finished.

Hiding/showing parameter fields

As in other windows, you can hide the column with the buttons, toolbox, layer menu and object parameter box by selecting **View > Parameters**. This will create more on-screen room for the Environment workspace display.



Hiding/showing the cables

By choosing View > **Cables** (*Hide/Show Cables*), you can toggle the view of the cables. This also hides/shows the positioning bar to the right of the individual objects (used for moving and sizing some Environment objects).



Protecting cabling and object positions

You can prevent the position, size and cable connections of all objects from being accidentally altered, by selecting **View > Protect Cabling/Positions**.



Background

If the cabling and object positions are protected, and the cables are hidden, the background color changes. This usually looks better for virtual mixing desks and fader setups.

You can also use Ornaments and Macros to clean-up and improve an Environment control panel's appearance.

List Display

Selecting **View > by Text** switches the graphic display of the layers to a list display. The cables are not shown in the list display. This type of display is most useful in the “All Objects” layer (see section *Spezialized Layers* on page 5-5), and for importing Environments from other songs (see section *Environment Exchange* on page 5-92).

Objects



Creating Objects

When you select an object type from the New menu, an object of this type is created in the current layer (for more details see section *Object Reference* on page 5-24). Clicking with the pencil on the background creates a new “Standard Instrument”.

You can copy existing objects by dragging an object while holding down *Alt* (see also the section *Moving Objects* on page 5-8). Cabling is preserved; so groups of objects copied in this way will be connected in the same way as the originals. (When you copy a single object, only its output cables are preserved.)

Deleting Objects

Click with the eraser, or press ***Backspace*** to delete all selected objects.

Adjusting the Size of Objects

You can adjust the size of fader, keyboard and monitor objects by grabbing and dragging the bottom right corner (just as with windows). Changing the size with multiple objects selected (by “rubber-banding” or clicking while holding down ) will alter



each object's size while preserving their sizes relative to each other.

Selecting **Options > Clean up > Size by Default** (Reset Size), sets the size each selected object to its default value.



The key commands *Object Width -1 Pixel* or *...+1 Pixel* and *Object Height -1 Pixel* or *...+1 Pixel* decrease or increase the width or height of all selected objects by 1 pixel.



Moving Objects

You can move objects by grabbing their icon or name, and dragging them to the desired position. The surface of keyboard and fader objects is used for their operation. You must therefore either grab the name (if available), or the positioning bar to the right of the object.



If you hold down , you can also grab keyboard or fader objects by their surface. Remember to first deselect any other selected objects by clicking in the background, to avoid moving them as well.

Grid

Select **View > Snap Positions** to align the objects to an invisible grid. It's a good idea to leave snap positions switched on. You only need to switch it off if you want to manually move an object by a few pixels.

You can use the key commands *Object move left/right/up/down* to move all selected objects one pixel in the relevant direction, even when the grid is switched on.



Aligning Objects . . .

...in a row or column

To align several selected objects horizontally or vertically, select the function **Options > Clean up > Align Objects**. The top left object stays where it is. The position of the next object



determines whether the objects are aligned in a column or a row. If it is to the right of the top left object, all the objects are aligned horizontally. If it is below the top left object, all the objects are aligned vertically.

...to grid positions

If you choose **Options > Clean up > Positions by Grid**, all selected objects are aligned to the invisible grid.



Moving Objects between Layers

...with the layer selection box

To alter the layer assignment of a group of Environment objects, first select the objects, and then select the desired layer, while holding down **Alt**. This moves the selected objects to the layer you've chosen.

...with a second Environment window

Another way of moving objects to a different layer is to open a second Environment window. Select the target layer and drag the relevant objects from one window to the other. You can also use this method to copy objects between layers (by holding



...via the clipboard

Another option is to move objects via the clipboard (**Edit** menu or the usual key commands):

Select the objects that you want to move and select **Edit > Cut** (default key command .

Switch to the destination layer.

Make sure that no objects are selected (by clicking in the background).

Select **Edit > Paste** (default key command .

Replacing Objects

If any objects are already selected when you go to add objects to the current layer using **Edit > Paste** a dialog box appears asking “Replace current selection? No/Replace”. If you press  or click Replace, the selected objects will be replaced by the objects in the clipboard. The existing cabling remains intact.

Assigning an Object to the Selected Track

You can assign any Environment object to the selected track in the Arrange window simply by clicking the object with the MIDI Thru tool.



You can also assign Environment objects to the selected track using the Arrange window’s Instrument List. Simply click and hold on the track name until the list appears, then scroll to the desired Environment object (see section *Icon/Display Filter for the Instrument List* on page 5-23).

Remember: the MIDI Thru function of the sequencer diverts all incoming MIDI events to the Environment object assigned to the selected track.

Special Selection Commands

Selecting All Used Instruments

With **Edit > Select Used Instruments**, you can select all objects which are assigned to tracks in the Arrange window or are connected to such objects by any series of cables.



Selecting All Unused Instruments

The **Edit > Select Unused Instruments** function allows you to select all the objects that are neither being used in the Arrange window, nor are connected to any such objects by a series of cables.



Selecting Cable Destination Objects

Edit > Select Cable Destination selects the destination object of a selected cable connection. This is particularly useful in two cases:



The destination object is in a different layer. You can use the function to select and display the destination object in its layer;

From the list display **View > by Text)** you can locate the destination object of a source object, because selecting the source object also selects its cable connections.

This function allows you to follow the cabling from one serially-cabled object to the next. If several objects are connected, or several cable connections are selected, the path of the top cable is followed.

Toggling your selection

Just as in the Arrange and editor windows, you can change the status of any selected objects in the current layer by choosing **Edit > Toggle Selection**.



5.2 The MIDI Signal Path

Before any MIDI events received at your computer's MIDI inputs can be recorded by the sequencer, there must be a connection between two Environment objects: the physical input and the sequencer input.



In the sequencer, the events are always directed to the selected track, where they can be recorded. The events played by the track are mixed with any incoming events and sent to the Environment object assigned to the track in the Arrange window's

Track List. From here, the events are directed to a MIDI output (see the section *Direct Output Assignment* below).

You can insert objects into the signal path between the physical input and the sequencer input as well as between the instrument and the MIDI Out port to accomplish other Environment processing (see section Cabling on page 5-13).

Direct Output Assignment

You can create a direct output connection from any instrument object to a physical MIDI output by setting the desired output in the **Port** parameter line underneath the channel parameter.

If you click and hold in this line, a flip menu will open that lists 63 individual ports and one general setting (0) for each of the three MIDI output connections as they have been set up in the communication preferences (**Options > Settings > MIDI-Communication**). Depending on the type of your computer (or rather its age...) these can be MIDI interfaces connected to the universal serial bus (USB) of your computer, to the modem port or to the printer port. Please note: Currently only USB connections to interfaces belonging to Emagic's Unitor family are supported. For more details please read section MIDI Interface Communication on page 17-11.

If you have a MIDI-interface with more than one output port (or even several interfaces which can be stacked, like products belonging to the Unitor family). you can set up a direct connection to one of the individual ports (1 to 63) of that or those interface/s (for example *"1-Modem 5"* or *"1-USB 23"*). If you select output 0 (zero) combined with a particular "connection" setting (e.g. *"1-Printer 0"*), the MIDI data of that instrument will be sent to **all** output ports of that particular connection.

Note: If you click in the right part of this line, the appearing flip menu will contain **all** available port settings (193!), even those which are not assigned to an output connection (labeled *"n/a"*). If you click on the left side, there are only four choices: Off and the same output number of each connection as it was set here

before (for example Off, *1-Modem 7*, *2-Printer 7* and *3-n/a 7*). This is useful if you want to switch off the direct output connection, or to switch between identical port numbers of different connections.

For more details about the MIDI connection settings and USB support please read section *MIDI Interface Communication* on page 17-11, and also look for "*ReadMe*" files and additional update info files on your program discs.

Any object with a directly assigned output is marked by a shaded triangle on its right side. When there is no direct assignment the triangle is hollow.



Breaking the Direct Output Assignment

To break the direct output assignment, set the the Port parameter to *off* which is the top entry in the flip menu there.

Cabling

The cabling between Environment objects gives you control over the entire MIDI signal path. A cable is normally shown as a gray line between a source and destination object.

Objects always have their input on the left side and their output on the right side. The output of an object is marked by a small triangle pointing to the right.



If you grab the triangle, the mouse pointer turns into a patch cord representing a cable connection coming from the output. Move the mouse to the destination object to connect it with a cable. Once you have selected the destination object (this happens automatically when you touch it), release the mouse button.



If the source object has already been directly assigned to an output, a dialog box will appear telling you this, and asking whether you want to replace the direct assignment. You have three options:

Cancel the connection is not made, and the direct output assignment remains intact.

No: your cabling is carried out, but the direct output assignment remains intact. This means that the object is connected in two places—one via the visible cable and one via the Direct Port assignment.

Remove (or *Backspace*): your cabling is carried out, and the direct output assignment is removed. (This is the default selection because you generally won't wish to have an object connected to two different destinations)

Cabling between Layers

If the destination object of the cable is on a different layer, you can connect it using the Instrument List. To do this, hold  as you grab the output triangle and scroll to the desired destination object when the Instrument List appears. (For more on the Instrument List see section *Icon/Display Filter for the Instrument List* on page 5 -23).

You can also open a second Environment window showing the destination layer, and connect the objects graphically between the windows.

A cable connection to another layer looks like this:



Multiple Cabling

Multiple inputs

There is no limit to the number of cables that you can “plug” into a target object. All the MIDI signals are then mixed at its input.

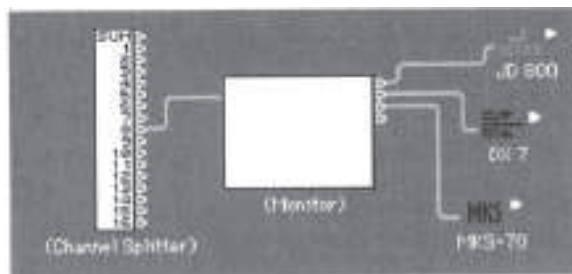
Multiple outputs

Once an output from an object is used (i.e. cabled to another object), another output triangle automatically appears. You can thus cable outputs from the object to as many other objects as you want.

Some objects have special outputs and are therefore exceptions to this rule.

Special Outputs

Some objects have several functionally *different* outputs, and in these cases, each output is only available once (e.g. channel splitter, cable switcher or physical input). If you want to route the signal from these outputs to several destinations you must first create a monitor (**New > Monitor**), plug the cable into it, then connect it to as many other destinations as you like. The monitor object also allows you to view the MIDI events flowing through it.)



To select a common destination for several existing cables, first select the cables. (If the cables are already leading to a common destination, the simplest way of doing this is to select the destination object.) Next grab one of the cables and plug it into the new destination object. A dialog box will appear asking “Do you want to connect all selected cables with new Destination? No/Connect”. Click Connect, or press .

Cabling Serially



To cable a group of objects serially, first select all the objects then select **Options > Cable serially**. The objects will be cabled in series, starting with the top-left object. This is handy for quickly connecting groups of faders in a mixing console, for example.

Cabling in Parallel



To cable a group of the same type of object (e.g. transformers, faders, etc.) to the same destination, first cable one of the objects to the destination. Next select that object and choose **Edit > Copy**. Finally, select the rest of the objects to be cabled to the same destination and choose **Options > Apply Buffer Template to > Cable(s)**. (You can copy more complex cabling configurations this way, too. Just ensure that the type(s) of objects in the group you select to copy, matches the type(s) of objects in the group you select to apply the buffer template to.)

Deleting Cables

To remove a cable connection, click the cable with the eraser or select the cable and press **Backspace**. You can also remove a cable by grabbing it and plugging it into the input of its own source object.

You can use Edit > Clear Cables only to remove all selected cables without also clearing any objects that happen to be selected. This is handy if you want to clear all the cables leading to or from one of more objects, because selecting objects also selects all their cable connections.

Cable color

Cables are given the same color as that of the object they came from (assuming, of course, that your monitor is set up to be used in color). You can switch off the cable coloring, and render them all gray with the **View > Colored Cables** function.

5.3 Environment Objects

Object Overview

Standard Instrument

This is the simplest track playback object. It provides a direct connection to a single MIDI port and channel, which can be set in its Parameter Box. You can also set a number of other playback parameters, including program bank & number, initial volume & pan, transpose, velocity scaling and MIDI delay (for details see page 3-12 and page 5-24).

Multi-Instrument

A multi-instrument is like a collection of 16 standard instruments (see above) in one package. There is one of these "sub-channels" for each of the 16 MIDI channels as indicated by the numbered squares on the multi-instrument icon. (Clicking one of these numbered squares selects the corresponding sub-instrument and displays its parameters in the Parameter Box.) All sub-channels of a multi-instrument use the same output port.

Multi-instruments have 1.5 name-banks each of which will hold 128 preset names (page 5-40). (Double-click the multi-instrument, or any of the sub-instrument numbers to open the name-banks window.) If the multi-instrument's program parameter (page 5-25) is checked, selecting names in the names-bank will send MIDI program change messages to the corresponding port and channel.

You will generally use multi-instruments to address multi-timbral synths or samplers in your studio. A multi-timbral synth is one that can receive on several MIDI channels at once - playing back a different sound on each channel. Since most newer MIDI devices are multi-timbral, the multi instrument

will likely be the most commonly used instrument in your Environment.

Mapped Instrument

This instrument is particularly useful for drum instruments or any drum-mode MIDI device. (A drum-mode device is one which assigns different sounds to different MIDI notes on the same MIDI channel.) Any input note may be routed to a different output note and assigned its own velocity setting, notation, name and output cable (see page 5-34). You can also protect Mapped Instruments from being accidentally transposed.

Faders

Faders are used to send MIDI events by clicking with the mouse on the screen. Faders come in different forms (e.g. knobs, sliders, numericals, and buttons). Faders will also respond to incoming MIDI events which makes them ideal for automated mixing (see page 5-78).

GM Mixer

The GM Mixer is a collection of 16 fader “modules” configured to emulate a virtual mixing desk for 16 MIDI channels. Each module has controls for volume, mute, preset, bank and four assignable knobs (one of which is typically used for pan). There are optional controls for standard XG and GS effects.

The GM Mixer comes in three styles and like standard instruments, is able to provide a direct connection to an output port which can be set in its Parameter Box.

You will most likely use Logic’s Adaptive Track Mixer for most of your MIDI & audio mixing but the GM Mixer can be handy when you want a dedicated mixer for a specific, multi-channel MIDI device.

Keyboard

The keyboard object is used for creating notes with the mouse. It will also display all notes passing through it. In this sense you can think of it as a real-time, note-on monitor (see page 5-40).

Monitor

The monitor object displays all MIDI events passing through it, and retains the last 32 such events. You will need to resize it to see more than 5 events at a time (see page 5-41).

Channel Splitter

The channel splitter routes MIDI events according to their channel. It has one output for each of the 16 MIDI channels and an additional “SUM” output. If a particular channel output is not connected to anything then events for that channel will appear at the SUM output (see page 5-41).

Transformer

The transformer is used to select and transform MIDI events passing through it. Its select-conditions allow it to act as a MIDI filter. Using its transform-operations along with the select-conditions allows you to transform some MIDI events while letting others pass through unchanged. Using what are known as “meta-events” (page 5-46) you can control many of the transformer’s parameters in real time (see page 5-42).

Voice Limiter

The voice limiter limits the number of MIDI notes (1 to 32) that can be held simultaneously. It does this by “note stealing” newly arriving notes will cause some currently held notes to be turned off once the voice limit is reached. The choice of what notes are turned off is controlled by the priority parameter, which can be set to last, top or bottom (see page 5-51).

Chord Memorizer

The chord memorizer maps single notes into chords containing up to 12 notes. It operates on a “pitch-class” basis, meaning that all notes with the same name (e.g. all C’s, all F#s, etc.) will be mapped to the same chord but transposed relative to the incoming note’s octave. The chord memorizer can also be used as a “scale filter” by mapping certain (non-scale) notes to nothing (see page 5-51).

Arpeggiator

The arpeggiator turns chords into arpeggios. It plays the currently held notes individually in a selectable pattern (up, down, random, etc.) and at a selectable speed between whole-notes and 768th-notes. It has a number of useful parameters including note-length, repeat on/off and octave doubling. All of its parameters can be changed in real time using MIDI controller messages (see page 5-47).

Logic’s transport must be running for the arpeggiator to work.

Delay Line

The delay line echoes MIDI events passing through it at intervals ranging from one tick to 256 whole notes. There are parameters for the number of echoes, as well as their transposition and velocity offset. The echoes of each incoming event will be cycled through each of the cabled outputs of the delay line (see page 5-50).

Logic’s transport must be running for the delay line to work.

Touch Tracks

The touch tracks object allows you to assign any selection of sequences and folders to a single MIDI note. You can then use your MIDI keyboard to trigger playback of these selections.

You can not use touch tracks to play audio regions.

Logic's transport must be running for the touch tracks to work.

Physical Input

The physical input represents the inputs of your computer's MIDI interface, and is normally connected to the sequencer input (see page 5-58).

Sequencer Input

The sequencer input represents the input to Logic's sequencer (i.e. to Arrange window tracks). MIDI events arriving here are routed to the selected track, where they may be recorded. They will also be sent to whatever Environment object is assigned to that track (see page 5-58).

MIDI Metronome Click

The MIDI metronome click generates MIDI note events at Logic's bar, beat and grid divisions. The pitch and velocity for each division can be set separately and the divisions can be turned on or off individually. The metronome itself is toggled on and off in Logic's transport (see page 5-59).

Logic's transport must be running for the MIDI metronome click to work.

Internal Objects

Internal objects allow MIDI to be routed to sound cards and software synthesizers inside your computer. The selection of devices (e.g. Quicktime, Wavemaker, Retro AS-1, etc.) depends on the hardware and software installed in your computer (see page 5-60).

Modem/Printer Port

Represents the physical outputs of the MIDI interfaces, which are connected to the port of the same name on your computer. If you are using an interface with several ports, you can create

one of these **objects** for each individual port, up to a maximum of 99 (see page 5-62).

Audio Objects

Audio objects are the only Environment objects that are not part of the MIDI signal flow. They allow MIDI controller messages to control aspects of Logic's audio playback such as volume, pan, muting and insert parameters. They are also used to create audio tracks in the Arrange window-audio tracks are just tracks with audio objects assigned to them.

Ornament Objects

The ornament object is not part of the MIDI signal path. It is simply a display object which allows you to create backgrounds behind other Environment objects. Its purpose is to allow you to create more attractive and useful control panels.

Macro Objects

Macros are not actually Environment objects at all. They are collections of Environment objects together with all their cables. Macros behave like single Environment objects. You can turn any reasonably sized (up to about 100) collection of Environment objects (including other macros) into a macro, then move it around, duplicate it and connect cables to and from it like any other object. You can also convert macros back into regular objects and cables for editing.

Aliases

As with sequences in the Arrange window, aliases of Environment objects represent the original object, but also have some properties of their own. Aliases are most often used for faders in which case they have their own value and cabling but send the same MIDI events as the original fader. In the case of text faders, aliases provide a significant savings in memory.

Object Parameters

Each Environment object has several parameters that control its operation. These parameters can be viewed and changed in the object's Parameter Box. The Parameter Box appears in the Arrange window when you select a track assigned to the object. It also appears in the Environment window when you select the object. These are the *same* parameters—changes in one location will be reflected in the other.

Opening/Closing the Object Parameter Box

By clicking the triangle in the top left, you can hide all of the parameters, except name and type. This reduces the box to its minimum vertical size.



Common Parameters

The following parameters are the same for all object types:

Name

The name of the object is shown next to the triangle, and can be selected for editing by clicking on it. You can also edit the name by clicking on the object with the text tool.



Object Type

The object type is shown in brackets, and cannot be edited.

Icon/Display Filter for the Instrument List

Click-holding on the icon in the Parameter Box opens a flip menu, where you can choose an icon to represent the object in the Environment and in the Arrange window's track list.

Placing a check in the box next to the icon means that the object will appear on the Instrument List, and can therefore be assigned to a track by using this menu. (Objects not on the track list can still be assigned to tracks by dragging them from the Environment window to the track list or by using the Environment's MIDI Thru tool.)



If the box is not checked, the object will not appear on the Instrument List, and a diagonal line will appear through the icon. This will not hide the object or its icon from view in the Environment.



You should only check the icon box if the object will be used as a track instrument, or you need to connect cables to its outputs using the instrument selection menu (see page 5-14).

5.4 Object Reference

Standard Instrument

To create a new standard instrument, select **New > Instrument**, or click in the background with the pencil tool. Instruments serve as the interface between Logic's tracks and your MIDI sound sources.



Please note: In older versions of Logic the name of this object was just "Instrument". The term "Standard Instrument" is now used both in the program itself and in this book to clearly point out the difference between the "Standard Instrument", "Multi Instrument" and "Mapped Instrument" objects.

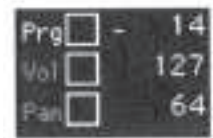
The Standard Instrument Object Parameter box will be familiar to you from the Arrange window. Here are the details of the individual parameters.

Cha Use the *Cha* parameter to set the MIDI channel for the instrument's output. All events at the output of the standard instrument will be sent with the channel that is set here. Exception: If you set this parameter to *All*, all events will be sent with their original channel settings as they appear in the Event List window.

Port Below the channel parameter, a direct connection to one of the MIDI output ports can be set. Remember that you can also directly cable the standard instrument to other Environment

objects for MIDI processing. If no MIDI output port is defined in the object parameter box for that instrument, the *Port* parameter will display "off". For details please read section *MIDI Interface Communication* on page 17-11 and section *Direct Output Assignment* on page 5 -12.

Prg, Vol and Pan The *Prg*, *Vol* and *Pan* parameters transmit program changes, volume controller (#7) and pan controller (#10). If there is no check in a box, you can edit its value, but it will not be sent until you check the box (by clicking on it). If the box is already checked, any value alterations will be sent immediately, and the values will also be sent whenever the track is selected.



To the left of the program number, directly to the right of the box, is the parameter for bank select. If your MIDI sound sources recognize Bank Select messages (check in their manuals), you will be able to switch between sound banks. If your sound sources respond to the standard Bank Select message (Controller #32), you will be able to use this parameter directly. If not, please read the section *Defining Your Own Bank Select Commands* on page 5-32.

Transpose The *Transpose* parameter allows you to define the number of semitones by which all the note events will be transposed as they are output. Negative values transpose downwards.

Velocity The *Velocity* parameter allows you to increase or decrease the “note on” velocities of all note events by an amount adjustable between -99 and 99.

Lim The two note values of the *Lim* parameter define a pitch range. All notes outside this range will be ignored by the instrument when it plays a sequence.

- VLim* The two values of the *VLim* parameter define a velocity range. All notes whose velocity is outside this range will not be played by the instrument.
- Delay* The *Delay* parameter causes all MIDI events to be sent early or late by from -99 to 99 ticks. This allows you to adjust for any differences in reaction time between your various MIDI devices. For delay effects use the sequence parameter of the same name, as this allows longer delay times.
- No Seq Trp* If the box next to the *No Sequence Transpose* parameter is checked, all sequences on any tracks played by this instrument are protected from transposition. In other words, the “transpose” sequence parameter is ignored. This is very useful for instruments assigned to drum or other multi-timbral samples, as transposition will cause the sounds rather than the pitches to change for these instruments.
- No Reset* If the box next to the *No Reset* parameter is checked, no reset messages will be sent to this instrument. This can be useful if controllers are being used for non-musical purposes, for example when an Instrument is used for mixer automation. The **Preferences > Reset Messages** window controls what reset messages are normally sent, but these messages are not sent to “No *Reset*” instruments.
- Default Score Style* The *Default Score Style* parameter is located at the bottom of the Parameter Box. It displays *Auto Style* by default but can be changed to any of the available score styles. Whenever a sequence is created on one of the instrument’s tracks it will be assigned the score style



displayed here. In the case of *Auto Style*, Logic will pick an appropriate style based on the pitch range of the notes in the sequence.

Don't forget: you can change a sequence's Score Style at any time in the Display Parameter box in the Score window.

You can change the Score Styles of all sequences on an instrument's tracks by holding down *Alt*, when selecting the instrument's default Score Style.

Multi-Instruments



To create a new multi-instrument, choose **New > Multi Instrument**.



A multi-instrument is intended for use with multi-channel MIDI sound devices. It combines 16 separate sub-instruments (one for each MIDI channel) with 15 name-banks, and a number of common bank-select options.

A multi-instrument consists of 16 sub-channels, each of which has a complete set of instrument parameters. The name, MIDI channel and output of each sub-channel is determined by the multi-instrument. All sub-channels share the 15 banks of program names, and use the same format for their bank select message.

Multi-Instrument Parameter Box

Once you have created a multi-instrument, you will see the multi-instrument parameter box shown here.

You should already be familiar with the meaning and operation of the individual parameters (if not refer to the description on page 5-34). The settings made here apply to all sub-channels and except for the output port setting, should be left at their neutral, default setting. Set the output port in the line below the "*Cha*" parameter line (parameter *Port*). For more information on this subject, please read the section *Direct Output Assign-*



ment on page 5-12. However, keep in mind that the direct output assignment can also be replaced by manual cabling to other environment objects.

Activating and Selecting Sub-channels

To select a sub-channel for a specific MIDI channel number, click the appropriately numbered button on the multi-instrument object. The first time you click it, the sub-channel is activated making it available from the Instrument List. To avoid menu clutter, you should only activate as many sub-channels as you need. As with any other Environment objects, you can remove sub-channels from the Instrument List, by unchecking the icon box. The button for that sub-channel will then be shown with a diagonal line through it. If you want to select the entire multi-instrument rather than a particular sub-channel, click on the top edge of the multi-instrument or next to the icon.

In this diagram, sub-channel 1 is selected, sub-channels 1-8 are activated and 9-16 have been removed from the instrument list.



The Sub-channel Parameter Box

The sub-channel parameters are the same as for Standard Instruments (see page 5-24). The only parameter you can not change is the MIDI channel. (If you change the output port, this will affect the whole multi-instrument and all other sub-channels.) If you try to change the channel from the Parameter Box in the Environment window you will get the following warning: “Channel protected! Please select a sub-channel from the Instrument flip menu.” However, if you change the channel in the Arrange window’s Parameter Box to a sub-channel that is on the instrument selection list (i.e. one without a slash through it on the multi-instrument), the track’s instrument will change to the selected sub-channel.

Object Reference

Cabling

To connect the output of an Environment object directly to the input of a sub-channel, hold down **Alt**, and connect it the via the Instrument List (see section *Cabling between Layers* on page 5-14).

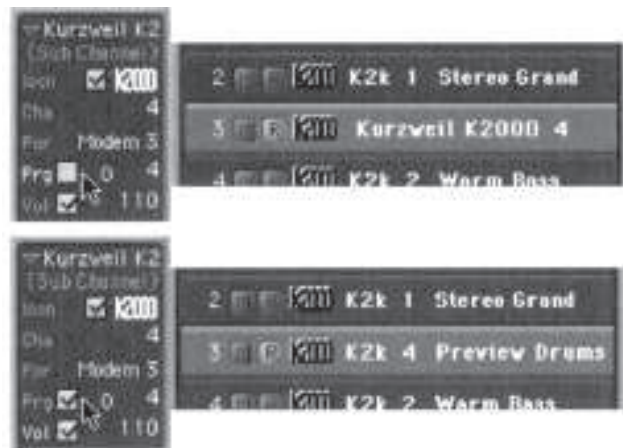
Multi-Instrument Window

Double-clicking on a multi-instrument opens the multi-instrument window:



Name and Short Name

In the top left, under *Device Name*, *you* can enter the full name of the multi-instrument. In the top right, under *short Device Name*, *you* can also enter a short name. This short name is used in the Arrange window track list when the program name is displayed (see below). The short name begins right next to the icon, so leave a space in front if you want uniform spacing in the track list.



Depending on whether or not you have checked the sub-channel's *Prg* box, the following information will appear in the Arrange window track list:

the name of the multi-instrument & channel number (if the box is not checked) or

the short name, channel number and program name (if the box is checked).

ProgramNames

There are 128 program names in the multi-instrument window. A total of 15 banks of 128 program names are available. There are several ways of entering the program names:

by double-clicking on the name (via the text input field),

by copying via the clipboard from a different multi-instrument, or from a word processing program. The clipboard functions for a whole sound bank are available in the text import flip menu (downwards arrow at the top right). First copy the program numbers or General MIDI names to the clipboard, and add them to a word processor document. You can then edit the names and copy the whole section back again.

If you want to use program numbers instead of names, select **Init Names as Numbers** from the text import menu.

if you want to use General MIDI program names, select **Init General MIDI Names**.

if you have Emagic's SoundDiver editor/librarian software: start SoundDiver then switch back to Logic and select **Update Using Autolink**. The current program names for the currently selected bank will then appear in the multi-instrument window automatically.

If the **Use GM Drum Names for Channel 10** box at the very bottom is checked, the standard GM names for drum sets will be shown in the multi-instrument window for sub-channel 10.

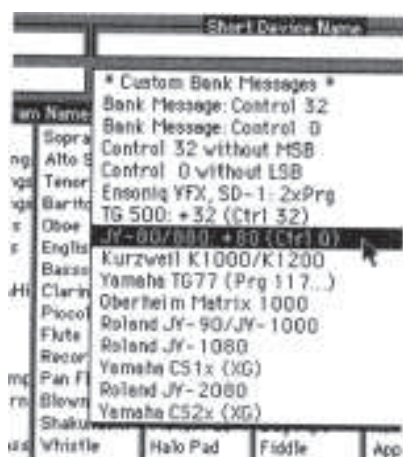
If the Prg box in the selected sub-channel's parameter box is checked, you can send a program change messages by selecting a program names in the multi-instrument window (either by clicking, or using your computer's arrow keys).

Banks

On the left, above the program names, you can choose one of 15 available sound banks (0-14) via a flip menu. The top item ("No Bank specified. Names of Bank 0 used.") can be used if that particular sound generator does not understand bank select messages, or only has 128 sound programs. Bank 0 is always initialized. The first time you choose one of the banks 1-14 you will be asked whether you want to initialize this bank. Press **OK** or **Cancel** if you do not want to initialize the bank. Non-initialized banks do not have their own program names, but use the names of the equivalent program numbers from bank 0. You should only initialize additional banks if you want to enter program names for those banks, because each initialized bank uses more memory.

Chapter 5

The Environment



In the flip menu to the right, labeled **Bank Message**, you can define the MIDI events which are sent when you switch banks on this multi-instrument. Different bank select formats are used by different MIDI instrument manufacturers—please consult your instrument’s manual to see whether it supports bank select messages and if so, using what format. With modern synths, there is a good chance the format will be one of the top items in the flip menu: either controller #32 or controller #0. There are also presets to accommodate several of the more common types of synth. If your synth does not use one of the formats on the list, don’t give up—all is not lost!

Defining Your Own Bank Select Commands

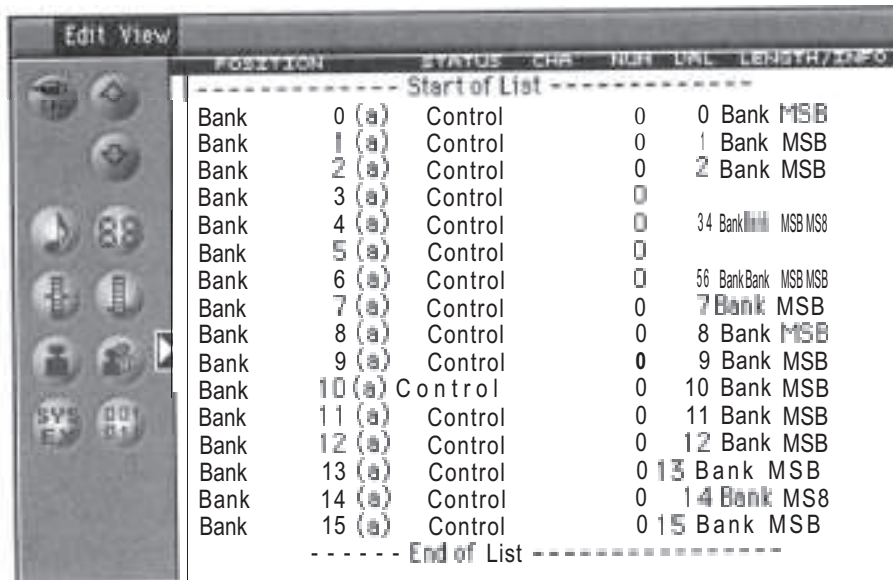
Logic provides 63 bank numbers (0 thru 62) for each single standard instrument, multi-instrument sub-channel, or mapped instrument. You can create custom lists of as many events of any kind (even Sysex) as you wish - for each of these banks. Whenever you change the bank manually or send a standard bank change from Logic, the entire list for that bank will be transmitted to your synth.

Note: if you need more than 63 banks for a particular MIDI device, use several instrument objects.

To set up custom bank select messages, first select the instrument, then select **Options > Define Custom Bank Messages...** A window similar to the event list will open, with a single,

Object Reference

default bank select message for banks 0 thru 15. (The default used is a MIDI controller #0 message with value equal to the bank number.)



POSITION	STATUS	CHN	RMT	DML	LENGTH	SYSEX
----- Start of List -----						
Bank 0 (a)	Control	0	0	0	1	Bank MSB
Bank 1 (a)	Control	0	0	1	1	Bank MSB
Bank 2 (a)	Control	0	0	2	1	Bank MSB
Bank 3 (a)	Control	0	0		1	Bank MSB
Bank 4 (a)	Control	0	0	34	1	Bank MSB
Bank 5 (a)	Control	0	0		1	Bank MSB
Bank 6 (a)	Control	0	0	56	1	Bank MSB
Bank 7 (a)	Control	0	0	7	1	Bank MSB
Bank 8 (a)	Control	0	0	8	1	Bank MSB
Bank 9 (a)	Control	0	0	9	1	Bank MSB
Bank 10 (a)	Control	0	0	10	1	Bank MSB
Bank 11 (a)	Control	0	0	11	1	Bank MSB
Bank 12 (a)	Control	0	0	12	1	Bank MSB
Bank 13 (a)	Control	0	0	13	1	Bank MSB
Bank 14 (a)	Control	0	0	14	1	Bank MSB
Bank 15 (a)	Control	0	0	15	1	Bank MSB
----- End of List -----						

You can create MIDI events here in exactly the same way as you do in the Event Editor, by cutting, copying, inserting and editing. The only difference is that you enter the bank number you want, instead of a time position.

The letter in parentheses next to the bank number allows you to control the order of transmission, when a bank uses more than one message.

For example:



POSITION	STATUS	CHN	RMT	DML	LENGTH	SYSEX
----- Start of List -----						
Bank 1 (a)	Control	32	0	0	1	Bank LSB
Bank 1 (b)	Control	0	0	1	1	Bank MSB
Bank 1 (c)	\$F0	\$01	\$0A			Sequential
----- End of List -----						

Here the Bank Select command "Bank 1" has been set up to send three events: first, (a) controller #32 message with the value 1, then (b) controller message #0 with value 0 and finally, (c) a Sysex message.

If there are no events defined for a particular bank, a standard Bank Select message will be sent for that bank instead.

The custom bank select information becomes part of the instrument, and will be automatically copied along with it.

For bank messages that need a channel (e.g. MIDI controller messages), the channel of the instrument is used. (This feature is especially useful for multi-instruments, since you only have to create one set of bank messages for all 16 sub-channels.) If an instrument's channel is set to *All*, then channel 1 is used.

On multi-instruments, the bank select message (whether it's a standard controller or a string of events you've configured) is also used by Sound Diver, if Autolink is switched on. This enables it to request the program names in the current bank.

Mapped Instruments

To create a new mapped instrument, select **New > Mapped Instrument**.

A mapped instrument is useful for drum programming. It is used just like a standard instrument, but has the following special features:

Each individual input note can be:

- named (e.g. snare, hi hat...);
- mapped to an output note (e.g. F# and G# to the same hi-hat sound to allow you to play rapid repeats);
- given a velocity offset;
- assigned its own MIDI channel;
- sent to one of up to 16 output cables (this allows you to create a single instrument that addresses multiple sound sources);
- given its own notation parameters:

- note head shape
- relative vertical position in the staff
- drum group assignment

Object Parameter Box

The mapped instrument's parameters are a subset of the parameters of a standard instrument (described on page 5-24). The



missing settings are available on a note-by-note basis in the mapped instrument window (see page 5 -37).

Cha Use the ***Cha*** parameter to set the MIDI channel for the instrument's output. All events at the output of the instrument will be sent with the channel that is set here. Exception: If you set this parameter to All, all events will be sent with their original channel settings as they appear in the Event List window.

Port Below the channel parameter, a direct connection to one of the MIDI output ports can be set. Remember that you can also directly cable the instrument to other Environment objects for MIDI processing. If no MIDI output port is defined in the object parameter box for that instrument, the ***Port*** parameter will display off.

Prg, Vol, Pan The ***Prg***, ***Vol*** and ***Pan*** parameters transmit program changes, volume controller (#7) and pan controller (#10) If there is no check in a box, you can edit its value, but it will not be sent until you check the box (by clicking on it). If the box is already checked, any value alterations will be sent immediately, and the values will also be sent whenever the track is selected.

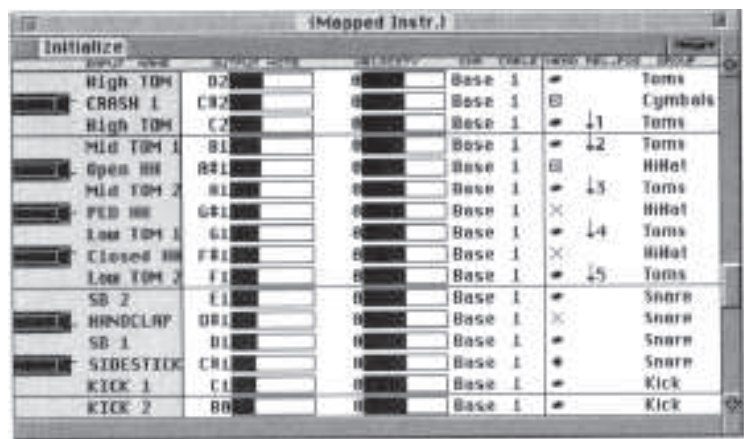
To the left of the program number, directly to the right of the box, is the parameter for bank select. If your MIDI sound sources recognize Bank Select messages (check in their manuals), you will be able to switch between sound banks. If your sound sources respond to the standard Bank Select message (Controller #32), you will be able to use this parameter directly. If not, please read the section section ***Defining Your Own Bank Select Commands*** on page 5-

- Delay** The *Delay* parameter causes all MIDI events to be sent early or late by from -99 to 99 ticks. This allows you to adjust for any differences in reaction time between your various MIDI devices. For delay effects use the sequence parameter of the same name, as this allows longer delay times.
- No Seq Trp** If the box next to the No *Sequence Transpose* parameter is checked, all sequences on any tracks played by this instrument are protected from transposition. In other words, the “transpose” sequence parameter is ignored. This parameter is typically checked for mapped instruments since they generally are used for drum- mode MIDI devices and transposition would cause the sounds rather than the pitches to change.
- No Reset** If the box next to the No *Reset* parameter is checked, no reset messages will be sent to this instrument. This can be useful if controllers are being used for non-musical purposes, for example when an instrument is used for mixer automation. The **Preferences > Reset Messages** window controls what reset messages are normally sent, but these messages are not sent to “*No Reset*” instruments.
- Default Score Style** The *Default Score Style* parameter is located at the bottom of the Parameter Box. It displays *Auto Style* by default but can be changed to any of the available score styles. Whenever a sequence is created on one of the instrument’s tracks, it will be assigned the score style displayed here. In the case of *Auto Style* Logic will pick an appropriate style based on the pitch range of the notes in the sequence.



Mapped Instrument Window

To open the mapped instrument window, double-click on the napped instrument icon in the Environment window. The rows correspond to input notes, and the columns contain the various parameters available for each note.



Monitor Keyboard (Selecting Notes)

The keyboard on the left represents the input notes. It can be played by clicking on it. You can also select individual notes or note ranges, by dragging the mouse over the notes you want. To select more notes, click them while holding . Any value alterations will apply to all selected notes.

Input Name

In the next column, you can click on the input note and give it a name of up to 12 characters. You can initialize the names of the selected notes to their pitch descriptions (e.g. (C#3) by selecting **Initialize > Names as Notes**, or to the names of the GM Standard drum sounds by selecting **Initialize > Names as General MIDI**.

Output Note

This column is where you set the output note either as text by double-clicking on the note description, or graphically by dragging the beam. When you change the value, MIDI notes are sent out so you can hear what you're doing. If you choose

Initialize > Output Notes, the output notes of the selected pitches are set to be the same as the input notes.

Velocity

This is where you set a velocity offset, which is added to or subtracted from the velocity of the incoming note. You can either grab the number and use the mouse as a slider, or click the beam at the desired point. **Initialize > Output Velocities** resets all velocity offsets to 0 (no offset).

Cha

Here you can set the MIDI channel of individual notes. This allows you to play individual sounds from different drum sets in the same sound generator. Normally you will use the *Base Setting* here. This means that the notes are sent on the channel set in the mapped instrument's object parameter box. If you choose *All*, the channel information of the incoming notes is used. This is useful if you want to insert the mapped instrument in the signal path after a multi-instrument or single instrument. To set all selected notes to *Base*, select **Initialize > Output Channels**. Please also refer to the section *Setting the MIDI Channel: Multi Instruments* on page 3-15.

Cable

You can send individual notes from the mapped instrument object to up to 16 different output cables, to enable you to play sounds from different sound sources. If you select a cable that does not exist, then the note will not be sent out. **Initialize > Output Cables** resets all cables to #1 (the top outlet of the mapped instrument).

NotationParameters

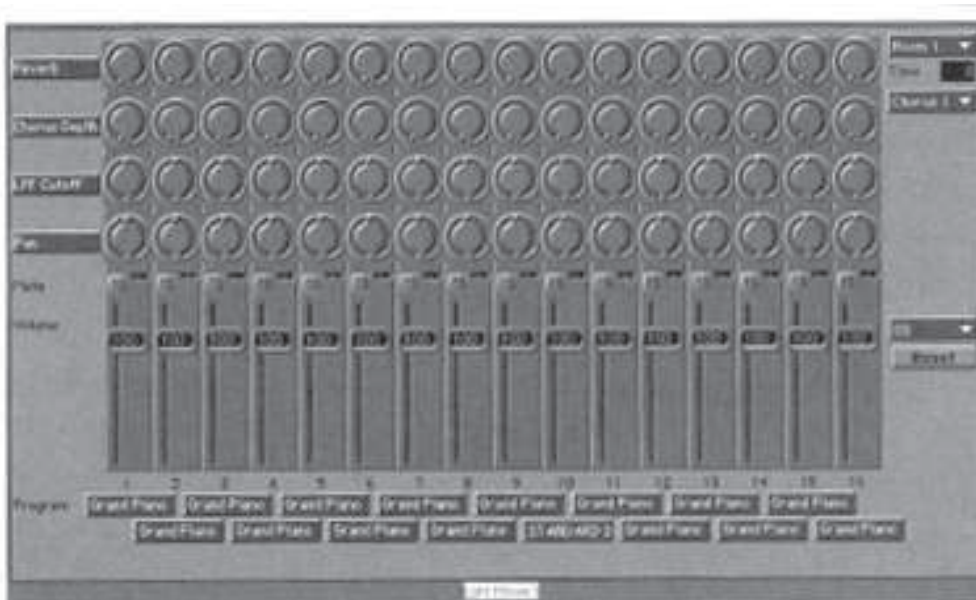
The last three columns define the notation parameters of the individual notes. Selecting **Initialize > Score Parameters** neutralizes the settings of the *Head*, *Rel Pos.* and *Group* parameters.

Faders

To create a new fader, select **New > Fader >...** A sub-menu appears, where you select the fader type you want. Faders are such a diverse and important part of the Environment, that we have devoted an entire section just to them. Please see the section *Faders* on page 5-68.

GM Mixer

To create a new GM Mixer, select New > **GM Mixer**.



The GM Mixer is intended for controlling the 16 channels of a MIDI device which conforms to the GM, XG or GS standard. This includes the GM set of 128 program names (with variation banks for XG or GS), MSB/LSB bank select messages, and standard controller names (#1 for mod wheel, #2 for breath, #7 for volume, #10 for pan, #11 for expression, etc.).

Most of the GM Mixer's set up is done in the mixer itself, but there are also a few parameters in its parameter box.

Cha/Port

Like an instrument object, the GM Mixer can also have a direct MIDI output connection. This is set in the line below the channel in the Parameter Box (see also page 5-12). The *Cha* para-



meter itself just sets the channel of the lowest GM Mixer module. You can use this, together with resizing the GM Mixer window, to create sub-mixers for any continuous string of channels (e.g. 3 thru 7).

Legend The *Legend* checkbox toggles the display of the legend along the left edge of the GM Mixer. The Legend not only indicates what the rows of controls do, it is used to set the function of the top four rows of knobs. After the GM Mixer is set up, you can hide the legend to save space.

Bank The *Bank* checkbox toggles bank MSB/LSB display below the program name display, at the bottom of the GM Mixer. (Bank display is off in the illustration above.) You can save space by hiding bank display, unless you need to select program banks numerically.

Style The GM Mixer comes in three flavors. Choose the one most pleasing to your eye.

Keyboard

You can create a virtual keyboard by selecting **New > Keyboard**.

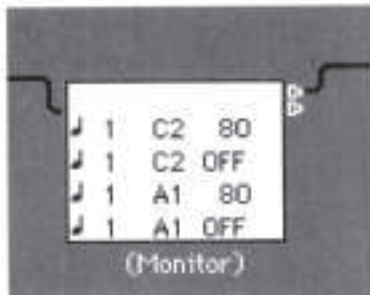


The keyboard object generates MIDI note events when clicked with the mouse. It also displays note events passing through it-when a note is being held, the corresponding key will reverse its color. Although you can record the output of the keyboard in Logic, its main purpose is for testing and monitoring in the Environment.

Apart from the standard parameters, you can also set the MIDI channel, a fixed velocity for all notes and the octave of the lowest note on the keyboard (which is always a C). You can re-lize the keyboard to control its range above the lowest note. Click the keyboard with  to select and move it.



Monitor



To create monitor, select **New > Monitor**. The monitor shows all events (MIDI and meta) passing through it. It remembers the last 32 events, with the newest events at the bottom. You can resize it to show from 1 to 32 events. (Actually you can make it bigger but the display above the 32nd line is gibberish.)



Monitors are very handy as both testing, and branching devices. In the case of objects that have functionally different outputs [e.g. the channel splitter and the mapped instrument), you can use them to attach more than one output cable with the same function.

Click in the body of the monitor to clear it.

Channel Splitter



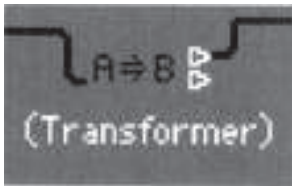
To create a new channel splitter, select **New > Channel Splitter**. You can use the channel splitter to reroute MIDI events to different cables, according to their MIDI channels. Every MIDI event received at the input of the channel splitter is first routed to the output corresponding to its MIDI



channel. If there is no cable connected there, the event is rerouted to the SUM output (at the top). The SUM output carries the events of those channels which have no cables from their individual outputs.

If you want to route the information for a single channel to more than one object, use a new transformer or a monitor as a junction (see above and page 5-14).

Transformer



You can create a new transformer object by selecting **New > Transformer**. The transformer allows you to both select and alter MIDI events in real time.



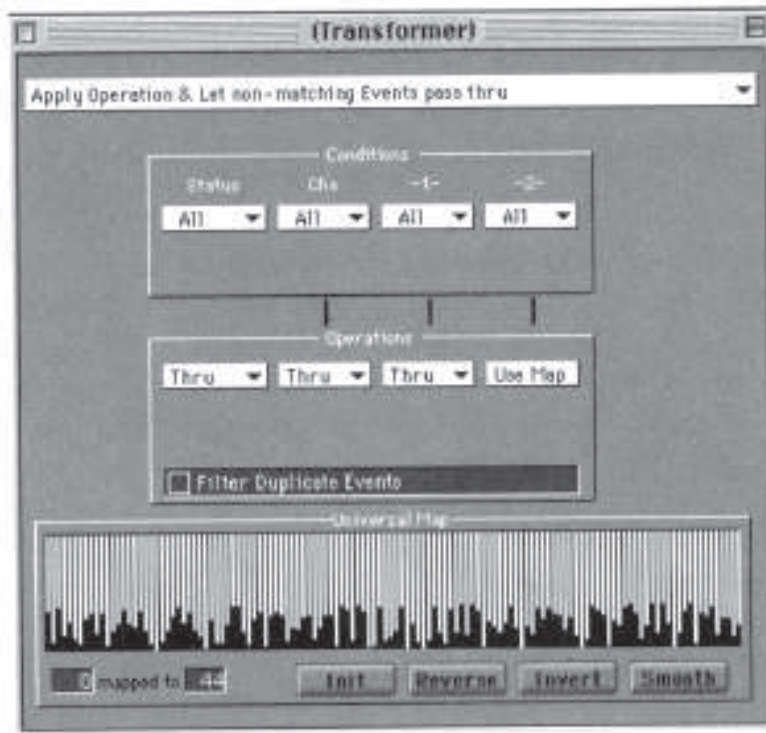
For the transformer to work, it must be in the MIDI signal path. If you use it to alter the output of an instrument, for example, cable it between the instrument and a port object (removing the instruments internal port assignment in the process).

Working Principle

The transformer first checks whether an event corresponds to certain *conditions*. If it does, then certain *operations* are carried out. Events which do not meet the conditions may pass unchanged to the output, or be thrown away. The transformer can also make copies of matching events before altering them, and can split the selected events from the unselected ones, using its two top cables.

Transformer Dialog Box

Double-clicking on a transformer opens a dialog box, where you can set the conditions and operations. You can open this box as a floating window, by double-clicking on the transformer while holding the **Alt** key.



Mode of operation

Use the flip menu at the top of the transformer window, to set the transformer's mode of operation.

Apply Operation & Let non-matching Events pass thru

MIDI events which conform to the condition are processed. MIDI events which do not conform to the condition are passed thru. This is the default setting.

Apply Operation & Filter non-matching Events

MIDI events which conform to the condition are processed. MIDI events which do not conform to the condition are not passed through.

Filter matching Events

All MIDI events which match the condition are filtered out. MIDI events which do not conform to the condition are passed through.

Copy matching Events & Apply Operation

All MIDI events which conform to the condition are copied and the copy is processed. The original *and* transformed copy (plus any MIDI events which do *not* conform to the condition) are passed thru. The unchanged original is passed *before* the transformed copy.

Copy matching Events & Apply Operation (rev. order)

This is the same as above except the original is passed *after* the processed copy. You might want to use this for example, when converting note events to pan controllers. This causes the pan message to be sent before the note. (Many synths do not alter the pan position of already sounding notes.)

Condition Splitter (true -> top cab/e)

Events which match the conditions are altered by the operations and sent to the transformer's top output. Events which do not match the conditions are sent, unaltered, to the transformer's second output. Do not use the other outputs-nothing ever appears there.

Alternating Split

Events entering the transformer will be alternated between the top two outlets. No conditions or operations apply.

SysEx Mapper ('1' -> position, '2' -> value)

This transformer mode is used to create and edit Sysex messages. You first enter the structure of the Sysex message in the transformer window-i.e. its length, whether a checksum is required, and the values of bytes you don't want to change in real time. Next you select which kind of MIDI event will affect the data bytes in the Sysex message. (Typically you will use MIDI controller events.) Incoming MIDI events will then change the data bytes in the Sysex message according to the following rules:

The -1- value sets the position of the data byte to change.

The -2- value sets the value of the data byte.

The channel controls the action to be taken:

Channel 1: The changed SysEx message is sent.

Channel 2: The changed SysEx message is not sent.

Channel 3: The unchanged SysEx message is sent.

Channel 4-16: No meaning (reserved for future use).

The purpose of the channel controls is to allow multiple bytes of a Sysex message to be modified before it is sent.

The “Filter non-matching” parameter in the sysex mapper transformer window prevents incoming MIDI events which do not control the Sysex message from being passed through. Typically you will want this feature turned on, to prevent inter-
locking controller data from invading the Sysex data stream.

Filter Duplicate Events

You can use this feature to save MIDI bandwidth. When this checkbox is selected, the transformer will filter out a MIDI event if:

- its type matches the transformer’s Status condition. (This Status condition must not be set to “All” for this to work.)
- the previous event matching the Status condition was identical (i.e. same channel, -1- and -2- value).

Note that you can use this feature with the Status condition set to “All” to filter out duplicates, but here duplicates will only be recognized if they are consecutive (i.e. there are no other events in between).

Conditions and Operations

The conditions and operations are the same as for the Transform window. The only differences arise from the fact that transformer objects work in real-time and therefore, position and note length have no useful meaning.

Pitch Bendediting

The transformer can process 14-bit pitch bend events: if byte is changed (e.g. by addition or scaling), the change will affect 1

the second byte too. Set the -2- operation to "THRU" to process 14-bit pitch bend data properly.

Remote Controlling the Condition and Operation Values

The value(s) of any transformer conditions or operations can be externally controlled using meta events.

Meta events are internal Logic messages-their form is similar to MIDI controller events, but they have no MIDI meaning and never leave Logic.

Like MIDI controller events, all Meta events have a two data values: the first one indicates the type of Meta event (49-127 but not all are used), and the second one is the event value (0-127).

Meta events number 122 thru 127 affect transformer parameters.

Meta event #127 sets the first (top) operation value,

Meta event #126 sets the second (bottom) operation value (if available).

Meta event #125 sets the first (top) condition value.

Meta event #124 sets the second (bottom) condition value (if available).

Note that **all** numerical condition and operation parameters that are used (i.e. conditions not set to All and operations not set to Thru) will be affected by the same meta event. (The status condition and operation are never affected by meta events.) If you want to set numerical parameters individually, or leave some fixed, use separate transformers in series.

Meta event #123 sets the transformer map value for the current map position. (Use meta event #122 to set the position first.)

Meta event #122 sets the transformer map position.

Note that you can use a transformers *MapSet* operation to create these two meta events simultaneously. When the operation status is set to *MapSet* then the -1- parameter specifies the map position and the -2- parameter specifies the map value at that position. (I.e. a meta event #122 is sent with the -1- value followed by a meta event #123 with the -2- value.)

You can create meta events with a fader; you can transform MIDI events into meta events with a transformer, and you can add meta events to a sequence using the Event List editor. When you use a fader, set the Out definition to *Switch*, then set the -1- value to the meta event number. (Cable switcher type faders and meta events share the same Outdefinition. -Z-values above 49 signify meta events.)

Since you can remote-control any fader by using its In definition, you can use faders as well as other transformers to convert MIDI events to meta events for remote-control of transformers.

Condition: Map

Most of the transformer conditions are numerical relations (=, unequal, inside, etc.), and the condition is met if the incoming MIDI event value satisfies the condition. The last condition, *Map*, works a little differently. In this case, two numerical parameters are specified, and the incoming value is first converted by the map, then the mapped value is compared to the two parameters to see if it is inside them.

Transforming Meta Events

Transformers can also process meta events, as long as they are not the meta events that affect transformers. These particular meta events change the transformer, rather than being processed by it.

Arpeggiator

To create a new arpeggiator, select **New > Arpeggiator**.





The arpeggiator cycles through all held notes (i.e. arpeggiates chords) within its range. You can set the range as wide or narrow as you wish. There are parameters for direction, velocity, speed, note length, start quantize (a.k.a. snap), repeats, octaves, and velocity offset. All the arpeggiator's parameters can be MIDI controlled in real time, which adds immensely to its versatility.

To use the arpeggiator, you must put it in the MIDI signal path and Logic's Transport must be running. (For technical reasons, the arpeggiator resets on cycle jumps.) Typically you would assign the arpeggiator to an Arrange track, and cable its output to an instrument or port object. Of course, you can insert it anywhere else in the MIDI signal path that you like.



You can then use the arpeggiator with live MIDI input, or for sequence playback. You can also record the output of the arpeggiator by cabling it back to the sequencer input object. If you do this, be sure you either record to a no output track, or break the arpeggiator's connection to the MIDI output.

Parameters

An arpeggiator has the following parameters (the numbers in parentheses are *CTRL Base* offsets-see below):

Direction (+0)

The direction of the arpeggiated chord:

- Up* Lowest note to highest note
- Down* Highest note to lowest note
- UpDn* Up and down-highest & lowest notes repeat
- Auto* Up or down depending on whether the second chord note arrived before or after the first chord note
- UpD2* Up and down-highest & lowest notes don't repeat
- Rand* Notes play in random order
- All* All notes play at once (useful when repeat is on)

Vel (+1)

Velocity values of the arpeggiated notes.

- 1-127** Fixed velocities,
Orig the velocities of the recorded notes are retained,
Rand random velocities between 1 and the original value.

Lim (Low: +2 High: +3)

This is where you can define the pitch range (between C-2 and G8) within which the chords are arpeggiated. Any chord notes outside this range are passed directly to the output. This is useful when you want to solo above, or play a bass line below an arpeggiated chord.

Res (+4)

This is where you set the rhythmic note value (i.e. the speed in note divisions) of the arpeggio via the flip menu shown here. None = arpeggiator switched off.

Len (+5)

This is where you define the length of the arpeggiated notes. The Orig setting means that the lengths of the recorded notes are retained.

Snap to (+6)

Unless this value is set to None, the arpeggiator will wait for Logic's transport to reach the next indicated note division before starting the arpeggio. This is useful for rhythmically syncing the arpeggio with other MIDI data.

Repeat (+7)

On continues the arpeggio for as long as the chord is held down.
Off finishes the arpeggio after one run.

Octaves (+8)

The arpeggio can be repeated over 1-10 octaves.

Crescendo (+9)

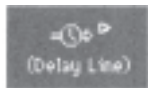
The velocity value set here (-99 to +99) is added every time the arpeggio is repeated (provided *Repeat* is set to On, of course).

Ctrl Base

All 10 parameters of the Arpeggiators can be remote controlled by controller events. With the parameter Ctrl Base, you can determine the controller number for the first parameter (Direction). The other parameters will be controlled by subsequent controller numbers. To find out which controller number will be used for a certain parameter, add the number shown in brackets behind the parameter's name in this documentation, to the Ctrl Base value. When *CTRL Base* is set to Off MIDI control of the arpeggiator is turned off.

Delay Line

To create a new delay object, select **New > Delay line**.



This object repeats any MIDI events, like a delay line. Pitch and velocity offsets can be added to note events. Like the arpeggiator, you will need to put the delay line in the MIDI signal path and you will need to start Logic's Transport for it to work.

The delay line can send up to 99 repeats, at intervals ranging from one tick to 256 whole notes. You can suppress the original note. Repeats leaving the delay line will cycle through all cabled outlets, so that you can play each repeat with a different sound.

Parameters

A delay object has the following parameters:

Thru Original: if this box is checked, the original signal is passed thru, otherwise it is suppressed.

Repeats: number of event repeats (1 to 99). Setting 0 = delay switched off.



- Del:** the delay time between the individual repeats. The left value is in divisions and the right value in ticks.
- Trp:** the transposition of note events per repeat (+ 99 semitones).
- Vel:** the change in the velocity values of note events per repeat ($\pm$ 99).

Voice Limiter

To create a new voice limiter, select **New > Voice imiter**.



The voice limiter limits the number of MIDI notes (1 to 32) that can be held simultaneously. It does this by “note stealing”- newly arriving notes will cause some currently held notes to be turned off, once the voice limit is reached (see the *Priority* parameter below).

Normally you will assign the voice limiter to an Arrange track, and cable its output to the instrument object whose voices you want to limit. Or, you could cable it between an instrument object (breaking its internal port connection) and a port object.

Parameters

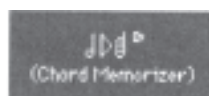
- Voices:** this is where you define the maximum number of simultaneously played voices, from 1-32.
- Priority** this is where you define which notes get stolen (i.e. turned off) when the number of held notes exceeds the limit. *Top* = lowest notes are turned off first. *Bot* = the highest notes are turned off first. *Last* = the earliest notes are turned off first.



Chord Memorizer

To create a new chord memorizer, select **New > Chord Memo-rizer**.





The chord memorizer maps individual notes to chords. You can assign one chord to each pitch class (i.e. to C, C#, D, etc.). The octave of the incoming note determines the octave of the resulting chord. A chord can have from 0 to 12 notes in it. (Zero and one note chords can be useful for creating scale-filters and scale-correctors).

The easiest way to use the chord memorizer is to connect its output to the instrument that you want to have play the chords, and assign it to an Arrange window track. You can, of course, put it anywhere else in the MIDI signal path that is convenient.

Parameters

Cha: All chord notes will be output to the channel set here.

Lim: Notes within this range are mapped to chords. Notes outside the range are passed through unaltered.

Trp: The output chords are transposed by the amount set here (+/- 99 semitones).

Example: if you map C to the chord CEG and set transpose to 1, then C will be mapped to C#FG#.

Key: The entire chord map is transposed by the amount set here (+/- 99 semitones).

Example: if you map C to CEG and set key to 1 then C# will be mapped to C#FG#.

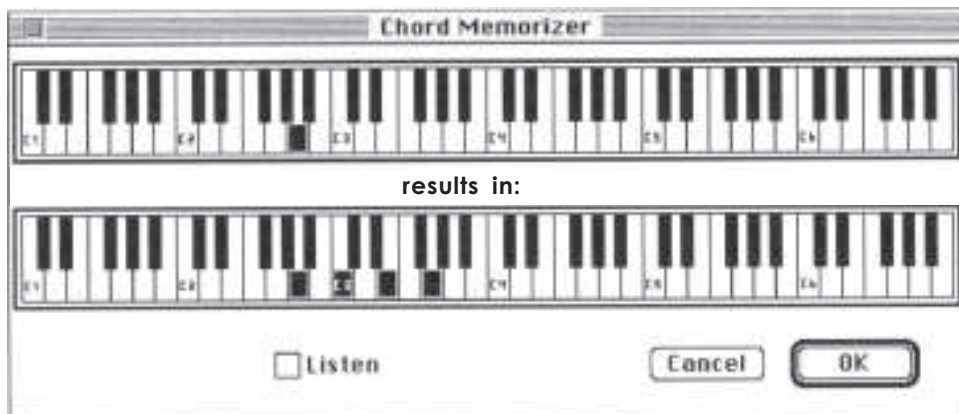
Example: if you want to map single notes to triads within a given key, first set the key then define each of the triad chords (see below).

When you want notes mapped to the triads in a different key, just change the key parameter.



Chord Memorizer Window

Double-clicking on the chord memorizer opens the chord memorizer window, where you can set up the chord definitions



The top keyboard is for inputting/displaying the incoming note, and the lower keyboard is for inputting/displaying the assigned chords. Notes can be input using the mouse, or a MIDI keyboard.

Remember to use notes on the top keyboard within the *Lim* range. Also remember that you can only define one chord for each of the 12 pitch classes (i.e. C, C#, D etc.). You *can* of course, use more than one chord memorizer.

Entering Chords with the Mouse

First click the input note on the top keyboard. All notes in the associated chord will become inverted on the bottom keyboard.

Next, click notes on the bottom keyboard until the inverted notes make up the chord you want. You can click notes either on or off on the bottom keyboard. Also, you don't have to stay within the same octave as the input note.

When you have the chord you want, select another input note, or close the chord memorizer window.

Entering Chords from Your MIDI Keyboard

First play the "input note".

Next click the *Listen* checkbox. The bottom screen keyboard is inverted to indicate it is ready to record.

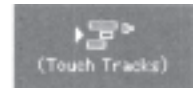
Now play the notes in the desired chord. (You can play them one at a time or as a chord.) If you make a mistake, uncheck the *Listen* checkbox, then check it again to start over.

When you have the chord you want, uncheck the *Listen* checkbox and play a new input note, or close the chord memorizer window.

Touch Tracks

Touch Tracks turns Logic into an interactive *live* sequencer. A touch tracks object allows you to trigger MIDI sequences or folders with single notes. (You can not use Touch Tracks to trigger audio.). (In the following section, any references to sequences means folders as well).

To create a Touch Tracks object, drag a sequence or folder from the Arrange window into the Environment window or select **New > Touch Tracks**“. To use a Touch Tracks object, simply assign it to an Arrange window track, or put it anywhere in the MIDI signal path.



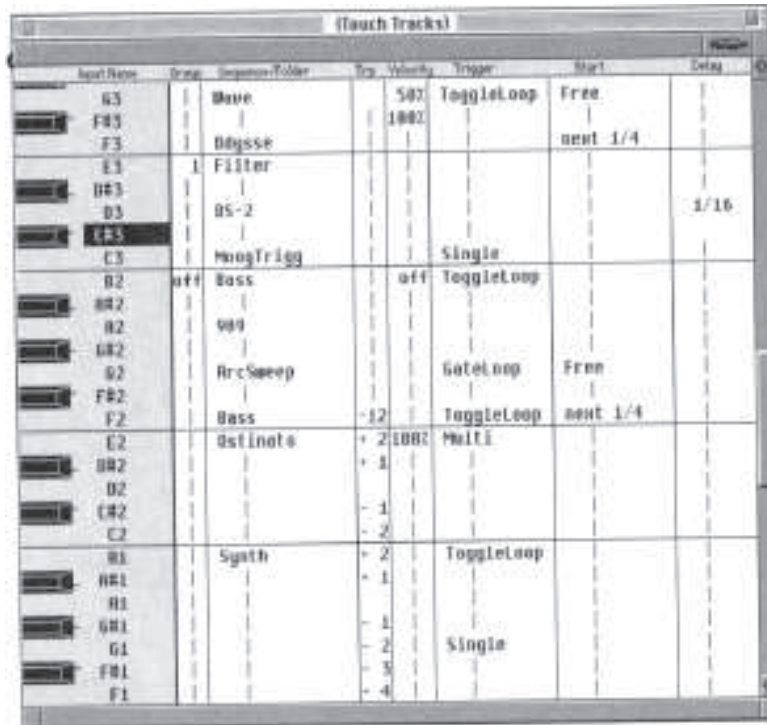
Here are some things to remember:

Only the Touch Tracks input has meaning-the trigger notes must appear here. Although there is an output triangle, it has no meaning-no events ever appear here.

Sequences triggered by Touch Tracks play exactly as they would from the Arrange window. In particular, they play back through the instruments assigned to their tracks.

Logic's Transport must be running for Touch Tracks to work.

The Touch Tracks



Input Name	Group	Sequence/Folder	Try	Velocity	Trigger	Start	Delay
G3		Bass		500	ToggleLoop	Free	
F#3		Odysse		1000		next 1/4	
E3	1	Filter					
D#3		BS-2					1/16
D3		HangTrigg			Single		
C3							
B2	off	Bass		off	ToggleLoop		
B#2							
A2		999					
G#2		ArcSweep			GateLoop	Free	
F#2		Bass	-12		ToggleLoop	next 1/4	
F2							
E2		Ostinato	+2	1000	Multi		
D#2			+1				
D2			-1				
C#2			-2				
C2							
B1		Synth	+2		ToggleLoop		
A#1			+1				
A1			-1				
G#1			-2		Single		
G1			-3				
F#1			-4				
F1							

You open the Touch Tracks window by double-clicking on the the Touch Tracks object. This window is set up similarly to the mapped instrument window (see page 5-37) - the input note is selected using the keyboard on the left, and the output sequence assignment and parameters are set in the columns of the corresponding row,

Sequence Note Assignment

Dragging a sequence into the Environment initializes a Touch Tracks object, so that initially all notes trigger this sequence. (Middle C plays it at its original pitch, and all other notes transpose it relative to middle C.)

In the **Input** column, you can see the input notes, and to the right, in the **Sequence/Folder** column, the names of the assigned sequences or folders. On the vertical keyboard to the left, you can select individual notes or whole pitch ranges, by grabbing and dragging. If you then drag a sequence from the Arrange window, it will only be assigned to the selected note(s).

Remember that with the following parameters, all settings also apply to multiple selections.

A **vertical** gray line means that the setting is the same as the line above. If you change a vertical gray line that is above another vertical gray line, the lower one will change to display its previous value. (I.e. it will no longer be the same as the line above.)

Group

To the left of the sequence name is the group column. Groups work the same way as they do in the Hyper Editor—when you trigger a sequence, any other sequences in the same group that were playing will stop.

The *off* setting means that the sequence is not assigned to any group.

Transposition

When you drag a sequence to a key in the Touch Tracks window, that key triggers the sequence at its original pitch (i.e. without transposition). If you want the sequence transposed, you can set the amount in the **Tip** column (± 99 semitones).

If a key range is selected when you drag the sequence to the Touch Tracks window, then incremental transpositions will automatically be set for adjacent keys.

When you create a Touch Tracks object by dragging a sequence into the Environment, C3 triggers the sequence at its normal pitch, and all other keys trigger it transposed relative to C3.

If you do not have Display *Middle C as C3 (Yamaha)* checked in the **Options > Settings > Display Preferences** window, then C4 rather than C3 will trigger the sequence at its original pitch. This will actually be the same MIDI note event, but Logic will use the Roland naming convention of C4 for middle C, rather than C3.

Velocity Sensitivity

In the **Velocity** column, you can set how much the velocity values of the played sequences are affected by the velocity value of the trigger note: by 100% (meaning a lot), 50% (meaning a little) or *off* (meaning not at all).

Object Reference

Trigger Modes

The Trigger column sets how sequence playback is handled:

<i>Multi</i>	Playing the trigger note starts the sequence. Playing it again starts the sequence again, without stopping the original version.
<i>Single</i>	Playing the trigger note once starts the sequence. Playing it again stops playback, and starts the sequence over.
<i>Gate</i>	The sequence plays until the trigger note is released (or until the sequence ends).
<i>Gate Loop</i>	The sequence loops until the trigger note is released.
<i>Toggle</i>	Playing the trigger note starts the sequence. Playing it again stops playback.
<i>Toggle Loop</i>	Playing the trigger note starts the sequence looping. Playing it again stops playback.

Start

In this column you can quantize how sequences start and stop.

Free means there is no quantization. The *next* 1/16, 1/4 or 1/1 settings start or stop the sequence at the next 1/16 or 1/4 note or at the start of the next bar, after a trigger note is input.

Delay

In this column you can assign the sequence a delay. As in the sequence parameter box, the delay is set on the right side of the column in ticks, or on the left side in note values.

You can use *Delay* and *Start* together to make sequences start at any position in the measure. E.g. Set *Start* to 1/1 and set *Delay* to 480 ticks to start playback at the second 8th note in the measure.

Physical Input/Sequencer Input

The Physical Input represents the MIDI inputs on your Macintosh MIDI interface; the Sequencer Input symbolizes Logic's MIDI input. You can (and should) have only one of each of these objects in the Environment.

To create an input object, select **New > Physical Input** or **>New > Sequencer Input**. To make use of an existing object, drag it onto the relevant layer. This will not affect its cabling.



Physical Input

The physical input represents the MIDI inputs on your Macintosh MIDI interface. Each Song can have only one physical input object. You only need to create a physical input object if you have deleted the one that was there when the Song was created. In this case, select **New > Physical Input**.

The physical input object is where the MIDI signals are received from the inputs of the MIDI interface(s) connected to your computer. This object has a total of 65 outputs. The outputs labeled M1 to M32 carry the MIDI data from the first 32 individual inputs of the interface(s) connected to "MIDI connection 1" and "MIDI connection 3", as set up in **Options > Settings > MIDI Communication Preferences**. The bottom 32 outputs on this object are labeled P1 to P32, Those carry the first 32 individual inputs of the interface(s) connected to "MIDI connection 2". Depending on the type (and age) of your computer, these interfaces can be connected to the modem or printer port, or to the *universal serial bus* (USB) The connection assignment is set up in the *communication preference*.



The top output (SUM) carries the MIDI events for all of the individual outputs which are not cabled separately.

MIDI Remote Control

Remote control events are intercepted at the physical input, and are not passed through to its outputs. Therefore, remote control events will not reach the Environment, or be recorded

on Arrange window tracks. If you want to trigger a MIDI Remote Control action from a fader or transformer, cable that object's output back into the physical input object.

Define your own MIDI Remote Commands, using the Key Command window. (section Assigning a Function to a MIDI Message on page 1-47).

Sequencer Input

MIDI events arrive at Arrange window tracks thru the sequencer input object. If nothing is cabled into the sequencer input object, nothing can be recorded in Logic. Typically, the physical input is cabled directly to the sequencer input but other objects can be inserted between them. Typical candidates are a monitor, a keyboard object or a cable switcher feeding various effects such as arpeggiator, a delay line, etc. (See the *Clicks & Ports* layer of the Logic default Song.) You can also cable the output of complex Environment processes back into the sequencer input object in order to record the processed signal.

Each Song can have only one sequencer input object. You only need to create a sequencer input object if you have deleted the one that was there when the Song was created. If you do need to create one, select **New > Sequencer Input**.

MIDI Metronome Click



The MIDI metronome click object is used to create note events at bar, beat and division intervals. There is a button on the Transport to turn the MIDI metronome click on and off. You can also set the MIDI metronome click to activate during recording and/or playback in the **Options > Settings > Recording Options window**.

Each Song can have only one MIDI metronome click object. You only need to create a MIDI metronome click object if you have deleted the one that was there when the Song was created. If you do need to create one, select **New > MIDI Metronome Click**.



The checkboxes after *Bar*, *Beat* and *Division* allow you to generate note events separately for bars, beats and divisions. The *Cha*, *Note* and *Vel* parameters define the MIDI channel, note # and velocity of the generated notes. The line directly below the *Cha* parameter for *Bar* (*parameter Port*) allows you to set a direct output port for the metronome.

Internal

“Internal” objects include instruments that are located inside your Macintosh (such as a Nubus card, PCI card and/or software synthesizers). Although communication with sound sources inside the computer does not actually use MIDI, it still conforms to the MIDI communication protocol. Internal objects have an input but no output. They therefore act like extra output ports which are hard-wired to the relevant sound generator.

Using internal objects

If you want to address internal sound sources, you must first create an equivalent internal object (see below). Then create a standard instrument, multi-instrument or mapped instrument for the device. Next connect the output of the instrument to the internal object. Then use the instrument as you normally would.

Use the Which parameter to select which (up to 16) internal device an internal object represents.

Digidesign SampleCell

Digidesign’s SampleCell II is a 32-voice sample player on a PCI or Nubus card. To create an internal object for the SampleCell II card, select **New > Internal > SampleCell**.

The options for the SampleCell II card are similar to those of any good stand-alone sampler. Because there are no delays due to MIDI transfer or decoding, the timing is far superior to that of external MIDI sound sources. There should be no need to set any delay for this multi instrument.



Apple QuickTime

The QuickTime system extension is a software GM sound generator. The "QuickTime™ Musical Instruments" file must be present in your System folder for it to work.

To create a virtual representation of the QuickTime synth, choose **New > Internal > QuickTime**.

This software synth is 16-part multi-timbral. As usual, you can switch sounds via program changes; these are arranged according to the GM table. Channel 10 is always set to drum sounds, with GM key assignments. The QuickTime synthesizer reacts to notes and program change commands, as well as to pitch bend (+/- 2 semitones) and the main volume controller (#7).

The first note is used to initialize the relevant MIDI channel of the QuickTime synthesizer and therefore is not played.

Notes

Don't expect too much in terms of sound quality from the QuickTime synthesizer. It is not intended for use in professional productions.

The QuickTime synthesizer has a noticeable delay, that will vary according to your hardware, and the number of initialized MIDI channels. You can compensate for the delay by setting negative *delay* values in the object parameter box of the relevant (multi) instrument.

Morning Star MacWavemaker

Logic can control Morning Star's MacWavemaker soundcard directly. You can create an object to address the MacWave maker in the Environment window by selecting **New > Internal > WaveMaker**. All other details of setting up a multi instrument to control it are the same as with Sample Cell.



Retro AS-1, Unity DS-1

These objects represent direct outputs to the Retro AS-1 software synth and the Unity DS-1 software sampler, if these are installed in your computer. The objects can be regarded similar to a MIDI output port, but supply a direct internal connection to those software sound sources. Therefore, it's not necessary (but still possible) to use OMS if you want to address the AS-1 or the DS-1 from within Logic.

AutoLink Object

This object is designed to supply a direct internal connection to other MIDI applications which support the AutoLink standard that is used by Logic and SoundDiver.

Modem/Printer Port

Modem or printer port objects represent the MIDI output(s) of a MIDI interface connected to the modem or printer port.

Please note: These objects are not available if you use a newer Macintosh computer with USB connections (see below).

To create a port object choose **New > Modem Port** or **New > Printer Port**.



You can use the lowest line in the parameter box to set the MIDI output (1-99) for the port object. The signals from all the MIDI cables which you plug into this object will then be sent via the MIDI output you've chosen. The All setting means that the incoming MIDI signals are sent simultaneously to all outputs of the MIDI interface,

If you want to address several MIDI outputs separately, you have to use an interface which is compatible with Mark Of The Unicorn's MIDI Time Piece (like Emagic's Unitor8, Mark of the Unicorn's MIDI Express or Opcode's Studio 4 or 5). Only this kind of interface will allow you to activate the ***MTP Cable Mode*** checkbox (**Preferences > Communication**). All this,

however, only applies to "old" interfaces which are connected to the modem or printer port of older Macintosh computers.

There is no separate port object for USB (Universal Serial Bus) connections. However, Emagic recommends (even for modem and printer port connections) not to use these port objects any more, but instead use a standard instrument object as a port object, set its Port parameter accordingly (see section Direct Output Assignment on page 5-12) and connect the desired cables to it.

Remember that you can also address any of your MIDI output ports directly from any of the instrument objects (and the GM mixer object as well).

MIDI interface with only one MIDI port

If your MIDI interface has just one MIDI port set the port object to *MTP port 0* or *All*. You still have the option of connecting a second independent MIDI bus to the other Macintosh port (printer or modem).

Aliases

You can create an alias of any Environment object, and it will behave exactly like the original. In the case of faders, the alias has its own value which can be different than the value of the original. This is particularly useful in the case of text faders which take up a lot of memory-if you need several text faders with the same names, use one original and make the rest aliases.

To create an alias of any object, first select the object, then select **New > Alias**.

Aliases can be reassigned using meta events, and there is a special fader for accomplishing this which works something like the cable switcher. You can create an *alias assigner* by selecting **New > Fader > Specials > Alias Assigner**. This will create a meta fader whose -1- definition is set to 46.

To use the alias assigner, cable its top outlet to the alias, and cable subsequent outlets to the various originals you want to assign to the alias. If you set the alias assigner's maximum range to the number of original's, minus 1 (we're counting from 0 here), you can then connect the next cable to a new alias and subsequent cables to originals for that alias. In this way, the alias assigner can be used to assign multiple aliases at the same time.

One situation where you might use an alias assigner, is to switch a delay line alias between different original delay lines with different delay times. This is one way of simulating MIDI control over delay time.

Alias Parameters

Although aliases share the parameters of the original objects from which they are made, they also have their own, special parameters:

Icon & Instrument List Checkbox

As usual, the checkbox next to the word *Icon* determines whether the alias appears on the Arrange window's Instrument List, and you can assign any of the usual icons to it.

Ref

The *Ref* menu is used to select the original of which the alias is an alias. It is the same as the Arrange window's Instrument List.

Channel

For everything except faders, the channel parameter causes all events leaving the alias to be channelized to the selected channel. (If the setting is *All*, then their channel will be unchanged.)

For faders, the channel parameter causes the alias to act as if the original fader's *In* and *Out* definitions both were set to that channel. (Again a setting of *All* results in no change.)

Share Name/Share Size

These two parameters do just what they say-when checked the alias shares the size and/or name of the original, and if not, it can be resized and given its own name.

Copying Aliases

If you copy an alias by itself, you will get a new alias of the same original. If you select an alias and its original and make a copy, you will get a copy of the original with its own alias.

Ornaments

You can create an ornament by selecting **New > Ornament**. Ornaments are simply solid backgrounds behind other Environment objects. An ornament never “covers” the objects behind it, and you can still select multiple objects by *rubber-banding*.

Macros

As mentioned above, macros are not actually Environment objects at all, they are collections of other Environment objects, together with their cabling. To create a new macro, first select all the Environment objects to be included, then select **New > Macro**. Logic will enclose all the selected objects in a frame with a gray background. The macro will automatically be named after its input and output objects, separated by a dash (see below).

You can create a “protected” macro by holding  while creating the macro. Protected macros *can not be unpacked* so be sure that you make a copy of the objects and cabling before creating a protected macro.

Macros have a lot in common with standard Environment objects-you can connect cables to and from them, they can be resized, they have their own parameters and icon, they can be placed on the Arrange window's Instrument List and they can

be copied or dragged between Environment layers (including between different songs).

Macro Input and Output

Since a macro is a collection of objects, individual objects need to be specified as the macro's input and output. This can be done in two ways: by name or by default. If you name one object "Macro-In" and name another object "Macro-Out" these will automatically become the macro's input and output. If there is no object named "Macro-In" then the upper-leftmost object will become the macro's input. If there is no object named "Macro-Out" then the lower-rightmost object will become the macro's output. Cables leading into the macro will deliver their events to the macro's input object, and cables leading from the macro's outlets will carry events leaving the macro's output object.

Macro parameters

Macros have four parameters:

Icon & Instrument List Checkbox

As usual, the checkbox next to the word Icon determines whether the macro appears on the Arrange window's Instrument List. If you use a macro as a track instrument, events played on the track, or incoming MIDI events when the track is selected, will be sent to the macro's input object (see above).

If you place a macro on the Instrument List, you can choose any of the available icons for it, using the pop-up menu as usual.

Auto Fader

If you select a group of faders, and create a macro from them, checking this box will make them act as if they were cabled serially.

Show All

This checkbox determines which objects will remain visible when collected in a macro. When it is checked, all objects will be visible (assuming the macro is not resized to hide some of them). When it is unchecked, only faders, monitors ornaments and keyboard objects will be visible. (You can toggle this checkbox at any time.)

Special Macro Features

Unpacking

You can turn any macro back into its component objects and cables, by double-clicking in any empty region of the macro object.

Size Limitations

Macros are limited in size-the limit depends on the memory taken by the individual objects within the macro. Typically this limit is between 100 and 200 objects.

Nesting Macros

You can nest macros-i.e. a macro can contain other macros as objects.

Cables Leading Outside of the Macro

If some of the objects you select when creating a macro have cables leading to objects not selected, these cables will be deleted when the macro is created, and a warning dialog will notify you of this. In this case, the macro is made from a copy of the original selection of objects and the original collection of objects remains unchanged. If no cables lead to unselected objects then the macro is made from the original objects.

5.5 Faders

To create a new fader, select **New > Fader > ...** A sub-menu appears, where you may select the style of fader you want. There's also a sub-menu at the bottom named *Special*, for selecting various special types of faders (e.g. cable switcher, meta-fader, etc.).



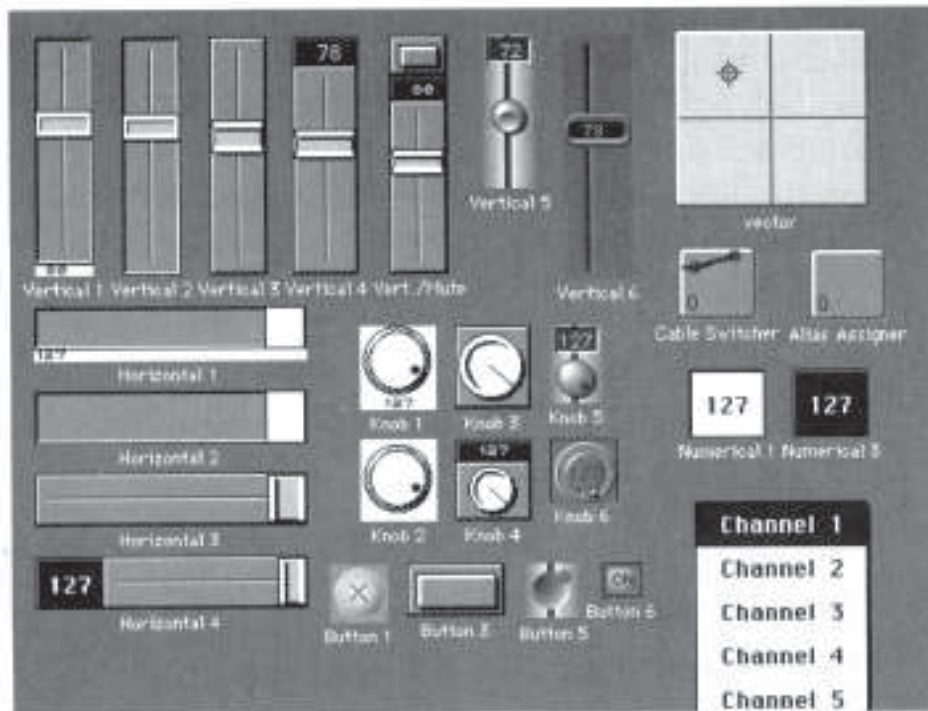
The style of a fader is how it looks on screen (i.e. slider, knob, menu, numerical...) and how it responds to the mouse (i.e. click & drag, double-click & type a number, click & select from a menu...). The *Auto* style fader, which is the default style if you create a fader with a key command, changes styles as you change its shape and size.

The *type* of fader determines what events the fader sends out and responds to. In two cases: the cable switcher and the alias assigner, the fader doesn't send out events at all. The cable switcher routes events to its different outlets, and the alias assigner changes the references of fader aliases.

In general, a fader's style is completely independent of its type. A cable switcher can look like a button, a knob can send out MIDI or meta or Sysex events, etc. The one exception is the Vector style fader-these behave in a special way by sending out two (or in special cases, four) messages at a time, depending on the mouse location within their 2-dimensional, vector window.

Don't worry about selecting the wrong fader style or type from the *New* menu-you have complete control of a fader's style and type from its parameter box.

The illustration below shows examples of each of the fader styles. Notice that some of the styles have numerical displays. When there is a numerical display, you can double-click it to enter exact values by typing. Otherwise, when you scroll any of the sliders or knobs, the series of values that are sent out depends on the fader size and scrolling speed-i.e. not every consecutive value is sent out. (If you click & drag slowly on the numerical you can get all consecutive values.)



The text-style fader on the right, behaves like a pop-up menu. If you double-click it a window will open, where you can type in the names for each of the menu positions. There is also a checkbox so that you can make the names scroll, rather than choosing from the menu.

The vector-style fader (upper right) allows you to scroll in two directions, and it sends out two values, one corresponding to the vertical position and the other to the horizontal position.

The *cable switcher*, shown below the vector, routes all incoming events to the selected output. As with most objects, cabling one outlet automatically produces another. Clicking the cable switcher causes it to change to the next output.

Operation

Normally you grab the surface of a fader with the mouse pointer or pencil, and drag it vertically or horizontally, depending on what style it is. Knobs can be dragged either vertically or horizontally.

When there is a numerical display, you can grab this number and use the mouse as a slider. The advantage of this is that if you drag slowly, it will move in increments of 1, regardless of the size of the object. When using the fader itself as a slider, values may jump in larger increments, depending on the size of the fader and speed of dragging. You can also change the values in steps of 1, by dragging on the slider or knob while holding down .

Buttons send out only two values, the lowest and highest of their range, which you may set in the parameter box. You change a button's state by clicking it.

Working with Object Groups

When building a virtual mixing desk or synthesizer control panel in the Environment, you often have to deal with large groups of fader objects, which have the same size, regular spacing and/or a similar definition. To save time with the definition and alignment of these groups you can define (one or more) objects *as prototypes (templates)* by copying them onto the clipboard (**Edit > Copy** or ). You may then apply certain characteristics of the *template* to selected objects.

Size and Alignment

The function **Options > Apply Buffer Template to > Size**, allows you to transfer the size of the prototypes to the selected objects.

To align a group of objects, choose 2 objects in a row, or 2 objects in a column, or 4 objects in 2 rows and 2 columns, whose horizontal and vertical alignment you want to transfer to the group, and copy them into the clipboard. You can then transfer this alignment by selecting the function **Options > Apply Buffer Template to > Position**, and the selected objects will be positioned at the top left corner, in the same way as in the prototype.

Options > Apply Buffer Template to > Position and Size combines both the above functions.

Definition

The function **Options > Apply Buffer Template to > Definition** transfers the parameters of a copied prototype to all selected objects. If several prototypes of the same type are available, the one with the most similar size is used.

...> **Definition, channel increment** increases the channel number from object to object, beginning with the top left object. It is not necessary for the selected objects to have the same In or Out definition as the prototype.

...> **Definition, number increment** increases the first data byte of the definition (e.g. controller #).

Cabling serially

Using **Options > Cable serially**, you can connect all selected objects in series, beginning with the object at the top left.

Names with Numbers

If you select a group of objects then label one of them with a name that ends in a numeral, the remaining objects will be renamed with the same name, but increasing numerals. (E.g. Selecting several objects and naming one of them "Object 1" will result in the names "Object 1", "Object 2", "Object 3", etc.)

Special Functions for Faders

Grouping Faders

If you select several faders and move one of them, all the selected faders will move proportionally. As long as all the faders remain selected, their proportional positions will be retained (even after one or more of the faders has reached its minimum or maximum position).

Moving the fader group with **Alt** pressed, changes all values linearly (the absolute value differences are maintained).

Moving the fader group with   pressed, changes all faders to the same value.

Sending Fader Values

You can use the functions **Options > Send All Fader Values** and **...> Send Selected Fader Values** to make all fader objects or all selected fader objects send their current values. Using this function in conjunction with Logic's *Record/Pause* mode, allows you to record a "snapshot" of all current fader positions. This is especially useful for virtual mixing desks, or synthesizer panels. (*Send All Fader Values* only sends values for faders whose outputs are cabled to other objects.) The option *Send All Fader Values after loading* in the **MIDI Options** page of the **Song Settings** causes all fader values to be automatically sent after a song is loaded.



Resetting Fader Values

With **Options > Reset Selected Faders** all selected faders are set to zero and transmitted.



Recording Fader Movements

You don't need any special cabling to record the data generated by a fader. All data generated by faders is recorded on the selected track when Logic is in *record* or *record/pause* mode. The recorded events correspond to the *out definitions* of the faders.

Playing Back Fader Movements

Any fader will react to incoming events matching its *In definition*. The fader must, of course, be in the MIDI signal path. Typically you accomplish this by cabling the track instrument into the fader. For purposes of automated mixing, it is useful to create a new (standard) instrument just for the purpose of automating the faders. Ensure that the instrument (which you might conveniently name *playback*) has no direct MIDI output, that its channel setting is *All* and that it is cabled into the first fader in any chain of faders cabled serially.



Mixer Automation

There are extended automation functions for all faders in the Environment. As with mixing desk automation, there are three modes: Merge, Replace and Update. Using these three entries under **Mixer Automation** > . . . , existing fader movements can be altered in real-time.

Merge

Newly-written, data is merged with existing data. (This is the mode used in Logic versions prior to 3.0.)

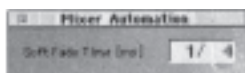
Update

In this mode, only changes in the fader values relative to existing data are recorded.

Replace

In this mode, existing data is replaced by the new data. The data is replaced for as long as the mouse button is held down.

Soft Fade Time



When you release the mouse button after carrying out a replace operation, there is a smooth transition between the current fader value and the recorded value, as determined by the *Soft Fade Time*. You can set this using **Options** > **Mixer Automation** > **Soft Fade Time...** This purpose of this is to ensure that there is no sudden jump in level.



Setting the Automation Mode

You can set the automation mode in either the Record Switch menu, or from the Environment (**Options > Mixer Automation >...**).

Correcting the Automation

If you want to change recorded fader values:

Select the sequence(s) containing the recorded fader values.
(Other MIDI data in these sequences will not be affected.)

Make sure you have chosen the desired automation mode.

Start Logic recording.

Move the fader(s) whose values you want to change.

(Remember that in *Replace* mode, the old values will be replaced for as long as the mouse button is held down.

Fader Styles

The fader's style is shown in the line under the icon. It can be changed by clicking on the current style name, and selecting the new style from the resulting flip-menu.

Remember that a fader's style does not usually affect its function-i.e. you can select the most convenient style for the desired application (see page 5-68).

Here are notes on some of the fader styles:

Vertical/Mute



This is exactly like the *Vertical 4* fader style, with an added mute button. When you press the mute button, the fader sends an event matching its Out definition with value 0. While the mute button is down, the fader's movements are not sent. When you release the mute button, the current fader value is sent.



Numericals



When a fader has a numerical display, you can double-click the numerical and type in a new value. If you use the numerical for slow scrolling (i.e. click and drag slowly on the numerical), the fader values will change in steps of 1.

Buttons

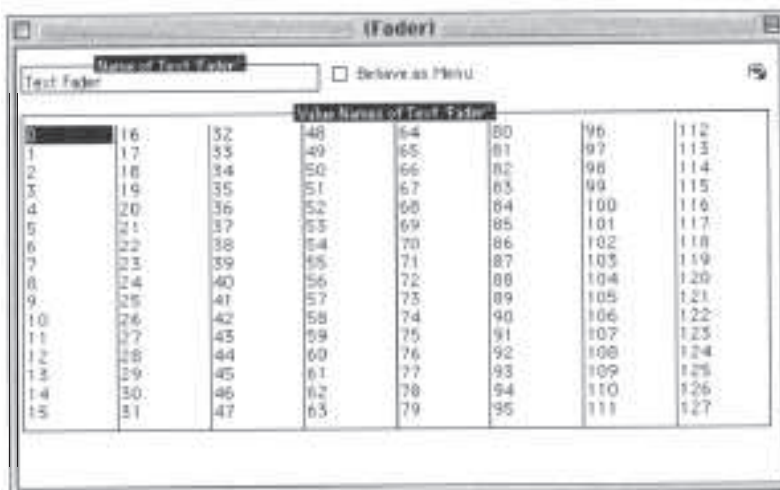


Button style faders send only two possible values corresponding to the minimum and maximum of their range. When you click the button on (hollow or X'ed in the illustration), the maximum value is sent, and when you click the button out (filled or not X'ed in the illustration) the minimum value is sent. If the minimum and maximum range parameters are set to the same value, then the button sends out this value each time it is clicked.

Text

Text faders function like *Numerical* faders, but can display text for each of the 128 possible MIDI values (0 thru 127).

Double-clicking on the surface of a text fader opens the text fader window.



Clicking on a position in this window will cause the corresponding fader value to be sent. (This is similar to selecting programs by name in the multi-instrument window.) Double-click on a position in this window to type in the text for that position. By default (i.e. when you create a text style fader) the text positions contain the numerical values. (You can use this feature to create numerical style faders with colored backgrounds.)

Clipboard Functions

The flip menu located at the top right of the text fader window provides cut, copy, and paste clipboard functions. You can use this to transfer the entire list of text names to a text editor (e.g. a marker text window in Logic), for more convenient editing. Note that empty lines, and lines containing only spaces are ignored, so to insert blank, positions in the list, use  .

Behave as Menu

Check the *Behave as menu* box to cause the text fader act as a flip menu. When this box is not checked, the text fader acts as a scrolling menu.

Text Fader Range

A text fader will only allow you to enter as many names as are in its range. (E.g. if you set a text style fader's range to 0,1 you can only enter two values.) You should always set the minimum range necessary for a text fader, as this saves memory. In any case, remember that the first name corresponds to the lowest value in the range (not necessarily 0) and the last name corresponds to the highest value (not necessarily 127).

If you force a text fader to a value outside its range (i.e. with MIDI input) it will display "---" for values below its range and "+++" for values above. The one exception is when the fader's range is 0,1 in which case all values above 1 will display the name for value 1.

Vector-Style Faders



Vector faders function like joysticks—they can be moved in two dimensions: up/down and right/left. Each dimension generates to its own MIDI events, so that each time you change the position of the crosshair with the mouse, two MIDI events

are sent.

Most faders have In and Out definitions (see page 5-78) which determine what MIDI events the fader sends (Out) and reacts to (In). The vector style fader replaces these with *Vert* and *Horz* definitions, which determine what MIDI events correspond to vertical and horizontal motion. (If you cable corresponding MIDI events into the vector fader, its crosshair display will update accordingly.)

4-Channel vector mode

If you set a vector fader's *Vert* and *Horz* definitions to the same MIDI event (e.g. the same MIDI controller & channel), the vector fader will send out four MIDI events each time the crosshair is moved. These will be the same MIDI event (e.g. controller) on four consecutive MIDI channels, starting with the channel set in the *Vert* definition.

The values of these MIDI events will correspond to the closeness of the crosshair to the vector fader's four corners—upper-left for the lowest channel then upper-right, lower-left and lower-right respectively for the next 3 channels. In the center, all channels receive value 32. At the corners the corresponding channel receives value 127 and all remaining channels receive value 0. (This is with the default range of 0 to 127. If you alter the range, the center and corners will behave differently but mysteriously, the four values will always add up to 12.5).

Fader Functions

MIDI Events

Each fader has an In and an Out definition. The In definition determines the kinds of MIDI events that can remote control the fader (i.e. the types of events it reacts to). The Out definition determines what kinds of MIDI events the fader sends out. (A fader can, therefore, convert one type of MIDI event to another.)

Most MIDI events consist of three bytes. The first one indicates the type and channel of the MIDI event (e.g. a note on channel 3). The second one indicates the first *data value* (e.g. the pitch in the case of note events) and the third one indicates the second *data value* (e.g. velocity in the case of note events). A few MIDI events use only two bytes (program change & aftertouch). Although faders (and transformers) always provide for 3 bytes, the second one is thrown away for these special, 2-byte messages.

Fader parameters are provided for setting the message type, MIDI channel and the first data value. (Note that the message type and MIDI channel are actually combined in the resulting MIDI event.) The second data value is determined by the fader setting, or in the case of remote control, by the incoming MIDI event.

Here are the In and Out definition parameters:

Out (or In): defines the event type (use the flip menu shown in the diagram).

Channel (1-16): defines the MIDI channel of the event.

-1- (0-127): defines the first *data byte* of the event. In some cases like pitch bend this is an actual data value. In other cases like MIDI controllers, this indicates the controller type (e.g. volume, pan, etc.). In still other cases like aftertouch, this byte is unused.

Here is the meaning of the -1- parameter and fader position for each of the In and Out definition choices:

Note On

The -1- parameter sets the pitch and the fader position sets the velocity. This is most useful as an In definition for trapping specific notes and converting them to other MIDI events, or simply monitoring their velocity. (If you move a fader whose Out definition is set to Note On, then a note-off MIDI event will immediately follow the note-on. This might be useful for creating on-screen drum pads from button-style faders, for example.)

P-Press

The -1- parameter sets the pitch and the fader position sets the Poly Pressure amount (i.e. the key pressure).

Control

The -I- parameter sets the MIDI controller number (i.e. the controller type) and the fader position sets the controller value. The controller type can actually be selected by name, using the flip menu that appears when you click and hold the -1- parameter.

Program Change

The -1- parameter is ignored. (In earlier versions of Logic, this sets the program bank.) The fader position determines the program number.

C-Press

The -1- parameter is ignored. The fader position sets the channel pressure (a.k.a. aftertouch) amount.

PitchBd

The -I- parameter sets the pitch bend LSB and the fader position sets the MSB. Typically you would set the -1- parameter to

0 and use the fader to control the “coarse” pitch bend amount. (No pitch bend corresponds to a -1- setting of 0 and a fader position of 64).

The Sysex and *Switcher/Meta* functions are described later (see both the section *Sysex Fader* on page 5-86 and the section *Cable Switchers* on page 5-83).

Range

The range parameter contains two numbers-the left one sets the lowest possible fader value, and the right one sets the highest. (Note that these limits can be exceeded using MIDI remote control.) When the fader style is a button, the range determines the in and out position values of the button.

For text style faders, the first name always corresponds to the low end of the range, and the subsequent names correspond to incrementing range values up to the top of the range. The number of names you can enter into the window is limited by the range.

Val as

This parameter determines the way the fader value is displayed numerically:

- Num:* the fader value is displayed directly as a number (0-127).
- Pan:* the fader value 64 is displayed as “0”; smaller values appear as negative numbers and larger values as positive numbers (-64 to 63).
- Hz, Oct, dB, ms:* These display formats are tailored to various DSP functions.
- Bpm:* An offset of 50 is added to the fader value. This displays the correct tempo settings for the special *Tempo Control* fader.

Faders

If none of the above formats is suitable, consider using a text style fader and entering the desired display values as text. Examples include percentages, note names and program names.

Filter

This parameter provides various filter options for MIDI events reaching the fader:

- Off:* All incoming MIDI events are allowed to pass through. All events that match the input definition are converted in accordance with the output definition.
- Other:* All MIDI events which do *not* correspond to the input definition are filtered. All the events that match the input definition are converted in accordance with the output definition, and allowed to pass through.
- Match:* All MIDI events which match the input definition are filtered, all the others are allowed to pass.
- All:* All incoming MIDI events are filtered.
- Thru:* All MIDI events coming from the Physical Input are filtered. This is the same as *off* for all events coming from Logic (i.e. from sequences or the Environment).

Use this filter mode to prevent MIDI feedback by blocking incoming MIDI events from being sent back out.

- Shot:* When the fader is moved with the mouse, only the final value (the value when the mouse button is released) is sent.
- 14Bit:* Used in conjunction with pitch bend, this allows Z-byte (i.e. fine tuning) pitch bend events. Used with controller messages, this causes the fader to send two MIDI controller messages—one for the MSB and one for the LSB.

For the 14Bit setting to work, the fader's *In* and *Out* definitions must be the same. For controllers, the MSB uses the In definition controller number and the LSB uses the controller number 32 higher. This conforms to the MIDI standard for sending 14-bit controller data.

Note that with this filter setting, the fader's range maximum can be set to 16383 and a fader value of 8192 represents no pitch bend.

Feedback

When the *Feedback* parameter is unchecked, the fader will automatically prevent feedback loops resulting from circular cabling. (I.e. the fader remembers when a specific MIDI event has passed through it and will not allow it to pass through again.) In some instances, you may want to allow this feedback-for example, to allow a MIDI event to change a *cable switcher?* position after it has passed through the switch. Checking the *Feedback* checkbox will allow this.

Switchers, Alias Assigners & Meta Faders



These are two types of faders that do not generate MIDI events. They share the same *Out* definition type-the one at the top of the definition menu. This selection will either display Switch or *Meta* depending on the current -1- value setting:

If the -1- value is 48 then the fader will be a cable switcher.

If the -1- value is 46 then the fader will be an alias assigner

If the -1- value is anything else then the fader will send out meta-messages whose type corresponds to the -1- value.

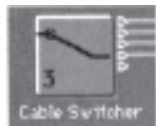
To create a cable switcher:, either select **New > Fader > Specials > Cable Switcher...** or change an existing fader's *Out* definition to *Switch/Meta*, and set its -1- value to 48.

To create an **alias assigner** either select **New > Fader > Specials > Alias Assigner** or change an existing fader's Out definition to **Switch/Meta**, and set its -1- value to 46.



To create a **Meta** type fader, either select it from the **New > Fader > Specials >...** menu, or change an existing fader's Out definition to **Switch/Meta** and set its -1- value to the desired meta event number (see below).

Cable Switchers



As stated above, cable switchers route events rather than generating them. Any kind of MIDI or meta event can be routed by a cable switcher. The only exception is events which match the cable switcher's In definition. These events will change the switch position (i.e. the routing) rather than passing through the switch themselves.

A cable switcher can have any fader style. In particular, using the text fader style allows you to label the switcher's various routes. Use the Auto-style to actually display the switch routing, as shown here.

A cable switcher can have up to 128 separate outlets-a new outlet is generated each time an existing one is cabled to something. You can click on an auto-style cable switcher to step through the outlets (including the last, uncabled one). Incoming events matching the cable switcher's In definition will change the switch position to match the incoming data value. (If the data value is greater than the number of switch positions, the last, uncabled outlet will be selected.)

Data values of 126 and 127 have a special effect:

An event with a value of 127 increases the output number. If you're on the switcher's last output when this event is received, you jump back to the first output. This is just like clicking on the fader.

An event with a value of 126 decreases the output number. If you're on the switcher's first output when this event is received, you jump forward to the last output.

Meta Event Faders

Meta faders generate special events called *meta events*, which are used to control certain Logic functions, but have no MIDI meaning, and are never sent to the MIDI output. In some cases (e.g. *Goto Screenset*, *Goto Song*, etc.), you do not need to cable meta faders to anything for them to work, but even in these cases you can use cabling to process **meta** events in the Environment, and therefore alter their effect. In the majority of cases (e.g. *Set fader range minimum*, *Bang!*, *Set TransformerOperation Minimum*, etc.) the meta-fader must be cabled to the object being affected.

Here's a quick summary of the currently implemented meta events that can be generated by faders:

46	Assign Alias
49	Goto Screenset
50	Goto Song
51	Goto marker
52	Stop playback
96	Set fader range minimum
97	Set fader range maximum
98	Set the fader value without sending.
99	Bang! Cause the fader to send its current value. The bang events data value can be used to cause the fader to increment-without-rollover (127), decrement-without-rollover (125) increment-with-rollover (123) or decrement-with-rollover (121). Use even values one less than those shown, to have the bang passed through to all connected faders.

Faders

- 100** Tempo Control (see the section section *The Real-time Tempo Fader* on page 15-14)
- 122** Set transformer map value for the currently selected map position.
- 223** Select transformer map position.
- 124** Set transformer condition maximum (bottom) parameter, if any. (This applies to all conditions not set to *ALL*)
- 125** Set transformer condition minimum (top) parameter, if any. (This applies to all conditions not set to *All*).
- 126** Set transformer operation maximum (bottom) parameter, if any. (This applies to all operations not set to *Thru*).
- 127** Set transformer operation minimum (top) parameter, if any. (This applies to all operations not set to *Thru*).

For more on Meta events 124-127, read the section *Remote Controlling the Condition and Operation Values* on page 5-46.

Goto Marker/Screen Set/Song

Selecting **New > Fader > Special > goto Marker** creates a fader which allows you to enter a marker number (Meta event 51), the Song position line then moves instantly to the marker with that number. Markers are numbered sequentially, throughout the song, even if you have renamed them from the default numbers they had when first created.

Selecting **New > Fader > Special > goto Screenset** creates a screenset fader (Meta event 49): you can then switch to the screenset with the number you select using the fader.

Meta event 50 has no effect within Logic, but you can use it to switch Songs on an external hardware sequencer.

Meta event 52 allows you to interrupt playback at any position.

To make the most efficient use of these faders, we recommend that you limit the value range to suit your particular circum-

stances, for example so that you limit the faders to operate on just the number of screensets, Songs or markers you're actually using.

Sysex Fader



The Sysex fader type (the last selection on the fader definition menu) is slightly different than the other fader types. It allows you to create a list of MIDI events that will be sent whenever the fader is moved or remote controlled. You enter the messages to be sent in a window that resembles Logic's Event editor, except that the time positions serve only to control the order in which the events in the list are sent - they are not sent at specific times, or with any delay between them.

The Sysex fader type was originally intended for sending MIDI System *Exclusive* (*SysEx*) messages-messages that are exclusive to individual manufacturers' MIDI devices. There are a number of built in features in the Sysex fader's window (see below) that facilitate creating Sysex messages. But, remember that Sysex faders can be used for any kind of MIDI event, and are therefore, very handy for sending ordered batches of messages, like mixer or control panel snapshots with one click of the mouse.

You can also specify Sysex as an In definition, but this has limited usefulness because incoming message (presumably Sysex) must be very short to be recognized. This is an artifact of the fact that Sysex messages can be of any length, and Logic must break them up in to small packets to avoid interrupting other MIDI activities.

Opening the Fader Event Editor window

When you set a fader's Out or In definition to Sysex, its event list window opens up. You can also open this window at any time by double-clicking on the word Sysex in the fader's parameter box.

Faders



An important thing to remember about the Sysex fader is that only those events that are selected when the event list window is closed will have their value altered by the fader value. Events which are not selected will be sent exactly as they appear in the window.

Those events that are selected will have the value in the *Val* column of the list altered if they are normal MIDI events (i.e. controllers, program change, aftertouch, etc.). For Sysex messages, you can set both which byte(s) are altered, and the format (e.g. MSB/LSB, BCD, nibbles, etc.) that the value will take. You can also specify a checksum format, if necessary.

Creating a Sysex message

There are two ways of entering Sysex messages in the Sysex fader's event list. One way is to have the fader "learn" the sysex string. To do this, you manually operate your synthesizer to have it send the corresponding Sysex message, using its onboard controls. Here's how:

If you haven't already done so, connect the MIDI output of the synthesizer to the MIDI input of your computer.



Click the MIDI In button in the top left. If the button is red, this means it is ready to receive incoming MIDI data.

Alter the parameter on your device. The corresponding Sysex message will be displayed and the MIDI In button will switch off automatically.

Another way is to type the sysex string into the event list. (Consult your MIDI devices manual for Sysex documentation.) Using this method qualifies you as a confirmed MIDI freak. Here's how:

Click the Sysex button with  held down. A generic Sysex message appears.



The first data byte in the top line (directly after the word Sysex in the *NUM* column) gives the manufacturer's ID. This may be several data bytes long (since there are more than 128 manufacturers of MIDI devices).

Usually the manufacturer's ID is followed by a device type ID, an individual device ID, a data type ID (which could be several bytes), if necessary the number of the multi-timbral sub channel, the identification number(s) of the sound parameters plus the value of the sound parameter. But there are no standards here-MIDI manufactures can and do adopt arcane schemes, presumably intended to make your life miserable (and protect the keys to the kingdom).

The data bytes are normally input as decimal numbers. To enter them in hex, place a "\$" in front of values when you type them in. **View > SysEx in Hex Format** allows you to see all the values in hexadecimal (for more on this, read the section on using Sysex displayed in hex, in Chapter 9).

The last data byte (the *EOX* indicating the end of the Sysex message) is entered automatically by Logic. The number of data bytes in the Sysex message can be decreased or increased by clicking one of the two arrows before and after the word < EOX >.

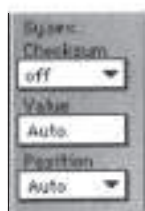


Although you can have any number of messages of any kind in the Sysex fader3 event list there is only one fader value, and all selected messages will adopt this same value.

Faders

 You can use a similar method to create any kind of MIDI or meta event in the Sysex fader window. Clicking any of the 8 event type buttons (note, program number, pitch bend, controller, channel pressure, poly pressure, Sysex and meta event) with  held down creates a new event of that type. (Meta events are created using the expanded view button with the 0's and 1's on it).

Setting the Sysex Data format



The terms "SUM" for the checksum and "VAL" for the fader value that's being sent are displayed within the Sysex string. The position and format of these bytes are set from the "Checksum", "*Value*" and "*Position*" flip menus. These settings may be found in the lower left-hand corner of the fader event list.

 Only *selected* Sysex events are affected by the above parameters, because only these events are affected by the fader value.

Checksum format

A checksum can be created in any of the following formats:

- Roland
- Yamaha
- Regular Checksum
- 2's complement
- 1's complement

 If you don't know which one works with your MIDI device, try "off" (= no checksum) first or "2's complement".

Value byte position

Position allows you to determine the position of the value byte. This position is specified in bytes counting from the end of the message: "last" refers to the position directly before the *EOX* byte, "Last-1" indicates the byte before that, etc.

"Auto" ensures that the value byte is inserted last in the Sysex string if no checksum has been selected, or -if a checksum value *has* been entered-that the value is inserted as the second to last byte.

Value byte format

Value can be set to the following:

<i>Auto</i>	The value is sent as one byte if the value range maximum is 127 or less. If the maximum is higher than 127, the value is sent as two bytes, the MSB (most significant byte) first.
<i>One Byte</i>	The value is sent as one byte.
<i>MSB/LSB</i>	The value is sent in two bytes, with the MSB (most significant byte) first.
<i>LSB/MSB</i>	The value is sent in two bytes, with the LSB (least significant byte) first.
<i>BCD 4 LSB</i>	The value is sent as “Binary Coded Decimal” in four bytes, in the order 1,10,100,1000.
<i>BCD 4 MSB</i>	The value is sent as “Binary Coded Decimal” in four bytes, in the order 1000,100,10,1.
<i>2 Nibbles L</i>	The value is sent in two nibbles, with the “least significant nibble” first.
<i>3 Nibbles L</i>	The value is sent in three nibbles, with the “least significant nibble” first.
<i>4 Nibbles L</i>	The value is sent in four nibbles, with the “least significant nibble” first.
<i>2 Nibbles M</i>	The value is sent in two nibbles, with the “most significant nibble” first.
<i>3 Nibbles M</i>	The value is sent in three nibbles, with the “most significant nibble” first.
<i>4 Nibbles M</i>	The value is sent in four nibbles, with the “most significant nibble” first.
<i>2 ASCII M</i>	The value is sent in two nibbles, the “most significant nibble” first; the nibbles are sent in ASCII format for the hex value. For example, the value \$7F (=127 in decimal) will be transmitted as a “7” and “F”.

3 ASCII M same as 2 *ASCII M* but in 3 nibbles.

4 *ASCII M* same as 2 *ASCII M* but in 4 nibbles.



The unused bits of the transfer in nibbles (X in 0XXXNNNN) are sent with the information at the relevant positions of the Sysex strings. If you want to transfer these deleted bits, you will have to enter zeros in the Sysex string.

Transforming Sysex messages

Logic only allows one Sysex string per fader. To transform one Sysex message into another, please use the following method:

Create a fader with the Sysex message you want transformed as the input definition. Select any controller you like as the output definition.

Create a second fader with the same controller as the input definition. Make the transformed Sysex message the output definition.

Connect the first fader to the second.

Remember that the incoming Sysex message must be very short for this to work properly. This is an artifact of the fact that Sysex messages can be of any length and Logic must break them up in to small packets to avoid interrupting other MIDI activities.

Meta Events for Faders



Meta are only used internally by Logic—they are not part of the MIDI signal flow, and have no MIDI meaning. But, they do have a similar form as MIDI events, and they are processed in the Environment.

Meta events serve several purposes in Logic. They are utilized extensively for formatting in the Score editor, and if you look at a scored sequence in the Event editor, you will see many meta events interspersed with the MIDI events that comprise the notes and controller messages.

Meta events are also used to affect the operation of faders and transformers in the Environment (see page 5-84). They can also be generated by faders and processed by transformers. The

only restriction is that meta events intended to affect transformers can not be modified by transformers. (Transformers can, however, convert MIDI events or other meta events in to ones that affect transformers, without being affected themselves.)

5.6 Environment Exchange

One of the main reasons for the Environment is to customize Logic to your MIDI studio. This can present a problem when sharing songs with other musicians, using different studio setups. It also presents a problem when you return to older songs after you've changed your studio. Logic offers several functions to make these changes as easy as possible.

Whenever you want to exchange Environments between songs, there is a source song containing the desired Environment, and a *destination* song whose Environment you want to change. The destination Environment must be in memory, and must be the active song (i.e. one of its windows must be active). The source song can also be in memory or it can be a file on your hard drive (or any other medium like CD, Zip, floppy, etc.).

If there are two songs in memory, Logic will assume the active song is the destination, and the other song is the source. If there are more than two songs in memory, Logic will assume the active song is the destination and the most recently active of the other songs is the source. If there is only one song in memory, Logic will present an *Open* dialog, for you to select the source song when you try to import an Environment.

Here are some tips on how to accomplish specific tasks:

Importing Single-purpose Environments

A single-purpose Environment *patch* might be an editor for a specific piece of MIDI equipment or an Environment patch for single MIDI processing task like a MIDI LFO, or complex

arpeggiator/delay line configuration. If the Environment patch is contained on a single layer, the simplest solution is to select **Options > Import Environment > Layer...**, and then select the desired layer in the dialog that appears. This layer and all of its objects will be inserted in the destination song, at the same layer position (i.e. the same place on the Layer's pop-up menu) as it occupied in the source song. Note that this inserts a new layer into the destination song-its original layers will be shifted as needed.

You can also move selections of Environment objects together with their cabling between songs by dragging or copy & pasting. This is often even simpler if you first collect the objects into a macro.

If the Environment patch you wish to import is complex and spread across several Environment layers, then you should use the merge feature by selecting **Options > Import Environment > Merge**. In this case, all Environment objects from the source song will be added to the Environment of the destination song. One thing to watch out for here is that the merged objects will be placed on the same layer as they came from, which can create a mess if there are already objects on that layer in the destination song. To avoid this, create blank layers in the destination song at the same positions as the layers to be brought in, before importing.

Something to be careful of in both cases, is the treatment of unique objects in the Environment like the Physical Input and Sequencer Input objects. If you import these objects, they will replace their counterparts in the destination Environment AND ANY CABLES LEADING INTO THEM WILL BE LOST. It is best to temporarily delete these from the source Environment, before importing.

Another thing to be aware of is that when you import an object, all objects cabled from the imported object are also imported.

Replacing an Older Environment with a Current One

If you have only added things to the Environment, but not deleted anything, then you can update older songs by choosing **Options > Import Environment > Update**. Only the new objects will be imported-the older objects along with their cabling and track assignments will remain in place.

Swapping Environments

In the complex situation where you want to swap the Environment for one MIDI setup for the Environment for another, Logic offers three options; *Replace by Port MIDI/Channel*, *Replace by Name* and *Total Replace*. But be aware that this complex task involves a lot of guess work on Logic's part and the results almost always require some manual fine tuning on your part.

Replace by Port/MIDI Channel

This option replaces all objects in the destination song for which there are objects in the source song addressing the same port and MIDI channel.

Replace by Name

This option replaces all objects in the destination song for which there are objects with the same name in the source song. This affords you some degree of control, by renaming objects in the source song accordingly.

Total Replace

This completely replaces the destination song's Environment with the source song's Environment. If you use this option, you will have plenty of work left reassigning objects to Arrange window tracks, but sometimes it's the only way.

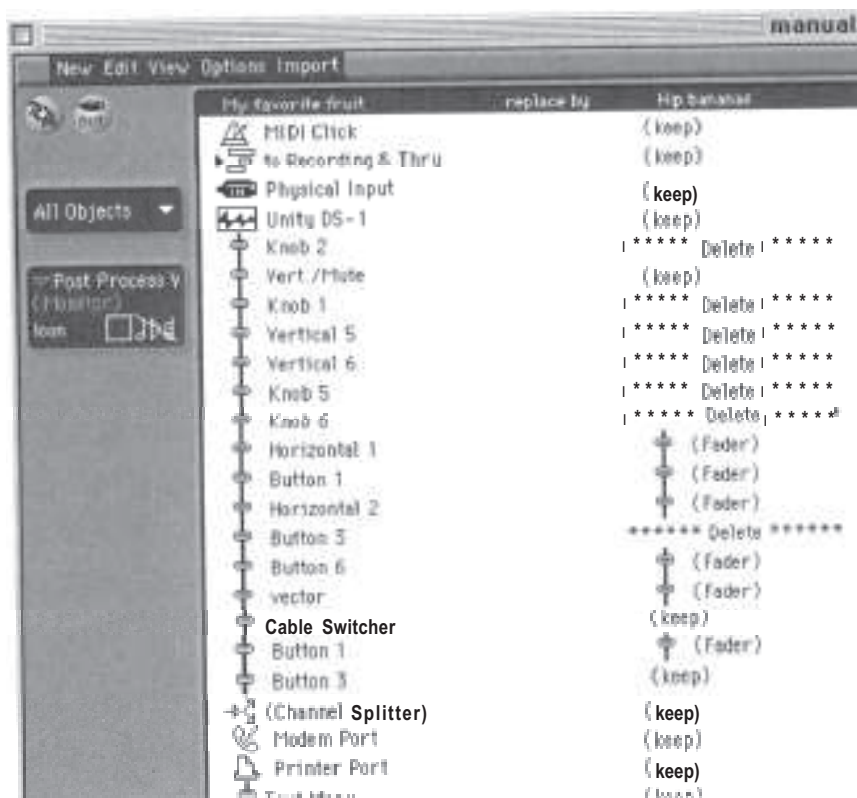
Replacement by Assignment

The most flexible and also most time consuming method of exchanging Environments is to indicate for each Environment

Environment Exchange

object whether it is kept, deleted or replaced and if it is replaced, what object replaces it. You do this by selecting **Options > Import Environment > Custom...**

When you select custom import, an Environment window appears listing all objects in the destination Environment in a column on the left, and listing the chosen action for each object in a column on the right.



You can assign the replacements one at a time, by clicking on items in the right hand list and selecting the desired replacement object from a pop-up menu. This menu includes all objects on the Instrument List of the source song.

Or, you can select one or more rows of the list, and then choose what to do from the Import menu. In addition to keep and delete, this menu offers several automated selection techniques described below.

The first selection on the Import menu is *Import Environment using current Assignment*. No importing takes place until you

make this selection. The other choices (including the pop-menus in the right column), only set up how things will be imported.

Import Options

The last two selections on the Import menu provide additional options in the importing process:

Import > Copy layer Names: the layer names of the second Environment are transferred to the current one.

Import > Copy selected Objects from 2nd Environment all objects which are selected in the source Environment are copied into the destination Environment, even if they don't appear on the assignment list.

Automatic Assignment Functions

The assignment functions described below are immediately applied to all selected rows in the import list. The assignments appear in the right column and all rows are then deselected.

Import > Assign as 'Keep': these objects are unaltered.

Import > Assign as 'Delete': these objects are deleted.

Import > Assign by Identical: these objects are assigned objects from the source song which match the type, icon, name, port and MIDI channel. Objects with the same Unique ID have priority.

Logic maintains an internal list of all Environment objects. When objects are deleted, their position in this list becomes available for a new object. Whenever an object is added to the Environment, it is put in the first available position on the list. (If there are no empty positions, it is added to the end of the list.) An object's position in the list is its Unique ID - as long as the object is not deleted, its Unique ID never changes.

Import > Assign by Unique ID: objects are assigned objects from the source song with the same Unique ID. (This option is

useful for updating an Environment when objects have only been added.)

Import > Assign by Port/MIDI Channel: objects are replaced by objects with the same port/MIDI channel. If a match can not be found for an object, an object set to port 0 and the same MIDI channel is assigned. If a match still can not be found, an object with the same MIDI channel is assigned.

import > Assign by Name: objects are replaced by objects with the most similar name. (At least 80% of the name must be the same.)

Import > Assign by Icon/Name: same as above except the objects' icons must also be the same.

5.7 The Environment and OMS

What is OMS?

OMS, or the “Open Music System” (or as it used to be known: “Opcode MIDI System”), is a System extension that acts as a logical interface between application programs and the physical MIDI ports of your Macintosh.

To work in this way, OMS lets you define so-called “nodes”, which represent the individual items in your MIDI setup. These nodes have a name and information about the MIDI device they represent: where it is connected, what MIDI channel(s) it receives on, and so on. You create the OMS nodes with the OMS setup program, and from there can then edit or erase them. For more information, see your OMS manual (although there are some hints to follow in this manual, in the section *Creating an OMS Setup for Logic* on page 5-101).

The OMS manual sometimes refers to nodes as “instruments”, but as this word has its own meaning in Logic, we will be referring to the virtual objects that you create in the OMS setup program as nodes-and only nodes.

When you are using OMS as a MIDI driver system, Logic is not aware of your MIDI interfaces, or the physical MIDI ports. All information concerning these is hidden in the OMS system, and cannot be accessed by applications like Logic. Instead, OMS gives you access to the above-mentioned nodes to communicate, and ensures that the flow of data is properly managed and maintained, if for example, strings of data need to be merged.

As with Emagic's Autolink (the connection between Logic and the universal editor/librarian SoundDiver), OMS also lets you run several programs at once.

When Should You Use OMS?

Logic offers very extensive support for OMS (from OMS version 1,2,3 onwards).

Nevertheless, you should only use OMS in Logic if you plan to run other OMS applications at the same time as Logic- for example Opcode's editor/librarian "Galaxy", or when you are using a multi-port MIDI interface which is not able to run in MTP emulation mode.



"MTP" stands for Mark of the Unicorn's MIDI *Time Piece*. Please note that you do NOT need to use OMS in Logic to run Opcode's Studio 4 or Studio 5 MIDI interface, Both of them can emulate an MTP (Mark of the Unicorn's MIDI *Time Piece*) without compromising on functionality at all. Only if you want to stripe analog tape with SMPTE using one of these interfaces, you will need the OMS setup program.

You can switch between using and not using OMS in Logic's Communication Preferences. It doesn't hurt to keep OMS installed on your system (i.e. as an active extension), whether or not you're using it for Logic-if you're not using it, it remains completely passive.

Please note that you do NOT need OMS to run an Opcode MIDI interface. In fact, you could run either a Studio 4 or Studio 5 in MTP emulation mode without compromising on functionality at all. However, if you want to stripe analog tape with

SMPTE using an Opcode interface, then you WILL need OMS.

Please note that because OMS adds an additional layer of hierarchy between Logic and the MIDI ports:

Logic's MIDI timing under OMS will not be as good as with its own MIDI driver.

Logic Environments can be more difficult to design and use under OMS.

Logic under OMS

To use OMS with Logic, you first need to install and set up OMS. Consult your OMS manual for further instructions.

Once OMS is installed on your system, you tell Logic to use it by checking the Use *OMS if available* checkbox in the **Preferences > Communication** window. Thereafter, whenever you start Logic, OMS will be used instead of Logic's own MIDI driver (unless you later uncheck the Use *OMS if available* checkbox).



You do not have to re-start Logic to activate and de-activate OMS. Each time you uncheck and re-check either the Modem Port or Printer Port checkbox, Logic will check its OMS status, and re-connect accordingly.

Assigning OMS nodes to Logic's inputs

You connect OMS nodes to Logic's inputs on the **Preferences > OMS Input Connections** page.

These connections are necessary if Logic is to receive MIDI events (from external keyboards, for example).

If the option *Connect all Real OMS Input Nodes with Logic* is active, all the information from the real, physical OMS nodes will be received by Logic. Normally, this means all the information that is received at the inputs of your MIDI interface.

If this option is switched off, Logic will only receive information from nodes that have been selected via the input flip

menus. If none of these have been assigned, Logic will be unable to receive any MIDI information at all.

You can select eight more OMS nodes from here to input data to Logic's inputs, but normally this will not be necessary if the **Connect all Real OMS Input Nodes with Logic** option is active.

The selection of OMS nodes for connection to the input becomes important when you wish to connect two simultaneously active programs via virtual OMS nodes. This is known as IAC-inter-application communication. (You can connect a node to more than one destination-superfluous connections are ignored.)

When you turn **Connect all Real OMS Input Nodes** option on or off, you'll have to switch OMS briefly off, then back on again via **Preferences > Communication**. This breaks the previous input connections, and establishes the new ones.



Input connections to OMS slow the computer down. You can speed things up - particularly if you have a slow Macintosh-by only making the connections you are really going to need.

Assigning Instruments to OMS nodes

You can assign any instrument's output directly to any OMS node. When OMS is active, there is a pop-up menu between the icon and the Cha parameter for selecting this assignment. If no OMS node has been assigned (*OMS Map used*) will be displayed here.

If you have not made a direct OMS node assignment, you can use the instrument's *Cha* parameter as usual to select an output port (*MU thru M63 PO thru P63*). But, be aware that these selections have a different meaning under OMS-they assign OMS nodes to the instrument via *OMS Output Maps*, which you set up in the Preferences window.

OMS Output Maps

Logic allows you to set up *OMS Output Maps* for both the printer and modem ports of your computer. (Of course you need multi-

port MIDI interfaces connected to these ports, for this to be useful.)

You create the OMS Output Maps using the **Preferences > OMS Output Map** pages. There are separate pages for the modem and printer ports, and each one contains pop-up menus for ports 0 thru 42. (The remaining instrument ports, 43 thru 63, are not used under OMS.) The assignments you set up in this table are stored in the Preferences, and are therefore shared by all songs.

There's a good reason for using OMS Output Maps instead of assigning instruments directly to OMS nodes. This allows you to switch between using and not using OMS without any further Environment adjustment. When OMS is in use, the map connects instruments to the correct OMS node and when OMS is not in use, the standard *Cha* port assignment selects the same MIDI ports directly.

Creating an OMS Setup for Logic

If you have not done so already, here is one logical way to create and use an OMS setup for Logic. Once you've created the Setup, everything becomes really quite simple:

This method describes a setup for older Macintosh computers, which still have a modem and printer port. The procedure for newer computers with a USB port should be similar-just use the appropriate names to label the OMS nodes.

1. Start your OMS Setup Program.
2. Create a node for each of the physical outputs your system has and name them in the same way that Logic does - "M1", "M2", . . . and/or "P1", "P2", etc.
3. Exit the OMS setup program. This OMS setup should now never be changed, unless you add another MIDI interface to your system. Of course, you can add OMS instruments at any time-but for work with Logic, that shouldn't be necessary. Just create virtual representations of new synths in the Environment in the usual way.
4. Boot up Logic, and activate OMS (via **Preferences > Communication**), if you haven't already. OMS must be up

and running as your MIDI driver.

5. Select **Preferences > OMS Output Map** for the desired output port (modem or printer).
6. Assign each OMS node to the corresponding output; for example, Modem 1 should be assigned to the “M1” node in the flip menu.
7. Quit Logic, so that the Logic Preferences file updates itself (or select **Options > Save Preferences** from the Key Commands window).

That’s it! Now you should be able to play any of your old songs and they should sound identical, with or without OMS.

Logic as an OMS node

You can create a virtual OMS node for Logic by checking the *Create virtual IAC OMS Instrument for Logic* checkbox in **Preferences > OMS Input Connections**.

Logic creates a virtual node in the OMS *IAC driver*. Double-click the driver in the OMS Setup window to assign the node a suggestive name like “Logic”. The node will be visible to connect cables to, in OMS applications which support IAC (inter-application communication).

Please remember that Logic can transmit all MIDI data (track events, MTC or MIDI clock, for example) to all OMS nodes, even virtual ones representing other programs which support IAC.

Points to Note when Working Under OMS

MIDI Clock/SPP and MTC Output

You can make any OMS node an output for MIDI Clock/SPP (Song Position Pointers) or MTC. Just make the necessary settings via **Song Settings > MIDI Options**.

Rectangular Output Symbols

Environment objects with direct OMS assignments have rectangular outlet symbols instead of the usual triangles. However, if the OMS Map is used, the outlets remain as triangles.

Initializing OMS settings

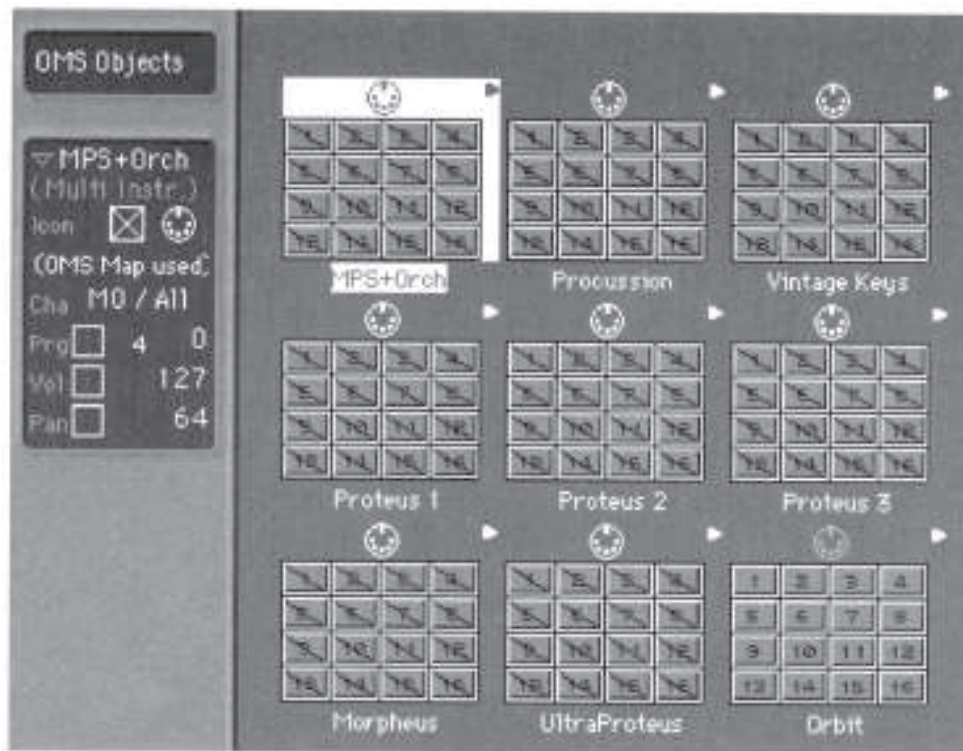
Select **Options > Initialize > just OMS settings** from the Key Commands window.



This initializes the OMS Output Map, all the input connections, and the settings for MIDI Clock, MTC and the CBX Ports.

Updating Logic to Your OMS Setup

If you already have OMS set up, or you make changes to your OMS setup, you can select **Options > Update > OMS Equivalents**, or use the associated key command to create Environment instruments for each of your OMS nodes. If you selected OMS's multi-timbral option when creating the node, Logic will create a multi-instrument for that node. (So, when making your OMS setup, de-select the multi-timbral option if you want regular instruments created by this process.)



The *OMS Objects* layer might look something like the above picture.

About the Created OMS Environment Objects

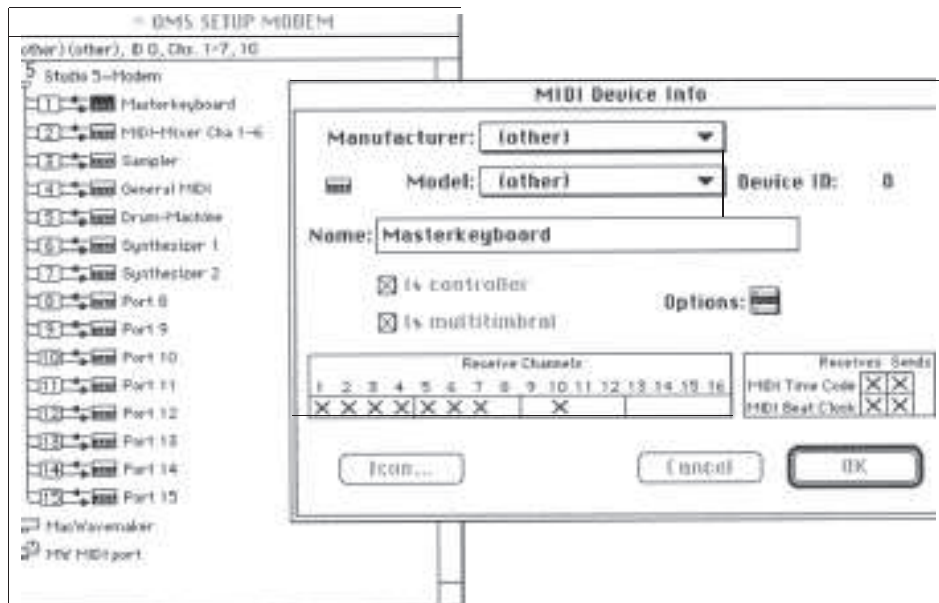
The following settings are taken from the OMS studio setup:

- The names of the OMS instruments,
- The number and assignment of the ports

In the example shown in the diagram, there are 15 instruments on Opcode Studio 5 ports and one Macintosh MIDI card with OMS driver.

The channels of the multi-mode instruments (as activated sub-channels of multi-instruments).

The OMS settings for the Environment layer shown above are as follows:



On the left is the OMS setup. This is where all the MIDI ports and internal MIDI cards are displayed. All external MIDI devices are connected to the ports by cables.

The right window (MIDI Device Info) is for making settings for the individual MIDI devices. Keep in mind that the receive channels which are active here are “translated” into the active sub-channels of the multi-instrument.

5.8 Tips and Comments

Here are a few helpful tips for creating your own Environments. For more detailed examples of Environment processes and how to create them, see the *Beginner's Guide to the Environment* PDF file included on your Logic CD-ROM.

Working with Notes

One of the most common applications of the Environment, beyond controlling your MIDI studio, is processing MIDI note events. This includes such things as transposition, scale correction & filtering, arpeggiation, echo & delay, voice doubling, etc. Some of these processes have Environment objects dedicated

to them, while others require patching several Environment objects together. In either case, it is helpful to keep in mind how MIDI handles notes, and what information they contain.

How to Avoid Hanging Notes

MIDI notes come in two parts: a note-on event and a note-off event. For live MIDI, there is an *indeterminate* time between these events (i.e. you don't know when the player is going to release the note). For notes recorded in sequences, the note length is known, and for this reason it is not uncommon for MIDI sequencers to combine a note-on and its matching note-off into a single event-as a note and its duration. Logic records notes this way and it is a good idea to keep this in mind when processing notes in the Environment.

With live MIDI input it's not hard to leave hanging notes. This can happen when you change a note-on event without making a similar change to the corresponding note-off event. For notes recorded in sequences this is not a problem, because notes are single events-Logic takes care of making the matching changes when it sends out the note-on and the note-off. The upshot is that processes that work fine when sequences are played through them, might leave hanging notes when live MIDI is processed. Logic tries to prevent this by not allowing some kinds of changes to note events (e.g. random pitch, changing velocity to 0 and changing 0 velocity to something else), but you can work around these limitations, and it's up to you to bear the consequences.

What's in a Note

Each MIDI note event contains 3 pieces of information: channel (1-16), note number (0-127) and velocity (0-127). (Note that in many places-transformer, instrument and fader parameters, for example-the MIDI channel is displayed using numerals 1-16.) Channel is often associated with the type of sound the note will produce, because it is typically used to select the MIDI device that will play the note (the MIDI channel in the case of

multi-channel MIDI devices). Note number is most often associated with pitch, but for drum-mode MIDI instruments such as drum synths, it determines what sound is played. When used for pitch, note number 60 typically corresponds to middle C (C3). Velocity is generally associated with accent, as it is often applied to a synth preset's volume or filter cutoff parameters (or both). Do not confuse velocity with volume, however—a preset's overall output volume is typically set by MIDI controller #7.

Note-on events with velocity 0 are interpreted as note-off events, and this is the only form of note-off event some older synths recognize. (This means that the note-on velocity range is really 1-127). Although Logic will record and play back true note-off events with their correct velocity, you can not change, use or even see note-off velocities in the Environment (or the Event editor). For Environment purposes, consider note-off events to be note-on events with velocity 0.

You have (almost) complete freedom within the Environment to change any of these three pieces of information: channel, note number and velocity. You could for example, use a note's number (pitch) and/or velocity to influence what channel it goes to (i.e. what sound it plays). Conversely, you could use its channel to influence its pitch and/or velocity (e.g. transpose and accent notes on channel 3). The things you are prevented from doing include randomizing pitch, and changing note-on events to note-off events or visa versa. The purpose of these restrictions is to prevent the occurrence of hanging notes, as described above. (With a little clever Environment plumbing you can work around both restrictions.)

Where to Insert Environment Processes

The following applies to Environment process that are intended to modify MIDI data passing through them. This includes processes that involve a single Environment object (e.g. an arpeggiator or a delay line), as well as complex processes that combine many objects. For complex processes it is a good idea to start and end the process with a monitor object. This allows you both to see what the process is doing, and to have an easily identifiable entry and exit point. (If you intend to

collect the process into a macro, name these monitors "Macro-In" and "Macro-Out".)

The three most common positions to insert an Environment process are: between the physical input object and the sequencer input object; before an instrument object; or between any instrument object and an output port. In the third case, remember to remove the instrument's direct output assignment (see page 5-13).

The *Clicks & Ports* layer of Logic's default song (see illustration on page 5-3) contains an example where objects are inserted in the cabling between the physical input and the sequencer input. The keyboard object shows note activity; the two monitor objects show the data passing through them before and after events and the cable switcher switches between: no process, an *octave doubler* and a *velocity enhancer*. Processing applied at this point will affect all incoming MIDI data, and will be recorded when Logic is recording. Note that processing of this sort is not easy to undo—you would have to apply a complimentary process to any recorded sequences.

If you want an Environment process to affect data going to a specific MIDI device, but you don't wish to record the process, simply assign the process's input object to an Arrange window track, and cable its output object to the Environment instrument or port object serving the desired MIDI device. Although this is the simplest solution, it has the disadvantage of not making the instrument parameters available when you select the track. If you wish to have these available, use the method below.

If you wish to process data recorded in sequences (and have the instrument parameters available from the Arrange track), then the third location—between an instrument and a port object—is the best choice. In this case the process can be switched in and out at will without making any permanent change to the data. In case you want to make the changes permanent, cable the output of the Environment process into the sequencer

input instead of the port object; record on a track assigned to “No Output”, then reassign the track to the port object.

Creating Control Panels

Many Environment processes involve on-screen control panels. Typical examples include simulated mixing desks, synth editors and analog style sequencers. In some cases you may want to automate, or at least take snapshots of the fader positions. For this you need some way of recording the fader positions, and of playing back the results through the controls. Here are two rules of thumb to make this simpler:

Cable the control panel faders in series.

Ensure that each fader’s **In** and **Out** definitions match.

With the faders cabled in series, you can either assign the first one to a track, or cable a track instrument into it, so that data played on that track will control the faders. For this to work, a fader’s In and Out definition must be the same, because the Out definition is what is automatically recorded by Logic when the fader is moved with the mouse, and the fader will only react to these events if they match its In definition. Keep in mind that each of the faders in the series must be unique in some way - i.e. it must have a different MIDI channel, or MIDI controller number than any of the other faders.

If you want to convert some set of recorded fader positions to a snapshot, open the Event list editor, then copy & paste the MIDI controller messages into a Sysex fader (see page 5-86). Cable the output of the Sysex fader into the first control panel fader in the series. Ensure that none of the events is selected when you close the Sysex fader window. Then clicking the Sysex fader (button styles work best here) will set all the controls to the snapshot values.

Chapter 5
The Environment

Chapter 6

Mixers, Effects and Audio Objects

6.1 The Basics

Your recording studio probably consists of one or more multi timbral sound modules with stereo analog outputs, microphones and guitar pre-amps or similar analog signal sources, as well as a mixing desk (analog or digital, at any rate a separate device). The Installation Handbook describes how to connect these devices to the audio In- and outputs of the sound card or audio system you are using with Logic. With your hardware mixing desk, you route the various signals that you want to record, using subgroups when possible, to Logic's recording inputs (i.e. the sound card). The mixing desk also typically offers separate microphone pre-amplifiers. With a simple setup, these tasks could also be done by separate microphone pre-amps (Voice Processor, for instance), or by sound modules with integrated mixing functions. A traditional mixing desk, however, performs an additional function, mixing the analog signals from your MIDI multi timbral sound module(s) with output signals from the soundcard.

Logic mixes the audio signals that are fed to the sound card with its own effects.

The instrumental part levels and the effect levels, that is, the entire mix of the signal output by your sound module, is remotely controlled by Logic.

The Adaptive Track Mixer performs both of the following functions simultaneously: The control of MIDI multi timbral sound modules (including level, pan, effects settings for the different parts), as well as mixing the audio tracks

within Logic (level controls, panning and so on). The actual combining of the audio signal from your various pieces of hardware (MIDI modules, sound card, etc.), resulting in the stereo signal that you will hear from the speakers, is once again the job of the mixing desk.

Any mixing action in Logic's mixers, that is: every movement of a fader or knob, can be recorded simply by pressing the record button. Afterwards, the passage with the recorded movements of the controls can be played back, and you will see how the knobs move just as you have recorded them. Since the data controlling these knobs are MIDI controller events, you can edit them like any other MIDI events—for instance with the HyperEditor.

You won't need a mastering device (two-track tape recorder, DAT or the like) to record the final mixdown of your production. By pressing the Bounce button, you automatically create a stereo file of the total mix, which can be burnt onto a CD without further editing. To do this, however, you will need to record the complete MIDI accompaniment as a stereo audio signal, so that this can be mixed together with the other audio tracks by Logic. No external mixer is needed for the Bounce procedure—except possibly to monitor as you work.

Different Mixers

Logic offers you a range of mixers. In most cases, we recommend that you use the adaptive Track Mixer. This configures itself automatically, so that only the tracks used in the Arrange window are shown, in sequential order. The Track Mixer is described in the section *Track Mixer* on page 6-4. Exceptions to this, for certain applications, are described in the section *GM Mixer* on page 6-100.

The Audio Mixer lends itself to working on projects that only require audio tracks, especially when few audio tracks are required. The automation works differently than with

the Track Mixer, in that all automation files are written to a separate track. This allows greater clarity in the Arrange window. This is also recommended when the Audio tracks are combined with several MIDI tracks, where you intend to do most of your mixing directly on your hardware mixer. You may also prefer to use the Audio mixer if you work with TDM Hardware, and it is not your usual working practice to always use the same virtual mixer. The Audio mixer is a layer of the environment, and is described in the section *Audio Mixer* on page 6-98.

If you are working only with multi timbral sound modules, and don't work with any audio tracks, (especially if you use General MIDI, (Roland) GS- or (Yamaha) XG sound modules, it is worth using the GM Mixer. If you are using two GM modules or one GM module with 32 Parts and two MIDI inputs, you can use two GM Mixers. The Automation works differently from the Track mixer, in that the Automation files are written to different tracks than the other MIDI data, which improves the clarity of the Arrange window. The GM mixer is described in the section *GM Mixer* on page 6-100.

If you are playing back songs that were created with an older version of Logic, which didn't contain the adaptive Track mixer or the Audio mixer, there will probably be different mixer configurations, made from separate Audio Objects, which will still work. If you wish to construct an individual, extremely flexible Modular mixer, you can use Audio Objects and other Environment Objects to build your own configuration.

Users of Former Versions

As a rule it is not necessary for you to build your own mixers out of Audio objects-at least not for a typical mixdown. The features, which you may be familiar with from older versions of Logic, such as the ability to link together Audio- and other Environment Objects, are of course still available. The

description of the functions of the mixers, with their various effects and output channels, as well as the automation, also applies for the most part to the separate Audio Objects. Any differences will be pointed out, as the features are explained.

If you are already familiar with the Audio mixer, but are new to the Track mixer, please note that the automation of the Track mixer works differently. The automation files are written into the individual audio and MIDI tracks-not just to the track (usually) labeled “A-Playback”.

The appearance of the Audio Objects has been modified for this version. When opening older songs, you may have to change the size of the Audio Objects they contain, so that the new format is visible. Newly created Audio Objects appear with the new look. You can switch the new look on or off in the Preferences (**Audio > Audio Preferences**), under the section “Display” with the option New Style Audio Objects.

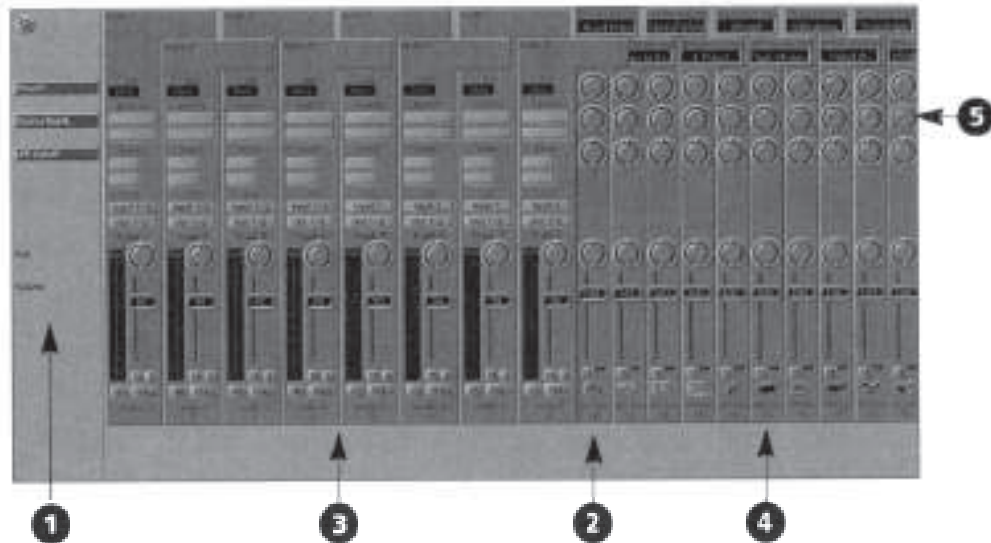
6.2 Track Mixer

The Track mixer is the simplest and most convenient tool for mixing audio and MIDI tracks. Logic creates a mixer based on all tracks currently assigned to the track list of the Arrange window. It contains as many channels as there are tracks. Unlike the GM mixer, or the Audio mixer, the adaptive mixer deals with both audio and MIDI tracks. These are laid out in the same order as in the Arrange window. Each channel represents an object in the Environment: the audio channels are audio objects, the MIDI channels are instruments (parts of multi instruments). The parameters which you adjust on the adaptive Track mixer are also available in the other displays of the Environment or Arrange window. To open the adaptive mixer go to the Arrange window and select **Windows > Open Track Mixer**.



If you press  while opening the track mixer, the track mixer will appear as a float window.

Track Mixer



On the left is the parameter display ❶ which tells you (among other things) which effects are assigned to the aux sends. Selected tracks are framed in red ❷. The operation and parameters of the audio track objects ❸ and the MIDI track objects ❹ are the same as normal. The number at the bottom of the channel is the track number. The graphic layout of the program names and bank select commands ❺ is slightly different from the design of the GM/GS/XG mixer (for clearer labeling). Unlike the audio mixer, you can adjust the size of the display vertically and horizontally using the telescope “zoom” gadgets.

The MIDI channel strips have little “LED”, which light up green for Note On events, and in yellow for all other MIDI information.

Tracks which do not have their own mixer parameters are not visible in the adaptive mixer. These include delay objects, arpeggiators, etc. If there are several consecutive tracks in the Arrange window with the same track instrument, this instrument is represented by a single channel in the adaptive mixer.

Display

The **Track** menu is where you define what types of track are displayed in the adaptive mixer. You can switch off MIDI or folder tracks, if necessary. Although the bus and master

(output) objects are not track instruments, they can also be displayed in the adaptive mixer.

The **View** menu is where you customize the display to suit your requirements. You can hide the labels for the user-defined knobs on the MIDI mixer channels, which saves a great deal of space. You can individually switch off the display of the instrument names, program numbers, bank numbers, the user-defined knobs on the MIDI mixer channels (*Assign*), the pan pots (*Pan*), the faders (*Volume*), and the track names and numbers, to make more space. If you want to remove the instrument names and program names from the display, the channel keeps its rectangular shape-without the tabs which are needed for long names. If you switch off just the display of the instrument name, the track name is shown instead. You can also switch off display of the equalizer, the aux sends, and Plugins for the audio channels.

Like the other editors, the adaptive mixer can be stored as part of any Screenset. You can switch between these using the number keys —, or you may call up two digit screensets by holding down  while entering the first digit.

Setting Knobs to their Defaults

Clicking a fader or knob in MIDI tracks while holding **Alt**, resets the value to a neutral value. The neutral value is zero for all controls except *Volume* (#7)= 100, and *Panorama* (#10), *Balance* (#8), *Resonance* (#71), *Release Time* (#72), *Attack Time* (#73) and *Cutoff Frequency* (#74), for all of which the default is 64 (center position).

-clicking a volume fader sets it to 90 (0 dB)

m-clicking a pan pot sets it to the middle

m-clicking a send pot sets it to 90 (0 dB)

Folders and the Track Mixer



The adaptive mixer can not only display the contents of the the highest level of the Arrange window, but also the contents of folders. The adaptive mixer automatically registers any change in the folder structure or instruments assigned to tracks. If a folder is selected, or if the top window is an Arrange display of a folder when you open the adaptive mixer, it will show only those tracks in the folder. If you click the directory close-box (the black button below the window close-box at the top left), you switch the adaptive mixer to the next-highest level, in this case the level containing the folder. Double-clicking the folder channel restores the display of the folder. Of course, the display of *Folder Tracks* must be switched on in the **Tracks** menu.



Options

Send all Mixer Data

By selecting **Options > Send All Mixer Data**, you can transmit all currently visible mixer knobs and faders to MIDI.

Track Selection when Playing or Recording

You can prevent the track selection from changing when you make an adjustment to a channel, while Logic is in recording- or playback mode. For each situation, there is an independant parameter. You can access these parameters by selecting **Options > Change Track in Record Mode** and **Options > Change Track in Play Mode**.

6.3 Inserts, Sends for Plug-ins

You can get CPU-based real-time effects with any audio hardware that can be addressed via Emagic's own audio drivers. This is applicable to:

AV (Apple Sound Manager)
Audiowerk8
ASIO (all Soundcards accessible via ASIO drivers)
Korg 1212 I/O card

With DSP Factory, DAE TDM or Akai DR hardware only the DSP-based real-time effects are available.



“DSP-based” means that these effects are calculated using digital signal processors on insert cards (or in the external audio hardware). “CPU-based” means that these effects are calculated in the computer’s processor. These two methods are also known as “host-based” or “Native Signal Processing”,



An upcoming update to Logic will allow up to two CPU based real-time effects to be inserted on each channel of the DSP Factory. This will be posted as a free update to version 4.0 at the Emagic web site.

Memory Allocation

A propriety Plug-in requires between 10 and 400 KB of RAM. A single VST Plug-in, with its own graphic interface (editing window), can require up to 2 MB. Please ensure that you have enough free memory.



Refer to the section on memory allocation in the Installation manual

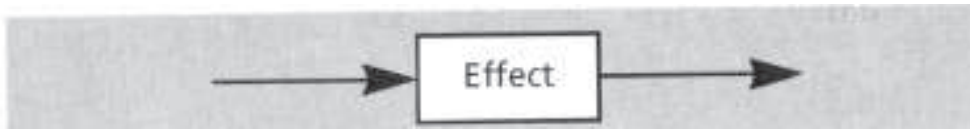
Introduction

Here are a few basics about signal flow in mixing desks. If you are already knowledgeable in this area, then go directly to the section *Plug-ins from Other Manufacturers* on page 6- 19.

In practice there are two ways of sending audio to effects: via an insert, or via a bus (also known as an “aux send”).

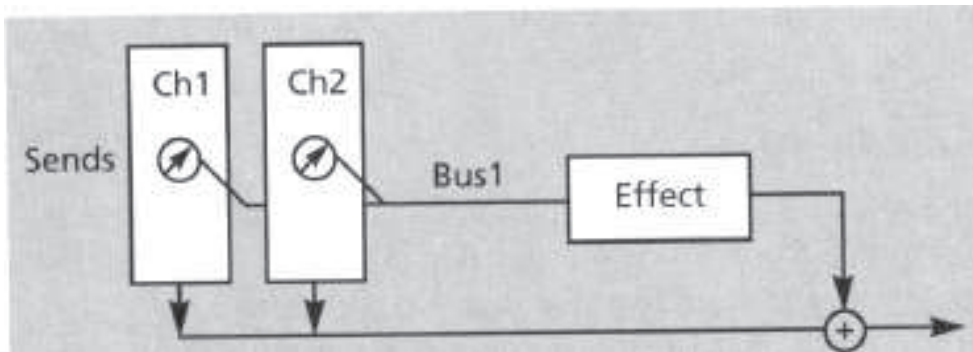
Insert Effects

With insert effects, the all of the signal is processed. This means that 100% of the signal flows through the effect. This is suitable for equalizers or dynamic effects. This also typically applies to pan knobs and faders.



Bus Effects

In this case, a controlled amount of the signal is sent to the effect. The classic example is reverb, where the most important effect parameter is the amount sent, as this determines where the signal source appears to be placed in the listener's audio field. Echo (delay) and the modulation effects (chorus, flanger ...) are also normally used in this way.



With Logic, the effect is positioned in the insert of a bus object. The signals of the individual tracks can each be sent to the bus with a send knob. The audio signal is then processed with the effect, and mixed with the stereo output.

The advantage of this, over inserting effects on tracks, is efficiency. This method allows as many tracks as you like to be processed by one inserted plug-in, saving CPU power as compared to inserting the same effect directly into multiple tracks.

With computation-intensive effects like reverb, it is always advisable to feed them via a bus. Chorus, flanger and delay should also always be fed via a bus, if they are going to be used on more than one track.

In some cases it may make sense to patch an effect such as a delay, directly into the insert of an individual track. There are

no restrictions in Logic as to where real-time effects may be used.

Equalizer (EQ) and Filter

Please note that the availability and extent of the EQ facilities depends on the hardware you are using. If you are using internal AV hardware, for example, you can insert up to 4 bands of EQ per track. (If you need more, you can also use the effects inserts for additional EQs). With a Yamaha CBX-D3, you will not have access to EQ at all. With Pro Tools II or AudioMedia III hardware, on the other hand, you can control up to two frequency bands, with a Yamaha CBX-D5, or DSP Factory, up to four.

An Alternative: Equalizer Plug-ins

The EQs included with Logic Audio Platinum's inserts are more flexible than those offered on most analog mixing consoles. Logic Audio Platinum features such Equalizer Plug-ins as the DJ EQ, and the Fat EQ (with a graphical representation of the frequency curve).

<i>BP/On</i>	the Bypass switch in the top right turns the EQ band off.
<i>Typ</i>	next to the BP switch you can use a flip menu to select the type of Eq, Thru, Low/High Shelf, Parametric, Low/Highpass Filter)
<i>HZ</i>	mid frequency or cut-off frequency
<i>dB</i>	cut/boost)-(n/a for filter)
<i>Q</i>	bandwidth-(n/a for filter and shelf)

If your HDR allows several EQ bands, the flip menu for the next band (labeled "Thru", until assigned) is always visible so that you can use it.

EQ adjustments can be stored and played back via MIDI (see the section *Automation* on page 6-81, for more details).





A long click on the "Thru" button allows you to select a new equalizer band, or a filter. The parameters of the new EQ are then displayed. You can add up to 4 EQs or filters.

EQs can also be added as insert effects, if you don't want to make the audio objects too long, or if you want to use more than 4 EQs/filters per track.

For details on the various types of EQ please refer to section *Parametric Equalizers* on page 6-11.

Parametric Equalizers

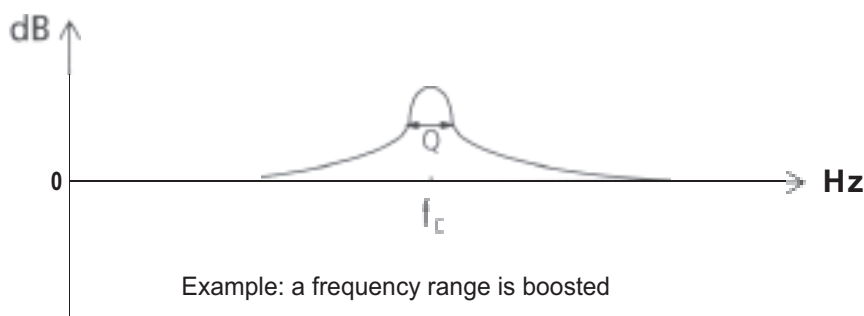
An equalizer (EQ) is used to boost or cut specific frequency ranges.

HQParEQ

The fully-parametric equalizer "HQParEQ" has the following three parameters:

Hz	Center frequency
dB	Cut/Boost
Q	Quality

Thus a symmetrical frequency range on either side of the center frequency is boosted or cut. You can adjust the width of this frequency range with the Q factor.



Bear in mind that the "HQParEQ" offers very high sound quality but therefore requires more processing power than any other EQ algorithm.

HQSweepEQ

Unlike the fully parametric EQ, the "HQSweepEQ" has no adjustable quality function or (Q) and doesn't require as much processing power.

The "HQSweepEQ" also offers high sound quality, and thus is still a CPU demanding algorithm.

ParEQ

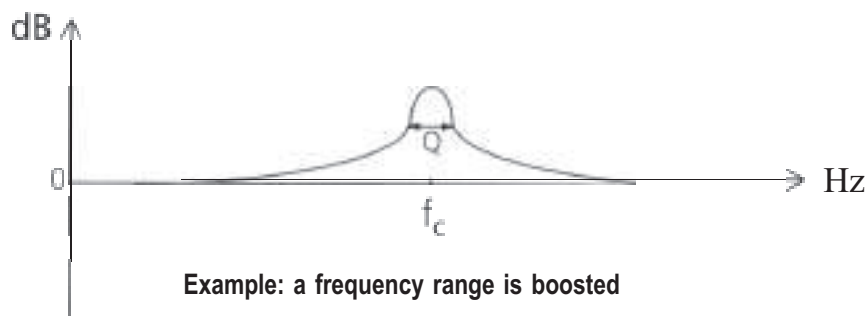
The fully-parametric equalizer "ParEQ" has the following three parameters:

Hz Center frequency

dB Cut/Boost

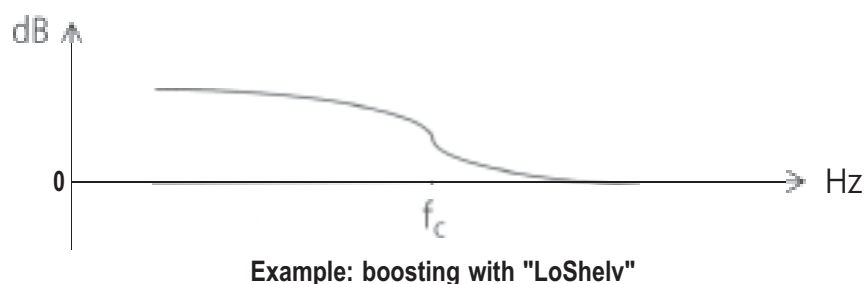
Q Quality

Thus a symmetrical frequency range on either side of the center frequency is boosted or cut. You can adjust the width of this frequency range with the Q factor.



LoShelv

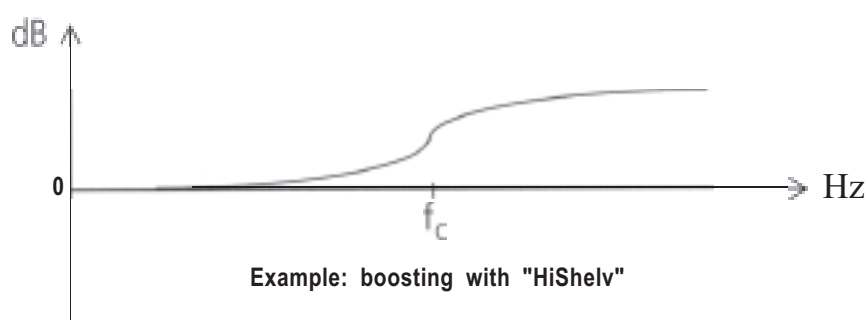
The "LoShelv" equalizer only affects the frequency range below the selected frequency.



The low shelving equalizer allows you to boost or cut the bass range.

HiShelv

The "HiShelv" equalizer affects only the frequency range above the selected frequency.

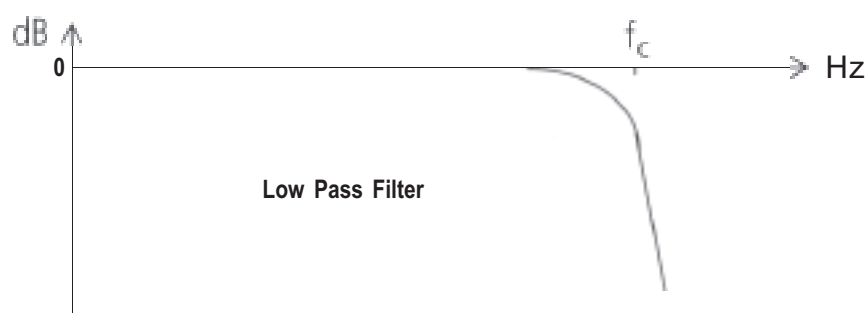


The high shelving equalizer allows you to boost or cut the treble range.

Filter

LowPass

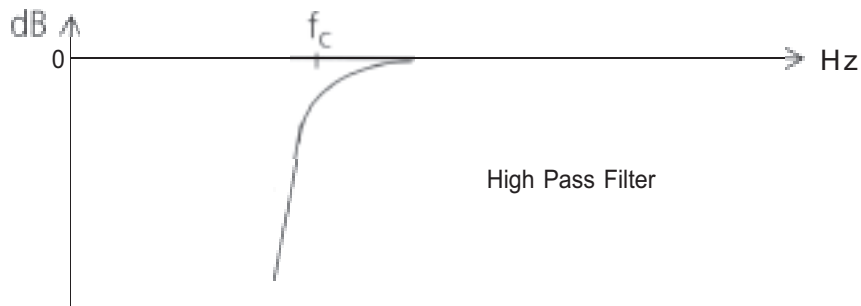
The "LowPass" filter affects the frequency range above the selected frequency. Lower frequencies pass through the filter.



You can use the low pass filter to completely get rid of the treble range above a selectable frequency.

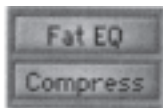
HighPass

The "HighPass" filter affects the frequency range below the set frequency. Higher frequencies pass through the filter.



You can use the high pass filter to completely get rid of the bass range below a selectable frequency.

Inserts



If you have enough processing capacity, you can use up to 8 inserts per audio object.

An extra blank Insert is created, as soon as all the currently displayed inserts are used, up to the maximum allowed.

The Insert flip menu displays all available propriety Logic Audio Plug-ins, as well as all correctly installed VST Plug-ins.

Mono/Stereo Input/Output

By holding **Alt**, you can also connect effects which have a mono input and a stereo output (m/s), or stereo I/O, in mono objects. If you add one of these (m/s) effects, you should usually only insert effects with stereo inputs (s/s) from this insert point onwards.

Don't forget that in general, stereo effects require twice as much processing power.

In stereo objects, usually only effects with stereo inputs and stereo outputs s/s), should be inserted (see picture).

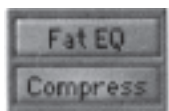
However, Logic can automatically insert conversion modules in the background to handle stereo→mono and mono→stereo transitions. This enables you to use any Plug-in in any order. But keep in mind that:

These conversion modules require extra processing power. During a stereo-→mono conversion, all spatial information is lost.

During a mono-→stereo conversion, no spatial information is added-the same mono signal is sent to both outputs.

All Plug-ins that may require such a conversion are marked with a dot in front of the I/O description, like this: *(m/s). For optimal performance, we recommend avoiding these “marked” Plug-ins.

Bypassing Inserts



Inserts can be bypassed by  clicking on the name.

The name of an active Plug-in is highlighted in blue. Bypassed Plug-ins are gray.

Sends (Aux Sends)



The Sends allow you to send a portion of any signal out of the signal’s audio object, to an internal bus or auxiliary output. The Send routing options available vary, depending on the hardware you are using.

NOTE: the CBX-D3, AudioMedia and Session 8 hardware (in external mix mode) offer no aux send facilities.

Click on an empty Send slot and select one of the possible send destinations, using the flip menu that appears. When you have selected the send destination, you can regulate the amount of signal supplied to the send using the Send Amount knob control. As you are adjusting the level, the relevant value appears in the box, to the left of the knob.

Remove Send

Open the flip menu of that Send slot again, and select No Send.

Normalizing the Send Level

You can reset the Sends to 0 dB (90) with a m-click on the knob.

Bypassing the Sends



You can switch Sends into bypass mode by *Alt*-clicking the name.

If a Send is switched on, its name will be green. If a Send is switched off, its name is gray (bypass mode):

Automatic Sends

The sends can be automated too. The first send responds to Controller #28, the second send to Controller #29, etc. Please refer to the section *Controllers for the Insert Effects* on page 6-85,

Post and Pre Fader Aux Sends

Auxiliary sends in analog mixers are positioned either before (“pre”) or after (“post”) the fader. To select from between these two options in Logic, click-hold the Send insert. The Bus menu opens, and at the top you may select either *Post* or *Pre*.

Post fader means the level going to the aux send changes along with the fader movements. If you are using the aux send to route signal from the channel to a reverb, this means that the relationship between the original signal and the effect signal remains constant. When you pull the fader all the way down, the reverb disappears as well. This setting is generally the most useful, and is the default setting in Logic.

If an aux send is set to be pre fader, the level going to the send remains constant regardless of any fader movements. Even if the fader is pulled completely down, the signal is still fed to the selected bus. Pre fader sends are mainly used for monitoring, either for headphone monitoring in the studio, or for foldback monitoring on stage. Pre fader aux sends are also useful when you have the fader completely down, and you want to hear just the effect signal (without the original signal). Normally you

would use inserts (see below), but with pre fader sends you can simultaneously process several audio objects with the same effect. In the “pre fader” setting, the equalizer still affects the sends (pre fader, post EQ).

Stereo Sends

When using Audiowerk, Direct I/O, ASIO or DAE with TDM hardware (ProTools III, 24 or 24MIX), Logic includes the ability to assign sends of mono tracks to stereo busses (or stereo outputs). It is not necessary to assign two sends to address a true stereo reverb plug-in. Select options like *Bus 1-2* or *Output 3-4* in the send flip menu.

Post Pan

This applies only to TDM hardware: when using stereo sends on mono objects, you can also use the option “Post Pan”. This means that the signal sent to the stereo send destination will be divided to the left and right channels, according to the setting of the pan fader. For example, this might be useful if the a reverb plug-in includes the processing of the pan position. To avoid confusion with the Pre/Post fader setting, this option appears only when using stereo sends in mono objects.

6.4 Plug-ins

Plug-in Window

You can add any of the real-time effects by means of a click-hold on an insert slot (see the section *Faders and Level Adjustment* on page 6-66).

Double-clicking an assigned insert slot opens the relevant plug-in window. This is where you can edit the parameters of the real-time effects.

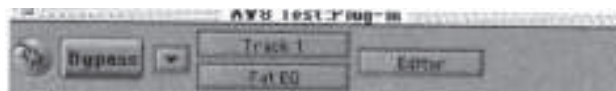
Inserting Mono/Stereo Plug-ins

With version 4.0, the plug-in menu (long click on the Insert Slot) only displays those plug-ins with a channel format that matches the given Audio object: In mono objects, only mono plug-ins (m/m) are displayed. The same is true for stereo objects and stereo-plug-ins (s/s).

If you want to insert plug-ins featuring a different channel format, e.g. a Tape Delay to a stereo track, then press and hold **Alt** when you're opening the plug-in menu.

Functions of the Plug-in Window

Every Plug-in window has a row of switches at the top whose functions are the same in all Plug-ins.



Link

If the Link button is switched off (default), you can open several Plug-in windows simultaneously.



If the Link button is switched on one Plug-in window is used to display all double-clicked Plug-ins. The window changes to show the controls for whatever plug-in was last double-clicked.



Bypass

The bypass switch takes the effect completely out of the signal flow. The effect's input signal is routed directly to the output, unaltered.



Arrow

The arrow menu is used to store and load the effects programs; see the section *All plug-in settings are stored within the songfile, and are automatically recalled the next time you Load that song*, on page 6-19.



Choosing an Audio Object

If you have used an effect algorithm in another audio bus object, for example in another track, you can simply swap between the objects. You can quickly compare settings or copy effects programs.



InsertSlotSelection



If you have installed plug-ins in other insert slots of the same audio object, you can quickly change to the other Plug-ins. This allows you to quickly balance the settings of all the Plug-ins within an object.

Plug-ins from Other Manufacturers

Logic supports the VST standard, which allows real-time effects made by different software manufacturers to be integrated into Logic's audio signal-flow.

Logic automatically searches for a folder called "VST PlugIns" or its alias, which must be in the same folder as the Logic program. In order for VST Plug Ins to work, Logic must be allocated at least 16 to 20 Megabytes of memory.

Dither	(s/s)
Auto Pan H-VST	(s/s)
EchoTrans™ H-VST	(s/s)
Flanger H-VST	(s/s)

The plug-ins are automatically available in the Plug In menu, along with Logic's own effects. This is the menu which opens when you open an audio object's Insert button (in a Bus or Track object).

Double click an inserted plug-in to open its editor. If the selected plug-in does not have its own editor, Logic's standard control surface will be used to access the plug-in's parameters.

All plug-in settings are stored within the song file, and are automatically recalled the next time you load that song.

You can also use Plug-in Setups to help organize plug-in parameters. These are analagous to the storage locations in an external effects unit, combined with a clipboard for effects parameters.

You can use the arrow at the top of the Plug-in window to open a flip menu with the following functions:

- Copy Setting*** copies all parameters onto its own “private” clipboard. They remain here until the next time you select ***Copy Setting***. This does not affect the global Logic clipboard.
- Paste Setting*** If you have opened a plug-in of the same type, you can add the parameter set from the “private” clipboard. This allows you to quickly set up several plug-ins of the same type, or exchange effects settings between songs.
- Save Setting*** This allows you to save all the plug-in parameters to disk. This is useful if you have created a special sound effect, which you want to have available for future use.
- Load Setting*** For loading stored parameter sets from disk. The file selector box shows only setups for compatible plug-in types.

Each real-time effect has its own set of parameters, and therefore its own file format.

***Proprietary Plug-in-Settings created with Logic Audio Platinum
Windows can be read by Logic Audio Platinum Mac, and vice versa.***



Logic Audio Platinum automatically creates a folder called “Plug-in Settings” in the folder that Logic itself is installed to, if one is not already present there. Inside that folder, a sub folder with the name of the current plug-in will be created. Load & Save brings the fileselector automatically to that location.



Waves plug-ins: Once a WaveShell has been installed properly, the next time you launch Logic, the program will ask you where the folder with the Waves plug-ins is located. Anytime you launch the program after you have told it where the folder is, it will automatically display the individual Waves plug-ins directly in the Plug-in selection menu.



The last TDM-Plug chosen will be displayed with a dot

6.5 Logic's Native Effects

This section deals with Logic Audio Platinum's own, proprietary real time effects. You'll learn about the integration of plug-ins by other manufacturers in the section *Plug-ins from Other Manufacturers* on page 6-19.

Some of the parameters shown are only available if you choose the simple display mode. You can reach this mode with the Editor switch on top of the Plug-in window.

General Attributes

Each of Logic's native plug-ins come with several preset settings, for instant use or as a starting point for your own edits. These settings are ordinary settings files in the folder named "Plug-in Settings" (see above), in the same location as your Logic Audio application program. Inside this folder are additional folders for each native plug-in type. You can simply add your own settings to the appropriate folder; it will then appear as a preset in the arrow menu in the title bar of the corresponding plug-in windows. This way you can build up your own effect program library, which can be accessed from all songs.

Rotary Controls

There are two ways to manipulate the rotary knobs:

1. Click on the center and drag the mouse up and down. This option lets you access all available value increments.
2. Click on the knob and move the mouse in a circular motion. Here the program responds to the angle of rotation. You'll find it easier to fine-tune values when the circular motion has a greater radius-in other words, drag it in bigger circles.

GeneralParameters

If you m-click any parameter, this resets it to a neutral value.

Mix

“Mix” defines the proportional mix of the original (dry) and effected (wet) signals.

If the effect is being fed from a bus, it nearly always makes sense to set the mix proportion to 100% (default for inserts in bus objects). Only then can the send knob on the individual channel use the entire possible range. Negative Values stand for a reversed phase.

Region Gate

“Region Gate” lets you define when the calculation of the effect should cease after the end of the last region. Short settings can save processing power, if the effect is not required for long passages in the Arrange window. During these passages the processing power saved could be made available for another effect. Longer settings may be needed if you find that the “tails” of reverbs or delays are getting cut off after regions end.

You can also use the Region Gate parameter as an effect parameter in and of itself, for creating gated reverb delay effects.

Introduced with Version 4.0

Update customers take note

The real-time effects of Version 4.0 have been fundamentally redesigned, and now supplant the real-time effects found in earlier versions. The only exception is the earlier version’s reverb plug-in, still available as “3.0 Reverb”. Although the quality of the new reverb algorithms is better by any objective measure, we nevertheless left the old reverb in the program’s arsenal of sound-sculpting tools, simply to give you more options to chose from.

To maintain the sonic integrity of your songs, the updated version does not tamper with your old songs. They remain fully intact, and when opened under Version 4.0 are played back with the old effects. However, when you want to insert plug-ins, only the new effects appear in the selection list. You might ask “Why?”: Simply because they sound better, and significantly enhance performance.

Autoload Song

If you have preconfigured effects plug-ins for your Autoload song, we recommend that you replace these with the new algorithms. If you have assigned equalizers on any channels, be sure to select the new algorithm for each EQ band, or insert a Fat EQ instead. Your best bet for replacing old flanger and chorus effects is the new Modulation Delay (Mod/Delay). Choose the new Tape Delay (m/m) or Stereo Delay (s/s) in place of the old delay, replace your basic reverb with a SilverVerb, and your main reverb with a GoldVerb or PlatinumVerb. If your machine is not one of the most powerful on the market, you can still achieve good performance, and good sound quality, even when you use a SilverVerb as your main reverb.

The Effect Algorithms

Please note that the entire text describing the parameters for each plug-in is available as from an on-line help menu, accessible from the “arrow” flip menu in the plug-in’s editor window. Familiarize yourself with the general capabilities of the plug-ins now, and refer to the help if you need a refresher on what any particular parameter does, later.

Fat EQ

Similar to the bass and treble controls on a stereo system, equalizers let you shape or voice the tone of a signal. The difference is, of course, that an EQ gives you much more precise and powerful options for influencing a sonic image.

The extremely high-quality Fat EQ offers up to 5 fully parametric bands—buttons 1 through 5 let you activate these individually; inactive bands would otherwise drain your computer's resources.

The icons above the graphic display let you determine whether Band 1 acts like a high-pass filter or a low shelving EQ. Similarly, Band 5 can be switched back and forth between a low-pass filter and a high shelving EQ. Bands 2 and 4 can be switched from their normal operating mode as fully parametric bell EQs to low or high shelving EQs. The center band (No. 3) always operates as a fully parametric bell EQ.

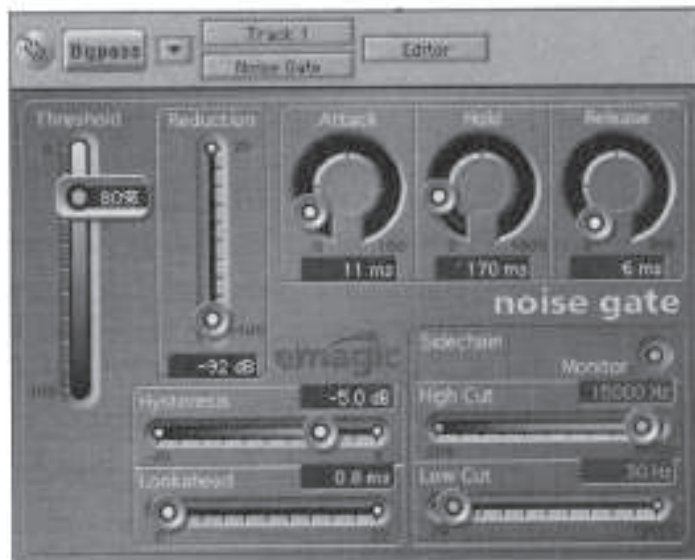
Go to the area directly below the graphic display depicting the frequency response curve, to select the frequency for the individual bands. You don't have to click on the arrows to change values in increments. Simply click on the number, and change the value with your mouse. You'll be able to hear, and thus audition a frequency better if you rotate the *Cut/Boost* knob located below it clockwise to turn it up. The same holds true for any frequency that you want to attenuate. Once you've located the frequency that you're hunting for, you can back off the **Cut/Boost** knob and set it to the desired value. Use the Q (band-



width) parameter located in the bottom display to determine to what extent the band influences neighboring frequencies. At low Q values, the EQ influences a wider frequency range, and at high Q values, the effect of the EQ band is limited to a very narrow frequency range. Please bear in mind that your perception of an attenuated or boosted frequency depends on the Q factor: If you're working with a narrow frequency band, generally you'll have to cut or boost it more drastically to notice a difference.

Noise Gate

Ordinarily, a noise gate suppresses unwanted noise that may become audible during a lull in the signal. However, you can also use it as a creative sound-sculpting tool.



Here's the basic principle behind a noise gate: Signals that lie above the threshold are allowed to pass unimpeded ("open" gate). Anything below the defined threshold (e.g. background noise, crosstalk from other signal sources, etc) is fully muted, "closed" gate). In other words, the Threshold slider determines the lowest level that a signal must have to open the gate—it separates the wanted or useful signal, from the unwanted or noise signal.



Actually, an open gate lets a mix of the wanted and unwanted signals pass. Fortunately, we all benefit from a nifty effect called "masking"—the noise signal is camouflaged by the louder useful signal, so that the listener is no longer able to perceive the background noise. You'll find that in practice this generally does the trick—the undesirable noise is rendered inaudible.

The *Reduction* slider lets you control the intensity of noise suppression. As a rule, you should set it to the lowest value and leave it there to make sure that the gate closes completely. If you prefer, you can dial in other values, so that the noise signal is reduced less dramatically. As an alternative, you can even boost the signal by up to 20 dB.

The three rotary knobs at the top influence the dynamic response of the noise gate. If you want the gate to open extremely quickly, say for percussive signals such as drums, set the *Attack* knob to the lowest value, by turning it as far as it will go counter-clockwise. If the signal fades in a bit more softly, as is the case with string pads and the like, a noise gate that opens too quickly can wreak havoc with the signal, causing it to sound unnatural. For this type of sonic scenario, set the *Attack* knob so that the gate emulates the attack of the original signal. Much the same holds true for the Release phase of signals. When you're working with signals that fade out gradually or have longer reverb tails, you should turn the *Release* knob up so that the noise gate allows the signal to fade naturally. The Hold knob determines the minimum amount of time that you want the gate to stay open. This knob lets you avoid the dreaded "chattering" effect, caused by a rapidly opening and closing noise gate. The Hysteresis slider gives you another option for avoiding chatter, without dialing in a minimum Hold time. Let's back up a bit for a brief explanation: Noise gates often begin chattering when-during the attack or release phase-the level of a signal fluctuates slightly but very rapidly. Instead of clearly exceeding or falling short of the threshold value, the level hovers around the threshold instead. The noise gate then rapidly switches on and off to compensate, producing the undesirable chattering effect. If, however, you were able to tell the noise gate to open at the determined threshold level and remain open until the level drops below another, lower pre-defined threshold, you'd be able to avoid chatter, as long as the sonic window formed by these two thresholds is large enough to contain the fluctuating level of the signal. This is exactly what this feature enables you to do-the value determined by

the *Hysteresis* slider is actually the difference between the Threshold values that open and close the gate. This value is always negative. Generally, -6dB is a good place to start.

If you're dealing with audio material featuring extremely sensitive transients, or attack phases that are critical to the overall sound, you may find it beneficial to have the noise gate open up a tad before the useful signal fades in. This is what the *Lookahead* slider is designed for. The program analyzes the signal level ahead of time, and anticipates the point at which it can open the gate before the signal actually reaches the threshold value. When you chose to use this feature, please make sure you set the *Attack*, *Hold* or *Hysteresis* controls to appropriate values.



As a rule, Lookahead only works with digital noise gates. Actually, at this point in time, the feature only works properly-i.e. without causing undesirable latency of the useful signal-in a native processing environment.

When you're working with noise gates, you'll run across scenarios where the useful signal and the noise signal have levels that are near enough to be perceived as identical. A typical example is the crosstalk of a hi-hat-its signal tends to bleed into the snare drum track when you're recording a drum set. If you're using a noise gate to isolate the snare, in many cases the hi-hat will also open the gate. To avoid this effect, the on-board noise gate ships with sidechain filters.

A sidechain is pretty much what it soundslike: the signal is tapped, routed out and analyzed to find out whether or not it falls below or exceeds the threshold. In other words, the sidechain signal is used to control the noise gate.

When you press and hold the *Monitor* button, you can audition the sidechain signal. Then you can set the filters so that only those frequencies at which the useful signal is particularly loud are allowed to pass. For our example, we'll use a high-cut filter that only allows the bottom end and mids of the snare to pass, and cuts the higher frequencies of the hi-hat. Now when you switch sidechain monitoring off, it will be much easier to dial in a suitable threshold. This will be a value that is only exceeded by the level of the louder useful signal-in our example the fre-

quencies that make up the snare's fundamental tone. Put simply, the noise gate only allows the sound of the snare to pass. Should the need arise, you can follow much the same procedure to isolate a kick or snare drum within an entire mixdown.

Compressor



A compressor tightens up the dynamics of a signal. This means that the difference in levels between loud and soft passages is reduced. With the peak remaining constant, the overall loudness—the perceived volume—of a track is increased. Next to an EQ, a compressor is your most valuable sound-shaping tool when you're mixing. A compressor is a universal effect, it has a virtually unlimited range of applications. You should definitely exploit it for vocal tracks, but a compressor can also often work wonders for entire mixes. When you use a compressor, be sure to route the entire signal through it, by inserting it directly into tracks. It should only be used in a bus when you want to compress a group of tracks (e.g. drums) simultaneously, by the same amount. Again, these should be routed to the bus in their entirety, as opposed to using Send knobs to route just part of each signal to the bus. You do this by selecting the appropriate Bus as the output destination for the tracks that you wish to compress.

The Emagic Compressor was designed to emulate the response of the finest analog compressors. It follows the following prin-

ciple: When a signal exceeds the defined threshold level, the compressor actually alters the response, so that is no longer linear. Instead, all levels that exceed the threshold are attenuated by the value that you set with the Ratio slider. A ratio of 4:1 means that an incoming level that lies 4 dB above the threshold is dampened, so that it comes out the other end of the compressor with a level just 1 dB above the threshold. On the flip side, if you route in a signal that is loud enough to double the output level of the compressor (+ 6dB), the input signal would have to have a level 24 dB greater than the threshold level. This tells us that a compression ratio of 4: 1 is a fairly drastic manipulation of the original signal's dynamics. Since the compressor lowers levels, the volume of its output signal is normally lower than that of the input signal. To compensate for this decrease in levels, the output of the compressor is equipped with a *Gain* slider. *Auto Gain* automatically sets the level of amplification to a value equivalent to the "sum of the threshold value minus the threshold value divided by the ratio" or put less confusingly $T - (T/R)$. This function ensures that a normalized input signal is amplified so that the output signal is also normalized, regardless of the values that you set for threshold and ratio-provided you are dealing with relatively static signals. Use the *Attack* and *Release* knobs to shape the dynamic response of the compressor. *Attack* determines the amount of time it takes for the compressor to react to signals that exceed the threshold. At higher values, the compressor does not fully dampen a signal until after it runs through its attack phase. This type of setting ensures the original attack, for example the sound of a pick or finger striking a guitar string, remains intact or clearly audible. If, on the other hand, you want to maximize the level of a master signal, set the *Attack* knob to low values; i.e. so that the compressor responds more swiftly. *Release* determines the amount of time it takes for the compressor to stop dampening louder passages, once the signal level falls below the threshold. If the compressor generates an ugly pumping sound, adjust the *Release* knob accordingly.

When you have configured a compressor so that it dampens the signal at or above the threshold value by the predetermined ratio, while the level just below the threshold is routed through at a 1: 1 ratio, an audio engineer would term the compression as “hard knee.” However, in many cases you’ll come up with a better sounding track by using a more gradual transition from the 1:1 ratio below the threshold, to the ratio that you entered for levels above the threshold. Here the characteristic curve is not as radical—it rises gradually from the bottom left to the top right, as seen in the graphic display. This type of compression is called “soft knee.” The *Knee* slider lets you dial in anything from hard to soft, and provides a useful number of increments. This wide range of options gives you the tools to shape the sound as you like, depending on whether you want to radically maximize loudness with absolutely no regard for the original dynamics (hard), or are going for the more musical compression that acoustic recordings typically require (soft). Keep in mind that *Knee* controls only the shape of the compression, not its intensity; use the Threshold and *Ratio* sliders for this purpose.

Incidentally, the *Gain Reduction Meter* indicates the intensity of compression with which you’re tightening up the original signal. This feature is a great help, particularly if you are not experienced with using compression. Keep an eye on it to make sure you’re not over compressing your tracks.

When the compressor has to decide whether or not the level exceeds the threshold (or for soft knee compression, if the level is getting close to the threshold), it can analyze either the peak or RMS level. The latter value is a better indication of how we perceive loudness. When you use the compressor primarily as a limiter, select the *Peak* button. When you’re compressing individual signals, use of the RMS button will often deliver better, more musical results.

If you activate *Auto Gain* and RMS simultaneously, the signal may be saturated. If you hear any distortion, switch *Auto Gain* off, and enter a suitable gain level manually.

Despite all of these handy tips for tweaking sounds, you should always keep one thing in mind-there are no hard and fast rules. Use your own taste and ears. If it sounds good, it is good.

Expander



The Expander is similar to the Compressor, with one fundamental difference-it increases, rather than reduces, the dynamic range above the threshold. The Ratio slider features a value range of 1:1 to 0.51. This means that the Emagic Expander is a genuine “upward expander” (as opposed to a “downward” expander that increases the dynamic range below the threshold). You can use this effect to emphasize the transients of highly compressed signals. This spices up the sonic image, making it sound livelier and fresher. Please bear in mind that you will perceive the signal as being softer, even when the peak level remains the same. In other words, the expander decreases loudness. If you manipulate the dynamics of a signal fairly radically (depending on the threshold and ratio values), you’ll find that you’ll have to back off the level via the Gain slider to avoid distortion. In most cases, Auto Gain will take care of this for you.

Envelope



The Enveloper is an unusual tool that lets you shape transients, i.e. the attack and release phases of signals. No other type of dynamic effect (compressor, expander) can achieve similar results-and they can be quite impressive indeed.

The most important Enveloper controls are the two **Gain** sliders that govern **Attack** (left) and **Release** (right). In the center position, the signal remains unprocessed. If you turn the gain up, the attack or release phase is emphasized. If you turn it down, the corresponding phase is attenuated. For example, boosting the attack lends a drum sound more snap, or amplifies the sound of a guitar string being plucked or picked. When you cut the attack, percussive signals fade in more softly. You can also mute the attack so that is virtually inaudible. This lets you come up with all kinds of interesting manipulations. Another handy application for this is maintaining friendships-it allows you to mask the poor timing of accompanying instruments, rather than tell your pals that they have all the groove of a horde of accountants at the office Christmas party.

Emphasizing the release also boosts the amount of any reverb on the affected track. Conversely, when you tone down the release phase, tracks originally drenched in reverb end up sounding drier. This effect is particularly useful when you're working with drumloops, but of course there are many more applications. Let your imagination be your guide.

When using the Expander, you should set the *Threshold* to the minimum value and leave it there. Only when you seriously crank the release phase, thus boosting the noise level of the original recording to a very high level, should you turn the Threshold slider up a bit to limit the Enveloper so that it influences the useful signal only.

Drastic boosting or cutting of the release or attack phase may change the overall level of the signal. The *Out Level* slider allows you to compensate for this effect manually.

The *Time* parameters for the attack and release phase (2 knobs below the graphic display) enable you to access the time-based intervals that the program interprets as the attack and release phases. Generally, you'll find values of around 20 ms (attack) and 1500 ms (release) are fine for starters. Adjust them according to the type of signal that you're processing.

Similar to its counterpart on the noise gate, the *Lookahead* slider lets you enter values that tell the Enveloper to anticipate what the signal will do in the very near future. Normally you won't need this feature, except possibly for signals with extremely sensitive transients. If you do decide to employ *Lookahead*, you may have to adjust the attack time accordingly.

To give you better insight into true nature of the Enveloper, here's a quick look at how it works: It is equipped with two internal envelope followers. One follows the amplitude of the input signal directly, whereas the other follows all changes generated by the variable delays (individually adjustable for attack and release). The difference between the two envelope followers is used to boost or cut the original signal by way of the corresponding *Gain* sliders (also individually adjustable for attack and release).

In contrast to a compressor or expander, the Enveloper operates independently of the absolute level of the input signal-provided the Threshold slider is set to the lowest possible value).

Volume

This plug-in enables you to influence the level and phase of the signal.

Norm	unchanged phase,
Inv	inverted phase.



The stereo version of this plug-in lets you invert the phase position of the two stereo sides separately:

L- R+	left inverted, right unchanged,
L+ R-	right inverted, left unchanged.

When you invert the phase of a signal, it sounds identical to the original. Only when the signal is heard in conjunction with other signals does phase inversion have an audible effect. For example, if you mic a snare drum from the bottom as well as from the top, you should invert the phase of the bottom microphone's signal so that it is in phase with the top mic signal.

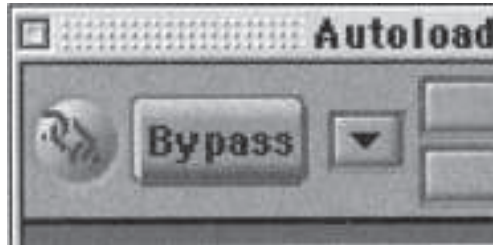


The phase of condenser microphones is generally opposite to that of dynamic microphones. If you are using one of each of these two types of microphone to record a snare, you won't have to invert the phase of one of them. On the other hand, if you mic a sound source from the same direction using two different types of microphones, you will have to invert the phase of one of the mics. This is done to avoid cancellations when the signals are mixed together.

Influencing levels with “Volume” makes sense when you're working with automated tracks during post-processing, and you want to quickly adjust master levels. This could be the case when you've inserted an additional plug-in that does not feature a dedicated gain knob, or you want to change the basic level of a track for a remix version.

Direction Mixer (DirMixer)

In our humble opinion, the Direction Mixer is a component of mixing consoles that undeservedly lost favor with the audio world. For decades it was a vital tool, particularly for German audio engineers, who used it to come up with great stereo recordings.



The Direction Mixer plug-in offers the following features:

MS Decoder

The option of influencing the stereo base

Variable pan positioning of a stereo recording

Input

This parameter is used to determine whether the input signal is a standard “left/right” signal, or if you’re dealing with an MS encoded (middle and side) signal, for example when the two sides of an MS stereo mic setup were recorded directly.

Direction

This parameter determines the direction from which the middle of the recorded stereo signal will emanate from within the mix, or in less complicated terms, its pan position.

At a value of "0", the middle of the stereo recording will be dead center within the mix. Positive values shift the middle of the stereo recording towards the left, negative values towards the right. At 90° the middle of the stereo recording is panned hard left, at -90° hard right. Higher values shift the recorded signal back to the center of the stereo mix, except that the stereo sides of the recording are swapped. At values of 180° or -180°, the middle of the recording is yet again dead center of the mix, although the left side of the recording is audible on the right side of the mix, and vice versa.

Base

This parameter determines the spread of the stereo base. At a neutral value of "1", the left side of the signal is positioned precisely on the left, and the right side precisely on the right. As the values decrease, the two sides increasingly move towards the center of the stereo image. A value of "0" produces a mono signal (both sides of the input signal are routed to the two outputs at the same level-i.e. a true middle signal). At values greater than 1, the stereo base is extended out to an imaginary point beyond the spatial limits of the speakers. In terms of MS levels, this is an involved way of saying that the level of the side signal is increased so that it is higher than the level of the middle signal. At a value of "2", only the side signal remains audible (on the left you'll hear "L-R" and on the right "R-L").

If you chose to use the Direction Mixer simply to spread the stereo base, please keep in mind that as the Base values increase beyond "1", monaural compatibility decreases accordingly. Once you process a stereo signal with an extreme setting of "2", when you play back the track in mono, the signal will be cancelled out completely-after all, "L-R" plus "R-L" doesn't leave you with much.

What is MS?

Relegated to obscurity for a good long while, recently MS stereo ("middle-side" as opposed to "left-right") has enjoyed a renaissance of sorts. Here, two microphones are stacked on a stand or suspended from the ceiling so that they are positioned as close together as possible. One of the two is a cardioid (or omnidirectional) mic which faces the sound source that you want to record directly, i.e. in straight alignment. The other is a bidirectional mic, the sensitive axes of which point to the left and right at 90° angles. The cardioid mic delivers the middle signal, the bidirectional mic the side signal. Simply record the middle signal to the left side of a stereo track and the side signal to the right. This configuration enables the Direction Mixer to easily decode MS recordings. The advantage that MS recordings have over XY recordings (two cardioid mics aligned so that

they are directed to a point halfway to the left and right of the sound source) is that the stereo middle is actually located on the on-axis (main recording direction) of the cardioid mic. This means that slight fluctuations in frequency response that occur off the on-axis-as is the case with every microphone-are less troublesome.

To avoid confusion and unpleasant surprises, please be sure to label MS recordings by adding “MS” to their names.

In principle, MS and LR signals are equivalent and can be converted at any time. When “-” signifies a phase inversion, then the following applies:

$$M = L + R$$

$$S = L - R$$

In addition, L can also be derived from the sum of and R from the difference between M and S. Here’s some interesting trivia for you: Radio (FM) broadcasts feature M and S stereo. The MS signal is actually converted to a signal suitable for the left and right speakers by the receiver.

Dither

When lower-order bits of high-resolution digital audio signals are lopped off-in audio jargon, truncated-it causes low-level signals to sound grainy when you’re monitoring them at higher volumes. These fairly ugly effects are most audible towards the end of a reverb tail. “Dithering” or “Noise Shaping” are processes that are used to mask these effects by adding noise to the signal.



The Dither plug-in is recommended for use as the final plug-in in an output object whenever the following two prerequisites are met:

The signal is routed out in digital form.

The signal is routed out via a card or recorded to a medium with a resolution lower than 24 bits, e.g. a 16-bit DAT recorder.

Please keep in mind that once you have patched a signal through the Dither module, you've reached the end of the line is far signal processing goes—you can't even adjust the Master Fader or Pan knob. If you want to program a fade out, please insert a "Volume" plug in front of the "Dither" plug-in.

Distortion Effects

Distortion

This distortion effect simulates the lo-fi dirt generated by a bipolar transistor. Move the *Drive* slider up to increasingly saturate the transistor. Generally, the distortion created by the plug-in tends to increase the signal level, an effect that you can compensate for with the Output slider. The *Tone* knob lets you filter the harmonics-laden distortion signal, which then delivers a somewhat less grating, softer tone. The Emagic Eye is watching—it visually represents the settings of the *Drive* and *Tone* parameters.



Overdrive

The Overdrive effect emulates the distortion of a field-effect transistor (FET). When saturated, FETs generate warmer-sounding distortion than bipolar transistors. Here too, the *Drive* slider pushes the transistor over the edge and into overdrive. Generally, the distortion created by the plug-in tends to increase signal levels, an effect that you can compensate for with the Output slider. The *Tone* knob lets you filter the harmonics-laden distortion signal, which delivers an even warmer sound. Once again, the Emagics Eye visually represents the settings of the *Drive* and *Tone* parameters.



BitCrusher

BitCrusher is the ultimate digital distortion box. You can do all kinds of wild stuff with it, such as recreate the 8-bit sound of the pioneering days of digital audio, create artificial aliasing by dividing the sample rate, or distort signals so radically that they are rendered unrecognizable. Warning: The BitCrusher can damage your hearing when operated at high volumes...



The *Drive* slider lets you boost the level at the input of the BitCrusher. Please note that this tends to excite the clipping stage located at the output of the BitCrusher as well.

The *Resolution* knob lets you reduce the resolution from 24 bits down to 1 bit.

The number of bits is always an exponent of two. The range of available values is equivalent to the exponents of two that a given sample rate can handle. For example, whereas 65,536 different values are possible for 16 bits, at 8 bits, you're left with just 256. The sonic image becomes ever more ragged as the values decrease because the number of sampling errors increases, thus generating more distortion. At extremely low bit resolutions, the amount of distortion can be greater than the level of the usable signal.

The *Downsampling* slider lets you lower the sample rate. For example, at a value of 2, the original 44.1 kHz signal is sampled at a rate of just 22.05 kHz. With a factor of 10, the rate is knocked all the way down to 4.41 kHz.



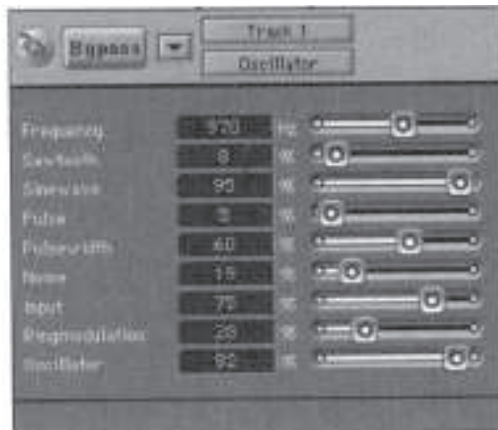
The Nyquist theorem of digital signal processing states that, as a rule, frequency data up to half the value of the sample rate remains intact. Any frequencies above this value are lost. However, the BitCrusher is not equipped with a low-pass filter designed to filter frequencies that lie above the value equivalent to half the sample rate. Indeed, you could use a simple low-pass filter for this purpose. Nonetheless, there is method behind the madness—it is precisely the distortion generated by these alias tones (hence the name aliasing that makes this effect so unusual. “What does it sound like,” you ask? Something similar to a short-wave radio with a station dialed in just a tad off the mark required for decent reception.

The *Clip Level* slider lets you define at what point below the normal threshold you want the signal to start clipping. The *Mode* buttons are used to determine whether the signal peaks that exceed the clip level are “Folded,” “Cut,” or “Displaced” (check out the graphics on the buttons and the resulting waveform in the display). The kind of clipping that occurs in standard digital systems is usually closest to that of the center mode (Cut). Internal distortion may generate clipping similar to the types generated by the other two modes.

Tone Generation, Ring Modulation

Oscillator

The Oscillator generates a static frequency, as determined by the *Frequency* parameter. The level of the oscillator is controlled by the Oscillator parameter, whereas the level of the original signal of the audio object is controlled by the Input parameter.



An oscillator wouldn't be much fun if all it could do was produce an interminable sine tone. One of the redeeming features that spares us equally interminable boredom is the ring modulator-it produces side band frequencies. The frequencies of these side bands depend on the original signal of the audio object, and the signal of the oscillator. In technical terms, the ring modulated signal is the product of real-time multiplication of the two signals. If you analyzed the resultant ring modulated signal, you'd find that this signal consists of the original signal as well as the sums and differences of its frequencies.

Here's an example: Let's say the audio object plays back a 100 Hz sine wave, and the oscillator generates a frequency of 1000 Hz. Ring modulating these signals generates side bands with 1200 Hz (sum) and 800 Hz (difference). The generated spectrum is not necessarily what most people would call harmonic. This is why ring modulators are often considered good tools for shaping enharmonic metallic sounds, bells being one of the more conventional sounds that they can produce. Bear in mind that the signal of your audio object will be much more complex than a simple sine wave. Moreover, the oscillator is no slouch either-it can do more than just generate sine waves. Consider the oscillator an additional source of distortion, rather than a simple perpetual note generating component.

Here's a quick run-down on the parameters: Frequency determines the frequency of the oscillator. The Frequency can either be adjusted in Hz or as "Note" and "Octave", with variable tuning "tune". You can switch between both modes, using *Freq. Source*. *Sawtooth* sets the level of the sawtooth wave, the spectrum of which contains all harmonics. *Sine Wave* defines the level of the sine wave, and *Pulse* the level of the pulse/square wave. *PulseWidth* controls the ratio of positive and negative pulses. A value of 50% generates a square wave, the spectrum of which consists solely of odd harmonics. Noise adds low-frequency noise. Input controls the level of the original signal of the audio object, *Ring Modulation* the level of the ring modulator, and *Oscillator* the level of the pure oscillator signal.

Delay and Pitch

Tape Delay

The Emagic Tape Delay simulates a vintage tape echo device, although with some very useful features that such old devices never offered. The first of these is that its delay settings are variable in musical increments. It is equipped with a high- and low-pass filter in the feedback circuit, as well as a circuit that simulates tape saturation effects. This plug-in is ideal for the dub delays invented by Jamaican toast masters, and used in many styles of music today.



Here you won't be wasting your time computing delays and typing in the values in milliseconds. Instead, all you have to do is enter the desired note value directly. Simply click on the desired button for half, quarter, eighth or sixteenth notes.



When it is computing the desired delay, the plug-in uses the internal tempo of the sequencer. Tempo information is updated in the plug-in window when you open it, and later, every time you execute a mouse operation. Please note that at this point the plug-in can't handle tempo changes. This is because we wanted to avoid the noise that is generated when a delay is switched. The Tempo box serves solely to display the current bpm value—you can't use it to change the tempo of the sequencer.

When you want to create dotted note values, move the Groove slider all the way to the right to “75%”; for triplets, select the jetting “33.33%”. Note that all intermediate values are possible. Conveniently, you can view the current delay value in the *Delay* display.

As you might expect, the *Feedback* slider determines feedback intensity; in other words, the amount of delayed and filtered signal that is routed back to the input of the Tape Delay. When you set it to the lowest possible value, the Tape Delay generates just a single echo. Turned all the way up, the echoes are repeated ad infinitum. Keep in mind that the levels of the original signal and its taps (echo repeats) tend to add up, and may cause distortion. Here's where the internal tape saturation circuit comes to the rescue—it can be used to ensure that these overdriven signals sound good.

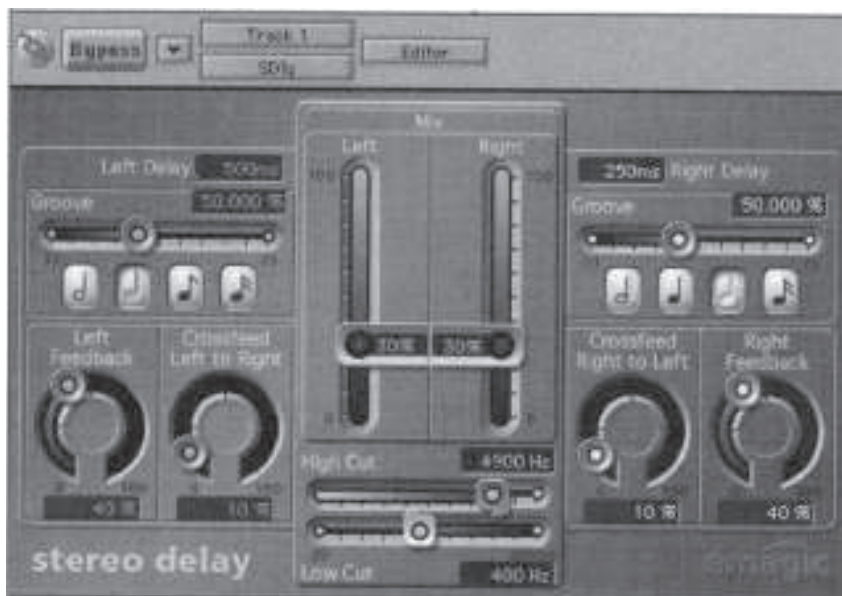
You can shape the sound of the echoes, using the on-board high-cut and low-cut filters. Although these filters are fairly flat, they're not located post-output. Instead, they are located in the feedback circuit, meaning that the effect achieved by these filters increases in intensity with each repeat. If you're in the mood for an increasingly muddy tone, move the high-cut filter slider towards the left. For ever thinner echoes, move the low-cut filter slider towards the right.

The *Mix* slider determines the balance between the original (dry) signal and effects (wet) signal. If you've inserted the Tape Delay to an individual track, you'll generally find that settings of up to 50% are desirable. If the Tape Delay is patched to the insert of a bus object, and you're routing the signals of a track to the plug-in with the *Send* controls, you should set the *Mix* slider to "100%", and leave it there.

If you're unable to hear the effect, even though you've set up a suitable configuration, be sure to check out not only the *Mix* knob, but also the filter settings: Move the High-cut filter slider to the far right, and the *Low-cut filter* slider to the far left.

Currently, there is only one Tape Delay (m/m), hence the monaural input and output. A Stereo Delay is your best bet for stereo tracks or busses.

Stereo Delay

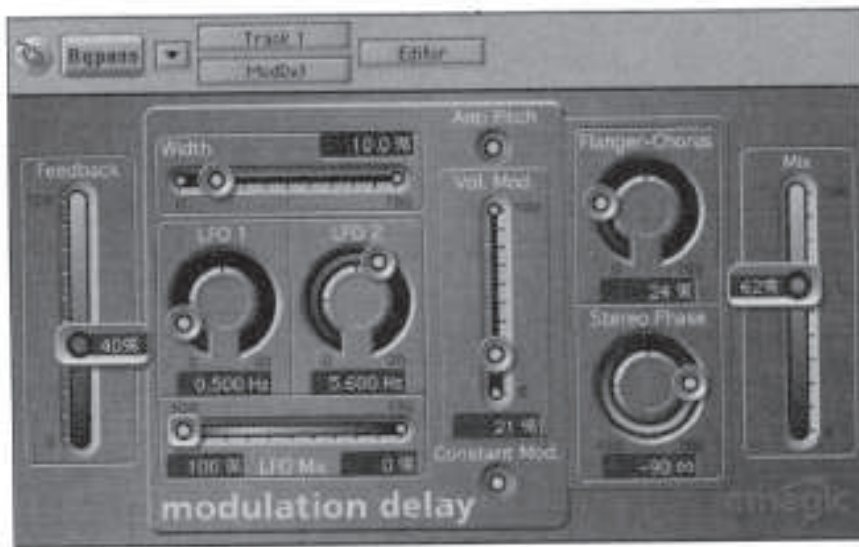


The Stereo Delay works much like the Tape Delay, which is why we'll skip the general info, and take a closer look at the differences between the two. There is just one Stereo Delay (s/s), hence the stereo input and output. You are free to use the Stereo Delay for monaural tracks or busses, when you want to create independent delays for the two stereo sides. Please bear in mind that if you use this option, the track or bus has two channels from the point of insertion forward. Unlike the Tape Delay, the Stereo Delay does not feature a circuit that replicates tape saturation.

You can set the *Delay* (using Note buttons and Groove sliders), *Feedback* and *Mix* values separately for the two sides. However, the settings for the *High-cut* and *Low-cut* filters apply to both

sides equally. In addition, the plug-in features a **Cross Feed** knob for each stereo side-it determines the feedback intensity; i.e. the level, at which each signal is routed to the opposite stereo side.

Modulation Delay



As its name implies, the Modulation Delay generates effects such as flanging or chorus, based on modulated short delays. It can also be used-without modulation-to create resonator or doubling effects.

The modulation section consists of two LFOs, with variable frequencies (0 to 20 Hz). The balance between these two is determined by the **LFO Mix** slider. Use the **Width** slider to enter the desired modulation width. When the Width slider is set to the far right position, delay modulation is switched off completely. The **Volume Modulation (Vol. Mod.)** slider lets you determine the intensity of the amplitude modulation (Tremolo). The **Constant Modulation (Constant Mod.)** button lets you do just that-ensure that the modulation width remains constant, regardless of the modulation rate. When this feature is switched off, higher modulation frequencies reduce the modulation width. In simple delay circuits, a delay modulation normally also modulates the pitch of the signal. Use the **Anti Pitch** button

to ensure the pitch of the modulated signal remains constant. This is exactly how high-end chorus and flanger effects work.

Set the basic delay time with the *Flanger-Chorus* knob. Set to the far left position, the Modulation Delay puts on its flanger cap. As you move towards the center position, it thinks it's a chorus. As you move the knob closer to the far right position, you will hear clearly audible delay taps. This latter type of setting is generally used without modulation (Width = 0), for doubling effects.

The *Stereo Phase* knob defines the phase of the modulation between the left and right stereo sides. At 0° , the extreme values of the modulation are achieved simultaneously on both sides, at 180° , the extreme values opposite each other are reached simultaneously.

The *Feedback* slider controls the intensity at which the effect's signal feedback is routed to the input. If you're going for radical flanging effects, enter a high *Feedback* value, if simple doubling is what you're after, you don't want any feedback at all. The Mix slider determines the balance between dry and wet signals.

Phaser



The Emagic Phaser emulates the effect of analog phaser circuits with from 4 to 1'2 orders (as in 4th order, 5th order etc.) Use the Order slider to set the desired number of orders. Note

that you are free to select odd numbered settings (5, 7, 9, 11) which, strictly speaking, don't generate actual phasing. However, the more subtle comb filtering effects produced by odd numbered settings can come in handy on occasion. As a rule, the more orders a phaser has, the heavier the effect. The settings "4, 6, 8, 10 and 12" put five different phaser algorithms at your fingertips, all of which replicate the analog circuits that they are modeled on, each designed for a specific application.

The modulation section comes complete with two LFOs, featuring individually variable frequencies, and freely variable mix options. Additionally, the frequency of LFO 1 can be modulated by the level of the input signal. Use the *Envelope Modulation* slider to set the desired modulation intensity. By staking out the limits of the modulation with its highest and lowest values, you can determine the modulation width and range. These high/low limits are controlled by the *Sweep Ceiling* and *Sweep Floor* sliders-you can enter values for them directly in the form of the desired frequency. This value also determines the maximum intensity of the comb filtering created by the phasing effect.

The *Stereo Phase* knob is used to define the phase for the left and right channels of a stereo phaser (s/s). When you're using a monaural phaser, this parameter is, of course, meaningless.

Negative Values of *Mix* result in a phase inverted mix of effect and direct signal.

As the icing on the phasing cake, you can tweak the *Color* slider to add just that to the effect. Here, the comb filtering effect is amplified via feedback.

Pitch Shifter

The first thing you may notice about the Pitch Shifter is its minimalist approach—just a few parameters are available.

Semi Tones is used to set the transposition in semi-tone increments within a range of one octave upwards or downwards. *Cents* controls detuning in increments equivalent to 1/100th of a semi-tone step.

Use the *Mix* slider to control the desired balance between the original and processed signals.



The *Drums*, *Speech* and *Vocals* buttons are used to set internal parameters, so that the program delivers the best results for a given application. When you select *Drums*, the groove of the original track remains intact. With *Vocals*, the intonation of the original is retained unaltered. Hence *Vocals* is well-suited for any signals that are inherently harmonic or melodious; for instance, string pads. *Speech* mode is a compromise between the two—the program attempts to retain both the rhythmic and harmonic aspects of the signal, which is desirable for complex signals, such as spoken-word recordings or rap music. *Speech* is thus also suitable for other “hybrid” signals, such as rhythm guitar.

When in doubt, *Speech* is a good place to start. A/B the options to compare them, and use the one that suits a given recording best. When auditioning and judging settings for quality, it's a good idea to temporarily turn the *Mix* knob up to 100%. Keep in mind that Pitch Shifter artifacts are a lot harder to hear when you mix a smaller percentage of a transposed audio to the overall signal.

Ensemble



The Emagic Ensemble is like a pitch shifter on steroids—it consists of eight internal, modulatable pitch shifters. Two standard LFOs and one random LFO enable you to come up with fairly complex pitch modulations, which—much like a natural chorus effect—conjure up the impression of an instrumental or vocal ensemble. A big Emagic Eye visually represents the number of voices, and their modulations.

Use the *Voices* slider to determine how many voices (1-8) are generated in addition to the original. Please note that the plug-in's appetite for computer resources increases proportionally to the number of voices: When you activate eight voices, the Ensemble requires roughly eight times the CPU power of a pitch shifter.

The two conventional LFOs and the random LFO (which generates random modulations), each feature a *Rate* knob that controls frequency, and an Intensity slider to determine the modulation width.

The *Phase* knob controls the phase relationship between the modulations of the individual voices. The value that you select

here depends on the number of voices, which is why it is indicated in percentages rather than degrees. The value 100 (or -100) is equal to the greatest possible distance between the modulation phase of all voices. Here the voices are distributed an equal distance apart over the full 360°.

The *Stereo Base* slider serves to distribute the voices across the stereo field. When you set a value of 100%, the stereo base is expanded artificially. Please bear in mind that monaural compatibility may suffer.

In addition to the familiar *Mix* slider that determines the balance of dry and wet signals, the Ensemble also features an *Effect Volume* knob. This lets you determine the level of the effects signal separately. This feature allows you to compensate for changes in volume caused by manipulating the *Voices* parameter.

Gainer

This plug-in allows a constant amplification or damping, by a specific amount, given in decibels, of a track or return object.



By using Phase you can invert the Phase of the signal (swap “Plus” and “Minus”). In this way you can combat time alignment problems, particularly those caused by running multiple microphones at the same time. The stereo version additionally offers a balance parameter.

Sample Delay

This simple plug-in allows the simple delaying of a channel by single Sample values, in the stereo version these are given separately for each channel. Used in conjunction



with the capability of the Gainer to invert the phase, this plug-in is particularly suitable for correcting run-time problems that may occur with multiple channel microphones.

For delay effects, such as echo, or modulation effects, use the Stereo Delay, Tape Delay or Modulation Delay plug-ins.

Every sample, at a frequency of 44.1 kHz is equivalent to the time taken for sound to travel 7.76 Millimeters. Looked at differently: If you delay a channel from a stereo microphone by 13 samples this will acoustically give an increase of the microphone separation of 10 centimeters.

Flanger

Intensity

The modulation amplitude or width. This defines how far from the “Offset” the comb-filter will be modulated.

Speed

The frequency or speed of the modulation. This determines how quickly the comb-filter is modulated.

Feedback

With “Feedback”, the delayed signal is routed back into the input. Negative values mean that the phase of the routed signal is inverted.

You can use “Feedback” to increase the intensity of the flanger, which can produce very dramatic effects.

Chorus

The Chorus effect can be used to make a signal “wider”, particularly the variety with stereo output (m/s).

Offset

The “Offset” of the chorus defines the normal value of the delay time, which is then modulated by the “Speed” and “Width” parameters.

Intensity

The modulation width of the chorus effect.

Speed

The modulation frequency of the chorus effect.

Feedback

With “Feedback”, the delayed signal is routed back to the input, which increases the intensity of the chorus effect. High values can produce an unnatural sound.

Reverberation

Reverb is the bread-and-butter effect, the most prevalently used, and the one you can least do without. In nature, every sonic event that reaches our ears is accompanied by varying amounts of reverb. As a rule, you could say that early reflections off walls and the floor are more significant than the actual reverb tail (diffuse reverberation). When you’re planning a mix, be aware that reverb is perceived as a natural environment for a signal to “live” in, whereas the total absence of reverb is perceived as an “effect”. The complete absence of reverb creates the kind of “in-your-face” proximity that would rarely be considered desirable. A mix in which all signals are perceived an equal distance from the listener lack spatial depth—which is an involved way of saying that it sounds flat or two-dimensional.

Reverb plug-ins should always be patched into busses. Route signals from individual tracks to the plug-in with the respective Send knobs. With this method, you’re making the most your machine’s resources, by using it to compute the reverb for several tracks at once. You can vary the *Send* knob setting, to stagger the signals so they appear to have different spatial depths.

If, however, you want to put a special type of reverb on just one track, you are, of course, free to insert the plug-in directly to it. In this case, use the *Mix* parameter, which is found on every reverb plug-in, to determine the signal’s spatial depth or, put

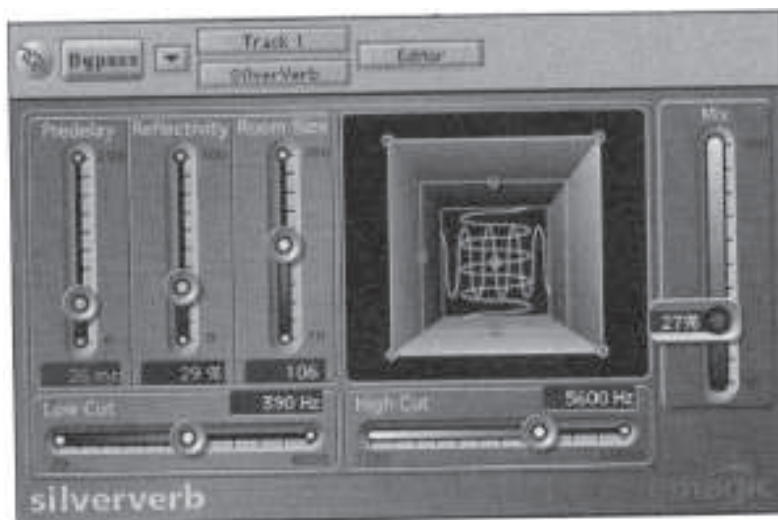
simply, how far back in a room it seems to be. The balance between the dry and wet signals is the single most important parameter when you're working with reverb.



Tip: Often you'll find it beneficial to insert an equalizer (Fat EQ) before or after a reverb plug-in. This type of setup lets you adapt the sound of the reverb to the requirements of the mix. Some audio engineers like to compress the reverb signal to create a tighter sonic image.

When you're trying come up with a great reverb sound using a plug-in, we recommend that you first try out the factory presets. Once you locate a reverb that sounds somewhat similar to what you're after, you can edit it to suit your taste.

SilverVerb



Although the Emagic SilverVerb is based on a simple reverb algorithm, it delivers remarkably good results.

The actual reverb algorithm is controlled by just three parameters: As its name implies, *Reflectivity* defines how reflective the imaginary walls, ceiling and floor will be. *Room Size* challenges your architectural skills-use it to define the dimensions of simulated rooms. The graphic display visually represents these parameter settings.

Pre-delay determines the delay between the original signal and the reverb tail.



Whereas high Pre-delay settings tend to generate something similar to an echo, low values often muddy the original signal. Ideally, you should go for as high a setting as possible before the plug-in begins generating something that sounds like a delay tap. With appropriate pre-delay settings, you can apply relatively generous amounts of reverb to percussive parts, while allowing the attacks to remain intelligible.

Low Cut and ***High Cut*** let you filter bass and treble frequencies out of the reverb tail.



In most cases this will unclutter your mix. The reason for this is that a long reverb with a great deal of bottom end generally makes for a flabby mix, and high frequencies in the reverb usually sound somewhat unpleasant, hamper speech intelligibility, or mask the overtones of the original signals.

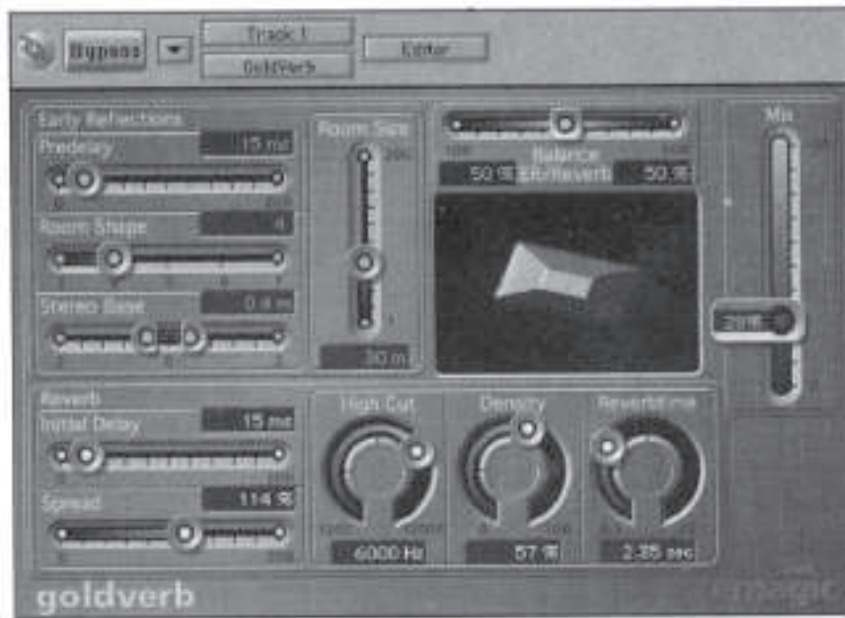
Density/Time controls the density of early reflections, and the reverb time.

Averb

Coming without the EQ parameters ***Low Cut*** and ***High Cut***, the Averb uses the same lean algorithm as the SilverVerb but utilizes even less processing power. The other parameters are the same as with SilverVerb (described above).



GoldVerb



The GoldVerb consists of two sections: *Early Reflections* and *Reverb* (diffuse reverberations). The balance between these two sections is controlled by the *Balance ER/Reverb* slider, located above the graphic. When you set this *Balance* slider to either of its extreme positions, the unused section is deactivated, to maximize performance.

Occasionally, you should try using just the *Early Reflections* section of the GoldVerb for ambient effects. It will be well worth your effort.

Early Reflections

This section emulates the original signal's first reflections as they bounce off the walls, ceiling and floor of a natural room. These early reflections are essential to how we perceive a room. All information about the size and shape of a room capable of being discerned by the human ear is contained in these early reflections.

Pre Delay

Pre-delay is the amount of time that elapses between the original signal, and the arrival of the early reflections. In any given room size and shape, *Pre-delay* determines the distance between the listener and the walls, ceiling and floor. When

used with artificially generated reverb, it has proven advantageous to allow this parameter to be manipulated separately from, and over a greater range than, what is considered natural for *Pre-delay*. In practice, too short a pre-delay tends to make it difficult to pinpoint the position of the signal. It can also color the sound of original signal. On the other hand, too long a pre-delay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, which leaves an audible gap. The ideal pre-delay setting depends on the properties or, more accurately, the envelope of the original signal. Percussive signals generally require shorter pre-delays than signals in which the attack fades in gradually. A good practice is to use the longest pre-delay possible before undesirable side effects, such as an audible echo, begin materializing.

Room Shape

Use this slider to define the geometric form of the room. The numeric value (3-7) represents the number of corners it has.

Room Size

Unsurprisingly, *Room Size* determines the dimensions of the room. The numeric value indicates the length of its walls, i.e. the distance between two corners.

Stereo Base

The *Stereo Base* parameter enables you to define the distance between the two virtual microphones that you are using to audition the simulated room. Spacing the microphones slightly further apart than the distance between two human ears generally delivers the best results. Of course, more realistic results can be obtained if you choose to use the distance between two ears located on opposite sides of the same head.

Reverb

This section generates diffuse reverberation,

Initial Delay

This is the delay between the original signal and the diffuse reverb tail. If you're going for a natural-sounding, harmonic reverb, the transition between the early reflections and the reverb tail should be as smooth and seamless as possible. Basically, what we said about the pre-delay (see the section: Early Reflections) holds true for this parameter:

Set the *Initial Delay* so that it is as long as possible without there being a perceptible gap between the early reflections and the reverb tail.

Density

This parameter controls the density of the diffuse reverb. Ordinarily, you want the signal to be as dense as possible. However, less *Density* means the plug-in eats up less computing power. Moreover, in rare instances, too great a *Density* can color the sound, which you can fix simply by backing off the *Density* knob. If you select a density value that is too low, the reverb tail will sound grainy.

Diffusion

Diffusion of the reverb tail. Sometimes the terms *Diffusion* and *Density* are confused. The density is the average number of reflections in a given period of time. The diffusion is the amount of irregularity of the density. High values of diffusion stand for a regular density, with few alterations in terms of level, times and panorama. With low values of *diffusion*, the reflections' density becomes more irregular and grainy. The stereo spectrum changes, too.

Reverb Time

Reverb Time is commonly considered the amount of time it takes for the level of a reverb signal to drop by 60 dB. This why the time is often indicated as RT60. Most natural rooms have a reverb time somewhere in the range of 1 to 3 seconds, a value which absorbent surfaces and furniture reduces. Large empty

halls or churches have reverb times of up to eight seconds, some cavernous or cathedral-like venues even beyond that.

High Cut

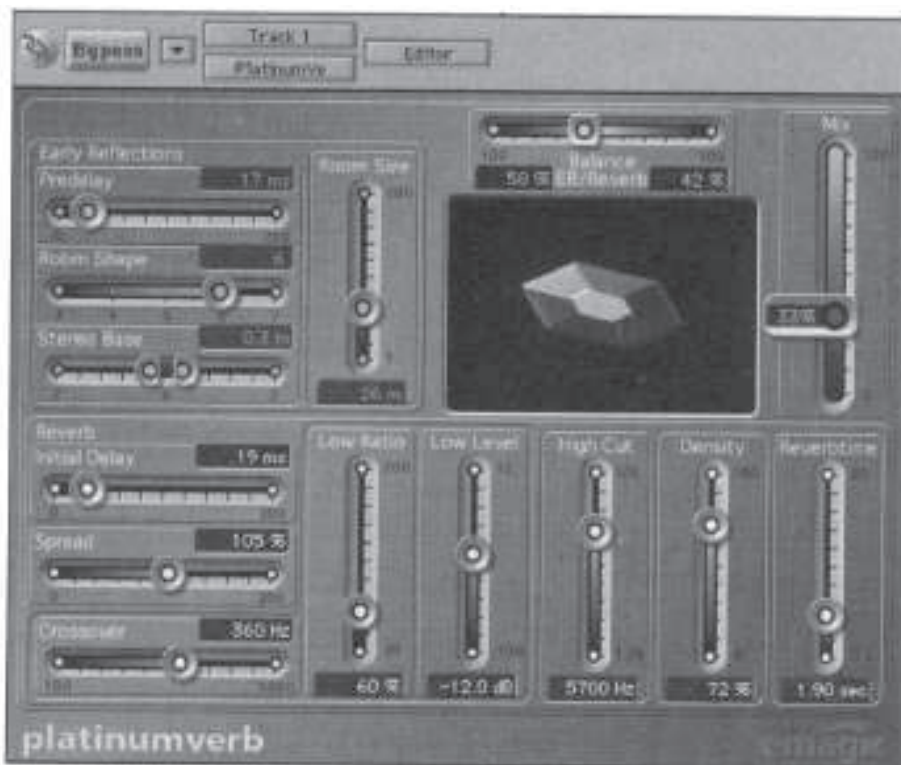
Uneven or absorbent surfaces (wallpaper, wood paneling, carpets etc.) tend to reflect lower frequencies better than higher frequencies. The High Cut filter lets you replicate this effect. If you set the *High Cut* filter so that it is wide open, the reverb will sound as if it is reflecting off of stone or glass.

Spread

This parameter controls the stereo image of the reverb. At 0%, the plug-in generates a monaural reverb, at 100%, the stereo base is artificially expanded-which, of course, makes the reverb sound monumental, but collapses in monaural playback.

Yes, the 21st century is loomin, but if you're mixing tracks that should still sound fairly decent when they're piped through tiny short-wave receivers, antiquated kitchen radios, bargain-basement clock radios, simple (monaural) TVs, etc, be shure to keep monaural compatibility in mind.

PlatinumVerb



The PlatinumVerb is only available with Logic Audio Platinum, not with the Gold version. The difference between the PlatinumVerb and the GoldVerb is the former's enhanced Reverb section. The *Early Reflections* sections of the two plugins are identical. For more information, please read the section covering the GoldVerb here we'll focus on the additional features offered by the PlatinumVerb.

The **Reverb** section of the PlatinumVerb is based on a genuine dual-band concept. This is to say that the on-board frequency crossover splits the incoming signal into two bands, which are then processed with reverb in two separate modules.

Crossover

This is the frequency at which the two frequency bands are split for separate processing.

Low Ratio

This parameter factors the reverb time of the bass band. It deviates from the setting for the **Reverb Time** parameter, which applies to the high band. At 100%, the reverb times for the two bands are identical. At lower values, the reverb time of the frequencies below the crossover frequency is shorter. At values greater than 100%, the reverb time for low frequencies is longer.

 Both of these phenomena occur in nature. In most mixes, a shorter reverb time for bass frequencies is preferable. For example, if you're using the PlatinumVerb to put reverb on a drumloop featuring kick drum and snare, a short reverb for the kick drum lets you dial in substantially more of the wet signal.

Low Level

This knob determines the level of the Bass reverb. At 0 dB, the volume of the two bands is equal. The Bass reverb level can be boosted by up to 12 dB and attenuated by up to 100 dB.

 In the vast majority of mixes, your best bet is to dial in a lower level for the low-frequency reverb signal. This lets you turn up the level of the bass instrument—it will sound punchier. This also helps counter bottom-end masking effects.

EnVerb



The Emagic EnVerb is based on a rather unusual and innovative reverb algorithm. It has a unique feature—you can adjust the envelope of the diffuse reverb tail freely. This gives you options that far exceed those of a conventional gated reverb.



The algorithm of the EnVerb requires quite a bit of computing power.

TimeParameters

With a concept as sophisticated as that of the EnVerb, you can well imagine that a single parameter for reverb time just won't do the trick.

Original Delay

This parameter enables you to delay the original signal. Of course, this delay is only perceptible when the Mix parameter is set to a value other than 100%. The starting point of the diffuse reverb tail is not influenced in any manner.

A delayed original signal is particularly handy when you want to generate reverse reverb: Set all envelope parameters to zero with the exception of **Attack** and **Original Delay**, which you should set to approximately the same value by which you want to pre-delay the given region or track.

Pre Delay

This is the delay between the (undelayed) original signal, and the starting point of the reverb attack phase.

Attack

This is the amount of time it takes for the reverb to climb to its peak level.

Decay

This is the amount time it takes for the level of the reverb to drop from its peak to the sustain level.

Sustain

This is the level of the reverb that remains constant throughout the sustain phase.

Hold

This is the duration of the sustain phase.

Release

This is the amount of time that the reverb takes to fade out completely, after it has run through its sustain phase.

SoundParameters

The following parameters shape the sound of the reverb. (For more information on these parameters, check out the in-depth descriptions of the GoldVerb or PlatinumVerb).

Density

Reverb density. Higher values generally sound better.

Spread

Stereo base of the reverb.

High Cut

High-frequency attenuation for the reverb.

Crossover

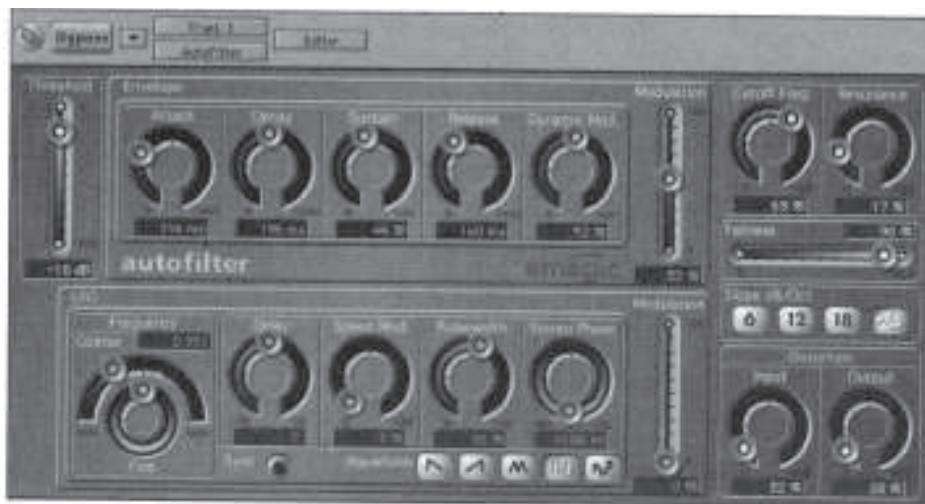
The crossover frequency for the following parameter..

Low Level

Relative reverb level of frequencies below the crossover-frequency. Although you are free to turn the level of these frequencies up, in most cases, you'll get better-sounding results when you set negative values for this parameter.

Special Effects

AutoFilter



The AutoFilter is an extremely versatile, resonance-capable low-pass filter, that comes with a couple of truly unique features. Its important parameters are located to the right side of the editor: The *Cutoff Frequency* knob determines the point at which the filter kicks in. Higher frequencies are attenuated, lower frequencies are allowed to pass unhindered.



This seems a good opportunity to share some good news: In case you were unaware of this, you can control all parameters of a plug-in with MIDI controllers. Depending on whether you insert the AutoFilter to Slot 1, 2, 3 or 4, the cutoff frequency is controlled via Controller Number 64, 80, 96 or 112. The MIDI channel must, of course, correspond to the settings of the track or bus object that you're dealing with. For live control applications, the given track of a mix object must be activated in the arrangement; e.g. "A-Playback".

The *Resonance* knob emphasizes the frequency range bordering the cutoff frequency. When you turn the resonance up suffi-

ciently, the filter itself begins oscillating (with the cutoff frequency). Just like on the legendary Minimoog, self-oscillation is initiated before you max out the resonance parameter. When you're working with resonance, ordinarily the manner in which the low-pass filter allows frequencies to pass changes: higher resonance values cause the filter to cull out the bottom end, so the signal ends up sounding thinner. The *Fatness* parameter compensates for this audio version of anorexia. When you turn *Fatness* up to maximum, the resonance setting has no effect on the response of the frequencies below the cutoff frequency.

The *Slope* buttons determine how steep you want your low-pass filter to be: frequencies above the cutoff frequency are dampened by 6, 12, 18 or 24 dB per octave (in audio jargon, these are called filters of the 1st, 2nd, 3rd and 4th order). Even though the 24 dB filter is largely the component of choice for synthesizer designers, be sure to experiment with the other options, they can also deliver pretty hip results. *Distortion Input* and *Output* lets you separately control the two distortion units - one pre-input and the other post-output. Although the two distortion modules are identical, their respective positions in the signal chain-before and after the filter, respectively-enable them to generate remarkably different sounds.

All other parameters of the AutoFilter are used to dynamically modulate the cutoff frequency. These come in two sections: *Envelope* (ADSR, Envelope Generator) and *LFO* (Low Frequency Oscillator, Modulation Generator).

The Threshold parameter applies to both sections, and analyzes the level of the input signal. If the input signal level exceeds that of the variable Threshold level, the envelope and LFO are retriggered. The *Modulation* slider of each section determines the intensity of control signal's effect on the cutoff frequency.

Envelope: when the Threshold level is exceeded, the control signal is triggered at the minimum value. After a variable interval, the length of which is determined by the *Attack* parameter, the signal reaches its maximum value. It drops off during the time it takes to run through an interval defined by the *Decay*

value, and ends up at the predetermined *Sustain* value. Once it drops below the threshold, it falls all the way to the minimum value in the amount of time determined by the *Release* parameter. If the input signal again falls below the *Threshold* level before the control signal has reached the *Sustain* level, the Release phase is triggered. The *Dynamic Modulation* parameter lets you modulate the peak value of the Envelope section, using the level of the input signal.

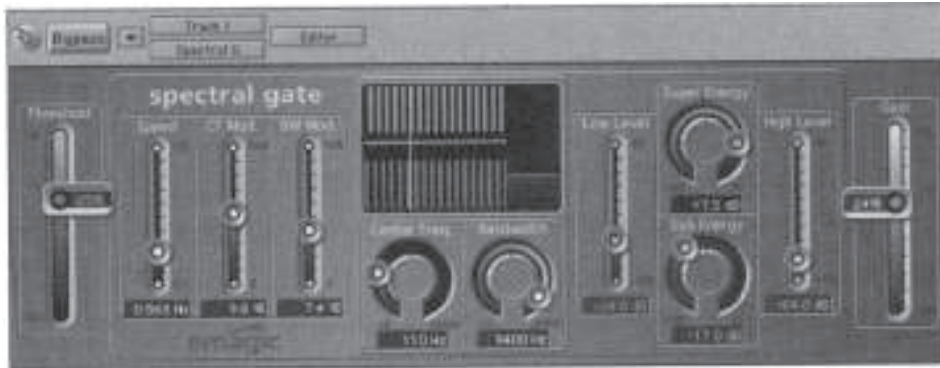
LFO: the LFO oscillates, using the waveform determined by the Waveform buttons. The choices are: descending sawtooth (saw down), ascending sawtooth (saw up), triangle, pulse wave or random (random values, “Sample & Hold”). Once you’ve selected a waveform, you can shape the curve with the *Pulse Width* knob. Use the Frequency knobs to dial in the desired LFO frequency: *Coarse* sets a value between 0.1 and 10000 Hz, *Fine* lets you adjust it in smaller increments. The *Speed Modulation* (Speed Mod.) knob is used to modulate the frequency of the LFO independently of the input signal level. If the input signal exceeds the threshold level, the modulation width of the LFO increases from zero to the value that you have entered for *Modulation*. You can also define the amount of time this process takes, by entering the desired value with the *Delay* knob. If the Sync button is activated, the waveform is started at 0° as soon as the threshold is exceeded.

Whenever you use the AutoFilter as a stereo plug-in, you can determine the phase relationship between the LFO modulation on the two stereo sides, with the Stereo *Phase* knob.

Volume

With the volume parameter, you can lower the “Volume” (up to 50 dB); e.g. to compensate a higher level when using *distortion*.

Spectral Gate



The Spectral Gate makes the signals above and below the threshold level audible, separately, and within a modulatable frequency range. It works with a Fourier transformation of the entire signal. Technical jargon aside, it is a tool that lets you come up with pretty wacky filtering effects, stuff that, until this plug-in saw the light of day, hadn't been possible.

The frequency range that you want to process is defined by the *Center Frequency* and *Bandwidth* knobs. This frequency band is separated by steep slopes. Within this band, you can use the *Threshold* slider to determine a level that separates the frequencies above and below it. The frequencies above the threshold are made audible with the *Super Energy* knob, the frequencies below it with *Sub Energy*. Additionally, the original signal outside the defined frequency band can be added to the mix: *Low Level* blends in the frequencies that lie below the frequency band (bass frequencies,) and *High Level*, the higher frequencies that lie above the defined frequency band.

The actual frequency band can be modulated by three parameters: *Speed* determines the modulation frequency, *Center Frequency Modulation (CF Mod.)* defines the intensity of the center frequency modulation, and *Band Width Modulation (BW Mod.)* controls the bandwidth modulation.

The *Gain* slider lets you adjust the level of the generated effects signal.

We suggest you use a drumloop when you begin experimenting with this plug-in. Set *Center Frequency* to the minimum, and

Band Width to the maximum values (the entire frequency range is processed). Turn up the *Super Energy* or *Sub Energy* knob one at a time, and experiment with the *Threshold* knob. You'll soon get a feel for how different threshold levels affect the sound of *Super Energy* and *Sub Energy*. When you've come across a sound that you consider particularly nifty (or even just useful), you can narrow down the bandwidth drastically, gradually increase the center frequency, and use the *Low Level* and *High Level* sliders to mix in some treble and bass from the original signal or, at lower Speed settings, turn up the *CF Mod.* or *BW Mod.* knobs. Enjoy!

6.6 Faders and Level Adjustment

Level Meter



Depending on the hardware and its drivers, the audio objects have level meters for displaying playback or input monitor level.

 **NOTE:** the Yamaha CBX hardware does not supply Logic with information about levels of any kind; please refer to the meters on the hardware itself. The meters in Logic will cycle up and down at a steady rate, to indicate the presence of signal. This is normal for the CBX.

When you arm a track in preparation for recording, the meter will display the input level.

The playback level is also displayed in the Arrange window to the left of the track number, if you select **View > Track Numbers/Level Meters**.



Peak Hold

As with conventional mixing desk meters, peak values are “held” on the display for a few seconds, so that they can be read more easily. Even if your CPU processing capacity is not sufficient for a smooth display, it reliably shows the last maximum level.

ClipDetector

If the signal overloads, and goes into clipping, the Clip detector (the red “virtual LED” at the top of the meter) will light and remain lit. You can reset the detector by clicking anywhere on the meter. This will reset the Clip detector for all channel strips.



Pre Fader Metering

With **Audio > Pre Fader Metering** engaged, the level displayed by the level meters is measured pre fader (that is: pre pan/balance), but post Equalizer and inserts. This way, the level meters will display the level of the track, even if the fader has been dragged all the way down,

Adjusting the Level

The fader on each track object sets its playback level. The recording level is not set in Logic, but at the source supplying the signal. You cannot adjust the level on the digital input. Logic’s fader shown here is used to control the monitor level.



The level reading of the audio objects and the aux sends can be shown in decibels (dB), or MIDI controller values. To change the scale, select the appropriate object ,and alter the “Val as” parameter. A MIDI volume of 90 is equivalent to 0 dB.

Please note that the Yamaha DSP Factory hardware cannot boost the audio signal. Therefore, DSP Factory tracks will output at 0 db when set to full volume (MIDI value 127).

The maximum boost is +6 dB. You can reset the volume fader to 0 dB (90) by **Alt**-clicking.

DuringRecording

An independent monitoring level is available when a Track object is record enabled. After arming the track, set the fader to the level you would like. When the object is again disarmed, the original level will be restored. For technical reasons this feature is not available when using TDM hardware

...via Key Commands

The global key command *Toggle (Mute) Audio Tracks* switches the levels of all audio objects between two different values. Each object may be set to any value for both toggle positions, so that you can quickly switch between two basic mixes.



In the same way, you can switch the other object types, using the global key commands *Toggle (Mute) Audio Input*, *Toggle (Mute) Audio Aux*, *Toggle (Mute) Audio Bus* and *Toggle (Mute) Audio Outputs*.



6.7 Routing

By holding **Alt**, the inputs or outputs of all selected audio objects can be changed in one go. Please note that the inputs are changed only on audio objects with the same mode: mono or stereo. This way, you can easily select one single input for all audio tracks.

Input Selection

In the *Input* field, you select the audio input of the audio hardware, which will supply signal to that track when recording to it. If *Monitoring* is switched on, the signal of this audio input will be routed to the output or bus selected in the field below, as long as the track is record enabled.

Sub Groups (Busses) and Output Routing

In the field directly above the pan knob, you determine where the signal of that audio object (meaning: that mixer channel) will be sent. You can select between output objects and bus objects, which can function as mixer subgroups. If the audio object is a stereo object, and if *Universal Track Mode* is activated (in **Audio > Audio Hardware and Drivers...**, also see the section *Universal Track Mode on page 8-8*), the bus will also be stereo. If the bus is a stereo object as well, the entire signal flow will be stereo-in each of the 16 busses which are available in Logic Audio Platinum (Gold: 8), if this is desired.



If Universal Track Mode is deactivated, two of the mono bus objects will always function as a stereo pair.

Bus Objects

The bus objects basically can be used in three different ways:

They can be used to route audio signals to external devices via the individual outputs of the audio hardware, allowing you to change their level and pan settings, and apply plug-ins, if you like.

They can be used as subgroup channels, which control the level of several audio channels at once. For example, all drum tracks or all different vocal takes could be routed to one stereo bus to control their overall level—still retaining the relative differences between the single channels.

Since busses can also be routed to other busses, several effect returns can be routed to the same subgroup, in order to adjust the level of all effects in the mix at once. It will be much easier to handle the mixdown (particularly if automated) this way.

To record only the signal of a particular subgroup for the purpose of using it as one individual stereo track in the arrangement, route it to an unused output object, and use the *Bounce* function there, as described in the section *Bouncing* on page 6-92.

The bus objects can be used as effect returns: in this case the plug-ins serve as internal effect processors. Effects like reverb and delay generally are applied to such a bus object, using the inserts. These bus objects are usually addressed with the send knobs of the audio channels, as described in the section *Bus Effects* on page 6-9.

These buses again will be routed to other busses, or to an output object.

Output Objects

Output Objects represent Logic's audio outputs. This is where level and stereo balance are adjusted; mastering plug-ins can also be inserted here, for example a compressor or an equalizer like the *Fat EQ*.

The output objects also provide the *Bounce* function (see the section *Bouncing* on page 6-92), which can either be used for recording all audio into one stereo audio file (comparable to a master tape), or to merge several tracks (including all plug-ins etc.) into one stereo or mono track (depending whether the output object is set to mono or stereo).

16 Busses

Logic Audio Platinum offers a maximum of 16 internal Busses (Logic Audio Gold: 8). A Preference decides whether the Bus objects 9-16 are actually set. For this, open the dialog **Audio > Audio Hardware & Drivers...** and activate the option "16 Busses" under the appropriate driver.

The default setting is off to save screen space when displaying your mixers.

Pan/Balance

Mono objects feature a Pan control which determines the position of a signal in the stereo image.



NOTE: panning is not possible if you are using the Yamaha CBX hardware, as each track is "tied" to a fixed output.

Stereo objects, on the other hand, possess Balance controls. The balance control differs from the pan control in that the former controls the relative levels of *two* signals (L/R) at their outputs. The latter merely shares one signal proportionally between two outputs.

You can reset the pan control to center (64) with a m-click.

Track Arming

The Rec control (see right) “arms” a track, making it ready for recording. Once the track has been armed, the control flashes red. During actual recording, it remains lit red all the time.



If the control is flashing but gray-red in color, no audio track has been selected in the Arrange window, and no recording can take place.

Rec controls only exist for Track objects. Please note that the number of Track objects which can be armed at the same time, is limited to the number of tracks which your audio hardware can record simultaneously. This would typically be the number of inputs offered by the audio hardware.

You can select which available input the Track object will record from, by using the input selector switch, found in the I/O section of the Track object. Click-holding on the switch will reveal a flip-menu, showing all available inputs for your hardware. If the record track is stereo, the inputs will be displayed in pairs (Input 1-2, Input 3-4, etc.). For mono tracks, they will be displayed individually.

With extended TDM systems you can record on all 32 or 48 audio tracks. In TDM systems there are now 32 audio busses, instead of the previous 16.

If you are using different audio hardware systems, you can record on all hardware simultaneously.

Solo

All audio objects have a solo button. This mutes all other audio objects of the same hardware type. The muted objects are marked with a flashing "M" in the mute button. Please note that this does not mute MIDI tracks.



You can solo several objects at once.

Alt-clicking a previously unsoloed object, releases other active solo buttons, so that you only hear one channel (“Interlocking Solo” or “Solo Toggle”).

Alt-clicking any activated solo button removes the solo status from all audio objects.

Solo Automation

The Automation uses Controller 3# for the Solo status of a given object.

Value 127	Solo (actually any value except 0 or 64)
Value 0	Switch off solo
Value 64	Toggle solo

Solo Safe

When you solo a channel, if you want to hear the signal together with its effects, the effect return channels should not be muted. Unlike conventional mixing desks, Logic can scan the entire signal path, and leave the effect return channels open.



The same applies when you solo an effect return signal. The channels feeding the effect are muted, but their effect sends remain open, so that the effect still receives a signal.

The automatic mute-suppression only applies to the internal effect returns. If you are using external effect units via bus objects, the scan will keep the effect master sends open. However, Logic cannot know which of the input objects you are using as effect returns for the external effect units. You need to manually switch these input objects to “solo safe”—just like on a conventional mixing desk.

m-clicking an inactive solo button makes this audio object “solo safe”. This stops it being muted when you solo another channel. “Solo safe” status is indicated by an X in the solo button.

m-clicking again defeats the solo safe status.

Mute Control

You can mute any audio object by pressing the button marked "M". Pressing the button again restores the previous level.



Mute Remote Control

Mutes used to be recorded as Volume Controller events in previous versions, but Controller #9 is now used instead of Controller #7. This allows you to control muting via Environment buttons, or hardware controllers.

Value 127	Mute (actually any value except 0 or 64)
Value 0	Switch off mute
Value 64	Toggle mute

Stereo Objects and Channel Strips

All the types of audio objects (Track, Input, and Output) may be configured as stereo. Logic Audio Platinum uses true stereo tracks, as long as *Universal Track Mode* is activated (see the section *Universal Track Mode* on page 8-8). This means that when you configure Track 1 as stereo, Track 2 will be left unaffected. Please note that DAE, DSP Factory, and CBX systems behave as with earlier versions, setting a track to stereo will “grab” the adjacent track.

Audio sequences on the tracks of stereo objects are always dealt with together, no matter what the operation being performed. If you open the Sample Edit window, both sides of the stereo pair are shown there, and are processed together.

Creating Stereo Objects

To do this, select the object named “*Audio 1*” (click on its name, at the bottom). Click on the Mono symbol, at the bottom left of the object, next to the REC button.



The button will now show two interlinked circles, indicating that it is now a stereo object



If you click-hold on this button, you will see a flip-menu, giving you four choices:

Mono	Sets the track to play mono files.
Stereo	Sets the track to play stereo files. These may be either “interleaved” or “split” stereo files.
Left	If a stereo region is assigned to the track, only the left side will be played. If a mono file is assigned to the track, it will play normally. In either case, the output of the track will be mono.
Right	If a stereo region is assigned to the track, only the right side will be played. If a mono file is assigned to the track, it will play normally. In either case, the output of the track will be mono.

The Effects

Whatever parameters you adjust for stereo objects, the changes affect both sides equally. The pan control becomes a balance control, and the auxiliaries, EQ and insert points (on TDM hardware only) operate in stereo.

Stereo objects have their own set of parameters for setting level, balance, aux send and EQ, (and with TDM hardware, insert points as well). As a result, you can toggle between stereo and mono object types, without losing the parameters for each type.

Any combination of stereo and mono recordings is permitted. For example, you can set tracks 1 and 2 each to stereo, and use tracks 3 and 4 for mono recordings, if you wish.

Songs on Different Kinds of Hardware

If you’ve recorded a song on (say) a 16-track Pro Tools system, and you wish to work on it further on a 4-track AudioMedia II-based system, you’ll only have access to the first four tracks, and will no longer be able to see the aux sends and the insert points. Tracks 5 to 16 appear as “empty” objects. Nevertheless,

you can carry on working with the AudioMedia II system, and still return to your larger system later. All of the larger system's options such as the various Inserts, Sends etc. will be restored to the audio objects automatically.

Audio Configuration Window

You can open this window from the Audio Window, by selecting **Options > Audio Configuration**. Here you can control all the routing settings (inserts etc.) for the song-even if the required HDR hardware is not currently installed. You can use the local **Device** menu to switch between the different hardware.

The **View** menu allows you to view only those elements which are of interest.

If you want to remove elements from the configuration, you can do so with the new **Edit > Remove...** option.

6.8 MIDI Channel Strips

This section deals with the MIDI channel strips of the Track Mixer, as well as those of the GM Mixer, which is described in the section *GM Mixer* on page 6-100.

The MIDI channel strips work as a remote control for the mixing parameters of your sound modules and synthesizers (e.g. volume, pan). This is the nature of a sequencer. The controls send MIDI Control Change Messages. They do not control any audio signal flow within Logic, as the audio channel strips do.

Switch on all options in the **View** menu of the Track Mixer, so you can see all the parameters described below.

Switching the Instruments Voice Program

Bank



If your sound source “understands” bank select events, you can choose the bank number for each of the 16 MIDI channels. Please remember that not all synthesizers support bank select events.

You can use several different bank select formats. See the section *Defining Your Own Bank Select Commands* on page 5-32.

Program

The Program flip menu is where you can select a sound by name. Each channel has its own flip menu. The top row is for selecting programs for odd-numbered MIDI channels (1, 3, 5 ... 15), and the bottom row is for even-numbered channels (2, 4 ... 16).

To select the sound for a MIDI channel:

- Click-hold the sound name.

A flip menu opens, containing all the GM sound names.

- Select a sound from the menu, and release the mouse button.

Logic selects the sound in the sound module.

Volume

You can use the volume fader to control the volume for each individual channel. Controller 7 is sent.

- Click the fader and move it, keeping the mouse button held down.

Mute

The mute button switches the volume of the channel between zero and the current fader position. In practice this means that if the button is “down”, the channel is muted. If you switch the mute off, the fader position becomes valid again.

If you want to mute a channel:

Click the mute button of that MIDI channel.

Click the button again to switch the channel back on.

Pan

The pan knob allows you to directly control the pan position of the sound.

Click-hold the knob.

Push the mouse up and down.

The knob moves according to the mouse position.

Other Controllers

You can send any controller data, to control different parameters of your sound source, with each of the upper rows of knobs. First, here’s a description of the predefined functions, which are defined in General MIDI. Not all MIDI instruments will understand these Controllers:

Reverb

This knob controls the reverb level. The further you turn the knob, the louder the effect signal becomes (in other words, more signal from the channel is fed into the effect processor).

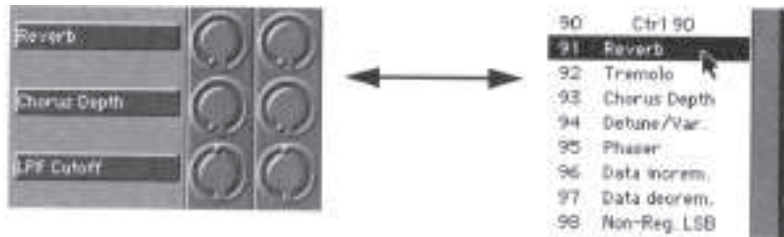
Chorus Depth

This knob controls the depth of the chorus effect. The further you turn the knob, the stronger the effect becomes.

Cutoff Frequency

This knob controls the overtone content of the sound. Higher values make the sound brighter.

Selecting Other Controllers



Here you can select the controller that you want to send with the adjacent row of knobs.

In the controller list, all controllers are listed by name, if the MIDI standard provides a certain function for their number. All controllers between 1 and 120 are accessible.

Summary of the MIDI Mixer Functions

These functions are always available:

Before a sound module can react to the events in the GM Mixer, it must be switched to receive the relevant controllers (see "Ctrl. No." column). For details refer to the sound module's manual.

MIDI Channel Strips

Sometimes the response to controllers can be disabled globally (e.g. MIDI-Menu, “Receive Control Change”. Switch this to “Enable”). Again, for details, see your MIDI module’s manual.

These functions can be controlled in all GS and XG synthesizers:

Name	Description	MIDI Event	Ctrl.No.
Resonance	Filter resonance (Q)	Resonance	71
Cutoff	Filter frequency (Fc)	Cutoff	74
Attack	Attack time of the envelope generator	Attack Time	73
Release	Release time of the envelope generator	Release Time	72
Reverb	Reverb depth	Effect Depth	91
Chorus	Chorus effect depth	Chorus Depth,	93
Phaser	Phasing effect depth	Phaser	95
Modulation	like Modulation Wheel (often Vibrato)	Modulation	1
Breath	like Breath Controller	Breath Controller	2
Foot Control	like Foot Control Pedal	Foot Controller	4
Portamento	Portamento time (pitch glide between notes)	Porta Time	5
Tremolo	Tremolo effect (volume modulation)	Tremolo Depth	9 2
Detune	Pitch deviation between the oscillators	Detune	9 4

Many non GS or XG synthesizers cannot respond to these controllers. Please refer to their manuals for more information.

Saving the Mixer Settings

The Mixer settings directly affect the Instrument Parameter box at the bottom left of the Arrange window. The latest settings here are saved with the song. If the memory in your sound module is not maintained by battery every time you switch it off, it will forget any settings you made with Logic. To restore the old settings after you load the song, go to the main menu and select **Options > Send to MIDI > Used Instrument MIDI Settings**.



Extended GM, GS and XG Functions

In addition to the GM Standard, there are extended standards set up by Roland (GS) and Yamaha (XG). In GS and XG mode you can select different effect programs, as well as controlling the level of the reverb and chorus effects

Make sure that **View > Legend** is switched on,

Click-hold “GM” on the right side of the Mixer window. A flip menu appears.

Select the extended standard (GS or XG).

Depending on your selection, the controllers for the extended effects then appear.

Select the desired reverb or chorus effect from the flip menu.

Program the desired reverb or delay time under “Time”, by holding down the mouse button, and setting the value.



Reset Button

The reset button transmits a “GS On” or “XG On” command, and resets all the controllers back to neutral positions. This allows you to reset all the connected sound modules to their standard settings, so that you can begin a mix from scratch.

There is a special General MIDI Mixer pre-defined as an environment layer. Read more about this in the section *GM Mixer* on page 6-100.

6.9 Automation

On a mixing desk, automation is the ability to record the changes to the levels of the various audio channels over time, and then recall and reproduce these changes automatically when mixing.

“Full” automation allows you to recall the adjustments to all of the parameters on a desk, such as the pan and EQ settings, as well as all the channel levels.

Naturally, the “virtual” mixing desks in Logic’s Environment are all fully automated consoles.

Logic’s Various Automation Concepts

Logic has different automation methods, depending on what mixer is used. If you are using Audio and MIDI tracks, there is the adaptive Track Mixer, which is covered at the end of this chapter. It is designed for simple operation and automation. Any mixer movements made during a recording are recorded to the individual tracks, as MIDI events, and can be played immediately-simple! You could almost ignore the following explanations. However, there are alternatives:

If you are using the audio mixer (it appears by default in the Audio Layer of the Default Song), which has been automatically configured to your hardware, all the automation data is recorded to a track instruments labeled “A-Playback“, “B-Playback“, etc. Each of these can automate up to 16 audio channels, so the number of “playback” mixing objects you need will depend on how many tracks of audio you use in a given session. Select the "A-Playback" track, start recording and move the controls on any channel of the audio mixer from 1 to 16. All the automation data is stored as MIDI events in the selected track. (With the adaptive Track Mixer, the automation data is spread over the relevant tracks). This mixer is useful for purely audio applications (hard disk recording with no MIDI instruments).

Automation data and audio tracks are conveniently separated, when it comes to individual editing.

The same applies to automation on the GM/GS/GX mixers. The track instrument for the automation is called “GM Mixer Modem” or “Printer”. Select this track if you are only using a General MIDI sound source (without audio).

The pure audio mixer and the pure GM/GS/GX mixer act like the mixers which you can construct yourself in the Environment. They are part of the standard Logic package, except that the audio mixer configures itself according to whatever audio hardware is installed on your computer. The following section explains automation for self-configured mixers and audio mixers.

Theory of Audio Automation

When you move the graphically represented controls (faders, switches, knobs etc.) on the audio objects with the mouse, you generate MIDI controller information, which can be recorded to a separate MIDI track.

If you then play these MIDI controller events back to the audio object(s), the controls will move in exactly the same way as when the controller information was recorded, and of course, the effect of these moving controls will be the same.

The controller events can then be manipulated in the Event List Editor, just like any other events. You can also *create* automated control sequences graphically by using Hyper Draw, or the Hyper Editor.

Instead of using the faders on the audio objects themselves, you can use other fader objects to generate the MIDI controller events. This approach has the advantage that you are not dependent on the default graphical arrangement of the audio objects’ controls. You can easily create your own mixing control surface, and tailor it to your specific needs.

Signal Flow

Automating a mixer is similar to recording the movements of ordinary fader objects in the Environment.

Playback Instrument (“Mixer Object”)

To record the movements of the controls, you need an object which will act as a track instrument in the Arrange window. A channel splitter is the obvious choice.

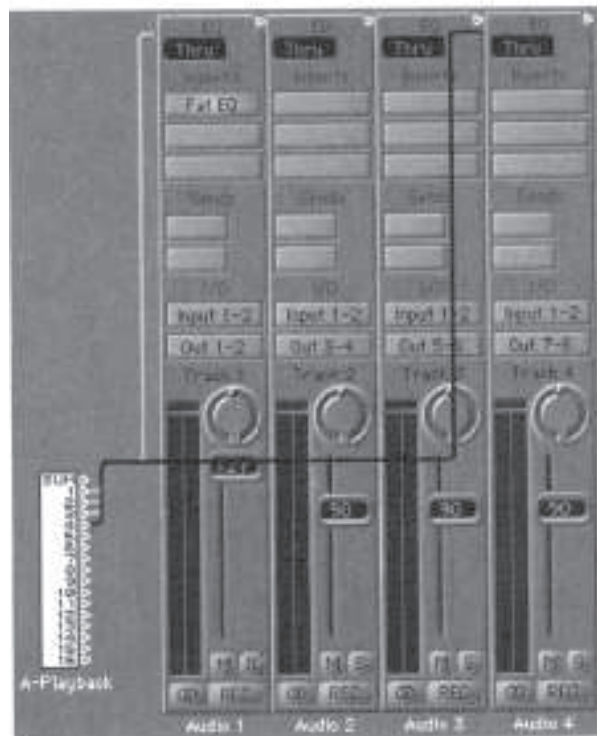
Start by connecting the individual channel splitter outputs with the “matching” fader objects. By “matching”, is meant that the MIDI channel of each splitter output should be the same as the MIDI Cha setting in the parameter box of each fader object.

So Channel Splitter Output 1 is linked to Track 1's fader object, splitter output 2 to track 2's object, etc. (assuming of course that you haven't altered the default values given to the MIDI Cha parameter before you do this).

The channel splitter is always present when you select a new Song, and is has the default name “A-Playback”

(for “Audio Playback”. You can, of course, rename this object as you wish. For simplicity's sake, we'll keep this name the same for now.

In the above picture, you see an example of 4 audio objects cabled to the Channel Splitter, with Channel 4's Output selected.



Signal Path and External Fader Objects

If you'd rather just use level controls, or you'd like to rearrange the controls to suit your own taste, you can create your own custom control surfaces.

You can do this because every function of an audio object (e.g. level, pan and EQ controls) can be controlled by MIDI, which means you can configure an *external* fader object to send the corresponding data needed to control each function.

To do this, these fader objects have to be able to send MIDI controller information (Out definition), and be "remote-controlled" by the same MIDI data when it is received ("In" definition).

There is a list which MIDI control numbers are used to control the various audio object controls in the section *Which Controller Numbers?* on page 6-84. The next section describes the signal path you need to set up.

Which Controller Numbers?

The functions available to you depend on the hardware you're using. Whatever you're running, you'll have control over level and pan (except for the CBX, which cannot be panned), but the EQ and aux sends will only be available to hardware that is suitably equipped.

Controller Number	Meaning	Controller Number	Meaning
3	Solo	28	Send 1
7	Volume	29	Send 2
8	Balance (only for Stereo Objects)	30	Send 3
9	Mute	31	Send 4
10	Panorama	32	Send 5
16	EQ #1 Frequency	33	Send 6
17	EQ#1 Bandwidth		

Automation

Controller Number	Meaning	Controller Number	Meaning
18	EQ #1 Gain	34	Send 7
19	EQ #1 Bypass	35	Send 8
20	EQ #2 Frequency	64-79	Plug-in #1
21	EQ #2 Bandwidth	80-95	Plug-in #2
22	EQ #2 Gain	96-111	Plug-in #3
23	EQ #2 Bypass	112-127	Plug-in #4
24	Cue-Session 8		

The descriptions given here are also shown in the Event List, if the hardware you're using has the relevant function.

Controllers for the Insert Effects

The operating controls (parameters) in the first four insert effects of each audio object can be automated. The first 16 parameters in each insert effect can be automated.

The allocation of Controller numbers to the insert effects parameters is shown in the table below. The numbering of the inserts begins at the top.

1. Insert	Effect parameter	Controller # 64 to # 79.
	Bypass	Controller # 56
2. Insert	Effect parameter	Controller # 80 to # 95.
	Bypass	Controller # 57
3. Insert	Effect parameter	Controller # 96 to # 111.
	Bypass	(Controller # 58)
4. Insert	Effect parameter	Controller # 112 to # 127.
	Bypass	Controller # 59

The automation of the Plug-in parameters works the same as with volume and panning, or as the Plug-ins in TDM systems. All of the parameters are recorded in

normal sequences, as controller events. This means that you can use all the usual edit options in Logic's Editors for automation data, as well.

Examples:

With AV as the destination, Logic shows *Volume, Pan/Balance, EQ1, EQ2, EQ3, EQ4, Send 1,...*

With CBX as the destination, Logic shows only *Volume* (plus *Pan/Balance* for stereo recordings).

With AM II as the destination, Logic shows *Volume, Pan/Balance* and *EQ1...*

With PTII as the destination, Logic shows *Volume, Pan/Balance, EQ1..., EQ2...* and *Sends*.

With PTIII as the destination, Logic shows *Volume, Pan/Balance, Sends* and *Plug-in..*

For ProTools III users:

For details on the automation of Plug-ins, please refer to the section on the TDM Extension.

Automation of the Plug-in Parameters

The automation of the plug-in parameters works in exactly the same way as the automation of volume, as previously outlined.

Try it for yourself. Begin by recording the to *A-playback* instrument. Open a plug-in for any track from 1 to 16, and move the control which you want to automate. These movements will be recorded as MIDI commands, which can then be played and edited.

This method can be used as an "automatic selection" technique, to set Controller numbers to be automated via graphical means in Hyperdraw. With the Hyper Draw Auto Define function, Hyper Draw automatically switches to the first Controller used.

You can even automate the plug-ins inserted on “Input”, “Aux”, “Output” or “Bus” objects. To do this, create a channel splitter in the Environment (**New > Channel Splitter**), and connect its outputs with cables to the relevant audio objects. Make sure that the **Cha** parameter setting for the audio objects match the MIDI channel number of the splitter output they are connected to. Name the splitter object appropriately (“Input Automation”, “Bus Automation”, etc.), and add the channel splitter to the Arrange window like a track instrument, as you did previously with the “A-Playback” instrument. When you want to automate tracks 1 to 16, use the “A-Playback”, if you want to automate a bus, or output, use the channel splitter that the object is cabled to.

Dynamic Controller Allocation

To allow more than 16 parameters to be automated in future Plug-ins (or TDM Plug-ins, etc.), Logic organizes the number of required Controllers dynamically.

The “basis addresses” (Controller numbers: 64, 80, 96, 112) are unaffected. However, if you have only switched on one plug-in in the first position (top insert slot), you can automate 64 parameters: 64 to 127.

If you add another plug-in to the second slot, Controllers:

64 to 79 (16 parameters) are used for the first Plug-in, and
80 to 127 (48 parameters) are used for the second.

If you use one plug-in in the first slot, and a second Plug-in in the third slot, you can automate 32 parameters in each (with Controllers 64 to 95 and 96 to 127).

If you want to see which Controller number controls which parameter in each Plug-in, just open the Event List, and click-hold the Controller number. You will see a list of all automated parameters in the flip menu. The parameter names of all the known plug-ins are shown as text in the Event List.

Snapshot Record

Select an “A-Playback” track,
at the desired song position, go into Record/Pause,
select the relevant audio object in the Environment,

Select **Options > Send Selected Fader Values**.

All the relevant controllers will now have been recorded.

Real-time Record

To record the adjustments made to audio object controls (or the faders of a custom “virtual mixer”) in real time, you have to set up the “A-Playback” instrument in one track of the Arrange window,

Start by bringing the Environment window containing the audio objects to the front on your monitor. (Here’s a quick tip: bring it up as a float window, by holding down **Alt** as you open it).

Start recording. As the song plays through, you can operate all the controls for channels 1-16 on your “virtual mixer”, and all the adjustments you make will be recorded on the “A-Playback” track as MIDI controller events. If you wish to automate tracks 17-32, assign the “B-Playback” object to a track in the Arrange window, and follow the same procedure. Channels 33-48 can be automated with “C-Playback”, and so on.

When you play the song back, you’ll be able to just sit and watch as all your adjustments from the previous run-throughs are faithfully reproduced. Keep in mind that if you have multiple mixers in your song, you must record the control data to the channel splitter that is actually connected to the mixer you are trying to automate.

Setting up Groups

You can move several faders at once by simply selecting all the desired objects (e.g. by Shift-clicking them).

Mixer Automation Parameters

This section explains how to record and update mixes in multiple passes.

Mixes made with the mouse have to be recorded in several passes, or “takes”. You have to build up the mix by recording different takes for level, panning and EQ on all the tracks. This brings up the question of how Logic handles updating already recorded control movements. Logic has three operating modes for updating automation data. You can access them from the audio mixer, from the adaptive Track Mixer, and from the GM/GS/GX mixer, by selecting **Options > Mixer Automation**.

The mixer automation parameters determine how Logic reacts when a controller on a track is overwritten by the same controller, on the same MIDI channel. If it is a different controller, or a different MIDI channel, there is no problem: a fader movement (Controller 7 = MIDI Volume) on audio track 3 (with MIDI channel 3) will not be affected by recording a fader movement on track 4 (with channel 4), nor by recording a pan movement on channel 3. All these control movements are not contradictory, and can be sent simultaneously. But what happens if you are recording, while a previously recorded fader movement is playing on track 3, and you then grab this fader with the mouse and move it? Of course the volume commands are always recorded but..

In **Mixer Automation > Merge** mode, Logic responds as with other recordings: the new commands are recorded in addition to the old ones. The result is that the fader waggles back and forth between the new and old values. This mode is rarely of any practical use, but is available for the sake of completeness.

In **Mixer Automation > Replace** mode, the old controller messages are deleted, and replaced by the new ones. This process resembles the replace function in the Transport panel, which erases previous recordings—just as with a tape recorder. However, **Mixer Automation > Replace** only



replaces the same controller on the same MIDI channel, allowing you to correct a prior recording of the same controller. Unlike the replace record function, a pan movement would not be affected by recording a fader movement. The new recording begins as soon as you grab the fader (press the mouse button). It ends soon after you release the mouse button-but not immediately. The new fader movement will end with a value which is very unlikely to match the old value at the same point. The undesirable consequence would be a sudden shift in volume (in the case of a fader). To compensate for this, you can define a *Soft Fade Time*, which is the time it takes for the end value of the new recording to reach the value of the old recording. *Soft Fade Time* is adjustable in increments of milliseconds, and insures a smooth transition.

In the third mode, **Mixer Automation > Update...** any movements in the old recording are retained, but are updated by the new recording to produce relative value alterations. For example, if there are fader movements which you are happy with, but which are too loud in general, you can use this mode to make them all ten points softer.

Erasing Recorded Fader Movements

Now and again, you'll need to erase the movements you've stored for a particular fader, and re-record them.

The Hyper Edit widow can be a very useful tool for this job. You will need to do some preparation by setting up a Hyper Set, containing the event definitions for all the controls you're using. For more about this, see the chapter on using Hyper Edit, in the first part of this manual.

If you don't have an appropriate hyper set at your disposal, you can also delete the data from the Event List window ().

If you know the MIDI channel and control numbers of the fader, knob or switch you want to change:

Select one of the events you wish to erase;
select **Edit > Select Similar Objects**;
press the *Backspace* key.



The MIDI channel containing the Controller information usually corresponds to the track number of the audio object (so the Track 3 object generates MIDI controller events on MIDI channel 3 - unless you change its default value, which is not recommended).

Volume fader adjustments generate controller #7 data, while moving the pan control sends controller #10 data.

If you don't know the MIDI channel or controller number for the fader, knob or switch you wish to edit:

Set the Event List window to “Contents Link”;
Set the song position to the start of the sequence;
hit PAUSE, then RECORD;
move the control whose movements you want to edit.

The Event List window will then show all the events generated by this control.

Hit STOP twice;
deselect all the events in the Event List window;
select all the events that have just appeared;
select **Edit > Select Similar Objects**;
hit the *Backspace* key.



Graphic Control Changes with Hyper Draw

If you need to make fader adjustments over a period of time, so that they happen as evenly and smoothly as possible (for example, a fade in or out, or a panning operation), you're best off using Hyper Draw. This is covered in the Arrange window chapter (in the section *Hyper Draw in the Arrange Window* on page 3-49). Please also see the section *Hyper Draw for Regions* on page 4-26.

6.10 Bouncing

Output objects have bounce switches ("BNCE"). This also applies to the output objects of the adaptive Track Mixer and the audio mixer, which are supplied by the default song, as an Environment layer. This allows you to create an audio file based on all the audio tracks assigned to this output. All parameters, including volume, pan, and effects are recorded as part of the "bounced" file. Bouncing takes place in real-time, so that the signals from any plug-ins, or MIDI devices routed into an input used can be recorded in the bounced file:

Press the bounce switch on the output object

Set the parameters in the Bounce dialog window (see below).

Press the bounce switch in the Bounce dialog window.

A file selector appears where you enter a destination folder and name for the bounced file.

The DSP Factory does not offer "Bnce" buttons on its output objects. To perform a bounce with this system, you must configure the "DS2416 Routing window" so that "Rec 1-2" are set to the output pair that you would like to bounce from. When done with the bounce, reset "Rec 1-2" to "Input 1-2".

Mastering with Plug-ins

The Inserts of the output objects allow signal processing during the mastering process (as well as during normal playback). Typical mastering tools are the compressors, de-essers and equalizers, like the Fat EQ. Please take note that for technical reasons, only plug-ins which need no mono/stereo conversions can be used, that is to say, (s/s) plug-ins on stereo outputs, and (m/m) plug-ins on mono outputs.

Mono Bouncing

Mono bouncing is possible when the Output object is set to "mono". In this case, a mono audio file is generated.

Options in the Bounce Dialog Window

Start and End Position

This is where you define the segment of the song which you want to be written to the bounced file. The default is the whole song, from the beginning of the first audio region to the end of the last. However, if the cycle function is engaged, the default will be the locator positions. In any case, you can adjust the start and end positions manually. For example, if you want to take account of a MIDI sound module mixed via an audio input object outside the preset region.



It is often a good idea to set the endpoint of the bounce a bit past the end of the last region to be included, so that reverb tails and echos from delay plug-ins are not chopped off. How much extra time is needed, will of course depend on the effect settings you use.

Beneath this, you are shown the memory requirement for the bounced file.

Resolution

This is where you define the resolution of the bounced file. The options are 16 Bit, 8 Bit (e.g. for multimedia productions) or 24 Bit (e.g. for mastering to DVD, Platinum only).

FileFormat

You can select from between SD II, AIFF or WAV format here.

Stereo File Type

Here you can select between split stereo (for use in ProTools), or interleaved stereo (for further use in Logic, a SampleCell II card, or for CD writing software).

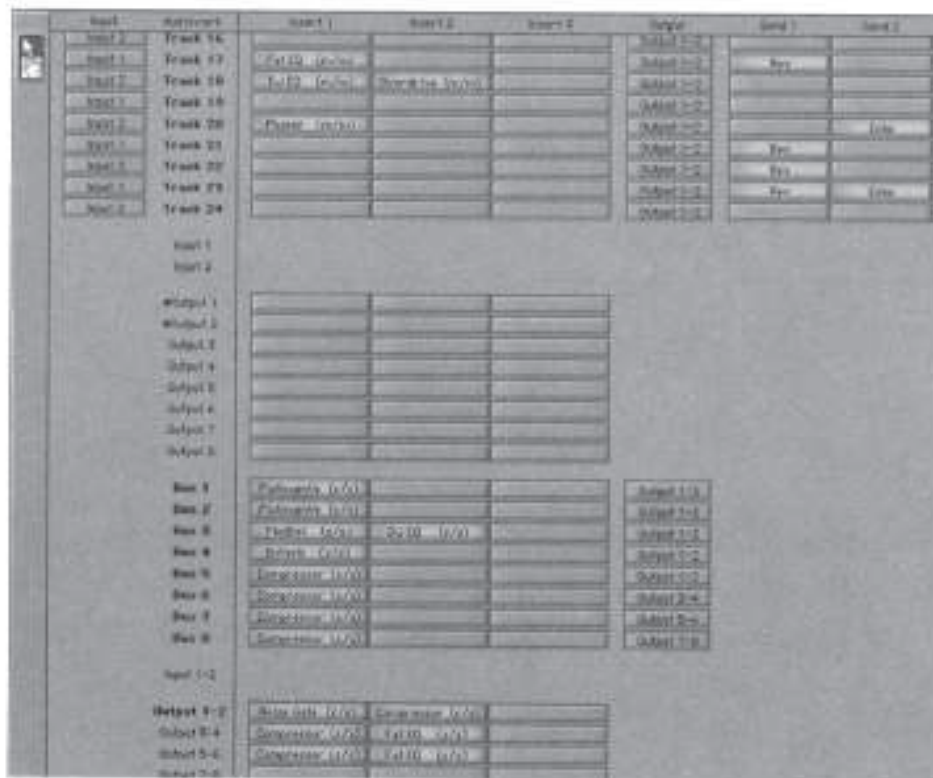
Executing a Bounce

Pressing **Bounce** performs the bouncing process. A file selector will appear, where you can define the recording path. You can

listen to the playback in real time while bouncing takes place. Pressing **Bounce & Add** adds the file to the audio file list of the Audio window, after performing the bounce.

6.11 Audio Configuration

The Audio Configuration window can be accessed under **Audio > Audio Configuration**, and under **Options > Audio Configuration** in the Audio window.



The Audio Configuration window is an overview of the Input/Output routing, as well as of all inserts and sends of all audio objects that exist in the environment for a given audio hardware device. The various object types are listed, with their names shown on the left side and their settings shown in columns. It makes no difference whether you change settings in the Audio Configuration window, in the track mixer or in the environment (audio mixer), the setup will be instantly reflected in all locations. Use whichever you prefer.

The Audio Configuration window also shows the settings of any audio objects which have been deleted or are unused. If you create an audio object in the environment (**Create > New...**), and then select the channel (Cha). it will appear with the settings that are currently assigned and can be edited in the Audio Configuration window.

You can even see the song's TDM settings if the song is opened without any TDM hardware installed.

Just as in the object type flip menu of the audio object parameter box (which opens when you click-hold the cha parameter), a channel setting displayed in bold type indicates that the channel is in use by an existing object.

A dot in front of the name indicates, that the object is part of a stereo pair-whther it is controlled by another object, or be it the controlling object itself.

Handling

Click-holding a parameter opens a flip menu with which you can change the settings, just as it is done with the audio objects in the environment (or audio mixer). The top value of the flip menu always assigns a neutral value for the parameter.

Moving and Copying Plug-Ins from one channel to another

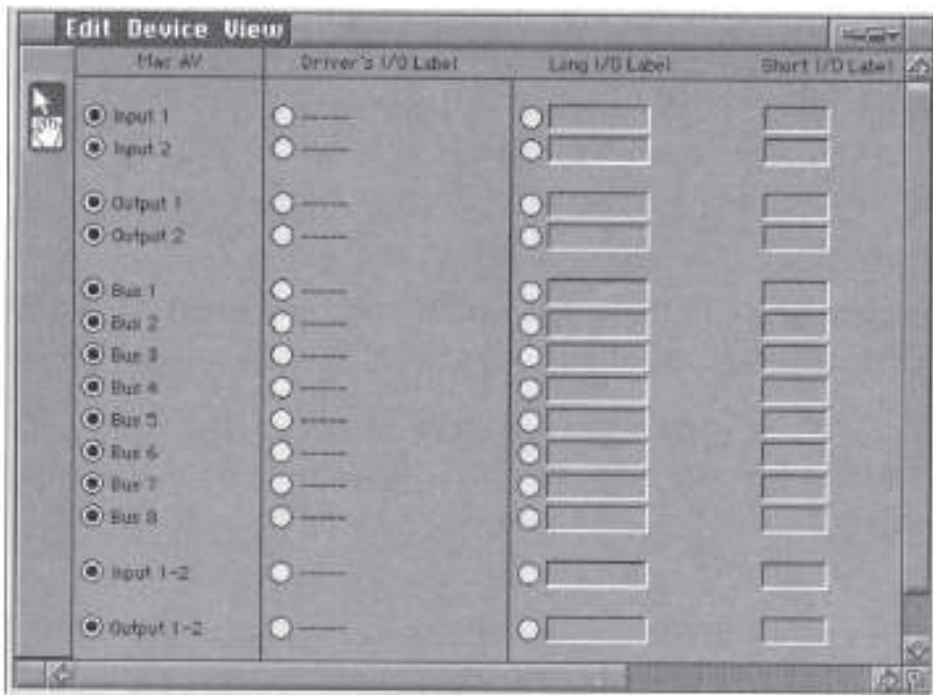
By dragging while holding , you can move plug-ins with all their parameter settings, from one insert to another. The cursor turns into a dragging hand symbol while doing so. Holding *Alt* as well, copies the plug-in settings.

Labeling the Inputs and Outputs (I/O Labels)

In the Audio Configuration window, you can define names for each audio object's I/O options, which can be used instead of defaults like "Input 1" or "Output 8". Select View > **I/O labels** to do so.

Chapter 6

Mixers, Effects and Audio Objects



 Be sure to select the right hardware under device. you can give any audio input, output or bus for each audio hardware type an individual name.

To the far left, you see the default names. Just to the right of this, the names delivered by the relevant audio driver (as long as there are any). In the column to the right of this you can double click and type in a name of your choice. The width of the name bar is the same as in the audio objects shown in the environment. Farther to the right, you can type in short labels, as used beneath the send knobs of the mixers.

With the radio buttons, you can individually define which name will be used for each I/O section for the selected type of hardware. As soon as you type in a name, it will be selected automatically.

Deleting I/O Labels

In order to reset all names to their defaults, select **Edit > Reset I/O labels**.

I/O Label File

The I/O Labels can be defined for each hardware type individually, but are valid for all songs. The labels are stored in a sep-

arate file located in the same folder as Logic. The file is called "IO Labels..." with the name of the hardware type.

Copy Entire Mixer Setup

In the Audio Configuration window, you can copy entire mixer setups. There are two ways of doing so:

You can copy settings between songs;

You can copy settings between different hardware devices.

The latter only is possible if the audio hardware devices are controlled by drivers utilizing the Emagic Audio Engine. These drivers are listed and discussed under section *Audio Engine* on page 8-5. This is how copying settings works:

Open Audio Configuration **Audio > Audio Configuration**, and select **View > All Components**. The devices are selected in the menu **Device**.

...between Songs

Select the song whose setup you wish to copy, and open its Audio Configuration window. Select **Edit > Copy**, select the target song and open its Audio Configuration window. Select **Edit > Paste** there.

...between different Devices

Select the source device (under **Device**). Select **Edit > Copy**, and change to the target device afterwards (**Device...**). Select **Edit > Paste**. The common key commands can be used as well.

Resetting the Mixer

In the Audio Configuration window, you can set all parameter types of the audio objects of the selected audio hardware device to a neutral default value. You can select from among **Edit > Remove EQs**, **Remove Plug-Ins** or **Remove Sends**.

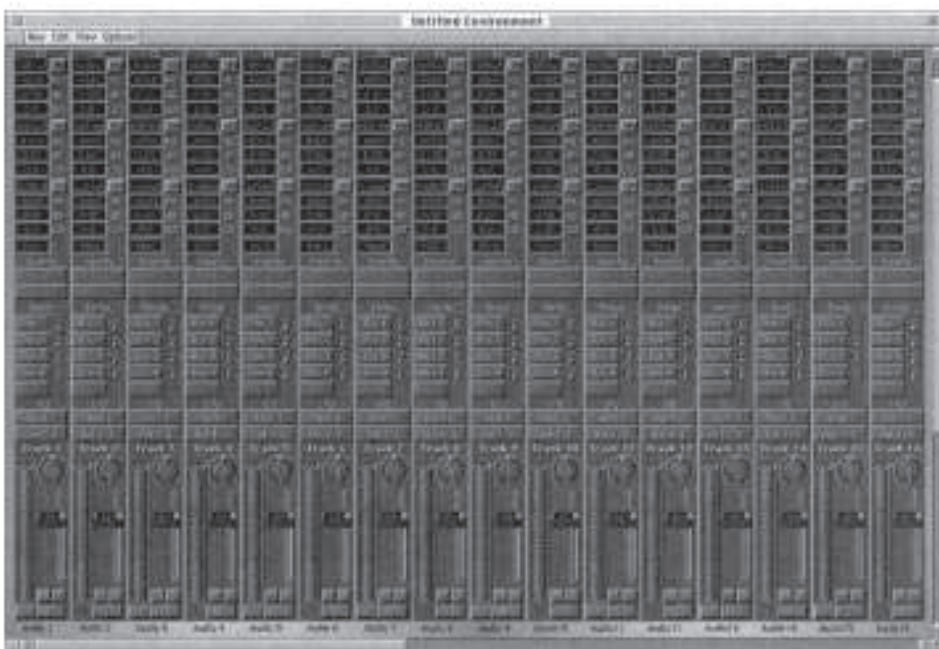
Resetting the Audio Recording Path

Select **Edit > Remove Audio Recording Path** in order to reset the the audio recording path.

6.12 Mixers in the Environment

Audio Mixer

One of the Environment layers in the default song (i.e. the song Logic loads after you boot it up) is the Audio Mixer. This mixer consists of audio objects which may be operated and automated as described above. To open it, select **Audio > Audio Mixer**, or go to the Environment (**Windows > Open Environment** or *Command -m*), and click the Layer button.



The diagram above shows 16 audio objects (channels 1-16) for 16 different tracks. To the right side of the Environment window are the effect returns and the output objects (diagram below).



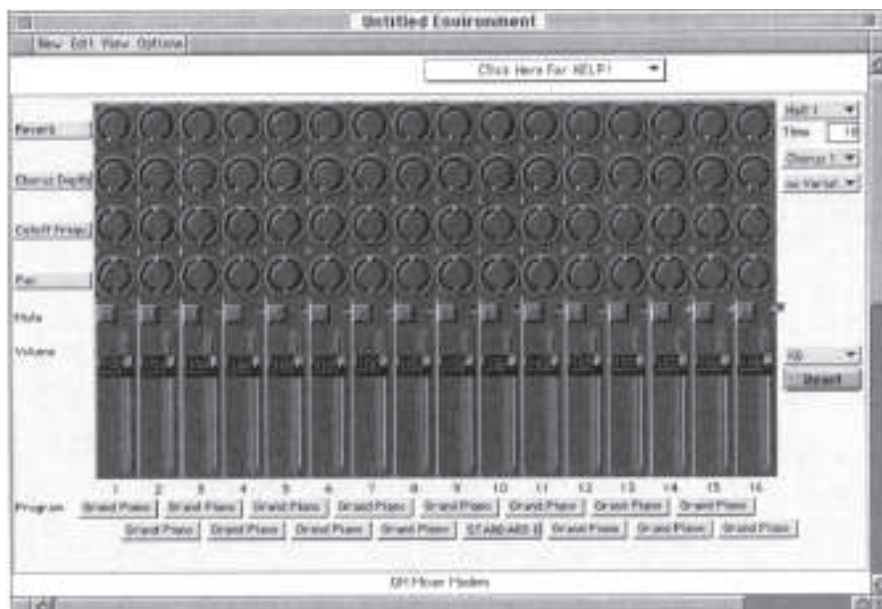
The specification and size of the Audio mixer depends on the hardware used; the audio mixer configures itself when you start Logic, or create a new song. The mixer takes up as little space as possible. If equalizers and auxiliary sends are included the size of the mixer increases automatically. Please note that once you save a song file, the Audio Mixer configuration is saved with it, and will not be automatically updated, even if you change audio hardware. You can do this yourself manually, or import an Audio Mixer from another song, should the need arise.

Automation of the Audio Mixer

Select a track in the Arrange window with the *A-Playback* track instrument. Start recording, and move the audio mixer controls. The movements on all channels from 1-16 are recorded to the one *A-Playback* track, where they can be edited. To automate channels 17-23, use the *B-Playback* track instrument. During playback, the movements are carried out, although if you mute the relevant *Playback* track, you can prevent this. For details, refer to the Automation section.

GM Mixer

One Environment layer in the default song which Logic loads at the start of the program is the GM mixer. This pure MIDI mixer has nothing to do with audio objects: it is for remote-controlling a General MIDI sound module (GM). There are also specialized adaptations for Roland's GS standard (General Standard) and Yamaha's XG standard (Extended General MIDI).



The GM Mixer is retained in the program mainly for compatibility reasons. In most cases, the use of the adaptive Track Mixer is recommended. It has the same functions, but is automated in a different way.

General MIDI is an extension of the MIDI Standard, and lays down certain requirements for sound modules. These include the following points:

- Able to receive on 16 MIDI channels simultaneously

- 128 fixed programs, with GM Standard sounds

- Chorus and reverb.

- Panning and volume for each channel controllable via MIDI.

The mixer functions are generally also available to sound modules which do not support the GM standard—as far as their MIDI implementation allows. Technically the GM mixer generates MIDI "control change" events, which generally can be recognized by all MIDI sound modules. However, only GM sound modules are guaranteed to respond to these events.



The object parameter box for the GM/GS/XG mixer is only visible if **View > Parameters** is switched on, *and* the GM/GS/XG Mixer is selected as an object (by clicking its name beneath the window).

Bank Select Format

The GM/GS/XG mixer sends bank select events in the standard format. However, if you connect the output of the mixer to the input of a multi instrument, bank select events are automatically converted into the multi instrument's format. For advice on setting the pre-defined bank select formats on multi instruments, please refer to the section *Defining Your Own Bank Select Commands* on page 5-32.

A bank is a group of up to 128 sound programs. You can call up a specific sound program inside a bank via the Program flip menu. For GS and XG sound modules, there is an inverted viewing method: within a sound program (reached via a MIDI program change command you can use the bank select command to select among several versions of the sound program (though not 128 versions).

Automation of the GM Mixer

Select a track assigned to the *GM Mixer Modem* or *Printer* track instrument in the Arrage window. Start recording, and move the GM/GS/XG mixer controls. The movements on all channels are recorded on the one GM mixer track, where they can be edited. During playback, the movements are carried

out, although if you mute the GM mixer track, you can prevent this. For details, refer to the Automation section.

Creating Your Own Mixers

Parameters of the Audio Objects

Each audio object represents one track of your hard disk recording (HDR) hardware. It can also be used to control input or group signals.

The way audio objects look, depends on the hard disk recording system you're using. The features you see on the audio objects will be a reflection of the features of the audio hardware you are using.

In order to address the HDR hardware, Logic automatically creates audio objects in the Environment. These appear as an *Environment Layer* called *Audio*. This is the audio mixer, which is covered in a separate section.

When Logic creates a new Song, it contains the maximum possible number of audio objects that your hard disk recording setup will allow.

Recalling the Audio or Track Mixer

Just as a reminder: To open Logic's adaptive Track Mixer, where MIDI and audio tracks are both available, go to the main menu and select **Windows > Track Mixer**. In order to open the environment layer with the Audio Mixer, select **Audio > Audio Mixer**.

Creating Audio Objects

Select **New > Audio Object**, and double-click the audio object icon which appears. You then determine which track of your hard disk recording system is controlled by this object with the *Cha* parameter in the object parameter box. If you are running more than one HDR system, you may select which system the object will control in the *Dev* parameter, in the object parameter box. Its exact appearance will vary, according to the hard disk recording hardware the audio object represents. Next to it, you can see the channel components of the audio mixer, here shown as it might look when using an Audiowerk 8.



Display

If you want to reduce the length of a channel, you can hide individual components, such as the inserts, or the I/O routing. Remember that you are not actually switching the functions off, but just changing the display of the audio object.

First select by **Shift** clicking, or by “rubber banding”) all audio objects whose display you wish to change.

In the parameter box, you can then switch off any of the following options:

- Show EQs
- Show Inserts
- Show Sends
- Show I/O



Keep in mind that some types of objects (such as outputs or busses) do not have all of these components. Their size is adapted to fit the track objects.

You can construct your own audio mixing desks, on different Layers, each with different display options. Thus, you can adapt the mixing

desks to fit within your screensets. You might create a layer where all the objects show just the level and pan controls to be part of a screenset that also displays an Arrange window, used for basic tracking. On another layer you might create a mixer that shows all controls, for use during mixdown. If you have duplicates of audio objects on different layers, they will always interact with each other, so that changes made to a "Track 1" object on one layer will be reflected on the "Track 1" object on another layer.

Erasing Audio Objects

You erase **objects** by selecting them, (just click on their upper edge or on the name), and pressing . This sets the level of the track controlled by this object to zero. If there is a duplicate of the object being erased, it will remain in place, and its level will not be set to zero.

The Object Parameter Box



If the Parameter column is visible (select **View > Parameters** if it's not), a box will appear in the column showing the parameters for the selected audio object. By clicking on the triangle at the top of this box, next to the name, you can conceal or reveal its contents.

The name in the uppermost line of this box can be changed by double-clicking on it. The check box on the left next to the icon tells you whether the audio object will appear on the instrument selection menu in the Arrange window. You can even change the icon itself with a flip menu, if you wish, although the default icon is usually suitable.

Dev

You use the *Device* parameter to select the hard disk recording system the audio object is connected to. The default setting here will be for one of the pieces of hard disk recording hard-

ware you have installed; entries referring to hardware not connected to your system will appear “grayed out”. You can read more about their drivers in the section *Audio Driver* on page 8-1.

Cha

The *Channel* parameter determines the *type and channel* of the signal controlled by an audio object.

You use this parameter to select which hard disk recording track an audio object will control; or you can set the audio object to control your recording hardware’s inputs, outputs, or even - depending on your hardware - its aux sends, or sub-group faders.

First, you select the type (*what* the audio object will control: a track, an input, an output etc.), then a number (Track 1, Output 4, etc.). You make all these selections from a flip menu:

Track

Selects the audio track number. The total number available depends on many tracks your hard disk system offers. *Track* is the typical setting for an audio object, making an object to which you may record audio, and which can play recorded tracks back.

Input

allows you to monitor signals connected to your hardware’s inputs. This can also be done via *Track* objects, provided the REC track arming switch is active. *Input* objects are therefore usually used to mix in external signals (like effects returns), through the system inputs at mixdown.

Please note that *Input* objects are not always available-it depends on the hardware you have connected.

Output

If you select this option, the audio object will control one of your hard disk recording system's outputs. You can then set the output level from this object, and it will act as a "master" fader for all signal passing through it-to the physical output(s) it controls.

Bus

This setting turns the audio object into a bus. You can then send portions of the signals from several tracks (Track objects) via their send controls to this bus, then process this submix with a real-time effect assigned to the insert of the bus object.

If your hardware has several individual outputs, like the Audiowerk8 the bus object can act as a master send for an external effects unit. Select an individual output for the bus object, and then connect it to the input of the external effects unit.

If you are using TDM-Hardware (with ProTools III), you may be able to select other parameters under the Channel setting.

It is entirely possible to define several objects that relate to the same channel. If you have a system with a lot of tracks but a small monitor, you could have trouble seeing all your tracks at once. To get around this, you could create a new layer to exist in parallel with the one showing tracks 1 to 16. You could, for example, design a layer showing just channels 3, 5, and 14, input 5, and outputs 1 and 2.

When you select audio object types, you'll notice the following text styles are used:

Outline: This indicates that the object is not available.
 This happens when a song has been produced
 on a system with more tracks than yours (for
 example track 9 on an 8-track system).

Mixers in the Environment

Normal: This is used for objects that haven't been defined in the Environment yet, and which aren't therefore in use.

Bold: Objects in bold have been defined in the Environment, and are in use.

MIDI Cha

You use the MIDI *Cha* parameter to determine which MIDI channel the audio object will respond to. You can then control the audio object remotely, using MIDI controller information.

This also works the other way around. Once a MIDI channel number has been set, you can generate controller information on this MIDI channel, by moving the sliders and knobs on the audio object itself.

When the Channel parameter is set, the MIDI channel is automatically set to the same number, as this usually makes things easier to remember. For channels above 16, 16 is subtracted from the track number, so that a *Cha* setting of 17 will default to MIDI channel 1, *Cha* 18 to MIDI channel 2. and so forth.

For more information about MIDI remote control, take a look at the section *Automation* on page 6-81.

Val As

You use the parameter Value as to select whether the scale of the fader on the audio object is given as:

Num a numerical value from 0 to 127, or

dB as a value in decibels.

The numerical value 90 is equivalent to 0 dB,

On the DSP Factory, a numerical value of 127 equals 0 db.



Functions

Icon Representation

Double-clicking on the upper edge of an audio object changes its appearance from that of a large fader, with graphic controls, to a small icon.

Double-clicking on the icon converts it back to a fader.

Scaling Objects

Clicking in the bottom right-hand side of an object and dragging with the mouse button held down allows you to change the size of the object. You then see the various available sizes as you drag.

The same is true for the labeling of TDM Plug-ins.

If you wish to resize several objects at once, do the following:

Resize one object;

Select just this object;

Select **Edit > Copy** (or hit 

Rubber-band all the objects you want to resize;

Select the function **Options > Apply Buffer Template to > Size**.

For background information, please refer to the section on Object Groups in the Environment chapter (first part of the manual).

Color

You can assign a color to selected Audio Objects with **View > Object Colors**.

This color will be assigned to recorded Audio sequences. If you prefer to see this color on the surface of the Audio Objects in

the Environment, activate the option **Options > Settings... > Audio > Colored Audio Objects**. Please note that the audio objects themselves can only be displayed in color if they are “old style” objects. Any audio object may be set to be shown as an “old style” fader by reducing, or enlarging its size. This has no effect on whether recorded audio sequences display the assigned color, they always do.

Chapter 6
Mixers, Effects and Audio Objects

Chapter 7

The Audio Window

Logic allows you to access any digitally stored audio recordings (audio files) on your hard drive, in the most common Macintosh formats: AIFF (Audio Interchange File Format) and SDII (Sound Designer II format). All recordings made using Logic are stored in the Sound Designer II format.

Audio files imported into a Logic song must be at a bit depth and sample rate supported by Logic and the audio hardware you are using. Logic Audio Gold supports 16 bit files at sample rates of 44.1kHz and 48 kHz. Logic Audio Platinum supports depths of 16, 20 and 24 bits, and sample rates of 44.1, 48, 88.2 and 96 kHz.

The Audio window is where you organize all the audio files on the hard disk that are used in a Logic song. It doesn't matter whether these audio files have just been recorded, or whether they were copied weeks ago from a CD-ROM onto the hard disk. The Audio window gives a Logic song access to any compatible data on the hard disk.

The Audio window contains no positioning information about the files it lists. In fact, files that are not yet in use in the arrangement of the song may be included in the Audio window. Files are assigned to the sequencer's time axis by arranging sections of the audio files, known as "regions", in the Arrange window, in the same way as MIDI sequences.

The Audio window is really like a catalog for audio files. It also gives you an overview of what regions have been defined for each audio file.

Here you can define new regions and edit, delete, or rename existing ones. When you edit them here, the accuracy is limited

to units of 256 sample words. To make more precise edits, use the Sample Edit window.

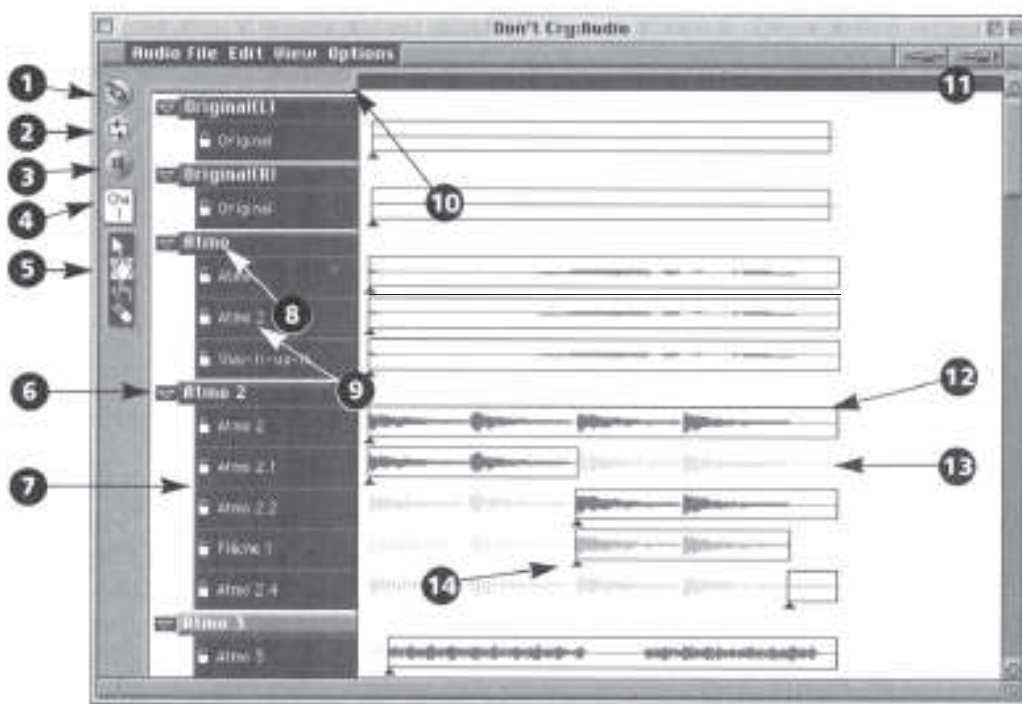
These regions can then be dragged directly into the Arrange window, where they now may be arranged as audio sequences.

The menus in the Audio window contain all the operations relating to the administration of audio files and regions, plus the System parameters of the hard disk recording hardware (HDR hardware).

Opening the Audio Window

To open the Audio window, select **Audio > Audio window**.

You can open several Audio windows at once (even within one Screenset) for example, if you want to use different zoom factors.



- ❶ Link button
- ❷ Region cycle button
- ❸ Monitor button
- ❹ Selection field for the output channel for monitoring.
Underneath this, Logic Audio Platinum has a selection

Layout

field for the output hardware for monitoring (only visible if there is more than one type of HDR hardware installed.)

- 5 Tool box for the Audio window
- 6 Click this small triangle to show or hide the regions in the relevant audio file
- 7 Click the lock symbol to protect the parameters of the region
- 8 The name of the audio file (large print)
- 9 The name of a region (small print)
- 10 Grab and drag this point to adjust the horizontal window division
- 11 Zoom telescope for adjusting the waveform display
- 12 Region box with waveform display
- 13 Display of the audio file data outside the current region
- 14 The Anchor-musical reference point in the region

If you add an audio file to the Audio window, Logic automatically creates a region encompassing the entire length of the file.

You can create as many regions as you want from the same audio file, and there are also no limits in terms of length, as long as the region is no longer than the source audio file.

Regions can be moved with the mouse into the Arrange window. The audio file is then played at the desired song position.

7.1 Layout

On the left side of the window is a thin column. At the top there are the mode buttons, and at the bottom are the tools.

The column to the right lists the audio files, and their regions by name.

Further to the right, the region waveforms are displayed graphically, in the large white area of the window. A region is shown as a boxed, black or colored area, while the rest of the audio file

is shown in light gray, indicating that this part of the audio file is not used for the region.

Zoom Functions

The two telescope symbols at the top right, just below the title bar can be used to enlarge or reduce the display (zoom function).

You can use the telescope symbol on the left to enlarge or reduce the vertical display area, thus altering the height of the region waveform display. (Preset key commands:    and   .

The telescope symbol on the right is for enlarging or reducing the horizontal display size, i.e., altering the time display of the regions (   and   .

If you hold down  as you click the telescope symbols with the mouse, you can alter the vertical and horizontal zoom simultaneously.

The Mode Buttons

The mode buttons allow you to select different operating modes. These affect both the display and the playback of regions in the Audio window.

Link

Link mode in the Audio window means that whenever you select an audio region in the Arrange window, the same region is automatically selected and displayed in the Audio window.

You can switch link mode on or off by clicking the button with the chain link icon.

“Hidden regions” cannot be displayed in link mode. If you want to display them you have to “Show” the regions (see section *Hiding and Showing Regions* on page 7-6).



Monitoring

You can play regions directly in the Audio window. This monitoring is not related to the time axis of the sequencer. There are several ways of playing a region:

Playing from a specific position

Click-hold the desired region in the waveform display with the mouse. Playback starts at the point where you clicked. This allows you to play specific sections.

Playback stops as soon as you release the button.

Playing the whole region

If you want to play an entire region, select it by clicking the name of the region in the audio list. Now click the button with the speaker symbol to start playback. Click the button again to stop playback.



You can also start or stop this type of playback using the “Play/Stop Region” key command.



Stereo Playback

If a region from a stereo file is selected, clicking the speaker icon plays both sides of the stereo file. However, if you click the icon while holding down  or , only the selected region (mono) is played. If you subsequently select a region from another stereo file, Logic goes back to playing in stereo.

Region Cycle

Regions may be looped continuously during monitoring. To switch “region cycle” on or off, click the button with the circular arrow symbols. This mode applies to all regions in the Audio window.



This cycle setting only affects monitoring in the Audio window, and has no effect on song playback. It should not be confused with the cycle function in the Arrange window.

You can adjust the start and end points of the region in real time, while the region is being cycled. This is useful for setting precise region lengths, for example when “polishing” drum loops.

Selecting the Audio Output for Monitoring

The actual routing of the audio signals to the different outputs is done in the Arrange window (using the audio object in the track list).

However, you can choose a specific channel of the audio hardware for monitoring in the Audio window. Set the desired output in the *Cha* button below the speaker icon.



In the *Dev* button below it you can choose between different audio hardware systems, if you have installed more than one. If you don't have more than one this button will not appear.



7.2 Display

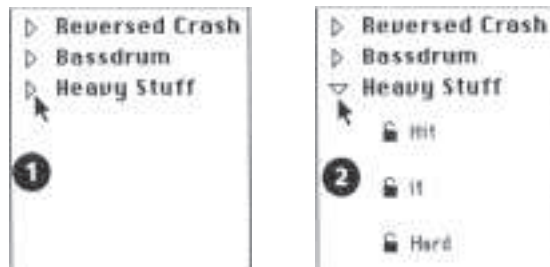
The Audio List

The list display gives an overview of all the audio files used in the current song, and the regions contained within them. This is where audio files can be added, removed, deleted, or renamed. Regions can also be created, deleted and renamed.

Hiding and Showing Regions

After you add an audio file, you will see the file name in the Audio window, in large print. This takes up minimal screen space, and shows you as many currently used files as your zoom

setting and monitor size will allow. There is a small triangular arrow directly next to the file name ❶.



Click the small triangle ❷ next to an audio file to reveal its regions. Like the list display of folders in the Macintosh file system, this shows you the “contents” of the audio files-i.e. the relevant regions. The arrow is now pointing downward toward the regions, which are shown in small print. Click the triangle again to hide the regions.

You can also press the  key while clicking on this triangle to quickly Hide/Show all regions.

Activating the Display of all Regions

By selecting **View > Show All Regions**, you can instantly display all regions, from all audio files contained in the Audio window.



Hiding the Display of all Regions

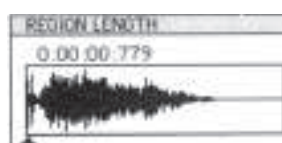
View > Hide all Regions instantly removes the display of the regions for all audio files.



If you want to view individual regions, just click the small triangle to the left of the file name.

Displaying the Lengths of the Regions

The lengths of the regions can be displayed in various units.



Select **View > Show length as...** at the lowest level of this hierarchical menu, you can choose the units for the display of the lengths of all regions.

Show Length as... > None

No display of length.

Show Length as... > Min:Sec:Ms

Absolute time length of the region in hours, minutes, seconds and milliseconds.



Show Length as... > Samples

Number of sample words in the region.

Show Length as... > SMPTE Time

SMPTE length, which unlike absolute time, gives frames and bits instead of milliseconds.

Show Length as... > Bars/Beats...

The region is displayed in musical units: bars:beats:divisions:ticks.

If the tempo, time signature or “divisions” settings are altered, the display is automatically recalculated.

Displaying Information about Audio Files

Information on every currently loaded audio file can be displayed from left to right, in the region waveform display as follows:

Display

Sample rate (Hz), Bit Depth (Bit), Mono or Stereo status (Cha), file size (kByte) and file path or directory (Location). Mono files have a circle symbol, and stereo files have a double circle symbol.

You can switch this display on or off by selecting **View > Show File Info**.



Sorting Audio Files

The audio files in the Audio list can be sorted according to various criteria by selecting **View > Files sorted by...**

Files sorted by... > None

The audio files are listed in the order in which they were loaded or recorded.

Files sorted by... > Name

The audio files are listed in alphabetical order.

Files sorted by... > Size

The audio files are listed according to size, in decreasing order.

Files sorted by... > Drive

The audio files are sorted according to the drive where they are stored (hard disk, removable drive, partition).

Sorting Regions

The display of the regions can be sorted within an audio file according to various criteria by choosing **View > Sort Regions by...**

Sort Regions by... > Start

The regions are sorted according to their time position in the audio file. This is the default setting.

Sort Regions by... > Length

The regions are sorted according to their lengths.

Sort Regions by... > Name

The regions are sorted alphabetically.

The Waveform Display outside the Region

Logic defaults to showing the waveforms outside the defined region area in light gray.

You can alter the display yourself by clicking inside the region while holding down ctrl and keeping the mouse button held down. A flip menu appears containing the following options:

None

No waveform display outside the regions.

Bright

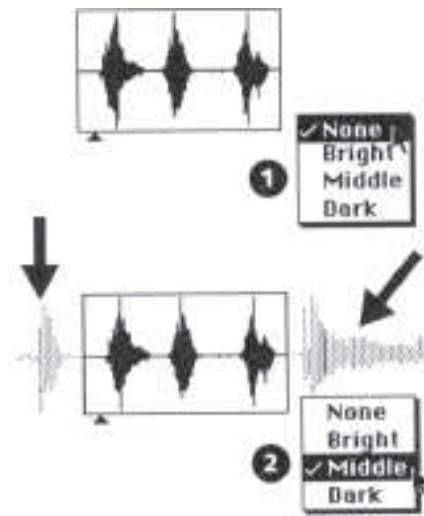
Waveform display outside the regions colored light gray.

Middle

Waveform display outside the regions colored medium gray.

Dark

Waveform display outside the regions colored dark gray.



Overview Calculation

In addition to the pure audio data, an audio file also contains data which is required for graphic display of the waveform in the Audio window and Sample Editor (“Overview” data).

Automatic Overview Calculation

If the option *Create Overviews after recording* is switched on in **Audio > Audio Preferences**, graphic overviews are automatically calculated right after audio recording.

If you switch off this option, overviews are not calculated automatically. However, you can start them manually.

Starting Overview Calculation manually

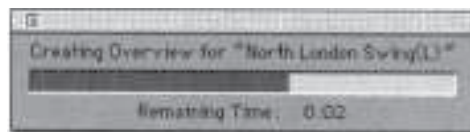
Sometimes when you load/import audio files, you have to carry out the calculation manually if the audio files don’t contain any overview data.

You can start the overview calculation of selected files with **Audio File > Refresh Overviews**.

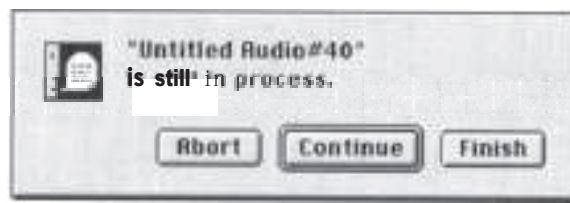
Controlling the Overview Calculation

The calculation of the overview data for an audio file is shown in a float window. This calculation is carried on in the background, so you can continue working with Logic.

You can position this float window wherever you want-the last position is saved in the Preferences.



Double-clicking this window opens the following dialog box:



If you stop the calculation by clicking *Abort*, you can still play the audio file-but bear in mind that without an overview, it will not be easy to edit.

Continue carries on the overview calculation in the background, as it normally would.

Clicking *Finish* transfers the calculation to the foreground and therefore speeds it up considerably. The disadvantage is that you cannot use your computer for anything else until it is finished.

7.3 Operation

Selection Techniques

There are several different ways of selecting audio files and regions in the Audio window.

To select a single item, either click its name in the audio list, or the waveform display.

If you hold down *shift* at the same time, you can select several items, even if they are not adjacent to one another. In the audio list, you can also “rubber band” select multiple adjacent files.

To select all items at once, select **Edit > Select All** ( .

Audio Files

Selecting the next Audio File

The key command *Select Next Audio File* selects the next audio file in the audio list.



Selecting the Previous Audio File

The key command *Select Previous Audio File* selects the previous audio file in the audio list.



Audio Files and Regions

Selecting used Regions

When you select **Edit > Select used**, all audio files and regions used in the arrangement of the song are selected.



Selecting unused Regions

Edit > Select unused selects all those audio files and regions which are not used in the arrangement of the current song.



For example, you can call up this function at the end of a production, to delete any items that are not required (.

Edit Commands

All the standard edit commands are available in the Audio window. As usual, they apply only to the currently selected items (whether audio files or regions).

The Cut, Copy and *Paste* commands only apply to exchanging items between two different songs. An audio file can only appear once in a song's audio list, and so cannot be copied within a song. For instructions on how to physically copy an audio file to the hard disk, please see the section *Copying Audio Files* on page 7-40

Cut

The selected audio files or regions are cut, i.e. moved to the clipboard. They are removed from the Audio window. Shortcut:



Copy

The selected audio files and their regions are copied to the clipboard. They remain in the Audio window. Shortcut:



Paste

The contents of the clipboard are added (providing they contain audio files and associated regions from another song).

Shortcut:

Clear

Any selected audio files or regions are deleted. You can achieve the same effect by pressing

Don't forget; the functions **Cut** or **Clear** do not delete audio files from the hard disk, they just remove them from the Audio window of the current song.

If regions from the audio files in the Arrange window are being used as audio sequences, these sequences will be deleted from the Arrange window as well.

If you want to delete the selected audio files from the hard disk, select **Audio File > Delete File(s)**.

Undo

Reverses the Previous Command (Shortcut: ). You should call up this function if you want to reverse an action.

Not all actions can be reversed using Undo. If you select a function which you cannot undo, you will be warned of this before you can execute the function.

Regions

Creating a Region

Select the desired audio file and select **Audio File > Add Region**. The new region will appear in the list, after the existing regions.



Copying a Region

If you select an existing region and then carry out the *Add Region* function, another region is created whose definition (start, end, Anchor) is exactly the same as the selected region.

Deleting a Region

You can use the Eraser tool to delete one or more regions. If any of these regions exist in the Arrange window, they will be removed from there, as well. You should therefore be careful when using this tool. However, the Undo function is always available, if you inadvertently remove a needed region.



You can also delete any selected regions with the  key.

Altering the Boundaries of a Region

You can directly alter the boundaries of a region in the graphic display, using the Region Edit (finger) tool.



You can also use the normal mouse pointer. Just make sure you grab the region by the lower third when carrying out the following actions:

Moving the Start Point

To adjust the start point of a region, grab the left border. The tool turns into a small hand pointing to the right. You can now adjust the start point.



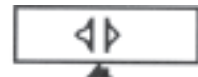
Moving the End point of a Region

To adjust the end point of a region grab, the right border. The tool turns into a small hand pointing to the left. You can now adjust the end point.



Moving the whole Region

You can also move an entire region within an audio file, by grabbing it in the middle. The tool turns into two arrows pointing left and right. If the region is very small, use the zoom function to enlarge the section. Make sure you can see the two arrows before making any alterations.



If you want to adjust the boundaries of the region without moving the Anchor, hold down  during the operation. This applies to moving the start or endpoints, as well as moving the whole region.

Moving the Anchor

To move the Anchor, grab the small Anchor triangle below the region. The tool turns into the Region Edit tool. A vertical guide line appears above the Anchor triangle.



With audio sequences in the Arrange window, the Anchor is marked by a vertical dashed line. In order to check the anchor's position, you can use the two key commands described in the section Checking the Position of the Anchor on page 9- 10.

Snapping to Zero Crossings

If **Edit > Search Zero Crossings** is switched on, adjustments to the start or end point of the region are snapped to the nearest point where the wave form crosses the zero axis. This ensures “glitch-free” playback. When adjusting the start point, the area before the selected point is searched. When adjusting the end point, the area after it. This option is useful when defining regions in the Audio window.

Search Zero Crossings also applies to all operations in the Arrange window.

Fine Movement

The graphic display in the Audio window is optimized for quick and simple organization of the audio files and regions. This involves displaying as many things as possible simultaneously in the window. Because of this, the finest resolution for positioning the start and end point, or the Anchor, is limited to units of 2,56 samples. This is usually adequate, particularly if search zero crossings is switched on.

However, sometimes you need to make precise adjustments down to the individual sample words. Drum loops are a good example. The Sample Edit window is better suited to these situations. To open the Sample Edit window for a region, double-click on that region.

Protecting the Region Parameters



Regions can be “locked”, to protect against accidentally altering the start and end points, or the Anchor position. The small lock symbol next to every region can be opened and closed by clicking it. If it is locked, you can play the region, but you cannot edit it.

This function should not be confused with delete protection: A protected region can still be deleted.

Important

Renaming Audio Files and Regions

To rename audio files and regions in the Audio window, just double-click the name in the audio list. A text input box appears where you can type in a new name.

Before renaming audio files the following warning appears:

Keep in mind other songs may use the same File! Do you still want to rename the File?

You should check whether the audio file that you want to rename is used by another song. If it is, don't rename the file, otherwise it will not be found or played by the other song.

Logic helps you in these situations:

Logic alters the name of an audio file in all currently opened songs which use this file.

If the file is half of a “split” stereo pair, Logic automatically assigns the new name to the other half of the SDII stereo pair.

Logic also renames any backup files on the same drive.

If you rename a stereo file, Logic automatically assigns the new name to up to five files (both the mono files used in Logic, their backups and the stereo file). In this case it is a good idea to store all these files in the same location. (Refer to the section on moving files.)

You can rename regions whenever you like.

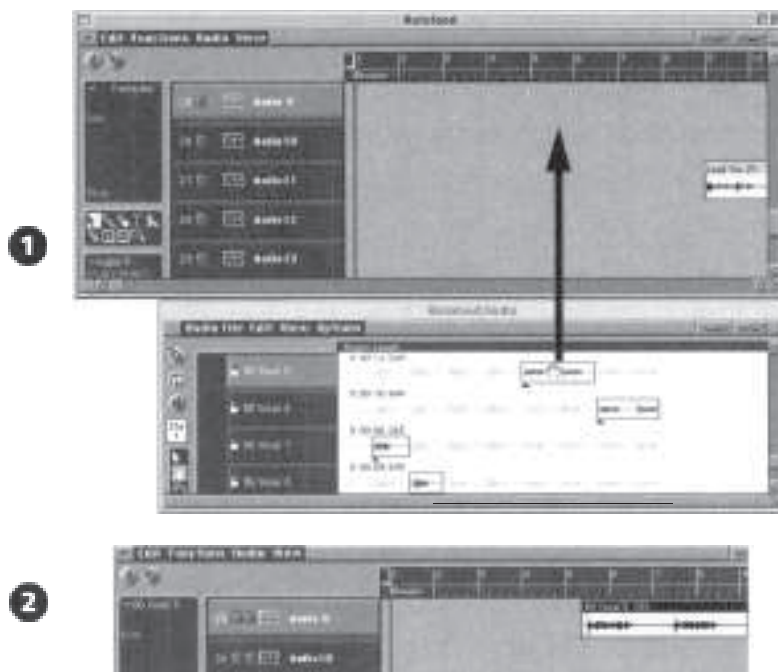
As long as regions have the same names as the audio files they refer to, any renaming of the audio files is automatically carried over to their associated regions.

Adding a Region to the Arrangement

To add a region to the song arrangement, drag its graphic display into an open Arrange window, to create an audio sequence.

Grab the middle of the graphic display of the region in the Audio window and drag it into the Arrange window. If you would like to insert split stereo files, please use the left (upper) Region.

Don't worry if the region starts to play. It will stop playing as soon as you move the mouse pointer outside the Audio window. You can prevent this by using the Region Drag tool (the small hand with the outstretched fingers), or by grabbing the name of the region in the Audio List, and dragging it into the Arrangement.



7.4 Strip Silence

The *Strip Silence* function allows you to create regions automatically. The amplitude level of the audio material is the basis used to create the regions.

Functional Principle

The principle is simple: all the amplitude values below a set threshold are interpreted as “silence,” and removed. New regions are created from the remaining passages, whose level is above the threshold.

You can choose to replace the original region in the Arrange window with these new regions, without altering the timing of any of the passages.

The main uses are as follows:

Removing background noise during gaps (noise gate).

Unlike analog noise gates, strip silence works “in advance”.

There are no problems with cutting off the attack of the region(s).

Dividing a long section into several convenient segments (e.g. for deleting or sorting).

For short percussive regions, (e.g. drum loops), you can simulate time compression/expansion by simply altering the tempo.

You can even quantize the individual segments in an audio recording-something which until now was only possible for MIDI events.

Calling up the Strip Silence Function

First, select the region in the Audio window on which *Strip Silence* is to be carried out. If you are using split stereo audio files, the corresponding region of the other channel is automatically selected too. Then select **Options > Strip Silence** in the Audio window. The Strip Silence window will open.

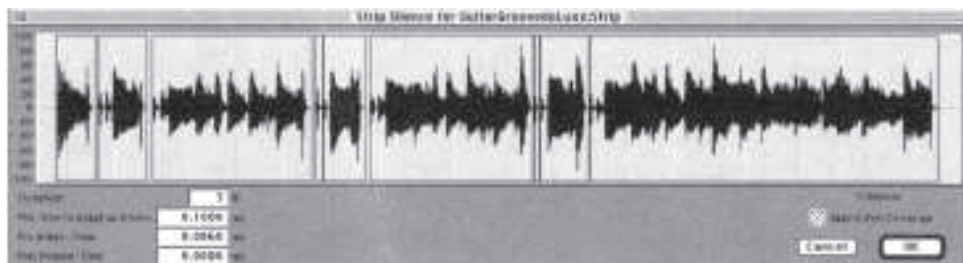


The Strip Silence Window

There are several settings in the Strip Silence window which directly affect the number and length of the newly created regions.

Strip Silence

Every time you alter a parameter in the top part of the window, the graphic display of the region changes accordingly.



Strip Silence Parameters

Depending on the audio material, all the parameters can have an effect on the number and division of the regions. To get the desired result you may need to experiment a little.

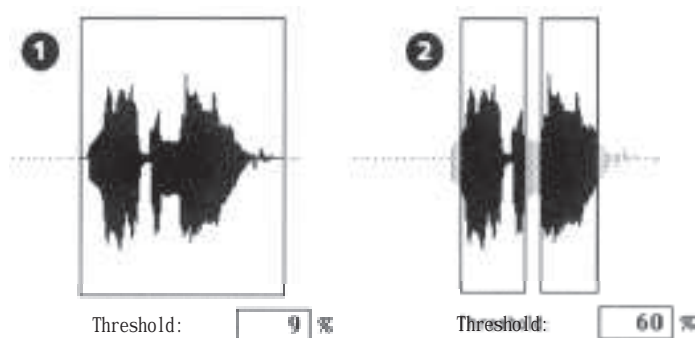
Threshold:	5 %
Min. time to accept as Silence:	0.1000 sec
Pre Attack-Time:	0.0060 sec
Post Release-Time:	0.0000 sec

Threshold

The Threshold defines the amplitude threshold that a passage has to exceed, in order to be defined as a region.

The units here are shown in percentages.

In terms of the number and length of the regions, this is the most important parameter. As a general rule: a higher Threshold leads to more short regions.



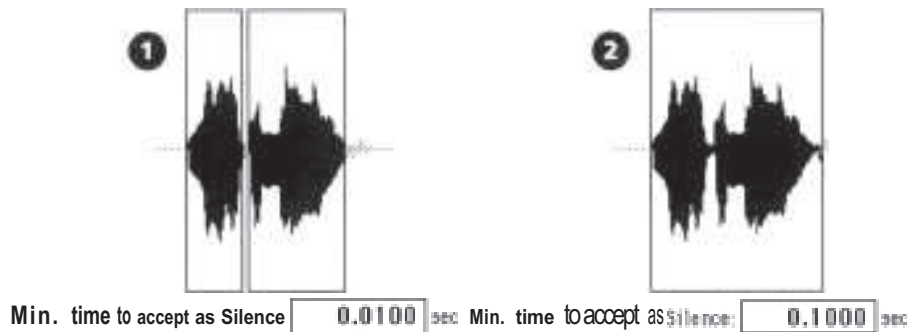
Example of "Threshold": low values ① create large, continuous regions, high values ② cause more short regions

Min. time to accept as Silence

You use *Min. time to accept as Silence* to define how long a section must be below the threshold value, before it is considered to be a gap.

The units here are shown in seconds. The smallest division is one tenth of a millisecond.

Very small values tend to increase the number of regions, because even short dips in amplitude are interpreted as silence. Higher values prevent sections of audio from being “interrupted” by fluctuating amplitudes.



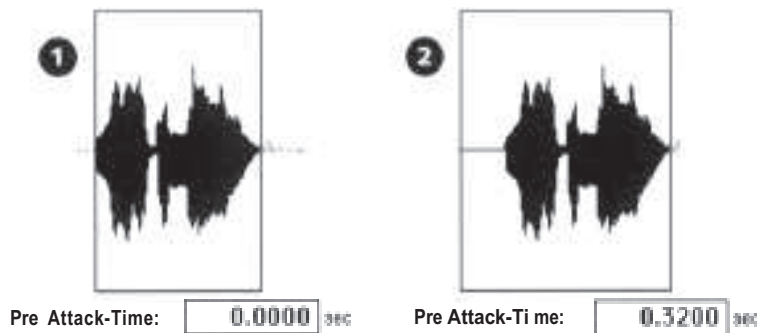
Example of “Min. time to accept as Silence”: with low values ❶ even short dips in amplitude are excluded, with higher values only longer stretches of lower amplitude are removed.

Pre Attack-Time

The *Pre Attack-Time* parameter allows you to add a pre-defined amount of “pre-roll” to the beginning of all regions created. With high threshold values, this prevents any amplitudes with slower attack times from being “chopped”. This function is useful for non-percussive material like vocals, wind instruments, strings, etc.

The units here are shown in seconds. The smallest division is one tenth of a millisecond.

Strip Silence



Example of “Pre Attack Time”: values higher than “0” pre-delay the start point.

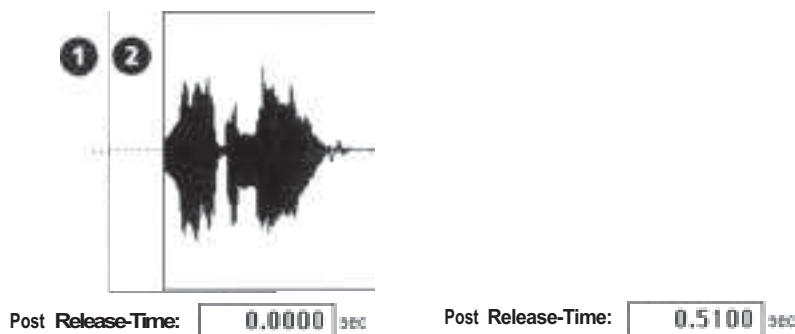
Overlaps are permitted, i.e. the pre-delayed start point of a region can extend back to the end of the previous region-but only as long as the threshold value is not exceeded.

Altering the Pre Attack Time has no effect on the Anchor position, or the absolute position of the audio data on the sequencer time axis.

Post Release-Time

Next to *Post Release-Time* you can define an automatic release time for the end points of all the regions.

The units here are shown in seconds. The smallest division is one tenth of a millisecond.



Example of “Post Release”: values greater than “0” lengthen all the affected regions

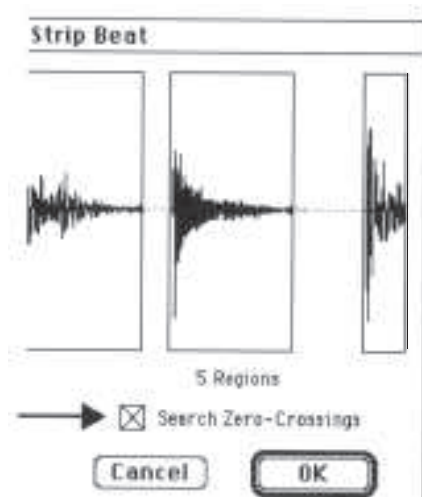
You can use this function to avoid cutting off amplitudes which fade out gradually (particularly if you have set a high threshold value). For example, cymbals, open hi hats, snares with long reverbs vocals, etc.

The *Post Release Time* parameter does not allow overlaps. This means that the region's end point cannot extend beyond the start of the next region.

Adjusting the Post Release Time does not affect the Anchor position or the absolute position of the audio data on the sequencer time axis.

Search Zero Crossings

If the *Search Zero-Crossings* option in the Strip Silence window is active, the region start and end points are always automatically snapped to the nearest waveform zero crossings.



Click the check box, to toggle the automatic positioning to zero crossings between on and off.

Replacing a Region in the Arrange

If you are editing a region which is being used as an audio sequence in the Arrange window, you can open a dialog window by clicking OK in the Strip Silence window.

If you want to replace the region in the Arrange window with the regions created by the Strip Silence function, click *Replace*. This means that the relative timing of the individual audio segments remains unaltered. You can listen to the result and if you are not satisfied you can Undo it.

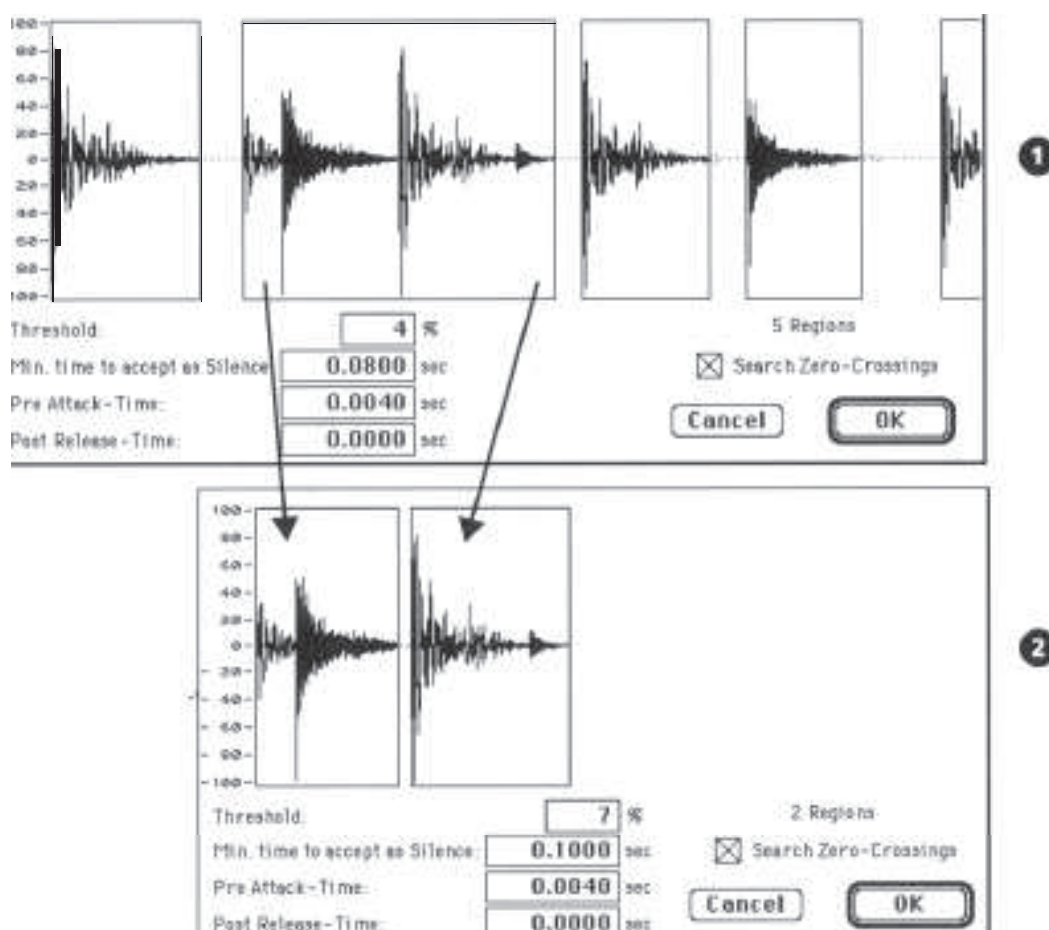
Strip Silence

If you want the new regions to appear only in the Audio window click No. You can then drag them individually into the Arrange window, in the usual way.

Multiple Editing Using Strip Silence

Strip Silence always affects the selected region in the Audio window. It doesn't matter if a region has been defined manually or whether it is a product of a previously executed *Strip Silence* function

or example, you can use this method to roughly split up a whole audio file, and then divide the new regions, using different parameters. The new regions can then be again processed with the Strip Silence function-and so on.



Example of the “Strip Silence” function applied in two stages: in the first stage ❶ a one bar drum loop is first split up into multiple sections, in the second stage ❷ this new region is divided yet again.

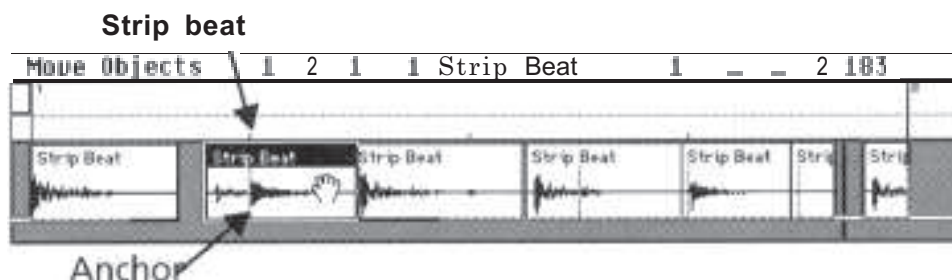
Reassigning Audio Sequences Using Strip Silence

Regions which already exist as audio sequences in the Arrange window can also be split up by the Strip Silence function. The position of the remaining audio data along the sequencer time axis is not affected. This means that a recorded audio sequence can be immediately edited using Strip Silence. Several new audio sequences are then created, which can be individually moved, copied, or deleted.

Automatic Positioning of the Anchors

When you create new regions using *Strip Silence*, a separate Anchor is automatically created for each new region. Logic always sets this to musical units, in accordance with the format value set in the Transport window. This means that when you are moving audio sequences to fit in with the music, there will be “rounded” values in the Info line.

It is recommended that you adapt the tempo, particularly for drum Loops, before carrying out the Strip Silence function (see the section Automatic Tempo Matching on page 4-21). This will ensure that the Anchors remain at musically logical positions. You should also set the desired format value, if necessary before using Strip Silence.



The Anchor is automatically set to musical units, in this case the “two” of the first bar. The *Info Line* displays musical values.

Uses for Strip Silence

Noise Gate

The most conventional use for Strip Silence is to simulate the classic noise gate effect. When used on long recordings with many gaps-such as vocals or instrumental solos-you can get good results by setting a low threshold value. Background noise is removed, without affecting the main signal.

Creating Segments of Spoken Recordings

With spoken passages, *Strip Silence* can divide a recording into sentences, words or syllables. For film synchronization or jingles, you can move or reposition the segments of speech by simply dragging them around in the Arrange window.

With tempo changes, you can also simulate a time compression/expansion effect, as the syllables automatically move closer together, or further apart.

Creating Segments for Drum Loops

Dividing drum loops into small segments is a good way of perfectly synchronizing them. For example, with audio passages where the bass drum and snare are completely separate, you can often use Strip silence to isolate each individual beat. The new regions then behave exactly like MIDI events, so you can alter the tempo to “quantize” them.

If you have a SampleCell II card you can import the regions of an audio file created by strip silence as individual samples. First you have to export (store as an SDII file) the region definitions of the audio files by selecting **Audio File > Export SDII Regions**.

Optimizing the Synchronization over long time-spans

Different computers, different synchronization sources (Internal or SMPTE code), different tape machines and - in theory - even different samplers or hard disk recording systems, will exhibit slight variations in clock speed. Changing just one component can lead to a loss of synchronization

between recorded audio material and MIDI. This particularly applies to long audio sequences.

This is another situation where the **Strip Silence** function can help, by creating several shorter audio sequences, with more trigger points between the audio and MIDI events.

Please also see the section *Audio Synchronization* on page 16-33.

7.5 File Administration

Record File

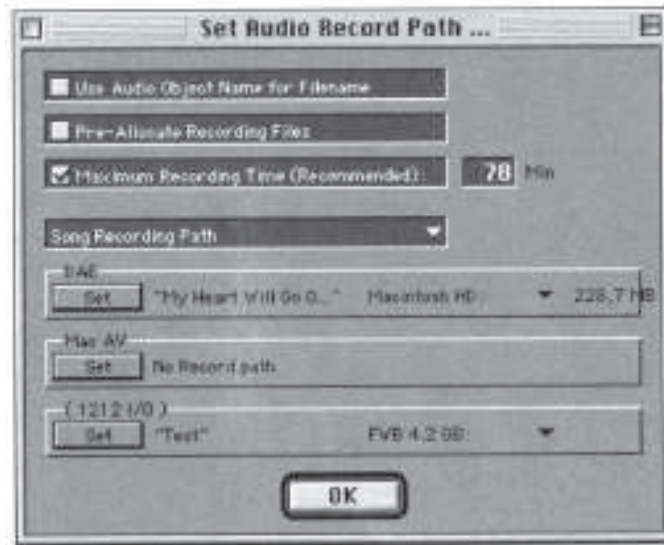
Defining the Record Path

Every recording in the Arrange window creates a new audio file. To keep track of your recordings, you should tell Logic where the data is to be written before you start recording.

You can also define a file name for the audio files that are recorded - a kind of working title - which Logic will automatically keep using, by appending a series of numbers to the file name for each subsequent recording.

The path (folder) for audio recordings can be set individually for each song. You can also have different paths for different audio hardware systems running simultaneously.

To call up the dialog window for these settings, select **Audio > Set Audio Record Path...** or open the Record menu (click-hold the Record button).



Use Audio Object Name for File Name

If this option selected, the name of the audio object is used as the file name for the recording.

This option can be very useful. You can name several audio objects for the type of parts that will be recorded during a session, such as "Bass Guitar", "Organ", etc. Logic will automatically name each recording appropriately, based on the name of the object it is recorded to.

Pre-Allocate Recording Files

If this option is selected, Logic creates the audio file for the next recording after you record-enable the track. In addition, at the end of every recording, the audio file for the next recording is immediately created. This allows you to start audio recording more quickly.

Unused recording files are deleted when you quit Logic.

Maximum Recording Time (Recommended): xxx Minutes

This parameter defines the maximum recording time for new audio files, in minutes; this determines the size of the temporary recording file.

Before any recording can take place, a file large enough to hold it must be created on the hard disk.

You can switch this option off, in which case a record file as big as the entire available space on the currently selected hard disk will be created.

Disadvantages:

the audio file may be heavily fragmented, if the actual recorded file is much shorter than the pre-created recording file.

if you are using DAE hardware there may be no space left for the Overview file. The Overview will then have to be calculated after the recording.

It is strongly recommended that you switch on this option, and set a maximum recording time.

If there is not enough storage space available to accomodate the time you have set, the maximum length of the recording is temporarily reduced. During the recording, you can see how much time is available in the Record Float window (with the red bar).

Global Record Path/Song Record Path

The Global Record Path was available in earlier versions. It is stored in Preferences, and applies to all songs. The advantage is that you don't have to define a record path in every new song before you can record, provided that you don't mind recording files for every song into the same folder.

The Song Record Path allows you to organize the audio recordings for each song into its own folder. The advantage here, is that when you switch from one song to another, any new recordings are automatically stored in the current song's folder, without having to switch the path manually.

Set

These buttons can be used to define the path for each separate hardware system. You can create a new folder (if required) in the dialog box.



To the right, you will see the current volume, and the remaining capacity (only if the volume is registered).

If you click-hold the volume, the entire path is displayed. This way you can see exactly which folder you are recording into.

Here is an example illustrating the individual steps:

It is usually advisable to store all the audio files from one recording session or song in a single folder for that song. Let's suppose the song you are working on is called "Morning Dude".

Select **Audio > Set Audio Record Path...**

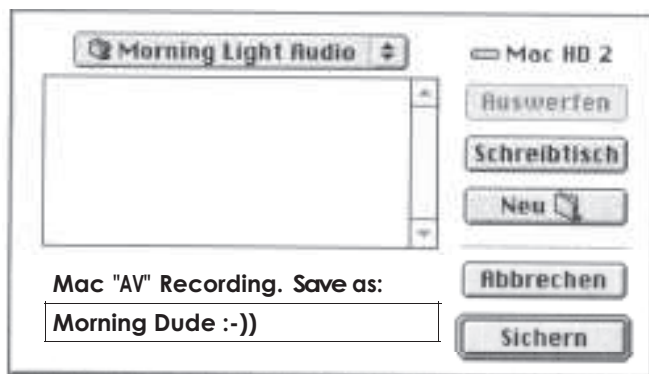
Create a new folder by clicking New **Folder** in the file selection box.

Call the folder something like "Morning Dude Audio" and click **Create**. The new folder opens automatically.



Now enter a working name for the audio files. It is a good idea to enter a name which mentions the song and the type of recording. If you are recording vocals for the song "Morning Dude" you could use something like "Morning LeadVoc" as a working name.

Then click **Save**.



For every new recording an incrementing number is added to the name. For example:

Morning LeadVoc. #01
Morning LeadVoc. #02
Morning LeadVoc. #03 . . .

You can change the path and name whenever you want, by calling up the Set *Record Path* function again.

Let's assume you want to record the lead guitar for our example song "Morning Dude". Enter a new name such as "Morning LeadGuit".

When You Choose the Path remember...

If the hard disk to which the path leads is not connected or is switched off, the path will be deleted.

If you rename the volume, Logic will not be able to find it.

When You Enter the Recording Time Remember...

Important: the display of the remaining memory and the maximum recording time depends on the following factors:

the number of tracks which are to be recorded simultaneously.

the sample rate.

If you are starting Logic for the first time, this display refers to recording a single mono track.

After every recording Logic makes any unused memory available for more recording.

It is always advisable to choose a much smaller value than the maximum possible recording time. Otherwise after the first recording there will not be enough space for further recordings, unless you reduce the value. A good practise is to set this value one minute higher than the longest single take you expect to record.

If you set a long recording time, and then increase the number of tracks, or change the sample rate, and there is not enough room on the hard disk, Logic will warn you first with an alert box.

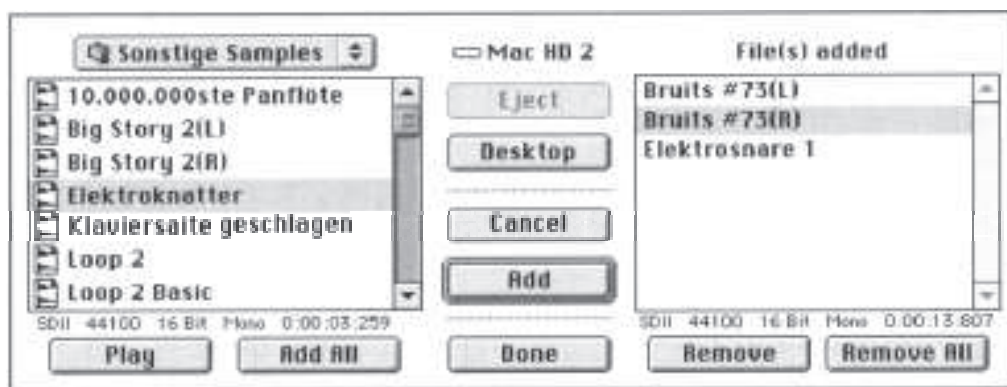
Adding Audio Files

If you want to use an audio file stored on the hard disk in the current song, select **Audio File > Add Audio File**.



An extended file selection box appears where, you can select one or more audio files at once.

Audio File Selector



Logic has a special file selector box for adding audio files, with the following features:

Adding several files at once.

Adding or removing all files in the folder.

The cursor stays at the last position after you have added a file-subsequent files can therefore be added by pressing “Enter” or “Return”.

Any files which have already been added are filtered from the view.

Auditioning audio files:

Channel 1 of the audio hardware (Dev) set in the Audio window is used for playback.

Playback can be stopped at any time.

Playback is not stopped if you change the folder or volume.

A short standby pause (a delay before the file can be auditioned) ensures that you can scroll quickly through directories.

“Done” means all selected audio files have been added to the song.

If you are using Digidesign DAE hardware, you can listen to an audio file before you load it. This window also contains extra information on the files, such as size, format and sample rate. In the “Audio Hardware & Drivers” page, you can decide whether you want to use this expanded box, or the file selector box from the MacOS.

Click *Cancel* in the file selection box when you have finished selecting files.

The file names will then be listed in the Audio window.

Drag & Drop using Audio Files

The Drag and Drop Manager of the MacOS is supported: you can drag any selection of audio files directly from the Finder into an open Audio window,

The end result is exactly the same as using the *Add Audio File...* function. It can save you time, especially with large multiple selections.

If you drag any audio file ending in "...(L)" or "...(R)" the other half of the split stereo recording is automatically moved with it.

Unfortunately, at the present time this feature works only with files in the SDII format (not AIFF). SDII stereo files are not converted automatically. (For SDII stereo or AIFF files use *Add Audio File...*).

Add Audio Files from CD

If files have been added from CD, or some other protected volume, Logic asks you for a path, so it can copy these files to the hard disk.

Importing Tracks from an Audio CD

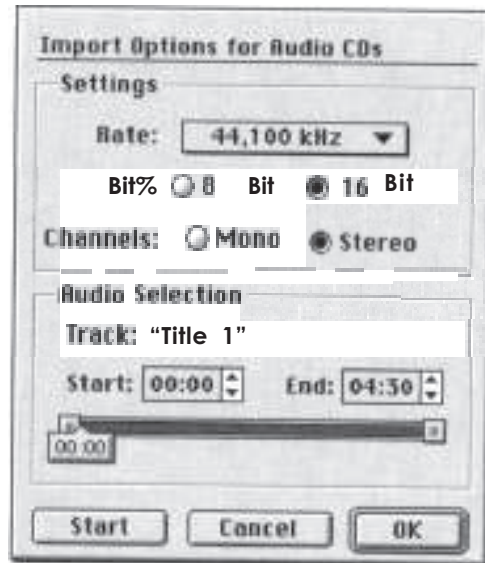
If you want to digitally import a track (or part of one) from an audio CD, you can make use of the fact that the QuickTime System expansion (contained in MacOS 7.5 or higher) stores "movies without pictures" as AIFF files. Logic can import AIFF files.

Here's how you do it

1. Go to the main Window and select **Options > Movie...**
2. In the file selection box, choose the track you want from the CD in the CD-ROM drive. (If the tracks are not listed, check that the *Audio CD* System expansion is switched on.)
3. Click *Convert...*
(If you want to create an AIFF file from the whole track go straight to step 8.)
4. Click *Options...* and this dialog box will appear:

Chapter 7

The Audio Window



5. Make the settings as shown above. (Important: Size = 16 bit. *Rate* must match the sample rate, *Stereo/Mono* depending on what you want.)
6. Define the section of the track you want, using the *Start* and *End* locators. You can enter the values either numerically, by clicking the up/down arrows, or by moving the marker on the bar. Click *Play* to preview your selection.
7. Click *OK*.
8. Enter a name for the AIFF file, and select a folder.
9. Click *Save*.
10. Close the Movie window, otherwise the audio file will play along with the Logic song as a “movie without pictures”, via the internal hardware.

Now that the audio file is stored on your hard disk, you can add it as usual in the Audio window by selecting **Audio File > Add Audio File...**

Keep in mind that it is usually necessary to have a SCSI CD-ROM drive in order to import tracks from audio CDs.

Removing Audio Files

To remove an audio file from the current song, select its name in the Audio window and press . This does not delete the file from the hard disk.

Creating Audio Files

Audio files are normally created by making a recording. After a digital mixdown of regions in the Arrange window, a new audio file is also created (see section *Digital Mixdown* on page 4-24).

Saving a Region as an Audio File

If you want to create one (or more) independent audio files from selected regions, use the function **Audio File > Save Region(s) as...** Use the file selector to select the folder where you want to store the new audio files. If you are only editing one region (or two paired regions from a stereo recording), you can also enter a name for the new audio file. If you are editing several regions simultaneously, it is a good idea to name the regions in advance, as the region names will apply to the audio files created from them.

A dialog box then appears:

Add Audio files to List of Audio Window? No / Add

If you want to add the new audio files immediately, confirm with **Return** or **Add**, otherwise click **No**.

There is a very similar function, which allows you to convert selected regions into individual audio files directly from the Arrange window (section *Converting several Regions into Audio Files* on page 4-4).

Deleting Audio Files

In theory, audio files are displayed and organized by the Macintosh operating system in exactly the same way as all the other

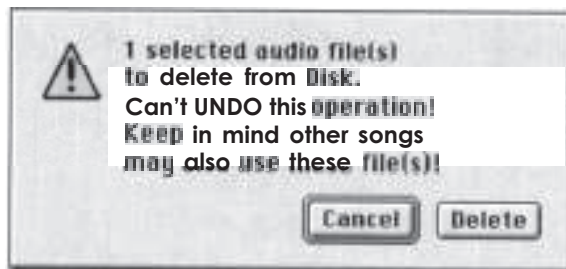
Macintosh files. You can therefore delete or copy them in the Finder. However, this has the following disadvantages:

If you delete an audio file, you may not know if it is needed in a song.

If you accidentally delete audio files, Logic warns you of this when you load the song. In addition, there will still be items in the Audio window and Audio sequences in the Arrange window which originally referred to the missing audio file, and have therefore been “orphaned”. This detracts from a clear overview of the song.

For this reason, it is safer to delete redundant audio files in the Audio window. First, select the audio files which you want to delete. Be careful, because once files have been deleted, they are gone for good. You cannot use the Undo function to recover deleted audio files.

Select **Audio File > Delete File(s)** An alert message informing you of the number of audio files about to be deleted will appear.



Click *Cancel* to abandon the deletion process, or click *Delete* to permanently delete the files.

Here too, you should make sure that the files about to be deleted are not being used in any other songs.

Optimizing Audio Files

The **Audio File > Optimize Files** function allows you to delete those sections of audio files which are no longer used in the

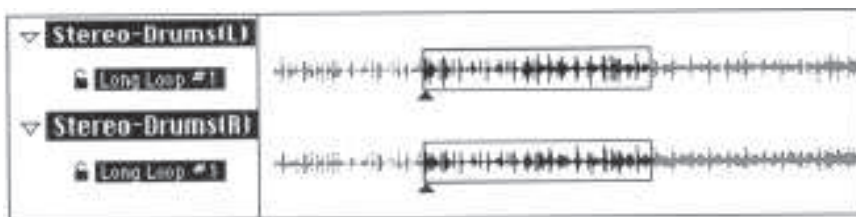


song. This can usually free up a lot of wasted space on the hard disk.

You can call up Optimize *Files* for any number of audio files. Optimize *Files* works as follows: for all selected audio files Logic determines which segments are not contained in any of the regions. These segments are then deleted and the used regions are stored next to each other in the audio file.

The regions in the Audio window must be redefined by Logic during this process. Therefore, the song is automatically saved after Optimize Files.

Because this process deletes physical data from the storage medium you cannot undo it.



Unused audio passages outside the regions often take up unnecessary storage space and can be removed using the function **Optimize File(s)**-even for several files at once.

For safety reasons, the function *Optimize Files* uses a Pre/Post Roll area. That is, before and after each used region, one second of original audio data is left.

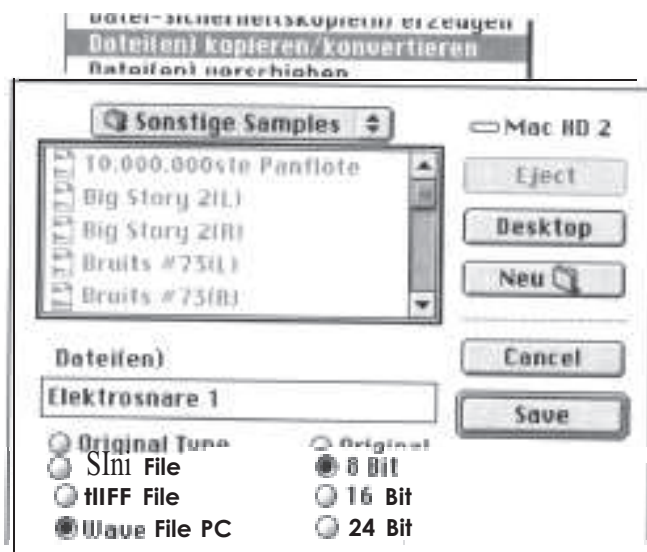
Making Backups

The **Audio File > Backup File(s)** function stores duplicates of files at the same storage location as the original(s), with the extension "dup".

Don't forget the backup options that are available in the Sample Edit window.

Copying Audio Files

The **Audio File > Copy File(s)** function copies files to a different location on your hard disk (or other storage medium). When you copy a file, you can enter a new name in the file selection box (similar to the **"Save A Copy As..."** command).



Logic checks whether there is enough space to copy the selected file(s) at the target location. If there is already a file with the same name there, Logic asks whether you want to replace it. Logic also gives you the opportunity to replace the audio file in the song with the file that you have just copied.

Moving Audio Files

The **Audio File > Move File(s)** function enables you to move audio files on your hard disk. Unlike the copy function, the source file disappears afterwards.



If the source drive/partition is the same as the target drive/partition, the files are simply moved to the other folder. This is a very quick and convenient way of organizing the drive and the song.



Example

Choose “Select Used” from the Edit menu of the Audio window (this refers to the files which are being used in the Arrange window), and move them to a new folder. This folder will then contain just the audio files from this song.

Be very careful when using this function. A different song may use the same audio files. The next time you start the other song, you will then have to locate the files that you have moved

Logic updates the information on the new path for all open songs which use this audio file. Thus, you can open every song which uses the audio file(s) you are moving. Now, move the files to the new location, and save the songs to transfer the storage reference(s).

Split and Interleaved Stereo Formats

The internal structure of multi-track HD recording systems is such that each individual audio track can be dealt with independently. Accordingly, a separate audio file is created for each track.

Logic can record stereo files in two different ways, one of which is by coupling two mono tracks and therefore two audio files. This is known as the “split stereo” format. Select **Audio > Audio Preferences > Force record & convert interleaved into split stereo file(s)** if you want Logic to use this format.

But normally, Logic creates single files containing both channels of a stereo recording. This is called the “interleaved stereo format”. These are composed of a sequence of alternate small packets of data from both tracks. Another example of this are stereo recordings made using the “Sound Designer II” program. Logic can record files directly in the interleaved format, giving stereo files recorded in Logic instant compatibility with other mastering, or CD recording applications. Please note that DAE and CBX hardware are not compatible with this

format, so all files recorded on a DAE or CBX HDR system will be in the Split Stereo format. Please read more about this topic in the section *Universal Track Mode* on page 8-8.

The sections below on “split” stereo files apply only to the above mentioned hardware systems:

Automatic Conversion of Stereo Files

When you are using DAE or CBX hardware, and add a stereo sound file to the Audio window, two mono files should be automatically created if this function is enabled.

When the automatic conversion of stereo files is selected, the original stereo sound file is not deleted-Logic generates two new independent mono files from the stereo sound file. Don't forget that this process takes up twice the amount of space on the hard disk.

The two new audio files are linked together in the Audio window to make a single “stereo unit”. Any edits which are carried out on one channel, automatically affect the other channel too.

Special Features of “Split” Stereo Files

There are a few special features for dealing with “split” stereo sound files:

In the file selection box “split” stereo audio files are treated as a single file, even though strictly speaking, they are two independent files.

In the audio list you can see both files. They have the same name apart from the channel markings(L) and (R). If you rename one channel of a stereo audio file, the file for the other channel is automatically renamed as well.

Their regions can also be renamed.

If you use the "Add Region" command to create a new region, Logic does this for both audio files.

Any alteration made to either region is automatically transferred to the other region. This applies to the start point, end point, and position of the Anchor.

If either of these regions is moved to the Arrange window to create an audio sequence, the other region is moved there, as well.

Converting Different Split Stereo Formats

Logic treats audio files whose name ends in ".L" or ".R" as stereo files. These “name extensions” are used by ProTools Session or SoundDesigner II to identify “split stereo” files. Logic’s file management functions such as *Rename*, *Create* or *Revert to Backup* can also identify these name extensions.

Convert All

When you add stereo files using DAE, or other hardware that does not support interleaved files, the files have to be converted into the split stereo format. If you want to add several stereo files in the File Selector box, Logic asks whether all the files should be converted at once (“Convert All”).

Disconnect Stereo Files

Both sides of a stereo audio file are normally edited together. However, sometimes you may wish to edit one side individually. This is now possible:

Select the stereo file

Select **Edit > Disconnect selected Stereo File**

You can now edit both sides of the recording as individual mono files.

This function will only disconnect one stereo file at a time. If you want to reconnect the files, don 't define any new regions.

Reconnect Stereo Files

The **Edit > Reconnect Stereo Files** command reconnects all disconnected stereo files in the song. Only files that were previously stereo are reconnected.

Manual Stereo Conversion

Convert to SDII Stereo

This command converts two (split-stereo) audio files into a single stereo file in the SDII format. This is useful if you want to edit a “split” stereo file using a different program, such as Digidesign's Turbosynth or Arboretum's Hyperprism.

Select the audio files and choose **Audio File > Convert to SDII Stereo**. Keep in mind that this only works for split stereo files. You can't use *Convert to SDII Stereo* to combine two mono files into a stereo file.

Logic automatically stores converted SDII files at the same location as the original split files (providing there is enough room on the drive).

Convert to AIFF Stereo

Split stereo files can be converted into an interleaved stereo file in the AIFF format using the **File > Convert to AIFF Stereo** command.

Reconvert from SDII Stereo

This command is for converting SDII stereo files back into two mono files so that after editing them with an external program you can use them in Logic again (only necessary with DAE hardware).

Precondition: the mono files must be already registered in the Audio window. (If not, just add the stereo file, which will be automatically converted).

Select the audio files (or one of them) and choose **File > Reconvert from SDII Stereo**.

The two audio files will be replaced by the externally edited stereo file.

Dealing with SDII Files

Logic allows you to access the regions in SoundDesignerII files. (The SDII data format lets you store region definitions).

Import SDII Regions

You can import regions of SDII files into the Audio window. This is useful if you want to play regions from a long recording made using Pro Tools. Select the audio file and choose **Audio File > Import SDII Regions**.

Export SDII Regions

This command allows you to export one or more selected regions from the Audio window into the SDII file. To export all the regions from an audio file, just select the name of the audio file and choose **Audio File > Export SDII Regions**.

Audio File Format Conversion

When you copy files using the **Audio File > Copy File(s)...** function, you can define the format of the destination file.

All selected audio files can be copied into any of the following formats:

...Original Type: The original format is used (SD11 or AIFF).

...SDII File: The copies are in the Sound Designer II format.

...AIFF File: The copies are in the Audio Interchange File Format

...Wave File PC: The copies are in the ".WAV" format.

Exchanging Audio Files between Mac and PC

When converting audio files to the Wave format on the Macintosh, or when converting AIFF files on the PC, you can use long file names.



Wave files can be directly played on the Macintosh, without problems, with the Emagic Audio Engine. SDII Files can also be played on the PC, as long as the files are recorded at 16 Bit 44.1 kHz. It is generally recommended, when transferring files from Mac to PC, that you convert the files to Wave format (while still on the Mac), as the Resource is lost in SDII files. (The Resource contains detailed information about format, Sample rate and Bit definition).



The use of DOS-formatted hard disks on your MacOS computer for audio is not recommended, since its audio performance is significantly worse than the performance of an HFS+ drive.

To transfer a whole Logic song from Logic Mac to Logic Windows, just go to the Audio window and copy all the audio files onto the MS-DOS drive or the PC net (if available), as .WAV files. The song file itself can be read immediately on both platforms.

Logic Windows will recognize a song which was created on a Mac and looks for "FileName.WAV" instead of the original SDII files.

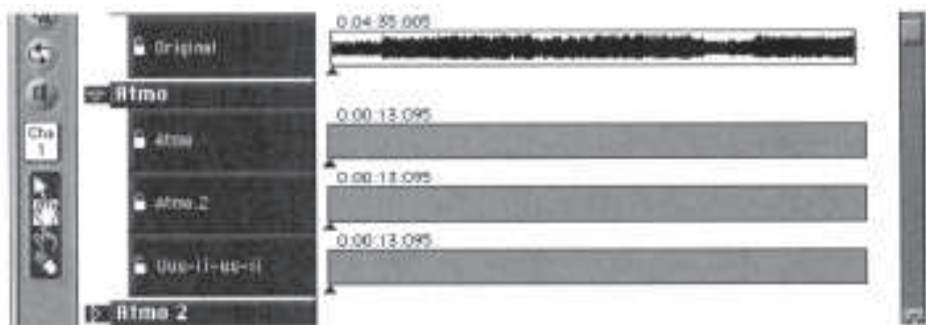
Likewise, to transfer from Logic Windows to Logic Mac, you have to export the audio files from Logic Windows as AIFF files (onto the Mac/HFS drive). Logic Mac then looks for "FileName.AIF" files instead of "FileName.WAV" files, which would be used on the PC.

Store the Logic song file in the same folder as the audio files. Then Logic will find the audio files immediately after loading.

Other Functions Involving Audio Files

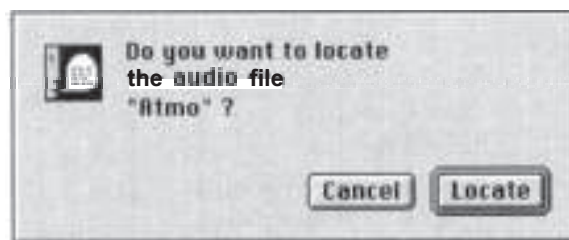
Reassigning Audio Files (Update File Information)

If Logic cannot find one or more audio files—for example when it opens a song—the regions are shown as gray areas in the Audio window.



If the files are available under a different name, or if you want to assign a “replacement” file, you can do so as follows:

Double-click the relevant region, or select **Audio File > Update File Information**.



Select *Locate* in the dialog box and a file selection box opens, where you can load the desired audio file.

What to Do if Logic Cannot Find an Audio File

Sometimes Logic cannot find a file which was previously used in the song. This could be due to one of the following

You have not connected the relevant hard disk, or you have renamed the volume.

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The Audio Window

You have stored the files in a different volume, or moved them to another volume.

You have renamed the files in the Finder, or renamed them in the Audio window of another song.

You have deleted the files.

In these cases Logic produces a dialog box. You can respond in any of the following ways:

Search

The current volume is searched for this name. If the search is unsuccessful, Logic asks you if you want to search other volumes for these files. This enables you to assemble songs, even if you have copied or moved the relevant files onto other media.

Skip

(Don't search for this file). Use this function if you know that this audio file no longer exists, or has been renamed. This button changes to "Skip All" if after the first skip, yet another audio file could not be found.

Skip All (for Several Files)

(Don't search for any more files). Use this function if you know that all audio files in this song no longer exist, or have been renamed.

Manually

You can manually define where to search for the file. A file selector appears, containing the names of the audio files which are being searched for.

If more than one file is found with a matching name you can pick out the correct file in a dialog box.

When searching for audio files (whether manually or because Logic cannot find a file at the expected position), Logic

Other Functions

searches only those hard disks to which the hardware has access. This is because:

Session 8 or ProTools II/III systems can only play audio files from hard disks connected to the SCSI bus of the disk I/O card,

CBX systems can only play audio files from hard disks connected to the Macintosh internal SCSI bus.

Fragmentation Analysis

The function **Audio File > Analyze Fragmentation** helps you analyze the fragmentation of selected audio files on the hard disk. Only the “data fork” of the audio files is examined, since this is what affects continuous playback. Defragmentation programs such as Norton’s “Speed Disk” also examine the fragmentation of the “resource fork”, which means that this program sometimes shows fragmentation where Logic shows none. However, this type of fragmentation does not affect audio playback.

7.6 Other Functions

Audio Record

You can call up the Environment Layer containing the audio objects from the Audio menu (**Audio > Audio Mixer**).

Calibration

For instructions on the calibration functions (**Options > Calibrate to Internal Hardware** please refer to section *Audio Synchronization* on page 16 - 33.

Setting the Sample Rate

In the Audio menu you can select one of the available sample rates, for example 44.1kHz 48kHz 88.2kHz or 96kHz. (If you

are not sure which one you need please refer to the Introduction manual).

The sample rate setting is global, and applies to the playback of all audio files in the current song. It is not possible to play different audio files simultaneously using different sample rates.

If you have loaded several songs, each with different sample rates, Logic can automatically take account of this when you switch songs. Each song can only contain files of one sample rate,

Some drivers and cards allow to work with sample rates like 88200 or 96000 Hz. Logic Audio Platinum supports these rates. High sample rates not only eat drive space, but also result in more CPU load, because everything has to be handled in half of the time.

Converting the Sample Rate

Logic can digitally convert sample rates:

Double-click a region in the Audio window to open the Sample Editor.

Press  to select the whole file.

select **Factory > Sample Rate Convert.**

Enter the desired sample rate in Hertz next to Destination (Hz) (e.g. 44100), and

Press .

For more detailed instructions see the section *Sample Rate Converter* on page 9-41.

Reconfiguring Your Audio Hardware

Due to faults in the driver software, the indicated settings of the audio hardware sometimes fail to reflect what is actually happening.

Other Functions

The result may be that some tracks are too loud or too soft, or it may be impossible to insert a Plug-in even though there is enough DSP capacity (on TDM systems).

You can rebuild your Audio Hardware Settings with the Key Command ***Refresh Audio Configuration***.



In the case of CBX, all Parameter settings will be sent again. In the case of ProTools, the complete virtual TDM mixer will be reset and rebuilt-including all Plug-ins. This can take several minutes.

Normally this command is not needed, but it's useful if you are working with “buggy” audio driver software.

Starting Logic without Audio Extensions

If you hold down  when you start Logic, a dialog box appears that allows you to switch off individual drivers, or all audio drivers for this particular start-up (this has been possible for DAE since Version 2.6).

You can start Logic more quickly without Audio drivers, for example if you just want to use the MIDI sequencer.

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Chapter 8

Audio Driver

8.1 Why Audio Drivers?

Logic automatically recognizes what audio hardware is available on your computer, and configures itself accordingly. Access to the hardware parameters is carried out using drivers, which are covered in the this chapter.

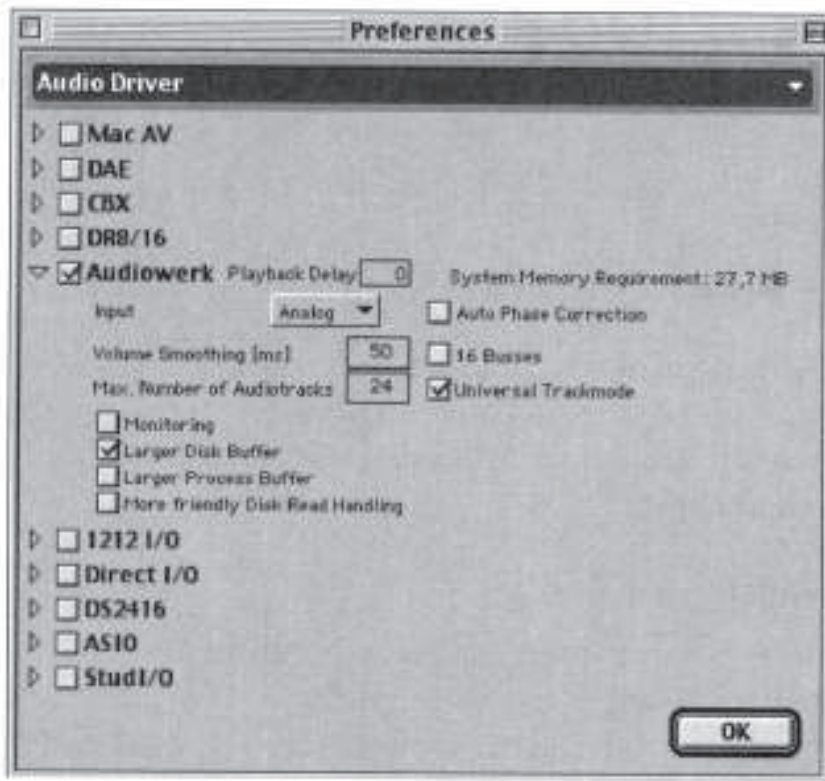
Using Logic Audio Platinum without “Audio”

If you don’t want to use a specific piece of audio hardware with Logic (for example, if you want to run Logic purely as a MIDI sequencer, or if you just want to use the internal computer hardware for listening to CDs), select **Audio > Audio Hardware & Driver**. Uncheck the box for the audio hardware you normally use.

You may have to restart Logic before any new settings become effective. When activating a driver (using the **Audio > Hardware & Drivers** option) there is an option in the dialog window (“Try Re-Launch”).

This process is always only an attempt to start, as it is possible that not enough RAM is available, Making memory available to the driver is simpler when starting Logic Audio.

If the attempt fails, you should leave Logic and restart. The driver will then be loaded with the program.



Available Drivers

Any Audio hardware requires an appropriate driver as a communication link to the Logic software. Some of the Drivers support several different types of Audio Hardware. Here is a brief list of the Drivers available. These are described in more detail later:

Mac AV

The driver for the AV hardware, i.e. the computer's audio inputs and outputs, is covered in the Installation manual. Please refer to it for the relevant settings in the Sound control panel. The Mac AV driver has several parameters which are described in the section *Common Parameters* on page 8-6.

DAE

DAE stands for *Digidesign Audio Engine*. This program is the driver software for many Digidesign cards. New cards, like "Project II" are not supported by the DAE, but by the driver

“Direct I/O”. If you use an older PCI card, such as an Audiome-
dia III or a ProTools System, please be aware that you may
control your system with either DAE, or the Direct I/O driver.
You’ll find detailed information about DAE in the section **DAE**
on page 8-9, and more about Direct I/O in the section **Direct I/O**
on page 8-38. It’s not possible to use DAE and Direct I/O
simultaneously. The DAE driver for Logic Audio Gold
supports all Digidesign audio cards except TDM hardware,
with Logic Audio Platinum it also supports TDM hardware.

CBX

Logic supports the CBX D3 and CBX D5 hard disk recorders
by Yamaha. You’ll find more information about the CBX driver
in the section **CBX** on page 8-15.

DR 8/16

Logic supports the DR 8 and DR 16 hard disk recorders by
Akai. You’ll find more information about the DR 8/16 driver in
the section **DR 8/16** on page 8-17.

Audiowerk

If you want to use Audiowerk8 or Audiowerk2-no problem. It
will be recognized by the program, and configured automati-
cally. **Input** selects between the digital S/PDIF - and the analog
Input. **Monitoring** selects whether a monitor signal is output or
not. The Audiowerk Driver supports the features of the Audio
engine, as described in the section **Audio Engine** on page 8-5.

With the Audiowerk2, “Output 1-2” are considered as the analog, and “Output 3-4”
as the digital S/PDIF outputs.

1212 I/O

The Korg 1212 I/O is a PCI-Card with 8-Channel ADAT-type I/
Os and two-channel analog and S/PDIF I/O. Its parameters are
described in the section 1212 I/O on page 8-20.

Direct I/O

This is the driver for newer Digidesign products like Project II, that can't be controlled by DAE. Direct I/O may also be used with other Digidesign-PCI-Cards, like Audiomedia III. In comparison to DAE, there are more tracks, there are native Effects and VST-Plug-ins. You also can control a ProTools III Core System and even ProTools 24 (d24) or ProTools 24 MIX-Systems via Direct I/O. But please keep in mind that using these system with Direct I/O disables all TDM functions. The Direct I/O Driver supports the features of the Audio engine, as described in the section *Audio Engine* on page 8-5. Read more about Direct I/O in the section *Direct I/O* on page 8-38, and more about DAE in the section *DAE* on page 8-9. It's not possible to use Direct I/O and DAE simultaneously.

Sonorus StudI/O

This hardware supports 24-Bit recording, high sample rates such as 88,2 or 96 kHz, when used in conjunction with Platinum, and has ADAT and S/PDIF-In/Outs. Read more about this driver in the section Sonorus *StudI/O* on page 8-40.

ASIO

ASIO stands for "Audio Stream In/Out", a standard introduced by Steinberg for communication between audio software like Logic, and audio hardware. The number of supported devices changes continuously. Please see the ReadMe file on the program CD. It contains hints how to adjust the preferences for each type of audio hardware. The ASIO Driver supports the features of the Audio engine as described in the section *Audio Engine* on page 8-5. You'll find more Information about ASIO in the section *ASIO* on page 8-40.

Yamaha DSP Factory (DS 2416)

You can read all about the driver for this digital mixing desk hardware in the section *Yamaha DSP Factory* on page 8-43.

8.2 Audio Engine

The features available vary somewhat from driver to driver. The Audio Engine used for Audiowerk, Mac AV, Korg 1212, Direct I/O, Sonorus StudI/O and ASIO offers the most flexibility. This covers the vast majority of available audio hardware:

16 and 24 bit audio files can be played back simultaneously with Logic Audio Platinum. Playback of 24 bit audio files naturally is only possible with 24 bit hardware, and therefore is not possible with the Audiowerk.

Windows Wave files (*.wav) can be played back directly. They appear in the file selector box, when adding an audio file, provided the file type is "WAVE", or the file name ends with the extension "*.wav" Important: Please avoid playing back from MS-DOS formatted storage media, as the performance is very poor under Mac OS. Please copy the files to an HFS drive (Mac OS formatted) beforehand.

Level meter for bus and output objects. The level meter displays the signal after the Insert Slots (Plug-ins).

Audio tracks can be routed to busses (in addition to the regular output routing.) This allows you to create sub groups (e.g. all drum tracks) and edit them together, e.g. with filters.

Busses can be routed to other busses. This extends the routing options, e.g. for processing combined sub group signals.

Bouncing to 8, 16 or 24 bit audio files using the SDII AIFF or Wave file formats.

The best fader response and volume smoothing possible.

Best performance possible with internal, or VST plug-ins.

Stereo Recording. Stereo recordings normally create an interleaved stereo file (single file for a stereo recording).

If the files have to be compatible with Digidesign DAE/TDM Systems, split stereo files have to be created. Select **Audio > Audio Preferences > force convert interleaved into split stereo files** for DAE/TDM compatibility.

Common Parameters

The drivers mentioned above not only have the audio engine, but therefore also the following parameters in common:

Volume Smoothing [ms]

This parameter defines the length of the fade between two consecutive volume values for an audio track. When setting this value to 0 you might hear “zipper noise” when moving a volume fader during playback. Higher values soften the volume changes and eliminate the zipper noise.

Max. Number of Audiotracks

The Audio Engine requires free system memory, which is not assigned to Logic or any other application. The amount of memory needed depends on the maximum number of tracks to be played, and on the number of I/O channels supplied by the driver. This setting allows you to reduce the amount of memory used by the driver, by reducing the number of tracks. This may be sensible when you want to run other applications or drivers simultaneously.

20/24 Bit Recording

With Logic Audio Platinum, you can record 24 bit files when this setting is active. Please keep in mind that this only makes sense if you are actually using a 20 or 24 bit interface. As long as your audio hardware is capable of this, you can select this option in the **Audio > Audio Hardware & Drivers** window. 20 or 24 Bit Recordings offer a significant improvements in the available dynamic range, however, they require high quality peripheral components.

96 kHz

Some audio cards allow high sampling rates such as 88.2 or 96 kHz. Logic Audio Platinum supports these high sample rates. The sample rates, as usual, can be selected in the **Audio > Sample Rate.** window. Keep in mind that with these higher sample

rates, not only that twice the space on the hard disk will be needed, but also that the Audio Engine will be required to perform twice as fast. Furthermore, you should consider that the improvement in audio quality of 96kHz recordings compared to 44.1 kHz in comparison with the differences between 16 and 24 bit recordings, is little. Many audio engineers consider 44.1 kHz 24-bit recording to be the best balance of sound quality, and efficient use of resources.

Monitoring

This option allows you to switch monitoring (i.e. listening to the actual input signal) on or off. Please note that monitoring is processed only via software - a certain delay is inevitable (see “Buffer Size”). If you are listening to the recorded signal through your mixing desk, you should switch this option off.

The Korg 12/12 I/O features latency-free monitoring.

Larger Disk Buffer

This option influences the amount of audio data that is read from the disk in advance. Since version 4.0, this option is switched off by default, matching the demands of fast hard drives and powerful computers. If you get frequent error messages while running Logic Audio in this mode, you should switch this setting on, so that you can play back more tracks, achieving higher reliability. However, more RAM is needed in this case.

Larger Process Buffer

This parameter determines the size of the native buffer used to compute mixers and effects. Do not activate this option if you own a fast computer (G3). This shortens response times to operations such as volume changes or Solo.

When you're working with older PowerPC CPUs, the smaller process buffer may cause problems. In this case, activate the option. This will increase the number of effects you can use simultaneously.

When in doubt - e.g. if you own an older Power Macintosh model but are using a G3 card - experiment. Experiment to find the setting that coaxes the best performance from your system.

More friendly Disk Read Handling

Activate this setting when using other Audio drivers simultaneously, if you experience playback problems. This gives the other drivers more time for their disk access. The maximum number of audio tracks for this driver may be reduced when this option is enabled.

Universal Track Mode

With **Audio > Audio Hardware & Drivers > Universal Track Mode** engaged, you can play back adjacent stereo and mono regions on a single track. The even-numbered audio objects won't be regarded as the right channels of the odd -numbered stereo audio objects to their left, and every audio object has it's own mono/stereo switch. Depending on whether a mono or stereo region is played back, the pan knob will behave as a balance or pan control. If you play back a mono region and the pan is set to the center position, both channels of the audio object will output the same level. Please note that the Universal Track Mode has limited routing capabilities.



If the Universal Track Mode is not engaged, mono regions will be played back only on the left or right channel of stereo tracks. You can only route the signal to single bus objects. The Non-Universal Track Mode has the advantage of more routing capabilities, such as additional output sends. You can also route the signal to stereo sends and pairs of busses (stereo busses), if you switch Universal Track Mode off.



The Non-Universal Track Mode is useful if you want to play different mono files for left and right mono channels of one audio object, even when it is linked to be a stereo track. An inserted stereo/stereo plug-in on that stereo linked track receives different signals for left and right, which is useful for vocoder like plug-ins.



In order to change tracks to or from DAE/TDM always switch Universal Track Mode off. Also, be aware that DAE/TDM currently does not work with interleaved stereo files, but most other drivers like Audiowerk8 Direct I/O or ASIO and others do. If

DAE

your tracks have to be switched between DAE/TDM and others, split stereo files should be used. To do so, enable the following switch: **Audio > Audio Preferences > Global > Force convert** *interleaved into split stereo file(s)*.

After changing the Universal Track Mode setting, you have to reboot Logic before the change will take effect.

8.3 DAE

With Logic Audio, the Digidesign Audio Engine is currently used to drive the following audio hardware:

AudioMedia II and III
Session 8
SoundTools
Protools II, III, and 24
ProTools Mix

As Digidesign TDM systems also use the DAE (Digidesign Audio Engine), you should read the first two sections of this chapter, if you have a TDM system. For information on the special features of TDM, see the section ***TDM Hardware*** on page 8-22.

Starting Logic Audio without DAE

If you hold down the  key when you start Logic, a dialog box appears, which allows you to switch off individual drivers, or all audio drivers, for this particular start-up. You can start Logic more quickly without Audio drivers, if for example, you want to use just the MIDI sequencer.

Digidesign Hardware

The **Digidesign Hardware Setup** dialog box contains all the important System settings for your Digidesign hardware.

This is where you determine whether Logic will use the analog or digital inputs when recording.

You can also check whether there are any communication problems with your hardware—for example if your hardware is not being recognized properly.

If you own several pieces of Digidesign hardware, you can select from among them here.

Opening the Setup Window

You can open the Setup window from within the Audio window by selecting **Options > Digidesign Hardware Setup**.



Select Card Type

This is where you select the card whose settings you want to adjust. Please note that Pro Tools III hardware can be treated like a Session 8 card, if you wish.

Cards to Use

If you have more than one card of the same type, this is where you select the one whose settings you want to change, by entering the relevant PCI or NuBus slot number from a flip menu, showing all the possible numbers.

Interface Options

This is where you make the settings for a given card.

Interface

Some Digidesign systems offer you a choice of audio interfaces, when you buy them. You set the kind of interface you're using from this box.

SampleRate

This is where you set the sample rate you will be using in your recordings. All the audio files you record with Logic (or any other software) will be recorded at the rate you enter here. You

can select either 44.1 or 48 kHz. This parameter corresponds to the setting in the Audio menu.

Sync Mode

This is where you select the source for the system's clock signal (or bit clock-pulses sent out at the sample rate frequency). You have the following options:

- Internal -

The Digidesign hardware's internal sync signal is used. It runs at the frequency you've selected in the *Sample Rate* box above.



The following note applies to ProTools only:

In a TDM system, only one of the cards determines the sync rate; the card in the NuBus or PCI slot with the lowest number. This *should* be the Disk I/O card. You can tell which is the "Master sync card" by its glowing red LED.

- Digital -

The sync signal received at the digital input is used.

Any digital output, whether AES/EBU or S/PDIF is constantly sending clock information, even if a signal is not passing through it. When this setting is on Digital, the sync signal is used by the Digidesign hardware, not only when recording, but also as a sync clock for the whole system for playback. If this box is set to Digital, please be sure that the digital device connected to the input is never turned off, or you will lose your system sync signal.

For example, if you are using a portable DAT recorder that saves power by turning itself off when not in use, you should select Sync Mode: Internal immediately after recording from the digital input.

CH 1-2 Input

This is where you choose whether you want to use the analog inputs (for which you select the *Analog* setting) or the digital inputs (for which you select *Digital*).

If you are using an audio interface with more than one pair of digital inputs (e.g. an 888 I/O), you will also be able to select any of the extra input pairs from here.

Swapping Digidesign Hardware

If you have more than one kind of Digidesign hardware installed in your system (for example Pro Tools and Session 8) you can swap between the different types by selecting **Options > Exchange Digidesign Hardware** from the Audio window (provided the Session 8 card is the first card in your computer).

Pro Tools can read audio files from the hard drive attached to the Session 8 card.

Setting a Level for Recording

In a studio, the recording level will normally be set at the bus fader on the mixing desk (or directly at the sound source). The recording machine should be set so that a level of 0 VU (Volume Units) corresponds to the relevant reference level of +6 dBu (radio), +4 dBu (professional studio) or -10 dBV (home recording).

However, if in certain situations, you need to be able to set the recording level at the input of Logic, some HDR hardware will allow you to do so:

Open the Audio window and select **Options > Digidesign Hardware Setup**,

Click **Other Options**.

You can now set the recording level for the AudioMedia card.

On a ProTools system with the 888 I/O audio interface, you can calibrate the recording level, using the “Calibration Tool” software supplied by Digidesign. For example, this is very useful if you want to set the reference level to give you greater headroom.

Session 8 / ProToolsProject

The first time you start the program with Session 8 or ProTools Project hardware, and DAE Version 2.95 or later, Logic asks which driver mode should be used: Project or Session 8.

The **Audio > Audio Hardware & Drivers > Use Project Driver Mode for Session 8 System** option means you can swap modes later.

Advantages of Project drivers:

- Allows scrubbing of up to 2 audio tracks
- I/O routing for every track, directly from the audio objects
- Up to 4 sends per track

Disadvantages of Project drivers:

- Internal/External mix modes not available
- Routing (with Studio Interface) not available
- Only 4 EQs available in total (spread over all 8 tracks).

To help you decide:

If you want scrubbing, use Project driver mode.

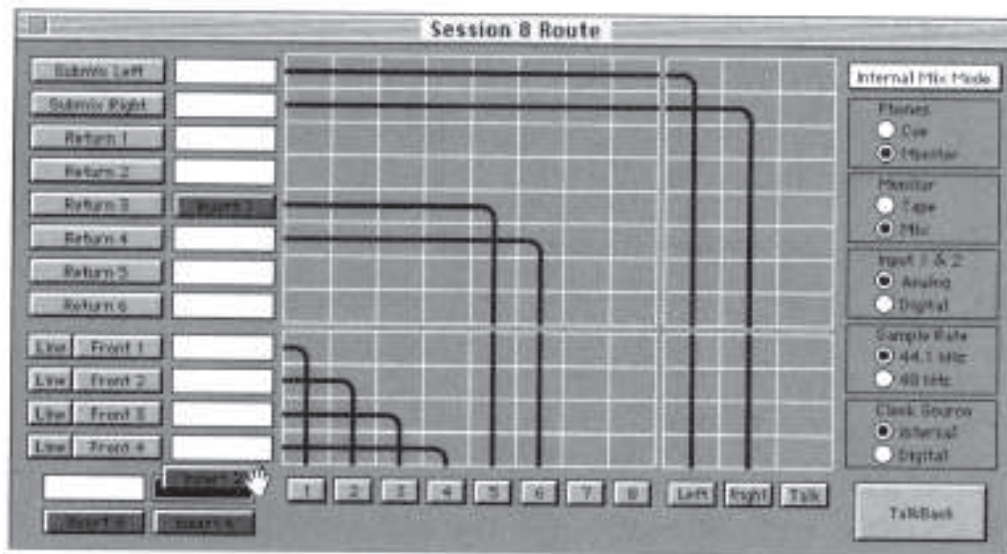
If you want 8 EQs and Session 8 routing, use Session 8 mode.

Plug-in Start Window

Whenever you start TDM Plug-ins, a small status window opens. If there is a problem, this window should show which Plug-in(s) are the cause. If you want to disable this window, you can switch it off, by selecting **Audio > Audio Hardware & Drivers > DAE/TDM > TDM Setup Indicator** (default: on).

Session 8 Routing

You can determine the Session > routing from the audio object.



You use this window to set the various Session 8 routing options. For more on these options, please refer to the Session 8 manual.

Audio Objects with Session 8 in Internal Mix Mode

As with Pro Tools II, additional sends have to be set up individually for each channel. Click on one of the fields under the word Sends. You use the pull-down menu that appears, to set the number of the required send. To the right of this is the control used for setting the aux send level.

Internal Mix Mode allows you to control the stereo master channels. Simply define audio objects with the setting **Output 1** for the left-hand channel, and **Output 2** for the right. The master channels can be controlled by one stereo fader, if they are set to **Output 1-2**. The **Output 3/4** settings can be used for fader control of the Cue L/R channels, while the **Output 5-8** settings can be used for Aux Master Send controls.

Session 8: Points to Note

The Session 8 Mode is available with Logic Audio Gold, but not with Platinum—for technical reasons. Platinum-customers can use the

Gold version without any disadvantages. Platinum supports TDM system instead.

Overviews

Session 8 hardware cannot calculate, or display any overviews (the Logic graphic files that represent the waveforms you store) while recording. The overviews are created immediately after recording stops. Once you've finished recording, however, the overview calculation is carried out "in the background", so you can engage other functions while the waveform display is being created-you don't have to sit and wait.

Scrubbing

Audio scrubbing is not possible with Session8.

Sampling Rate

Session8 has two fixed sampling rates: 44.1 kHz and 48 kHz. This means that you cannot calibrate it to external timecode, as this function varies the sample rate to match the existing timecode. It's possible that Digidesign will revise this part of the DAE software, to make Session 8 compatible with this function in a future version. As a user, you should remain in close contact with Digidesign, so you can enjoy the benefits of the latest available software updates.

8.4 CBX

Logic Audio Platinum supports the h CBX-D3 and CBX-D5 hard disk recording systems by Yamaha.

CBX-D3

If the CBX-D3 operating system software is not yet installed, LOGIC Audio automatically installs it. While it does this (it

happens after booting up LOGIC Audio), the message "*Initializing CBX*" will appear on the screen.

The CBX-D3's sample rate can be calibrated to an external SMPTE signal, or external MTC. Please note that the external clock source should run at the same speed when calibrating, as it will at mixdown. The best way to insure this is to calibrate immediately before mixdown.

Using a CBX-D3 and -D5 together

Please give the CBX-D3 the lower SCSI-ID.

In the **Audio > Audio Hardware & Drivers** window, select *Continuous Playback* under *CBX Driver Mode*. If the hard disk then seems too slow, change the setting to *Mixed Mode*.

Restrictions

Level Metering while Recording

The current version of the CBX operating system software does not allow level metering while recording. This is not the fault of LOGIC Audio. Instead of the correct level, the meters on CBX audio objects will cycle up and down to show activity. Please use the level meters on the CBX itself.

Overview

The calculation of the Overview waveform display in the Sample Edit window only takes place after recording.

Just to remind you: the overview calculation always runs in the background, so you can continue to work while it is calculated. Double-clicking on the overview float window opens a dialog box, which can be used to do the following:

break off the calculation (Abort),

carry on calculating (Continue),

or complete the calculation more quickly by bringing it into the foreground (Finish). If you select this option, you will not be able to perform other tasks while the overview is calculated.

Pan

CBX-D3 hardware does not have panning facilities. Individual tracks are hardwired to the outputs with the same number as the track.

Short Regions

Short sections (either regions, or selected areas in the Sample Edit window when the playback monitor and cycle mode are both engaged) should not be shorter than about 7000 samples, if you want them to play back correctly. If you have a badly fragmented or slow drive, this minimum length could be as much as twice as large. Perfect reproduction of sections shorter than this cannot be guaranteed.

8.5 DR 8/16

Logic supports the DR 8 and DR 16 hard disk recording systems by Akai. Please note the following limitations:

The DR is the clock master, meaning Logic Audio Platinum is slaved to MIDI Time Code (MTC). Under **Options > Settings > MIDI Options**, you should set "MIDI Machine Control (MMC)". MIDI and SCSI connections are required.

Logic Audio Platinum saves the current playlist found in the DR. When quitting Logic Audio Platinum, it will prompt you to send back the original playlist found when Logic Audio Platinum was launched. This option can be switched off under **Audio > Audio Hardware & Drivers > DR8/16 > Restore original playlist before quit**.

Logic Audio Platinum is able to display the output levels for each audio track, reading them from the DR through SCSI.

This may cause SCSI noise. You can switch off this function under **Audio > Audio Hardware & Drivers > Request Level-meter thru SCSI**.

The DR has its own disk(s), the contents of which are invisible to the Mac. For this reason, the DR file references are mirrored on the Mac bootdrive in a folder named "(DR-Files)". This folder is automatically updated to the current state found in the DR everytime Logic Audio Platinum is launched, so you don't need to be concerned about it. These "Alias" files are needed to save basic information, and the overview data. The files are small.

Arrange:

Logic Audio Platinum allows you to grab the whole playlist from the DR, and adds all regions to the Arrange in one operation. Select **Audio > Akai DR8 > Get complete playlist from DR8/16**. This is useful if you have recorded into the DR outside of Logic, and wish to begin working with that recording in Logic. This is an update, existing regions are replaced by the new ones.



The same is possible for one, selected audio track. (**Arrange > Options > Audio Setup > Get playlist of selected track from DR8/16**).



Any new recording is to be handled on the DR itself. When a recording is finished it is easy to grab the new recorded regions using the above described functions.

To update the DR's playlist to the new or changed regions, the whole playlist must be sent from Logic Audio Platinum to the DR. This happens automatically, after a short delay. The DR stops if this happens during playback.

View:

Logic Audio Platinum gets the overview data from the DR for each file. The resolution is ten times smaller than is normally the case in Logic, but it is enough to see what's going on.

Audio window:

All DR files used in the Arrange are automatically added and displayed in the Audio window as usual.

The Audio window allows all DR Files to be added in one operation, using the menu entry **Audio File > Add all DR8/16 Files. New** regions can be created, edited and added to the arrange as usual.

The DR files can be renamed with the DR limitation of 10 characters. (Note, that sessions created within the DR itself might use these files).

Selected audio files can be deleted using **Audio File > Delete File(s)**. This helps to recover disk space within the DR. (Note: this operation can't be undone).

Selected audio files can be copied to a Mac drive by using **Audio File > Copy File(s) > SDII** or AIFF. Note, that this can take a while, due to the SCSI performance of the DR (see the section *Sample Edit* below).

Environment:

As with other audio systems, audio objects are created for the DR tracks.

With the current DR firmware, volume and panning changes are sent to the DR thru SCSI. This is slow, but this behavior may be changed when the MIDI Sysex structure is made available by Akai.

Sample Edit:

It is possible to open the Sample editor, but you will be warned (an alert is given the first time): that the read/write performance (speed) is very poor. If you start a process, it can be aborted by typing  . This behavior is not due to any a problem with the Mac or Logic Audio Platinum, but is a limitation in the DR's architecture.

8.6 1212 I/O

System Requirements

You need a PowerMac, or MacOS compatible computer with PCI expansion slots to run a Korg 1212 I/O. Be sure to consult the manufacturer's documentation to ensure that all equipment is installed and functioning correctly.

Please make sure that the System extension "1212 I/O" is in the folder "Extensions ", inside your System folder Please use version 1.1 (or higher). Ask your dealer or Korg distributor an update, if necessary.

Features

1212 with Logic

Logic Audio Platinum currently supports operation with one 1212 I/O card. This card has 8 inputs and outputs in ADAT Optical Link Format, 2 inputs and outputs in S/P-DIF format, and 2 analog inputs and outputs. All inputs and outputs can be used in parallel.

Audio Tracks

If the 1212 driver is switched on, whenever you create a new song, all the necessary audio objects are created automatically. You can copy this Environment layer into your Autoload Song.

Bit Depth

The Korg 1212 S/PDIF in and outs support 20 Bit-the other inputs and outputs (Analog, ADAT) only deliver a 16 Bit signal. Logic Audio Platinum supports 20 Bit recording and playback, by recording 24 Bit files, the last 4 bits of which are ignored.

De/Reactivating Drivers

If you wish to run Logic without the 1212 Extension, or any other audio drivers-in other words, simply as a MIDI program-do the following:

Open **Audio > Audio Hardware & Drivers**

Deactivate the check box next to “Korg 1212 I/O”.

Quit Logic.

The 1212 Extension will be deactivated on the next program start. Less RAM is required when the driver is deactivated. To reactivate any installed driver, simply activate the appropriate check box.

Special Functions

Input Gain

The Korg 1212 I/O allows the regulation of input gain. Open **Audio > Audio Hardware & Drivers**, and select an appropriate value between 0 (off) and 255 (maximum).

Synchronization

In the dialog window **Audio > Audio Hardware & Drivers** you can select between these Word Clock sources: Internal, S/P-DIF, ADAT.

Audio Objects

Input Routing

In every Track Object, you can select the input, by using the top button in the I/O section.

Output Routing

In every Track Object, you can choose the output pair, by using the lower button in the I/O section. Select either the left (odd) or right (even) channel of the output pair, by using the Pan control.

8.7 TDM Hardware

Logic Audio Platinum (not Gold) allows you to make use of the TDM functions supported by ProTools III, ProTools 24 or Mix Plus.

What is TDM?

TDM stands for “Time Division Multiplexing”, the time-interlaced transmission of several digital audio signals through one data bus. This bus system is physically isolated from the computer system bus, and runs between the individual TDM-capable plug-in boards. 256 digital audio signals with a word length of 24 bits each, can be transmitted on the TDM bus, comprising the signal paths within a virtual mixer. These signal paths are necessary to insert plug-ins, which are calculated on the DSP card (DSP = Digital Signal Processor), into the individual channels, or to select them through auxiliary buses, or to direct the 8 digital outputs of a SampleCell II TDM card to the virtual mixer.

Please note, that with the ProTools III hardware, you can only record if the audio hard drive is connected to the disk I/O card. A hard drive connected to the computer’s SCSI bus cannot be used. The disk I/O card has its own, independent SCSI-Bus, even though the “Mount Digi” software included with Digidesign or the “Digidesign System INIT” show the connected disk(s) on the MacOS desktop. The newer ProTools D24 and ProTools Mix cards no longer use their own dedicated SCSI bus, and can use any drive mounted to the System bus.

However, you should always use a drive from the approved list provided by Digidesign.

System Requirements

To operate with TDM support, you will need a Macintosh, Power Macintosh, or a MacOS compatible computer with properly installed ProTools III (PCI or NuBus), ProTools 24 or ProTools Mix audio hardware.

In order to utilize the 24-bit capacity of ProTools 24 or ProTools Mix, Logic Audio Version 3.012 (with an installed TDM Extension), or Logic Audio Platinum 3.5 or higher is needed.

Logic Audio Gold is not compatible with ProTools 24 or ProTools Mix hardware. ProTools III-Hardware, however, can be run with Logic Audio Gold (without TDM capacity- like a Session 8 System).

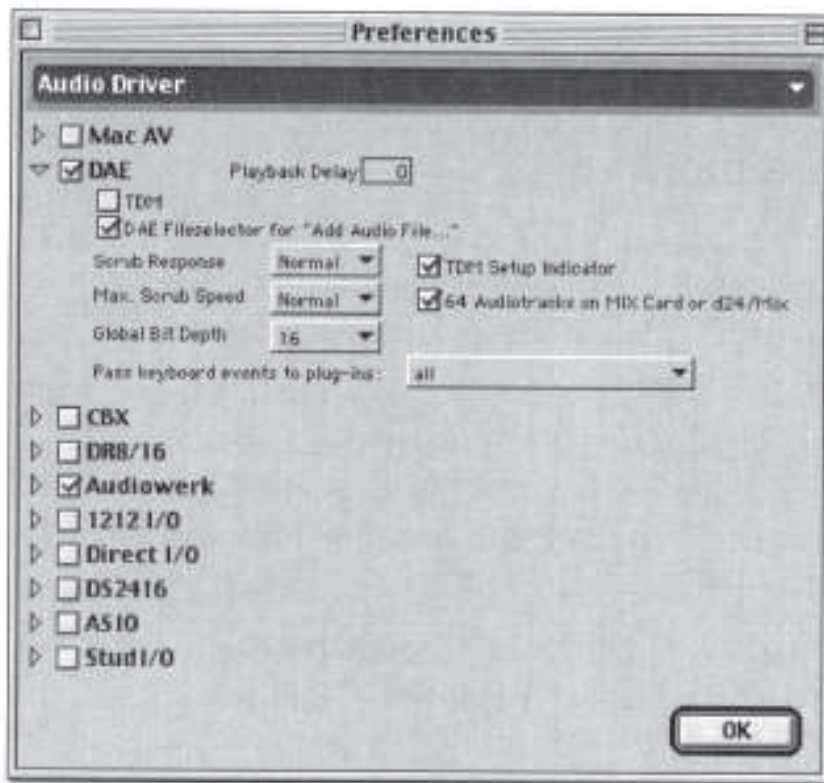
We recommend that you ask your Digidesign dealer to install the ProTools hardware, and to configure your computer.

Turning Your Extensions On and Off

If, for any reason, you should wish to run Logic without its TDM functions, you can switch off the TDM driver at any time.

This facility can be particularly useful if you are booting up Logic with only a small amount of RAM allocated for its use, if you are using a slow computer or if the necessary hardware is temporarily unavailable (in another studio, being repaired, etc.). If you switch off the driver you will also prevent any associated error messages from appearing when you boot up. These always appear if an activated driver cannot find its associated hardware anywhere in your system.

Select **Audio > Audio Hardware & Drivers.**



Click on the check box by the word "TDM" (see above).

The "x" will disappear, and you can now use your ProTools III hardware like a Session 8 system.

You may disable the entire DAE driver, using the same steps, except that you should click on the check box to the left of the word DAE in the line above. After disabling all audio drivers, you can use Logic exclusively as a MIDI program. This is particularly useful if the computer has no audio hardware, or if the existing audio hardware will be controlled by other software during sequencer operation.

Switching the Driver Back On

To turn the TDM or DAE drivers back on, just click again on the same checkboxes. However, before you can make use of TDM or DAE again, you will have to restart Logic.

Object Parameters of TDM Audio Objects

For general information on the audio object parameters, see the section *The Object Parameter Box* on page 6-104.

Cha

The Channel parameters on TDM audio objects have a few extra settings:

Bus

When this is selected, the audio object becomes a bus fader, for example when in use as part of a subgroup. You could for instance, route several track objects' outputs to buses 7/8, and then set the **Bus-7/8** object to use the physical outputs 1/2. You can then use the **Bus** object to adjust the level of the subgroup (7/8), which will be sent to the master output (1/2).

Aux

The **Aux** object is a variable (input and output) object. If you use one of these objects to handle the signal from Input 1, this will have the same effect as using an **Input/object**.

The advantages of an Aux object are that you can:

- assign various insert effects to several Aux objects, and apply them to Input 1 (whereas an Input-1 object could be used only once),
- change the signal source for the Aux object at any time (all other objects have a fixed signal source; e.g., Track, Input or Bus).

Initially, only Aux 1 and Aux 2 are available in the Selection Menu. Once you have used all of the available Aux objects, Logic Audio will make more available as needed.

You will probably use an **Aux** object primarily to set a **Bus** as an Input, on which Aux Sends of various audio tracks will be transmitted.

You can also use the **Aux Object** with a bus input as Aux Master Send for an external effects device. Select an output of the audio interface to be used as Aux object output. Then connect the input of the external effects device with this output. Now, you can direct the output of the external effects device back to the virtual mixer.

Editing Plug-ins

Plug-in Key Commands

With Logic 3.0 and later versions, you can control Plug-ins with their own key commands. For example, if you activated an area of the TDM Plug-in with the mouse, you can enter numeric values in that section. To avoid accidental entries, you can click on the background of the Plug-in window, or press the  key, after you have changed a parameter.

Any key commands that are not used by the Plug-in, will be passed on to Logic. Under normal conditions, this ensures that functions like Start and Stop will continue to work. If this is not the case, you may use several new options to define that only certain key combinations will be passed to the Plug-in, while all other key presses on the computer keyboard will be received by Logic. These options are located in **Audio > Audio Hardware & Drivers**, in the DAE section (“Pass keyboard events...”).

Plug-in Setups (Settings, Effect Programs)

All TDM Plug-in settings are saved with the song, and will be automatically restored when the song is loaded.

You can also use Plug-in Setups as a practical means to manage the parameters of a Plug-in. This method is comparable to the storage locations of an external effects device, combined with a clipboard for effects parameters.

In the header line of the Plug-in window, you can open the Setups pull-down menu by clicking on the arrow button:



- Copy Setup* copies all parameters onto its own independent Plug-in clipboard. They remain here, until the next time you select *Copy Setup*. This does not affect the global Logic clipboard.
- Paste Setup* If you have opened a plug-in of the same type, you can paste the parameter set from the plug-in clipboard. This allows you to quickly set up several plug-ins of the same type, or exchange effects settings between songs.
- Save Setup* This allows you to save all the plug-in parameters to disk. This is useful if you have created a special sound effect, which you want to have available for future use.
- Load Setup* For loading stored parameter sets from disk. The file selector box shows only setups for compatible plug-in types. It can also read the Digidesign format for TDM plug-in settings.

The parameter setups of a plug-in can also be copied between the plug-in's mono and stereo versions, and the Digidesign format for plug-in Settings can be read (the supplied effects programs of most plug-ins are saved in this format). When loading Setups, through the arrow menu in the plug-in windows, Logic automatically displays the corresponding files in the file selection box. Settings files are usually located in a "Settings" folder, within the DAE folder. The name of the last preset chosen will be indicated by a dot in the menu listing.

DAE Plug-in Folders

Starting with DAE 3.1, the DAE folder contains only one "Plug-ins" folder which can store both AudioSuite and TDM Plug-ins.

Disabled Plug-ins

If you haven't done so, create a folder with the name "Plug-ins (unused)" in your DAE folder (or in any other location). Then move all TDM Plug-ins not used on a daily basis, into this folder. This will speed up the DAE (and Logic) boot process.

Support of TDM Plug-in Sidechains

Some TDM Plug-ins, such as Noise Gates or compressors, allow Sidechain Signals from other Tracks or Busses. This option is available in Logic Audio.

Automation

For detailed information on this subject, read the chapter Chapter 6: *Mixers, Effects and Audio Objects*.

As with other audio objects, TDM objects can be automated, although only adjustments made to the controls on the first four plug-ins can be recorded. Up to 16 parameters may be remote-controlled on each of these four plug-ins.

A fixed hierarchy has been adopted for the controller numbers used to automate plug-in parameters, based on the position of the plug-in. Plug-ins are counted from the top downwards.

1.Plug In	Controller # 64 to # 79
2.Plug In	Controller # 80 to # 95
3. Plug In	Controller # 96 to #111
4.Plug In	Controller #112 to #127

Dynamic Controller Assignment

To automate the parameters of plug-ins with more than 16 parameters (e.g., D-Verb, DPP1 or JVP), Logic Audio manages the number of required controllers dynamically.

The "base addresses" (the first controller numbers: 64/80/96/112) remain unchanged. However, if you have activated only

one plug-in, in the first position (first slot), you can automate 64 parameters in this location: 64-127.

If you insert an additional plug-in on the second slot, controllers 64-79 (16 parameters) will be available for the first plug-in, and

80-127 (48 parameters) will be available for the second plug-in.

If you use one plug-in in the first slot and one plug-in in the third slot, 32 controllers (64-95 and 96-127) will be available for each of the two plug-ins.

To find out which controller number controls which parameter for a given plug-in, open the Event List and click on a controller to select it. A list of all parameters that can be automated will appear.

In the Event List, the parameter names of all known plug-ins are shown in plain text.

Caution: Not all TDM Plug-ins (in particular older plug-ins) allow for problem-free real-time automation of parameters. In such cases, the basic settings still can be saved together with the song, however, the dynamic automation of the parameters during song playback is not possible.

Automation Process

The automation of plug-in parameters works the same as the regular automation described in the manual.

Just try it yourself. Select a track assigned to the *A-playback* instrument, and start recording. Open a plug-in, on an audio object insert, and move an operating control of the plug-in. Your moves will be recorded as MIDI commands, which can be recalled and edited.

This procedure can be used to detect which controller numbers can be automated graphically with Hyperdraw Using the Auto Define Hyper-

draw function (which also is available as a key command), Hyperdraw will set itself automatically for the first used controller.

Some plug-ins, like the Digidesign Plug-in “Dynamics”, respond to the controllers they receive “from the outside”, but they will not automatically update the operating control display. Therefore, Logic Audio will redraw the complete plug-in after a certain period.

Plug-ins in objects like “Input”, “Aux”, “Output”, or “Bus” can be automated as follows: Create a channel splitter in the environment **New > Channel Splitter**), and use cables to connect its channels with the respective audio objects. Insert the channel splitter into the Arrange window just like a track instrument (the same way you inserted the *A-playback* instrument earlier). Name the channel splitter something like “*Bus Playback*”, so that you can distinguish it from the existing *A-playback* instrument.

Functions, Special Features

Deleting Objects

If you delete an audio object in the environment, by selecting it and pressing , the level of this object will be set to “Zero” (minus infinite dB). However, “Output” or “Bus” objects are an exception: When deleting these, the level will be set to 90 (0 db); otherwise, the tracks routed to this output would not be audible any longer. The other settings (send paths, plug-ins, and sound control) are not deleted but merely muted. This way, Logic Audio automatically ensures that DSP processor capacity will not be wasted. As soon as the object is displayed again in the environment, the last selected level will automatically be restored. In the TDM System, the Send, and Plug-in will be restored.

Special Features of ProTools III with 32 or 48 Tracks

Logic Audio supports 32 or 48 track ProTools III systems, with more than one Disk I/O card, and 64 tracks with the ProTools Mix system.. Please remember the following, when operating a ProTools III system with more than 16 tracks:

Every Disk I/O card has its own SCSI bus. This means that tracks 1-16 can only play back audio files from hard drives which are connected to card 1, tracks 17-23 only from disks connected to card 2 etc. The same applies to recording. The recording path of every Disk I/O card is managed separately in the Audio window through the dialog box **Audio File > Set Record Path...**

Logic will notify you in the Audio or Sample Edit window if you accidentally try to playback an audio file on a track from another card. Such “erroneously assigned” audio files will not play in the Arrange window.

Since MIDI cannot differentiate more than 16 channels, you will need additional Channel Splitter objects (“A-Playback” instruments) for the automation of more than 16 tracks on any TDM system. If you want to create a completely new song (instead of opening a copy of your Autoload Song), e.g., with alt, Logic will automatically create the following Environment Setup:

Tracks 1-16 are connected with channels 1-16 of the Channel Splitter *A-Playback*,

Tracks 17-32 are connected with channels 1-16 of the Channel Splitter *B-Playback*

Tracks 33-48 are connected with channels 1-16 of the Channel Splitter *C-Playback*

(Initially “A-Playback” stood for “Audio-Playback”. If you don’t like the new name, you can easily rename the channel splitter objects.)

If you operate more than one audio interface (through additional Bridge I/O cards), the additional interfaces are called "b", "c", etc.; for example, "Out 882b 1-2".

Special Features of ProTools 24

The most significant new features of ProTools 24 are:

Under Logic you now can also record and playback with 24 bit word length.

Provides up to 32 audio tracks. For system requirements, please see the Digidesign documentation.

A core system may have 24 inputs and outputs (versus 16 with ProTools III PCI, or 8 with ProTools III NuBus).

ProTools 24 does not have its own SCSI bus. It uses the computer's SCSI bus. Therefore, you should read QT movies and audio data not only from separate hard disks, but also from hard disks connected to different SCSI busses.

For the operation of ProTools 24 with TDM, Logic Audio Platinum is required. Unlike ProTools III, it cannot be run as if it were a Project or Session8 system, with Logic Audio Gold. However, you can run a ProTools 24 system in Logic Audio Gold, using the Direct I/O driver. This gives you no access to TDM functionality, so if you have such a system, it is highly recommended that you upgrade to Logic Audio Platinum.

TDM Mixer Plug-in

Please be sure that there is only one Mixer Plug-in in the "Plug-ins" folder (within the DAE folder), which will be either the 16-bit version or the 24-bit version.

24-bit files can also be played back through the "16-bit optimized mixer" (or vice versa). However, only the "24-bit optimized mixer" can adequately handle the resolution of recordings in 24-bit format.

The 24-bit mixer requires more DSP capacity than the 16-bit mixer. Therefore, you may get a DAE error message ("DSPs maxed out"), when opening older songs with maximum DSP load, after the upgrade to ProTools24. In this case, you should use the 16-bit mixer plug-in.

Changing Global Bit Depth

If ProTools 24 hardware is being used, Logic will ask at the initial program start, whether you want to use 16-bit or 24-bit word length. Later, you may change the "Global Bit Depth" in the dialog box **Audio > Audio Hardware & Drivers** from 16-bit to 24-bit, and vice versa.



In most cases, a resolution of 16-bit is adequate for the individual tracks of a regular multi-track production. This setting requires less space on the storage medium, and also less data throughput capacity on the hard disk, to play the same number of tracks. The 24-bit format is mainly advantageous with highly dynamic material (e.g. classical music productions),

Program Start

If you are loading a song with different word length audio files, Logic will display a warning. You can change the word length (see above), or you may convert the audio files.

Please note that the conversion of 16-bit recordings to 24-bit recordings does not bring any sound quality improvement, while a conversion from 24-bit to 16-bit results in the irrevocable loss of the added dynamics possible with 24-bit files.

Converting the Word Length

In the Audio window, select all of the files to be converted.

For the selection, it would be useful to sort the audio files by their word length (bit depth) (see above). Then select all files to be used in the arrangement with [Edit > Select used].

Select **Audio File > Copy File(s)**.

In the following dialog box, you have the option to set the word length (bit depth).

After confirming the dialog, the audio files will be copied-the copied files will have the selected word length.

Logic now will ask you if the references in the current song should be changed to reflect the copied files, instead of the original files.

If you want to work immediately with the current song, you should confirm this dialog.

Save the song with a different name by selecting **File > Save as...** if you want to keep the references to the original files in the original song.

Sample Editor

All functions of the Sample Editor can be used with 24-bit audio files. You can even exchange sound material in either direction between 16-bit files, and 24-bit files by using the Copy and Paste commands.

Please note that Premiere Plug-ins use only 16-bit resolution (you can still edit 24-bit files). AudioSuite Plug-ins support 24 bits.

Simultaneous Operation of Different Hardware

Please note the following, if you want to use a TDM System and other audio hardware at the same time.

Generate Audio Mixer

New Song

With the selection of the command **File > New** (or ) , a copy of your Autoload Song will be opened. If you want to open a completely new song with standard defaults (Default Song), just press  .

Corresponding Audio Objects

Based on the connected hardware, and the installed extensions, this song will include audio objects that correspond to the currently active audio hardware, in the Audio layer of the Environment. It also includes the audio playback instrument, which you will set as the current track instrument for the recording of fader movements.

You may select the complete layer ( ) , copy it to the Clipboard ( ) , and then insert it into a new layer in your Autoload Song ( ). Or, you can drag the required objects to the environment of your Autoload Song.

If you want to integrate the new “TDM Mixer” into your existing Audio layer, make sure that the position of the audio objects will be maintained during the copy process. To avoid new objects being placed on top of the existing ones, select the existing objects prior to inserting the new ones and drag them to the side or to the bottom (“grab” the object at its right edge, at the object name or with the •J key). Don’t forget to deselect all objects by clicking on the background; otherwise all selected objects will be replaced with the inserted objects.

Finally, don’t forget to save your new Autoload Song ( ).

Control Playback

To control playback in the Audio window or in the Sample Edit window, make sure to select the correct corresponding hardware and channel, under Channel and Device (at the left side of the window).

For example, it is not possible to play back an audio file located on a hard disk on the SCSI bus of a ProTools III System through the internal Mac hardware, since only it can access the SCSI bus of the I/O card.

In order to play such a file, the audio file would have to be physically copied or moved to a disk on the system SCSI bus (using the Copy or Move command).

Total Number of Audio Tracks

When operating different audio hardware simultaneously, the number of possible tracks cannot be determined by merely adding them up. This is especially true if the connected systems have to use the CPU processor's power. This is the case primarily in AV mode, but also when using DAE.

The simultaneous operation of DAE hardware and AV audio tracks will result in a significant reduction in the number of usable AV tracks. However, stand-alone systems, like the Akai DR-8/16, cause only a minor load increase on the CPU.

The arithmetically added number of tracks may also be reduced, if the connected systems access the same SCSI bus. This is the case, for example, when using AV tracks and CBX hardware at the same time.

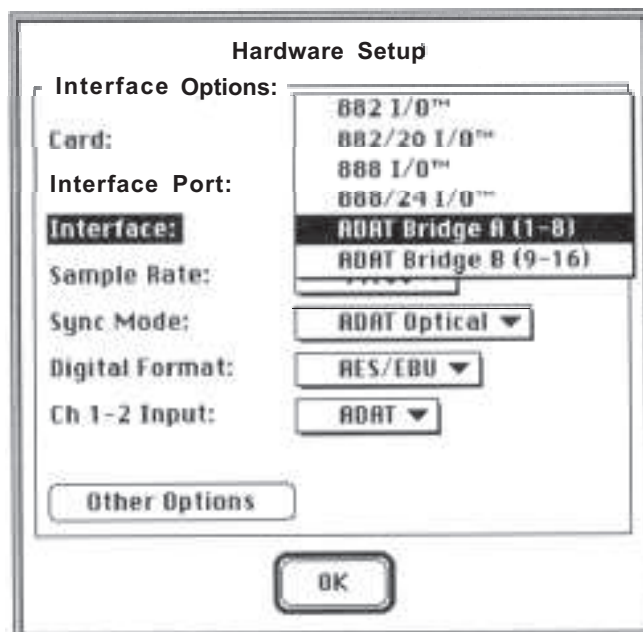
This effect may be significantly reduced (i.e., the total number of tracks increased) by using different hard drives for the various systems (e.g., AV reads from an internal drive, and CBX from an external drive).

On the other hand, the simultaneous operation of ProTools III and CBX (=4 tracks or 8 tracks with two CBXs) presents no problem at all.

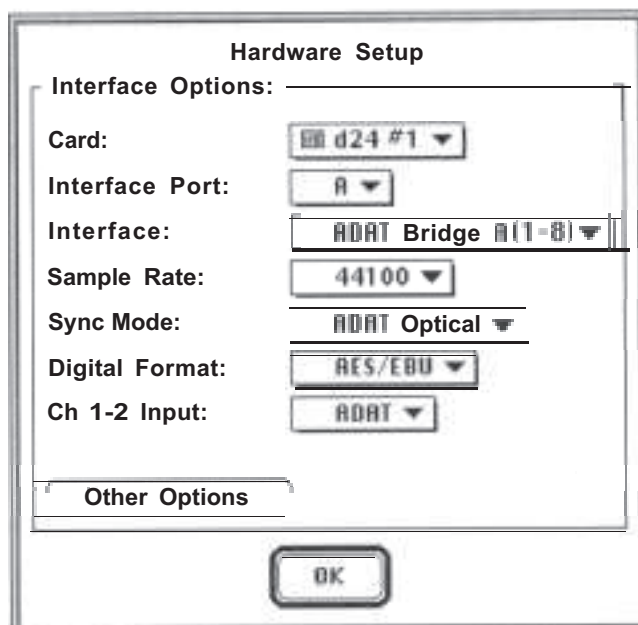
We recommend, however, that you avoid the use of an internal hard disk for hard disk recording, if possible.

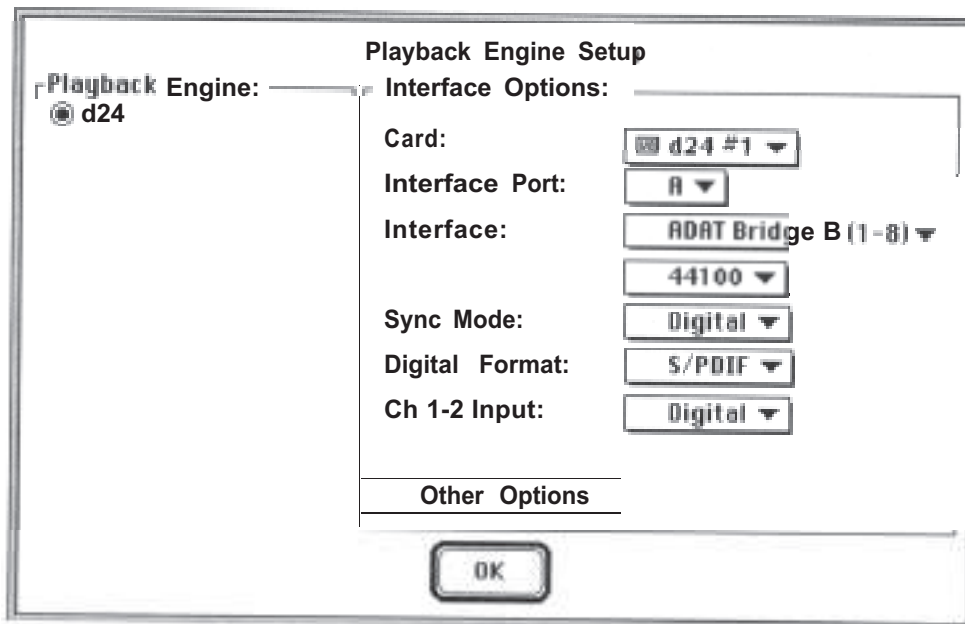
Digidesign Adat Bridge Interface

If you want to use the Digidesign Adat Bridge Interface, you need DAE Version 3.2.3 (or higher). The Adat Bridge Interface is a 19" unit with two optical Adat inputs and outputs providing 16 channels, where channels 1 and 2 also act as AES/EBU and S/PDIF ports.



Each of the two groups of up to eight channels can be individually connected to a Disk I/O or a PCI DSP Farm. The PT24 Disk I/O has two interface sockets. You can select the bridge interface in the **Hardware Setup** menu.





8.8 Direct I/O

Until now, Digidesign hardware has always been addressed via DAE (Digidesign Audio Engine). Digidesign has now released hardware like “Project II”, that is not supported by DAE. The new Audio Engine has been developed to use these cards with Logic Audio. As the Direct I/O interface supports other Digidesign hardware as well, the driver is called “Direct I/O” instead of “Project II”.

Please note that the system extension Digisystem INIT 3.4 or higher is required. This system extension is shipped with the card, but is also available from the Digidesign distributor in your country, or from the Digidesign web site www.digidesign.com).

By the way: users of other Digidesign PCI cards, (e.g. Audio-media III or ProTools Project) may also benefit from Direct I/O. You achieve a higher number of tracks (if your CPU is fast enough), and gain access to Logic Audio’s internal effects and VST Plug-ins. Even if you are using a ProTools III Core system, you may use Direct I/O, at least if you have a powerful CPU. However, please note that no TDM functions will be

available. Furthermore, you should only play back audio files from the hard drive attached to SCSI bus of your Power Macintosh, and NOT from any drive attached to the SCSI bus of the Digidesign card when using Direct I/O. Newer and larger ProTools 24 systems can be also be run via Direct I/O, instead of DAE. However, as you then lose all TDM functions, it does not really make sense.

In order to activate the audio driver for Digidesign Project II, or other hardware, go to the main menu and select **Audio > Audio Hardware & Drivers > Direct I/O**. Here are descriptions of the parameters you will find in the “Direct I/O” section for the new Audio Engine:

I/O Buffer Size

This parameter, which affects Direct I/O, governs the latency and the performance of the card. A value of "1" is an ideal setting for good performance and reasonable fader response. Smaller values reduce the monitoring delay and the fader response time, but also reduce the number of audio tracks and the performance of plug-ins.

Use 16 Ins & Outs

The Project II card allows you to use two audio interfaces, with 8 inputs and outputs each. If you have only one interface, please switch this setting off. This saves memory, and optimizes overall performance. You should only switch this setting on when actually using two interfaces.

Digidesign Hardware Setup

In the audio window, with **Options > Digidesign Hardware Setup**, you can open a dialog that allows you to select the audio interface, and to switch between analog and digital input. The equivalent dialog box for setting up DAE is only available when actually using DAE to run your Digidesign hardware. The Direct I/O driver cannot be used simultaneously with DAE.



8.9 Sonorus StudI/O

In comparison with the ASIO driver, this direct support of the StudI/O offers the following advantages:

Better performance

More precise start when recording and playing back.

Operating Modes

The StudI/O can be operated in 5 different modes. All modes allow 24 bit:

2x ADAT

16 In/Output channels always available in ADAT format.

ADAT + SPDIF

8 I/O channels in ADAT format, and two channels of S/PDIF I/O.

ADAT + BNC

8 I/O channels in ADAT format, and ADAT-sync

2x SPDIF

Two twin channel S/PDIF I/O.

96K SMUX

8 I/O channels over ADAT with 88.2 or 96 kHz format (Platinum only).



With regards the other parameters that appear here, please refer to the manual of the StudI/O

8.10 ASIO

ASIO (Audio Stream In/Out) is a common driver format introduced by Steinberg to allow audio hardware and audio software

ASIO

(in this case Logic Audio Platinum) to work together. An ASIO driver translates the input data of the hardware into a specific format, and transfers the data to the software (Logic Audio Platinum), which receives it as recording data. In turn, the software (Logic Audio Platinum) transmits the final mixed signals to the ASIO driver, which then transfers it to the hardware.

The ASIO driver is generally provided by the manufacturer of the audio hardware.

The advantage of this sort of industry standard interface is:

New Audio hardware is compatible with Logic Audio right away, as long as it is equipped with an ASIO driver.

The disadvantage of this sort of industry standard interface is:

To software each driver (and the hardware supported with it) looks identical. There is no flexibility to support specific features of the hardware. Some manufacturers ship their own control panels, the settings of which are “invisible” to Logic Audio.

Just like Direct I/O, ASIO support is handled by the Audio Engine as described in the section Audio *Engine* on page 8-5. The settings for the audio engine and for the ASIO driver can be found in the **Audio > Audio Hardware > Drivers > ASIO** menu.

Current Driver

This is where you select the ASIO driver you would like to use. When changing the Audio driver, it is necessary to quit Logic and open it again.

Logic expects to find the ASIO driver in a folder named “ASIO Drivers”, in the directory of the Logic program file.

! This also can be an alias to an ASIO folder somewhere else

ControlPanel

This switch allows you open the control panel of the active ASIO driver, if one is available. Whether or not this control

panel exists depends on the manufacturer of the hardware being supported.

Clock Source

This setting allows you to use a digital clock source from an external device, if the hardware and the driver you are using supports this.

ASIO Buffer Delay

Some ASIO drivers give incorrect information regarding input and output delay. The difference from the actual value may vary between different versions of the driver, therefore it is necessary to allow changes of this setting. Please check our recommendations for specific hardware systems in the ReadMe-File on the CD-ROM.

Max. I/O Streams

This parameter is used to define a maximum of inputs and outputs. This is needed especially for the DS2416 card, which does not run with its 8 ins & 16 outs on machines slower than a G3/333. The default is set to 4 ins and 8 outs for the DS2416. For others it is set to *Max*, which means that you can use all ins and outs.

It's recommended that you switch off unused I/O channels in the driver (control panel), if you are using an ASIO driver that supplies one.

System Memory Requirement

Changing some of the above settings increases or decreases the required memory. The required memory is displayed on the right side of the "ASIO" switch.

The Key Command *Open System Performance...* will open a window that displays the CPU and Hard Disk Bus usage as a percentage (with 100% being the maximum capacity).

8.11 Yamaha DSP Factory

The currently supported components of the Yamaha DSP Factory System include one DS2416 card and, optionally, up to two AX44 extension bays (4 analog ins and outs each) or an AX16T (2 x 8 ADAT Ins and Outs). Furthermore, Logic Audio Platinum is also ready to support up to 8 channels of one SW 1000XG soundcard and future I/O extensions whose specifications are yet unknown. In order to use the card, select **Audio** > **Audio Hardware & Drivers** and check DS 2416.

Features

(If you use one DS2416 only)

Up to 16 audio tracks

24 channel digital mixing disk, including

- up to 4 channels for two stereo effect returns of the two internal multi-effect processors

- up to 4 Input channels. Further inputs are only available with the AX44 hardware extensions and by sacrificing audio tracks (there is a total limit of 24 channels).

Variable output routing for the stereo sum and the 4 Aux sends. In addition, the two internal effects sends can be used in parallel. This results in a maximum of 6 individual outputs.

Each of the 24 channels offers:

- 4 Equalizer Plug-ins (fully parametric, shelving)

- Dynamics Plug-in (Compressor, Limiter, Expander, Gate)

- Channel Delay

In addition, the two effect return channels each offer:

- One Multi-Effect Plug-in (choose from a list of 40 algorithms)

Mixer

The mixer directly reflects the available DSP and routing of the DSP Factory, as long as you use the ASIO Driver DS 2416 as the only driver. It is defined by the feature set of the hardware, and therefore is not completely variable. Please note that Logic does not utilize its own audio driver, but uses Yamaha's ASIO driver, and the real-time effects of the digital mixer integrated in the DSP Factory.

The default configuration of the mixer offers 16 audio tracks, 4 inputs (analog and digital input), two stereo effect returns and two stereo output objects (master fader) for the stereo analog and digital output of the DS 2416 card. The mixer offers 24 channels. Additional inputs always reduce the number of audio tracks.

Fader

The faders have a different range than the faders which control other hardware: 0 dB equals the volume controller value 127. The value 90 equals -6dB. Unlike other audio drivers, the fader for the DSP Factory can only decrease the volume, but not increase it.

Inserts

Each mixer channel holds up to 4 EQs, one dynamic module, one channel delay and one phase inverter.

EQs, dynamics modules, channel delay, and the phase inverter are used as inserts and can be controlled via plug-in windows. The actual signal path is not variable and goes like this: EQs -> Dynamics -> Delay.

The plug-ins of the audio tracks are used-as is common-on playback. However, please note that effects on the input channels (i.e. channel 17 to 20 in the standard configuration) are also recorded. For unaffected recording you might want to set the inserts of the input channels to bypass. (-Click). On the

other hand, it might be useful to directly record EQs and dynamics.

On the effect return channel, the internal multi-effects devices always use the first insert slot, but EQs and dynamics can be used in the following ones.

Sends

Track channels (audio tracks) and input channels have up to 8 sends. The sends labeled "Effect 1" and "Effect 2" address the two internal effect busses.

In addition, the sends of the available outputs of the installed hardware can also be used; e.g. output 1-4, if one DS2416 card is installed (using an AX44/AX16 extends the choice to outputs 1-6). Please also refer to the following section on input/output routing.

As is common with Logic's fader, the effect sends are processed after the fader (post-fader), but can be switched to pre-fader using the flip menu of the active sends.

Input/Output Routing

Located above the pan fader is the output routing for all 24 channels. This allows you to direct the output signal of a track to one of the outputs. The track channels ("Audio 1-16") also feature input routing; i.e. the ability to select from which input you would like to record.

The number of available inputs and outputs depends on whether you have only a DS2416 card installed, or whether you are also using the AX44 I/O box (or AX 16):

DS2416 4 inputs and 4 outputs

DS2416 + AX44: 8 inputs and 6 outputs (the two additional outputs can only be used parallel to the internal effect busses).

DS2416 + AX16: 8 Inputs and 6 Outputs (2 more Outputs can be used in parallel with the internal effect busses; Recording busses are restricted to a maximum of 8 Inputs.

A second AX44 may be used but does not increase the number of outputs you can address simultaneously (max. 8)

Default:

Input 1 = Channel 17 = Analog In L

Input 2 = Channel 18 = Analog In R

Input 3 = Channel 19 = Digital In L

Input 4 = Channel 20 = Digital In R

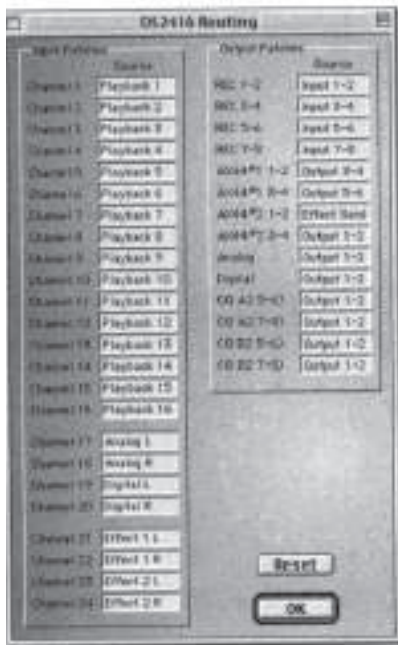
Both the Analog Out and the Digital Out are addressed by "Output 1-2".

The first AX44 (if installed) is addressed by "Output 3-4" and "Output 5-6".

The second AX44 (installed) mirrors effect sends 1 and 2 on its first output pair.

These default settings can be edited in the "DS2416 Routing" window.

DS2416 Routing



You can open this window with a key command or from the audio window **Options > DS2416 Routing**. The left half is a sort of digital patchbay for selecting the inputs of the mixer channels 1-24. The right half is basically a patchbay for the outputs of the hardware.

The input patches can be configured with certain limitations; e.g. the 4 inputs of the first AX44 can only be inserted on channels 17-20 (thereby sacrificing the inputs of the DS2416) or on channels 13-16 (thereby sacrificing 4 audio tracks).

The inputs of the second AX44 can only be inserted on channels 9 to 12 (losing another 4 tracks). The 8 outputs of the SW1000XG soundcard ("Sub 1-8") can only be inserted via channels 9-16 or 17-24.



Please note that this window already displays some options which are not yet available as hardware, e.g. the hardware for the options "IO-A2" and "IO-B2". Logic Audio should run with this hardware, although we were not yet able to test it as of the release of Version 4.0.

Clock Source

The parameter *Clock Source* under **Audio > Audio Hardware > Drivers > ASIO** allows you to switch between the use of internal synchronisation or an external digital synchronisation signal. The setting *serial in* is needed in case you use a DS 2416 together with a SW1000XG.

If you select *Digital* Logic Audio will continuously check the sample rate of the incoming S/P DIF signal. If it detects a difference from the currently set sample rate (**Sample Rate** in the **Audio** menu) a warning message will appear. If you confirm

this message with USE, the internal sample rate will be set to the detected rate.

Chapter 9

The Sample Edit Window

The Sample Edit window offers an enormous number of data-editing functions, which are used to process individual audio files. You can edit mono, as well as stereo files.

You can set the lengths of audio regions with extreme precision (down to single-sample resolution), by making use of the window's adjustable zoom resolution. The anchor points can also be positioned here, with the same degree of accuracy.

Opening the Sample Edit Window

There are various ways of opening a Sample Edit window:

Select **Audio > Sample Editor**.

Press  .

Double-click on any audio region in the Arrange window; this opens that region in the Sample Editor.

Double-click on any region in the Audio window; this takes that region into the Sample Editor.

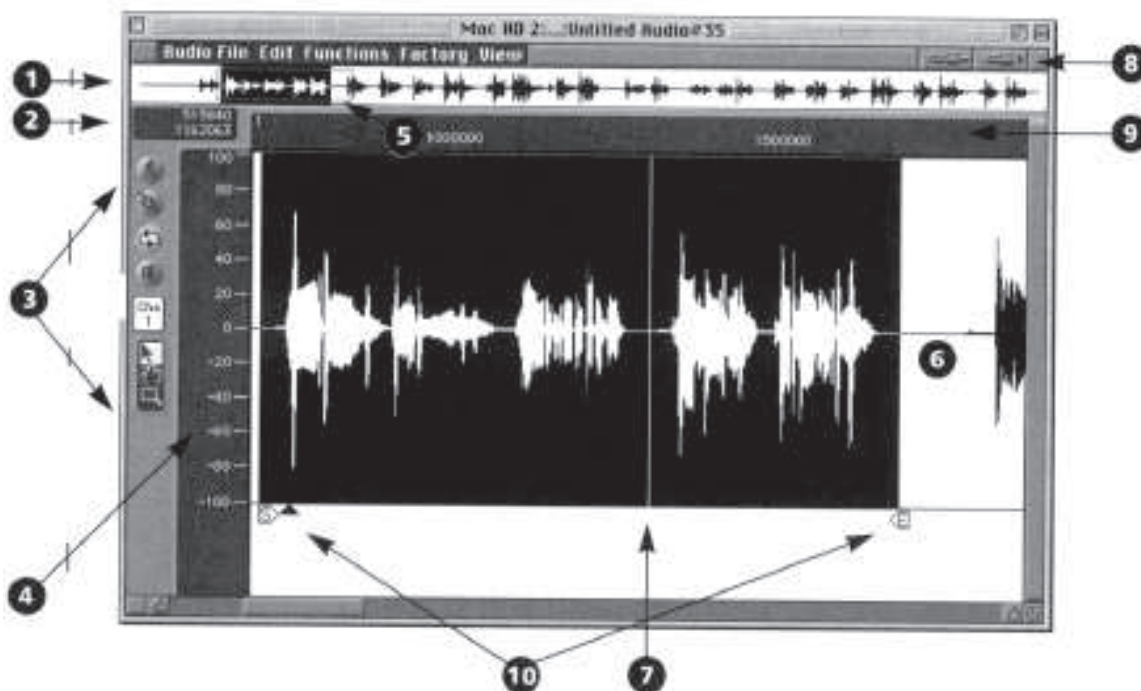
If no region is selected, Logic will ask if you'd like to load a new file into the Sample Edit window.

If you're dealing with an interleaved, or half of a "split" stereo region or audio sequence, both channels/audio files will be displayed in the Sample Edit window, with the left side on top, and the right side below.

If you open the Sample Edit window from the Arrange window, as opposed to the Audio window, the bar ruler is able to reference the region's position in the song. You can tell all of this

from the position marker lines in the Bar Ruler. A dotted line indicates no time connection (Audio window), while a broken one indicates a time connection for the region, which is being used as an audio region in the Arrange window.

9.1 Layout



- ❶ This is the overview; the entire audio file is depicted here.
- ❷ This is the parameter box for the current selection, and shows the start point and length of the selected area.
- ❸ When the Catch (“walking man” symbol) button is activated, it insures that the playback position is always visible in the window. The other controls operate as in the Audio window.
- ❹ Amplitude Scale (readings as percentage, or 16-bit decimal values).
- ❺ The dotted frame shows the extent of the section visible in the display area.
- ❻ Detailed waveform display.
- ❼ Playback position line (also visible in the overview).

- 8 The zoom telescopes are used to set the magnification used on the detailed waveform display.
- 9 The horizontal time ruler displays the name of the edited region at the top left. Beneath that is the time position in the audio file, in various formats.
- 10 The Start point, Anchor and End point of the currently selected region can be changed by simply grabbing, and dragging them.

9.2 Display

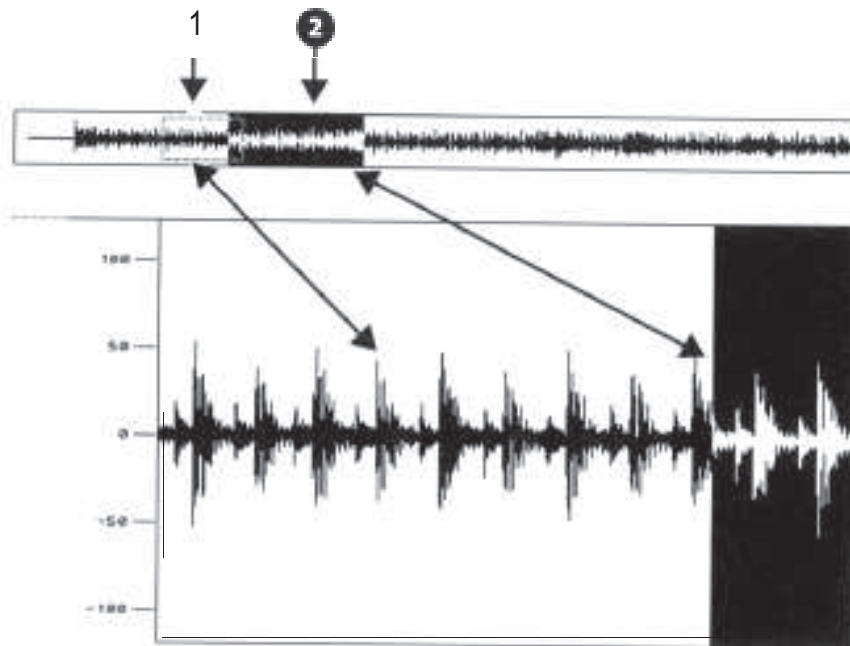
Overview

Between the Sample Edit window's title strip and Bar Ruler is the "Overview". This display always shows the full length of the currently selected audio file, regardless of the zoom resolution set by the telescope symbols. Please note that no idea of scale is given in the overview; a kick drum sample lasting 0.3 seconds could take up the same space here as a choral passage lasting 1.5 minutes.

During playback, the current position is indicated by a vertical line, moving in real time. This is visible in both the overview and the detailed waveform display.

The current selection is also displayed in the overview.

The section visible in the detailed waveform display is shown in the overview as a dotted rectangle.



The overview: the dotted box ❶ shows the section of window currently displayed in the detailed waveform display. Part of the selection ❷ is also visible.

Functions in the Overview

A short mouse click on the overview brings the area clicked on into the detailed waveform display.

A long click with the mouse resumes playback from this position. Releasing the mouse button halts playback once more.

Double-clicking lets you listen to the sample from the position clicked.

Window Functions

The Catch and Link functions work in pretty much the same way as in the other edit windows.

Catch Mode

Catch mode insures that the Sample Edit window always displays the area around the current playback position (whether



you are playing the song, or monitoring the sample). *Catch* is turned on and off either by clicking the switch with the “walking man” symbol on it, or with the key command.

Link Mode

Link mode insures that any audio sequences selected in the Arrange window are displayed in the Sample Edit window. *Link* mode is turned off and on by clicking the switch marked with the “linked chain” symbol, or with the key command.



If you often work in Link mode, give this a try: open a Sample Edit window and switch on Link mode. Now close the window. Logic now leaves Link mode permanently on.

Double-click on the audio region you wish to view. You can see immediately that the display in both windows is linked.

The Detailed Waveform Display

Display Scale

Zoom Telescopes

The zooming with the telescopes works as described in the section *Zooming* on page 1-14 in the sample editor as well.

The Zoom Tool

Just as in the other windows, there is a zoom tool in the Sample Edit window toolbox (the magnifying glass). If you use this tool to draw a rectangle, the selected area of the window will be magnified so that it fills the whole screen. You can also repeat the action. Clicking the mouse once (with the tool) returns you to the previous zoom resolution.

If you hold down  as you click the telescope symbols with the mouse, you can alter the vertical and horizontal zoom simultaneously.

X- and Y-Axes Scales

The Y-axis has a vertical scale showing the waveform amplitude in percentage units (obtained by selecting **View > Amplitude Percentage**). On the other hand, if you select **View > Amplitude Sample Value**, the scale will be displayed in sample units.

The X-axis (the time ruler) shows the course of the audio file over time. You can select various units for this scale by using the **View** menu.

Don't forget that this display format also affects the figures shown in the Info Line, and the selection parameter field:

View > Samples

Displays the sample word number from the beginning of the song or audio file.

View > Min:Sec:Ms

Gives the scale in Hours:Minutes:Seconds:Milliseconds, from the beginning of the song or audio file.

View > SMPTE Time

Gives the scale in SMPTE time (in Hours:Minutes:Seconds:Frames). The time scale begins at the song start, with the SMPTE offset. When you use this scale, the absolute SMPTE value of the source clock is shown on the X-axis.

View > Bars/Beats...

Gives the scale in Bars, Beats, Divisions and Ticks, like the bar ruler in the other time-related windows. The "Zero point" is represented by "1 1 1 1", but lengths are measured from "0 0 0 0".

Absolute and Relative Time

The different axis scales in the Sample Edit window can be displayed on the basis of one of two different reference values:

Display

- | by reference to the time axis of the song (absolute position)
- | by reference to the beginning of the audio file (relative position).

RelativePosition

The units in the time axis (between the Overview and the waveform display) will be displayed with dotted lines when it is displaying the Relative Position. You will be in this mode if you open the Sample Editor from the Audio window, or if the Sample Editor is in Link mode, and you select a region in the Audio window.

The beginning the section is automatically assigned the value zero, or in Bar/Beat terms, "1 1 1 1".

This does not necessarily match the actual song position. The calculation of all remaining musical sections is then done using the current song tempo.

Absolute Position

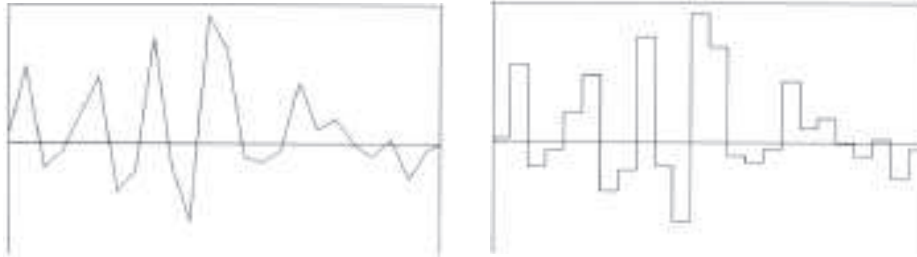
You can recognize this format by the broken line below the units in the time axis. There will be a broken line below the units in the time axis, when it is displaying the Absolute Position. You will be in this mode if you open the Sample Editor from the Arrange window, or if the Sample Editor is in Link mode, and you select a region in the Arrange window.

Time is measured from the start of the Song, which is given the value zero (or 1 1 1 1 in Bar/Beat terms). In this instance, the time axis shows the absolute (song) time, and the figures do not refer to the audio file.

Display Waveform as Sample Bits

At high magnification on the detailed waveform display, you can switch from the usual representation of the waveform, to one that shows the structure of the digital data you have recorded. You do this by selecting **View > Show as Sample &**

Hold. This way of displaying the waveform can be useful, for example, when eliminating clicks and pops from your recordings.



Thanks to the clever use of oversampling techniques and low-pass filters at the D/A conversion stage, the stored sample bits (on the right) are ultimately converted into a signal waveform more like the one shown on the left.

9.3 The Sample Edit Window in Use

Monitoring Sample Playback

There are various ways of playing back the sections of audio visible in the Sample Edit window, making it possible to hear audio as you edit it. Playback occurs independently of the position of the sequencer in a Song. If, on the other hand, you'd like to hear the selected audio passage in the context of the whole song, you'll have to use the Transport controls as usual.

Selecting the Audio Output

The small boxes under the loudspeaker symbol allow you to select the hardware and output channel to be used for monitoring. If you're editing a stereo file on a DAE, or DSP Factory system, the right channel will be sent to the next-highest available output channel.

Set the number of the output you want to use in the *Cha* box under the loudspeaker symbol. Use the *Dev* box just below it to choose between different kinds of hardware, if you have more than one type installed (if you do not, this switch will not appear.).



Logic Audio Silver does not have the device switch, since it supports only one type of audio hardware at a time.



On a DAE, or DSP Factory system, make sure that the pan positions of both channels are set as widely as possible, or that the matching Audio object is set to be a stereo object.

Playback from the Overview

Monitoring playback from the overview display is carried out in exactly the same way as with regions in the Audio window. Simply click-hold the mouse at the point at which you wish to begin playback. Releasing the mouse causes playback to stop. You can also start monitoring from any position by double-clicking there.

Playing the Whole Audio File

The key command *Play/StopAll* allows you to play back a whole file, regardless of the current selection.



Playing the Current Selection

To play back the current selection, click on the loudspeaker switch.



You can also perform this function with the key command *Play/Stop Selection*.



Playback from a Certain Position

If you double-click at any point on the time axis, the audio file will play back from this point, to the end of the current selection. If you double-click on a point beyond the selected area, the audio file will play right to its end.

Checking the Position of the Anchor

With these two key commands, you can check the position of the region anchor, by listening to the part before, or past the anchor. These key commands are: *Play/Stop Region to Anchor* and *Play/Stop Region from Anchor*.



Cycle Playback Mode

On the left-hand side of the Sample Edit window, just above the Loudspeaker button, is the Cycle button. If you turn this on, the currently selected section of audio will cycle continually when Sample Edit window playback is engaged.



*You can change the start and endpoints of the selected area, while monitoring it in cycle playback mode. In this way, you could edit the start and endpoints of (say) a drum loop, until loops perfectly. When you're satisfied, you make the selection into a new region with the function **Edit > Selection > Region**.*

Playing Regions

In general, you can only play currently selected parts of files from within the Sample Edit window. But there is a key command which toggles between playback of the whole region and stop—no matter what the selection is. This key command can be found under the name *Start/Stop Region*.



When you open the Sample Editor by double-clicking an audio region, this region will automatically be selected in the Sample Editor. The same thing happens when you click a region, while the Sample Editor is in Link mode.

Scrubbing

Scrubbing is only possible at present if you have DAE hardware.

Click on the time axis and hold down the mouse button. If you now move the mouse, the recording should play back, following the speed and position of the mouse.

More parameters can be set on the **Audio Hardware & Drivers** page (**Audio > Audio Hardware & Drivers**). For more information, see the chapter Chapter 8 *Audio Driver*).

Automatic Scrolling

You use the scroll bars along the bottom and right edges of the Sample Edit window, to scroll through the detailed waveform display in the usual way. However, if you're trying to reach one of the following points in the audio file, some of the keyboard short-cuts shown below should be of use to you. You can define your own keys for these commands from the Key Commands window (see Chapter 1 for more information), if you wish.

These commands bring the required point to the center of the screen.

Making Selections

Selecting the Whole Audio File

You can select the entire audio file with the function **Edit > Select All** ( .

Manual Selection

To select a particular section of an audio file, click on the start or end of the area you want to select, hold down the mouse button, and move the mouse to the right or left.

To Change the Boundaries of a Selection

By clicking on a selection while holding down the *shift* key, you can change its existing start and end points at any time. Whether you change the start point or end point by doing this,

is determined entirely by whether the point you clicked on was nearer to the start or end of the selection. The closest one wins..

Changing the Further Selection Limit

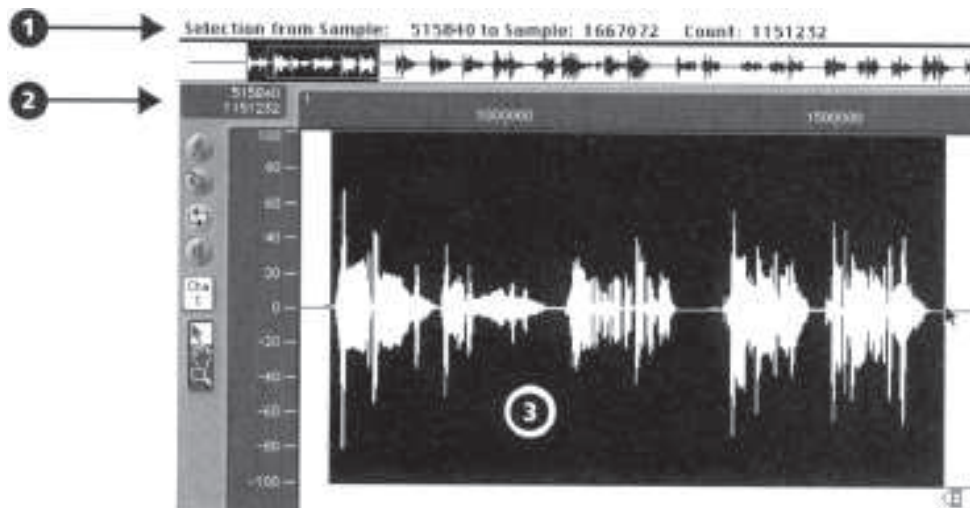
If you hold down *alt+shift*, the more distant selection boundary is changed (rather than the nearer).

Moving a Selection

If you hold down *alt*, you can shift the whole selection, without changing its length.

The Selection Parameter Box

The start point and length of the current selection are shown in the selection parameter box at all times.



The Relationship between Selections and Regions

When you open the Sample Edit window by double-clicking on a region, the Sample Edit waveform display will open with the entire region selected. Changing the selection has no direct effect on the borders of the actual audio region. The playback monitor only plays back the current selection.

Logic offers two functions that govern the interaction between selections and audio regions. These will allow you to create and edit regions with the minimum of fuss.

Selecting the Region

By selecting **Edit > Region -> Selection**, you select the entire region currently in the Sample Edit window. The current region is the one selected in the Audio window (or the region belonging to the audio sequence selected in the Arrange window).



This function is useful if, for example, after editing it in various ways, you want to select the whole region again, so that you can put it into Cycle Playback Mode.

Turning a Selection into a Region

If you wish to make your current selection into an audio region, select **Edit > Selection -> Region**. In this way, you can take an audio passage, define it as a selection, and then convert it to a region, in place of the original region you selected.



Creating New Regions

To define a new region from an area you've just selected, select **Edit > Create New Region**. You can define this function as a key command.



SoundDesigner II users, take note: this command is the equivalent of 

Editing Regions in the Sample Editor

If accuracy is what you need, you should edit the start and end points of regions in the Sample Edit window, not the Audio window.

The same goes for any adjustments you make to the anchor, which in many cases should really be placed on the amplitude peaks, rather than at the start of the attack phase of the sound.

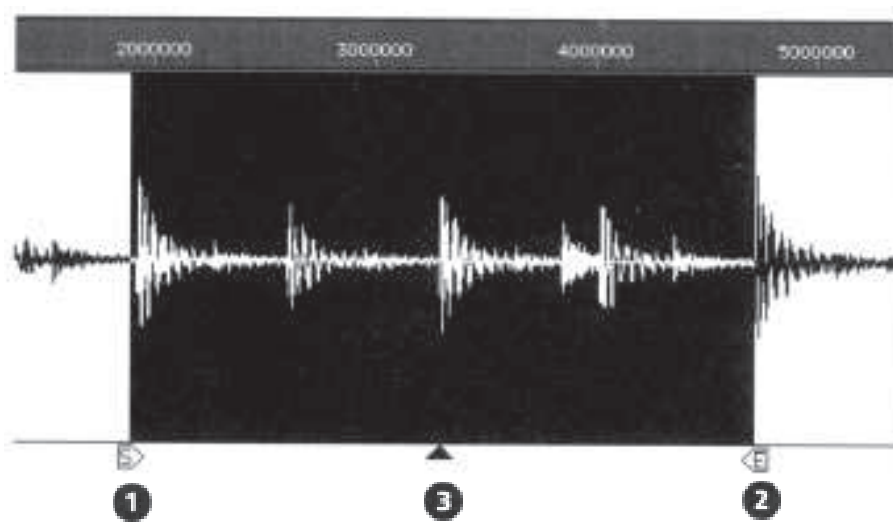
Chapter 9

The Sample Edit Window

A good example would be recordings of brass instruments, which may take some time to build to a peak. By moving the anchor to these peaks, the region will snap to the grid in your arrangement, using the anchor as the pivot point. The flexible zoom settings allow you to be as precise as you like, going right down to the level of single bits, at the highest magnification.

Be Careful! Any changes to the position of the Anchor point will change the relative position of that audio region in the song. Since the region start is the default position for the anchor you must also take care when changing a region's start point.

The small markers on the lower edge of the waveform display allow you direct access to the boundaries of the region and the anchor. As usual, you can just grab them, and pull.



- ❶ Region Start
- ❷ Region End
- ❸ Anchor

Protecting the Anchor point

If you move the start or end points of a region past the anchor point, the anchor point will also move. This is quite often not what you want to happen.

Holding down the  key while you move the start or end markers of a piece of audio prevents the anchor from moving.

Keep in mind though, that the anchor can never be to the left of the start point of its region, so it will move, if you drag the start of the region past the anchor to the right, even with *alt* pressed.

Search Zero Crossings

If this editing option is switched on, Logic will search for the nearest point where the waveform crosses the zero amplitude axis, whenever the start or end points of a selection are changed, to avoid glitches in playback. The program looks before the start point, and beyond the end point.

Drawing Waveforms with the Pencil

Using the pencil tool, clicks and pops or clipping can be corrected, by manually drawing in the waveform. The edges of drawn-in waveform sections will be automatically smoothed.

As long as the mouse button is held, the drawn in waveform can be replaced by the original by moving the mouse back.

When the -key is pressed, the drawn-in changes will be effective on both stereo channels.

Editing Commands

Like the other windows, the Sample Edit window features the usual edit commands **Cut**, **Copy**, **Paste**, **Clear**, and **Undo** under its Edit menu (or from the keyboard).

*In the Sample Edit window all these commands (except **Copy**) change the data on the audio files itself, in other words, they behave destructively. Consequently, they cannot be reversed using the "Undo" command.*

Cut

cuts a selected **passage out of an audio file, and** copies it to the Clipboard. All the following sections of audio move forward to fill the gap.

Copy

copies a selected passage to the Clipboard, leaving the selected area in its original location.

Paste

Inserts the contents of the Clipboard at the cursor position, or start point of the selection. If there is no selection available, the cursor acts as the paste point (it is shown as a thin dotted line). If audio data is present behind the paste point, it is moved back to make room for the Clipboard contents. If anything is selected at the time of the paste, it is deleted and replaced by the Clipboard contents.

Clear

Erases the selection without placing it in the clipboard. All data beyond the deleted passage is pulled forward to fill the gap.

Undo

Cancels the last edit command used, and reverses its effect. This also works with the destructive editing commands described in the section *Functions* on page 9-18, or described in the section *The Digital Factory™* on page 9-25.

Remember that the Undo function in the Sample Editor operates separately from the rest of the program. This allows you to try out the edit in the Arrangement. If you don't Like it you can go back to the Sample Editor and reverse the edit using the Undo function.

Space is reserved on the hard disk for armed tracks, which is not available for undo files. Logic therefore automatically switches off record-ready status for audio tracks, if the disk is nearly full, and this would make an edit possible in the Sample Editor.

9.4 File Management

Backup Copies

Automatic Backups

Before you carry out a destructive edit on an audio file in the Sample Editor, Logic will ask you if you wish to make a backup copy of the file you're working on, unless one exists already.

The **No** Dialog button ensures that this question will not be asked again, while you are editing in the current window.

You can even turn off the dialog altogether on the page **Audio > Audio Preferences**. In this case, you will be asked only when the program is first loaded, and when you carry out your first edit. You can reply as follows:

Process

No backup is made and the edit goes ahead. You won't be asked the question again until the next time you launch Logic.

Cancel

Stops the Edit.

Preferences

Opens the Audio Preferences window and gives you the chance to reinstate the Backup dialog box, so that it appears every time you go to make a destructive edit.

Manual Backups

You can make **manual backups** of the file you're editing, or replace it with a backup version at any time you like-and you can do it from the Sample Edit window with a variety of functions:

Audio File > Create Backup

Creates a duplicate of the audio file you're working on (with the extension ".dup"), and places it in the same folder, on the same level.



Audio File > Revert to Backup

This function completely replaces the current audio file with the backup (provided one exists, of course). A warning message informs you of the creation date of the backup file, before the current file is replaced.



You cannot reverse this function with Undo.

Audio File > Save A Copy As...

Copies the current audio file to the location of your choice.



Audio File > Save Selection As...

Saves the current selection as an independent audio file. After it has been saved, you can choose whether you wish to bring the file into the Audio window.



An interleaved stereo file is created when the source file is an interleaved stereo file.

9.5 Functions

The following section describes various useful functions available in the Sample Edit window, for polishing audio recordings. You can use these to add the finishing touches to your work.

Each of the commands affects only the currently selected audio. If you want to use them to alter the whole audio file, you have to use the **Select All** function beforehand.

All of the following functions are destructive, so they change files stored on your hard drive. You can use the Undo function-but only until you make another destructive edit. So you could, for example, change the start and end points of the selected audio in between destructive edits, without losing your undo facility.

Since the Undo function in the Sample Editor works independently of the rest of the program, you can try out an edit in the Arrangement and make changes there. As soon as you open the Sample Editor again (or bring it into the foreground), the Undo function is available for the last destructive sample edit.

Before these functions are executed, you are asked to confirm them as a safety measure. This confirmation dialog box can be turned off by selecting **Audio > Audio Preferences > Warning** before process Function in Sample Edit (Menu).

Data altering processes can be cancelled while underway with . The audio file will be retained in its original state, even if it appears that some of the editing function has been carried out.

Logic's more complex editing functions are found in the **Factory** menu, which is described in section *The Digital Factory*TM on page 9-25.

Normalize

Normalization is the process by which the maximum level of a digital signal is raised to a certain amount-typically to its highest possible level, without introducing distortion. Normalizing is activated in Logic by selecting **Functions > Normalize**.

This is done in the following way; Logic finds the point with the highest volume (-xdB) in the currently selected audio, and determines how far this is from the maximum possible level.

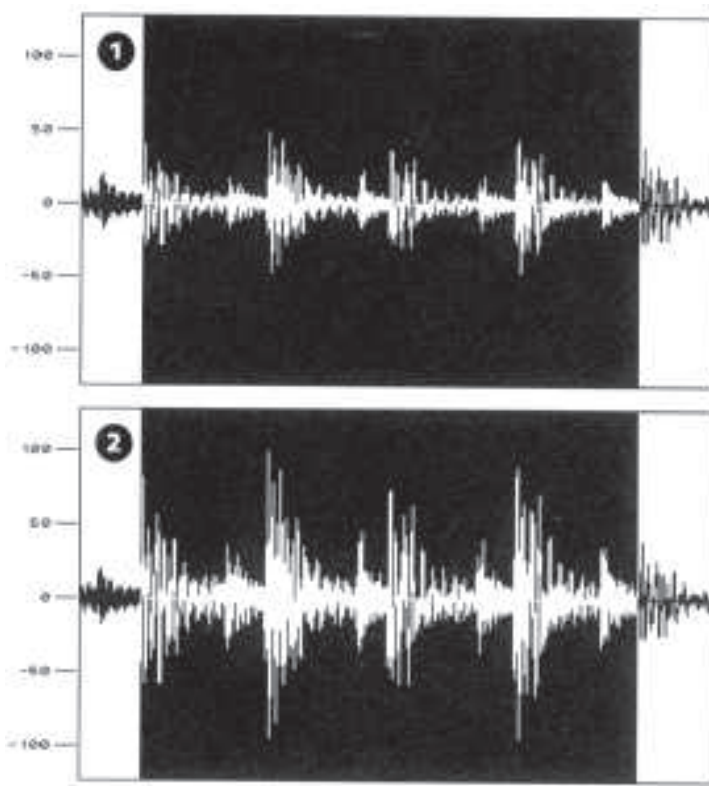


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The level of the whole selection is then raised by this amount. The dynamic balance of the audio passage remains unaltered it merely gets louder.

Under **Functions > Settings** you can select the desired maximum level in percent and decibels. It might be useful not to select 100 % (0 dB), for instance if you want to increase the gain of the overall signal afterwards, say with an EQ. This might result in clipping, if you set the normalize level too high.



 Example showing the “Normalize” function. ❶ Before ❷ After

The start and endpoints for the section being normalized should generally not fall within a continuous section of audio, as this will result in abrupt increases in volume after normalization. The start and end points should therefore be located in sections that also contain pauses. Occasionally, you should remove any unwanted, audible noises that fall in gaps in the music with the aid of the "Silence" function.

Change Gain

You can use **Functions > Change Gain** to raise or lower the level of a passage of audio by a specific amount.



A dialog box appears, in which you can set the required level change in percent (**Change relative:**) or decibels.

If you click on **Search Maximum**, the highest peak level is determined and the value is then calculated that would be used to normalize the audio file.

The value **results in absolute:** displays the maximum level that would be achieved by changing the gain by the amount shown in the **Change relative:** box.

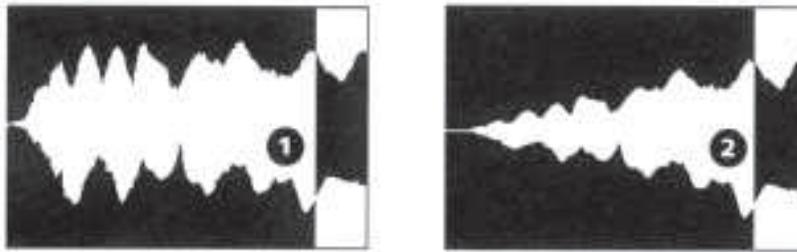
You should never make a gain change that results in a value over 100% as this would create digital clipping.

The gain change is effected by clicking **Change** (or).

Fade In

You use **Functions > Fade In** to create a fade in. You set up the period of time over which the fade-in will occur with the help of the currently selected audio (as shown in 1 and 2). Volume is set to zero at the left start point of the selection, and the fade-in occurs over the length of the selection.

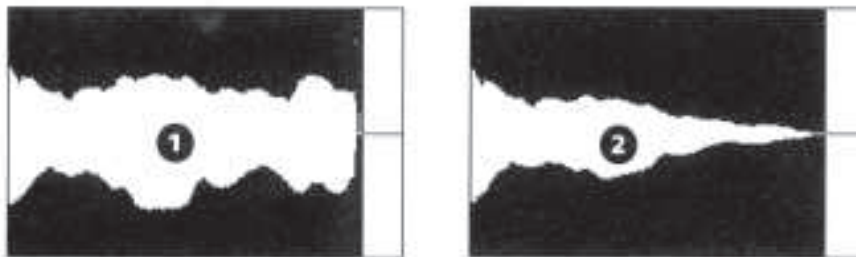




 Example showing the “Fade In” function. ❶ Before ❷ After

Fade Out

Functions > Fade Out works in the same way as Fade In, except that the fade works in the opposite direction. This lets you fade passages out automatically. this function is destructive. The following hints name non-destructive alternatives.



 Example showing the “Fade-Out” function. ❶ Before ❷ After

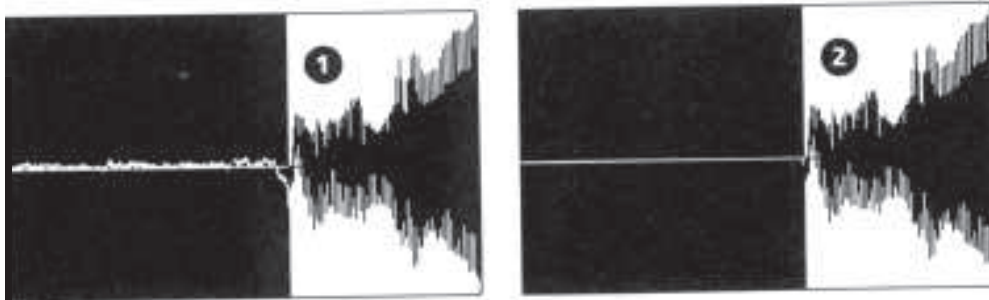
Common fades (like the typical fade-out at the end of a track) can also be achieved with the mixers, or with Hyper Draw. The “crossfade tool” in the Arrange window also offers a very flexible, non-destructive fade option.



If you use the Silence function (see below) to remove unwanted background noise from silent passages, small jumps in volume can sometimes appear at the start and endpoints of selections, as well as on the flanks of the audio signal. In this case, select only a small area (e.g. within and just in front of the flank of the signal,) and then use the “Fade-in” function.

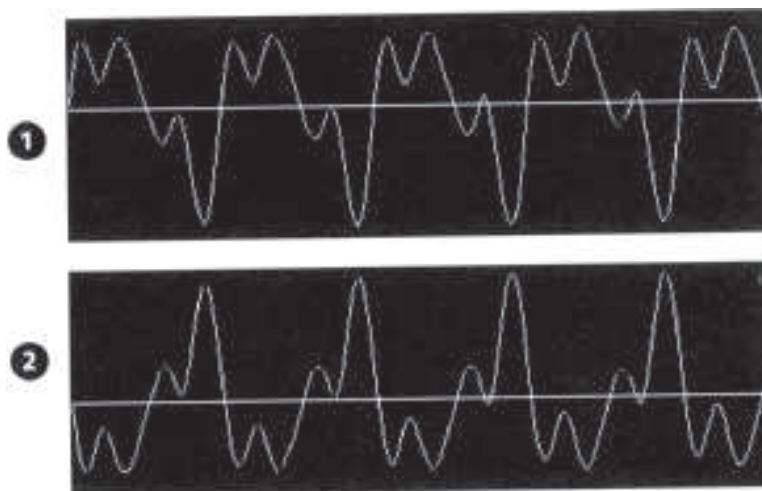
Silence

You use **Functions > Silence** to remove all data from a selected area. The waveform material contained in the selected audio passage and the corresponding amplitude values are all set to zero. You can use this function to silence unwanted background noise in quiet passages.



Invert

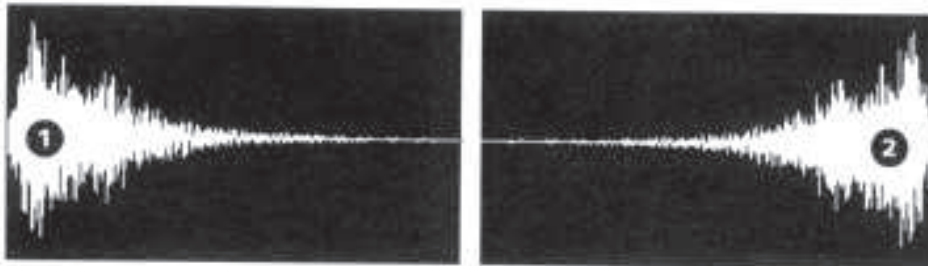
Functions > Invert completely reverses the phase of all the currently-selected audio material. All negative amplitude values become positive, and vice versa. While this doesn't change the file audibly, if it is heard in isolation, you can use Invert to correct phase cancellation errors, particularly if you're mixing down to mono. This is particularly valuable when several out of tune signals (or several signals processed with chorus pedals), are to be mixed down to mono together. The effect depends on the audio material.



You can also use the Invert function to decode MS recordings. But it is easier to use the Plug-in DirMixer for this.

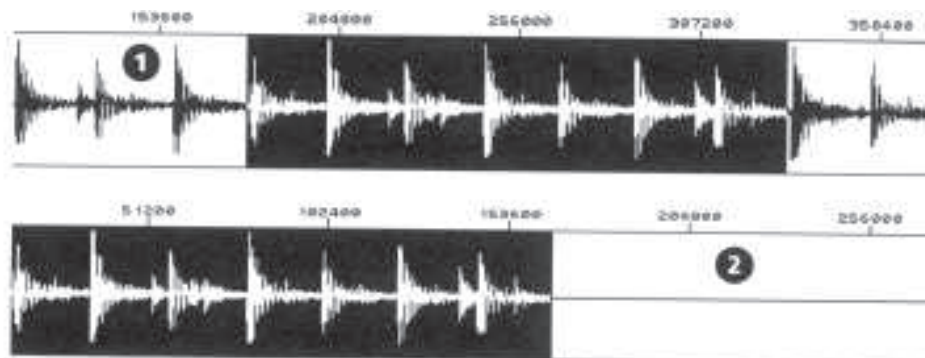
Reverse

You reverse the selected audio passage by selecting **Functions > Reverse**.



Trim

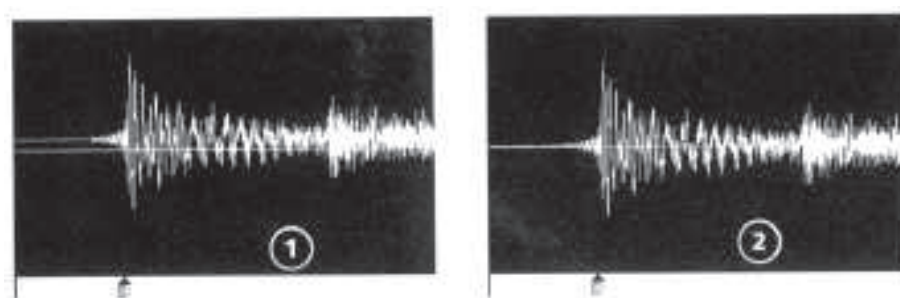
By selecting **Functions > Trim**, you can erase all the regions that *aren't* selected. Use “Trim” to remove unimportant passages from the start and end of your Audio Files.



Make sure that the areas you are about to delete do not contain any regions which you may need. Regions outside the selection will be lost, and regions which are partly outside will be shortened. If any such regions are being used in the Arrange, an alert box appears, giving you the option of cancelling the trim function.

Remove DC Offset

When using poorly constructed audio hardware, direct current (DC) can be undesirably layered over the audio signal. This results in a vertical shift in the waveform position, which can be clearly seen in the Sample Editor. During playback, this can cause crackling sounds at the start and end of the audio region.



With **Functions > Remove DC Offset** it is possible to center the waveform around the zero amplitude line, to avoid crackling at cut points.

Search Peak

When you select **Functions > Search Peak**, the currently selected audio is searched for the sample bit with the greatest amplitude value. The cursor in the waveform display is then placed on this bit.



Search Silence

When **Functions > Search Silence** is chosen, the selected audio is searched for sections containing silence (“digital zero”). The cursor is then placed at the start of the first section found that fits this description.



9.6 The Digital Factory™

Along with the real-time effects (see section *Plug-ins from Other Manufacturers* on page 9- 19), the Digital Factory™ is one of the most remarkable, and innovative features of Logic. This allows

you to edit the selected areas in audio files (e.g. regions or whole audio files), using a whole range of complex functions.

Nearly all these functions are destructive, which means that the data in an audio file on the hard disk will be permanently altered. However, the **Undo** function is available, if you need it. You can also ask for a backup file to be automatically created before you edit the file, or do it manually.

All functions in the Digital Factory can be performed on any size of mono, or stereo files, in the interleaved or split stereo format.

Please refer to the supplied Digital Factory tutorial files for a *hands on* demonstration of its various functions.

Overview

Time Machine

Independent alteration of the pitch and/or length of a recording (pitch shifting and time compression/expansion).

Groove Machine

Rhythmic-dynamic time compression/expansion, based on a definable swing grid.

Audio Energizer

Increase of the perceived volume, even with recordings already at a maximum *Normalized* level.

Sample Rate Converter

For adjusting the sample rate, or creating unusual pitch effects.

Silencer

Digital single ended noise reduction (dynamic filter and expander), plus spike reduction (declicking).

Audio to MIDI Groove Template

Creates a MIDI groove template (quantization grid) from rhythmic audio material.

Audio to Score

Generates MIDI data from monophonic audio material.

Quantize Engine

Rhythmic-dynamic time compression/expansion, based on all available quantize values.

Operation

The functions of the Digital Factory™ are available from the **Factory** menu in the Sample Edit window.

- Open the Sample Edit window by double-clicking an audio sequence in the Arrange window, or a region in the Audio window (or for a selected audio sequence/region by pressing  ).



This selects the region to be edited,

Or: select the area to be edited.

Or: press  , to edit the whole audio file.

The functions always affect the selected area.

Select **Factory** > . . .

Make the desired settings in the float window that appears.

With some Digital Factory functions, you can use the *Prelisten* button to get a rough idea of the expected result. (Not available for all digital audio hardware.). (In V 3.0 onwards, the prelisten function in the TimeMachine II also works with Audiowerk8 and Digidesign's AudioMedia III).

You can start playback if you wish to. Even though the functions alter the data in the audio file, they can still be applied while the file is playing.

Click the edit button at the bottom (e.g. *Process & Paste*).

The label on the edit button varies, depending on the function.

The top line of the Sample Editwindow keeps you informed about the functions progress.

The selected area of the audio file is replaced by the edited audio material.

Options

By selecting **Edit > Undo** (or ) you can compare the edit with the original whenever you want. In the meantime you can work in the Arrange window, because the Undo function of the Sample Edit window operates independently of the rest of the program.

You can repeat the edit with other regions or audio files, without having to close the chosen Factory “Functions” window and reopen it. The function will always be carried out on the selected material.

You can also use the flip menu at the top edge of the window to switch directly between the individual functions of every section of the Digital Factory. The two sides of the **Factory** menu (separated by a horizontal line) divide the Digital Factory into “Machines” (large float window) and “Functions” (small float window).

If you are playing your song (using MIDI and audio data) while you are editing the audio material, owners of slower computers may experience slightly “jerky” playback of the audio material, although the MIDI playback will continue to function correctly. The editing time also increases slightly, if the song is playing.

9.7 Machines

The top section of the Factory menu contains all the functions for altering audio data within a file.

Time & Pitch Machine

The Time Machine allows you to radically alter the time structure of audio files, including time compression/expansion, and pitch transposition. When changing the pitch, you have the additional opportunity to correct the alteration of the formants. Pitch shifting without formant correction causes a phenomenon commonly known as the “Mickey Mouse” effect.

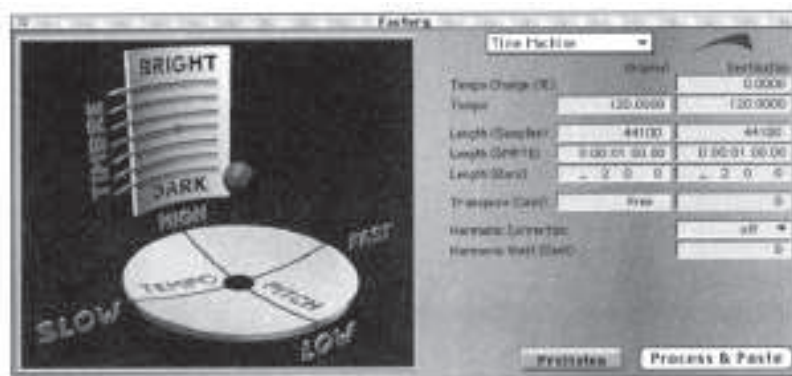
Overview of the features of the Time & Pitch Machine:

- Time compression or expansion without transposition,
- Pitch transposition with or without altering the length and tempo,
- Correction or alteration of the timbre (sonic character),
- Any combination of these functions.

Consider the Time Machine to be a universal tool for the control of pitch, tempo, and the sonic character of digital recordings. Each of these parameters can be addressed independently. The current settings are visually represented by the position of a ball in a 3-dimensional graphic display, where the axes are time, pitch, and timbre (sonic character).

Opening the Time & Pitch Machine

To open the Time Machine, select **Factory > Time & Pitch Machine**. For instructions on selecting the affected area, and basic operation please refer to the section **Operation** on page 9-27.



Parameters

Most of the parameters are mutually dependent; you don't need to enter them all to get a good result. We suggest that you don't adjust a parameter, unless you know what the value should be.

On the left side under *Original* you can see the present value, and on the right under *Destination*, enter the desired target value for the edit.

Tempo Change (%)

Tempo alteration in percent. (There is no original value here).

Tempo

Tempo in bpm (beats per minute). Make sure you set the right length in bars, otherwise the correct original tempo will not appear here on the left!

Length(Samples)

Length in samples.

Length (SMPTE)

Length in SMPTE time.

Length (Bars)

Length in bars. If you have already adjusted the song tempo to the region which you are about to edit, the original value will automatically be set correctly. Otherwise, you have to enter the original length manually here.

Transpose (Cent)

Transposition of the sound material in 1/100 semitone units (cents).

Free Transposition

Next to the transposition parameter you will see the description *Free*.

This will probably be the most common setting when you use the Time Machine. It means that the program carries out “free” compression/expansion or transposition. In this situation, the pitch and tempo of the audio material to be edited are completely independent of one another.

Classic(Correlated) Transposition

Here you can switch, using a flip menu from *Free* to *Classic*. The *Classic* mode is for situations where you want to transpose a selected region, and also affect its tempo. This produces an effect which you are probably familiar with, as it simulates the sound of changing the speed of a tape. In this case, the pitch, sonic character, and playback speed change together.

On the left of the Time Machine’s dialog window, you can see a graphic representation of the current settings. You can grab the ball within the graphic, and freely move it to adjust the compression/expansion, and pitch shift. The further the ball deviates from the center position, the harder the algorithm has to work, and the lower the expected sound quality will be.

Remember also, that the quality of the result depends greatly on the source material used.

Don’t let this keep you from experimenting. Feel free to try extreme settings for compression/expansion or transposition. In such cases, the result may not always be what you expect, but it might be just the “effect” you’re looking for.

Harmonic Correction

By activating *Harmonic Correction*, the formants which define the timbre (sonic character) are corrected. With *Harmonic Correction* set to On, the formants in the transposed material remain unchanged. This means that the original timbre (or the

physical size of the resonance body) will be maintained, and the transposition will therefore sound more natural. The only trade off is that the calculation takes more time.

By switching the *Harmonic Correction* to Off, the complete spectral structure (including formants) of a sound will be shifted according to the transposition value. This is the “normal” type of transposition, as seen in the previous Time Machine, other programs, or the pitch shift algorithms of effect processors. These algorithms are also included within Time Machine II, and can still be used for much faster calculations, or when you want the “effect” of pitch shifted formants.

Harmonic Shift

If *Harmonic Correction* is activated, you can also use the *Harmonic Shift* parameter to independently alter timbre. The units are Cents—just like with Transposition. 100 Cents = 1 semitone.

If you select the same value in both *Harmonic Shift* and Transposition, then no correction will be carried out, and the result is as if *Harmonic Correction* is switched off.

If you set *Harmonic Shift* to zero, then the formants do not change. This avoids the unwanted side effects of traditional pitch shift algorithms.

If you set *Harmonic Shift* to -300, *for example*, and the Transpose value to zero, the sonic character of the material will be changed as though it has been transposed three semitones down—but without an actual transposition in pitch. This means a musical “C” remains a “C”, but the timbre of the sound (for example, a voice) becomes darker.

Using the Graphic Display

You can move the ball in the 3-dimensional display with the mouse. Naturally, only two dimensions can be accessed at once; use *shift* to switch between two different Z-dimensional planes

(so you can reach every point of the 3D graph). By holding *ctrl*, *alt*, or *cmd* you can move the ball on one axis to independently change timbre, transposition or length.

You can also grab and move the shadows of the ball. These are its projections on the timbre axis, and the transposition/time plane, respectively

The position of the ball directly effects the numerical values, and vice versa.

You can reset the ball and all numerical values to center (neutral) positions by simply double clicking in the graphic display.

About Harmonic Correction

The whole spectral structure is usually shifted when audio is transposed-whether the method used is digital (editing samples), or analog (changing the tape speed). In this process, not only the fundamental pitch is transposed, but all the sound source's resonances (formants) are shifted as well-as though the whole instrument or singer is getting smaller or larger. This, of course, is not natural. Voices transposed up by "normal" methods will sound like Mickey Mouse, or conversely, if transposed down, like Darth Vader.

The Harmonic Correction of the Time Machine II allows you to correct this unnatural spectral shift of the formants.

You can also shift the formants without transposing. This means you can alter the physical size of the sound source's resonance body-for example to give female voices a male character and vice versa, while the pitch stays in tune. This is a remarkable effect, allowing you to change sounds so that they appear to have been made by unusually small or large instruments.

Harmonic Correction is completely different from the effect of an equalizer. You could never use an equalizer to change the apparent size of a resonant cavity.

The quality of the Harmonic Correction depends strongly on the source material, because the algorithm has to make “intelligent” decisions between tonal and atonal components of the recording, and handle them separately. This decision is not always definite, and is handled “subjectively” by the algorithm. These decisions are more simple when dealing with monophonic material than with complex stereo material, but the algorithm of the Time Machine II is able to handle a complete mix. Even the phase correlation of stereo recordings is maintained. At this point in time, this ability is unique to Time Machine II. Nevertheless, it may be possible that some material will not be processed in a satisfactory way. We urge you to experiment, in order to obtain the best results

The Harmonic Correction is a highly sophisticated DSP process, much more complex than time compression, and therefore needs more calculation time. Considering the complexity, the algorithm is extremely powerful, highly optimized and very quick.

If you need to save time, you can receive exactly the same result by first doing a normal transposition and then a Harmonic Correction afterwards. If you have to find the exact transpose value by trial and error, simply switch off the Harmonic Correction. As soon as you have found the right transposition, you can carry out an independent Harmonic Correction with the same value in a second step. With this method you can even do Harmonic Correction for audio files that were transposed previously with the "old" Time Machine.

Harmonic Correction in the TimeMachine II was developed in collaboration with Prosoniq.

Remarks on Computer Models

The PowerPC processor used in PowerMacs and other new PPC MacOS computers is very suitable for Harmonic Correction. On these PPC machines, the calculation times are comparatively short.

On computers with a 68040 processor (e.g. Quadra), the FPU-special calculation hardware-is supported. Nevertheless, this processor type is significantly slower than the PPC types, especially for these type of operations. Remember: The 68040 "LC" processor (like in the Performa 475) does not even have a FPU and is therefore much slower. The Harmonic Correction is the only process where the missing FPU is a disadvantage.

On computers with a 68030, the calculation time increases dramatically. Please remember that despite these warnings, the sonic quality of the result is independent of the processor type.

The Technology of the Time Machine

The Time Machine analyzes the spectral components and dynamics of the digital audio material, and then processes the result. The high-grade algorithm endeavors to retain as much spectral and dynamic information as possible, and minimizes phase variations. In stereo files, the phase relation between the left and right channels is fixed, and not altered. Doubled sound events are kept to a minimum. All this produces a high quality result, which despite the speed of the process, is less grainy than what you're probably used to hearing from other products.

However, you should bear in mind that apart from resampling (transposition), the Time Machine has to achieve the "physically impossible": when a sample is lengthened information has to be "invented." This should be as realistic as possible, and conversely when a sample is shortened information has to be cut out, which should be as unimportant to the overall character of the sound, as possible. Lengthening is more difficult than shortening, and if you have a choice it is better to speed up a drum loop that is too slow than vice versa.

There is always a small deviation between the set stretch or compression factor, and the actual result. This is because the algorithm needs a bit of freedom to optimize the spectral and dynamic integrity (the sound quality). The deviation from the set value is only a few milliseconds (or fractions of a bpm). This shouldn't present a problem, since the abso-

the deviation is independent of the Length of the processed section, meaning the deviation is no greater in longer files.

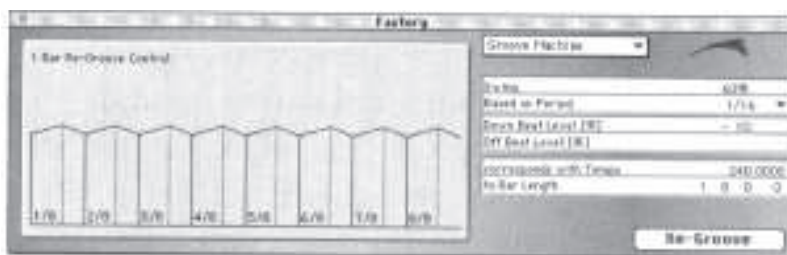
Groove Machine

The Groove Machine allows you alter the feel, swing, or groove of the digital audio material, in percentage steps. We call it “digital re-groove”.

Opening the Groove Machine

First, make sure that the tempo of Logic exactly matches that of the selected audio material.

Select **Factory > Groove Machine** in the Sample Editor. The function is executed by clicking the **Re-Groove** button.



Please see the section *Operation* on page 9-27.

Parameters

Next to *Swing*, you set the percentage for the swing factor. At 50% there is no change-you normally get the best results at 55% to 65%.

The *Based On Period* parameter lets you tell the Groove Machine whether the audio material should be edited on the basis of eighth note, or sixteenth note swing.

The *Down Beat Level* and *Off Beat Level* parameters are where you select whether the down beats (or off beats) in the audio material should be raised or lowered. Positive values increase the level, negative values reduce the level.

Bear in mind that raising the level of normalized audio material can cause distortion (particularly on the down beats, which are usually louder),

The *Corresponds to Tempo* parameter automatically transfers Logic's current tempo to the groove machine.

The length or tempo of the selected audiomaterial must be defined, otherwise the Groove Machine cannot work accurately. The length can be defined in musical values (bars, beats, divisions and ticks) under to Bar Length, or you can set the tempo directly, using Corresponds to Tempo.

Audio Energizer

The purpose of the **Audio Energizer** function is to increase the perceived volume of the audio material, while altering the sound as little as possible, and without causing clipping. Digital distortion (which would be the inevitable result of simply increasing the level, and which sounds very unpleasant) is avoided using this algorithm.

You could compare the effect to that of an analog tape which is saturated by a high recording level. However, the distortion factor and effect on the audio material by the audio energizer is much lower.

Here's an example of the effect. If an already normalized audio file (audio data which is already at the maximum dynamic range) is edited, the effect is as follows: a VU meter will show a higher level, indicating increased average energy in the signal. However, a peak display will show the same level as before, since the maximum signal level has not been exceeded.

Opening the Audio Energizer

Select **Factory > Audio Energizer** in the Sample Editor. For information on the effective range, and basic operation, please refer to the section *Operation* on page 9-27. The function is executed by clicking the *Energize* button.



Parameters

The main parameter is *Factor*. This is where you select the amount of average level boost. 0% means no alteration, while higher values produce an increase in energy. The setting you make here will depend on the audio material, situation and personal taste. Begin by trying values in the 40-100% range. Values below 10% will hardly have any effect, values over 100% can lead to undesirable alterations in the sound, depending on the material. Values over 200% are not recommended with normalized files, because they will have detrimental effects on the sound and dynamics. They can also greatly increase the required computation time. On non-normalized audio data even high values can be effective because initially the overall level is increased to the maximum, without affecting the dynamic range.

The *Attack* and *Release* parameters affect the algorithm for controlling the steepness of the filter. You can try increasing these values to double or four times the default, if the result sounds too “digital” or “raw”. This can happen if small elements previously hidden among the “main events” in the original are boosted. For example, the reverb sometimes gets louder.

The perceived loudness of the overall audio material is increased. if the material contains anomalies such as noise, these will also be increased, and sometimes becomes audible. If necessary, you can edit the result using the noise reduction function (Silencer) on a low setting.

Silencer

The Silencer consists of two component functions, which can either be used separately, or together:

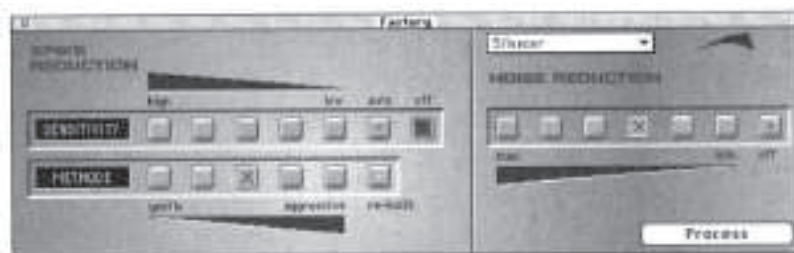
The “Noise Reduction” lowers the level of any noise in the signal, such as tape noise.

The “Spike Reduction” tries to identify and reduce anomalous signals such as pops or clicks.

You will probably find that you will use the Noise Reduction more often than the Spike Reduction.

Opening the Silencer

Select **Factory > Silencer** in the Sample Edit window. For information on its effective range, and basic operation please refer to the section *Operation* on page 9-27.



Noise Reduction

The purpose of the Noise Reduction is to reduce the noise components of the signal, and also affect the main signal, particularly the high frequency elements.

The process is “single ended”, meaning it affects material which is already recorded, and there is no need to “decode” the

signal during playback. This has the advantage that even noise which was already present in the original signal can be processed.

Operation

Set the **Spike Reduction** function to *off*, so that only the **Noise Reduction** function is used. Click the *Process* button to execute the function.

There is only one parameter for controlling the intensity of the process. Off means no edit, Min is the least possible edit, small numbers have a small effect and *Max* produces the maximum possible reduction of the noise component. The “correct” value depends on the quality of the material, and your own personal taste. Good quality material should only be edited using *Min* or 2. You will probably notice only minor changes in the sound. Bad (noisy) quality material should be processed using higher values or even Max. If the setting is too high, the treble component of the main signal will also be reduced.

The Silencer function is optimized normalized data. Low level material should be normalized first.

Spike Reduction

The aim of Spike Reduction is to identify and reduce “abnormal” signals such as pops, clicks, or digital “spikes”, in order to reconstruct a hypothetical “original” signal.

Operation

Set the Noise Reduction switch to off, so that only the Spike Reduction function applies. Click the *Process* button to execute the function.

The *Sensitivity* parameter controls the automatic recognition of the “spikes”. When set to High (sensitivity) instead of *Low*, even relatively small spikes are identified. When set to *Auto*, the threshold is controlled automatically by the program.

The *Method* parameter controls the intensity, i.e the way these spikes are processed. *Gentle* is the weakest algorithm, and only slightly smooths the original signal, while *Aggressive* is the strongest. All these settings “filter” the original signal at the identified position.

By contrast, the *re-build* setting completely replaces the often unusable original signal with an artificially generated estimate of the original signal, at the identified points. This synthesized signal is created by analyzing the audio material surrounding these points.

Experiment with the parameters. If the *Sensitivity* is too high, there is the danger that very sharp transients in the main signal will be identified as spikes. Depending on the audio material, it can sometimes be impossible for the program to distinguish between the two. This is particularly the case if the decision “spike or main signal” is merely a matter of taste, for example with some “click-like” bass drum sounds. In their attack phase, these sounds can have similar characteristics to static on vinyl.

If the automatic identifacition does not produce satisfactory results, pops and clicks can also be removed manually. Select the relevant area in the Sample Editor, and carry out a spike redaction using the settings “High Sensitivity” and e.g. “Filter5”. This method is easier than manually “drawing” waveforms.

Sample Rate Converter

The Sample Rate Converter is used for converting the sample frequency. For example, audio files which have a 48kHz sample rate (recorded on a Hi-Fi DAT recorder and digitally transferred into the computer), can be converted to 44.1 kHz.

Opening the Sample Rate Converter

Normally you will want to convert the sample rate of a whole audio file. To do this, select the whole audio file in the Sample Editor (cmd+A).

Select **Factory > Sample Rate Convert...** in the Sample Edit window. For information on the effective range and basic operation please refer to the section *Operation* on page 9-27. To execute the function, click the Convert button.



Parameters

Source (Hz)

This shows the previous sample rate of the audio file. However, to enable you to change incorrectly stored sample rate formats (for example after editing in other programs & or for effects you can enter any value you like here). You should only change this parameter if you know what you are doing.

Destination (Hz)

Here you can enter the desired sample rate to which the selected area is to be converted.

In most professional audio circles, only the 44.1 kHz sample rate is used. There is little audible advantage in using 48kHz. This format is mostly used with older DAT machines.

9.8 Functions

The second section of the **Factory** menu contains functions involving an interaction between MIDI and audio data.

Audio to MIDI Groove Template . . .

This function enables Logic to create MIDI groove templates from digital audio material. This means you can extract the feeling (the precise rhythmic references) from drum loops or other samples containing rhythmic passages, and use the resulting groove template to quantize MIDI sequences.

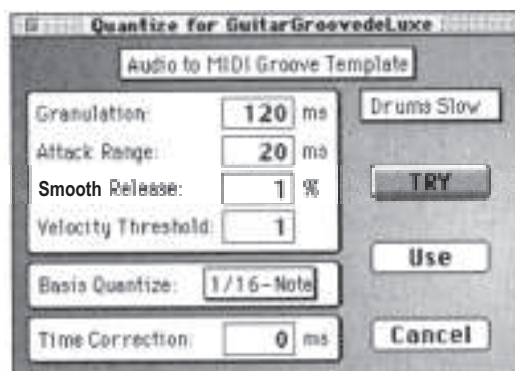
The following example explains how to do this using a one bar drum-loop:

Load the drum loop into Logic by selecting **File > Add Audio File** (Audio window).

▮ Drag the new region into the Arrange window.

▮ Now, open the Sample Editor by double-clicking the region.

Select **Factory > Audio to MIDI Groove Template...**



Parameters

The first four parameters control the analysis algorithm. They are listed in order of importance:

Granulation (ms)

This determines the time span of louder components of the audio material, from which Logic can derive information for

“velocity points” in the groove template. The most useful values are usually between 50 and 200 milliseconds, depending on the tempo of the audio material.

Attack Range (ms)

This tells the program how long the attack phases of the sounds in the audio material are. For example, drum and percussion instruments have short attack times of less than 20 ms, while string instruments have longer attack phases. The best values for most instruments are usually between 5 and 40 ms. Many instruments are near 20 ms.

Smooth Release (%)

This parameter is specifically for processing audio material containing sounds with a long release and/or reverb. This makes it easier to convert these sounds into quantization points. The setting here should generally be between 0 and 5%, except when processing passages with long, distorted guitars, or similar sounds.

Velocity Threshold

This parameter sets the threshold below which the individual sounds in the audio material are to be ignored. In most cases the value “1” is suitable, except when processing very dense, loud material with soft background noises.

There are more options containing the following parameters:

BasisQuantize

This function enables you to add artificial trigger points at positions in the audio material where there are no trigger points present. The groove templates obtained using this method are suitable for situations where you need more quantization points than are contained in the audio material.

The identification of trigger points in the audio material is not affected by this parameter.

*After selecting the **Audio to MIDI Groove Template** function, you can then determine the hits of the template. Move the selected material in the Sample Editor so that it fits the musical bar divisions of your song. The small "triangular lines" allow you to precisely compare the trigger points in the audio material with the musical positions of "basis quantize". Remember: the selection can be moved in the Sample Editor by holding down the Alt key, and dragging it.*

TimeCorrection

This parameter allows you to compensate for any time delay which may occur when external samplers or synthesizers are triggered by MIDI notes. These time delays are sometimes very noticeable, if the connected device is playing a sequence which was quantized using an Audio-to-MIDI groove template at the same time as the original audio material is being played. You should be able to compensate for this effect by using settings between -20 ms and 0 ms.

Instrument Type

This flip menu contains various presets for the Audio-to-MIDI parameters, which are suited to specific types of audio material. You can use these presets as starting points for your own processing.

Your own parameter settings are stored in the Logic Preferences file.

At the bottom of the Sample Editor are three fields labeled: *Audio Qua*, *Basis Qua* and *Result Qua*.

Audio Qua

Displays the quantization points which Logic has identified in the audio file.

Chapter 9

The Sample Edit Window

Basis Qua

Shows the quantization points which you have selected under *Basis Quantize*.

Result Qua

Shows the quantization positions in the new groove template, which are produced by the combination of the two values above.

Clicking on an audio trigger point will keep it from being transferred to the template. These manually selected points will be “grayed out”.

When you click *Try*, Logic will apply the new groove template to all selected sequences, allowing you to preview the result.

Adjust the parameters until you are happy with the result, and then click *Use*.

Clicking *Use* saves the new groove template, and installs it in the current song. It then appears at the bottom of the list in the *Quantize* flip menu (the menu found in the sequence parameters of the *Arrange* window). This means your new groove template is available for use whenever you need it.

*You can also select a region in the Arrange window and select **Factory > Make Groove Template**, which completes the whole process in one go.*

*When using this function, you should switch off the **Edit > Search Zero Crossings** option in the Edit menu of the Sample Editor*

Audio to Score Streamer

This function allows you to transform a monophonic audio recording into musical notation, by creating a MIDI sequence corresponding to the recorded melody. This allows you to have

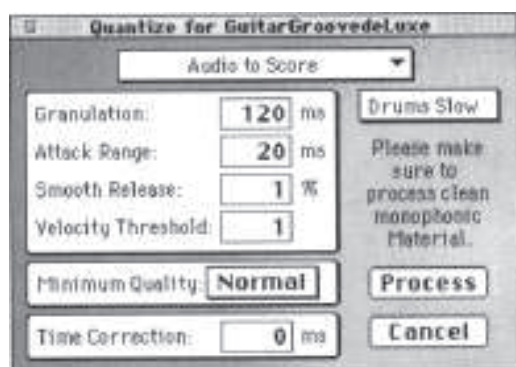
a MIDI sound module playing in unison with an audio recording or -transposed- as a second voice. This function is best-suited to producing melody notes from a clearly sung vocal line. It also works best with non-legato performances.

The Audio to Score Streamer creates a sequence in the Arrange window containing the optimum interpretation of the audio data. The notes are displayed in the Score window, immediately after the conversion.

Opening the Audio to Score Streamer

Begin by selecting the MIDI track in the Arrange window where the new MIDI sequence should be placed.

Select **Factory > Audio to Score...** For general information on operating the Digital Factory functions, refer to the section *Operation* on page 9-27.



The parameters for the Audio to Score function are similar to those used by the Audio to MIDI function used to create a groove template. For details refer to the section *Parameters* on page 9-43.

Minimum Quality is the only extra parameter. It lets you select between normal, or high quality processing, as the preset value of the program. This allows you to deal with situations which demand the highest possible processing quality. The *normal* Setting is more tolerant of audio trigger points which are not perfectly in tune. However, this setting can also generate wrong notes. If the minimum quality is set to **High**, Logic only accepts

audio trigger points which have a clearly identifiable pitch. Indistinct trigger points are replaced by notes with a pitch of C3 and MIDI channel 3, labeled as an *undetected mark*.

When analyzing audio material containing clearly identifiable pitches, you will find that you get good results from either setting.

Click **Process** to execute the function.

A MIDI sequence containing the generated data is automatically created in the Arrange window, on the selected MIDI track.

A Score window opens automatically containing the transcription of the audio file generated by the function. The Score Editor uses a score style called **Audio To Score** with three staves. This style can be edited just like a normal score style. The three staves contain the following information:

Stave 1 contains the pitches that were most clearly recognizable. All these are assigned MIDI channel 1.

Stave 2 contains the pitches that were less clear, for example, if the pitch is wrong, or could not be clearly identified by the first analysis. These notes are assigned MIDI channel 2.

Stave 3 contains all other information provided by the function, on MIDI channel 3. For example, detuned pitches produce a “dummy” note (C3 on MIDI channel 3). This can happen if the trigger point has a very short attack, or if the note has a high interference element (such as click noises or buzzing strings produced by a guitar).

*Depending on the selected file, notes can be detected more or less reliably. If the detection is certain, the MIDI notes are assigned to MIDI channel 1. Otherwise they get a C3 with MIDI channel 2, so that at least the starting point and length information will be retrieved. The threshold for “certain identification” is defined with the **quality** parameter. What's more, Logic offers the option to define pauses as notes. This especially is useful with MIDI Groove templates, so that these won't have*

gaps. These gap-filling notes appear as C3 with MIDI channel 3. They do not represent sound events, but breaks.

A few tips on how to get the best results using the Audio to Score function:

You will only get good results with clearly identifiable monophonic audio material. Solo voices, strings, and piano can be easily analyzed, as can any recordings with distinct pitches. Experiment with different parameter settings for processing the audio data. Sometimes, you will need to make several attempts to determine the optimum settings for a particular recording.

Quantize Engine . . .

This function allows you to use a MIDI sequence template (within reason), to quantize an audio recording.

This is done by using a dynamic time compression/expansion algorithm. The function is similar to the Groove Machine, except that the quantize template can reflect values other than just 8th or 16th note swing grooves. Even user defined MIDI grooves can be used.

The Quantize Engine is therefore the inverse of the Audio to MIDI Groove Template function.

Opening the Quantize Engine...

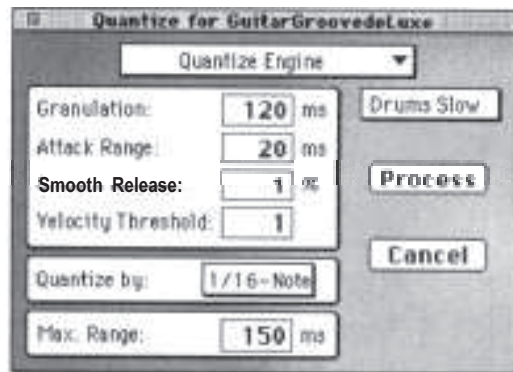
First select the audio sequence in the Arrange window to which you want to apply the groove template.

Select **Factory > Quantize Engine...** For general instructions on operating the Digital Factory functions, refer to the section *Operation* on page 9-27.



Chapter 9

The Sample Edit Window



The parameters for analyzing the audio material are identical to the ones used for the function, **Audio to MIDI Groove Template**. Please refer to the section Parameters on page 9-43.

Quantize by

This flip menu is where you set the required quantization. The same templates available for MIDI data are available here.

Max. Range

This is where you set the maximum time in milliseconds by which a peak in the audio material can deviate from the relevant quantization point in the groove template, while still being interpreted as “quantizable”.

Small values are suitable for editing audio material which already has a similar groove.

Large values allow you to use a groove template which deviates more from the original groove. However, this increases the risk of misinterpretation.

Operating tips:

First use the flip menu at the top right to select a set of parameters which correspond to the characteristics of the audio material. Start by experimenting with smaller values for the *Max. Range* parameter (“as small as possible, and as large as necessary”).

Chapter 10

The Event List

The Event List displays MIDI data in the form of a “list”, combining all the data-editing functions of the other editors, (except for the graphic operations and expressive options offered by the Score Editor).

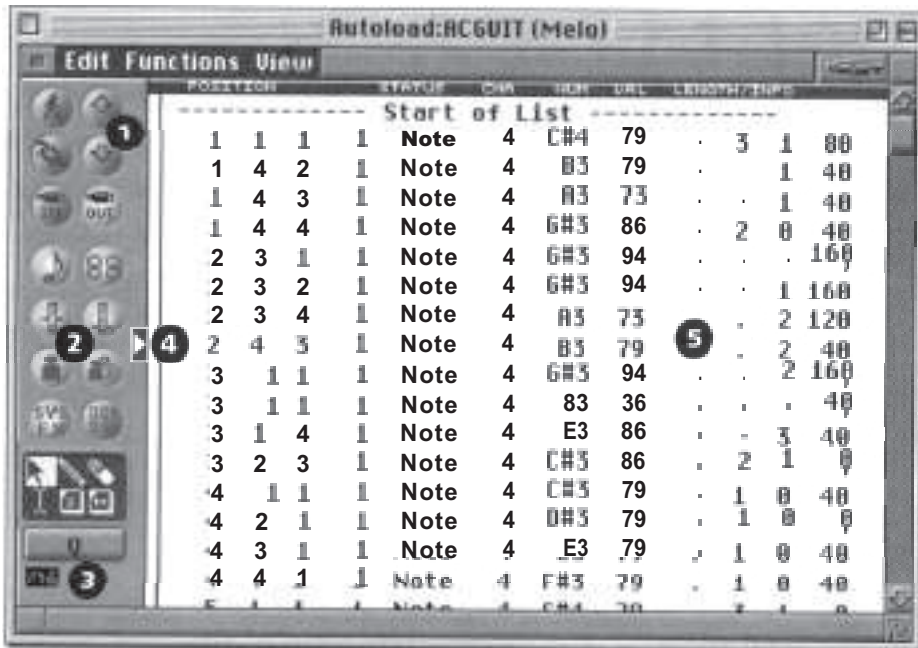
Usage

The Event List is used whenever you need to make precise alterations to recorded data, where the graphic display of the other editors is not as well suited to the task. It is the only editor which gives you access to all recorded event data. You can also restrict what you see, allowing you to edit only specific event types.

Opening the Event List

To open the Event List and view the contents of the selected sequence, select **Windows > Open Event list** (default *Cmd+2*): or use a user-defined key command (Open *Event Editor*..





You can also open it by double-clicking on a sequence if you have selected *Double Click Sequence to open Event Edit in Arrange* under **Options > Settings > Global Preferences** (or by double-clicking on the sequence while holding down the *alt* key, if another editor is selected there).

Structure

The standard buttons are supplemented by two scroll arrows ❶ to help you move through the list. The event type buttons below them ❷ allow you to filter specific event types from the display, and access or add them (by clicking on them while holding down the  key). Beneath the toolbox is the quantization grid selection field for the event quantize function (above ❸). There is also a field for defining the Division value (to the left of ❸), which corresponds to the Division value as set in the Transport window. If the Catch function is switched on, the arrow-shaped position marker ❹ will always point to the current event. The structure of the actual list display ❺ is described on page 10-8.

10.1 Display

The event type buttons allow you to “filter” the display, to remove individual event types from the Event List, so that you can view only the specific types of events you are interested in. Click the desired button with any tool (except the pencil).

If a button is grayed out, that event type will not be displayed.

All the functions affect only the events displayed, so non-displayed events are protected from any alterations you make.

Here is a short overview. For more detailed information on the individual event types, see page 10-11.

The note symbol stands for note events.



The symbol with the dual-digit, seven-segment display stands for *program change events*.

The hand wheel symbol with a marker in the middle (pitch bend wheel) stands for *pitch bend events*.



The hand wheel symbol with a marker at the bottom (modulation wheel) stands for all *control change events*.

The single weight symbol stands for *aftertouch events* (channel pressure).



The multiple weights symbol stands for *polyphonic key pressure events* (polyphonic aftertouch).

This symbol stands for *SysEx events*.



The symbol with a row of zeros and ones is called the *full message* button. This does not filter out any type of event, but affects the amount of detail displayed for all event types.

Normally the display in the Event List is restricted to one line per event. When the full message display is active, all information stored along with the event is shown too. This is particularly important for editing SysEx messages. When examining note events in the Event List, you will also notice Logic’s internal score layout information included in the list.

You can edit this in the Event List if you want, but it serves little purpose.

10.2 Operation

Scrolling

Clicking either of the scroll arrows moves the display up or down by one event. The event at the position marker is always selected (so the existing selection changes as you scroll). The scrolling speed can be varied as you scroll, by vertically moving the mouse. The key command *Scroll to Next/Previous Event* is also available from the Event window, and has the same end result.

Remember that if the MIDI Out button is switched on, every newly selected event will be played. This means you can scroll through the list and audibly monitor the events as you go.

If you want to keep the selected event where it is, use the usual scroll bar functions.



Selection Techniques

When selecting events with the mouse, you should click near the status column, to avoid any unintentional parameter alterations.

You can use any of the standard selection techniques here: individual selection by clicking on objects, multiple selection using the rubber band, or both of these (without altering the previous selection), in conjunction with the *shift* key. Don't forget you can also make selections according to specific criteria with the **Edit** menu (read the section *Selection Techniques* on page 1-23).

Any events which you remove from the display, by clicking on the event type buttons, are immediately deselected. This ensures that all the functions affect only the displayed (selected) events.

Special Selection Functions

Some selection commands (which can be accessed from within all the Editor windows via their Edit menus), can also be utilized in the Event List by selecting an item with the mouse, while holding down additional modifier keys.

If you click on an event while holding down the  key, you select all events between the last event that was selected and the current one.

If you hold down the  key and select an event, all similar events will also be selected.

If you select an event while holding down the  and *alt* keys, all **identical** events will be selected.

Event Editing

Adding Events

To add an event, click on the desired event type button with the pencil (or with any other tool, while holding down the  key). The event is then added at the current song position, and is automatically selected.

Duplicating Events

To duplicate an existing event (for example, so you can alter the parameter value of the copy) click on the original event with the pencil. An input box appears-this is where you enter the position for the newly-duplicated event. If you just hit *Enter*, the duplicate will appear at the same place as the original.

Pasting from the Clipboard

When adding events using the clipboard's **Paste** function, a position input box will appear, allowing you to type in the position of the first event. Once again, if you confirm by just pressing , the original position of the event is retained. The rela-

tive positions of other copied events are also always maintained.

This means that the events are not added at the song position, as in the graphical editors. For more details on the way the clipboard works, please refer to page 1-27.

Moving Events

To move an event in time, alter its position indicator. As soon as you alter its position, the list is automatically sorted, even though the currently selected event remains the same.

Altering Values

Event positions or parameter values can be altered in the usual way by using the mouse like a slider (grabbing and dragging), or with text input (just double-click on the parameter value).

You cannot alter the type of events using this method, however. To alter the event type, you must either open a transform window, or add an event of the desired type, and then delete the original event.

Altering the Values of Several Events

If a parameter of a selected event is altered, it will affect the same parameter in all selected events.

Relative Value Alteration

When you alter parameter values in a group of selected events, the relative differences between the parameter values remain unchanged. This means that the parameter values that you grab or double-click on can only be altered until the value of one of the selected events has reached its maximum or minimum value.

Flexible Relative Value Alteration

If you want to continue altering a parameter value in a multiple selection (even if one of the values in the selection has reached its maximum or minimum), hold down the *alt* key while you move the mouse, or press *enter* to confirm a numerical input.

Absolute Value Alteration

If you want to set a parameter to the same value for all selected events, hold down the  and *alt* keys, while you use the mouse as a slider, or press  to confirm a numerical input.

Numerical Value Input

If you want to directly input a number, double-click the relevant parameter.

There are a few special points that need mentioning concerning numerical editing of events in the Event List:

When entering event time positions and lengths, you can jump from part to part within the value you're inputting, using the  key;

New events can be created simply by using the ASCII keyboard. When you're numerically editing events, you can duplicate them by simply hitting the  key.

The key command "*Duplicate Event and Numerical Edit*" also allows you to create new events.



10.3 Event List Structure

The individual columns in the list have the following meanings:

POSITION	STATUS	CH	NR	DAL	LENGTH/INFO

Position

The position of the events in the song; for note events this means the beginning of the note. The units represent bars, beats, divisions, and ticks (see page 2-17).

Counting begins at 1 for each unit (first bar, first beat, first division, first tick: 1 1 1 1), and continues until it is carried over to the next largest unit.

Numerical inputs start from the left (which means you can enter just the bar number if you want). The units can be separated by either spaces, dots, or commas.

Position within the Sequence

If you select **View > Local Position**, the event positions do not refer to the absolute location within the song but to the relative position within the sequence.



Position and Length in SMPTE Units

If you select **View > Position & Length in SMPTE units**, the position and length columns switch to SMPTE units.



Status

This is where you can see the event type, as specified by the status byte of a MIDI message (for details, refer to the section *Event Type Structure* on page 11. You cannot directly edit this parameter.

Cha

The MIDI channel used to record an event.

Remember that during playback this MIDI channel will be replaced by the *Cha* parameter of the playback instrument. The event is only output to the recorded MIDI channel when the *Cha* parameter is set to *All*.

You should also keep in mind that with notation, the recorded MIDI channel is used to assign a polyphonic voice to the note. (for more on this, read Chapter 13).

Num, Val

These columns contain event data bytes. Their meaning depends on the event type:

Status	Num	Val
Note	Pitch	Velocity
Control	Controller number	Value
Pitch	LSB	MSB
C-Press	(not used)	Value
P-Press	Pitch	Value
Program	Bank Select	Program number

Length/Info

With controller events, this column shows the controller name, and with SysEx events, the manufacturer's name.

With pitch bend events, a 14-bit value is displayed here, which is composed of the first (Num) and second (Val) Data bytes combined. This value can be edited directly from here.

With notes or sequences, the length is displayed here.

Here too, the units are bars, beats, divisions, and ticks. For the sake of clarity, when the length begins with one or more zeros, the "-" symbol is used instead. The minimum length is 1 tick

(---1) not 0 ticks, because it makes no sense to simultaneously switch a note on and off.

Numerical input starts on the right, working to the left-and you can enter just ticks if you want. The units can be separated by either spaces, dots, or commas.

End instead of Length

If you select **View > length as absolute Position**, you can make the length display show the absolute position of the note off event, instead of its length from the note on.



List Structure on the Arrange Level

When you click on this symbol, or double-click on any entry, you move up one level in the folder/sequence hierarchy. You may also move up a level by double-clicking the white area above or below the list entries (if visible).



POSITION	STATUS	CH	HLS	URL	LENGTH/DIF
----- Start of Event List -----					
----- End of Event List -----					

This allows you to see all the arrange objects (sequences or folders) in the current song (or folder). On the arrange level, the list has the following columns:

POSITION	NAME	TRACK	LENGTH
----- Start of Event List -----			
3 1 1	Main Theme	1	14 2
3 1 1	Main Theme	2	14 2
3 1 1	Main Theme	3	14 2
3 1 1	Main Theme	4	14 2
----- End of Event List -----			

Position

The start of the arrange object (see page 10-8).

Name

Name of the sequence or folder. Double-clicking on the object name switches you to the display of its contents. With a sequence, you return to event-level editing.

You can edit the name with the text tool.

Track

Displays the track number. You cannot edit this value from here (you might totally destroy your arrangement if you could). If you want to move arrange objects to other tracks, it is better to do it graphically from the Arrange window.

Length

The length of the arrange object (see page 10-9).

10.4 Event Type Structure

Note Events

POSITION	STATUS	CHN	NUM	VAL	LENGTH/PRP
1 1 1	1 NOTE	1	82	80	1 1 232

Num

MIDI note number (note #). The range is from C-Z (note #: 0) to G8 (note #: 127). Middle C is note #60 and in MIDI terminology is called C3.



On some keyboard/synth modules (notably those made by Korg and Roland), the note range is C-1 (#0) to G9 (#127). In these cases middle C is called C4.

In the **Options > Settings > Display Preferences** page, you can reference the display to the description used most frequently on your devices (*Display middle C as..*).

Val

Velocity of a note from 1-127. The value 0 carries the note off message, and thus is not available.

Length/Info

Length of the note. Although MIDI can only transfer note on or note off messages, LOGIC actually stores the position and length of all the notes, which makes them easier to access musically. The note off message is generated automatically during playback.

Mapped Instrument Notes

POSITION	STATUS	CH	NUM	VAL	LENGTH/INFO
25 1 1	1	High TOM 1	1	.	1 57
25 1 3	1	High TOM 2	1	.	190
25 2 1	1	Mid TOM 1	1	.	1 17
25 2 3	1	Mid TOM 2	1	.	210
25 3 1	1	CHINA	1	.	232
25 4 1	1	SD 2	1	.	207
25 4 3	1	KICK 1	1	.	230
26 1 1	1	KICK 1	1	.	107

If the edited sequence is played through a mapped instrument, the defined names of the individual notes appear in the Status line. For the sake of clarity, there is a small note symbol to the left of each name.

Program Change Events

POSITION	STATUS	CH	NUM	VAL	LENGTH/INFO
1 1 1	1	Program 1		0	Grand Piano

Program change events can be transmitted to connected MIDI devices to call up different patches. These may be sounds in a synthesizer, programs in an effects unit, or snapshots in an automated mixing desk.

Val

A program number between 0 and 127.

Some manufacturers (e.g. Yamaha) number the programs in their devices from 1 to 128, not 0 to 127. In this case, you have

Event Type Structure

to subtract 1 from the program number given in the device itself.

Other manufacturers use various methods of dividing into groups (or banks) and sounds. The most common is dividing into 8 groups of 8 sounds, each numbered 1 to 8. These devices respond to program numbers 0-63 by calling up storage locations 11-88. The instruction manuals for these devices should contain conversion tables to assist you.

Num

Bank select. Normally you will see this symbol, which means no bank select will be sent. If you assign a number between 0 and 62, a bank select event is sent before the program change event. This allows you to address different sound banks (e.g. preset, internal, card) inside your synthesizer. The synthesizer must be able to recognize controller 32 as bank select, but unfortunately this standard is not yet widely used. If you have any problems with bank select, check your synthesizer's manual to see whether, and how it responds to bank select commands. You can change the type of bank select message Logic sends for each Multi instrument in your setup. See section *Defining Your Own Bank Select Commands* on page 5-32 for details.



Pitch Bend Events

POSITION	STATUS	CH	PRG	SR	LENGTH	INFO
1	1	1	PitchBd	1	0	0

Pitch bend events are used to continuously vary the pitch. They are usually generated by a centered pitchbend wheel, or a joystick on your keyboard.



Num

Fine pitch bend division (LSB). Many keyboards just transmit the value 0. If the pitch bend wheel has an 8-bit resolution, you will see the value 0 or 64 here.

Val

The effective pitch value (MSB) of 0-127. The value 64 corresponds to the mid-way setting of the wheel.

Length/Info

The 14-bit value is displayed in this column as a decimal figure ranging from -8192 to 8191. The value in this column may be edited in the usual way.

Control Change Events

POSITION	STATUS	CHN	NUM	VAL	LENGTH/INFO
1 1 1	1	Control 1	71	127	Resonance
1 1 1	1	Control 1	10	127	Pan
1 1 1	1	Control 1	64	127	Sustain
1 1 1	1	Control 1	1	127	Modulation
1 1 1	1	Control 1	7	127	Volume
1 1 1	1	Control 1	127	127	Poly Mode On

These event types are used to transfer MIDI controllers (e.g. modulation, sustain, volume and pan).



Num

The number of the controller. All the various MIDI controllers (such as the modulation wheel or sustain pedal) have their own numbers (#1 or #64 respectively). Some other controllers are also defined, such as volume (#7) or pan (#10).

Controllers that are defined in the MIDI Standard are described in the **Length/Info** field.

Val

Value of the controller. Continuous controllers have a range of 0-127. Switch controllers (#64-#90), transfer only two states; off (val=0) and on (val anything between 1 and 127).

Aftertouch Events

POSITION	STATUS	CHR	NUM	VAL	LENGTH/INFO
1 1 1	1 C-Press	1		0	

Aftertouch (or channel pressure) events are generated by a mechanical pressure sensor beneath the keyboard. The resulting sound modulation affects all the notes on that particular MIDI channel.



Num

This column is empty with aftertouch events, since they have only one data byte.

Val

Strength of the pressure on the keyboard (0 to 127).

Poly Pressure Events

POSITION	STATUS	CHR	NUM	VAL	LENGTH/INFO
1 1 1	1 P-Press	1	E3	0	

Poly pressure events are generated by mechanical pressure sensors beneath each individual key. The resulting sound modulation affects only that particular note.



Only a few keyboards currently support this capability.

Num

MIDI note.

Val

Strength of the pressure on the key.

SysEx Events

POSITION	STATUS	CHR	NUM	VAL	LENGTH/INFO
1 1 1	1 SysEx		4	0	Moog
		0	0	0	0
		0	+	-	+
					< EOH >

For more on how to program a SysEx message manually, check out the section *SysEx Fader* on page 5-86, which explains how to



Chapter 10

The Event List

program SysEx faders. When programming SysEx faders, it is important point to remember that the arrows in front of and behind the word EOX are used to add or remove bytes.

SysEx Data in Hex Format

You can select whether SysEx data is displayed in the Event List or the SysEx fader editors in hexadecimal or decimal, using the **View > SysEx in Hex Format** command.

Hexadecimal digits are preceded by a \$ symbol in Logic.

Here are a couple of tips for the real power users among you:

Numerical input (by double-clicking) is always independent of the current display mode, or event type and can be used in many other areas of the program. Here are some of the possible methods. You can:

Format	Examples
Decimals	1, 01, 2, 3, 4, 127, . . .
Hexadecimals	\$1, \$01, \$2, \$3, \$A, \$0A, \$7F
Notes	"C3", "C#3", "Cb3", "C##2" (equivalent to D2), "Dbb2" (equivalent to C2)

If you double-click on the note "E3#", you can enter a decimal value such as "64" or a hex value of "\$40" instead of the note itself.

In many parts of Logic, mathematical operations can also be used to change values, for example:

Function	Example
adding two existing values	" +5", "+\$10"
subtracting from t h e existing value	" -5"
summing two values	"38+17"
subtracting two values	"38-17"
multiplying two values	"7*8"
dividing two values	"80/5"

Meta Events

POSITION	STATUS	CH	NUM	VAL	LENGTH/INFO
1 1 1	1 Meta	1	48	0	Switch Fader
1 1 1	1 Meta	1	49	0	go Screenset
1 1 1	1 Meta	1	50	0	Song Select
1 1 1	1 Meta	1	73	0	Arpeggio
1 1 1	1 Meta	1	99	0	Fader Bang
1 1 1	1 Meta	1	100	0	Tempo Control

Meta events are not MIDI events, but are control messages specific to Logic. They are used to automate specific Logic functions, and organize objects in the notation which cannot be represented by MIDI events.



To create a meta event, hold down the  key and click on the Full Message button.

Num and Val

Num determines the function of the Meta Events you create, and *Val* is the value that is sent. In the Event List window you should only ever insert and edit the following Num values:

- Num = 47 Send Byte to MIDI. This sends the track instrument any byte value (*Val*) between 0 and 255 (\$00-\$FF). As an example of how to use this Meta Event, if you send 246 as the byte, this is equivalent to a MIDI tuning request message. The display will show “Send Byte \$F6”. Only use this Meta Event if you know what you’re doing—if you don’t, your sound modules and synths may start to behave very oddly indeed..
- Num = 48 Switch Fader. This will send MIDI events to a particular output number (*Val*) on a cable switcher. You have to first connect a cable between a track instrument (e.g. “M-Playback”) and the cable switcher in question.
- Num = 49 Goto Screenset. This event calls up a screenset (*Val* determines the number). The track instrument is irrelevant.

- Num = 50** Song Select. This event will switch songs on a MIDI data file (Val = the stored song number), if you have one connected. The track for this event is irrelevant.
- Num = 51** Goto Marker. When this event is sent, playback will jump to another marker (*Val* determines which marker number). Once again, it doesn't matter which track this event appears on.

Meta Events can also be generated by Fader objects in the Environment window. There's more about this in the chapter on the Environment (check out the section *SysEx Fader on* page 5-86).

10.5 Event Float Window

The event float window can be accessed from most other windows.

Select **Options > Event Float...** (or the key command **Open Event Float**) to open an event float window. This gives information on the currently-selected object, and can be compared to a single line of the Event List. You can edit all the parameters in this window.



The following parameters are displayed:

the start point of the selected object in bars, beats, divisions, and ticks,

the type or name of the selected object,

the recorded MIDI channel and first data byte (if the selected object is an event) - for notes, the name-and, if there is one, the second data byte -

length of the selected object in bars, beats, divisions, and ticks.

Clicking on the film symbol (on the extreme left) toggles the position and length display between SMPTE time and the normal display.

Holding down the  key while clicking on the SMPTE symbol turns the “MIDI Out” button on or off, and with it the monitor that gives MIDI playback in the Event float window).

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The Event List

Chapter 11

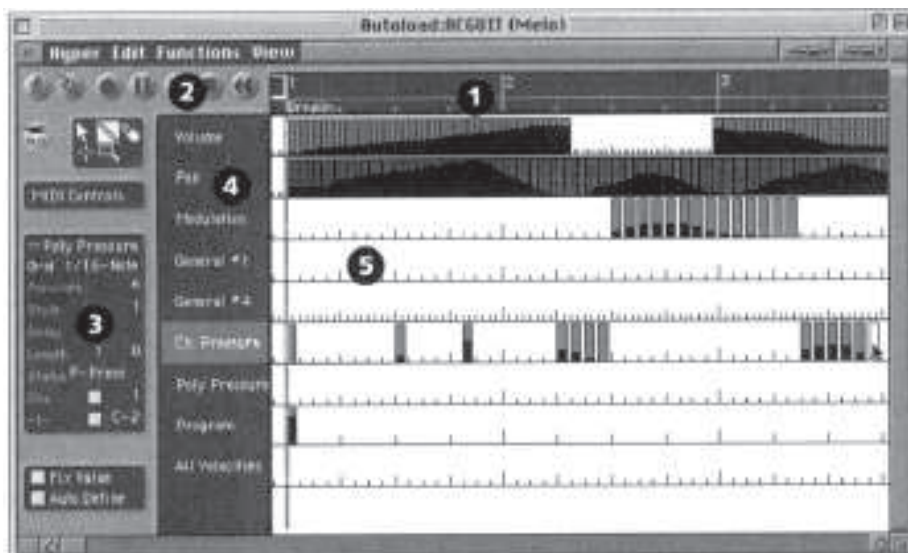
The Hyper Editor

Usage

The Hyper Editor has many uses. The main ones are creating and editing drum sequences, and graphically editing controller data. Despite some similarities, don't confuse the Hyper Editor with the *Hyper Draw* function.

Opening the Hyper Editor

To open a Hyper Editor window, select **Windows > Open Hyper Edit** (*open Hyper Editor*, default  **5**). You will then be able to see the contents of the sequence selected in the Arrange window.



Structure

The structure of the Hyper Editor window resembles that of the Arrange window: there is the horizontal bar ruler at the top (1) and the optional transport panel (2) in the top left corner. On the left is the optional parameter area (3), next to it the

event name column (4) and to the right of that the editing area (5), which can be likened to the area containing the tracks in the Arrange window.

The individual lines in the Hyper Editor are functionally similar to the tracks in the Arrange window, in that each line has an event definition, which determines the event type displayed in this line (much as tracks in the Arrange window contain an instrument). When you select a line in the name column, its event definition is shown in the parameter box.

The display of the events takes the form of a horizontal row of vertical beams, whose height indicates the value of the relevant event. You can directly alter the value by grabbing the beam and sliding the mouse up and down. It is possible to store a combination of simultaneously displayed event definitions as a hyper set.

11.1 Hyper Sets

A hyper set is a way of saving a combination of event definitions. When a hyper set is saved, the vertical zoom setting of the Hyper Editor is stored (this determines the number of event definitions that can be displayed simultaneously). You can save as many hyper sets as you need in a song.

Selecting a Hyper Set

Directly above the event definition parameter field is the hyper set selection field. Grabbing it opens a pull-down menu, which allows you to select from among the available hyper sets.

Creating a Hyper Set

To create a new hyper set, select **Hyper > Create Hyper Set**. The preset event definition in the new hyper set is the volume controller (#7). Any alterations that you make, such as adding or redefining event definitions, or adjusting the vertical zoom



setting, are automatically stored in the current hyper set (as with screen sets).

Create GM Drum Set

General MIDI has an established keyboard assignment pattern for drum and percussion sounds. This is often even the case with sound modules, samplers and professional synthesizers which do not carry a GM, GS or XG emblem. Since the Hyper Editor is ideally suited for the editing of drum tracks, you may from time to time wish to create a Hyper Set which is specifically tailored for GM Drum Sets, including the correct Names and Hi-Hat-Modes. Select **Hyper > Create GM Drum Set** to do so.

Naming the Hyper Set

To name the current hyper set, double-click on the hyper set selection button.

Clearing a Hyper Set

To clear the current hyper set (to delete its contents), select **Hyper > Clear Hyper Set**.

11.2 Event Definitions

The event definition of a hyper edit line determines which events it displays. This selection usually affects the status byte and the first data byte of the event. The second data byte then contains the adjustable value, which is represented by the height of the beam. For example, in an event definition line, the value of a controller or the velocity value of a note is shown as a beam. Don't worry if this seems a bit confusing at first: when setting up the hyper definition parameters, Logic helps you by providing pull-down menus containing written definitions of the status byte and (as far as possible) the first data byte. There is an even simpler way to set up hyper definitions: you

can automatically create event definitions for selected events - see below.

There are many ways of altering the way the beams are displayed, and adjusting them to the particular event types. You can also use a grid to align the display of existing events, and add new events. This grid can be set separately for each event definition in a hyper set. The height of the lines in the hyper set is adjusted using the Hyper Edit window's vertical zoom function.

Selecting the Event Definition

As with tracks in the Arrange window, you can select an event definition by clicking on the name column. This allows you to view its parameters in the event definition parameter box. Unlike with arrange tracks, however, it is possible to make a multiple selection.

Creating an Event Definition

When you select **Hyper > Create Event Definition**, a new event definition is added, at the position of the currently selected event definition. Initially, it has the same parameters. The event definitions beneath it are moved downwards.

Automatically Creating Event Definitions

If you select an event in another opened editor window, you can automatically create an equivalent event definition by checking the *Auto Define* checkbox. You can switch this function on/off with the *Auto Define Toggle* key command.



If the current hyper set already contains an event definition which corresponds to the type of the selected event, Logic does not create a superfluous double definition. Instead, this event definition is moved to the visible area.

Don't forget to switch off Auto Define immediately after completing input.

Creating Several Event Definitions Automatically

You can also automatically create several event definitions. Just select the events on which you want to base the event definitions. You can use the same function to create event definitions for all the event types in the selected sequence.

Before creating a group of new definitions, it is a good idea to create a new hyper set first.

If you select **Hyper > Multi Create Event Definition...** a dialog box appears asking you whether you want to create event definitions for all event types (select *All* or hit *Return*) or just for the selected events (Selected). Press *Cancel* to abort.

Deleting an Event Definition

Select **Hyper > Delete Event Definition** to remove the selected event definition.

Hyper Sets: Copying Event Definitions

Select the event definition that you want to copy, and select **Hyper > Copy Event Definition**. Switch to the destination Hyper Set, and select **Hyper > Paste Event Definition**.

Converting Event Definitions

You can redefine an event definition with all its events. The values of the events are retained, but the event type is changed to the new event definition. Select **Hyper > Convert Event Definition...** , or double-click on the name column of the event definition which you want to convert.



The box shown above appears. On the left (convert), you can see the parameters of the selected event definition, and on the right you can define the parameters of the destination event type. The current settings are used as default values.

If you place an "x" in the *Quantize Events* checkbox, the event positions are quantized according to the Grid setting in the right-hand parameter field. If you make no more alterations, this allows you to just quantize the positions of the events. If there is any *Delay* value on the right side, this is taken into account by the quantization.

Sorting Event Definitions

To change the vertical order of the various event definitions, just grab the event definition you want to move by its name column, and drag it to the desired position.

Event Definition Parameter Box

The event definition parameter box is where you make the settings for the currently selected event definition line. The most important parameters are *Status* and *-1-* (page 11-9).



Opening the event definition parameter box.

You can show (or hide) the entire left parameter area by checking (or unchecking) **View > Parameters**. You can close or open the parameter box by clicking the triangle in the top left corner.

Name of the Event Definition

By clicking the name next to the triangle, you can determine what appears in the name column. If you define a named MIDI controller, or a note from a mapped instrument in the Status and **-1-** parameters, the relevant name will be offered as a preset. Altering a note name in the event definition also alters the corresponding note name in the mapped instrument.

You can also display, add, and edit Meta Events in the Hyper Editor's event definition.

Display and Editing Grid

The Grid parameter is set with the typical pull-down quantization menu. New events can then be added at the set grid positions. The positions of existing events are not affected.

If you want to quantize the positions of existing events, you can do so using event quantization (see page 1-40), or more simply with event definition conversion (page 11-5).

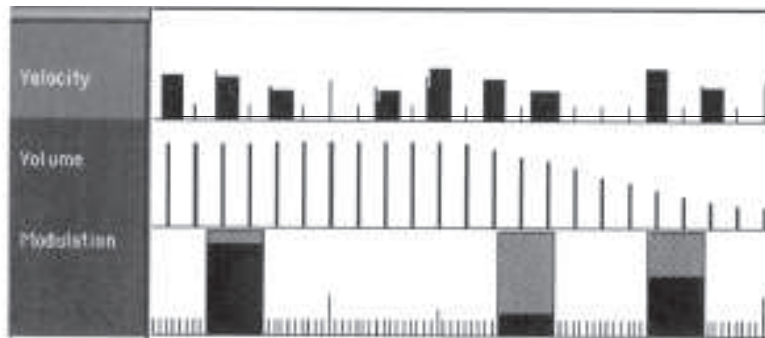
The same quantization templates are available as for the playback parameters, including the Groove Templates you define yourself.

When editing complex drum rhythms, it can be useful to create several hyper definitions for one drum note, each with a different quantization grid. For example, if there are two lines for a snare drum, one with a 1/16 and the other with a 1/96 grid, you can use the pencil to add individual hits in the coarser grid, and rolls in the finer grid.



Beam Width

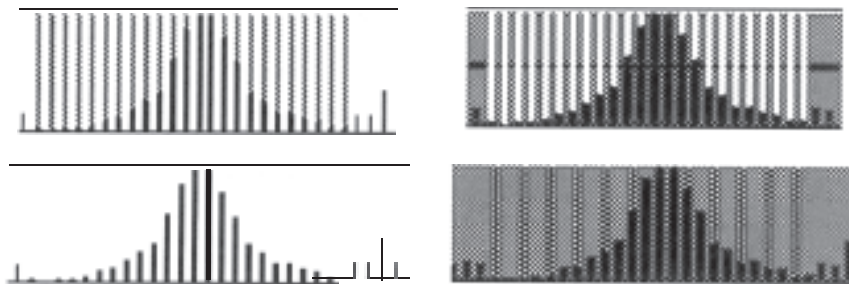
You can set the width of the event beams from 1-16, using the *Pen-width* parameter. Regardless of the beam width, the exact event position is always marked by the left edge of the beam.



When set to the maximum value (*Penwidth* = 16), note events are displayed with their actual length.

Beam Display

The *Style* parameter allows you to select from four different color display variations for the beams. Styles 5 to 8 are the same as 1 to 4, but the selected events flash when you highlight them.



Delay

The *Delay* parameter can be used to delay (with positive values), or advance (negative values) all the events in an event definition line by an adjustable number of ticks. The change in position is visible immediately. Even new events are offset from the grid positions by the *Delay* value. If you click between the word *Delay* and the parameter value, you can use a pull-down menu to enter note values (e.g. 1/16 etc.) directly.

Unlike the sequence playback parameters' delay option, this affects only individual event types or note numbers, which is very useful for drum programming.

In general, it is useful to transmit controller data slightly before or after the exact grid positions, to improve the timing of the notes lying on the grid positions.

Length of Notes to be Added

You can use the *Length* parameter to set the length of notes to be added, measured in divisions (the left number), and ticks (the right number).

To guarantee optimum timing when drum programming, you should ensure that note off events are never transmitted at the same time as note on events. Use note lengths which are not close to straight note values, e.g. 100 ticks 1/48 note = 80 ticks, 1/32 note = 120 ticks). Very small tick values are also unsuitable, because the note off events may sometimes be transmitted at the same time as note on events at the same position.

Event Status

If you grab the right side of the *Status* line, a pull-down menu opens, where you can set the event status for this event definition line.

MIDI Channel

Next to *Cha*, there is a checkbox, and to the right of this you can set a MIDI channel number. If you check this box, the display is restricted to events on the set channel. If the box is unchecked, the channel parameter is ignored, and matching events on all channels are displayed.

First Data Byte

In the bottom line is the -1- parameter, where you can determine the first data byte of the displayed event.

For example, if Note is set as the event status, you can determine here which pitch (note #) should be displayed in this event definition line. If the edited sequence is played by a mapped instrument, a pull-down menu appears here with the names of the input notes (drum sound names).

As with the MIDI channel, the checkbox determines whether (if checked) only events with the defined first data byte should be displayed, or whether (if unchecked) the setting should be ignored. In the case of note events, the velocity values of all notes (regardless of pitch) would be displayed in the relevant event definition line.

If **Control** is set as the event status, you can use a pull-down menu to determine the controller type (controller #).

With channel aftertouch (status **C-Press**), or program change events (status **Program**) the setting of the -1- parameter is always ignored, since the first data byte is displayed as a beam height. With pitch bend data (status **PitchBd**), the setting is also ignored, because both data bytes are used to display the beam height.

Making Different Definitions Simultaneously

Just as you can adjust parameters at once for several sequences, you can set the parameters of several event definitions simultaneously. Select several event definitions in the name column, by holding down  and clicking on them. You can also use **Hyper > Select All Event Definitions** to select all event definitions, and then deselect some of them by holding down the  key and clicking on them.

Any alterations which you make in the event definition parameter box affect all selected event definitions absolutely.

Hi-Hat Mode

In hi-hat mode, event definitions can be gathered together in groups within which a only one event from each group can be

played at any time position. A typical use is for preventing different hi-hat notes (open, closed, pedal .) from being inserted at the same position.

To define a hi-hat group, click in any of the lines at the left edge of the name column. Click here once again to switch hi-hat mode off for that line. All lines in a hi-hat group must be vertically adjacent to each other.

If you add an event within a hi-hat group, all existing events at this time position are deleted.

You can create as many hi-hat groups as you want in a hyper set, but they must all be separated by at least one line in which hi-hat mode is switched off.



11.3 Operation

The grid defined by the Grid parameter in the event definition parameter box is very important:

During selection, all events within a grid section are selected

When you change the values of existing events, all events within a grid section are altered, and

New events are added at the grid positions, in accordance with the value set as the *Delay* parameter

Selection Techniques

You can use any of the usual selection techniques, as described in the section Selection Techniques on page 1-23. However, there are the following differences:

To select individual events hold down the **Shift** key, as you would when selecting several events that are scattered across the screen, and;

When making a rubber band selection, hold down the **Shift** key (be careful not to click any events).

Moving and Copying Events

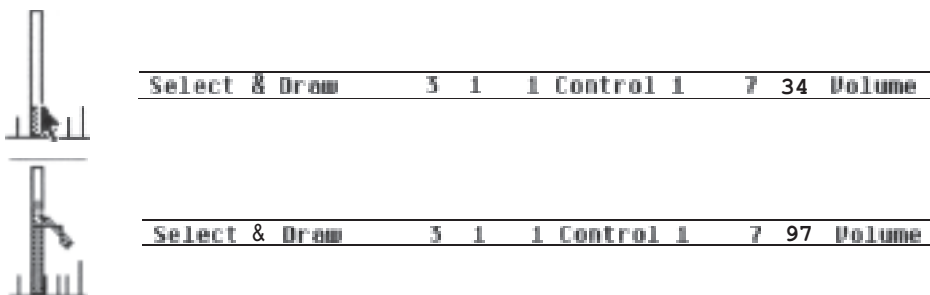
When moving selected events, you also have to hold down the  key.

To copy events, use the  key as usual.

With both operations, you can move events to other event definition lines. The events will be automatically converted.

Altering Event Values

You can alter individual values by grabbing the event beam (with the mouse pointer or pencil). As you change the event value, the current value is shown in the top line.



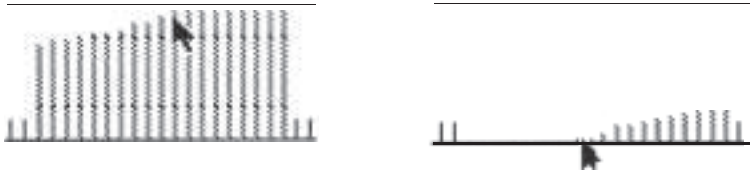
Altering Several Events



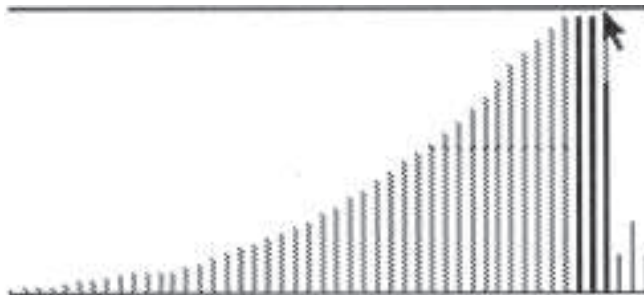
If several events are selected, you can alter all the values relatively, by grabbing just one of these event beams. The absolute differences between the event values remain the same. If one of the beams reaches the top or bottom, you will not be able to go any further in that direction.

However, if you hold down the  key as well, you can keep raising or lowering the values of the selected event group, until the event beam which you have grabbed reaches the top or bottom.

Operation



Setting up a Manual Series



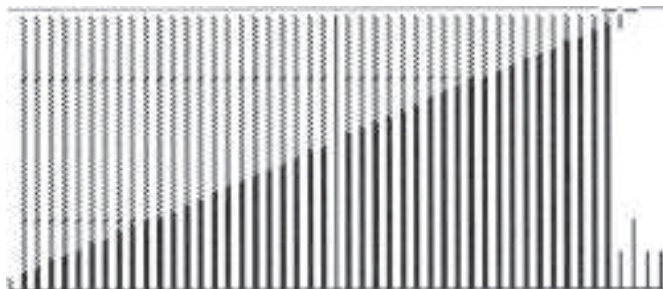
Existing Events

Draw the events with the mouse pointer, while holding down the mouse button.

Creating New Events

Draw the events with the pencil, while holding down the mouse button. The pencil is preset as the second tool ( key).

Setting up a Linear Series



Select the crosshair tool.

Click anywhere in the beam display area, and hold down the mouse button.

Keep an eye on the display showing the exact position and event value in the top line. Move the mouse to the start point of the line you want.

Release the mouse button.

Move the mouse to the desired end point of the line (to the right or left). You can also keep an eye on the position and event value in the top line.

Click to align the event beams along the line.

If there were already events in the area between the start and end points, they are aligned along the line.

If there were no events in the region, new events are created at the grid positions (in accordance with the Grid parameter).

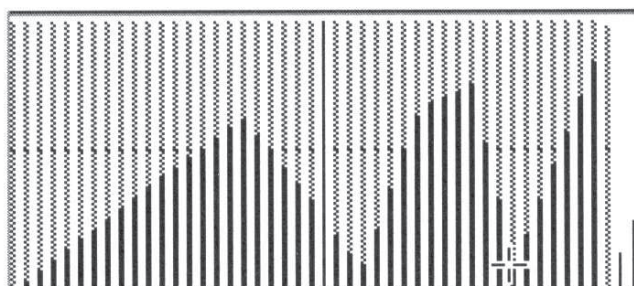
If you definitely want to create new events, hold down the  key when you click for the last time.

This creates one event at every grid value. With very flat slopes, or very fine grids, the individual grid positions always remain unoccupied if the value to be added there is the same as the value of the previous event. This reduces the data-flow along the MIDI bus, without reducing the resolution of the controller.

When defining the Grid parameter to add successions of controllers use the motto, “as coarse as possible and as fine as necessary” to keep down the data output. Logic can deal with very large amounts of data, but unfortunately the same cannot be said of MIDI.

Linear Series in Sections

The last time you click (regardless of whether or not you hold down the  key to add new events) if you also hold down the **Shift** key, you can immediately draw another line from the end point of this line.

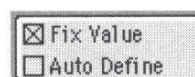


Adding Individual Events

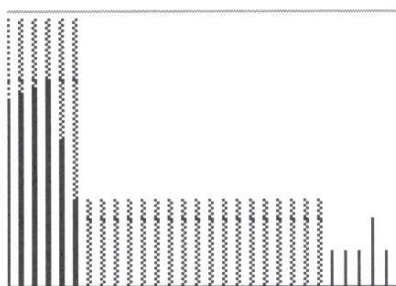
Use the pencil to add individual events. You can alter the added value even before you release the mouse button. The pencil is preset as the second tool ( key).

Fix Value

If you place a check in the *Fix Value* checkbox, you can prevent the height of any event beam being altered with the mouse pointer or pencil.



When adding events with the pencil, all the added events are given the value of the previously-selected event. This allows you to draw a succession of events with the same value.



By selecting an event with the pencil, you can adopt this value as a preset, because in fix value mode it is impossible to alter the value by clicking on it.

When adding events with the crosshair tool, the preset value is always used as the start point of the line.

Chapter 11
The Hyper Editor

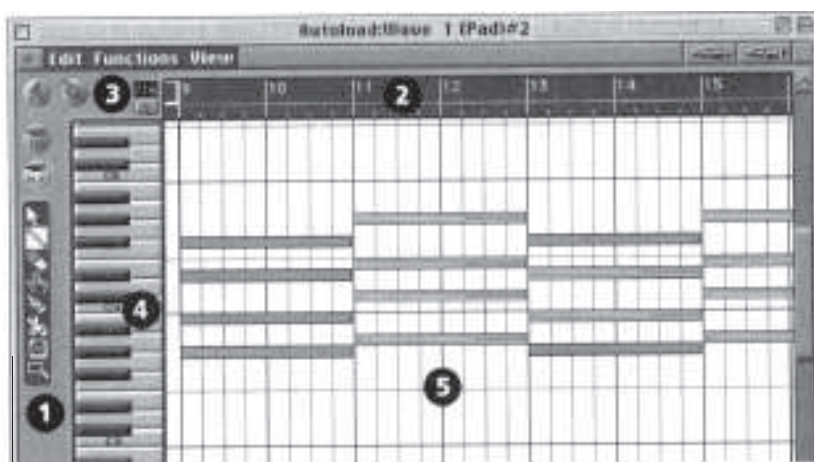
Chapter 12

The Matrix Editor

The Matrix Editor is used to graphically edit note events. Its advantage over the other editors is its ability to provide a more precise display of the length, position and velocity of the notes. The Matrix Editor is designed for fine-tuning the positions and lengths of notes.

Opening the Matrix Editor

To open a Matrix Editor window showing the contents of the selected sequence, select **Windows > Open Matrix Edit** ( ) or a user-defined key command (*Open Matrix Editor*).



The optional parameter field **1** contains the standard buttons and the toolbox. Like the Arrange window (and the Hyper and Score Editors), there is a Bar ruler at the top edge **2**. To the left of this (just as with the Transport window), you can define the sub-division **3**. The vertical screen keyboard **4** indicates the pitch of the notes, which are displayed as beams **5**.

12.1 Display

Notes are the only events shown in the Matrix Editor. The beams (and the way they can be edited) are very similar to the sequences in the Arrange window.

Pitch/Note Names

The pitch is indicated by the vertical keyboard on the left side. To help you with positioning, there are horizontal lines running across the screen between notes B and C, and notes E and E

Remember that you can adapt the octave numbering to Roland/Korg devices (page 17 - 19).

If a mapped instrument sequence is being displayed, the *names* of the notes being played will appear on the vertical keyboard instead of the notes "C" and "F".

When a mapped instrument note is being moved, its name is displayed in the Info Line.

Position

You can read the position of a note in the bar ruler. The background is marked by vertical lines to assist with positioning:

A straight line at every bar,

A dashed line at every beat, and

A dotted line at every division (you can alter the division value in the Transport window, or up on the left in the parameter area).

Velocity

The velocity of a note is portrayed by a horizontal white line in the note. The length of the line in relation to the total length of

the note corresponds to the velocity value of the note (in relation to the maximum value of 127).

In addition, if **View > Sequence colors** is disabled, the Matrix Editor now shows the velocity of notes by using different colors.

This makes it very easy to see at a glance what range of velocities are contained in a sequence.

If you change the velocity of MIDI notes with the V tool, you will notice that the color of the notes change as you do it.



Sequence Colors

When **View > Sequence Colors** is activated, the notes will be displayed in the color of the sequences that contain them. In this way, as a rule, the Notes can be edited more easily.

Background

By selecting **View > Change Background**, you can switch to a very attractive custom Emagic background. However, with this type of display, there are no vertical dotted lines marking the division positions.

Other Display Options

All other display options are covered in the section *Display Functions* on page 1-42. In the Matrix Editor, you will often want to move the visible section (using the scroll bars), and alter the display zoom (with the telescope buttons and magnifying glass).

12.2 Editing Notes

You can use the same intuitive editing functions as for the sequences in the Arrange window.

There are also some editing options which are available in all editor windows. These are covered in the section *Edit Functions* on page 1-33.

Creating Notes

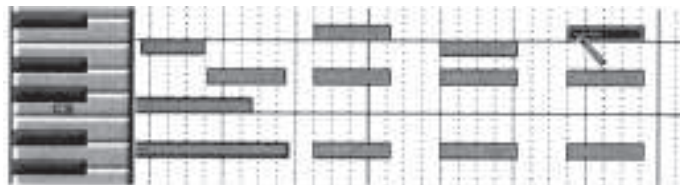
To create a note, click with the pencil at the desired point in the background.



Duplicating Notes

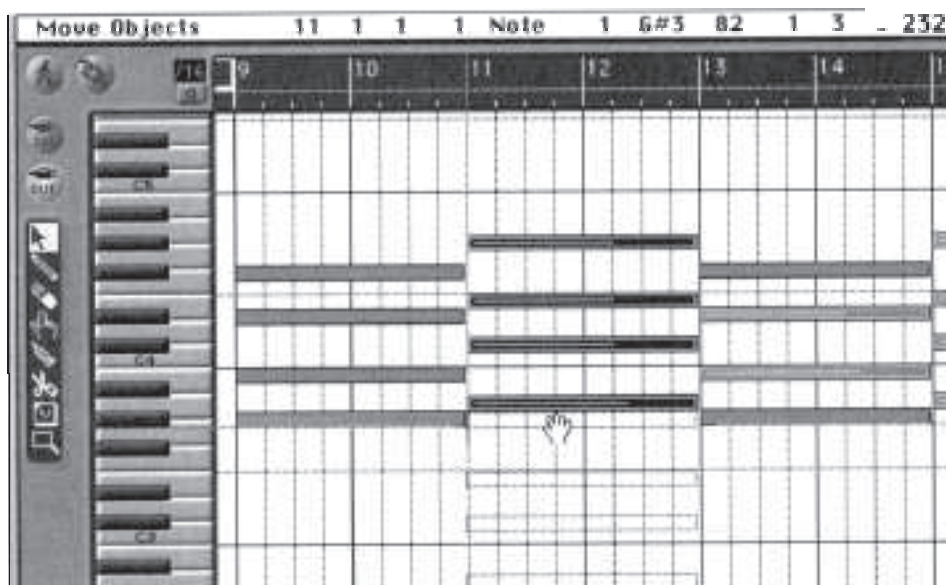
To copy an existing note to another position or pitch, first click the original note with the pencil (near the middle).

Now, any notes which you create by clicking the background will have exactly the same length and velocity as the original note.



Moving Notes

You can move selected notes by grabbing them (near the middle) and dragging them. If you move notes vertically, they will be transposed, and if you move them horizontally, they will be moved in time. While you keep the mouse button held down, the target position and pitch are shown in the Info Line at the top.



When you move notes horizontally, they snap to the division positions (you can alter the division setting whenever you want).

Remember that you can limit movement to one direction to avoid accidentally transposing notes when moving them vertically (and vice versa). (**Options > Settings > Global Preferences: Limit Dragging to one direction in Matrix and Score**).

Setting a Finer Grid

When moving notes in the Matrix window, you can make fine adjustments at a high resolution, by holding down the **ctrl** key as you drag. The exact resolution will depend on the current Zoom setting of the window.

If you hold down the **ctrl** + **home** keys, you can move the notes in tick steps, completely independently of the zoom setting and the time grid.

Copying Notes

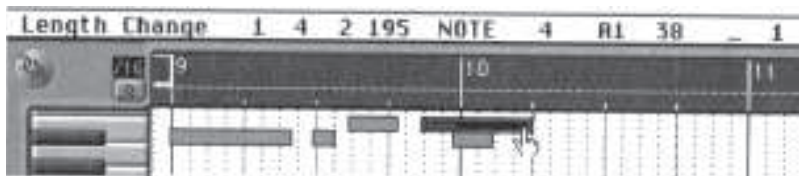
Hold down the **ctrl** + **alt** key as you move the notes.

This allows you to copy or move notes between two Matrix Editor windows, even if the windows belong to different songs.

Of course you can also copy (**Edit > Copy** or ) or move (**Edit > Cut** or ) the notes onto the clipboard, and then paste them at the current song position, with their original pitch (choose **Edit > Paste** or ; see page 1-28).

Altering Lengths

To alter the length of a note, grab it by its bottom right-hand corner, and drag it to the required length. While you are altering the length, the Info Line will keep you informed of the precise length of the note.



Altering the Lengths of Several Notes Simultaneously

You can adjust the lengths of several selected notes at once (e.g. a chord), by simply altering the length of one of them. The relative differences in lengths of the selected notes will be maintained.

Making Notes End Simultaneously

Hold  while altering the length of a note, to make all selected notes end at the same time.

Making Notes the Same Length

If you want to make several selected notes the same length, hold down   (as in the Event List, when making several parameters the same value).

What Tools to Use

When altering note lengths, you can use either the mouse pointer or the pencil. However, it is better to use the pointer,

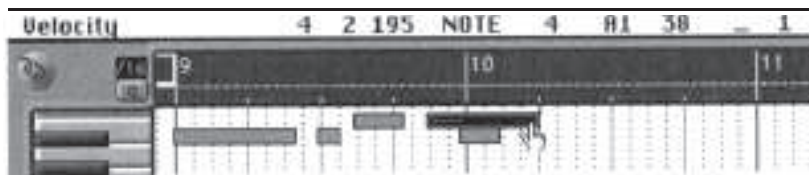
because you might accidentally create new notes with the pencil.

With very short notes, it can sometimes be difficult to grab the bottom right corner. In this case, you should use the index finger tool, which allows you to grab notes anywhere to alter their length.



Altering the Velocity

To alter the velocity value, click on the note with the V tool. The Info Line will then indicate the velocity value of the note you have clicked on.



If you hold down the mouse button, you can alter the velocity by vertically moving the mouse. If the MIDI Out function is switched on, the note will be output every time you alter the velocity.

Altering the Velocity of Several Notes

All selected notes can be altered simultaneously; the differences in the velocity values will be retained. If the velocity value of one of the selected notes reaches an extreme value (0 or 127), you can't go any further. However, by simultaneously pressing the **Alt** key, you can carry on altering the velocity values until the clicked note reaches an extreme value.

To give all the notes the same velocity value, hold down the **Alt + Shift** keys as you alter the value.

12.3 Folders

The Matrix Editor can display all sequences from a folder, or those for an entire song, simultaneously.

Changing Presentation Layers

Display of all Sequences

Double-clicking on the background, or by clicking on the window close symbol, changes the Matrix Editor to show all sequences. The beginning of each sequence is shown by a vertical line, in the color for that sequence.

A Matrix Editor window with activated (Same Level) link function (single click on the chain link button) shows all the sequences in the folder shown in the Arrange window, or songs.

Hide Notes of Non-Selected Sequences

By selecting **View > Show selected Sequences Only**, you can limit the display to notes in sequences that are selected.

Display of Individual Sequences

Double-clicking on a note changes the display level again, and will show only the sequence containing that note.

A repeated double-click on a note opens an Event List window (with : Score Editor).

A Matrix Editor window in Show Contents link mode (double-click on the chain link button), shows the contents of a single sequence selected in the Arrange window.

The following text assumes that all sequences are displayed.

Selection

You can create a selection over multiple sequences, move events, lengthen, or apply functions as required.

Selecting a note in the Arrange window causes the relevant Sequence to be chosen.

Info-Bars

When an individual sequence is represented, the Info-Bar will show the number of selected Notes.

When all sequences from a folder or song are displayed, the Info-Bar shows, next to the number of selected Notes, the number of sequences, from which the Notes are selected. For instance:

"8/2" 8 Notes are selected from 2 Sequences.

When notes from only one sequence are selected, the sequence name is shown next to the number of selected Notes. For instance:

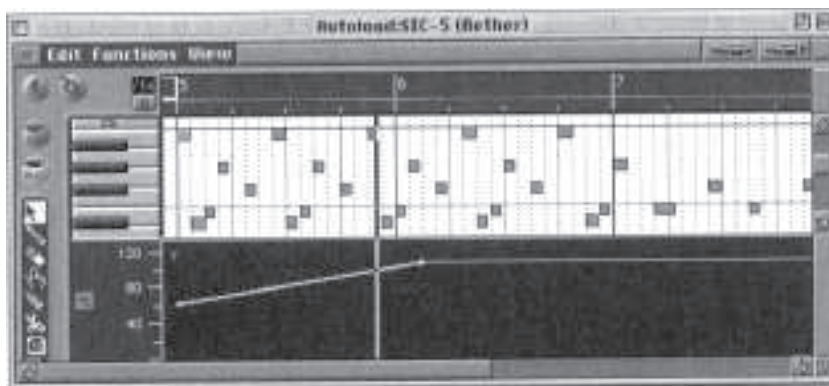
"3/Piano" 3 Notes are selected from the sequence "Piano".

When no Notes are selected, the Info-Bar shows the name of the last selected sequence. Using the Pencil will insert Notes into this sequence. For instance:

"Piano" No notes have been selected. Using the Pencil will insert notes into this sequence.

12.4 HyperDraw in the Matrix Editor

In the Matrix and Score Editors you can use the **View > HyperDraw...** command to open a HyperDraw display. This allows quick and easy editing of controller data in relation to note data.



Once you have opened the HyperDraw display, you can select the type of event you want displayed, using the arrow menu on the left.

You can adjust the size of the HyperDraw display by grabbing, and vertically moving the dividing line.

Hyper Draw-Functions

Quick Delete

You can delete all events displayed in Hyper Draw for a sequence by mm-double-clicking in the blue HyperDraw area.

Adding new points

Just click on the blue background of the Hyper Draw display - you can use either the pointer or the pencil, and a point will appear. As long as you hold down the mouse, you can move any of these points. The position and value are displayed in the Info line at the upper edge of the Arrange window.

No events will be inserted into the sequence until you release the mouse button.

Deleting points

Click briefly on any point to delete it.

Moving an existing point

Just grab a point with the mouse and move it. If you hold down the mouse button while you do this, you can restore deleted curve points, by dragging the mouse over where they used to be.

Fine-tuning curve points

While you move a point, hold down the  key. You will then be limited to moving the point vertically, but be able to select all 127 possible steps.

Deleting

You can delete selected notes by pressing the  key, or clicking on them with the eraser.



Selection Techniques

In addition to the usual selection options, you can select all notes of the same pitch throughout the whole sequence by clicking the corresponding key on the screen keyboard.

If you hold down the mouse button, you can also draw (rubber-band) a pitch range over the screen keyboard, and select all the notes within it.

Don't forget the specialized selection options available from the **Edit** menu (or key commands). These are covered on page 1-27.

Quick Opening of Event or Score Editor

A double click without any modifier opens the event editor. A double click while holding  opens the score editor.

12.5 Functions

For a description of all the other functions of the Matrix Editor, such as automatic length correction, and selective deleting copying, please refer to the section *General Functions of the Editors* on page 1-29.

Chapter 12
The Matrix Editor

Chapter 13

The Score Edit Window

Before getting started with this chapter, we would like to remind you that there is a Logic Score Tutorial on the Emagic Tutorial CD, which is shipped with every Logic Audio Platinum package. This tutorial provides an ideal way of getting to know all the basic functions, especially if you have never used Logic? Score Editor before. The tutorial uses a very practical, task oriented approach. Each chapter is complemented by a Logic song file, where you can work step-by-step following the description in the text. The results can be printed out, and are also audible via MIDI.

Preface

Logic's Score Edit window basically serves two purposes:

1. As a MIDI editor for users who prefer to use traditional music notation as an interface for MIDI sequencing. Input, editing, copying or deleting of MIDI events and changing their musical position can be done directly in the Score window. One can also change between different display levels. (i.e., view single or multiple parts at once)
2. To prepare and produce printed music. The number of staves in a score is only limited by the paper size. Also, each staff can contain up to 16 independent ("polyphonic") voices.

In addition to the notes (which always represent MIDI events), numerous musical symbols can be inserted. Text (and thereby any symbol which is part of a text font) can also be integrated into the music.

Single pages, or parts thereof, can be exported to graphic or text based programs, using the camera tool. The selected area can be saved as a graphic file, or copied to the clipboard, and then imported or pasted as a picture in another program. This is especially useful for preparing teaching material.



MIDI sequencing and preparing music for printout are closely related in Logic. Recorded MIDI notes are immediately displayed in the score. On the other hand, notes which are inserted with the mouse on the screen, or changes to existing notes are immediately reflected in the MIDI playback.

How this Chapter Is Organized

This chapter begins with a few survey sections on the following topics:

- The Score window itself.

- The general concept of Logic's notation facilities.

- The different methods for input.

- The various display options.

- The procedure for printing a score.

Following these are detailed sections on input and editing methods (13.9 to 13.14), and display options (13.15 to 13.20).

The last two sections cover text and chord symbols.

If you are looking for a particular function or detail, please use the index at the end of the book to find the corresponding description. Please note that in the index, many score-related topics can be found under the keyword "Notation".

Please keep in mind that all parts of Logic work together. Many functions are available in more than one window. This is especially true of the functions described in the section General Functions of the Editors on page 1-29 which are very useful for working in the Score window.

Opening the Score Edit Window

There are several ways to open the Score window:

Main menu **Windows > Open Score**

Key command *Open Score Window* (Default:   **O**).

Double click on a sequence in the Arrange window, if *Double Click Sequence to Open Score in Arrange* is selected in (main menu) **Options > Settings > Global Preferences**.

To recall an already opened window (which is hidden by other windows), select that window's name in the main **Windows** menu, where all currently opened windows are listed.



Several Score windows can be open at once. This can be used for example to display different groups of instruments, by using different Instrument Sets.

13.1 Elements of the Score Edit Window

Here's an overview of the Score Edit window's elements and their names, as used in this manual.



Detailed descriptions of the terms described below can be found later in this chapter. Tool Box, Bar Ruler, Zoom symbols, the switches for Catch, Link, MIDI-In and MIDI-Out and the different functions of the Edit and Functions menus can be found in chapters 4 and 8, since all these exist in other windows as well. The functions found in the other menus are all described in this chapter.

Chapter 13

The Score Edit Window



Along the left-hand side:

- 1** Buttons for Catch, Link, MIDI In, MIDI Out and Page Edit view
- 2** Instrument Set Box
- 3** Display Parameter Box
- 4** Event Parameter Box
- 5** Toolbox
- 6** + **7** Partbox

Along the top margin:

Bar Ruler (linear view only)
 Zoom symbols
 Menus

The main working area: **8**

optionally with:

Instrument name display column on the left side

Hyper Draw area at the bottom (only possible in single sequence view and with linear display)

Page Rulers (only in page view)

The various options for changing the display of these elements are described in the section *General Appearance of the Score Edit Window* on page 13-13.

13.2 Logic's Notation Concept

MIDI sequencing and score display are closely related within Logic. Every note in the score represents a MIDI note event. MIDI sequences are interpreted for display by Logic, using methods that can be influenced by the user to a great extent. The goal is to retain the feel of real time recordings, while still rendering a readable score.

If notes are put into the score using the mouse or step input, they are also played back via MIDI, since they represent MIDI events.

Basics

Recorded sequences are automatically displayed in the Score window. If you like the sound of the playback, but not the score display, you can usually get the desired result by changing some of the settings in the *Display Parameter Box*, in the different *Song Settings* dialog windows, or *Note Attributes*. You can control the display even further by selecting or creating Score Styles, *Instrument Sets*, and *Text Styles*, which best fit your music.

If you generally prefer mouse input, you first have to create one or more empty sequences in the Arrange window, using the Pencil tool. These will then be displayed as empty staves in the Score window. Rests are usually displayed automatically in Logic, and don't need to be inserted.

If you want to use the mouse to insert new notes into a previously recorded sequence, you can select a rhythmic note value

from the Partbox, and drag it to the desired bar and pitch position, in any of the visible staves.

No matter which input method you decide to use, the selected display settings will always be applied to the inserted MIDI events. This means for example, that a thirty-second note can only be displayed if the *Qua* parameter in the *Display Parameter Box* of the corresponding sequence is 1/32, or shorter. If that value is set to 1/8, an inserted 32nd note will be displayed as an eighth note (but will still play back as a 32nd). The display is also affected by the *Interpretation* setting in the sequence's Display Parameter Box. These settings can be different for each sequence, even for sequences on the same track. To use different display parameter settings in the same staff, just cut the sequence with the Scissors tool. Now you can assign different display settings to the resulting shorter sequences, although they will still be displayed as one uninterrupted staff. MIDI playback will not be affected by this procedure.

The Partbox contains many symbols which can be inserted into the score, in the same way as notes, by dragging them into position with the mouse. For details about this, please read the section *The Partbox* on page 13-32.

The printed result always corresponds to what you see in the selected Score window (Page Edit view must be activated). Layout work should be done in page and print view, *after* all the notes and most of the symbols have been recorded or inserted.

Staff Display in Relation to Position and Length of Sequences in the Arrange Window

Sequences in the Arrange window represent parts played by certain instruments (symbolized by the tracks), and are displayed as staves in the Score window. However, depending on which Score Style is used, an instrument's notes can also be displayed in a staff system containing two or more staves (such as a two staff system for a piano part).

Each of these staves or (in the Arrange window) tracks, can contain an (almost) unlimited number of sequences. If the sequences on one track directly follow one another in the Arrange window (with no gaps between them), they are displayed as one continuous staff in the score, unless you have limited the view to a single sequence, by engaging *content linked mode*. In linear view, you will still see clefs, time and key signatures at the beginning of each sequence, but these disappear as soon as you change to Page Edit view.

The Display Parameters for each of these sequences can be set independently. This allows you to influence the visual display of sequences recorded in real time in a very flexible way, especially concerning the display of rhythmic values. If necessary, sequences can be divided into several pieces with the *Scissors tool* (available directly in the Score window), allowing you to assign different display parameters to each resulting part, because each is now an independent sequence.

It is always very obvious which sequences are selected: Selected sequences are displayed with blue stavelines (unless this option is deactivated in the Score Preferences).

Logic's Basic Elements for Score Display

The following elements and their parameters form the basic concept for the score display in Logic. (Detailed explanations will follow later in this chapter)

Instrument Sets



Instrument Sets allow you to control which instruments will be displayed (and printed) in a score. You can include as many (or few) of the existing track instruments into a score as you like. The order of the instruments can be arranged independently of their order in the Arrange window. You can assign names and abbreviated names to each track instrument independently from the actual track instrument names in the Arrange window. These will then be

displayed in the score. You can also define which staves in the score will be connected with brackets and bar lines, to form groups of instruments in the score.

Any single instrument can quickly be extracted from the score, by selecting it from the flip menu which appears when you click on the Instrument Set Box, while holding down the -key. This way, you don't have to create an Instrument Set for every single instrument in order to produce parts. The automatic line breaks and the individual stave margins can be changed for each Instrument Set individually. (See the section *Instrument Sets* on page 13-111)

When you open a new Score window with several sequences selected in the Arrange window, an Instrument Set will automatically be created and displayed in that Score window, which only contains the instruments belonging to the selected sequences.

Score Styles

Each Score Style contains a collection of score settings. Their function could be compared to that of text formats in word processors and layout programs. Instead of text settings however, Score Styles contain the staff size, display transposition, stave distance, number of independent voices etc. So, instead of having to define all these settings each time a new sequence is created or recorded, any desired Score Style can be assigned to that sequence. Any number of Score Styles may be created and assigned to individual sequences. There are also a lot of Score Styles already included in the template files that come with Logic, and in Logic's default song. For details see the section *Score Styles* on page 13-77.

Text Styles

These are predefinable text formats for all types of displayed text. See the section **Text Styles** on page 13-146.

Display Parameter Box



The parameters in this box mainly affect the rhythmic display of the selected sequences. They can be different for each sequence. Some of these parameters can be overridden for individual notes, using *Note Attributes*. For details see the section *The Display Parameter*

Box on page 13-68.

Song Settings

Here you can set general display options, which affect the whole song. These include spacing between notes, page margins, line thickness, display of instrument names, bar and page numbers, and a lot more. For details see the section *Song Settings for Score Display* on page 13 - 123.

Note Attributes

These affect single notes or-when assigning them to a group of notes-all selected notes at once. For each note, the following parameters can be defined individually (for details see the section *Note Attributes* on page 13-105):

- Enharmonic changes.

- Settings for Interpretation and Syncopation, independent from the settings in the Display Parameter Box.

- Stem direction, tie direction, and beaming, independent from the settings in the used Score Style.

- Display as an Independent Note, which excludes the note from the measure's overall rhythmic calculation.

- Horizontal graphic offset from the default position.

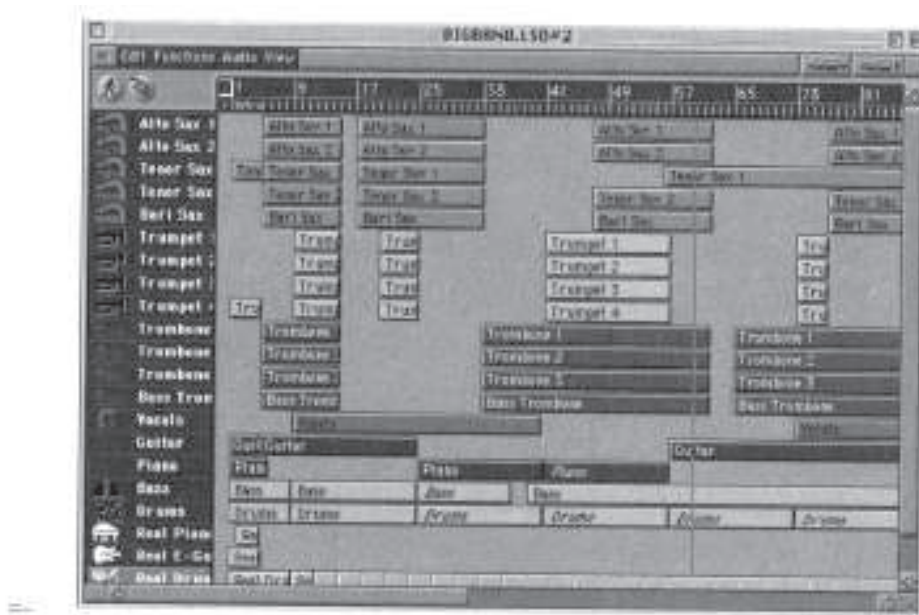
- Change of accidental distance.

- Change of note size, and form of the note head.

13.13 Adapting MIDI Sequences for Score Printout

If a song's MIDI sequences have been recorded in real time, the Arrange window often looks similar to the illustration below. Sequences are spread all over the place, with gaps between them. Sometimes, sequences belonging to the same instrument are on different tracks. Sometimes, *Aliases* and *Loops* are also used:

Please note: the following procedure is also described in the Logic Score Tutorial which can be found as a pdf file on the Emagic Tutorial CD (chapter 5.3). There, you can follow the description step by step in the included Logic song file ("Score Tutorial Song 5").



Reminder: The Score printout always corresponds to what is displayed in the active Score window. The staves which represent the sequences shown above will therefore be spread over the pages, with lots of space between them.

However, if you want to printout the arrangement shown above, you will need continuous staves for each instrument (track). This is achieved by changing the length of the sequences in the Arrange window in such a way that they

connect with one another on each track (except for those that are not intended for printout).

If you have used *Aliases* and *Loops*, you first have to select all these, and (in the Arrange window) use **Functions > Alias > Turn To Real Copies**, then **Functions > Sequence Parameters > Turn Loops To Real Copies** (both functions are also available as key commands), in order to convert these to “real” sequences, which - unlike aliases and loops - are displayed in the score.



In the Arrange window shown above, there is a looped sequence on a track called Real Drums (bottom track). Sequences like these hardly ever are written out in scores, so there is also another instrument (*Drums*) created for the score display Instruments which should not be displayed in the score (like Real Drums, Real Piano, Real Guitar in the above example) can be excluded by creating an Instrument Set without them.

All sequences which are intended to be displayed as one instrument have to be on the same track in the Arrange window. If these sequences are still on different tracks, either -click them all, or on all track names containing them, to select them. When all the sequences have been selected, hold down , and click on the track name of the destination track. Now, select **Functions > Object > Move Selected Objects to current Track** to move them there. (This has already been done in the above example)

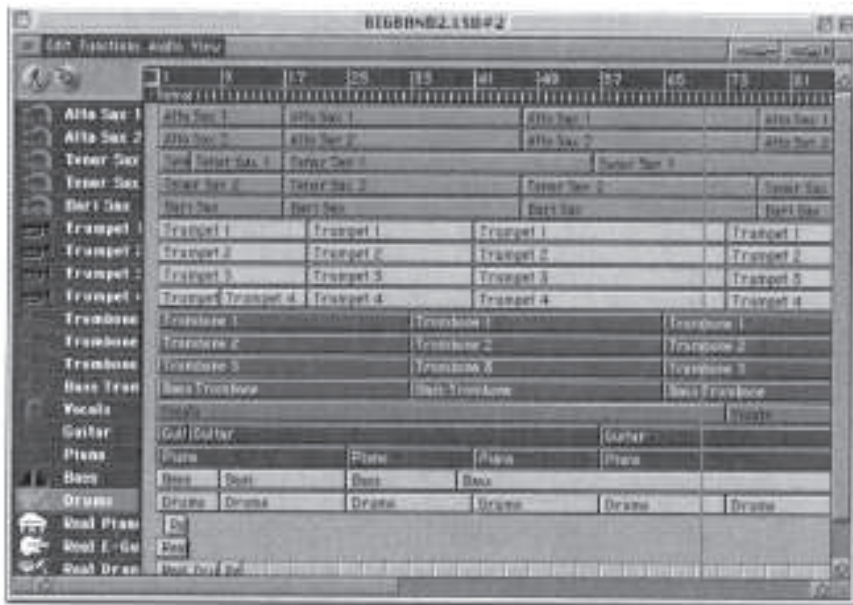


Hint for selecting the sequences and their destination track: when you click on the name of a track, all sequences on this track will be selected. With *Shift*-click (also on another track name) you can add other sequences to the selection. After you have selected all desired sequences, select the destination track by clicking on its track name while holding the *Alt* key (Warning: if you don't press *Alt* during this process, the sequence selection will change again.)

Next, you have to get rid of the gaps and overlaps between sequences, by changing their length. First, set the Locator Positions in the Transport window exactly to the beginning and end points of the song (This can also be done by dragging the mouse in the upper part of the bar ruler between those positions, to mark the whole length as a Cycle area). Now, select all sequences and select **Functions > Object > Tie Sequences within Locators** in the Arrange window. This lengthens each



selected sequence, until its right boundary meets the left boundary of the following one. This function also moves the starting points of the first sequences, and the end points of the last sequences on each track, so that they match the corresponding locator points. The position of events within each sequence are not altered in any way, so the MIDI playback will not be affected. After this procedure, the Arrange window in the example above would look like this:



In the Score window, these sequences are now displayed as continuous staves (except in Content Link mode, where only single sequences are displayed), since they are now continuous in the Arrange window. In linear view, you still can see clefs, time, and key signatures at the beginning of each sequence, but in Page and Print view these are not displayed any more, so nothing indicates the sequence borders.

Another Method: Merging Sequences

Instead of changing the sequence borders as described above, it is also possible to simply **Merge** all the sequences of each track, thereby creating one long sequence per track.

You can do this directly in the Score window by using the Glue tool: first, select all desired sequences, then click on one of them (anywhere within the staff) with the Glue tool. Note: if

sequences on different tracks are selected when you use the Glue tool, they will be merged separately by track.

In more complex arrangements, it will probably be easier to accomplish this in the Arrange window, especially if sequences on different tracks have to be merged. Use either the *Merge Objects* key command or menu **Functions > Merge > Objects**.

Note: there is also *Merge Objects per Tracks* available as a menu or key command. In both cases, it is important to select the desired sequences first.

Please note: although this method seems to be easier and faster than adjusting all the sequence borders as described above, you lose the ability to assign different display settings (e.g. display quantization) to the different sections (sequences) of each track instrument.



13.4 General Appearance of the Score Edit Window

There are quite a few options for changing the general appearance of the Score window. The settings for these options can be found in various places:

View Menu



Page Edit

Here you can activate and deactivate the following display options by selecting the corresponding menu entries. Activation is indicated by a check mark:

Note: a quicker way of changing between Linear and Page Edit view is to click on the -button in the upper left corner, or use the corresponding key command.

For editing, you should use Linear view, since screen redraws are much faster, especially on slower computers. Page view should only be used for layout work in preparation for the actual printout.



When you work in Page view mode, you can jump directly to any page, using the key command Goto Page. The Song Position Line will be automatically set to the beginning of the selected page, unless the sequencer is running when you execute this command. In this case, Catch mode will be turned off.



Print View

Print view will also show the page margins on the screen (only if Page view is activated). Note: the yellow area on the screen represents *the printable area* (which also depends on the printer used), not the actual sheet of paper.

Explode Folders

Displays the contents (sequences) of all lower display levels (folders), together with sequences of the current level.

If this option is not activated, folders are displayed as gray beams in linear view. In Page view, they are not visible at all. As with the other windows, the contents of a folder will appear when you double-click the gray beam that represents it.



Explode Polyphony

Displays every voice of a polyphonic Score Style on a separate staff. In general, this is not intended for printout, but for convenient editing of polyphonic voices.

Scroll in Play

Changes Logic's scrolling behavior when *Catch* is activated: the current song position is always kept in the middle of the window, so instead of the Song Position Line, the score display itself is moved around the SPL, which stays in the center of the screen.

Partbox

Show All Groups: If this is deactivated, only the Partbox Group currently selected in the top section of the Partbox appears below it.

Lock Group Positions: This allows you to lock the order of the Partbox Groups. Otherwise, the group of the last selected object is always moved up to the top of the lower section of the Partbox.

Toolbox

Hides/Shows the Toolbox on the left side of the Score window. As in all Logic windows, the Toolbox also pops up at the current cursor position any time you use the corresponding key command (default: ). After selecting a tool, or pressing  once more, it will disappear again.



SMPTE Time Ruler

Displays SMPTE timecode positions in the bar ruler (only in linear view). If there is enough vertical space, it will show both bar and timecode positions.

Parameters

Hides/Shows the left column of the Score window (Display Parameter Box, Event Parameter Box, Partbox etc.).

Instrument Names

Hides/Shows the names of the used track instruments, to the left of the score display.

PageRulers

Hides/Shows rulers in Page Edit view. Measurement units can be centimeters or inches.

White Background

Switches between a white and colored background.

Hyper Draw

If you activate one of these options, a Hyper Draw area appears at the bottom of the main working area. You can change this area's height, by dragging the borderline vertically with the mouse. Note that Hyper Draw in the score is only available when single sequences are displayed in linear view. For more on Hyperdraw, see page 13-31 and page 3-49.

Adjusting the Borders between the Different Areas

You can move the left and upper borders of the main working area, by positioning the cursor on one of the border lines, and click-dragging it. During these operations the cursor becomes a tool with two or four arrows. There are three possibilities:

On the left side, you can alter the width of the columns containing the parameters, (mainly changing the appearance of the Partbox Groups), and the instrument names.



Along the upper border, you can create *vertical headroom*. This is useful for working on parts containing very high notes, or symbols above high notes (like slurs), which might not be visible otherwise, and therefore could not be edited.



In the upper left corner of the main working area, you can change the width of the parameter column, and the height of the bar or page ruler. If **View -> Instrument Names** is activated, the width of the instrument name column can also be changed.



Parameter Boxes

Both the Display Parameter Box and the Event Parameter Box can be reduced to display just their title lines, by clicking on the triangle in the upper left corner of each parameter box.



The Main Working Area

The appearance of the score display depends on:

The settings of the *Link* and *Catch* buttons (see page 13-109)

The selected Instrument Set (see page 13-111)

Page or *Linear view* (menu **View**, and/or button to the right of the MIDI Out button)

Zoom level

Explode Folders on or off (Menu **View** see above)

Explode Polyphony on or off (Menu **View**, see above)

Hyper Draw on or off (Menu **View** see above)

13.5 Input Methods in the Score Edit Window

There are several ways to input notes and symbols in the Score window:

Real time MIDI recording (notes and sustain pedal markings only)

MIDI step input (notes only)

Mouse input from the Partbox (notes, text, and all symbols)

Insertion of copied events (all, except global objects)

Fast input of symbols by assignment to selected notes with key commands. These are available for some symbols which are attached to notes (accents, Jazz symbols etc., see page 13-45), and slurs or crescendi (see page 13-47).



With *Hyper Draw*, you can also insert and edit MIDI events which are not normally displayed in the Score window at all, like continuous controller data (i.e. volume, pitchbend etc.). More about this can be found in the section *Hyper Draw in the Score Edit Window* on page 13-31.

The various input and insertion methods can be combined. With all input methods, however, you should always keep in

mind that MIDI playback is also influenced by the settings in the *Sequence Parameter Box* and *Instrument Parameter Box* in the Arrange window. Settings in the *Environment* may also affect MIDI playback, if you have configured it to do so.

13.6 Score Display Options - a Survey

With a few exceptions (see section *Printout* on page 13-120), a printed score is always identical to the Page Edit view display in the currently active Score window.

The display, again, depends not only on what has been recorded or inserted, but to a great extent on the chosen display settings. Since there are a quite a few, each affecting only certain aspects of a song's display, here is an overview of the various settings. These are organized into groups, according to how much they influence the overall display. Detailed explanations of the following terms are given on the indicated pages.

Display parameters are also demonstrated using an example file in the first chapter of the Emagic-Score-Tutorial (to be found on the Emagic Tutorial CD),

Settings which affect all song files:

The Score Preferences (page 13-143)

Settings which affect the score display of one song file:

Page format, and printer driver

Song Settings (menu layout):

Global Format (page 13-123)

Clefs & Signatures (page 13-133)

Extended Layout Parameters (page 13-135)

Numbers & Names (page 13-128)

Guitar Tabulature... (page 13-137)

Text Styles... (page 13-146)

Score Styles... (page 13-77)

Settings which can be assigned differently for each Score window, and therefore allow different score displays within the same song file:

- Display levels: Content Linked, Linked or Not Linked (page 13-109)
- Instrument Set: selection and order of the displayed instruments (page 13-111)
- Explode Folders (page 13-14)
- Explode Polyphony (page 13-14)
- Linear view or Page Edit view (printout can only be started from Page Edit view)
- Zoom levels (does not affect the printout)

Settings which affect the score display of a particular sequence:

The parameters in the Display Parameter Box (page 13-68):

- Score Styles...-selection
- Display quantization ("Qua")
- Interpretation
- Syncopation
- No Overlap
- Max. Dots

Special case: *Score Off* in the Extended Sequence Parameters (page 13-77)

Settings which can be changed for each note individually, regardless of the default settings (Note Attributes, page 13-105):

- Accidentals: Enharmonic Shift
- Accidental Position (distance to note)
- Beaming of notes
- Stem Direction
- Interpretation
- Syncopation
- Tie Direction
- Treating a note as Independent Note or Independent Grace Note
- Note head appearance (Head)
- Note size (Size)

Horizontal Position: Graphic positioning
N-Tuplets settings (Tuplets)

13.7 Printout

For printing, it is important that you first choose the appropriate paper size and format in the menu **File > Page Setup...** The Score window with the intended print view has to be the active window. You start printing by selecting **File > Print...** or use the corresponding key command (default setting ). There will be a dialog box, which will vary slightly from printer to printer. Depending on the printer, there may be some additional options, such as which pages you would like to print, or the number of copies.

The printout is always identical to the Page Edit view display in the Score window. The following items however are *not printed* but are only visible on the screen:

- The dashed lines which represent page and header margins, and borders between staff systems
- Cursor, Song Position Line
- Colors (page margins, stave lines of selected sequences)
- Hidden time signature changes
- Hidden note heads (displayed grey on the screen), and their ties
- Hidden bar lines (displayed as dashed lines on the screen)
- Hidden N-Tuplet numbers

Printers work with higher resolution than monitors. Therefore, you should work in high zoom levels when doing layout work that requires precision, such as positioning symbols and text elements.

It is very easy to change back and fourth between normal and extremely enlarged view of the display, by using the Magnifying Glass tool (shortcut: rubber band selection while holding down , to enlarge the selected area).

Sonata Font for Display and Printout of Notes and Symbols

You can use the Sonata font by Adobe (not included with Logic), instead of Logic's internal score font, for display and printout of notes and symbols. The Sonata font must be in your *Fonts* folder (inside the system folder), in order to make use of this option. Use Adobe Sonata Font (if available) in Options > Score Preferences also has to be activated.

Since screen display with this font is rather hard to read (especially at smaller zoom levels), there is also an option to use Sonata just for the printout. Select only for printout in the applicable Score Preferences option.

13.8 Graphic Export

There are two ways to export Logic's notation into other programs, as graphic images

Postscript Export (EPS files):

Just as with most other Macintosh programs, you can use the Apple Laser Printer Driver to save a whole page as an EPS (Postscript) file. To use this option, you have to select that driver in the Apple-menu's *Chooser*, as if you were going to use it for printing. Start the regular printing procedure. In the printer dialog, select the page that you want to export, and then select the option *Save as Postscript File*. This file can only be imported into programs which can handle Postscript files. For this option, you need the Apple Laser Printer Driver, which can be found at Apple's web site. You do not need a laser printer, just the printer driver, since you will not actually print the page.

PICT Export Using the Camera Tool

The second possibility is to save a part of a Logic score page (or a whole page) as an image (PICT format), and paste it into

documents created with other programs (like word processing and DTP software). Use the *Camera tool* to select the desired score section, using the “rubber band” selection technique. As soon as you release the mouse button, Logic creates a PICT file, (or copies the selection to the clipboard-see option below.) encompassing the area you dragged the *Camera Tool* over.



This function is available only in Page Edit view. There are some related settings in **Options > Score Preferences:**

Graphic Export Resolution: Determines the resolution of the exported image file. This can be set between 72 and 600 dpi.

If the image is intended for use on a computer screen only, use the low resolution (72 dpi).

If the image is intended for a document that will be printed, select your printer’s resolution.

Graphic Export to (flip menu)

Clipboard: The image is copied to the clipboard and can be pasted directly into other documents, without saving it as a separate file.

PICT-File: The image will be saved as a PICT file. You’ll be given a file selector box, to assign a file name and path.

PICT File Creator ID: This allows you to determine which program will be used to open the file when you double-click it in the Finder. The default setting is SimpleText (text). This setting, however, has no influence on the file format itself.

13.9 Real-time MIDI Recording in the Score Window

Although real time MIDI recording is usually done in the Arrange window, you can also start recording directly while working in the Score window. You must first select the destination track in the Arrange window’s track column. In the Score window, this can be done by clicking on a staff which belongs to

that track. If there are no sequences on the track to be recorded, you will have to switch to the Arrange window for the recording, since the track is not yet displayed in the Score.

If you want your recording to become part of a previously existing sequence (i.e. because you use score template files), select the global menu **Options > Settings > Recording Options...**, and activate **Merge New Recording With Selected Sequences**, before starting to record. Also, make sure that the destination sequence is selected, otherwise Logic will create a new sequence on the same track, which will probably overlap the existing one.

The display of the recorded notes depends on:

1. The score display settings, especially those in the *Display Parameter Box* and the *Note Attributes*, Please read the corresponding sections in this chapter.
2. The *Quantize* and *Transpose* settings in the Arrange window's *Sequence Parameter Box*:

Logic creates a score display based on the MIDI data, including any changes caused by these two playback parameters.

An example: if *Transpose* in the Sequence Parameter Box is set to +2, a note that initially (when recorded) was a C will be played back and displayed as a D. The same applies to playback quantization with the exception of notes that were “dequantized” using the Quantize tool, or the corresponding key command.

Note: the Score window's Info Line always shows the original pitch of a note, independent from any playback, or display transposition parameters. Transposition parameters in the track Parameter Box (Arrange window), however, have no impact on the display.

13.10 MIDI Step Input in the Score Window

MIDI Step Input in the Score window works the same way as in the other editor windows. This function is described in the chapter Editor windows, on page 1-38. Please read this section first. The Score Editor window, however, is the only editor (besides the Matrix editor) where the content of several sequences can be displayed and edited at once. This is why there are some differences concerning Step Input in the Score window:



The sequence into which you intend to insert notes has to be selected (click on the staff). If there is no staff yet, you must first create an empty sequence in the Arrange window.

Simultaneous step input into several sequences is not possible. If more than one sequence is selected, the notes will be inserted into only one sequence.

Logic's full step-time functionality, as described in the section *Step Recording* on page 1-38, is only possible if just one sequence is displayed. It is possible to work with step input in full scores, but the potentially important *Back* function, with its fixed key command , does not work in this case.

If several sequences on the same track in the Arrange window directly follow one another, without any space or overlaps between them, you can insert notes using step input from one sequence to the next. As soon as the Song Position Line reaches the beginning of the next sequence, Logic will insert the notes into that sequence automatically. All of these sequences have to be selected. In the Arrange window, this would look somewhat like:



In the Score window, the above would look like:



Key Commands for Step Input

Reminder: if you assigned the keys used for the step input key commands (**A**, **S**, **D**, **Q**, **W**, **E**, **R**, **B**, **N**, **M**, **→** and **←**, see page 1-38) to other user definable key commands, these commands will not be available for step input anymore.

There are however some other user-definable key commands with the same, or very similar functions:



M corresponds to *Forward* the Song Position Line proceeds to the beginning of the next measure.

E and **R** correspond to *Set next higher format / Set next lower format*

A, **S**, **D** and **Q** in the Score window can be substituted by the key commands *Partbox: 1/4 Note..., ...1/32 Note*. In addition, there are also key commands for half notes and whole notes. These commands select the corresponding note values in the Partbox. When MIDI-In/Step Input is activated, the global display format in the transport window is always set to the matching value. There are a few things to keep in mind when using these key commands:

Using any of these key commands twice in succession changes the Partbox selection *and* the global display format to the next higher triplet value.

Using them three times in succession changes the Partbox selection to the next higher dotted value, the global display format however, retains the former triplet value, which will also be inserted now.

With the fourth identical command in succession, both the Partbox selection and global display format are reset to the original value.

If a key command for a value different from the currently selected value is executed, Logic always selects the directly corresponding binary value. Therefore, triplet values can only be selected by executing one of these key commands twice, or by using the key commands Set next higher format / Set next lower format.

Notes created with Step Input are interpreted and displayed according to the settings in the Display Parameter Box. Here are some hints:

Set *Qua* to the smallest rhythmic value that should be displayed.

If you want to see the precise note values that you insert, switch off *Interpretation*, and *No Overlap* in the Display Parameter Box, and select the highest possible value for *Qua*. *Syncopation* can be set according to personal preference. (Details about this can be found in the section *The Display Parameter Box* on page 13-68)

Use a Score Style with a clef which matches the register of the inserted notes.

Positioning the SPL Directly in the Score

The Song Position Line can be set directly in the Score window by clicking into a staff at the desired position, while pressing the -key. This function is only available when the sequencer is stopped. It is especially helpful in Page view, where no bar ruler is displayed. The selected position is displayed in the infoline as long as the mousebutton is pressed.

13.11 Mouse Input

In order to insert notes, symbols and text elements (called *Objects* in the following sections) with the mouse, there has to be a staff into which these objects can be inserted. Each staff (even an empty one) is the direct representation of one or more sequences. Empty sequences can be created with the Pencil

tool in the Arrange window, where their length and position can also be changed.

These empty sequences are displayed in the Score window as empty staves, with automatically displayed rests (unless the automatic rest display is deactivated in the assigned Score Style)

Input

For mouse input, you first have to select an object in the Partbox, and then drag it to the desired position in the main working area. As a positioning aid, you will see an *Info Line*, instead of the local window menus, as long as you are “holding“ the object with the mouse. Now move the mouse with the button pressed, until the desired position is shown in the Info Line. Release the mouse button. At smaller zoom levels, it’s especially important to use the *Info Line* to find the correct position.

Insert 2 3 1 1 NOTE 1 03 64 _ _ 3 232

When you are working in full score view, and insert objects between the staves (such as dynamic symbols), you should make sure that these symbols have been inserted into the desired sequence/staff. Immediately after inserting an object, the sequence into which you inserted the object will be selected, and therefore be displayed with blue stavelines. You can also see its name in the top line of the Display Parameter Box.



In general, it is better to change to single staff display (double-click on the staff), for inserting objects which are positioned between staves (especially lyrics). Although in full score mode it does not matter to which sequence an object belongs, since they stay where you put them, such objects can be displayed with the wrong instrument, when single instrument parts are displayed and printed later.

All Partbox Objects and their particular features are described in detail in the section *The Partbox* on page 13-32.

In the Event List notes and sustain pedal markings are listed as **MIDI Events**, most other symbols and text objects as **Meta Events**. Global symbols, global text elements, time, and key signature changes are not listed in the Event List. Symbols

which are attached to notes (accents, noteheads etc.), are not listed as events in the Event List, but are saved with the notes they belong to.

MIDI Channel and Velocity of Inserted Objects



The MIDI channel and velocity of objects inserted with the mouse, are determined by the *Insert Defaults* settings in the Event Parameter Box. These *Insert Defaults* are visible (and can be edited), when no object is selected. Click anywhere in the background of the working area to display this status. Make sure that “Insert Defaults” is displayed in the top line of the Event Parameter Box.

When inserting events into a staff which uses a Polyphonic Score Style, only MIDI channels corresponding to one of the voices can be used.



Many symbols besides notes (all Meta Events), have a MIDI channel in the Event List. Some of them even have a velocity parameter. These parameters vary in the Event Parameter Box: for most objects, the MIDI channel results in a **Staff** parameter, which for Polyphonic Score Styles determines to which staff (theoretically up to 16), the symbol belongs. BE CAREFUL: if this parameter is set to a number higher than the number of staves in the Score Style, the symbol will become invisible.

Insert Quantization

To make positioning of notes and symbols easier, objects can only be inserted at certain bar positions:

There is an invisible grid for positioning of notes. You can imagine its resolution as if the whole bar-beginning at beat one-were filled with notes of the chosen value. An example: in an empty 4/4 measure, half notes can only be inserted on one and three, eighth note triplets only on the first, second or third eighth note triplet of every beat.

In addition, all binary and dotted values can be inserted at any position of an automatically displayed rest. This means that if there is an eighth note on one, you can insert a quarter or a dotted quarter note at the second eighth's position. (In this case *Interpretation* has to be turned off in the Display

Parameter Box, to allow the eighth note rest on the “and” of beat one to be displayed). The *Syncopation* setting will determine how the inserted note in this example is displayed. Symbols (except those directly attached to notes) can be inserted at any position which corresponds to the grid defined by the *Qua* parameter. For hybrid quantizations (i.e. 8,12), the insert quantization is determined by the binary value (in this example 1/S).

However, you can move any note, and most symbols, to any desired position, by editing its numeric bar position in the Event List, or the *Event Float Window*. For example, if you want to insert a half note on “two” in 4/4 time, place it on “one” or “three”, and change the position to “two” in the Event Float.

Simultaneous Input of Objects into Several Sequences (Insert Multi)

If you hold down the -key while inserting an object from the Partbox, the object is inserted at the same position into all currently selected sequences. The *Info Line* will display *Insert Multi* during this process. With this technique, you can insert notes, symbols, and even text elements into several sequences at once. (Text elements will not appear until you have finished the text input procedure).



You may make the multiple sequence selection in either the Arrange or the Score window. -click all desired sequences/staves, or use the rubber band selection technique with the mouse. In the Score window, you don't have to select events, just drag the cursor vertically across the staves that you wish to select. After selection, the Display Parameter Box will indicate how many staves (sequences) are selected, in its top line.

Key changes, signature changes and global signs automatically are inserted into all staves of a song, and cannot be inserted into selected staves only.

Exact Graphic Position of Symbols

When you insert a symbol which is not automatically attached to a note, you should pay attention to the following details:

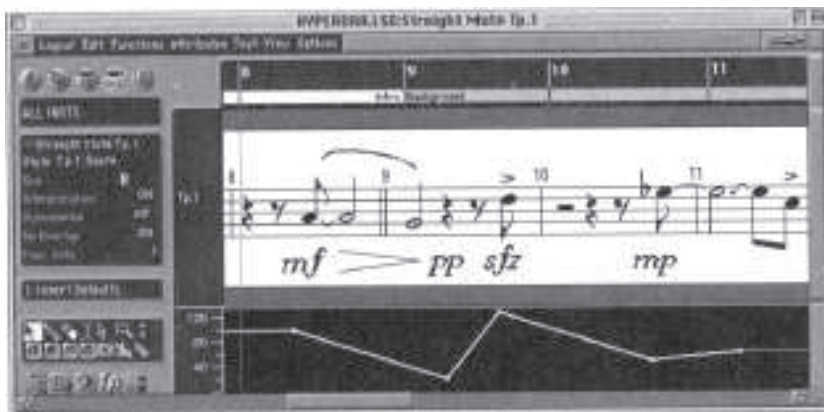


All symbols are inserted at the bar position which is indicated in the *Info Line*, when you release the mouse button at the end of the input process. This will also be the position indicated in the corresponding Event List line. Apart from the bar position, there are also parameters that affect the graphic offset from the inserted bar position. These are shown in the Event Parameter Box, where they also can be edited (vert. pos. hor. pos). The measurement unit of these two parameters is one sixteenth of the distance between two lines in a staff, so these distances will be scaled accordingly when the *Size* of the staff (defined in the selected Score Style) is changed.

When you insert symbols, make sure that the bar position really is identical with the graphic display position. If, for example, you work with small Spacing parameters (in **Layout > Global Format**), a “Forte” symbol which you put under a note on *one*, may be interpreted as being on two, with a negative hor. pos value, because you placed it a little bit too far to the right. This does not make any difference as long as you stay on that particular display level. However, if you switch to a different display level (i.e. for extracting the single instruments’ parts), this can result in an undesired display, since those graphic position values stand for an absolute distance (defined by vertical staff size). If later, you change the Score Style, the line breaks, or some other layout parameter, a different number of bars may be displayed in one line. Since the horizontal offset (*hor. pos*) is calculated as an absolute distance from the new (relative) graphic display of the bar position, the Forte symbol might now

be displayed at a position closer to the second beat than to the first.

Hyper Draw in the Score Edit Window



If you activate **View > Hyper Draw** you can insert and edit MIDI events which otherwise are not displayed in the Score window at all, like continuous controller data (i.e. volume, pitchbend etc.). The type and channel of the MIDI data which are displayed in the Hyper Draw area can be selected and changed from the **View** menu. Hyper Draw in the Score window operates in the same way as in the Arrange window. A complete description of Hyper Draw and its functions can be found in the section *Hyper Draw in the Arrange Window* on page 3-49.

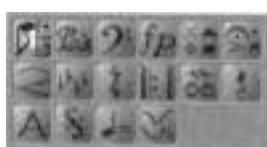
In the Score window, Hyper Draw can only be activated in Linear view, and if only one sequence is displayed. If you switch to another mode (Page Edit view, or full score), the Hyper Draw area disappears. However, the settings are stored, so when you return to single sequence and Linear view, the previous Hyper Draw setting is recalled.

In the Score window, the Hyper Draw display is horizontally scaled in relation to the score display above it. Its height can be changed, by dragging the horizontal borderline with the mouse. A corresponding value scale will be displayed in the instrument name column if **View > Instrument Names** is activated. As in the Arrange window, you may select and/or change the event

type and MIDI channel displayed in the Hyper draw area, from the menu **View > Hyper Draw**.

 Hint: With Hyper Draw set to *Note Velocity*, note velocities can be displayed and edited graphically.

13.12 The Partbox



The objects available in the Partbox (notes, symbols, text objects) are organized in groups. Its top section contains small squares, each one a representation of a group. Below that, there are all the groups with all available symbols. The arrangement of groups can change. If you click on one of the squares in the top section, the corresponding group is brought up right below it, for easy access to its symbols.

However, every displayed symbol can be selected and inserted anytime. The described function only makes it easier to work with the Partbox especially on smaller screens, or in small sized Score windows.

Selecting and Inserting Partbox Objects

Details on Mouse Input from the Partbox are described in the section *Mouse Input* on page 13-26. There are various ways to select and insert objects.

Selecting an object with the mouse from one of the object groups, and dragging it from there to the desired position in the score display. If a particular object group is not visible, click on its symbol in the top section of the Partbox. This places the selected object group above all the other groups.



Selection from flip menus in the top part: when you click-hold on one of the squares representing the Partbox groups, a flip

menu containing all the symbols of that group will appear. Move the mouse up or down, until the desired object is selected (inverted display). Now pull the cursor to the right, which causes the object to be dragged out of the Partbox. The rest of the procedure is identical to the one described above.

Direct Insertion of an already selected symbol: if the desired symbol is already selected (indicated by its inverted display in the Partbox), there is a shortcut. Click into the staff with the *Pencil tool*. If you click-hold the mouse button, you will be holding the previously selected symbol. Drag it to the destination position, watching the Info *Line* for positioning information.



Hint: when you press and hold the -key, the cursor takes on form and function of the alternative tool. If the pencil tool is assigned to that function, you can switch back and forth very quickly between the Pencil (for input), and the Arrow tool (for editing and making parameter changes) by pressing and releasing the -key, when inputting events.

Using Floating Palettes



A double-click on one of the squares in the Partbox' top section opens a small floating window, containing all the objects of the corresponding group. This palette can be moved to any place on the screen. Objects can be taken directly from these palettes, and dragged into position in the Score. Note that unlike in the regular Partbox groups, the object selection is not visible in the palettes.

The palettes can have different shapes, depending on which modifier key is pressed while you open a floating palette with a double click

Normal (with )

Vertical (with )

Horizontal (with )

If you don't use a modifier key when you open a palette, the shape depends on the corresponding setting in the Score Preferences.

As many palettes as needed can be simultaneously opened, and positioned around the screen.

Object Selection by Key Commands

The following (user definable) key commands will select the corresponding note values inside the Partbox. These can then be inserted quickly, using the pencil tool:

Partbox 1/1 Note

Partbox 1/2 Note

Partbox 1/4 Note

Partbox 1/8 Note

Partbox 1/16 Note

Partbox 1/32 Note



If you execute any one of these twice in succession, the corresponding triplet value is selected, three times brings you to the dotted value, and four times back to the original value.

There are also other key commands which change the selection inside the Partbox:

Next Partbox Symbol

Previous Partbox Symbol



“Next” moves the selection (similar to a text cursor) to the right, “previous” to the left. After reaching a group’s last object, the selection cycles back to the beginning of the group.

Next Partbox Group

Previous Partbox Group



Here “next” moves the selection down to the object group below the current one, “previous” up, as determined by the current order of the groups. When changing to another group, the selection within will always be the object which was last selected in that particular group.

Display Size of Inserted Objects

The size of inserted objects (except text and chord symbols) always corresponds to the size of the staff into which they are inserted (see Score Styles), and changes correspondingly if that staff's size is changed with the Score Style.

Most Objects can be made larger or smaller after having been inserted, using the Sizing tool (as can text elements, except for chord symbols). Just grab the object with the tool and move the mouse up or down. The changes will be immediately visible.

Here are detailed descriptions of the Partbox Object Groups:

Notes



The square representing the notes, in the top section of the Partbox, is divided into three parts, since the note values are organized into three sub-groups. Each of these can be brought to the top by clicking on its corresponding symbol (note, dot, or 3):



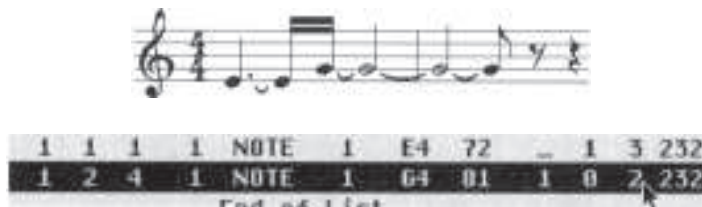
Inserting any of those symbols, creates a MIDI note event, with exactly the length of that value. For display however, all these MIDI notes are interpreted by Logic as if they had been recorded in real time. Therefore, the Display Parameter Box' settings have to be set in a way that allows the inserted notes to be displayed correctly. Please see the section *Rhythmic Interpretation of Sequences* on page 13-70.

If you only want to enter notes with the mouse, without using real time recording at all, switch off *Interpretation* and No *Overlap* in the Display Parameter Box, and choose a very high *Qua* value (i.e. 128, 384). *Syncopation* can be set depending on style, and personal preference.

Ties

Notes in Logic which are displayed as several graphical notes, connected with ties, always represent only one MIDI note. To insert such notes, you first insert a shorter note value, and

change its length afterwards in the Event Parameter Box, the Event List, or the Event Float window.



*Ties cannot be inserted in Logic, but are created and displayed automatically, according to the length of the MIDI note. The up/down direction of each tie can be set with **Attributes > Ties**.*

Changing the Way Notes with Ties Are Displayed

If you don't like the way particular tied notes are displayed, you can change the display by inserting a user rest from the Partbox, at the position where you want to see a graphic note. The rest's length is not really important, preferably a shorter one like an eighth note rest should be used. Watch the *Info Line*, for exact positioning. As soon as the rest is inserted, it will no longer be visible in the score display, however the note display will have changed accordingly. If you want to change the position of such a rest, or delete it, you may do so from the Event List. In the following example an eighth note rest was inserted into the example shown above at position 1.2.1.1, in order to make the second beat visible (*Interpretation* is switched off):



Allowing Diatonic Input only

Select **Options > Diatonic Insert** to limit input to notes that are diatonically correct for the current key. Notes can be altered chromatically, once they have been inserted.

Please note: if you protect (“lock”) your screensets (see the section *Protecting Screensets* on page 1-21), this function’s setting will also be “locked”.

N-Tuplets

Inside the triplet group there is the symbol for N-tuplets.



Depending on the display quantization selected, and the particular situation, Logic usually recognizes and notates regular triplets correctly. This requires a corresponding *Qua* value in the Display Parameter Box (see the section *Qua (Display Quantization)* on page 13-71). For the display of other N-tuplets, it is necessary to use the N-tuplet object.

Definition of Previously Recorded Notes as N-Tuplets



For notes that already exist (as part of a MIDI recording), but are not displayed correctly, the N-tuplet symbol has to be dragged on top of the first note. This note has to be indicated in the *Info Line*, before you release the mouse button. A dialog box will open, where you can define the N-tuplet's attributes:

In the score, the above would correspond to:



The dialog box can be explained most easily by using the example above: five tuplet eighth notes should be displayed in the place of four regular eighth notes. In addition, there are the following options:

Hide Bracket: the bracket is not displayed.

Hide Numbers: the N-tuplet's number will not be printed. On the screen, it will be displayed in parenthesis. This is necessary to be able to edit the N-tuplet further.

Show Denominator: the N-tuplet display also contains the denominator (the second number in the dialog box). In the example above this would be 5:4.

Direction: the direction of bracket and number.

Auto: according to the setting in the used Score Style.

Up/Down

Allow Double Values: if this option is activated (which is the default setting), N-tuplets can contain note values which are twice as long as the “regular” tuplet value. An example for this can be found on page 13-39.

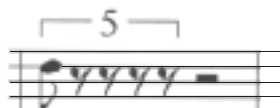
A second example, a quadruplet in 3/4 time:



Compare the settings in the dialog box with the score display: the tuplet's **Direction** is forced **down**, to avoid overlaps with the phrase markings. Show **Denominator** is activated, to emphasize the “four against three” rhythm.

N-Tuplet Input with the Mouse

If you want to insert a tuplet, including all of its notes using only the mouse, insert the first note at the desired position, then drag the N-tuplet symbol on top of it. In the automatically opened dialog box, define the settings. If you make the same settings as in the first example (eighth note quintuplet), and close the box, you will see the following result:



The Partbox

Initially, the display will be of a series of rests, their number corresponding to the N-tuplet value. You may now insert the other notes at those rests' positions. Select a note value which approximately matches the length of the desired N-tuplet notes. In this example, this would be dotted sixteenth notes.

*If you want to hear these notes played back with legato phrasing, and the notes are still too long or too short, you will have to change their (MIDI) lengths, The fastest way to do this is to select all the notes, and then **Functions > Note Events > Note Force Legato (Selected/Any)**, or use the corresponding key command.*



With the method described above, you can also create N-tuplets which contain *rests* or *syncopated note values* (double length):



If you want an N-tuplet to begin with a rest, you first have to insert a user rest from the Partbox, then drag the N-tuplet symbol on top of that rest. Automatically displayed rests cannot function as starting points for N-tuplets.

Changing or Deleting Existing Tuplets

A double click on the N-tuplet number opens its configuration box, where you can make changes, or use the **Delete** button to erase the tuplet. N-tuplets also can be deleted by clicking on the tuplet number with the Eraser tool.

Fast Input of Several Identical N-Tuplets

If you hold down **Alt** as you insert the N-tuplet symbol, the last inserted N-tuplet's settings will be applied to it, without the usual dialog.

Editing Automatically Displayed Triplets

Triplets are usually displayed automatically (depending on the *Qua* parameter in the Display Parameter Box). If you want to change the way these triplets are displayed, you can simply double-click the "3", which opens the N-tuplet dialog box (where you can change the settings), and converts the automatic triplet to an N-tuplet. This is also useful for hiding the number and brackets of "normal" triplets.

"Artificial" N-Tuplets to Bypass Display Quantization



You can bypass the display quantization value (*Qua*) by using hidden, "artificial" N-tuplets. For example, if there is a single 32nd note run in a sequence that otherwise only contains eighth note or longer values, you can set *Qua* to 1/8, and drag the N-tuplet symbol onto the first note of that 32nd figure (in the following example this figure

starts on the fourth beat). In the resulting dialog box, use the following settings:

This will show in the score display (on the screen) as:



On the printout the number (8) will not be visible anymore.

Although this is not usually what you would use tuplets for, it can be a handy tool for forcing the display of runs of shorter notes in a sequence (1/32nd notes in this case) whose display quantize value is set for longer note values (1/16nd notes in this case).

Grace Notes, Independent Notes

Independent notes are notes which are not included in the automatic rhythmic and graphic display calculation of a measure, but are still played back via MIDI at the position where they were inserted. There are two types:

Independent: these are displayed with their original length, as long as they are binary or dotted note values which can be displayed as a single (not tied) note. After insertion, you can change the stem direction, enharmonic interpretation, accidental distance and note head, using the other *note attributes* functions.

Independent Grace: these always are displayed as miniature eighth notes, with a slash across the stem, and with a tie. Tie direction, stem direction, enharmonic interpretation, accidental distance and note head can be changed.



Both types of Independent Notes can only appear as single notes, without any beams to other notes.

Creating Independent Notes:

Existing regular notes can be converted to Independent Notes with **Attributes > Independent > Independent** or **Independent Grace**. There are also corresponding key commands. **Attributes > Independent > Not Independent** converts notes back into regular notes.



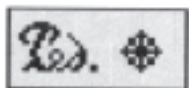
If you want to insert an *Independent Note* directly, select a note with the mouse from the Partbox while holding down **Alt** and drag it to the desired position.

Independent Grace Notes cannot be inserted directly, but have to be created by converting regular or Independent notes.

In most situations, you will have to create additional graphic distance between the Grace Note and the note it is associated with. The best way to do this is to use the Layout tool to move the main note away from the grace note.

Hint: If you want to create beamed grace notes, try using a polyphonic Score Style (no rests displayed for second voice). Assign these notes to the second voice, and change their size with the Sizing tool.

Sustain Pedal Symbols



These two symbols are the only score symbols-except for the notes themselves - which directly represent MIDI events (MIDI controller #64 sustain pedal on and off), and thus affect MIDI playback. Logic intelligently inserts the On or Off version of the symbol, depending on which type preceded it. (i.e., a Sustain Off always follows a Sustain On and vice versa)



Clefs



The clefs in the Partbox generally are intended for clef changes in the middle of a part, for a short section of the music. An example would be cello or bassoon parts, when the part changes to a higher register. Inserted clefs change the score display, starting at the precise insertion position (which can even be in the middle of a bar).



For changing the clef of a whole sequence, you should not use the part-box clefs, but choose an appropriate Score Style for that sequence.

When you insert a “C-clef”, please watch the **Info Line** carefully, since this clef can be positioned at five different heights, depending on the register (alto, tenor clef etc.).



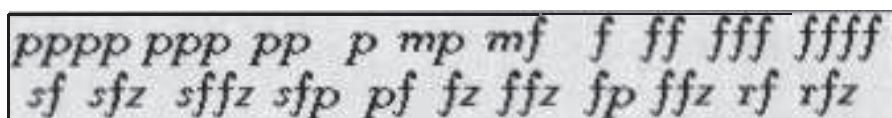


If you want to change an already inserted clef, double-click on it. This opens the following selection box. (The same applies to the “basic” clef

of a sequence, however, changing this clef affects the Score Style itself, as well as all sequences that use the same Score Style.)

The option **Smaller Clef Changes** in **Layout > Clefs & Signatures** also affects clef changes. All clefs, except the first one in every staff, will be reduced in size, depending on the value set here. This will happen, regardless of whether the clef changes are caused by using a different Score Style in the same staff, or by inserting clefs from the Partbox.

Dynamic Symbols



These symbols can be inserted at any position in the score. They are graphic symbols only, and thus have no effect on the playback of the sequences they are placed in.



Note Heads



If you drag one of these symbols onto a note (watch the *Info Line*: the destination note must be indicated there), the form of its note head changes to that of the selected symbol. If multiple notes are selected first, dragging the note head onto any one of the notes will cause it to be attached to all the notes in the selection. These note heads do not affect MIDI playback.



If you want an altered note to have a standard note head again, use the same method, this time using the black round note head.

Invisible Note Heads

Using the grey note head will make a note's head and associated ties invisible. On the screen this note is displayed with a grey notehead, so that it can still be edited. In the printout however, nothing will be visible, except the stem.

*If you combine this with **Attributes > Stem > Hide**, you can hide notes in the printout altogether.*

Symbols Attached to Notes

(Fermatas, accents, phrasing symbols, mute & bow markings, etc.)



This group contains symbols which usually refer to only one note at a time (Trills, however, are in a separate group). These symbols are positioned automatically, as soon as they are assigned to a note. You do this by dragging the symbol onto the note head (watch the *Info Line* to ensure accurate placement). These symbols always move along with the notes they belong to, whether they are transposed or otherwise graphically altered.



MIDI Meaning

The symbols of this Partbox group *can* affect the MIDI playback of the notes to which they are assigned. The effect can be on their velocity and playback length. If you want these symbols to affect the playback, you have to define the effect for each symbol separately in the *MIDI Meaning* dialog box. This can be opened from the menu **Layout > MIDI Meaning**. A detailed description of these functions can be found on page 13-141.

Quick Assignment with Key Commands

For some of these symbols there are user definable key commands, which automatically assign the corresponding symbol to all currently selected notes:



- ☰ *Attach Symbol Fermata Up*
- ☷ *Attach Symbol Fermata Down*
- *Attach Symbol: Staccato*
- ‡ *Attach Symbol: Staccatissimo*
- *Attach Symbol: Accent*
- ▲ *Attach Symbol: Strong Accent*
- ≧ *Attach Symbol: Long Accent*
- *Attach Symbol: Tenuto*
- ▼ *Attach Symbol: Up-Bow*
- ▣ *Attach Symbol: Down-Bow*
- ⊙ *Attach Symbol: Flageolet*
- + *Attach Symbol: Pizzicato*
- *Attach Symbol: Left Hand*
- | *Attach Symbol: Right Hand*
- ☪ *Attach Symbol: Soft*
- ♯ *Attach Symbol: Heavy*

Hint: working with these key commands can be even faster when you also use the key commands *Select Next Event / Select Previous Event*, and *Toggle next/previous Object (Event)*. The two latter commands *include* the following or previous event in the current selection, or reverse that inclusion.

If you want to insert one of these symbols so that it remains independent of any note, you can do so by holding *Alt* as you insert it with the mouse. This method was necessary to insert the fermatas in the following example:



If the first fermata were a regular inserted (attached) fermata, it would be placed above the beginning of the note on the offbeat of four (The tied note on beat one

is only a graphic note, not a MIDI note). For the second fermata, there is no note or symbol for it to attach to, since the quarter rest on three is an automatically displayed rest. (You could, as an alternative, attach these symbols to manually inserted rests.)

Jazz Symbols



Jazz Symbol input is similar to the input of accents etc.: if you drag a Jazz Symbol onto a note head, it will be assigned to that note, and positioned automatically. Additional space will be created automatically, to avoid overlaps with other notes or symbols. These symbols always move together with the notes they belong to, whether they are transposed or otherwise graphically altered.



Jazz Symbols can also be inserted using the key commands *Attach Symbol: Jazz* 1-6. The numbers 1 to 6 stand for the position of the symbols in the partbox. In the partbox display, black dots are used to indicate the position of each symbol in relation to the note head (compare the two illustrations above).



Jazz symbols may also be inserted independently from notes as graphic objects, by inserting them with the mouse while holding the m-key. These unattached symbols can be moved freely; and unlike normally inserted Jazz Symbols, they do not affect the spacing of notes.

Slurs, Crescendi, Lines, Arrows



Input

Insert these objects at the intended left starting point. They can then be edited to their desired shape. After insertion, the

object remains selected and small black square “handles” are visible at its end points, or, in the case of slurs, along its length as well. You can drag these “handles” with the mouse, to change the object’s shape.



Fast Slur and Crescendo Input with Key Commands

For slurs, crescendo, and decrescendo symbols, there are also user definable key commands for fast input:



Insert Slur Up
Insert Slur Down
Insert Crescendo
Insert Decrescendo

These commands place the corresponding object below or above (*Slur Up*) the currently selected note(s). The start point is determined by the first selected note, the length of the object by the last selected note. The shape and vertical position of each object is determined by the settings of the last edited object of the same type used.

Altering the Shape

To edit the shape of these symbols, grab and drag the black square “handles” at any time, even if they are not visible. To make them visible, select the object by clicking on one of its ends, or by rubber band selection.

Changing the Position

All but the last six objects can be moved by simply click-dragging the mouse from the object’s left hand starting point. The last six objects in the group are moved by click-dragging each of their corners (“handles”) separately.

Change of Length

All these symbols can be as long as you like, and can therefore be displayed across line and page breaks in several parts.

If, for example, you want to extend a slur to a position which is not visible in the Score window any more (in Linear view), you can grab it by its right end, and pull the cursor to the right border of the Score window - the score will scroll automatically. As soon as the desired end position for the slur appears in the window, move the mouse a little bit to the left to stop the scrolling. Now position the end of the slur. It is important that you do not release the mouse button during the entire process.

In Page Edit view, you can simply drag the right end to its destination in one of the lower staves. The slur will be displayed in several sections. The same principle applies to the other symbols in this group.

The length of these objects can also be changed in the Event List or Event Float, by changing the numerical length parameter. The end point is moved horizontally by this procedure. Slanted objects like arrows will be displayed at a correspondingly varied angle.

The length can exceed the end of the sequence, i.e. it is possible to drag the right end of a crescendo with the mouse to a position in the next sequence on the same track.

Particular Attributes of Certain Symbols

Crescendo: the tip determines the general position. At the upper right end, length and opening angle can be adjusted. At the lower right, length and overall angle can be altered.

Decrescendo: at the upper left end you can alter the bar position, and opening angle. At the lower left end, bar position and overall angle can be set. At the tip (right end), vertical position and length are adjusted.

Lines and arrows: can be moved in their entirety by dragging their starting point. At the other end, length and overall angle can be altered. The slanted line symbol stands for a line that can be altered freely. The other two line objects (vertical and horizontal) can only be extended in the corresponding direction.

Slurs: slurs have five editing points (the black square “handles” are visible when the object is selected) to change

The Partbox

their shape. The last two slur symbols in the slur group are mainly intended for slurs across staves (i.e. in piano parts).

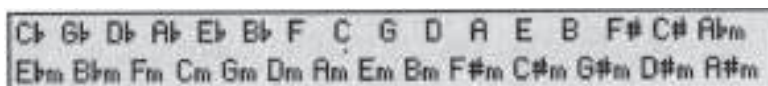
Reminder: these are slurs, not ties. Ties cannot be inserted manually, but are displayed automatically if a MIDI note's length requires it. (see the section **Ties** on page 13-35).

These symbols are graphic only, and have no impact on MIDI playback.

Precise Positioning

It is very helpful to work in high zoom levels when positioning slurs. You can quickly switch to an extremely zoomed display by holding down  while rubber band selecting a certain area of the score. In this case, the mouse becomes the magnifying tool, and the selected area is displayed as large as possible. Do all your positioning and layout work at this zoom level. Return to regular view by clicking on any free spot while holding  again.

Key Signatures and Key Signature Changes



Key signature changes can be inserted at any point in a song, by dragging one of the key symbols from the Partbox, as shown above, onto any staff. Be sure to watch the **Info Line**, to ensure accurate placement. Remember that key signatures and key signature changes are effective globally, for all instruments, on all display levels.



Natural symbols are displayed automatically with a new key signature when necessary (e.g. when changing from A flat major to F major or A major).

Display transposition of the key signature (for notation of non C instrument parts) is achieved by selecting an appropriate Score Style. For atonal music, there is the option to deactivate

Automatic Key Transposition for transposing Score Styles, in **Layout > Clefs & Signatures**. This is effective for the whole song. If it is deactivated, all sharps and flats are displayed directly with their notes. There is a similar option in the Score Style window (parameter "Key") which allows you to switch off the key signature for that particular Score Style, or even for individual staves in multi-staff Score Styles.

The basic key signature of a piece should, however, be set by double-clicking on the first staff, between clef and time signature. This opens a dialog box where you can set any major or minor key. This window also allows you to disable the use of double flats and sharps.



This method applies not only to the first staff, but for all staves. However, if the piece already contains other key signature changes, any newly inserted key signature only changes the key from the insertion point up until the next key signature change. As with the starting key signature, inserted key signatures affect all sequences, on all levels.

Deleting a Key Signature Change

This is done by either clicking directly on the key signature change with the Eraser tool, or by selecting it and pressing **Delete**.

Moving and Copying Key Signature Changes

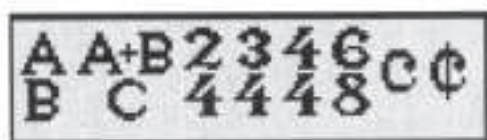
Since key signatures are global symbols, they cannot be moved or copied as easily as other objects.

Moving key and time signature changes is possible to a certain extent in the Arrange window. To do so, use the **Functions > Cut/Insert Time** functions, but do *not* select any sequences while performing the edit. This results in a movement of all time and key signature changes and all other global symbols (repeat symbols, double bar lines etc.), while the sequences and their data (notes, other symbols etc.) stay where they were. For a detailed description of this function please see the section *Adding or Removing Passages* on page 3-75.

It is also possible to copy *all* time and key signatures from one song to another. This is described in the section *Copying Time Signatures between Songs* on page 2-10.

The display of key signatures and key signature changes also depends on the settings in **Layout > Clefs & Signatures**, and on the *Key* parameter in the Score Style window for each Score Style. Please also read the section *Clefs & Signatures* on page 13-133.

Time Signatures and Time Signature Changes

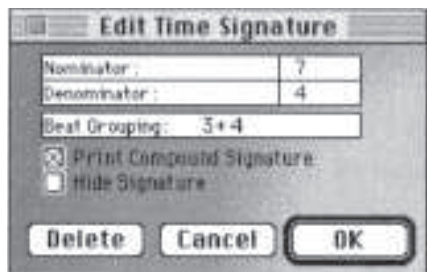


Time signature changes can be inserted at the beginning of any bar, either by inserting signatures from the Partbox, or by changing the time signature setting in the Transport window. In the latter case, the signature change will be inserted at the beginning of the measure containing the current song position



Time signatures and time signature changes are effective globally for all instruments, on all display levels. It is not possible to insert different time signatures for different instruments, or tracks. Time signature changes affect only the display, not the playback. The tempo indication in the Transport window always relates to quarter notes, even if eighth notes are chosen as the denominator for the Time signature. You can find more information about this in the section *Time Signature and Divisions (6)* on page 2-8.

Compound Time Signatures



Apart from 2/4, 3/4, 4/4, 6/8 and the alternative display options for 4/4 and 2/2 there are also Partbox objects for freely definable time signatures (A/B), and compound time signatures ($A+B/C$). When you insert one of these, a dialog

box opens:

The *Nominator* can be anything between 1 and 32, the Denominator can be 1, 2, 4, 8, 16 or 32.

The ***Beat Grouping*** entry field will open when you select $A+B/C$: here the grouping of the beats in compound time signatures can be determined. It is sufficient to just enter the numbers, i.e. "223". As soon as the entry field is closed, this becomes "2+2+3" automatically. The total number of beats in the bar is calculated automatically by the sum in this field. ***Print Compound Signature*** actually displays the defined grouping as the nominator. If this is not activated, only the total sum will be displayed as the nominator. ***Beat Grouping*** however, in both cases, affects the way the notes are beamed together.



With this, you can also change the automatic beaming in "normal" time signatures. In 4/4-time "1+1+1+1" produces four beamed groups (one for every beat), instead of the two groups displayed as a default

Invisible Time Signatures

With ***Hide Signature***, the corresponding time signature change will be hidden in the printout. The signature will still be displayed on the screen, but it will be "crossed out", and will not be included in the calculation of the spacing between notes and symbols. Because of this, the signatures may overlap notes on the screen, but not in the printout, since they are invisible there.



This function can be used for notating cadenza-like passages that contain more notes than would normally fit in accordance with the time signature.

Editing Time Signatures

All time signatures can be edited by double-clicking on them. The same dialog box as above opens, where you can insert the desired changes. There is also a *Delete* option, to delete the corresponding time signature change. Another way to delete them is by using the Eraser tool, or selecting them and pressing .

The display of time signatures and time signature changes also depends on the settings in **Layout > Clefs & Signatures**. Refer to the section *Clefs & Signatures* on page 13-133.

Moving and Copying Time Signature Changes

Since time signature changes are global, they cannot be moved or copied as easily as other objects.

Moving time and key signature changes is possible, to a certain extent in the Arrange window. To do so, use the **Functions > Cut/Insert Time** functions, but do *not* select any sequences while performing this edit. This results in the movement of all time and key signature changes, and all other global symbols (repeat symbols, double bar lines etc.), while the sequences and their data (notes, other symbols etc.) stay where they were. For a detailed description of this function, please read the section *Adding or Removing Passages* on page 3-75.

All time and key signatures from one song can be copied into another song. This is described in the section *Copying Time Signatures between Songs* on page 2-10.

Repeat Signs and Bar Lines



These objects can be inserted at any “visible” bar position: at the position of any graphically displayed note or rest). Usually, however, these objects are used at the beginning or end of bars.



In this case, they replace the regular bar line (except first and second ending symbols). They are global symbols, so they always are displayed in all staves, on **all** display levels.

Moving, Deleting

Repeat signs and barlines cannot be moved with the mouse. To delete them, you have to use the Eraser tool. These symbols can be moved, together with time and key signatures to a certain extent, in the Arrange window (see the two preceding sections).

The “double repeat sign” automatically is displayed as two “back to back” repeat signs, if it is positioned at a line break.



Invisible Barlines

The dotted bar line becomes invisible in the printout. In conjunction with hidden time signature changes, this enables you to create bars and musical passages which seemingly do not have a fixed meter. Another use for the invisible bar line is to force a graphic split of a note into two notes connected with ties, in all staves of a score simultaneously, if it is inserted in the middle of a bar.

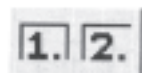


To hide *all* automatically displayed barlines of a song, there is the option *Hide All Barlines* in **Layout > Clefs & Signatures**.

However, manually inserted barlines, double barlines etc. will still be displayed and printed.

1st & 2nd Endings

After inserting one of the repeat endings, text entry mode is automatically activated. The default text is "1." and "2.". If you don't need anything else, confirm with . However, it is possible to enter any text into these brackets. The bracket which defaults to "2." is open-without a vertical end line, like the "1." ending. Since the text can be changed, you also can create a 2nd ending with a vertical line, for example.



Font, text size and style for these numbers or text elements are set in the Text Style window (Text Style *Repeat Endings*).

To edit the text, double-click directly on it, which activates the text entry mode, indicated by a flashing text cursor.

A 1st or 2nd ending can be moved by dragging its upper left corner. The length can be changed by dragging the right end. If a repeat ending is selected, it can be deleted with .

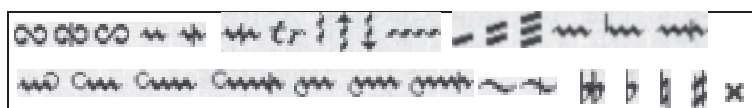
Between the staves in scores, or multi-staff Score Styles, repeat signs and the various special bar lines are connected across the staves, according to the settings in the Instrument Set and/or Score Style windows. First and second endings are only displayed above staves whose bar lines are not connected to the next higher staff.

For the display of repeat signs, there is an option *Alternate Repeat Symbols* (for the whole song) in **Layout > Global Settings**. This activates a “Real Book Style” display of repeat signs.



 Note: these symbols have no impact on MIDI playback

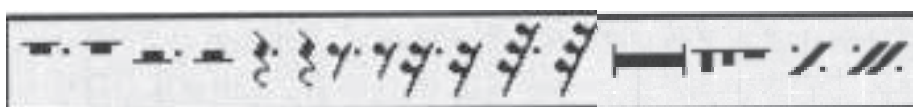
Trills, Tremolo, Arpeggio etc.



The symbols in this group can be positioned anywhere, and are not attached to notes. The trill line and the arpeggio lines can be set to any length, by dragging their end points. The trill line also can go across line breaks (see lengthening of lines, page 13-47). The accidental symbols in this partbox group are purely graphic, and do not affect MIDI playback of notes. They are intended mainly as a supplement to trill symbols.



Rests, Bar Repeat Signs



Automatic Rest Display

Rests are displayed automatically in Logic, in such a way that existing notes and automatic rests together always add up to complete bars, according to the given time signature. The only exception to this is, when a Score Style with deactivated rest display (*Hide Rest*) or with *Beat Slashes* instead of rests is used.

Manually Inserted Rests

In certain situations, you might want to insert rests manually. These replace the automatic rests, but unlike automatic rests, are listed as events in the Event List. These events have a MIDI channel and a velocity value. The velocity value determines their vertical position in the staff (normal middle position is 64).

Manually inserted rests, unlike automatic rests, can be moved higher or lower. Their size can be changed with the Sizing tool.

Example

In the following example, a polyphonic Score Style is used, whose second Voice's rest parameter is set to hide. This is because the polyphonic passages only appear occasionally, and one does not want to see double rests all the time. In this case, the manually inserted rests have to have the same MIDI channel as the second voice:



Two rests have been inserted manually in this example: the eighth note rest in the pickup (with the second voice's MIDI channel), and the eighth note rest on the fourth beat of the first bar (with the first voice's MIDI channel: an automatic rest here would be displayed in the middle of the staff, like the quarter note rest in the second bar)

You can also use manually inserted rests (also called “User Rests”), if you don’t like the way a particular rest is displayed

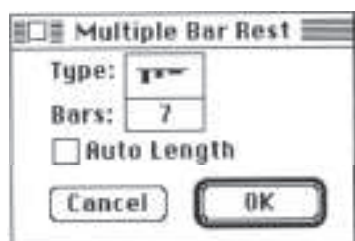
automatically, for example to avoid dotted rests, or for syncopated display of rests despite a *Syncopation off* setting. In terms of rhythmic display, user rests behave like notes, with the attributes *Syncopation ON* and *Interpretation off*. Remember that user rests are interpreted according to the display quantize setting (*Qua*), just as notes are.

Multi Bar Rests

Rests with a duration of several bars, can be indicated with these two symbols. These are suitable mainly for single instrument parts. In the full score, they are visible only when all staves contain multi bar rests at the same position.



“Church rests” (the older form of display) default to a length value of 2 bars. The default length of “normal” multi bar rests is *Auto* (which in the Event List corresponds to value 0). This means that the rest’s duration is calculated automatically according to the position of the next note, user rest or global symbol (i.e. repeat signs, double bar lines etc.).



Changes can be made in the dialog box, which opens when you double click the symbol:

Here you can select the shape (“modern” or “church”), activate or deactivate the automatic length function, and determine the length, if *Auto Length* is switched off. Note: “Church rests” cannot be longer than nine bars. If this is exceeded, the “modern” symbol is used automatically.

Moving Multi Bar Rests

These rests cannot be moved with the mouse. However, since they are displayed in the Event List, you can change their bar position there, or in the Event Float window. You can also alter the length of the rest (VAL) in this manner.

Bar Repeat Signs

The symbols for one or two bar repeats replace all notes and rests in the corresponding measures. All other symbols remain visible. This does not affect MIDI playback, so the hidden notes can still be heard. These signs can only be moved in the Event List or Event Float.



Text Objects, Chord Symbols

TEXT LYRIC CHORD SEQ INSTR SONG DATE

Since the description of text objects and chord symbols is quite extensive, you will find it in the section *Text: Input and Display* on page 13-146 and the section Chord Symbols on page 13-156.



D.S., D.C., Segno, Coda Signs



These symbols can be inserted at any position in the score. They are graphic symbols, and do not affect MIDI playback. Their size can be changed with the Sizing tool.



Tempo and Swing Symbols



These symbols can be inserted at any position. The font, size and style of the tempo indication itself is determined in the Text Style window (*Text Style Tempo Symbols*). The tempo indication is derived automatically from the playback tempo at the corresponding bar position. Note that the tempo indications in the Transport window and in the Tempo List always



refer to quarter notes, even if a time signature with another denominator is used. So, the displayed tempo is different depending which symbol you use. If in 12/8 time the tempo in the Transport window is 144, this will result in the following tempo symbol:



13.13 Move, Copy and Paste in the Score Edit Window

Moving or copying of complete sequences, or major parts of a sequence should generally be done in the Arrange window.

Still, most of the following described functions in the Score window work similarly to Logic's other editor windows.

However, it is NOT possible to copy, or move the following symbols:

Global symbols, like repeat signs, time, and key signature changes (except in the Arrange window as described on page 13-50) and global text elements (headers etc.)

Symbols directly attached to notes, like accents, fermatas, note heads etc.

Available Methods

Moving/Copying with the Mouse

You can move/copy by dragging the selected objects (with  for copying). This allows you to move or copy any number of notes and symbols to other bar positions and/or pitches. If you want to restrict the movement to just one direction (vertical or horizontal), hold down **Shift** during this process. There is also a

general preference for this: **Options > Settings > Global Preferences > Limit Dragging to one direction in Matrix and Score.**

It is important to watch the *Info Line* during this process. It shows the status (Drag or Copy), the current difference from the original position (*Transpose* and *Timeshift*), and the current cursor position (bar position and-for notes-pitch).



Positioning Grid

Movement along the time axis occurs along a quantization grid, corresponding to the particular sequence's assigned display quantization (*Qua*) in the Display Parameter Box. If a hybrid quantize value is selected (i.e. 16, 24) the grid corresponds to the binary value (26 in this example). If note positions are not aligned with this time grid (for example, because they were recorded in real time), their relative offset is still kept when moving or copying them. This allows you to move or copy unquantized passages within the Score Editor, without losing the original feel of the performance.

Please note: copying and moving notes with the mouse is only possible within the same sequence.

Moving and Copying Several Objects at Once

To move or copy (with *Alt*) several objects at once (notes and symbols), just select all desired objects and drag one of them (preferably the first one, or one positioned at exactly the beginning of a measure) to its destination position. Again, watch the Info Line for exact positioning. The other selected objects will be moved or copied in exact relation to the first one.

It is also possible to simultaneously move or copy objects which are in different sequences/staves. Each event will be moved/copied to its new position, within its original sequence.

Move/Copy with Cut, Copy & Paste via the Clipboard



These functions work the same way as in many other software programs, and in the other Logic windows (see the section *Edit Operations* on page 1-27).

Paste inserts the clipboard's contents (i.e. a note) into the selected sequence at the current song position (as indicated in the Transport window). So, before pasting, you have to set the Song Position Line to the desired destination position.

When you copy and paste several objects at once, the first object is pasted at the indicated bar position, the others in relation to it.

Note: if you copy the contents of a whole bar which starts with a rest on the first beat, the first object is not on one, since automatically displayed rests are not objects, and therefore cannot be copied. In this case, you have to set the paste position accordingly, (The exception is when there is an object such as a dynamic symbol at the beginning of the selection, say at beat one, which is also copied.)

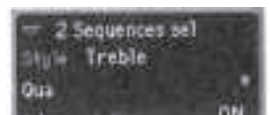
Paste Multiple



The clipboard's contents can be pasted into several sequences at once, at the original position. Select all desired sequences, and use either **Edit > Paste Multiple**, or the key command *Paste Multiple*.



The multiple selection of sequences can be done either in the Arrange, or directly in the Score window: *Shift-click* all desired staves, or "rubber band" the adjacent sequences. Note: the "Display Parameter Box" title line always indicates how many sequences are currently selected.



Automatic Insert Quantization

When you *Paste* (or *Paste Multiple*) objects from the clipboard, the position of the first inserted event is automatically quantized according to the current global display format (in the Transport window). If, for example, the current bar position is 4.1.1.37, and the global display format is set to 1/16, the pasted object will be inserted at position 4.1.1.1, whether it's a note or a symbol. When the clipboard contains more than one object,

the other objects' positions are not quantized, but their exact relative position from the first object is kept.

If you don't want this (i.e. because you want to keep the unquantized timing of the original preserved), it is better to use one of the other methods of copying.

Paste at Original Position / Paste Multiple at Original Position



These two commands (**Edit** menu, or key commands) allow you to quickly copy passages from one sequence to another, at exactly the same time position. Here, you do not have to think about the insert position at all, and the automatic insert quantization will not be applied.

This is the ideal method for copying a whole musical phrase, including all symbols, to the staves of other instruments. Remember that for Paste Multiple..., all destination staves have to be selected.



These commands can be very useful when used in combination with Select Similar, for example when copying all dynamic symbols from one staff into other staves/instruments.

Repeat Objects



This dialog box is described in detail on page 3-21. It can be opened from the local **Edit** menu, or with a key command. In the Score window, it is especially useful for copying complete bars (or multiple copies), without having to set any bar positions. For example, in the case described above, where a bar begins with a rest on one. In that situation, select all notes and symbols from that bar (rubber band selection), open **Repeat Objects**, insert the number of desired copies, set **Adjustment** to **Bar**, and press

Copy MIDI Events

This dialog box is described in detail on page 3-77. It can be opened with a key command, and gives you many options, such

is merely moving, rather than copying events. In the Score Editor, these operations affect all selected sequences.

Nudge Key Commands



Under the name *Nudge Event Position by...*, there is a whole group of key commands, which move all selected objects by various steps along the time axis. For Score work, the following Nudge commands are the most useful:

Nudge Event Position by Bar +1/-1: move event(s) forward/back one bar

Nudge Event Position by Beat +1/-1: move event(s) forward/back one beat

Nudge Event Position by Format +1/-1: move event(s) forward/back one Format step (as set in the Transport window)

13.14 Editing or Deleting Notes and Symbols

This section primarily covers the editing of actual MIDI playback parameters, and object positions. It does not cover score display changes caused by alteration of the different display options (Score Styles, Instrument Sets, Song Settings, Note Attributes). Those are described in detail in subsequent sections of this chapter.

Besides editing bar position and pitch (or vertical height) directly using the mouse, there are some other methods for editing the parameters of notes and symbols:

Editing Using Tools

Velocity Tool



You can grab one or several notes with the Velocity tool, and move the mouse up or down, you change the notes' velocity values.

The value change can be seen in the *Info Line*, and can also be heard if the MIDI Out button is activated.

You don't have to go to the Toolbox to get the Velocity tool: when you grab a note while holding down  , the cursor becomes the Velocity Tool. Once you release these modifier keys, the cursor takes on its previous shape and function again.

Quantize Tool



For Quantizing and De-Quantizing the MIDI playback of selected notes. Read the section *Note Quantization* on page 1-41 for details.

Layout Tool



With the Layout tool, you can change the horizontal graphic position of a note in relation to its bar position. Pulling the note to the right or left with that tool changes the displayed distance from the adjacent notes or rests. The bar position (and the playback) will remain unaltered. This is used for example, to create space for a grace note, or for an arpeggio line. Also, when symbols are moved with the Layout tool, their bar position will not be altered. Instead, the hor.pos and vert.pos parameters are changed, which is indicated in the *Info Line* during this process.

There is a shortcut for grabbing this tool: pressing  while grabbing the note converts the cursor into the Layout tool.

Sizing Tool



With the Sizing tool, you can change the size of any note, and of most symbols. If you want to reset the object to its original size, double-click on it with the Sizing tool. A dialog will appear asking if you want to reset the size of the object.

Changes in the Event Parameter Box

In the Event Parameter Box, you can edit the following parameters (please also note the following explanations regarding editing several selected objects simultaneously):

Notes:

MIDI channel, pitch, velocity, and note length

Note: to change pitch, length and MIDI channel of selected notes, there are also some key commands:

Event Transpose +1/-1: half step up/down

Nudge Event length by Format +1/-1: lengthens/shortens the note by the current global display format value

Nudge Event Channel +1/-1: changes the note's MIDI channel accordingly.



Symbols:

Staff:

Staff number: this parameter is only important when you are using a polyphonic Score Style with multiple staves (i.e. for Piano). It determines which staff a symbol belongs to. Note: if this value is higher than the number of staves in the Score Style, the symbol disappears!



Symbol:

The internal number of a symbol in its Partbox Group: once you know these numbers, you can edit the symbol by changing this number, instead of deleting a symbol and inserting a new one. This is especially useful for dynamic symbols: *0 = ppp*, *1 = pp*, *2 = p*, *3 = mp*, *4 = mf*, *5 = f*, *6 = ff*, *7 = fff* etc.

Examples

Imagine that in a certain bar, you want to change all *Mezzopiano* symbols in all staves of a score to *Mezzoforte*. Select one of the

mp symbols in the score, then select **Edit > Select Similar Objects**. Next, mark this one bar as a cycle using the mouse in the bar ruler, then select **Edit > Deselect Outside Locators**: now that all *mp* symbols in the marked bar are selected, go to the Event Parameter Box, change Symbol from 3 to 4, and you're done. (It takes longer to read this, than to perform this operation, especially when using key commands...).

Vert. pos / Hor.pos

precise graphic positioning (see also page 13-30):

vert. pos determines the height/vertical position. Zero in most cases means a position exactly on the top line of the staff, positive values are above it, negative values below it.

hor. pos determines the horizontal graphical offset of the object's bar position. Zero here corresponds to the left edge of a note at that particular position.

Additional Parameters for Text Objects:

Text Style

Size of the used font

Align: horizontal alignment

Concerning these parameters, please read the section *Text: Input and Display* on page 13-146 for detailed descriptions.



Most parameters in the Event Parameter Box can be edited using the mouse as a fader (hold down the mouse button while moving it up or down). Or, you can double-click on a parameter (except Text Styles and alignment), which opens an entry field. Here, you insert the desired value as a number, or in case of notes, also as a pitch, like D#4. You can also insert relative changes into this field, such as +4 to raise all selected notes' pitches by 4 half steps, or to increase a numerical value of another parameter by 4.



Changing Several Selected Objects Simultaneously in the Event Parameter Box

If several objects are selected, this is indicated in the Event Parameter Box' title line *by X Notes/Symbols/Events/Texts/Chords Selected*. *X* stands for the number of selected objects. The terms *Notes*, *Texts* and *Chords* are used only if the current selection contains only objects of that particular type.



If-as in most cases-parameter fields are visible while multiple events are selected, all corresponding values can be edited at once, "*" as a value means that the selected objects have different values for that particular parameter, There are different options for changing these values:

Relative Change (Preserving Differences):

Grab the "*" with the mouse and use the mouse as a fader. The value shown during this process belongs to the first of the selected objects. The other objects are altered by the same amount. You can also double click the value and insert the desired change (a number with a plus or minus in front of it) into the resulting entry field (see above).

Absolute Change (All Values Set Equal):

Press  during the above described procedure: as soon as you move the mouse while holding the modifier key, the parameter is set to the same value for all selected objects. This way you can set all notes of a chord to the same length or velocity, or set all selected objects to the same height (*vert. pos*).

There are also key commands that set the *vert. pos* and *hor. pos* parameters of all selected objects to the same value:



Align Object Positions Vertically

Align Object Positions Horizontally

Align Object Positions (affects both parameters)

These key commands set all objects to the value of the first selected object.

If only one object is selected when you use one of those key commands, a dialog appears where you can decide if the parameters of all similar objects should be set to the same value. (Note: this affects all displayed sequences and instruments.)



Hint: This is very useful for aligning lyrics and chord symbols.

Deleting Objects in the Score Window

Finally, a short survey of the methods for deleting objects in the Score window:

With the Eraser tool (works for all objects)



With (Backspace) or **Edit > Clear**, while the corresponding objects are selected (flashing). Exception: repeat signs and special barlines cannot be selected, so they can only be deleted with the Eraser tool.



You can only delete events in the Score window which are displayed there. If, for example, you erase notes from a real time recording in the Score window, MIDI controller events or pitchbend data which were recorded together with those notes, will not be deleted.

13.15 The Display Parameter Box

The settings in the Display Parameter Box (with the exception of Style) form the basis for the *rhythmic interpretation and display* of the notes. These settings can be different for every sequence. The displayed settings always refer to the currently selected sequence(s). These settings do not affect the MIDI playback of notes, only the score display.



Different Settings within the Same Staff

If you cannot seem to find the appropriate display parameters for a particular sequence, keep in mind that you can divide a sequence, using the Scissors tool (directly in the Score



window). This will leave the MIDI playback unaltered, but gives you the option to assign different display parameters (e.g. display quantization) for the resulting shorter sequences. Since these divided sequences still connect to each other, they will be displayed as a continuous staff.

Default Settings for New Sequences

If a new sequence is created with either the Pencil tool in the Arrange window, or through MIDI recording, Logic always applies the *Default* parameters to that sequence. These can be seen and edited in the Display Parameter Box, when no sequence is selected (The title line says Insert *Defaults* to indicate this status). These settings will remain as you set them until you quit Logic, but can be changed any time. Click on any empty spot in the main working area (below or above the score), to display the *Insert Defaults*. You can now edit any of the parameters. (for example, to set *Qua* to the display quantize setting that you will use for most sequences in that song) From this point on, all new sequences will get these settings as soon as they are created. If *Qua* is set to *default* here, the *Qua* setting of newly recorded or created sequences will depend on the *Global Format Value* in the Transport window (for details, see below).



The default setting for Score Styles (Style) is not defined here, but separately for each Track Instrument, in the bottom line of the Instrument Parameter Box, in the Arrange window.

Changing the Settings for Several Sequences Simultaneously

This is possible, if all desired sequences are selected. In the top line of the Display Parameter box, you will see an indication of how many sequences or objects (if the selection includes folders) are currently selected (for example 3 *Sequences sel.*), instead of the sequence name. If these sequences have differing values for any of the parameters, a "*" will be displayed in the corre-

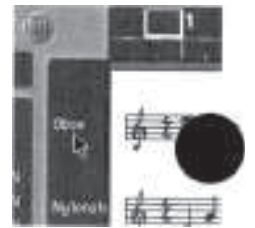


sponding line. If you change this value, all selected sequences will now have the same value for that particular parameter.

There are two special functions for multiple selection of sequences in the Score window:

Clicking on the title line in the Event Parameter Box selects all sequences that are included in the current display.

Clicking on an instrument name in the column between the parameters and score, selects all sequences for that particular Track Instrument (even sequences spread across different tracks assigned to that instrument), at the current display level. Remember, the instrument name column is only displayed, if **View > Instrument Names** is activated.



Rhythmic Interpretation of Sequences

Traditional music notation is only an approximate description the musical content in a piece. The actual performance depends heavily on the interpretation of the notes by the performing musicians. Quarter notes for example, hardly ever are held exactly for the duration of one beat.

The rhythmic interpretation of MIDI sequences recorded in real time to a click presents a similar problem, especially if one considers that Logic records MIDI events with a resolution of 960 ticks per quarter note. Nobody will play a downbeat at the exact time of the click. In some styles of music, it is common to play a bit “laid back”, or behind the beat. So, it is important that you have some idea of what your printed music should look like. This idea will be help you to determine Display Parameter Box settings which will fit your music.

Title Line: Name of the Sequence

A screenshot of a software interface showing a parameter box with the text "Lead Guitar Intro".

The name displayed here is identical to the name of the same sequence in the Arrange window. It can also be edited from here. A double click on the name opens an text entry field. If more than one sequence is selected, this line reads X Sequences selected, if nothing is selected, *Insert Defaults* (the default settings for new sequences).

Style

A screenshot of a software interface showing a parameter box with the text "Treble".

The Score Style used for the score display of the sequence. This is selected from a flip menu, which contains all available Score Styles for the current song.

Score Styles have no influence on the rhythmic interpretation and display, but are essential for Logic's notation concept in general. Their features are explained in the section *Score Styles* on page 13-77.

Qua (Display Quantization)

A screenshot of a software interface showing a parameter box with the text "8, 12".

This parameter determines the shortest note value which can be displayed in the currently selected sequence (exception: "artificial" N-tuplets, see the section *"Artificial" N-Tuplets to Bypass Display Quantization* on page 13=40).

The *Qua* value is selected from a flip menu, which contains the available display quantizations. Among these there are binary quantizations (displayed as *one* binary value like 16 or 128), and hybrid quantizations (two values combined, a binary and a ternary value, i.e. 16, 24 or 32, 96).

The binary values always correspond to the note value with the same denominator, i.e. 32 = thirty-second note etc. Ternary values refer to triplets. Here is a list of the ternary values, and their corresponding triplet values:

"Qua" Setting	Corresponding Note Length
3	1/2-note triplets
6	1/4-note triplets
12	1/8-note triplets
24	1/16-note triplets
48	1/32-note triplets
96	1/64-note triplets
192	1/128-note triplets
384	1/256-note triplets

Automatic Display of Triplets

When binary display quantizations are used, automatic triplets will not be displayed at all. (except for triplets which are inserted with the mouse, using an N-tuplet object). Therefore a "hybrid" quantization value must be assigned to the Qua parameter, to enable the automatic display of triplets.

Default Setting

The *Qua* parameter cannot be set to default in existing sequences, as it could in older versions of Logic. The *default* option can only be used with the Insert *Defaults* (see above), which determine the default settings for newly recorded or created sequences. If *default* is chosen here, the *Qua* setting of any new sequence will depend on the *Global Display Value* in the Transport window. In this case, the *Qua* value will always be the hybrid value which contains the *global display value* currently set in the Transport window plus-in case of a binary global value-the next higher ternary value, or-in case of a ternary global value-the binary value which is divisible by that particular ternary value.

A few examples: *a global value* of 1/8 will result in an 8,12. *Qua* setting for new sequences, global value 1/12 will become Qua = 4,12, global value 1/16 will result in *Qua* / 16,24, 1/24 in 8,24, and so on.

The Display Parameter Box

However, if a particular *Qua* value has already been set in the *Insert Defaults* (which are displayed in the Display Parameter Box whenever no sequence is selected), all new sequences will be assigned this value, regardless of the Global Display Value in the Transport window. Still, any of these values can be changed any time.

Swing Notation

For regular swing notation, 8,12 should be used as the display quantize parameter. This enables the display of eighth-note triplets, and also displays two uneven notes on one beat (dotted eighth and sixteenth), as regular eighth notes. For double-time passages with sixteenth notes, you either have to cut the sequence in the Arrange window, and assign a higher quantize value to the new sequence containing the double-time figure, or use hidden “artificial” N-tuplets for the sixteenth notes (see the section *"Artificial" N-Tuplets to Bypass Display Quantization* on page 13-40).

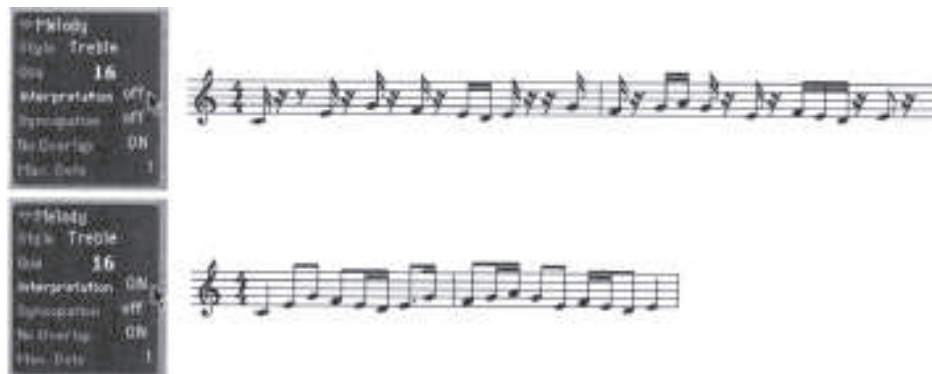
For swinging sixteenth notes (shuffle funk, hiphop etc.), the same principle applies. In this case, *Qua* would be set to 16,24.

Interpretation



If Interpretation is ON, notes are generally displayed with longer length values than their actual length, in order to avoid short rests. Short notes on a downbeat in 4/4 time for example, are displayed as quarter notes. The score becomes less precise, but easier to read.

If Interpretation is *off*, note lengths are displayed as close as possible to their “real” values, as determined by the *Qua* value. In the following example, the same sequence is shown twice, the first time with Interpretation off, the second time ON:



The Interpretation function is intended to produce an easy-to-read score display of real time recordings. When you use step input or mouse input, you should generally switch it off.

Interpretation actually can be switched on and off for every single note, independent from the setting in the Display Parameter Box. This can be done with a menu option, a key command, or using the *Note Attributes* box (see the section *Note Attributes* on page 13-105).

Syncopation



If *Syncopation* is switched ON, each (MIDI) note will be notated as one graphic note (i.e., not displayed as several tied notes), regardless of its position-if this possible. (This also depends on the *Max. Dots* setting, see below) If this is not possible, the note is (only graphically) divided into the minimum possible number of notes, connected by ties.

The usual application for this is the display of syncopated notes. The following example shows the same two bars displayed differently, first with Syncopation off, then ON:

The Display Parameter Box



Syncopation can also be switched on and off for every single note, independently from the setting in the Display Parameter Box. This can be done with a menu option, a key command, or in the *Note Attributes Window* (see the section *Note Attributes* on page 13-105).

If an unwanted display of ties and notes results from the Syncopation function, it is possible to change the graphic display of notes connected with ties. Take a User Rest from the Partbox (preferably a short one, but the length is not really important), and insert it at the barposition where you wish for the tie to be “subdivided”. Once the rest is inserted, it will disappear, but the note display will change. The inserted rest can only be seen and edited in the Event List.

This trick works not only for syncopated notes, but for all notes (see the section *Changing the Way Notes with Ties Are Displayed* on page 13-36). In polyphonic Score Styles, the MIDI channel of the rest, and the corresponding note has to be the same.

No Overlap



No Overlap ON prevents the overlapped display of notes in melodies that are played with an exaggerated legato. The effect can be seen in the following example, which shows the same sequence, first without, then with **Overlap** correction:



No Overlap should only be deactivated in rare cases. If, for example, a piano player arpeggiates a chord and holds the notes, the score displays the following result with *No Overlap* **ON**:



Although you can't see here that all notes continue to sound, if you deactivate *No Overlap*, the result looks like this (which is not much better, really):



The solution typically is to switch *No Overlap* **ON**, and use sustain pedal markings. If the notes are recorded in real time with a MIDI keyboard, using a sustain pedal, Logic will display the pedal markings automatically.

Max. Dots



This parameter determines how many dots Logic will allow for the display of single notes. Unwanted dotted notes or rests can be changed by inserting User Rests (which will stay invisible in the case of notes, see the section *Changing the Way Notes with Ties are Displayed* on page 13-36).

Special Parameter: Score Off



This parameter is not included in the Display Parameter Box. You will find it in the Extended Sequence Parameters, which you open via the global menu **Options > Extended Sequence Parameters** (or key command). If Score is set to *off* here, the sequence will not be displayed in the score at all. This is mainly used to prevent the score display of sequences containing only MIDI events which would not be displayed in the score anyway, like controller or Sys Ex data.

Please keep in mind for all these settings:

The Display Parameter Box settings not only affect the display of notes recorded in real time, but also that of notes which were inserted with the mouse, or using step input. If you insert a short note, such as a 32nd note it can only be displayed at its original length if Qua is set to 32 or shorter. If Qua is set to 8, the 32nd note will be displayed as an eighth note (although it will still be played back as a 32nd note). A note inserted with the mouse may also display as a longer value if Interpretation is ON for that sequence.

13.16 Score Styles

The Concept of Score Styles

Score Styles could be compared to text formats in word processing and desk top publishing software. In this case however, it is score attributes that are stored, not font settings. This makes it very easy to assign multiple attributes; such as clef, staff size, vertical distance between staves, instrument transposition, etc., to any existing or newly created sequence. A Score Style includes the following parameters:

Number of staves (for the display of *one* sequence)

For polyphonic, multi-staff Score Styles: configuration of brackets and bar lines connecting the staves.

For every staff

- Staff size

- Distance to the next higher and lower staves

- Number of independent (polyphonic) voices in that staff

- Clef

- Display transposition

- Key signature on/off

For every independent (polyphonic) voice:

- Automatic rest display on/off

- Stem direction of notes

- Tie direction

- Direction of N-tuplet brackets and numbers

- Beaming

- MIDI channel assignment for that voice and/or definition of a split point pitch for separation of the different voices.

Score Styles do not affect the rhythmic display, unlike the other parameters in the Display Parameter Box.

Also, they do not affect the horizontal distance (spacing) of notes (apart from changes caused by different stave sizes). This is determined in **Layout > Global Format...**, for details see the section *Constant/Proportional Spacing: Note distance* on page 13 - 125).

Score Styles are saved together with the corresponding song file, so there can be different Score Styles in different songs. Hint: create some empty template songs with (among other things) the Score Styles and other score settings that you normally use as a basis for your work with Logic. The demo and template songs from the Logic Support Collection already include many different Score Styles.

Assigning Score Styles

Each individual sequence can be assigned a different Score Style in the Display Parameter Box. This enables you (among other things) to quickly

- create parts for transposing instruments.

- display the same sequence in different sizes, for example, for printout of a full score, and parts for the different instruments.

- change between different forms of display in one staff. This would be done by cutting a sequence, and assigning different Score Styles to the resulting shorter sequences. (This might be used, for example, to alternate between passages that are completely written out, and improvised passages using only *Beat Slashes* and chord symbols)

Working with Existing Score Styles

Assigning Score Styles to Sequences

When you start Logic, the default song that appears already contains many Score Styles that you can use. You assign a Score Style to a sequence by selecting the sequence, then selecting the desired Score Style from the flip menu which appears when you click-hold on the Style Parameter line in the Display Parameter Box.

You can also assign a Score Style to several selected sequences at once (see the section *Changing the Settings for Several Sequences Simultaneously* on page 13-69).

Score Style assignments can be changed anytime.

Reminder: sequences on the same track in the Arrange window which directly follow each other without gaps between them, are displayed as one continuous staff in Page Edit view, but still can use different Score Styles.

Automatic Score Style Assignment for New Sequences



When you create a new sequence, either by making a real time recording, or with the Pencil tool in the Arrange window, the sequence will be assigned the Score Style which is selected in the bottom line of the Instrument Parameter Box of the active track instrument in the Arrange window.

When you create template songs, or when you begin to work on a new song, you may wish to change these settings for those instruments that are likely to always use the same Score Style. That way, those instruments will immediately be displayed with the correct Score Style.

Auto Style

If *Auto Style* is chosen in the bottom line of the Instrument Parameter Box, every new sequence will be automatically assigned a Score Style that fits the register of the recorded notes. *Auto Style* can only be chosen in the Instrument Parameter Box, and does not appear in the list of Score Styles in the Score window's Display Parameter Box.

The Score Style Window

There are several ways to open the Score Style window:

- Double click on the Style parameter in the Display Parameter Box

- Key command *Open Score Style Window*

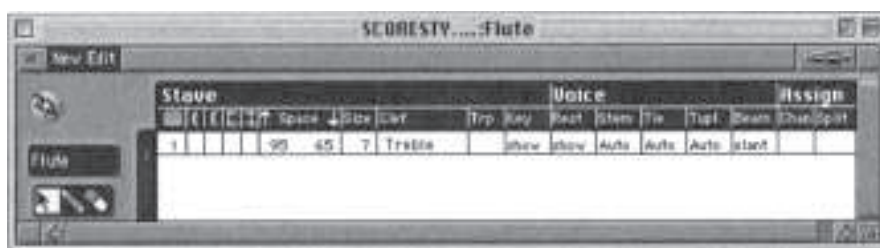
- Layout > Score Styles...**

- Occasionally in the Arrange or Environment window:

- Double click in the bottom line (*Default Style*) of the Instrument Parameter Box.



Score Styles



Along the left side of the Score Style window there are some elements which also exist in other Logic windows:

Link Button

If the Link button is activated while the Score Style window is open, you can always see the parameters of the currently selected sequence's Score Style. If you switch to a different sequence, the Score Style window will show the new selection.



Tool box

This Toolbox contains the Pointer tool (same as the regular mouse cursor), the Pencil tool (for creating new staves and Voices) and the Eraser tool (for deleting Score Styles, staves, or Voices).



Name Box / Style Selector

The field below the Toolbox shows the name of the currently selected Score Style. If you click and hold on it, a flip menu appears, listing all Styles that are available in the current song. With this, you can switch to another Score Style.



The Score Style *Parameters* are grouped horizontally in three main sections (top line): Staff, Voice and Assign.

Staff: The Parameters for staves. Each line below the word **Staff** represents one staff in the score display.

Voice: The Parameters for independent *Voices*. Each line below the word **Voice** represents one independent voice. The term **Voice** here only relates to the simultaneous display of rhythmically

independent, polyphonic passages in the music, not how many notes can be displayed simultaneously. One Voice can still consist of as many simultaneous notes (chords) as desired. Only notes which should be displayed as rhythmically independent from the other notes in the same staff have to be assigned to separate *Voices*.

Assign: The parameters for the assignment of notes to the different *Voices*. For details, see below.

Every Staff can contain several independent Voices (up to 16), however, one Voice can not be displayed across different staves. Therefore, a Score Style has to contain at least as many Voices as Staves.

List View



Besides the single Score Style view shown above, there is also a *list view* option in the Score Style window. The list view is mainly intended for copying Score Styles between songs (see below), or for deleting styles.

To change to the list view, double click in the empty space below the parameters. Another double click on one of the Score Styles in the list brings you back to the single display of that particular Style. Another way to change between single and list view is to click on the small button to the left of the menu line (see illustration).



Copying Score Styles between Songs

At some point, you will probably want to use Score Styles that already exist in another song. There are two methods for copying Score Styles between songs:

Importing All Score Styles from another Song



In the Score window, select **Options > Import Settings**. In the resulting dialog box, activate Score Styles only, and click on Import. This imports all the Score Styles from another song into the current one.

If another song file is already open during this procedure, that song's Score Styles will be imported. If no other song is open, a file selector box will appear, where you can select the song whose Score Styles you want to import into your current song.

Copying only Particular Score Styles from other Songs

One or several Score Styles can be selected in the Score Style window's list view, and copied between songs, using the usual copy/paste commands (**Edit** menu or key commands). To be able to do this, you have to open Score Style windows in both songs.

Automatic Score Style Copying

If you copy a sequence from one song to another, and the Score Style used by the original sequence does not yet exist in the other song file, it is automatically copied into that song, along with the sequence.

Deleting Score Styles

Score Styles can be deleted in both the single view (**New > Delete Style**), and in list view (with **Backspace** or **Edit > Clear**). In list view, you can also select several Score Styles at once for this purpose. For example, select **Edit > Select Unused** in list view, and press **Backspace** to delete all Score Styles that are currently not assigned to any sequences or Track Instruments in the song.

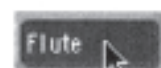
Creating New Score Styles

In single view, you can create new Score Styles using **New > Single Staff Style** or **New > Dual Stave Style**. The parameters in the Styles created this way are very basic, and in most cases have to be edited according to your individual requirements (see below).



Note: there is also an option **New > Mapped Style**, which mainly concerns drum notation, and is described in detail starting on page 13-99.

The default name of Styles created in this way is "***New Style**". A double click on the name box opens a text entry field, where any name can be typed in.



Creating New Score Styles Based on Existing ones

Often a new Score Style is needed which is almost identical to an existing one. In this situation, just make a copy of a Style and edit that copy. Copying can be done:

in the Score Style window (single view) with **New > Duplicate Style**.

with the Score Style window's flip menu (name box), by selecting the bottom entry (*****DUPLICATE!*****) in the list.

Directly in the Score window, by using the Score Style flip menu in the Display Parameter Box, again selecting *****DUPLICATE!***** at the bottom of the list. This is the fastest method, since the new Score Style automatically is assigned to the currently selected sequence. (see below)

In any case, the new Style will get the name of the Score Style it is derived from, with "***copied**" appended to its name. This text can be edited in the name box of the Score Style window.



If the only difference from the original Style is the staff distance or the Style's clef, creating a variation of the currently assigned Score Style can be done very quickly from inside the Score window: with the desired sequence

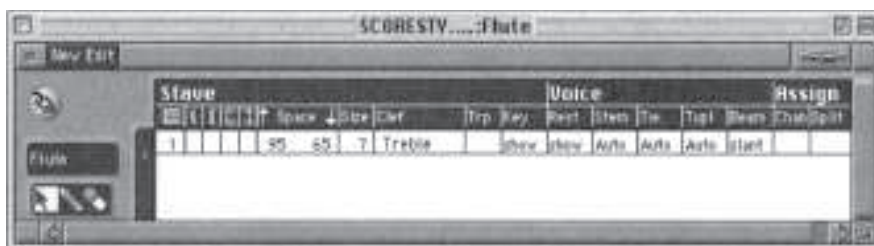
selected, select the ******DUPLICATE!****** command in the Score Style flip menu. This copies the current Score Style, and assigns the copy to the selected sequence. Now, change the staff distance by dragging the clef with the mouse (as in the opposite illustration), or double-click on the clef to select a different clef from the resulting box.

Editing Score Styles

Apart from the situation described above, you will normally have to open the Score Style window to edit Score Styles. The following examples show different types of Score Styles (with increasing complexity) to demonstrate the parameters in the Score Style window:

Simple One-Staff Score Styles

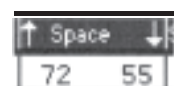
For a simple one-staff Score Style, the Score Style window will look similar to this:



Here only the following parameters are relevant:

Space

These values determine the distance to the next higher and lower staves or, for the top and bottom staves of a full score, to the page margins.

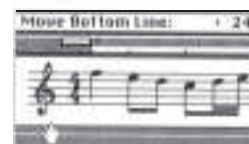


This also creates additional space belonging to the staff above and below the actual staff. If, for example, symbols are inserted between staves in full score mode, it is important which staff an inserted symbol is assigned to. When parts are extracted from the full score, the symbols will go with whichever staff they were inserted to. Hint: watch the **Info Line** in this situation.).

The vertical distance above the staff can also be changed directly in the score, by dragging the staff up or down at its clef.



The distance below the staff can be edited in the score as well, but only for the bottom staff in the score display, or if only one staff is displayed. Here, the bottom line of the score display area is dragged with the mouse, as shown in the opposite illustration.



These editing procedures change the settings of the corresponding Score Style itself. This means that all sequences which use the same Score Style will be affected which is not always what you might want. To prevent this, use the fast method for creating new Styles, based on existing ones, described on page 13-84).

Size

The size of the staff and the notes and symbols it contains. There are eleven available sizes. Here are some size recommendations: for regular instrument parts and lead sheets, use size 7 or 8. The staff size you should use in full scores depends on the number of staves in the score, and the size and format of the paper used. When using A4 or US letter paper: full orchestra: 2, big band: 3, wind quintet: 4.



However, the size of all staves in an Instrument Set can also be affected by the Scale parameter in the Instrument Set window (for details see the section Scaling Parameter for Instrument Sets on page 13- 118). Therefore it is possible to use the same sizes both for score and parts of a piece.

Clef

Here you select the clef which is used in the Score Style from a flip menu. Besides the usual clefs, this menu also offers some special options:



Drum.0 to **Drum.8**: staves with 0 to 8 lines and a “neutral” percussion clef. The relation of MIDI note pitches to the top line of all these staves corresponds to the top line in regular bass clef (AZ). However, no accidentals will be displayed in

these drum staffs. **Drum.0** also omits any ledger lines. These clefs will often be used in Mapped Drum Styles, where the height of a note does not correspond to pitch, but is determined by different Drum Map parameters. More on drum notation can be found in the section **Drum Notation with Mapped Score Styles** on page 1399.

- *no clef.0* to *no clef.8*: like *Drum...*, but without any clef.
- *TAB Guitar..* and *TAB Bass..* :12 different options for displaying notes as guitar or bass tablature. These tablature **TuningSets** are defined in the tablature window (**Layout > Guitar Tablature...**), where they also can be edited. For more information, please read the section **Guitar Tablature** on page 13-137.

Trans

The display transposition, measured in half steps up or down. This parameter does not affect MIDI playback. If a staff contains chord symbols, these too will be transposed accordingly. If **Automatic Key Transposition** in **Layout > Clefs & Signatures** is activated (which is the default setting), the key signatures will also be transposed (exception see next paragraph).



Key

If this parameter is set to **Hide**, the corresponding staff is displayed without any key signature. Instead, all sharps and Flats are indicated directly next to the notes. This parameter will usually be set to Show the key signature. **Hide** is an option mainly used for transposed french horn parts, which are sometimes written without any key signature.



 Note: the above mentioned option **Automatic Key Transposition** has a similar effect on the whole song. There however, the untransposed basic key signature is still indicated, whereas Hide Key suppresses any key signature.

Rest

The automatic rest display can be switched off here (HIDE), or set to display **Beat Slashes** instead of rests. This is useful for Rhythm section parts and improvised solos. In this case, the



number of slashes per bar will be determined by the nominator in the time signature (i.e. 4 slashes in 4/4-time, 6 slashes in 6/8-time etc.). Inserted notes and symbols will still be displayed, but automatic rests are only shown if they are shorter than one beat. You can insert User Rests manually, to replace the Beat Slashes at certain points.



Stem

This controls the direction of stems. The default setting is *Auto*. *Up* or *Down* forces all stems, regardless of pitch, into the corresponding direction. *Hide* makes all stems (and with them beams and flags) invisible.



Tie

The vertical direction of automatically displayed ties. The default setting is *Auto*. *Up* or *Down* forces all ties into the corresponding direction.



Tupl.

Controls the direction of N-tuplet brackets and numbers. The default setting is *Auto*. *Up* or *Down* forces all N-tuplet brackets and numbers into the corresponding direction. *Hide* prevents the automatic display of triplet brackets and numbers. In this case, the numbers are still displayed in brackets on the screen, to allow editing of these triplets. On the printout however, these bracketed numbers will not appear.



If you set this parameter to *Hide*, you can still make particular triplets visible: double-click on the bracketed number (3) to open a triplet dialog box. If you now close this box with OK, you convert the automatically displayed triplet into a “forced” triplet, which will be displayed according to the parameters set in the dialog box.

Beam

Controls the appearance of beams. The default setting *Slant* allows slanted beams (corresponding settings can be found in **Layout > Global Format**, see page 13-127). *Horiz.* allows only horizontal beams. *Vocal* prevents any display of beams, as is



Score Styles

typical in “classical” vocal parts, where notes are displayed with flags only.

The Name of a Score Style can be edited by double clicking on the name box, which opens an entry field.



The other parameters are only relevant if the Score Style consists of more than one Voice, which does not refer to the display of chords etc., but to rhythmically independent, *polyphonic* voices.

Simple Two-Stave Score Styles

(2 staves with one Voice each)

The most obvious example for a multi-staff Score Style is a two-staff piano system. Here is a simple version of that setup:



When working with multiple staves and/or Voices in a Score Style, the following parameters also become relevant:

Number of Staves

In the first column each staff is represented by a number. These numbers are assigned automatically and cannot be changed. An *additional staff* can be created in two ways:



New > Insert Staff. The new line is inserted at the position of the insert mark (>), which can be positioned with a mouse click in the narrow column to the left of the staff numbers.

Click on the lower left portion of the empty space below the parameters (below the Staff parameters) with the Pencil tool.

Brackets and Bar Line Connections

In the second to fifth columns of the Score Style window, you can determine which staves are bracketed (two bracket types available) and/or connected by bar lines (either at the beginning of each staff line only, or at every barline). For any of these connections, just click-drag the mouse in the corresponding column until you see the appropriate display. If a Score Style consists of more than two staves, these brackets and bar lines can be set to connect only certain parts of the whole system, i.e., they can be interrupted in between staves. If you want to delete any of these, grab the symbol at its end, and drag it up to the beginning, until it disappears.



The bar line connections can also be edited directly in the score. Just click on the upper end of a bar line to connect it to the next higher staff. To disconnect the bar lines, simply click on the same spot again.

The *Voice*-Parameters (which have been explained in the previous section) in the middle of the window (Space, Size, Clef, Rest, Stem etc.) can be set individually for each Voice (which in this case is identical with staff). However, in a simple two-stave system, you probably will just select two different clefs and adjust the staff distances as needed. The Space parameters now also refer to the staff distances inside the Score Style itself.

↑ Space	↓ Size	Clef
90	30	5 Treble
30	85	5 Bass

Voice Separation Methods

Additional staves are always treated as additional Voices. There are two different methods for assigning notes to the different voices. The parameters for these are to the far right side of the Score Style window, below the *Assign* header:

A fixed split point (between two lines in the *Split* column).

This is a defined pitch, which determines the absolute border between two Voices. Notes above it or exactly at that pitch are assigned to the upper Voice, notes below it are assigned to the lower Voice.

This method is very simple, but there is the disadvantage



that none of the Voices, even briefly, can ever cross that border. Quite often in “real life”, the border between voices needs to be flexible. A piano piece’s right hand part may dip below middle “C”, or the left hand may go above. It is not possible to notate this properly using a fixed split point.

Voice assignment according to the notes’ MIDI channels (Chan): for each Voice, a different MIDI channel can be set in the Chan column.

Assign	
Chan	Split
1	
3	

Every note is now assigned to a Voice according to its MIDI channel. (The note’s MIDI channel can be seen and edited in the Event List, Event Float or in the Score window’s Event Parameter Box, when that note is selected.) The different MIDI channels have no impact on MIDI playback, since the playback channel is determined by the setting in the Instrument Parameter Box (Arrange window).

Since MIDI channels are used for voice assignment, notes that do not have one of the channels used for Voices are not going to be displayed at all. This can, however, be used to your advantage, as it allows you to exclude certain notes from the display (i.e. improvised parts, trill notes etc.)

There are two useful functions that can speed up the process of changing the MIDI channel settings of notes, in order to assign them to the desired Voices:

In the Score Preferences (**Options > Score Preferences**) there is a parameter called *Auto Split Notes at ...* If this is activated, and the default Score Style in the Instrument Parameter Box for the selected instrument is a polyphonic Style, the played notes are automatically assigned MIDI channels according to the Score Style’s Voice assignments. Notes at and above the split point get the channel of the first Voice, those below get to the second Voice’s channel. (This feature only works for two Voices at a time) This creates a useable rough Voice assignment, which can be edited afterwards. For sequences that have already been recorded, use **Functions > Note Events > Split to Channels**, or the key command



Split to Channels, to assign notes to Voices in the Score Style according to the *Auto Split* pitch in the Score Preferences.

In most cases, it will still be necessary to change the Voice assignment (the MIDI channel) for particular notes, or groups of notes. There are several methods for doing this:

Editing the MIDI channel in the Event Parameter Box, the Event List or the Event Float window (works also with multiple selection).

Changing the MIDI channel using the key command *Nudge Event Channel +1* and *Nudge Event Channel -1*.

Hint: these can be especially useful when used in combination with the *Select Next/Previous Event* commands, which allow you to move the selection from note to note.

Using the *Voice Separation tool*: this tool allows you to draw a separation line between the notes, thereby assigning them to the different MIDI channels of the Voices. Place the tool between the staves, and draw a line where you want to separate the Voices, while holding down the mouse button. The following example shows the resulting separation line. If you make a mistake, just move the mouse back to the left a little bit (with the button still pressed). This erases the separation line to the right of the tool. Then try it again.



The result of the procedure will look like this:



Note: the Voice Separation tool can only be used if MIDI channels are defined for each of the Voices that you want to separate with it, since Logic has to know which MIDI channels to assign to the notes,

Voice Assignment when Using Mouse Input

If you insert notes with the mouse into a multi-staff Score Style, these are assigned automatically to the staff/Voice/channel into which they are placed. If that staff contains several Voices (see below), the MIDI channel corresponds to the channel setting in the *Insert Defaults* (but only if that particular channel is used by one of the Voices of that staff).

If you insert *Symbols* into multi-staff Score Styles, there is a *Staff* parameter in the Event Parameter Box for most of them. It determines which staff the symbol belongs to. This is also relevant for the *vert. pos* setting of that symbol.

Single-Staff Polyphonic Score Styles

(With Several Independent Voices)

Polyphonic Score Styles are needed when rhythmically different melodic lines should be displayed in the same staff. Theoretically, up to 16 independent Voices are possible.

There are two possible ways to add another Voice to a staff in the Score Style window:

Menu New > **Insert Voice**. The new Voice will be inserted at the position of the insert mark (>), which can be set with the mouse in the narrow column to the left of the staff numbers. (This is important if you want to insert a new Voice between existing Voices)

By clicking with the Pencil tool into the (empty) lower right portion of the Score Style window. In this case, the new Voice is simply added at the bottom.

In both cases, the newly inserted Voice will have the same parameters as the Voice above it, which can now be edited



process. This is indicated by two horizontal arrows between the numbers of the Voices (see illustration below), and is only relevant if there are more than two Voices. To switch to the lower Voices, the arrows can be moved with the  key while the mouse button is pressed:

Voice Separation: 1  2 3 Press "Shift" to choose voice

Input and Recording of Polyphonic Voices

Via MIDI

If you record polyphonic voices as separate passes in real time, you should set your keyboard or MIDI controller to the appropriate MIDI channel for each Voice. This way, you don't have to edit the channels afterwards. The same applies to step input.

Note: the original MIDI channel will not be recorded properly, if in the Environment window, the Sequencer Input Object's Channelize function is activated.

Reminder: if you want to record an additional Voice into an existing sequence, that sequence has to be selected before the recording is started, and in the **Recording Options** (global menu **Options > Settings > Recording Options**), the option *Merge New Recording With Selected Sequence* has to be activated. You could also record two separate sequences, using different MIDI channels, and merge them afterwards.

Mouse input

Mouse input into polyphonic Score Styles is very easy, if you activate **View > Explode Polyphony** (also available as a key command). This causes all Voices to be displayed in separate staves, no matter what the Score Style Staff settings are (the other Voice parameters remain valid). If you insert a note into one of these staves, it is automatically assigned the corresponding MIDI channel. After you finish input, turn off the *Explode Polyphony* option. Now the voices will be displayed correctly in one staff, according to the Score Style settings. This mode is generally useful for all editing in polyphonic staves. The



following illustration shows the same sequence as the previous two examples, this time with *Explode Polyphony* activated:



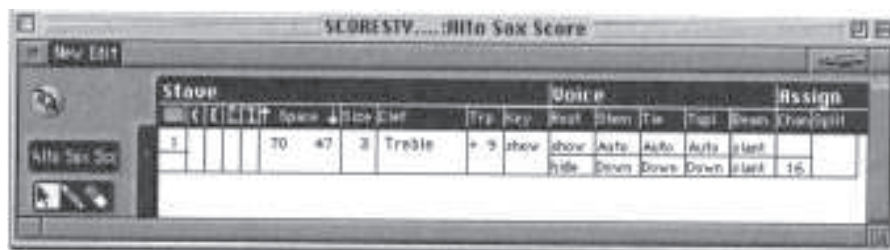
Of course you can also insert notes directly into polyphonic staves without activating the Explode Polyphony option. In this case, an inserted note gets the MIDI channel which has been set in the *Insert Defaults* (Event Parameter Box). (See the section *MIDI Channel and Velocity of Inserted Objects* on page13-28).

Single-Staff Score Style with Optional Polyphony

Sometimes you need to add a second Voice to a staff that usually only requires one Voice, in order to properly notate a short rhythmically independent passage. This is not uncommon with orchestral scores, where the unison part for a group of instruments will temporarily separate.

One solution is to cut the sequences and use different variations of the same Score Style, with and without polyphony, for the different sequences on the same staff.

Another solution is to use a Score Style with a “Main Voice” and a “Secondary Voice”. The parameters for the Main Voice should be set like in a non-polyphonic Style, the Secondary Voice is displayed without automatic rests:



Score Styles



The stem, tie and tuplet directions for the Main Voice are set to **Auto**, which is necessary for the proper notation of the unison parts. No MIDI channel is defined for the Main Voice, so all notes with a channel other than 16 will be assigned to the Main Voice. As long as no notes or rests with MIDI channel 16 are used, the score display looks the same as with a simple non-polyphonic Score Style.

For the second Voice, the stem, tie and tuplet direction parameters are set to down, and the automatic rest display is deactivated (Hide). Only notes with channel 16 will be assigned to this Voice. To achieve a score display like the one in the previous example, the following additional steps are necessary:

Rests in the second Voice have to be inserted manually where needed. In the example, this was done on the fourth beat of the second measure. The rest's MIDI channel has to be the same as the second Voice's channel (in this case, 16). This can be edited in the Event List.

In the polyphonic passages, the stems of the Main Voice have to be forced upwards by selecting the corresponding note's and changing their individual stem parameters (see *Note Attributes*).

The advantage of this approach is that you don't have to worry that much about MIDI channels during recording or input. Also, notes will not disappear unintentionally. There is, however, the disadvantage that the Voice Separation tool cannot be used in this case to assign notes to Voices.

Other applications for this method would include the display of beamed grace notes, occasional polyphony in a guitar-part, or the indication of rhythmic accents above or below beat slashes in rhythm section parts.

Optionally, you can add still more Voices to a staff to be able to display more complex parts (up to 16).

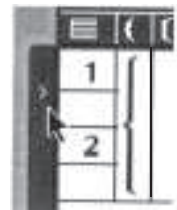
Multi-Stave Polyphonic Score Styles

(With more Voices than staves)

These Score Styles are used mainly for notating keyboard music. If you start Logic without using an Autoload-Song or template song files, an empty song appears, which contains some polyphonic Score Styles for piano (*Piano 1+2/3+4*) and church organ (*Organ 1+2/3+4/5*, with three staves)

These Score Styles are simply a combination of the Score Style types covered in this section so far. To insert an additional Voice or staff, place the Insert Mark (>) at the desired position with the mouse (see opposite illustration), and select **New > Insert Voice** or **New > Insert Staff**.

In these complex Score Styles, it is often best to use the *Explode Polyphony* mode when editing. (see page 13-95).



Copying Voices or Staves into other Score Styles

With the Copy&Paste-commands (**Edit** menu or key commands), you can transfer one or several voices and/or staves including their parameters, into other Score Styles. Select the Voices you want to copy, by dragging the mouse vertically in the margin column to the left of the stave numbers. The selection will be indicated by a black vertical beam (see illustration). Now, select **Edit > Copy** (⌘C). Switch to the Score Style into which you want to paste these voices (this also works between different songs). Finally, set the Insert mark (>) and select **Edit > Paste** (⌘V). Note that if a Voice is selected in the destination Score Style (indicated by a black beam in the left column), it will be replaced by the pasted voices.



Deleting Voices in a Score Style

To delete Voices or staves from a Score Style, select them (black beam, see above) and press **Backspace**, or select **Edit > Clear**.

Combination of Split Point and MIDI Channel Assignment for Voice Separation

If a Score Style contains more than two Voices, the different Voice assignment methods can be combined, as in the following example:



There is a *split point* (C3) between the two staves. *All* notes with channel 16 however (independent of their pitch), are assigned to a separate voice, which is displayed in the lower staff. The MIDI channel always has priority over the split point.

Drum Notation with Mapped Score Styles

Nowadays, most MIDI instruments contain a variety of drum and percussion sounds. Each MIDI note usually triggers a different sound. If a sequence containing a drum part is displayed with a regular Score Style, you see notes with no apparent relation (except that these MIDI notes trigger the corresponding sounds) to the sounds represented by them.

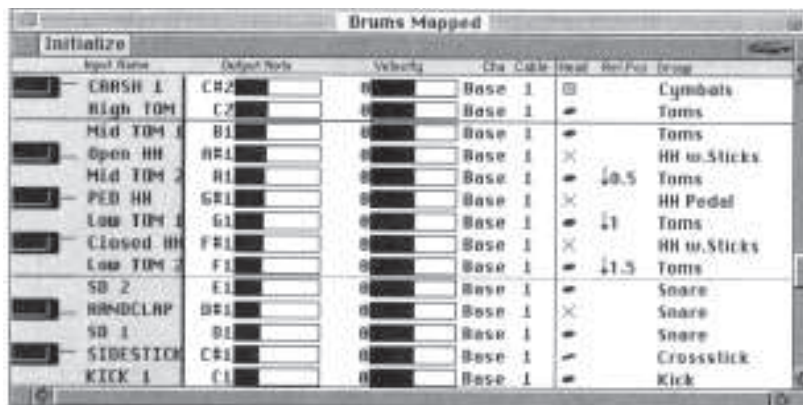
If you wish to notate these musically meaningless pitches as a readable drum part which also uses special percussion note-heads, then you will want to utilize Logic's *Mapped Instruments* and *Mapped Score Styles*.

The Basics of Drum Notation with Logic

For the purpose of drum notation, you have to use both a *Mapped Instruments* as a Track Instrument, and a *Mapped Score Style*, for the notation to be displayed properly. We'll start with the *Mapped Instrument*.

Open an Environment window, and create a new *Mapped Instrument* and double click its icon.(For more information on the MIDI features of Mapped Instruments, read the section *Mapped Instruments* on page 5-34).

You will see the *Drum Map Editor* for the instrument. The default settings correspond to the *General MIDI* drum note assignments, but they can be edited, as they have been in the following example:



In the column to the far right, there are three parameters which are relevant for notation:

Head

Determines the shape of the note head for notes triggered by this particular pitch. You can choose the shape from the flip menu, which appears when you click and hold on any of the note heads displayed here.



Group

Here, each MIDI note can be assigned to a *Drum Group*, using a flip menu. For the most commonly used drum sounds, there are already some groups defined in this menu: **Kick**, **Snare**,

Hihat, Toms, Cymbals etc.). If you want to define a new **Drum Group** for another instrument sound (for example *Tambourine*), select one of the New **Groups** in the flip menu, and double click on it. This opens an entry field, where you can determine a name for the new **Drum Group**.



The Drum Groups play an essential role in “Mapped Score Styles”: a note has to be assigned to a Drum Group to be displayed with a Mapped Score Style. If its not, it won’t be visible.

Rel. Pos.

This parameter assigns the note to a line in the staff. It defines the note’s position in relation to the top line of the staff. Integer values will make the note fall onto a line, fractional values result in a note position between two lines. (The measurement unit of this parameter is the distance of one staff line to the next)

Rel. Pos.



This position parameter can be set individually for each MIDI note. In the Score Style window, the positions can also be influenced, but only for all notes of a particular Drum Group together.

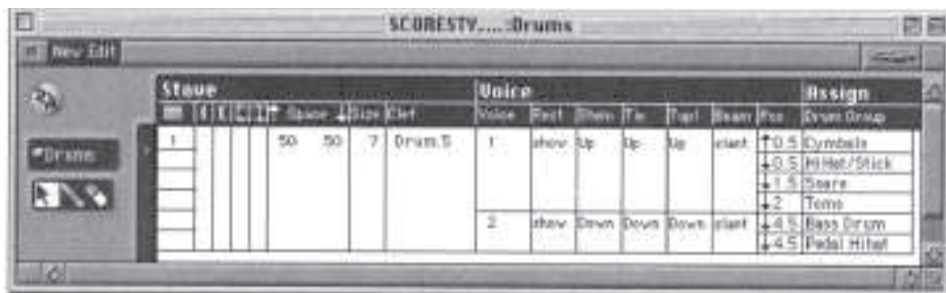
These options allow you to have two different MIDI notes (i.e. different bass drum sounds) displayed in the same way in the score, or on the same line, but with different note heads.

Different Drum Maps within the same Song

If you are using various MIDI instruments with different drum assignments, you can create a separate **Drum Map** (in the **Drum Map Editor**) for each instrument. The list of **Drum Groups** however, is the same for all instruments in a song. This enables you to display different drum tracks with the same **Mapped Score Style**. All **Snare Drums**, for example, will be displayed in the same way. If you don’t want this, just create additional drum groups like **Kicks**, **Snare 2** etc. and create a second **Mapped Score Style** for the display of these **Drum Groups**.

Mapped Score Styles

Take a look at a typical Mapped Score Style in the Score Style window first:



To the left side (*Staff*), everything is identical to non-mapped Score Styles (with the exception of the missing *Tram-pose* and *Key* parameters, which wouldn't make sense here). The chosen clef is Drum.5, a five-line staff with a percussion clef.

Below *Voice* (in the top header line), there is a separate Voice column, where the different Voices are numbered automatically. The Score Style in the example above contains one staff with two independent Voices.

The hierarchy *Staves-Voices-Drum Groups*, goes from left to right, and is shown by horizontal lines, which indicate the borders between these elements: each *Drum Group* belongs to the Voice at the same horizontal position. The *Pos.* parameter affects the vertical positioning of all notes of the corresponding *Drum Group*. The value shown here is an offset, which is added/subtracted to the relative positions defined for the individual notes in the *Drum-Map-Editor*.

The following example (shown both as a score excerpt and Event List) uses the settings of both the *Drum Map Editor*, and the *Mapped Score Style* shown in the examples above:

Score Styles

PART	INST	NAME
1	1	KICK 1
1	1	CRASH 1
1	1	Closed HH
1	2	SD 1
1	2	Closed HH
1	2	Closed HH
1	2	KICK 1
1	3	KICK 1
1	3	Closed HH
1	3	KICK 1
1	3	Closed HH
1	4	SD 1
1	4	Closed HH
1	4	Mid TOM 1
1	4	Mid TOM 2
2	1	KICK 1
2	1	Closed HH
2	1	Closed HH

Take a close look at this example, and compare the score display, the Event list, the *Drum Map Editor* on page 13-100 and the Mapped Score Style on page 13-102:

The corresponding Mapped Score Style consists of two Voices in one staff. *Voice 1* contains Drum Groups *Cymbals*, *HH w. Sticks*, *Snare* and *Toms*, its stems are forced upwards. *Voice 2* contains Drum Groups *Kick* and *HH Pedal*. Its stems are forced down. Both voices display *Beams* horizontally only (*Horiz*). Automatic rest display is activated for both.

The Hihat (*Closed HH*) is displayed with x-shaped note heads, between the two top lines, since the Score Style setting for Drum Group *HH w. Sticks* is -0,5. The Crash Cymbal (*CRASH 1*, note head "x" in a circle) according to its Drum Map Editor setting would be displayed on the top line, but because the Score Style setting for Drum Group *Cymbals* is +0,5, the entire group is moved further up. The Drum Group *Tom* in the Score Style is at position -2 (= middle line), however *Mid TOM 2* in the Drum Map Editor has the relative position -0,5, which places it lower than *Mid TOM 1*.

The Bass Drum (*KICK1*) would be displayed at the top line if it only referenced the Drum Map Editor, but the Score Style position parameter for Drum Group *Kick* is -4,5, which results in a bass drum display below the bottom line. The pedal hihat in this example would be displayed at the same position as the bass drum (with a different note head), but it is not in use in this particular score example.

Note: the accent above the crash cymbal and the symbols for open/closed hihat were inserted from the Partbox

Creating and Editing Your Own Mapped Score Styles

This is very similar to creating non-mapped polyphonic Score Styles (see preceding sections). Select **New > Mapped Score Style** in the Score Style window, and insert all desired staves, Voices and Drum Groups using the menu functions (New) or the Pencil tool.

The copy, insert and paste functions for single staves, Voices and Drum Groups also work as described in the section *Copying Voices or Staves into other Score Styles* on page 13-98.

Before defining the details of a Mapped Score Style, you should define all *Drum Groups*, note head shapes, and relative note position parameters in the Drum Map Editor.

The best method for this is to use an already recorded drum sequence, and open the Score window displaying that sequence, the Drum Map Editor, and the Score Style window. This way, you can directly see how parameter changes affect the score display.

Using Mapped Score Styles for Non-Mapped Instruments

If you use a Mapped Score Style for the display of a regular (non-mapped) instrument, Logic will simply use the default settings for Mapped Instruments (which correspond to the General MIDI drum sound assignments), to determine the mapping of notes to Drum Groups. However, this is not recommended, since this method is less flexible (for example there will be only the default Drum Groups available).

13.17 Note Attributes

The term Note Attributes is used for all note display parameters that can be set individually per note, regardless of the settings in other windows and boxes.



The Note Attribute window opens when you double-click on a note's head (presuming that in **Options > Settings > Score Preferences, Double Click To Open: Note Attributes** is selected).

The settings shown in this window always correspond to the note whose head has been

double-clicked.

Using the **Attributes** menu, the different settings are assigned to a note by selecting the corresponding submenu, with the note selected. This attribute assignment can also be made for a group of selected notes at once. The same applies to the assignment of Note Attributes using *key commands*.



Reset Note Attributes: with this command (**Attributes** menu or key command) you can reset all Note Attributes to their default settings. Note that this also affects symbols directly attached to notes (accents, fermatas, Jazz symbols etc.), which will be erased by a reset of Note Attributes.

Below is a complete list of all available Note Attributes, and all possible ways to assign them to notes:

Special Note Heads

(Default setting "normal" = round and black):

Insertion from the Partbox with the mouse (see page 13-43)

In the Note Attributes window (*Head*)



Altering Note Size

(Default: Size in the Score Style window)

With the Sizing tool (see page 13-64)

In the Note Attributes window (*Size*)



Horizontal Position Change

(Default: as defined by the note's bar position)

With the Layout tool (see page 13-64)

In the Note Attributes window (*Horizontal Position*)



Accidental Distance from the Note

(Deviation from the default setting in the Extended Layout Parameters window)

In the Note Attributes window (*Accidental Position*)



Display of Accidentals

(Default: display according to key signature)

In the Note Attributes window (*Enharmonic Shift / Accidental Type*).



Menu **Attributes > Accidentals**

With Key Commands:

Default Accidentals: according to the default setting

Enharmonic Shift: #: converts for example G flat to F sharp,
also F to E sharp

Enharmonic Shift: b: converts for example D sharp to E flat,
also B to C flat



Flats To Sharps: similar to *Enharmonic Shift #*, but leaves notes without flats unaltered (useful for multiple selection)

Sharps To Flats: similar to *Enharmonic Shift b*, but leaves notes without sharps unaltered (useful for multiple selection)

Force Accidental: forces the display of accidentals (and naturals)

Hide Accidental: displays the note without accidentals (MIDI playback is not affected)

Guide Accidental: forces the display of the accidental (also naturals) in parenthesis



The display caused by Force and Guide can still be affected with the Enharmonic Shift functions.



An Example: it is very easy to quickly convert all D sharps to E flats on the current display level. Select a D sharp, use the key command *Select Similar*, then apply the command *Enharmonic Shift: #*.

Stem Direction

(Default setting in the Score Style window)



In the Note Attributes window (*Stem Direction*)

Menu **Attributes > Stems**

With Key Commands:

Stems: default: according to the default setting

Stems: up

Stems: down

Stems: hide: hides a note's stem, and the corresponding beam or flag



Beaming of Notes

(Default depends on the chosen time signature, the "Beat Grouping" setting in the time signature dialog box, and on the Score Style Parameter "Beam")



Menu **Attributes > Beaming**

With Key Commands:

Beam Selected Notes: forces a beam to the following note
Unbeam Selected Notes: interrupts the beam to the following note
Default Beams: according to the default setting



Tie Direction

(Default setting in the Score Style window)



In the Note Attributes window (*Tie Direction*)

Menu **Attributes > Ties**

With Key Commands:

Ties: defaults: According to the default setting
Ties: up
Ties: down



Syncopation

(Default setting in the Display Parameter Box)



In the Note Attributes window (*Syncopation*)

Menu **Attributes > Syncopation**

With Key Commands:

Default Syncopation: according to the default setting
Force Syncopation: forces syncopated display
Defeat Syncopation: defeats syncopated display



Interpretation

(Default setting in the Display Parameter Box)

In the Note Attributes window (*Interpretation*)

Menu **Attributes > Interpretation**

With Key Commands:

Default Interpretation: according to the default setting



Force Interpretation: forces Interpretation mode

Defeat Interpretation: defeats Interpretation mode

Independent: Display as Independent Note or Single Grace Note

(Default setting "Not Independent", details on Independent Notes see page 13-41).



Menu **Attributes > Independent**

With Key Commands:

Not Independent the note is displayed in the regular rhythmic context

Independent the note is displayed independent of the rhythmic context

Independent Grace: the note is displayed as a single (independent) grace note



13.18 Instrument Sets and Score Display Levels

In Logic's Score Edit window, there are basically two parallel systems determining which sequences are displayed, and which aren't (linear view versus page edit view does not play a role here). The hierarchical system which controls the *Display Levels* is basically the same in all Logic windows, but has a few special aspects in the Score Edit window:

Display Levels

In the Score Edit window you can work with the following Display Level settings:

Content Linked activated by a double click on the *Link symbol*.

In this mode only the *object currently selected* in another window (which most of the time is a single sequence, but can also be a folder) is displayed in the Score window.



Content Catch: similar to *Content Linked*, but *Catch* is also activated. Only one sequence is displayed at a time, but during playback the display changes to the next sequence on the same track, according to the current bar position.



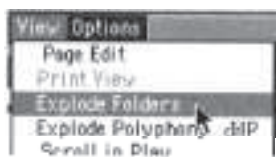
Linked (single click on the Link-symbol) The Score window displays the same display (folder) level as the Arrange window. This is most relevant if you work with Folders.



Not Linked (Link button deactivated): in this mode, Logic simply stays at the currently visible Display Level and doesn't react to sequence selection in other windows. For example, if you always want to see the full score in a particular Score window, choose this mode once the full score is displayed in that window (see below).



Explode Folders



If the current display level allows the display of more than one sequence at a time (*Linked* or *Not Linked*), the option *Explode Folders* in the **View** menu will affect how things are displayed: unlike the other editor windows, the Score window can display (or even print) the contents of different display (folder) levels at once. If the current Display Level contains folders, their contents will also be displayed in the score. (If this option is not activated, Folders will appear as grey beams in linear view. In page edit view, they won't be visible at all.)

Changing between Single Sequence and Full Score Display

At all Display Levels, a doubleclick on a sequence or folder will take you to the next lower level (*into* the Folder or to the display of only that particular sequence). A double click in the empty space below or above the staves will bring you up to the next higher level (the next higher Folder, or in case of the highest level to the full score display of all sequences of that song). Basically, the same principle applies even if you don't use Fold-

ers at all: the highest of all levels (the complete song) could also be regarded as a folder, which contains all the sequences.

Special Cases Concerning Score Display

In some cases, certain sequences will not be displayed in the Score window, no matter what the *Display Level* or *Instrument Set-settings* are. These cases include:

Muted Sequences, if *Hide Muted Sequences* is activated in **layout > Global Format**.



Sequences on muted tracks, if *Hide Muted Tracks* is activated in **layout > Global Format**.



Sequences with Score set to "off" in their *Extended Sequence Parameters* (global menu **Options > Extended Sequence Parameters** or key command).



Instrument Sets

Apart from the system of display and Folder levels, there is also the system of *Instrument Sets* in the Score window. This allows you to determine which (track) instruments should be included in the score display of a particular Instrument Set. Each song can contain as many Instrument Sets as you like. Instrument Sets allow you to:

arrange the (vertical) order of instruments in the score, independently from the order of tracks in the Arrange window.

assign a name and an abbreviated name to each track instrument, which will then be displayed and printed as instrument names in the score.

determine which of the staves will be connected by brackets and/or bar lines.

display or print only a certain part of the score, such as a particular group of instruments. This also allows faster editing in very large files, since screen redraws are much faster when less instruments are displayed.

exclude tracks from display that are only relevant for playback, but not for the score (i.e. tracks containing only MIDI controller data).

Reduce or enlarge all staves which are part of that Instrument Set at once, using the *Scale* parameter. (This allows you to use the same Score Styles for printout of full scores and parts).

Select from two different parameter sets (*Score* or *Part*) for page margins, header space height, vertical distance between stave systems, and the maximum number of bars per line.

An Instrument Set is selected in the *Instrument Set Box*, below the Catch and Link buttons. If Instrument Sets already exist in the current song, a flip menu with all available Instrument Sets will appear when you click-hold on the box. *New* songs will only show the entry *ALL INSTS.* which-depending on the display level-will display all instruments. Double-click this box to open the Instrument Set window.



The Instrument Set Window



The Instrument Set window can be opened with:

Double click on the Instrument Set Box



Menu **Layout > Instrument Sets...**

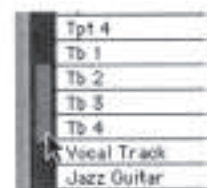
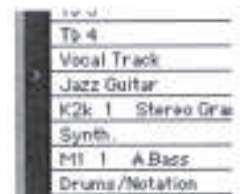
Key command *Open Instrument Set Window*



If you have opened the Instrument Set window while the default set *ALL INSTS.* was displayed, you will see a note saying that the *ALL INSTS.* Instrument Set cannot be edited. In the Instrument Set window, select **New > New Complete Set.** This will create an Instrument Set which contains all track instruments currently used in the Arrange window. Although for the moment this results in the same display as *ALL INSTS.*, you can now edit this Instrument Set. (In the *ALL INSTS.* Set you cannot define instrument names or change the order of instruments; and the bar lines are always connected throughout the whole score). On the yellow background you will see columns with the following parameters (from left to right):

Instrument

In this column, you determine which (track) instruments are included in the Instrument Set, and in which vertical order they will appear in the score. As in the Score Style window, there is a narrow column at the left margin, where you can set an insert mark (>) with a mouse click, or selection marks by dragging the mouse vertically. With these selection marks, instrument entries are selected. With the usual cut/copy/paste commands (**Edit** menu or key commands), you can move entries to change their order, delete them, or copy them into other Instrument Sets. If you *Paste* them, they will be pasted at the position of the insert mark, which you have to set to the desired position beforehand.



If you double-click below the instrument list, a new instrument entry is added at the bottom of the list. If you want to add an instrument somewhere in the middle of the list, set the insert mark accordingly, and select **New > Add Instrument Entry.**

Click-hold on the name of this new entry (*Instrument* column). A flip menu with all available track instruments will appear, that you can select from.

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If you **keep the -key pressed** while you select a track instrument with the flip menu in the Instrument Set window, it has the same effect as in the Arrange window: the former track instrument is replaced by the new one throughout the song, wherever it has been used,



In most cases, it is convenient to create a new Instrument Set with **New > New Complete Set**, since all instruments currently used in the Arrange window will be included automatically.

You can, however, also select **New > New Empty Set**, into which you then can insert the desired instruments one by one. Other options in this menu are **Duplicate Set**; which creates a copy of the currently visible Instrument Set, that you can use as a basis for another set, and **New Set of Selected Insts.**, which creates a new Instrument Set containing all currently selected instruments.

At the highest zoom level, the Track Instrument icons are also displayed in this column (as in the illustration above).

Please note: you will not see a new Instrument Set in the Score window until it has been chosen from the flip menu in the Instrument Set Box.

FullName

If you click in this column on any line, an entry field opens with the default entry **@(reference)**. If you don't change this, the Track Instrument name will be used as the name for that instrument in the score. However, since the track instrument name often is identical with the corresponding synthesizer patch (i.e. Solos-trings *high*), you will probably wish to assign another name here. This name will automatically be displayed in the score as the **full name**, for example "Violin 1". The display parameters for the instrument names are set in **Layout > Numbers & Names** (see page 13-128).

Short Name

Here you can define an abbreviated name for each instrument, which is used if in the **Layout > Numbers & Names** dialog window `shortnames` is selected for the instrument name display.

Brackets and Bar Lines

You can define which staves in the Instrument Set will be connected by brackets (two bracket types available), and/or connected by bar lines (either at the beginning of each staff, or throughout) using the last four columns. For any of these connections, just drag the mouse vertically in the column until you see the appropriate display.



These brackets and bar lines can also be interrupted between staves, which allows you to form groups of connected instruments in a score. If you want to delete one of these lines or brackets, grab the corresponding symbol at its end point, and drag it up until it disappears. You can use the same method to shorten the lines.

The bar line connections can also be edited directly in the score. Just click on the upper end of a bar line to connect it to the next higher staff. The same procedure applied once more disconnects the bar lines again.



Name of Instrument Set

The Name of an Instrument Set can be edited by double clicking on the name in the left column of the Instrument Set window (see opposite illustration), which opens a text entry field. If you click-hold on the name box, you can switch to another Instrument Set by selecting it from the resulting flip menu.



Scaling, Format

These two parameters below the Instrument Set's name make it possible to create both the full score and the parts for all instruments from within the same song file, without having to

change any settings. For details please read the section *Full Score and Part Layout* on page 13 - 117.

Importing Instrument Sets from Other Songs



Select **Options > Import Settings**. In the resulting dialog box, activate *Instrument Sets* only, and click on *Import*. This imports all Instrument Sets from another song into the current song.

If another song file is already open during this procedure, that song's Instrument Sets will be imported. If no other song is open, a file selector box will appear, where you can select the song whose Instrument Sets you want to import into your current song.

Additional Information on Instrument Sets

Relation to Display Levels

An Instrument Set will only be displayed as defined, if an appropriate *Display Level* is chosen (see page 13-109). *Content Linked* and *Content Catch* only allow the display of one sequence at a time, so they are not suitable for working with Instrument Sets. It's usually best to deactivate *Link* once the desired Instrument Set is fully displayed. If only one sequence is displayed, double-click below the staff, which will bring up the full score. After that, deactivate *Link*.



Please note: it is possible to work with several open Score windows showing different Display Levels and/or different Instrument Sets.

New Instrument Sets for Selected Instruments

In the Score window, **layout > Create Instrument Set from Selection** (also key command) will immediately create and display a new Instrument Set, which consists of all the instru-



ments which are used by the currently selected sequences. For this function, it does not matter if the sequences are selected in the Score or in the Arrange window. Since screen redraws are much faster if less instruments are displayed, this is a good way to speed up editing. Instrument Sets created this way are automatically named according to the instruments they contain. If you want to delete the set again after editing, select **New > Delete Set** in the Instrument Set window while the particular Instrument Set is shown there.

Automatic Display of Selected Instruments when Opening the Score Window

When you open a new Score window while several sequences are selected in the Arrange window, an Instrument Set will automatically be created and displayed in that Score window which only contains the instruments belonging to the selected sequences. In other words, the function *Create Instrument Set From Selection* is automatically performed whenever a Score window is opened (using   or global menu **Windows > Open Score**) and sequences on more than one track are selected in the Arrange window.

Instrument Sets created this way are saved with the song file, and are also available for later use.

Different Instruments Using the Same Sound

If several instruments or staves use the same MIDI sound for playback (with the same MIDI channel on the same MIDI instrument), and if you wish to be able to display those staves with different instrument names in the score, you have to create a separate track instrument for each staff in the Arrange or Environment window.

Full Score and Part Layout

Some functions in the Instrument Set window and some settings in the *Global Format* window (**Layout** menu) make it possible to produce both the full score and the individual parts

for all instruments from within the same Logic song file, without having to adjust any settings.

Part Extraction (“Instrument Filter”)

Hold the -key as you click and hold on the Instrument Set Box in the Score window. A flip menu will appear, which contains all Track Instruments currently used in the Arrange window. If you choose one of them, only this particular instrument will be displayed. This method allows you to quickly display (and print) instrument parts. However, in this special case, only the track instrument names will be used as instrument names in the score (In the example above, the drums would be *Drums/Notation*, instead of just *Drums*). So, if you use this method for printout of your parts, you have to be careful to use the “real” instrument names as names for the Track Instruments in the Arrange window.



Scaling Parameter for Instrument Sets

In the Instrument Set window, below the name of the Instrument Set, there is a *Scaling Parameter*, which can be set to any desired size between 50 and 200 percent of the original size. This scaling will affect all staves, the distance between staves, all symbols which are part of the staves (i.e. all notes, rests and other symbols), and all local text including lyrics. All these elements will be changed in size according to the scaling parameter of the currently displayed Instrument Set.

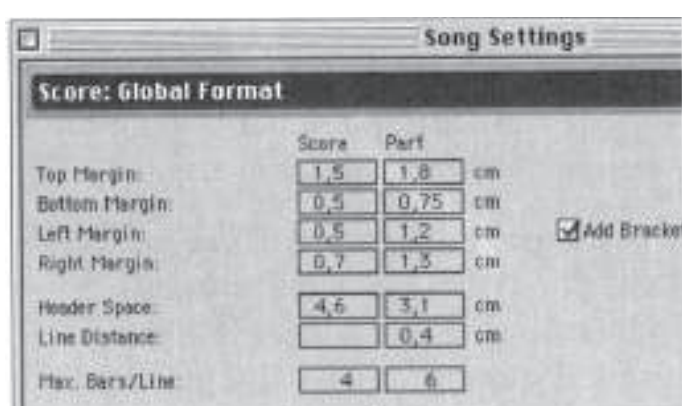


The scaling parameter does not affect Global Text objects (i.e. header text, text which has been inserted outside the margins) and text like page numbers, etc.

The scaling is *not* applied to parts which are extracted using the Instrument Filter function described above. Therefore, you can use the same Score Styles for full scores and parts: in extracted parts, the staves will appear with their original size, in the full score their size will be diminished according to the scaling parameter.

Separate Layout Options for Parts and Score

In the *Global Format* dialog window (**Layout** menu) there are separate page layout parameters for score and parts. Apart from left, right, top *and bottom margins*, the (vertical) size of the *Header* area the *Line Distance* (additional vertical distance between stave systems), and the maximum number of bars per system *Max. Bars/Line*) can also be set separately for *Score* and *Parts*.



Which of these parameters are used for the display and printout of a particular Instrument Set, depends on the setting of the *Format* parameter in the Instrument Set window (below the Scaling parameter, see opposite illustration). You can switch this setting by clicking on it.



When you use the *Instrument Filter* method (see page 13-118) to extract and print the parts for single instruments, the *Part* settings are applied automatically.



For the default *ALL INSTS* set, the *Score* settings will always be used, unless only a single sequence is displayed. In this case, the *Part* settings will be applied. Please note: the *ALL INSTS* set cannot be edited (and therefore not be scaled), so it will be necessary in most cases to create a dedicated Instrument Set for the full score.

The margin and header values can also be changed directly with the mouse, by dragging the margin (green) and header lines (blue) in page view. This will only affect the *Format* setting which is currently in use for display (*Score* or *Part*).

How to Use these Options

The best way to make use of this feature is probably to work with Score Styles which are basically fit for printing parts (i.e. size 7 or 8, as in the default Score Styles), and create an Instrument Set for the complete score which is scaled down enough to fit the (vertical) length of the page you are printing on. For example, 65% might work, but the actual scale percentage would depend on the size of the ensemble, and whether you want to fit several stave systems on each page. Adjust the page margins and the header directly in Page view, by dragging the lines with the mouse, and use this Instrument Set to print the full score (The *Format* parameter should be set to Score).

To create the individual parts, use the Instrument Filter function described on page 13-118. Since parts extracted with this method are treated as individual (but non-scaleable) Instrument Sets, they will be printed with the original size of the used Score and Text Styles (“100%“) and automatically use the *Part* parameters for margins etc.. When you work on the parts, you might want to adjust the page and header margins (this will be reflected in the *Part* settings).

You gain even more flexibility if you create instrument Sets which only contain one instrument each: in this case a different scaling factor can be chosen for each instrument, and the printed instrument name does not depend on the track instrument name.

To create additional vertical distance between the staves, open the *Global Format* window, and adjust the *Line Distance* setting in the *Part* column there. The same goes for distance between stave systems in the full score, with the parameter in the Score column. The maximum number of bars per line or system can also be set separately for score and parts in this window.

Line breaks and individual stave margins can be determined individually for each Instrument Set, and also for each of the extracted parts, as described in the following section.

Line Breaks

In Page Edit view, you can use the Layout tool to edit line breaks. The automatic line breaks are calculated based on the settings in **Layout > Global Format** (Spacing and *Max.bars/line*), but can be changed with the following procedure. To move the last bar of a stave system down into the next staff, grab that bar with the Layout tool, drag it down until the resulting hand symbol points down with the thumb, then release the mouse button. The bar is moved to the next stave system, and the remaining bars in the line are distributed evenly over the entire width of the page. The same method works for several bars at once. If you grab the third to last bar of a staff, and drag it down, three bars will be moved to the next staff. You also can move the first bar(s) of a staff upwards into the preceding staff, by dragging them upwards (symbol: hand pointing up).



Logic stores the line breaks for each Instrument Set individually. This also applies to all parts created with the above described Instrument Filter function. These are all stored together with the song file.

Restriction to Local Movement

When you move a bar with the Layout tool as described, all manually edited line breaks in subsequent lines are deleted, and the (automatic) line breaks are calculated again automatically from that line downwards. If you want to prevent this, perhaps because you only want to change one detail and want to keep the other staves as they were, hold the **ctrl**-key while you move the bars with the Layout tool. Line breaks edited this way will only affect the two directly concerned stave systems. All other stave systems will remain unchanged.

Combining *Max.Bars/Line* and Line Breaks

In the **Layout > Global Format** dialog window, there is a parameter called *Max.Bars/Line*. It determines the maximum number of bars Logic will allow in one staff or stave system. With the Layout tool, you can override this setting. For exam-

ple, you can set *Max. Bars* to 6, and still move additional bars into a line with the **Layout Tool**. The subsequent lines will again contain no more than six bars. The full number of bars set here will only be displayed, if the spacing settings are low enough to allow the proper display of that many bars.)

Please note: if you drag more bars into a line than would be displayed when employing the automatic line break function, this can result in overlapping notes and symbols!

Local Margins

(Indent for Individual Stave Systems)

The left and right margin of each individual stave system can be moved with the Layout tool, so that they do not align horizontally with the page margins anymore: just click and hold with the Layout tool a little bit inside the beginning or end of the staff, and pull the mouse to the left or right side (see illustration). During this process, the indication "*Left/Right Stave Margin*" will appear in the *Info Line*.



In this case Logic also adapts the number of bars per line automatically (unless the line breaks have been adjusted manually beforehand): if the line is shortened, bars might be moved down to the next line, depending on the *Spacing* parameters in the Global Format window.

To reset such a changed staff margin to line up with the other staves again, doubleclick at its end (or beginning). A dialog will appear, asking *Reset Local Margin?* Confirming this dialog will reset the margin.

Hint: like the manual line breaks, these local margins are also saved as part of the current Instrument Set, so they can be set differently in every Instrument Set.

Deleting all Line Breaks and Local Margins

To delete all line breaks and all locally changed margins in the current Instrument Set, select **Layout > Reset Line Layout**.

13.19 Song Settings for Score Display

In the Song Settings dialog box, there are six pages with score settings. You can change between these and the other pages of the Song Settings dialog box directly using the pop-up menu on that page. In addition to the methods mentioned below for opening these pages, you can select these pages in the global menu **Options > Settings >...**

The settings on these pages affect the whole song (in all Instrument Sets). They are saved with the song file, so they can be different in each song.

Please note: in most cases, values of “zero” are not displayed in these settings, the parameter box is empty instead.

Importing All Score Settings



Select **Options > Import Settings** in the Score window. In the resulting dialog box, activate Score Settings only, and click on *Import*. This imports all Score Settings relevant to the score display (i.e. the settings on these six pages) of another song, into the current song.

If another song file is already open during this procedure, that song's Song Settings (score only) will be imported. If no other song is open, a file selector box will appear, where you can select the song whose Score Settings you want to import into your current song.

Global Format

Here you can find the settings for page margins, spacing between notes, slanting of beams and format settings.

This window can be opened by:

Layout > Global Format...

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The Score Edit Window

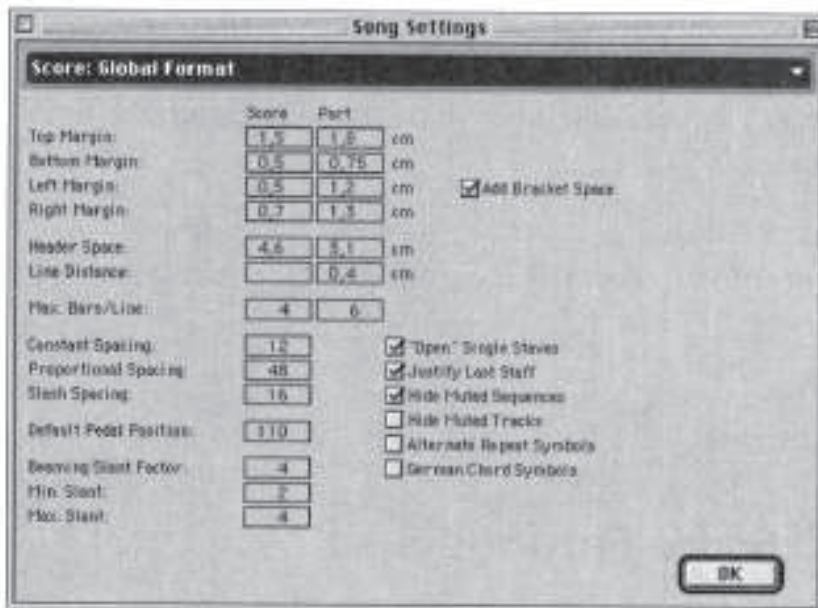
Key command *Settings: Global Format*



Double click on the page margins in Page Edit view



Note: distance values can be displayed in inches or centimeters. To change between these, just click on one of the depictions of the units being used.



The settings in detail:

Top/Bottom/Left/Right Margin

The page margins

These values show the margin distances to the outer border of *the printable area* on the page. So "*Top Margin 0,0 inches*" will mean that printing will start as close to the top of the paper as the selected printer driver will allow. This means that, with different printers, the size of the printable area can be different, although the difference should be minimal.

The margins are visible only in Page Edit view, where they are displayed as green lines on the screen. These lines are not printed. It is also possible to change them directly, by dragging them with the mouse in the score.



Unlike the global page margins, the left and right borders of stave systems can also be edited individually. To change the left/right border of a staff system, simply use the Layout tool to



click and drag the border to the desired position. The local margin can be reset by double-clicking the same position with the Layout tool. (see the section *Local Margins* on page 13-122)

Add Bracket Space

Creates additional space between the left margin line, and the beginning of the staves for braces and brackets. If this option is deactivated, staves will be aligned directly along the left margin line.

HeaderSpace

Defines the height reserved for headers, between the first page's top margin, and the first staff's top margin (as defined in its assigned Score Style).

Note: text elements inserted directly into this area automatically become Global Text, and will be displayed as headers in all Instrument Sets (scores and parts).

This value can also be changed directly in the score, by dragging the dotted line above the first staff with the mouse.

Line Distance

Defines additional vertical distance between stave systems on the same page. It applies to full scores, as well as single staff parts.

Max. Bars/Line

This parameter can be useful, when using small spacing values (see below) to prevent too many bars from being displayed in one line. The number set here limits the number of bars that can be displayed per line.

Note: this setting can be overridden when using the Layout tool for changing the line breaks. (see page 13-121).

Constant/Proportional Spacing: Note distance

These parameters determine how much horizontal distance Logic leaves between notes. *Constant Spacing* affects the

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distance from note to note, regardless of rhythmic value. *Proportional Spacing* also takes the note durations into consideration. If you only use *Proportional Spacing* (and set *Constant* to 0), every bar more or less gets the same amount of (horizontal) space. A whole note uses as much space as four quarter notes. In the opposite situation (high constant value, proportional value = 0), the distance from one note to the next is always the same regardless of note duration. A half note takes the same amount of space as an eighth note. Other factors, like accidentals, ties, etc. are also considered for calculating the note distances.

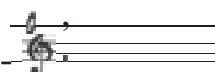
Here is an example of two display versions of the same measure, using different spacing parameters. You can see that in the first version, the distance between the different notes is almost the same throughout, whereas in the second version the individual note lengths play an essential role in the graphic distribution of the notes:



Which settings you use, depends both on personal preference, and the style of the piece. Certainly, one should aim for a good balance between these two parameters. For best results, try different combinations when you work on the final layout of your score.

Slash Spacing: Spacing of Beat Slashes

This is the distance parameter for slashes used in Score Styles that display *Beat Slashes* instead of automatic rests. This is especially important if *notes* are inserted into staves displayed with slashes. For example, for musical styles such as funk, which use a lot of sixteenth notes, you will likely choose a higher slash distance than for music that doesn't go beyond eighth notes.



This way, the proper relation between notes and passages with only slashes and chord symbols can be kept.

Default Pedal Position

This determines the vertical position of pedal symbols which are created and displayed automatically when you use the MIDI sustain pedal during a real time recording. If this parameter is set to zero, the recorded MIDI sustain pedal controller events will be hidden in the score. Pedal marks inserted from the partbox are not affected by this parameter.



Beaming Slant Factor | Min. Slant | Max. Slant

These three parameters affect the slant angle of beams. As with the *Spacing* parameters, these parameters work together. Appropriate settings have to be found by trying different combinations. The final settings will vary, depending on the style of the music, and on personal preference.



Beaming Slant Factor determines the general amount of beam slanting in relation to the intervals of the notes connected by beams.

Min. Slant determines the minimum interval of notes that causes beams to be slanted.

Max. Slant determines the maximum beaming slant angle.

These parameters always work relative to a particular score situation, which is why no absolute instructions on how to set them can be given. Again, it is essential to try different combinations when working on the final layout of a piece.

“Open” Single Staves

Displays single staves without vertical lines at their start point (left side of the clef).



Justify Last Staff

lengthens the last line (in full scores the last staff system) to the right page margin.

Hide Muted Sequences I Tracks

Excludes muted sequences, or sequences on muted tracks, from the score display. If these options are not activated, muted tracks and sequences will be displayed in the score even though they won't be heard during MIDI playback.



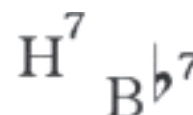
Alternate Repeat Symbols

If this option is activated, all repeat signs in the song will be displayed with brackets as shown in the opposite illustration ("Real Book Style").



German Chord Symbols

If this option is activated, the note *B* in chord symbols is generally displayed with its German name *H*. *Bb* will then be written as "B", according to the German note name. This option is also activated automatically, as soon as the letter *H* is used during chord symbol input.



Numbers & Names

These settings affect the automatic display of page numbers, bar numbers and instrument names in the score.

To open this window, select:

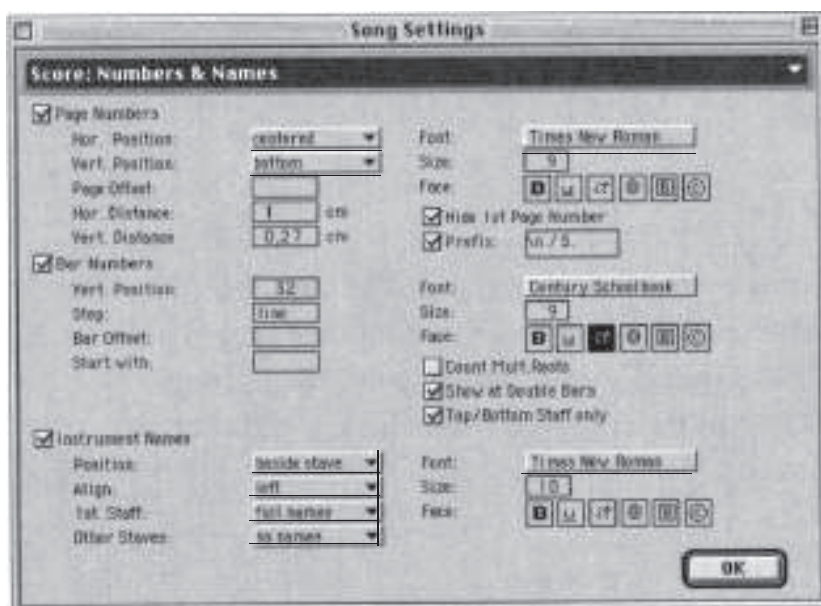
Layout > Numbers & Names...

Key command Settings: *Numbers & Names*

Double-click on any bar or page number.



Song Settings for Score Display



Automatic page and bar number display, and automatic display of instrument names can be switched on and off for the whole song, by activating/deactivating the corresponding check boxes in this window.

For each of these three you can set *Font*, *Size* and *Face*, (bold, underlined, *italic*, *outlined*, **boxed** and **circled**. Each one is activated by clicking on the appropriate symbol. The “circled” attribute is similar to “boxed”, but a circle is used instead of a rectangle. For wide text, an ellipse is used.

If you edit these settings, the corresponding Text Styles in the Text Style window will automatically be updated.

Page Numbers

Hor. Position

The horizontal alignment on the page

alternating: alternating, beginning on the right side

left / right / centered

rev. alternating alternating, beginning on the left side

Vert.Position

The vertical alignment on the page

top/bottom

Page Offset

This value is added to each actual page number for display purposes. This can be useful when you write a piece consisting of several parts, which you save as separate song files. To still have continuous page numbers throughout the score, you can set this parameter to the number of pages contained in all preceding parts.

Hor.Distance

The horizontal distance from the outmost possible printing position on the page. This is only relevant for page numbers with the *Hor. Position* parameter *alternating* or *rev. alternating*.

Vert Distance

The vertical distance to the highest or lowest possible printing position on the page (depends whether *Vert. Position* is set to *top* or *bottom*).

Hide 1st Page Number

Prevents display and printout of the page number on the first page (all others will be displayed and printed).

Prefix

Here you can enter text which will be displayed with every page number.

Example: if you use the prefix "*Page*", the word "Page", then a space and the actual page number will be displayed on each page. This means that page three will display "*Page 3*". If you want the page number in the middle of the prefix, you have to use the "#" symbol as a variable for the page number. An example: "*pg.#-*" on the second page prints as "*-pg.2-*".

You can also use the following text symbols to include automatically generated text in the prefix:

`\i` for the Instrument Set name

`\n` for the song (file) name

`\s` for the name of the currently shown Display Level, which can be the sequence or folder name (on the highest level even the song name)

`\d` for the current date (at the time of the printout)

Additional information on *Automatic Text Objects* see page 13-135.

Another example: in a song file named “Blue Shark” the prefix text `"- \n \i, Pg# -"` might result in the following display on the second trumpet part: *"- AD's Cool! Tp2, Pg.1 -"*. (If *Hide 1st Page Number* is activated, this will only be displayed from the second page onwards).

Bar Numbers

Bar numbers are placed above or below the barlines, and at the beginning of the line, above or below the clef (they are automatically moved to a position where they won't overlap the clef).

Vert. Position

The height above the staff. If a *negative value* is set here, the bar numbers are placed below the *staff*.

Step

This value controls the interval at which bar numbers are repeated. Value 4, for example, results in bar numbers at the first, fifth, ninth bar, etc. If you want bar numbers to be displayed only at the beginning of each staff line, set this parameter to line (which appears instead of zero).

Bar Offset

This value is added to all actual bar numbers for display.

Start with

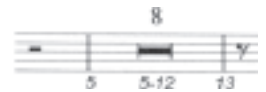
The automatic numbering will only be displayed starting at the bar indicated here.

Top/Bottom Staff only

In full scores, the bar number is only displayed above the top staff if this option is activated (with a negative *Vert. Pos.* value the number will be below the bottom staff).

Count Mult. Rests

Displays the first and last bar number below multiple rests as shown in the opposite illustration. This makes sense if a negative *Vert. Position* value is chosen, causing the bar numbers to be shown below the staves.



Show at Double Bars

Displays bar numbers at every double bar line and repeat sign, independent from the chosen *Step* setting. For example, *Step = 1* and *Double Bars* activated will display bar numbers at the beginning of each staff line, and above or below all double bar lines and repeat signs.



Instrument Names

This determines the appearance of the instrument names that have been defined for the Track Instruments in the *Instrument Set window*, in the columns *full names* and *short name*. If nothing is defined here, or *if ALL INSTR.* is used as the current Instrument Set, the names of the Track Instruments (as displayed in the Arrange window) will be used in the score. (Short names do not exist in this case)

Position

above/beside staves

Align

Left at the left margin

right at the right margin (refers to the space in front of the staves)

1st Staff

no names / short names / full names

Other Staves

no names / short names / full names

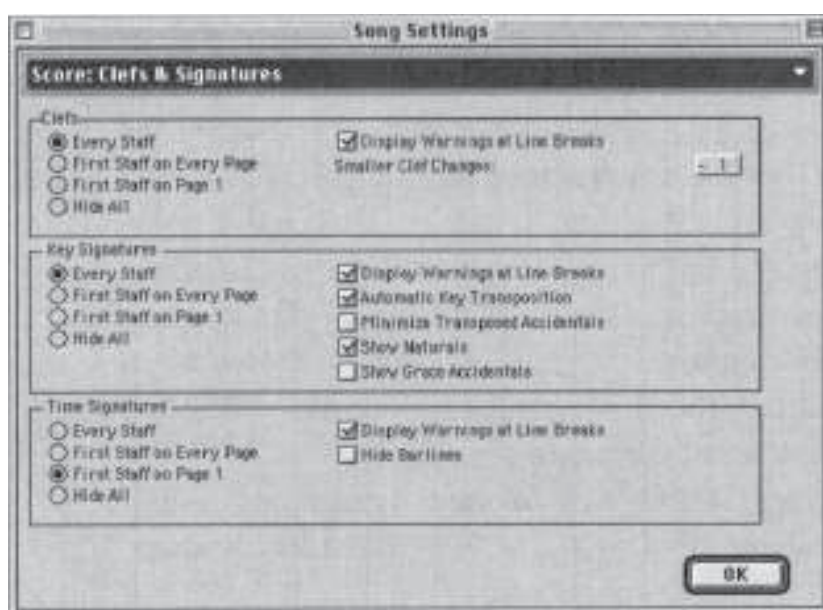
Clefs & Signatures

These are the options for the general display of clefs, key signatures and time signatures. (Concerning these topics please read the section *Clefs* on page 13-42, the section *Key Signatures and Key Signature Changes* on page 13-49 and the section *Time Signatures and Time Signature Changes* on page 13-51.

How to open this window:

Layout > Clefs & Signatures...

Alt-double-click on any clef or time signature in the score.



The following display options are available for Clefs, Key Signatures, and Time Signatures as well:

Every Staff

First Staff on Every Page

First Staff on Page 1

Hide All

Display Warnings at Line Breaks: this causes “warning” clefs, time signatures, or key signature changes to be displayed at the end of a staff or stave system, if the actual change is at the beginning of the subsequent staff or stave system.

Apart from these, there are the following options:

For Clefs:

Smaller Clef Changes (flip menu): clef changes (whether caused by a change of Score Style, or by a new clef inserted from the Partbox) can be displayed smaller than the clefs at the beginning of the staves, depending on this setting.

For Key Signatures:

Automatic Key Transposition: this enables the automatic transposition of key signatures in transposing Score Styles.

Usually, this will be activated. An exception would be for the notation of atonal music. If it is deactivated, all sharps and flats are displayed directly with the notes. Note that there is a similar option in the Score Style window (parameter “Key”), which allows you to switch off the key signature for individual Score Styles, or even for individual staves in multi-stave Score Styles.

Minimize Transposed Accidentals: this enables the use of enharmonically changed key signatures, if this results in a signature with less accidentals. For example, if a piece is written in B major (5 sharps), the Bb-instruments are notated in Db major (5 flats) instead of C# major (7 sharps).

Please note that Logic does not display any key signatures with more than seven flats or sharps, whether this option is activated or not.

Show Naturals: displays naturals at key signature changes, where preceding accidentals are no longer valid, for example when changing from E major to G major, or to C minor).

Show Grace Accidentals: if this option is activated, courtesy accidentals will be displayed automatically. This means: wherever a diatonic note is altered by an accidental, another accidental (i.e. in many cases a natural symbol) will be displayed when that note appears again unaltered in a subsequent bar.

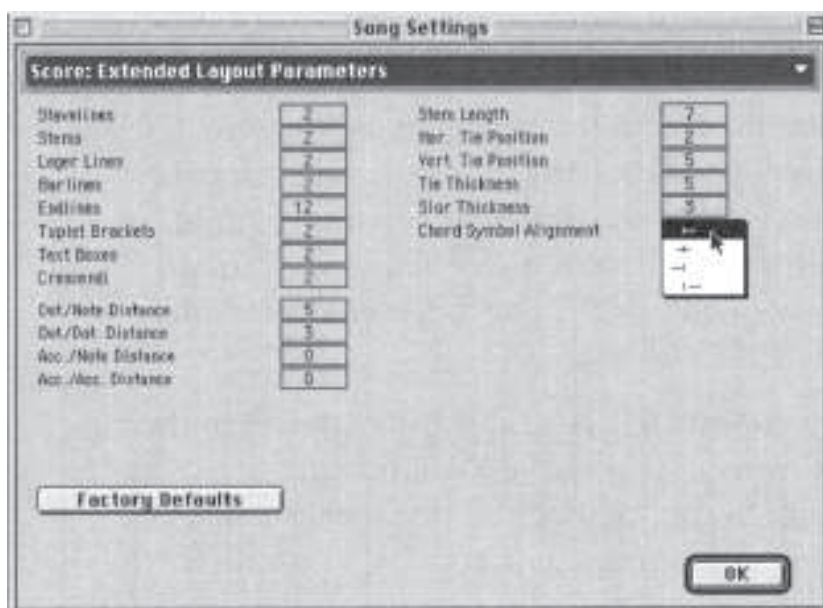
For Time Signatures / Barlines

Hide Barlines: this option allows you to hide *all* barlines, which can be useful for educational material, gregorian plain chant notation, and other special situations. Only automatically displayed barlines will be hidden, so it is still possible to insert barlines from the Partbox including the “regular“ barline, which will then be displayed and printed.

Extended Layout Parameters

This is where you define certain display settings for the whole song, like staff line thickness, stem length, distance between notes, and ties, or notes and dots etc.

To open this window, select **Layout > Extended layout Parameters**.



The first eight parameters determine the Line *thickness* of the following objects: stave lines, stems, leger lines, bar lines, repeat and end lines, tuplet brackets, text boxes (for boxed Text Styles) and crescendo/decrescendo (this last parameter also affects line objects and arrows).

Dot/Note Distance determines the distance between the note heads and dots for dotted notes. *Dot/Dot Distance* refers to double dotted notes.

Acc./Note Distance allows you to globally alter the horizontal distance of accidentals from the corresponding noteheads. The default setting should only be changed if e.g., very small values are used for the spacing parameters, which results in a very small distance between notes. This global setting can also be combined with the local *Accidental Distance* parameter of individual notes, in the *Note Attribute window*.

Acc./Acc. Distance allows you to globally alter the horizontal distance between several accidentals in chords.

Stem length is the default setting for stem length.

Hor. Tie Position and Vert. Tie Position determine the distances between note heads and related ties (which are displayed automatically). There is also a parameter for *Tie Thickness*.

Chord Symbol Alignment determines the general horizontal alignment of chord symbols whose *Align* parameter is set to def(default) in relation to their bar position. For details please read the section *Precise Graphical Position of Chord Symbols* on page 13-159.

Clicking on *Factory Defaults* resets all these parameters to their default settings. However, if you have a printer with high resolution, you should try to use smaller line thickness settings (2, or maybe even 1). Smaller staves, in particular, look much

better and more professional that way. The other parameters' settings are more a matter of personal preference.

On the screen these changes are only visible at the highest zoom levels. To really be able to judge the results it is necessary to try some printouts with different settings.

Guitar Tablature

Guitar tablature is an alternative method of notating music for fretted string instruments, especially for guitar and electric bass, but also for other fretted instruments with four to six strings or courses of strings.

In this system, the horizontal lines represent the strings of the instrument. Notes are always written on the line/string at which they are to be played. Instead of regular note heads, the numbers of the frets are shown.

Logic converts notes automatically into tablature, if a Score Style is used whose Clef parameter is set to one of these ***Tuning Sets***. The exact characteristics of these Tuning Sets are determined in the Guitar Tablature window.

How to open this window:

Layout > Guitar Tablature

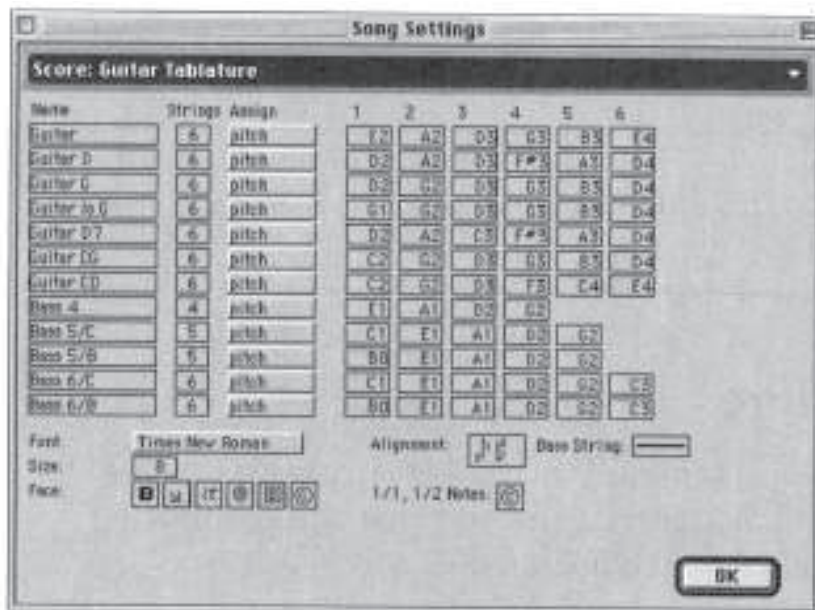
Key command ***Settings: Guitar Tablature***

Double-click on the TAB clef, at the beginning of any staff that already uses tablature in the score.



Chapter 13

The Score Edit Window



Here, twelve different *Tuning Sets* can be defined. Each of them corresponds to one line in this window. The regular guitar and bass tunings are already included as defaults (first line and last five lines), as are some of the more often used special tunings for guitar.

For every Tuning Set, there are the following parameters (from left to right):

Name: can be changed in the text entry field that opens with a double click on the name field.

Strings: number of strings (four, five or six).

Assign: the method Logic uses for automatically assigning notes to strings. (see below)

1 to 6: the pitches that the “open” strings are tuned to.

Other common parameters:

Font, *Size* and *Face* for the display of the fret numbers.

Alignment of the numbers to the stem: to the side, or centered (can be switched by clicking on this box).

Bass String: the display of the lowest string: may be the same as the other strings, or a bit thicker. (can be switched by clicking on this box)

1/1, 1/2 Notes: if this option is activated, half and whole notes will be displayed with a circle around the fret number. Since the note head is always a number, there would be no displayed difference between a half note and a quarter note otherwise.

Assign

Since most notes can be played at different positions (frets) on different strings, the *Assign* parameter plays an essential role in this system. In most cases the MIDI channels of the individual notes determine the string assignment. (Remember: the MIDI channel parameter of an individual note has no impact on MIDI playback. The playback channel is determined in the Instrument Parameter Box in the Arrange window.) There are the following options for string assignment:

Pitch: Logic assigns each note to the string on which it is playable, at the lowest possible position. However, to still be able to influence the string assignment, there is the rule (in *Pitch* mode only) that a note's fret position cannot be lower than its MIDI channel. This method is generally used when the notes are recorded from a MIDI keyboard:

An A3 with MIDI channel 1 is displayed at the second fret of the G string. If you want the note to be displayed at a higher position, to correctly display the proper fingering of a passage, you have to set its channel to 3 or higher. Now, the fret position cannot be lower than 3 any more, so the note will be displayed at the D string's seventh fret. If the channel is between 8 and 12, the A3 is assigned to the A string on the 12th fret, with channel 13 to 16, to the low E string (17th fret). If you want to force even higher positions, you'll have to use one of the other assignment modes.. .

Channel: in this mode, the number of the string (1 to 6, counted from the lowest string up) simply corresponds to the

MIDI channel. Channels 7 to 16 are also assigned to the highest string.

Inv. Chan: like *Channel*, but counted from the highest string (1) to the lowest (6 and above). This way of numbering the strings is generally used in traditional guitar literature. Since most Guitar-To-MIDI-Converters send notes on different MIDI channels, according to the strings they are played on, the last two modes are very suitable if you use such an instrument for recording MIDI sequences. In this case, the tablature notation displays the exact way the music was played during the recording.

Inv.Ch-1, Inv.Ch-2: these modes have been created for the purpose of recording and notating electric bass parts with Guitar-To-MIDI-Converters. "-2" is designed for four-string bass, "-1" for five-string bass (with an additional high B string). The principle is the same as *Inv. Chan*, but the number of the MIDI channel is reduced by 1 or 2, in order to convert the channels properly for bass string assignment.

The string assignment for a four-string bass will be 3 to 6, for a five-string bass 2 to 6 (counted from highest to lowest string in both cases), just as with guitar strings that sound one octave higher. Notes on channel 1 will be assigned to the highest string.

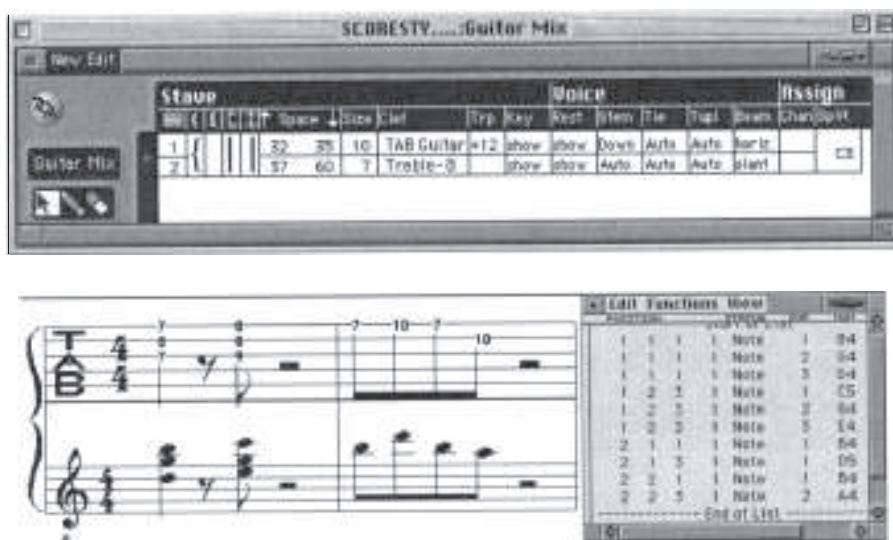
In all of these assign modes, notes below the lowest string will not be displayed at all. Also, notes generally can only be assigned to strings on which they are actually playable. (An F2 can only be played on the lowest string of a guitar, so the channel assignment is completely ignored in this case.)

You can create a two-stave Score Style that displays the sequence twice: once using regular music notation, and once as tablature. The indicated split point can be ignored in this case. There is a default style for this (named *Guitar Mix*), in the Logic default song.

Here is an example, where you can compare the two notation methods, along with the corresponding event list. The Assign mode used is *Inv. Chan.*, the Tuning Set is the first default set in

Song Settings for Score Display

the Tablature window, the regular *Guitar* tuning. If the *Pitch* mode were used here, almost all notes would be assigned to the highest string, which wouldn't make sense:



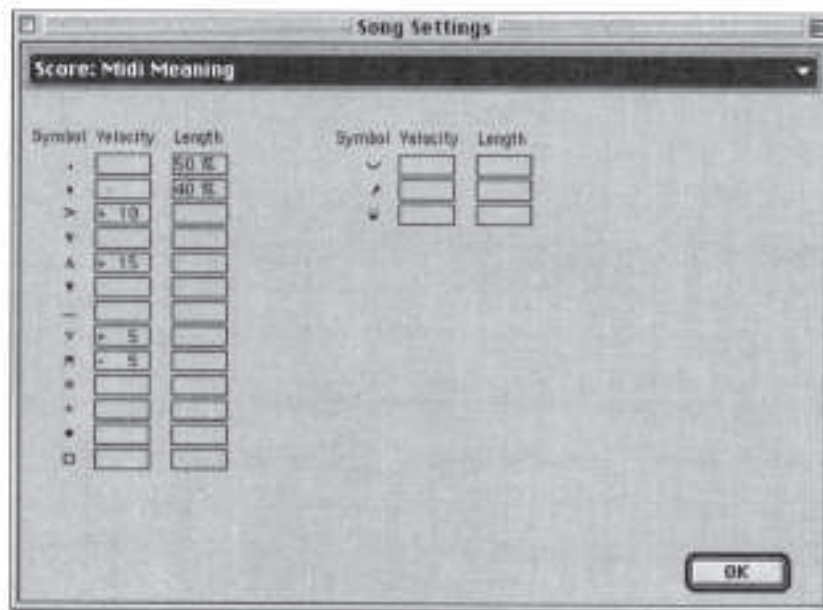
Some more information concerning tablature notation in Logic:

You cannot insert clef changes in staves using tablature. If you want to change between tablature and regular notation in the same staff, you have to cut the sequence at that particular point, and assign the desired Score Styles to those sequences.

Guitar tablature relates to regular treble clef. If you use the octave-transposing treble clef (*Treble-8*) for guitar notation (as it is done in most guitar music, and also in the above example) , you have to set the tablature's *Transpose* parameter in the Score Style window to +12.

MIDI Meaning

The settings in this window determine, if and to what extent, the insertion of the symbols listed here have an effect on the MIDI playback of the notes to which these symbols are attached. The window is opened from the menu **Layout > MIDI Meaning**.



For each symbol listed here, two parameters can be determined, which change the MIDI output of all notes to which these symbols are attached:

Velocity: This value is added to, or subtracted from the original velocity value of the corresponding note.

Length: The original note length is shortened according to the percentage chosen here. This only affects the playback of the note, not the score display. The note length change can for example be seen in the Event Parameter Box or in the Event Editor window.

How it Works

The MIDI Meaning functions basically make most sense for users who insert notes using the mouse. This makes it possible to work as if writing music on paper. First you write/insert the notes. They will all have the same velocity, and identical rhythmic note values will have exactly the same length. Accents and phrasing marks are then attached to some of the notes, which changes the velocity and (playback) length of those notes. This makes the playback sound much more realistic and “alive”.

The default settings in this window (i.e. in new songs) are 0 for velocity and 100% (no change) for length. If you don't change these, the symbols will remain purely graphic and will not affect MIDI playback. If you record your sequences in real-time it's better to leave these settings at the defaults, since the notes probably already sound the way you want them to.

Important: if you do use MIDI Meaning, you have to adjust the settings *before* you begin to insert accents etc., since the settings have no influence on already inserted accents and phrasing marks.

Once set, the velocity and length of notes will be changed as soon as you attach one of these symbols to a note. When you delete the symbol, note velocity and length are reset to their initial values.

One way to erase these symbols is to select **Reset all Note Attributes** from the *Attributes* menu, or as key command. This method, however, also resets all other Note Attributes.)

13.20 Score Preferences

The Score Preference settings are global settings, which are saved when you quit Logic, and are effective for all song files. They can, however, be changed anytime, with immediate effect on all open songs.

You can open the Score Preferences with:

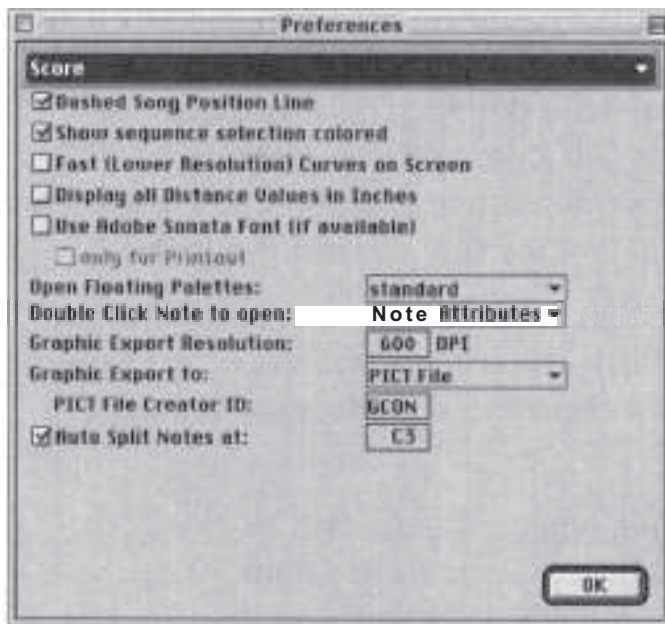
Local menu **Options > Score Preferences**

Global menu **Options > Settings > Score Preferences**

Key command **Score Preferences**

Directly from one of the other pages in the Preferences window with the flip menu.





Dashed Song Position Line

Determines if the Song Position Line in the Score Editor window is displayed as a solid line, or as a dashed line.

Show Sequence Selection Colored

If this option is activated, the stavelines of the currently selected sequence are displayed in color (blue), all other sequences will be displayed with black lines. This can be useful when changing the display parameters of sequences, since the parameters in the Display Parameter Box always only refer to the currently selected sequence/s. If it is not activated, all staves will be displayed black, whether selected or not.

Fast (Lower Resolution) Curves on Screen

This is mainly relevant for slower computers. The curves in slurs and score brackets will be displayed with a lower resolution on the screen, which speeds up screen redraws. This setting does not affect the printout.

Display All Distance Values in Inches

This relates to the measurement units in the Songs Settings and in the page rulers (Page Edit view only), which can be inches or centimeters.

Use Adobe Sonata Font (if Available) Only for Printout

This option activates the use of the Sonata font by Adobe for display and printing of the whole score. Please read the section *Sonata Font for Display and Printout of Notes and Symbols* on page 13-21.

Open Floating Palettes

Default Setting for the shape of Partbox Floating Palettes. Please read page 13-33.

Double Click Note to Open...

This setting determines, which window will open when you double click on a note head: Note Attributes, Event List, Hyper Editor or Matrix Editor.

Graphic Export Resolution, Graphic Export to..., PICT File Creator ID



Parameters concerning the PICT-Export function with the Camera tool. Please read the section *Graphic Exports* on page 13-21 for details.

Auto Split Notes at...

This parameter is only relevant if the chosen default Score Style on the recording track is polyphonic, and is using MIDI channels for the Voice assignment. In this case, newly recorded notes automatically are saved with the different MIDI channels used in the corresponding Score Style. They are allocated according to the split point set here.

13.21 Text: Input and Display

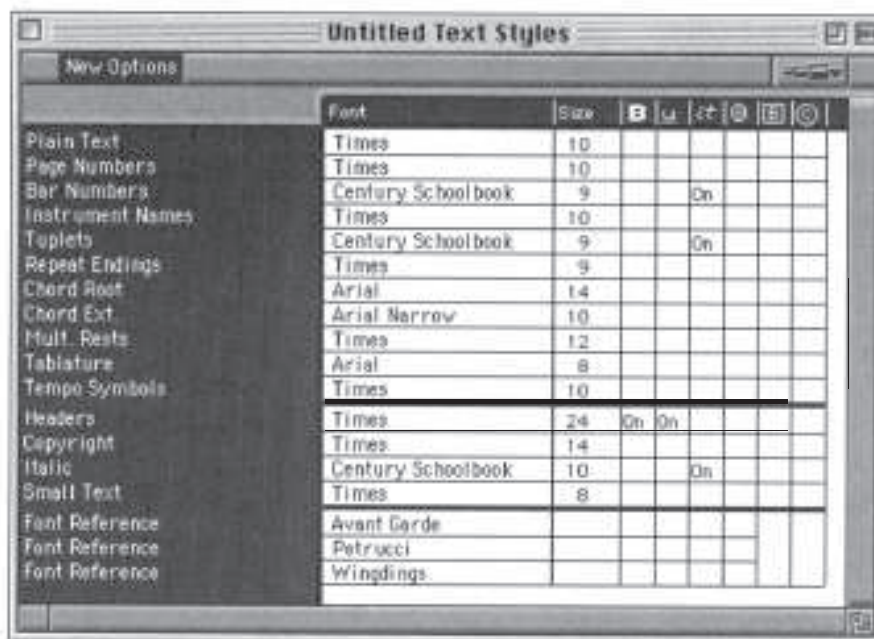
All text elements appearing in Logic Audio Platinum refer to user-definable default settings called *Text Styles*. When you insert text into the score, you can assign any of the predefined Text Styles to it. This way, you don't have to set all text attributes like font, size, style etc. again, every time you insert a new text object.

Text Styles

All Text Styles which are used in a song are listed in the *Text Style window*. There, you can edit the existing Text Styles, or create new ones. You can open this window the following ways:

Menu **Text > Text Styles...**

Double-click on the Text Style name in the Event Parameter Box (only visible if either text objects or no objects at all are selected)



The first eleven lines contain the System Text Styles, which exist in every song (but can be edited according to personal preference). Below these (separated by the horizontal line which is a bit thicker), there are the user-created Text Styles. Text Styles

are saved with the song file, so they can be different for each song.

Each line first shows the *Name* of the Text Style, then the *Font* (selectable with the flip menu), its *Size*, and the different columns for the style *attributes* (activated with a click in the corresponding field): **bold**, underline, *italic*, **outline**, **boxed** and **Circled**. Circled draws a circle around the text, which with longer texts is stretched horizontally according to the text.

You can edit these settings for both the System Text Styles and the User Text Styles. Any text fonts which are installed on your system can be used.

The System Text Styles

Plain Text

The default setting for “regular” text

Page Numbers, Bar Numbers, Instrument Names

The Text Styles for automatic page and bar numbering, and display of instrument names. These functions are described in detail in the section Numbers & Names on page 13- 128.



These Styles can also be edited in the **Layout > Numbers & Names** window. Changes in one of the two windows automatically update the settings in the other window, and affect the whole song.

Tuplets

The Text Style for triplet and other tuplet numbers.

Repeat Endings

The Text Style for anything which is written into the repeat ending boxes (usually just numbers, but text can also be entered there).

Chord Root, Chord Extension

For the display of chord symbols (see page 13- 156). ***Root*** concerns the root and the (optional) extra bass note in chord symbols, ***Extension*** is anything else.

Multiple Rests

The Text Style for the number above multiple rests.

Tabulature

For the display of the fret numbers in guitar tablature (see page 13-137)

Tempo Symbols

For the display of the numbers in tempo symbols, which can be inserted from the Partbox (page 13 -58).

Creating Additional Text Styles

Select *New > New Text Style*: under the lowest Text Style entry, a new line will appear, where you can define the attributes of the new Text Style(s). The names of any new style can be edited in the text entry field which opens when you click on the name of the Text Style.

Changing the Same Font in Several Text Styles Simultaneously

If you change a Text Style's font (flip menu) in the Text Style window while you hold the -key, this change affects all Text Styles which use the same font. This is a quick method, for example, to change the Logic default Text Styles to the fonts you prefer.

This method is also especially useful if you load a Logic song which was created on another computer, and uses text fonts that you don't have in your system (if this is the case, you will get a warning, message when you open the song). The names of

unavailable fonts will be displayed in parentheses in the Text Style window.

Font References

When you activate **Options > Show Font References**, any Text Styles which were created by directly editing a text object's attributes in the Event Parameter Box, or from the Text menu, are listed as *Font References*. **Delete Unused Font References** deletes any References which are not in use anymore. Editing the Font References in the Text Style window is not possible, except for changing the font. This display option mainly allows you to replace missing fonts in such Font References (see above).

Music Fonts as Fonts for Text Styles

In the above picture of the Text Style window, you can see a Text Style which uses the "Petrucci" font (not included with Logic). Fonts like that do not contain letters, but musical symbols only. Text objects using these Text Styles can be used for free positioning of musical symbols anywhere in the score. (Such as special percussion symbols).

Text Input

To insert text in the score, you have to get a Text Object from the Partbox. and drag it to the desired position with the mouse. During this procedure, the mouse cursor becomes the Text tool. You can also grab the Text tool from the Toolbox, and use it directly for text input. Each *Text Object* (except text in the Header, directly at the page margins, or outside the margins) is saved as a *Meta Event* within a particular sequence, at a certain bar position. This position can be seen in the *Info Line*, as you insert the text object. Text events are also visible in the Event Editor window, where you can change their position, but not the text itself.



After you have inserted the text object at the desired position and released the mouse button, a flashing text cursor appears at that position. Now you can enter text with the computer keyboard. The basic functions for moving the cursor, deleting parts of the text etc. are the same as in most word processors. As long as you are in text entry mode (indicated by the flashing text cursor), you can click on any position in the text to place the cursor there. You can also select parts of the text by dragging the mouse, and applying the usual cut/copy/paste commands. To leave text entry mode, press **Return**, or click anywhere outside the text.

After this, the newly inserted or edited text object is selected (the entire text flashes), and you can see its parameters in the Event Parameter Box:

Stave is only relevant in multi-stave Score Styles. It determines which staff the Text Object is assigned to. The *vert.pos* parameter measures the vertical distance to the top line of the staff, which is indicated here. *hor.pos* indicates the graphic horizontal deviation of the selected text object from its actual bar position (also in relation to the *Align* parameter).

Lyric ON defines the text object as a *Lyric Event* (song text). If a *Lyric Event* is placed at the same bar position as a note (above or below it), the horizontal distance to the previous and subsequent notes or rests is automatically expanded, so that there is enough space for the text.

Style: The Text Style of the selected text object,

Align: Determines the horizontal alignment of the Text Object. The first three settings relate to the bar position, the last three to the position on the page (each left, centered or right).



- : Here you can change the size and font for each individual Text Object, independent of the initially chosen Text Style, which serves as the default setting.

You can also change the settings for font, size, style and alignment of Text Objects with the sub-menus in the **Text** menu, while the desired Text Objects are selected.

Generally, you should not make those changes here, but rather create new Text Styles in the Text Style window, which then are also available for other Text Objects.

Editing Text

If a Text Object is selected, parameters can be edited in the Event Parameter Box. If you want to edit the text itself, either click on it with the Text tool or double-click on it with the Pointer tool, which will cause the flashing text cursor to appear again.

Deleting Text Objects

Like all other objects, Text Objects (or selected text inside a Text Object) can be deleted by pressing **Backspace**, as soon as they are selected.

Copying Text Objects

Copying whole Text Objects can be done with the usual methods:

Copy/Paste also *Multiple Paste* and *Paste at original Position*



Dragging with the mouse while holding the **Alt**-key. This, however, only works inside the same sequence.

Simultaneous Text Input into Several Staves

This works the same way with Text Objects as with other objects and is described in the section *Simultaneous Input of Objects into Several Sequences (Insert Multi)* on page 13-29.

 Example: insertion of the text “*accelerando a/ fine*” into all instruments at once.

Global Text

Global Text Objects appear in all *Instrument Sets* of a song (score, parts etc.), although they are inserted just once. The position of Global Text does not relate to bar positions (unlike regular Text Objects), but is defined as a graphic position on the page. Because of this, Global Text can only be inserted and seen in Page Edit view. The most obvious example of a Global Text object is a header line with the song’s name.

Input of Global Text

A Text Object is automatically global if it is inserted into one of the following areas on the page:

In the Header space (which is set in **Layout > Global Format**) Outside, or directly at one of the page margin lines

After you have inserted a Text Object in one of these areas, and finished text entry, you will see an *Event Parameter Box* with some additional parameters:

Pages: The pages on which the text will be displayed:
1 = first page,
2... = all pages except the first one, *Odd* = all pages with odd numbers, *Even* = all pages with even numbers, *All* = all pages.

Zone: The margin area which the text belongs to: *Top*, *Header*, *Side*, *Footer*.

Align: You’ll find two additional alignment options in this flip menu: for right and left alignment, you may select between alignment at the set page



margin, or at the outermost possible position. Although they are shown in the flip menu, the alignments relating to bar positions cannot be selected here.

Style, Size: These work the same way as for regular Text Objects.

Positioning of Global Text

After having been inserted in one of the margin areas, Global Text can be moved anywhere on the page with the mouse. With the regular mouse cursor (the Pointer tool), you are only able to move a Global Text Object vertically. To be able to also move it horizontally, hold down **ctrl** as you drag it, or just use the Layout tool. But remember: the position of Global Text objects is always measured in relation to the page (not to staves or to a bar position)!



Automatic Text Objects

In the Text group of the Partbox there are four objects which represent certain names that are automatically displayed, if these objects are inserted:



SEQ: Sequence/Folder: displays the name of the current Display Level. This can be a sequence, a folder or (on the highest level) even the name of the song file.

INSTR: The name of the currently displayed Instrument Set

SONG: The name of the song file

DATE The current date (at the time of the printout)

These automatic Text Objects can be inserted both as Global Text (in the margin areas), or as Text Objects relating to a bar position (in one of the staves/sequences). The settings for their appearance and exact positioning can be edited in the Event Parameter Box.

Example of Automatic Text

Insert the *INSTR* object above the top margin (centered, Pages set to 2...). It will automatically be treated as a Global Text Object. Now, create an Instrument Set named *Score* in the Instrument Set window. If you use this Instrument Set, there will be a line saying *Score* on the top of every page, except the first one. If you now extract parts for the individual instruments using the method described on page 13-118, each part will display the particular (track) instrument's name on top of every page, except the first one.

Lyrics

For song lyrics, there is a Partbox object named *LYRIC*. Before you start with lyric input, select the Text Style that you want to use for the lyrics, in the Event Parameter Box, with no object selected (indicated by *Default Inserts* in the title line of that box). Just click on an empty spot, anywhere in the Score window, to deselect everything.



Now drag the *LYRIC* object below the first note of the melody, or just click the Text tool  at that position. Watch the *Info Line*: each *LYRIC*-Object has to be at the same bar position as the note it belongs to (taking into account display quantization). Now, enter the text for the first note. Now, don't press , but , which moves the text cursor automatically to the beginning of the next MIDI note. If a MIDI note is displayed as several tied (graphic) notes, it is also possible to move the cursor only to the next *graphic* note with  , in order to be able to write several syllables below one longer note.

With this method, you can enter all the lyrics in one process, although each syllable is saved as an independent *LYRIC* object.



Editing lyrics is the same as editing regular text (see above). The Event Parameter Box displays the same parameters (*LYRIC* is set to ON here).

Apart from the fast input mode using the -key, the only difference from regular Text Objects is that Lyric Objects affect the distance between the notes to which they are assigned. This is to create enough space for the text to be displayed properly, without overlap. If the text (or the Text Style) is changed later, the note distance will again be calculated accordingly.

Some Hints for Working with Lyrics

Setting all Words and Syllables to the Same Height

If you don't enter the lyrics in one continuous process as described, the Lyric Objects probably will not have the exact same vertical position. To quickly set all *LYRIC* Objects to the same height (vert.pos), select **Edit > Select Similar Objects** after having selected one object, select the key command *Align Object Positions Vertically*. Note: you do not want to use this method if there are multiple lines of lyrics for different verses, since all of them would be set to the same height.



Several Verses Written below each other

There can be several *LYRIC* objects assigned to the same note, so you can insert several verses for a song, one below the other. Start with the first verse and make sure the vertical distance to the notes is correct. Then start with the input of the second verse. As long as you stay in fast lyric input mode, all Lyric Objects for the second verse will stay at the same height as the object you started with, in this case the first word of the second verse.

Note: it is important that Lyric Objects are assigned to the right sequence, if you work in a full score display (Remember: selected staves are displayed with blue stavelines). Generally it is recommended that you insert lyrics in linear view, and single sequence display mode, especially if you insert more than one

verse. In linear view, it is also easier, for example, to rubber band select several Lyric Objects, to change all their positions. (Please read the section *Changing Several Selected Objects Simultaneously in the Event Parameter Box* on page 13-67.

13.22 Chord Symbols

Chord symbols (text object **CHORD**) are inserted into the score like regular Text Objects, with the mouse from the Partbox or directly with the Text tool **I**, while holding the **Alt**-key. A text cursor will appear. First you enter the root of the chord, and then the additional information.

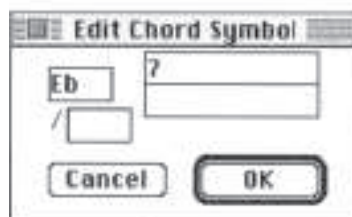


Important: when you insert chord symbols, be careful to insert them at the correct bar position. Watch the *InfoLine* as you insert the chord object.

A chord symbol can include a root note, a separate bass note and two lines of *extensions*. Try to insert a simple chord symbol first. Type “Eb7” and press Return. The result will look like this:

E^b7

Now double-click on the root Eb. The following edit window will open:



There are four entry fields for the different parts: one for the root, one for an additional bass note and two for *extensions*, which are placed above each other. If you click on any of them, an entry field opens where you can edit the chord symbol, or enter the desired additional information.

Chord Symbols

As you enter the text for the chord in the Score window, you can influence the way the inserted text is distributed to the different fields of the Chord Symbol Edit Box:

First, enter the root, then the extensions, then-after typing a slash-the (optional) additional bass note.

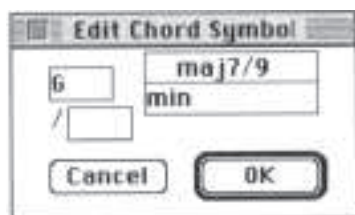
As soon as you enter a number, everything following that number will be placed in the upper extension line. If you write "(G7b9/b13)", the whole "7(b9/b13)" will be placed in the upper line, the lower line remains blank.

If you write letters before a number (for example FMAJ7/9), Logic puts the letters in the lower line, and everything after the first number in the upper line.

You can influence this automatic placement with a comma in the text: everything before the comma will be put into the lower line, everything after it into the upper line. If you use more than one comma, only the first one is relevant, the others are ignored. To illustrate these principles, here are a few examples in both forms, first the way they are entered as text, then the resulting chord symbol as displayed in the score:

Ab,add9	D7(#4)/C	C# sus4,7	Gmin, maj7/9
A ^b add9	D ^{7(#4)} / _C	C ^{#7} sus4	G ^{min} maj7/9

Note: in the last two chord symbols, blanks were entered to create additional horizontal space between the particular extension and the root. The editor window of the last chord symbol, for example, looks like this:



If you want to change a chord symbol, you can either double-click on the root, and edit the chord symbol in the resulting edit

window, or you can double-click directly on one of the extensions. A text cursor will appear in the extension line, and you can edit the extension like regular text.

The appearance of chord symbols can be edited in the *Text Style window*, where you can choose different Text Styles for the root (*Chord Root*) and the extensions (*Chord Extensions*).

Fast Input of Several Chord Symbols in One Process

If you insert several chord symbols into the same staff, there is a shortcut similar to the fast lyric input mode. After writing the first chord, don't press , but **Tab**. The text cursor automatically moves to the position of the next displayed note or rest. Enter the next chord and repeat the process, or press  several times again, to move to the position where you want to place the next chord symbol.

Hint: if you want to insert exactly one chord per bar into a very busy part, it is much faster to create an empty sequence, and enter the desired chords there (the cursor always moves to the next bar when you press  in empty sequences). Afterwards, you can either merge that sequence with the original sequence in the Arrange window, or copy and paste all chords at once.

Enharmonic Change of Chord Symbols

The roots of chord symbols can be enharmonically changed with the same commands as notes (**Attributes > Accidentals** or key commands). An example: "Gb7" becomes "F#7" by selecting **Attributes > Accidentals > Enharmonic Shift**. Only additional bass notes have to be edited directly in the Chord Edit window.



Transposition of Chord Symbols

Chord symbols in transposing Score Styles are transposed automatically, just like notes. Also, if the *Transpose* parameter (for playback transposition) in the *Sequence Parameter Box* (Arrange window) is changed, all chord symbols in that sequence will be affected accordingly,

German Chord Symbols: “H” instead of “B”

If the *German Chord Symbols* option is activated in the *Global Format* dialog window, the note **B** in chord symbols is globally displayed with its German name H. The English *Bflat* is written as *B*, according to its German name. This option is activated automatically, as soon as the letter **H** is used during chord symbol input.

Copying Chord Symbols

Chord symbols can be copied and pasted with the usual *Copy/Paste*, *Multiple Paste* and *Paste at original Position* commands, or (only inside the same sequence) by dragging them with the mouse, while holding the -key.



Precise Graphical Position of Chord Symbols

Horizontal Position: Bar Position, Align, Hor. Pos.

Each chord symbol is stored as part of a sequence at a certain *bar position* (Often there will also be a note at that same bar position). This is the main parameter for the horizontal placement in the staff.

The *Align* parameter determines which part of the chord symbol is aligned to that bar position: the left edge, center or right edge of the letter representing the chord root. This is set with the flip menu in the Event Parameter Box (see illustration). The setting for newly inserted chord symbols is always def.: in this case the alignment depends on the global *Chord Symbol Alignment* setting in the *Extended Layout Parameters* window. There you will find the same four alignment options: left, centered, right and " - ". The last option in this list should only be used when you load a song with chord symbols that have been inserted or modified with an older Logic version (2.6 or earlier). In these versions the alignment parameter was not available yet. To achieve identical positions as with the old version, set the *global* chord symbol alignment to " - ". Since

this setting is for backward compatibility only, it should not be used in new songs.

In addition to bar position and alignment, there is also the *hor. pos* parameter. It determines the additional graphical deviation of the chord symbol from the bar position

Be careful: if you move a chord symbol with the Arrow tool, it is easy to unintentionally change its bar position. To avoid this, press the -key during this process: the cursor becomes the Layout tool, so only the graphical parameters *event. pos* and *hor. pos* will be changed, but not the bar position itself.

Vertical Position: *Vert. Pos*, *Stave*

The *vert. pos* parameter determines the vertical position or height of the chord symbol above or below the staff. In Score Styles with more than one staff, the *Stave* parameter determines the staff which the *vert. pos* parameter refers to.

*A hint: to adjust a number of chord symbols to the same height above (or below) the staff, select them and use the key command **Align Object Positions Vertically**.*

Rhythm Section Parts

To create typical rhythm section parts with chord symbols and *Beat Slashes*, as shown in the example below, select a Score Style whose *Rest* parameter is set to “*Slash*”.



Chapter 14

The Transform Window

Usage

The Transform window can be used to alter existing events, according to definable parameters. These parameters can be saved as “transform sets”, and called up again for later use (for more on this, read the section Calling up *Presets and your own Sets* on page 14-13).

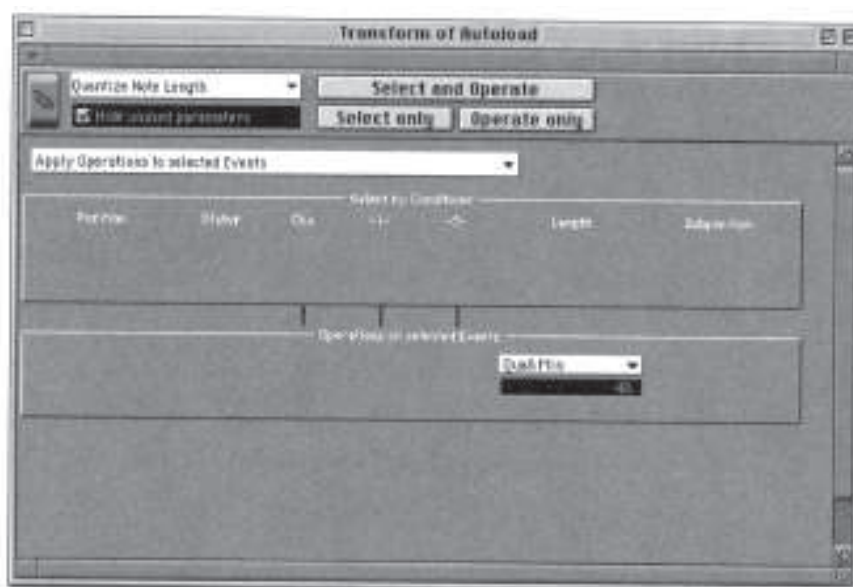
The Environment contains a similar transformer object for real-time editing of MIDI data (see the section *Transformer* on page 5-42).

For those of you who may be used to sequencers such as Vision and Performer, the Transform Window will likely be one of the more initially confusing areas of Logic to learn. Those who are experienced with Cubase, Cakewalk, and Notator SL will be in familiar territory, because the Transform window is similar to certain functions found in those programs.

Although it may initially be hard to understand, the Transform window provides an extremely high degree of flexibility and power, when you need to perform complex edits and transformations of MIDI data. A short time spent mastering its operation will yield substantial rewards during your sequencing sessions. You will find that there is practically no edit operation, or transformation that you can’t accomplish here.

Opening the Transform Window

Select **Windows > Open Transform** ( ) to open a new Transform window.



Effective Range

The Transform window is used to edit events only. As with other editors in Logic, the operations are only carried out on selected events. These might be:

- All selected events (in one of the editors),
- All events in selected sequences, or
- All events in selected folders.

You can use the link button in the top left corner to link the effective range with other windows.

All selected events on the same display level fall within the effective transform range, as do all the events in the selected arrange objects.

The title bar of the Transform window indicates the effective range, by giving the name of the song, and, if applicable, the name of the first selected sequence:



Purpose of the Transform Window

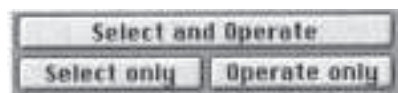
Conditions

This part of a transform function defines which events will be edited. If the *Select and Operate* or *Select only* options (see below) are used, the transformer checks to see whether the individual parts of the events (e.g. Position, Status, Channel, etc.) fall within the effective range, as defined in the pull-down menus under **Select by Conditions**.

Operations

The Operations section of the Transform window is where you define the edit operations to be carried out. These operations are defined in the pull-down menus under **Operations on selected Events**.

Actions



These three buttons implement the actions in the Transform window:

1

Select only

All events in the effective range which fulfill the Conditions will be selected, but not transformed. You can use this action to refine your own selection commands, and insure that your Conditions are set to affect only the events you wish to act upon.).

Operate only

All selected events are transformed according to the *Operations settings* (Conditions has no effect). This is useful if you want to edit events you've already selected manually).

Select and Operate

This is a combination of both the above actions: first, events are selected according to the *Conditions*, and then they are transformed, according to the *Operations* settings.

After each action, you can see how many events have been selected and/or transformed in the title bar, as shown below:



*After activating **select only**, you can alter the selection in an editor, before carrying out the transformation by activating **Operate only**.*

Display



The *Hide unused Parameters* checkbox allows you to remove all the unused pull-down menus in the conditions and operations boxes. This gives you a better overview of the settings being used, and also protects the hidden menus from any accidental alterations.

14.1 Transformation

Mode

You define the basic operating modes in the pull-down menu above the *Conditions* box:

Apply Operations to selected Events

As described above, the *Operations* are carried out on the selected events (default setting).

Apply Operations & Delete unselected Events

The selected events are edited, but in addition, all non-selected events are deleted. This ensures that the only events

remaining after the transform operation are the ones that matched the Conditions.

If you set the **Operations** up neutrally, you can use the Transform window in this mode as a programmable filter-only events which match the Conditions survive.

Delete selected Events

The selected events are deleted.

In this mode you can use the Transform window as a programmable erase function. All the events which match the **Condition** are deleted, and all other events remain unchanged. The **Operations** setting is irrelevant in this mode.

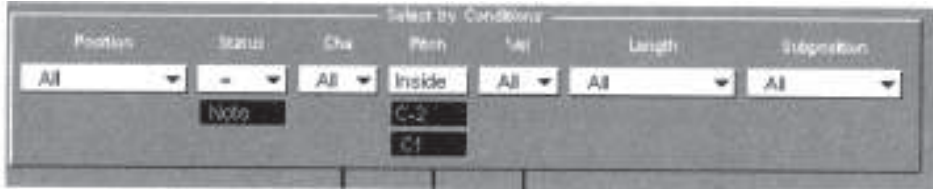
Copy selected Events & Apply Operations

The selected events are retained in their original form, as well as being copied and edited.

You can use this mode for applications like the following:

Imagine you want to add a mod wheel controller event (#1) 10 ticks before each note, with a value that matches the note's velocity. To do this, you set **Status** under **Conditions** as Note. Under **Operations**, set the **Position** to "Add" and below that to "-10". Set **Status** as **Fix Control** and **Pitch (-1-)** as **Fix 1**. The last of these operations means "the first data byte (- 1 -) receives the value 1" (the first data byte defines the controller number for controller events, and mod wheel events happen to be #1). If we analyse these settings, we see that all note events will be selected for operation. A copy of each will be made, and then moved 10 ticks earlier than the original, and it will then be converted to a CC#1 (Mod wheel), with a value that corresponds to the velocity of the original note, because the second data byte (Vel) is left unaltered.

Event Parameters



The individual MIDI event parameters have their own columns in the *Conditions* and *Operations* fields:

Position

Time position of the event, referenced to the start of the sequence. (not of the Song)

Status

Type of event.

Cha

Recorded MIDI channel of the event.

-1-Pitch

First data byte (note or controller number).

-2- Vel

Second data byte (velocity or controller value).

Length

Length of the note.

Subposition

Time position of the event, within a bar.

Conditions

The middle section of the window (*Conditions*) is where you define the conditions, which determine which events are

selected for the edit operation. Each column represents a different MIDI event parameter.

The condition is considered to be fulfilled when an event matches the defined effective range of all the event parameters. These ranges are defined in the pull-down menus found under each of the event parameter columns. (except the Status column) The possible value conditions are listed below.

The *All* setting in the top box means that all values fulfill the conditions, which makes all the other boxes in the same column superfluous.

Conditions for the Status

You have a choice of just two settings here: *All* (means all event types fulfill the condition) and “=“. Clicking in the box below, opens a pull-down menu where you can select from among note, poly pressure, control change, program change, channel pressure or pitch bend.

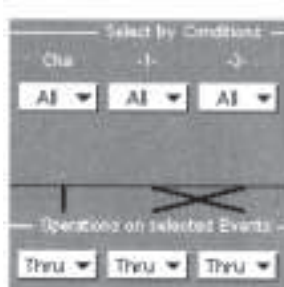
Conditions for the values of Position, Cha, -1- Pitch, -2- Vel, Length, and Subposition

You can assign one of the following value conditions for each of the parameters (except the status):

=	The value in the event, and the value in the box below it must be equal for the condition to be fulfilled.
<i>Unequal</i>	The value in the event, and the value in the box below it must be <i>unequal for</i> the condition to be fulfilled.
<	The value in the event must be smaller than the value in the box below it, for the condition to be fulfilled.
>	The value in the event must be larger than the value in the box below it, for the condition to be fulfilled.

- Inside** The value in the event must be within the value range of both the boxes below it, for the condition to be fulfilled.
- Outside** The value in the event must be outside the value range of both the boxes below it, for the condition to be fulfilled.
- Map** With this condition, incoming event components are compared to the map before they are evaluated. The two “Range” parameters then work the same as “Inside”. Incoming events whose mapped value lies within the range fulfill the condition—all others do not.

Exchanging Parameter Values

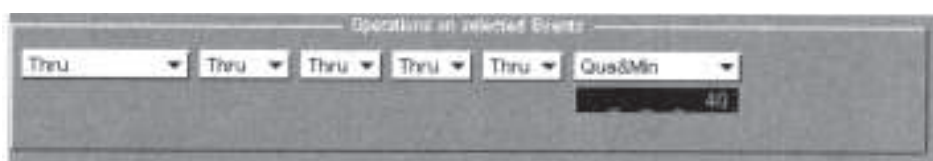


You can replace the value of each of the three event parameters *Cha*, *-1-* / *Pitch* and *-2-* / *Ve* with the value of a different parameter from this group. Click on the lines between the *Conditions* and *Operations* fields to change the source for the parameter value.

First the value is exchanged, and then the operation is carried out.

Operations

The bottom part of the window (*Operations*) is where you define any changes that will be carried out on events that meet the *Conditions*, with separate columns for each MIDI event parameter.



The *Thru* setting in the top box means that the relevant event part remains unaltered, which makes the other boxes in the same column superfluous.

Operations on the Status

<i>Thru</i>	The event type remains unaltered.
<i>Fix</i>	The event type is altered. Clicking in the box below, opens a pull-down menu where you can choose between note, poly pressure, control change, program change, channel pressure or pitch bend.
<i>Map Set</i>	This operation for the status of an event allows the universal map of a subsequent transformer to be controlled. The value of “-1- “ selects the position in the map. The value of “-2-” determines the value at this position of the map. Internally Logic sends a meta event pair: #123 with the position, and #122 with the value at this position. These meta events for altering a Transformer’s map can also be created in other ways.

Operations on the values of Position, Cha, -1- Pitch, -2- Vel, Length, Subposition

For all parameters (except status) the top box determines the type of operation which is carried out, using the values in the boxes below (referred to as the “set value”). The following operations are identical for the channel, and the first or second data byte, but the value ranges in the boxes below are automatically adjusted.

<i>Fix</i>	The parameter is fixed to the set value.
<i>Add</i>	The set value is added.
<i>Sub</i>	The set value is subtracted.
<i>Min</i>	Parameter values which are less than the set value are replaced by it. Larger parameter values remain unaltered.

<i>Max</i>	Parameter values which exceed the set value are replaced by it. Smaller parameter values remain unaltered.
<i>Flip</i>	The parameter values are reversed around a set point. Values above this amount are moved to the same distance below it and vice versa.
<i>Mul</i>	The parameter value is multiplied by the set value (4 decimal places).
<i>Div</i>	The parameter value is divided by the set value (4 decimal places).
<i>Scale</i>	The parameter value is multiplied by the top value and the bottom value is then added. This is a combination of <i>Mul</i> and <i>Add</i> . Negative values can be used to generate an inversion of the plus/minus sign and subtraction.
<i>Range</i>	meter values outside the set value range are replaced by the values of the limits (combination of <i>Min</i> and <i>Max</i>).
<i>Random</i>	Random values are generated within the set limits.
<i>+ - Rand.</i>	A random value between zero and the set value (positive or negative) is added.
<i>Reverse</i>	The parameter value is reversed within its value range (no value can be set here).
<i>Quantize</i>	The parameter value is quantized to a multiple of the set value.
<i>Qua&Min</i>	Like <i>Quantize</i> , but the quantization does not fall below the set value (a combination of the <i>Quantize</i> & <i>Min</i> functions with the same set value).
<i>Exponent.</i>	The parameter value is scaled according to an exponential function. The extreme values (0 and 127) remain unaltered. The set value determines the shape of the curve. Positive values

have an exponential scaling (increasing input values remain lower for longer, and then rise faster), and negative values have a logarithmic scaling (decreasing input values remain higher for longer, and then drop off faster).

Crescendo This only works if the position condition “Inside” has been selected (crescendos have to have some sort of start and end points.). Crescendo creates a smooth alteration in the current parameters, between the set value boundaries.

Rel. Cres This also only works if “Inside” has been selected as the position condition. The effect is similar to that achieved by Crescendo, but here the previous values of the parameters you’re changing are taken into account when the crescendo is created, preserving the relative feel of the original.

Here’s an example of creating a crescendo in note velocity values: First select “Inside” as the position condition, and “Status = Note”, then set “Crescendo” as the operation, with the minimum and maximum value, in the “-2-” or “Vel” column. Depending on whether the first value you set is smaller or larger than the second, you will get either a crescendo or a decrescendo. The original values of the events you change are irrelevant after the transformation, as they are completely changed by the crescendo function. In contrast, with the Rel.Cres. option, the original dynamics assigned to the notes will remain, even after Crescendo has been applied.

Use Map When this operation is selected, the set Map will be used.

14.2 Transform Parameter Sets

The settings for all the parameters described in the section *Transformation* on page 14-4 can be stored in transform parameter sets, which are available from the pull-down menu at the top left.

Presets

The first twelve items in the pull-down menu are write-protected presets. If you try to alter any of their parameters, Logic reminds you that you will not be able to save these alterations. However, you can alter the presets temporarily, but any changes you make will be discarded when you leave the preset. It is good to create your own transform sets, which can be configured and saved as part of the song.

Try calling up some of the presets, to acquaint yourself with the Transform window and its parameters.

Crescendo (see page 14-11).

Scale 14bit PitchBd (Pitch Bend data will be scaled so that the 14 bit information remains intact)

Double Speed (doubles the tempo by halving positions and lengths)

Half Speed (halves the tempo by doubling positions and lengths)

Humanize (adds a random value to the position, velocity and length of notes)

Reverse Position (reverses the positions of notes within a section)

Reverse Pitch (inverts the pitch)

Transpose (transposes the notes)

Exponential Velocity (alters the scaling of the velocity curve)

Velocity Limiter (limits the velocity)

Fixed Note Length (creates constant note lengths)

Maximum Note Length (limits the maximum note length)

Minimum Note Length (limits the minimum note length)

Quantize Note Length (quantizes the note length).

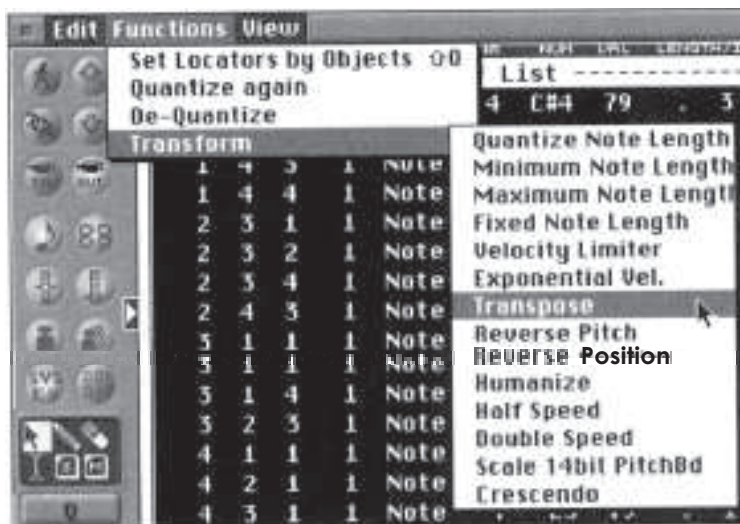
Your Own Sets

To store your own settings in a user set, select the bottom menu item: **** Create User Set! ****.

You can name the new transform parameter set by double-clicking the selection field. All settings will be stored immediately.

Calling up Presets and your own Sets

In the Event, Matrix and Hyper Editor windows, a sub-menu can be opened via **Functions > Transform...** from which the individual Transform sets can be selected directly (see below).



Chapter 14
The Transform Window

Chapter 15

Tempo

Tempo Display

If your song has a tempo that stays the same throughout, you can set this constant tempo in the Transport window.



The current tempo will always be displayed here, even if you are using programmed tempo changes or external synchronization.

You can set the tempo display format on the **Options > Settings... > Display Preferences** page (see page 17-21).

Tempo Track

Tempo changes are controlled by tempo events, which are stored in a special tempo track. This track is not visible in the Arrange window.

The tempo track applies to the whole song.

For details on creating polyrhythmic and polymetric structures, please refer to the section *Polyrhythms* on page 15-19. To find out more about changing the tempo of individual sequences, please read the section *by Time-Compressing or -Expanding* on page 3-23.

The tempo track also determines the relationship between incoming time code, and the current song position (see page 16-24).

Overview

Information on Changing Song Tempo

The easiest way to set *individual tempo changes* is from the Tempo List (see page 15-3). The Tempo List is a specialized Event List editor for tempo events.

Logic offers a simple way of creating *accelerandi* or *ritardandi* (steadily increasing or decreasing tempos): try using **Create Tempo Curve** (see page 15-8). Another way is to record tempo changes with the mouse, from the Transport window (see page 15- 15). You can do this even more accurately using the Graphic Tempo editor (see page 15-5). The Graphic Tempo editor is a hyper editor, with a fixed event definition for tempo events, and is really useful for manually editing existing tempo *changes* (see page 15-5).

You can also create *tempo changes with MIDI controller messages* (and thereby record these changes), with the help of the real-time Tempo fader (for more about this, read the section *The Real-time Tempo Fader* on page 15- 14).

If you want to change the *basic tempo of a song* which already contains tempo changes, the function to use is **Scale Existing Tempo Changes**, under **Tempo Operations** (see page 15-9). This is a special transform window for tempo events.

For more on manually generating a tempo map (by inputting MIDI notes rhythmically throughout the song-so-called “Human Sync”), please read the section *Tempo Interpreter* on page 15-11.

You can also record a “rubato” solo performance without a metronome, and then *create a tempo track that "fits" the recording*. This enables you to quantize accompaniment arrangements, or create a readable score. Creating a tempo track after a performance is carried out using the Reclock function (for more on this, read the section *Setting the Tempo after Recording (Reclock Song)* on page 15-16).

The Tempo Operations window contains a few other functions: for *stretching/compressing an existing Tempo curve* (read the section *Stretch Existing Tempo Changes on page 15 -9*), or for *replacing a passage containing several small variations in tempo, with one average tempo* (see the section *Creating a Constant Tempo on page 15-9*).

To find out how *to lock a particular bar of a song to a particular SMPTE timeframe* (for all of you out there who are using Logic to synchronize music to picture), read the section *Positioning Bars to Frames* on page 16-20.

15.1 Tempo List Editor

You can open the Tempo List by clicking and holding on the Transport's Sync button with the mouse. A pull-down menu will appear, from which you select **Open Tempo list...** or, you can simply select **Options > Tempo >Tempo List Editor**, or the global key command *Set Tempo Alternative..*



The layout, and way you use the Tempo List are very similar to the Event List (see page 10-1).

Creating Tempo Changes

Set the song position to the required point.



Click *Create*.

A tempo event appears showing the current tempo. You can alter the tempo in the tempo column.

If you click with the Pencil tool on the word “Tempo” in an existing tempo event, a new tempo event will appear, with an Input box for the position. Enter the required bar position and press .

Deleting Tempo Changes

You can delete tempo events by clicking them with the eraser, or pressing the *Backspace* key.



Copying the Tempo Changes from a Passage

Set the locators to the passage containing the correct tempo change.

Select **Edit > Select inside Locators**.

Copy the tempo events to the clipboard ( .

Deselect all tempo events (by clicking the background).

Paste the tempo events from the clipboard ( .

A position input box appears at the first tempo event, where you can alter the bar position. If the first tempo change is not at the start of the bar in the passage, be sure to alter the number of the bar, and leave the “fractional values” unaltered.

Press . The copied tempo changes will be selected, and you can undo the operation if necessary.

Alternative Tempo Lists

You can use up to 9 different tempo lists. This is useful in songs with tempo changes, so that you can;

temporarily slow down the tempo when recording MIDI sequences,
temporarily “switch off” the tempo changes,
try out different tempo variations.

To switch between the different tempo lists:

Open the tempo list (see above).
Choose **Options > Tempo Alternatives > 1 . . . 9**, and select the desired tempo list.

Other Functions

The entries in the tempo list interact with, and affect each other. For details, please refer to the section *Positioning Bars to Frames* on page 16-20. You can also make several synchronization settings here. All other functions are identical to those in the Event List (see page 10-1).

You can move any selected tempo event to the current song position using the key command *Pickup Clock Position*.

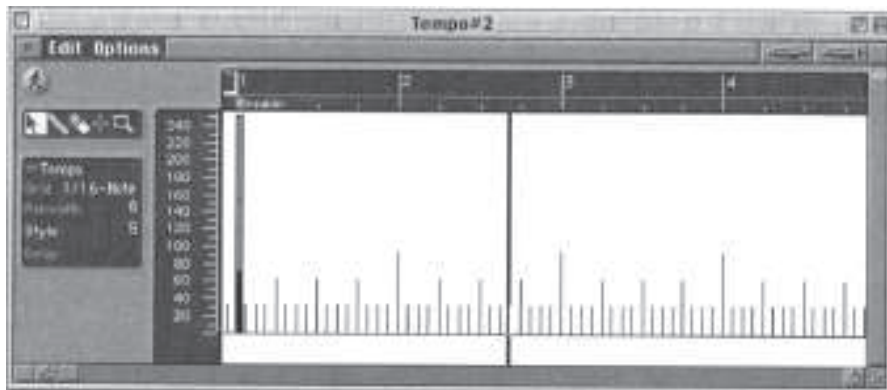


15.2 The Graphic Tempo Editor

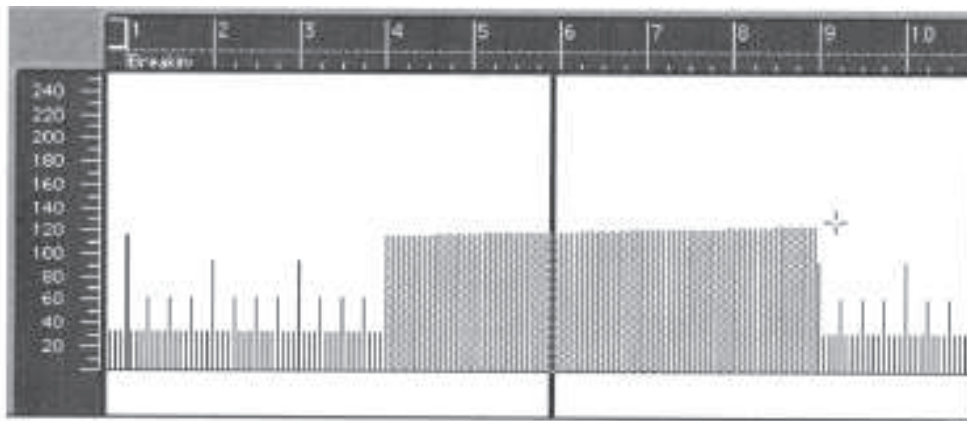
The Graphic Tempo editor is a specialized Hyper editor, that only allows you to draw and edit tempo events.

To open the Graphic Tempo editor, grab the Sync button in the Transport window and select **Open Graphic Tempo...** from the pull-down menu. Or, select **Options > Tempo > Tempo Graphic Editor**.





Creating Continuous Tempo Changes



Select the Crosshair tool.

Click on the required section in the working area, and hold down the mouse button.

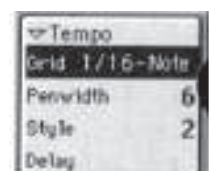
Keep an eye on the Info Line at the bottom of the window, which tells you the exact position and tempo. Set the start (or end) of the tempo change.

Release the mouse button.

Now set the end (or start) of the tempo change. If you want to create new tempo events (instead of altering existing ones), hold down the **I** key.

Click the mouse button.

Remember that the spacing of the added tempo events depends on the setting of the Grid parameter.



In most cases, the 1/16 note setting is enough to create the impression of a continuous tempo change.

Other Functions

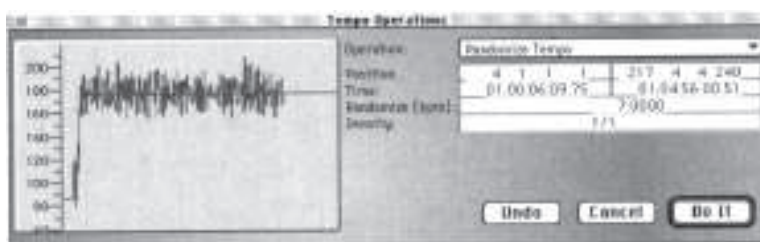
The operation of other functions are identical to those of the Hyper editor (see page 11-3).

15.3 Tempo Operations

The Tempo Operations window is used to edit existing tempo changes, as well as to create new ones. The window automatically displays the area you've selected for editing, and as the tempo curve is processed, the changes are displayed graphically in real-time, so you can see what's happening. The displayed tempo curve (which is calculated from the tempo settings you've made), is shown in red, or as a dotted line on black and white monitors).

Opening the Tempo Operations Window

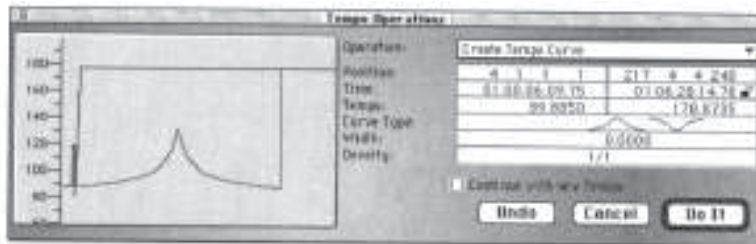
The Tempo Operations window can be opened from the Arrange window using **Options > Tempo > Tempo Operations**, from the Transport using the Sync button's pull-down menu, or by means of the key command *Open Tempo Operations*.



The different functions (*Operation*) offer you several interesting options for editing tempo changes, and the sync reference. The lock symbol lets you fix certain parameters so they don't change.

Create Tempo Curve

This function lets you create a lot of tempo changes with just a few parameters, and gives you a smooth overall change in tempo.



Three kinds of tempo curves are available from the Curve *Type* pull-down menu. Each of these classes has a *Curvature* parameter, which lets you determine whether the tempo should speed up or slow down, depending on whether you enter positive or negative curvature values.

Select the Curve *Type* you want.

Use the *Position* or *Time* settings to determine the region over which the tempo change should extend: the left entry is the start point, the right the end point.

Enter the start and end tempos you want in the *Tempo* line.

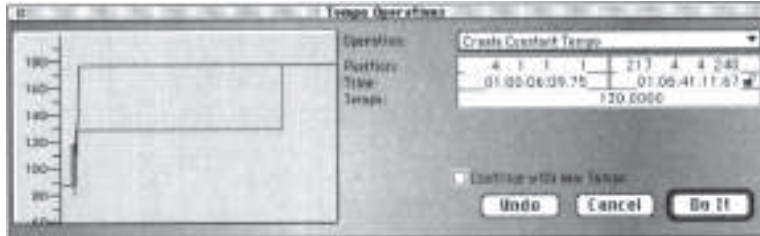
Finally, set the *Curvature*, and look at the graphic display, as the tempo changes.

The concentration of the tempo events created graphically can be altered using the *Density* parameter. Values of 1/8 and finer should only be used if really necessary (for example with very slow tempos and fast tempo changes). Even finer settings (1/16 and 1/32) can be entered, by holding down the  key.

Don't select a resolution higher than 1/8 or finer, just to make the curve Look smoother.

Operation:
Position:
Time:
Tempo:
Curve Type:
Width:
Density;

Creating a Constant Tempo



This function creates a constant tempo in any area you select. The default setting assigns the average tempo for the selected region.

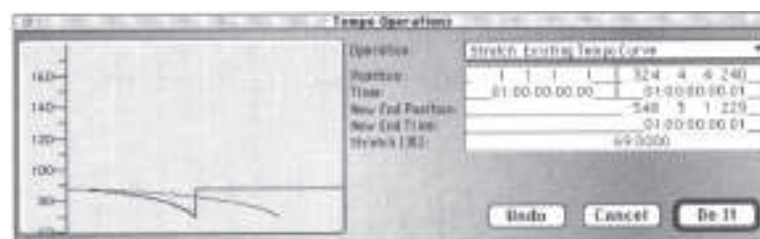
If you want to remove all variations in tempo between two points, without changing the SMPTE time for the right-most position, just click **Do It**.

Scale Existing Tempo Changes



With this handy function, you can alter existing tempo changes proportionally, or scale them. This lets you speed up an entire section of a song, without losing the relationships between any subsequent tempo changes within that region. You carry out the scaling by percentage. Positive percentage values speed up sections, and negative ones slow them down.

Stretch Existing Tempo Changes



The Stretch function lets you stretch or compress an existing tempo curve. The *Position* or Time values let you define the start point of the region you want to change. You then enter either the new end point of the tempo curve as a bar position, or SMPTE value, or set a *Stretch* value for the curve (in percent-age).

Thin Out Existing Tempo Changes



By selecting **Options > Tempo > Tempo Operations**, or by key command (*Open tempo operations*), you can access the Thin Out Existing Tempo Changes window.

Under *Position* or *Time*, you define the start and end of the area to be processed. *Density* defines, how many tempo events will remain per bar after processing. You can select values between one per bar (1/1), and 8 times per bar (1/8).

Randomize Tempo



By selecting **Options > Tempo > Tempo Operations**, or by key command (*Open tempo operations*), you can access the Randomize Tempo window.

Under *Position* or *Time*, you define the start and end of the area to be processed. *Randomize* defines the deviation from the

current tempo in beats per minute (bpm). *Density* defines how often the new tempo events will occur. You can select values between one per bar (1/1), and 8 times per bar (1/8).

15.4 Tempo Interpreter

The computer keyboard and/or incoming MIDI events can be used to set the sequencer tempo. Put another way, Logic allows manual synchronization (so-called “Human Sync”).

To do this, the key command *Tap Tempo* must be assigned to a particular key (any one you like), or to a particular MIDI event in the Key Commands window.



The sync is fairly loose, in that if the manual sync impulses stop coming in, the sequencer continues at the last tempo it received.

If you have an Unitor, a Mark Of The Unicorn interface (a MIDI Time Piece or MIDI Express), you can convert percussive audio signals into MIDI notes. This feature lets you use (say) a bass drum track (either live or from tape), to control Logic’s tempo.

Switching to Manual Sync

You can get into manual sync mode in the following ways:

manually, using the Sync menu;

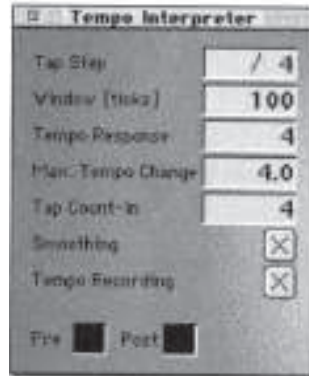
automatically in auto sync mode, if Logic encounters a *Tap Tempo* command.



The parameters for controlling Logic’s response to tap tempo commands can be found in the Tempo Interpreter window.

Opening the Tempo Interpreter

The Tempo Interpreter can either be opened from the Sync button's pull-down menu on the Transport, by using the key command *Open Tempo Interpreter*, or from the Arrange window, by selecting **Options > Tempo > Tempo Interpreter**.



Tap Step

Sets the note value that Logic will assign to the manual taps. The best results are obtained when using larger values, rather than smaller ones. *1/4 note* is usually a good choice.

Window

This sets the size of the time region displayed in the window, in which the tempo-determining tap notes will be entered (the size of the window is set in ticks, i.e. 1/3840 note). Only taps made within this window are used to determine the tempo. Any taps outside it are ignored.

The narrower the window, the more effectively Logic will be able to withstand the influence on the tempo of any taps that fall between the “defined” note values (as set with the *Tap Step* parameter).

The larger the window is set, the easier it becomes to create drastic tempo changes.

If you set the window parameter to 0, the tap window will disappear altogether, and all taps will be accepted as tempo-determining information. The sequencer will also come to a halt, if taps stop being generated.

Tempo Response

This sets the sensitivity for tempo changes: the larger the value, the greater the sensitivity. Suggested value: 4.

Tempo Interpreter

If the tap timing is imprecise, but Logic is supposed to be playing at a constant tempo, reduce this value to (say) 2.

Max Tempo Change

This sets the maximum tempo change possible per tap (in bpm). To obtain as regular a tempo curve as possible, follow this rule: set as small a value as possible (and only as large as necessary).

Tap Count-In

This sets the number of taps used as a count-in. Logic starts on the tap *after* this number of taps has been entered.

Smoothing

Switching on this function (the button will have an “x”), smooths out jumps in tempo in the taps being input (which could prove useful when using Logic live). You need to switch it off (no “x” in the button), if you want the sequencer’s tempo to follow your taps as precisely, and quickly as possible.

Tempo Recording

The “Tempo Recording” function creates a tempo list in record mode, to follow your taps. This function should normally be switched off.

Pre and Post

These are the ways your tap signals are displayed:

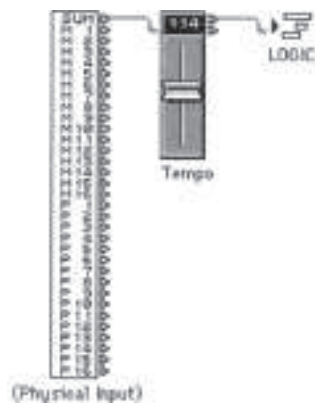
Pre: displays every tap that is input;

Post: displays every accepted Tap, i.e. each tap that occurs within the tap window (as defined above), on or around a tap step (again, as defined above).

15.5 The Real-time Tempo Fader

You can create a real-time Tempo fader in the Environment window by selecting **NEW > Fader > Special > Tempo Control**. This can control Logic's tempo (using Meta event #100). There is no need to connect this fader to anything, unless you want to record its output.

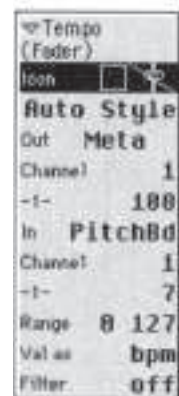
The tempo fader has a range of possible values from 50-177 bpm. External MIDI data controls it as you might expect, with an input value of 0 causing a tempo of 50, and an input value of 127 a tempo of 177.



If you place the tempo fader between the physical input and the sequencer input, you will be able to:

- record tempo changes;
- control the tempo with any MIDI events you like:

The tempo fader parameter box (shown right) displays Pitch Bend events as the In-definition (this is the default). This allows you to control Logic's tempo from the Pitch Bend wheel.



The Effect on Recording

This way, new data can be recorded for the tempo track “intelligently”. The original tempo data will only be erased in areas where there were tempo changes originally.

The Effect on Playback

If the Tempo fader is moved during playback or recording, all the data defined in the tempo list is switched off, until the next time the sequencer is stopped.

The Effect on External Sync

The movements of the tempo fader are ignored when external SMPTE sync is in operation.

The tempo fader can therefore be used to deactivate the sync reference during playback, or recording if you wish.

15.6 Tempo Functions

Recording Tempo Changes

Go to **Options > Settings > Recording Options**, and activate the Allow Tempo Change Recording checkbox. All tempo alterations which you make during recording will now be recorded as tempo events on the tempo track. You can then edit them in one of the tempo editors.

Adjusting the Tempo to Fit Regions

“Adjust Tempo using Object Length and Locators”: The MIDI tempo is adjusted so that the length of the selected audio region exactly matches the distance between the locators (cycle strip). If you want to use this function to adapt the MIDI tempo to match the tempo of a drum loop, make sure that the drum loop is cycling smoothly (with the help of the Sample Editor), and that the length of the region corresponds exactly to a specific musical length of the audio material. For example, so that the region counts out to exactly 2 bars when played back. Then, set a cycle zone to match the musical length of the region



(2 bars in this case), and select **Options > Tempo > Adjust Tempo using Object length...**

Setting the Tempo after Recording (Reclock Song)

You can record a sequence without using a metronome, and match Logic's tempo to the sequence afterwards.

This allows you to improvise freely, without any tempo restrictions, perhaps at the beginning of a new song. Previously, it would have been difficult to add further accompaniment to this improvised section later, as the improvised part would have no relationship to any meter that could be followed by other parts. This problem can be overcome, using the *Reclock Song* feature.

This function creates tempo changes automatically, in such a way that the improvised notes can be fit to a bar pattern that makes sense. What you actually hear played back is not altered by this process.

If you'd like to know how this process works in detail, here goes: the notes are fixed to their absolute time positions, and then the bar pattern is altered to fit these, by means of one or more tempo changes. Finally, the notes are released from their absolute fixed positions, so they line up with the newly created bar pattern.

This process can require a bit of manual tidying up afterwards. For this reason, it's usually better to try and establish a link between what you play and the musical meter as you actually play, if at all possible. In other words, play to the click if you can...

If this isn't possible, you should first see if you can tap a footswitch at quarter-note intervals as you play your improvised section. If you can, you could then assign the footswitch tap to the Tap Tempo function, and record the tempo changes in your section as you play, with the help of the tempo interpreter? tempo recording function.

If none of the above will work, *then* it's worth trying out the Reclock function.

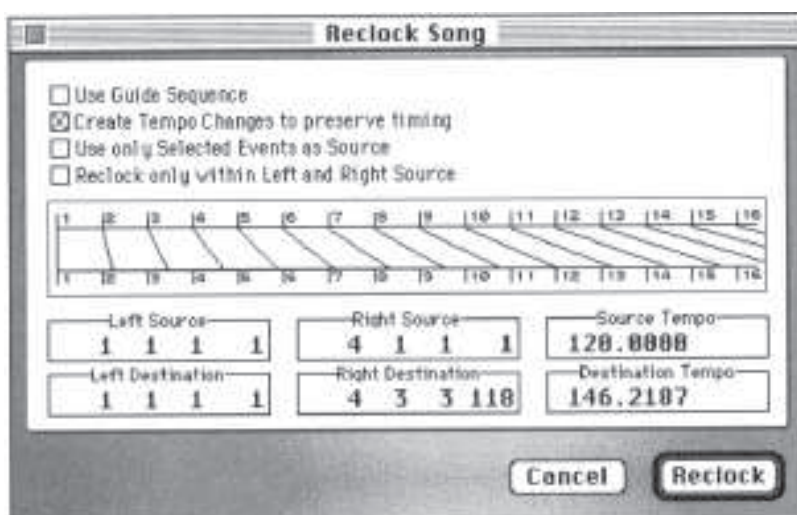
There are two ways of using the Reclock function:

- You may apply Reclock directly to the sequence you've played in:
 Advantage: no guide sequence is needed.
 Disadvantage: You have to know the precise position (in bars; quite tough to work out for the end note) of two notes, one at the end and one at the beginning of the sequence, as well as the length of the freely-played sequence in bars. Or...
- ...you create an additional guide sequence containing 1/4- or 1/8- notes, with the same timing as the freely-played sequence.
 Disadvantage: You have to record another sequence of the same length as the original (i.e. the guide sequence),
 Advantage: You only have to give the starting position.

We advise using the guide sequence method, because this saves you the hassle of having to enter the exact position parameters of the two guide notes.

Select **Options > Tempo > Reclock Song** to open the Reclock Song window.

You need to know the number of bars in the recorded sequence, and the exact destination bar position of one note right at the beginning, and one right at the end of the freely-played sequence.



The graphics in the middle of the reclock window show the relationship between the old (top), and new bar grids (below).

Left Source, Right Source

This is where you set the current bar position of the start and end notes in the source sequence. Open an Event List, if you want to find out what these are.

Left Destination, Right Destination

This is where you set the desired bar position (destination position) of the start and end notes.

Create Tempo Changes to preserve timing

If this option is checked, a set of tempo changes is created by the Reclock function. If you want a fixed target tempo, do not check this box, otherwise a series of tempo changes will be created to compensate for any timing deviations from the average tempo. If you want the recording to remain unaltered, you have to manually set the *Destination Tempo* derived by the function.

Use only Selected Events as Source

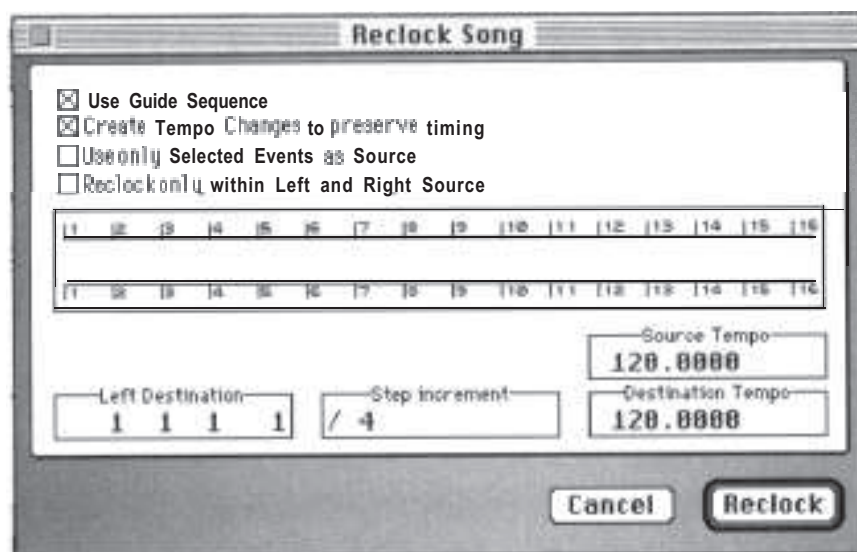
If this option is checked, only the selected events of the source sequence are used as a basis for any tempo changes.

Reclock only within Left and Right Source

If this box is checked, tempo changes are only created within the defined region.

Guide Sequence

If there are a lot of tempo changes in the recorded sequence, it is advisable to use a guide sequence. This is another sequence, in which you record quarter notes or eighth notes of any pitch, in the same meter as the freely-played sequence. Then select the guide sequence before you call up the reclock function.



Use Guide Sequence

You should check this box if the selected sequence is not the source sequence, but a recorded guide sequence with constant note values. In this case, only the following parameters appear below the bar display:

Left Destination

Where you enter the current position of the desired first beat in the guide sequence.

Step Increment

This is where you enter the note value of the beats in the guide sequence.

15.7 Tips and Suggestions

Polyrhythms

The Problem:

You want to simultaneously play parts of the song which have different tempos, or at least merge them over long stretches, but Logic only allows one global tempo at a time.

The Solution:

First record one of the parts at a suitable tempo.

Select all the relevant arrange objects and select **Functions > Object > Lock SMPTE Position** .

You can now alter the tempo, without affecting the position of the fixed objects. Any alteration in the time signature will not affect the playback.

Now record the other part.

You can select all the objects at any time, and use the same function to fix them to SMPTE time, and then restore the original tempo. Make a note of the different tempos.

Count-in for a Tempo Change

The Problem:

Logic is synchronized to a tape machine via time code. The song has a slower instrumental intro, and then the drums begin at a faster tempo. At the end of the intro, the drummer needs a count-in at the faster tempo.

Comments:

This problem is a difficult one, because with external synchronization you can't alter the tempo without either losing the correlation to the tape track, or recalculating the SMPTE offset. Even the simplest solution is relatively complex.

The Solution:

First record the count-in, e.g. as a 4/4 cowbell sequence on the first bar of the drums at the faster tempo. Make sure you record the first beat of the following bar as well; this point is important. Set a suitable quantization.

Lengthen the count-in sequence at the front, or create an empty sequence at least two bars in front of it, and join the two sequences.

Open an Event List by double-clicking on the count-in sequence. Here, select **View > Position & Length in SMPTE Units**.

Double-click on the position of the first note, and copy its position to the clipboard by pressing  .

Open the tempo list. Temporarily set the first tempo event (intro) to that of the following value. Make a note of the original intro tempo.

Now select all notes in the Event List of the count-in sequence ( .

Double-click on the position of the last count-in note (the one on the first beat of the following bar), and paste in the position from the clipboard by pressing  .

Now select **Functions > Object > lock SMPTE Position**.

Set the original tempo of the intro in the tempo list.

During the intro with the slower tempo, you will now hear the correct count-in for the following passage at the faster tempo, without having to make any complex calculations.

Scene-orientated Film Music

The Problem:

Film music often contains segments at different tempos, which all have to commence at certain absolute times in the film (i.e. at certain SMPTE positions). However, each segment has to have its own tempo, and some references to bars and **beats**—this is how Logic’s position grid and quantize functions work, after all.

Example:

Imagine you’re working on some music for a TV series. The time code start point is 00.00.00.00, and the director’s require-

ments are for a 20-second, slow intro, then 12 seconds without music, and then a fast theme to accompany a chase.

Basic Rule:

Use tempo changes to ensure that the bar grid in the sequencer agrees with the real one.

Solution to the Example:

First, play the slow intro at any tempo you like. A tempo event will appear at Bar 1, that will apply to the whole song at first.

Then comes the clever bit. Add a second tempo change at the end of the long intro. (if the tempo is 120 bpm, the tempo change would come at bar 11) The new tempo is unimportant, it will be replaced in a minute anyway.

Then comes the pause: in this example 12 seconds, a reasonably long time. Here is where you enter the third tempo change. All you need to do is type in the SMPTE time you want this to happen at (in the example 32 seconds). The bar number can be almost anything you like, as long as there is a reasonable distance between the second and third change (there's more than enough in this example, as Logic can cope with very small distances between tempo changes). Logic now automatically changes the tempo in the second tempo change to make the third happen at the right SMPTE time. Place the third tempo change on a bar line (say bar 15) so you can use the metronome and quantize functions later.

That's it! The second tempo change is what makes it all work: it "protects" the tempo of the first section, and ensures that the third tempo change can fall on a round bar line number.

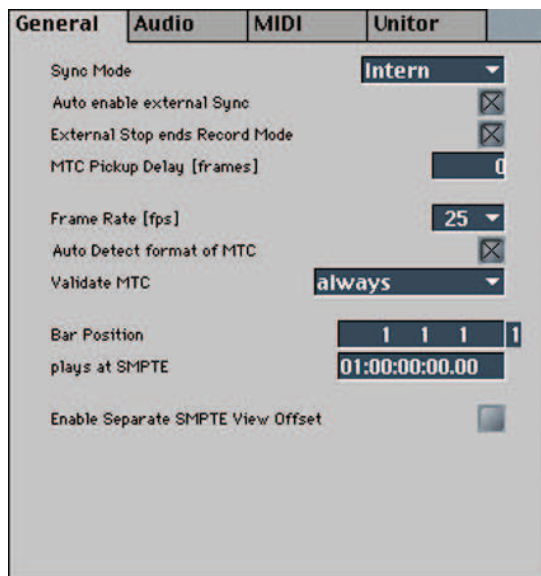
Chapter 16

Synchronization

This chapter begins by describing the synchronization window. This is where all Logic's synchronization settings can be made.

The remaining sections deal with other functions in the realm of synchronization (page 16-13), synchronizing a digitized film stored as a file (page 16-20), the basics of synchronization in general (page 16-24), and finally possible problems and their solutions (page 16-33).

16.1 Synchronization Window



You can open this window from the Arrange window using **Options > Settings > Synchronization Settings...**, or by click-holding the Sync button in the Transport window.



At the top, there are tabs for selecting the various pages: **General**, **Audio**, **MIDI**, **Unitor**.

General

This page of the synchronization window contains the major synchronization parameters for running Logic as a slave.

Sync Mode

This parameter defines the master to which Logic is to be synchronized:

Internal

Logic's internal timer. Logic is the master. External devices can be synchronized with MIDI Clock or MTC (the relevant settings are made on the "MIDI" page).

MTC

"MIDI Time Code". Logic runs as a slave. The MIDI Time Code can either arrive at a MIDI In port, or be generated by a MIDI interface "translating" it from incoming SMPTE code.

MIDI Clock

MIDI Clock and Song Position Pointer. Logic runs as a slave. Clock and SPP can be received at any MIDI input.

Manual

Tempo Interpreter. Manual synchronization or "Human Sync Mode". Logic runs as a slave to impulses, which the tempo interpreter recognizes as valid beats. The tempo interpreter can be controlled by any MIDI events, or by a key on the computer keyboard.

Auto Enable External Sync

When this option is activated, Logic runs as the master (Sync Mode Internal), until it receives a synchronization signal—either in the form of MTC, Clock/SPP, or from the Tempo Interpreter.

Logic automatically locks to the first synchronization signal it receives. Please be sure that different synchronization signals don't arrive simultaneously—because *there can be only one* time code master.

External Stop ends Record Mode

This option means that during external synchronization, recording stops whenever the time code ceases.

If the option is switched off, Logic stops, but remains in record mode (Record + Pause).

MTC Pickup Delay [frames]

Usually the parameter should be set to zero for the quickest possible pickup time in MTC sync. However, there are some devices which seem to transmit imprecise MTC commands when they start. As a result, the sync might not be perfectly reliable, and there could be a changing offset every time the sync is established. In such cases, you can set a delay for MTC pickup, and Logic will ignore the first MTC commands. A value of 25 to 30 corresponds to a delay of about a second, depending of the frame rate. Use this parameter when synchronized to certain hard disk recorders, **if the sync doesn't appears to be consistently reliable.**

Frame Rate

This is where you set the frame rate (in “fps”, frames per second). This frame rate applies to both transmitted and received time code.

Frame rate Typical applications

24	Film
25	Audio (Europe) and PAL Video
(30 d)	Unusable (not real-time)
30	Audio (USA) and NTSC Video (s/w)
29,97 d	Audio (USA) and NTSC Video (color)
29,97	Extremely rare (not real-time)



"d" stands for “drop frame”. In “drop frame” formats, certain frames are left out, following a regular pattern. To distinguish between them, formats without drop frame are sometimes referred to as “nd” or “non drop”.

Auto Detect Format of MTC

With this option, the incoming time code is analyzed and the correct frame rate is set automatically. In general you should leave this option switched on.

Please note that it is not possible to automatically distinguish between MTC frame rates 29.97 and 30, because;

the MTC standard does not allow a distinction, and

a measured rate of 30 fps could also be 29.97 fps time code running too fast, and vice versa.

Logic automatically interprets frame rates with “approx. 30 fps” as either 29.97 df or 30 nd, depending on whether or not the drop frame format is used. This interpretation will usually be correct, because only these two formats are actually used as a standard.

“Auto Detect...” only switches to 29.97 df or 30 nd if one of the other conventional formats was previously set. If you want to synchronize Logic to one of the unconventional frame rates, you have to define the format manually. This setting will not be altered by “Auto Detect.....”

SMPTE Offset

This is where you set the SMPTE offset for the song. Because songs do not always have to start precisely at bar 1, you can select any bar position to be played at the set SMPTE time.

The preset is 1/1/1 at 1:00:00:00. The SMPTE offset 1:00:00:00 is normally used, because it allows you to pre-roll some time code.

SMPTE View Offset

If you want to see the absolute time from the start of the song in Logic’s time display, rather than the actual SMPTE time, click Enable separate SMPTE View Offset. Then choose *Bar Position* 1/1/1 displayed as SMPTE 00:00:00:00 (preset).

Synchronization Window

If necessary, you can set other view offset values here, instead.

The Tempo List always shows the real SMPTE time, never the **SMPTE View Offset**. However, the **SMPTE View Offset** is used in all other windows.

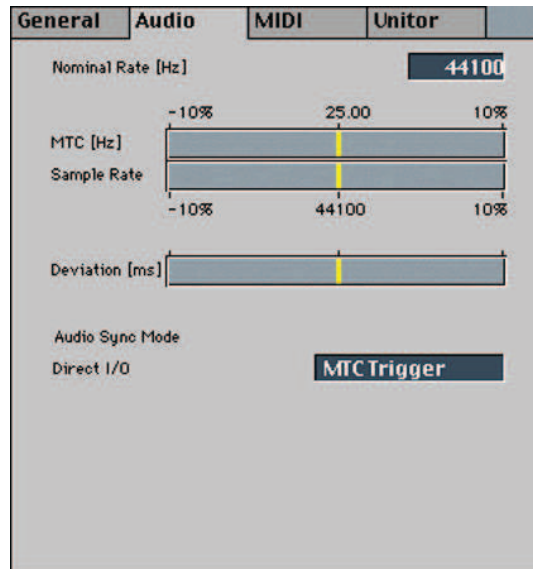
You can individually set whether the SMPTE time is shown with or without **SMPTE View Offset** for each open Transport window. In the Transport window “arrow” menu, there is a Use **SMPTE View Offset** option.

Audio

This page of the synchronization dialog window contains all the parameters relevant to synchronizing Audio and MIDI.

Nominal Sample Rate

This is where the selected nominal sample rate is shown.



You can change the sample rate in the Audio menu of the Arrange window, if your hardware supports several different sample rates.

MTC [Hz]

This display shows the deviation between the incoming MTC, and its nominal frame rate.

If the deviation is too large, please make sure that you have set the tight frame rate (on the “General” page of the synchronization window). If

in doubt, set the frame rate to 24fps and switch on “Auto Detect format of MTC”.

If the frame rate is correct, you can use this display to adjust the tape speed of the master machine to the nominal value (i.e. the same speed as used when the time code was recorded). Adjust the varipitch control on the master machine, until the vertical yellow line is exactly in the middle.

Sample Rate

This display shows the deviation of the sample rate from its nominal value.

Bear in mind that some audio hardware (e.g. SoundManager) will not allow any variation in the sample rate.

Deviation

This display shows the current phase deviation of the word clock from the time code master, in other words, the deviation between audio and MIDI.



With varying time code you can see in this display how Logic Audio regulates the sample rate of the hardware in “MTC continuous” sync mode. Even with large time code variations, there is no deviation between audio and MIDI. Your audio hardware must be capable of continuously variable sample rate, for this to function.

Small deviations between audio and MIDI are unavoidable, because MIDI can (and should) follow the time code master directly.

Sync Mode

This is where you define how each individual piece of audio hardware should be synchronised to an external time code master.

Depending on its design, not all audio hardware can work in every sync mode described below. This depends particularly on whether or not the hardware’s sample rate can be controlled.

MTC Continuous

Audio regions are started in sync, and the sample rate is continuously regulated according to the variations in the time code master. With this method, even very long audio regions stay in sync.

MTC Trigger

Audio regions are started in sync, but are then played with a constant sample rate, regardless of any variations in the time code master. Logic Audio always uses the set nominal sample rate (44.1, 48 kHz, 88.2 or 96kHz).

Sample rates of 88.2 and 96kHz are only supported by Logic Audio Platinum.

This mode is suitable when it is vital to retain the absolute pitch of a recording. If the speed of the time code master deviates from the nominal value, you may have to split long regions into shorter sections.

MTC Trigger/Auto Speed Detection

Similar to “MTC Trigger”, but in addition, the Speed Detection constantly monitors the speed of the time code master, while Logic is running. The next time you start Logic, it uses an adapted sample rate.

This mode keeps long regions in better sync with the time code master, although not as closely as “MTC Continuous”. However, it does use a constant sample rate, which is not affected by variations in the time code master.

External/Free

Logic has no influence on the sample rate. The audio hardware has to ensure that the position and sample rate of the audio regions match. This sync mode is only advisable if you can be sure that the word clock and time code master are running in

sync, for example by using an external SMPTE/wordclock synchronizer.

Digital

Similar to “External/Free”, but the hardware is also set up to synchronize to the sample rate of the incoming digital signal (with DAE hardware you have to make this setting in the DAE Hardware Setup).

SMPTE Slave Driver

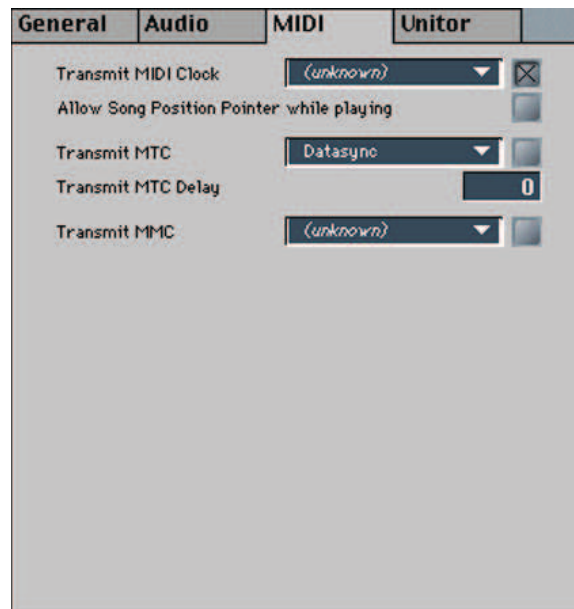
Exactly the same as “Digital”, but in addition, a SMPTE slave driver is set to the same sample and frame rates as in Logic. The SMPTE slave driver is a device for synchronizing ProTools hardware to time code or word clock.

MIDI

This page is where you configure Logic to send timecode via MIDI, when the sequencer is running. This enables you to synchronize external devices as slaves to Logic, which acts as the master.

Transmit MIDI Clock

The switch on the right activates transmission of MIDI Clock. In the display next to it, you can choose the port from which MIDI Clock is sent.



Every time you start, “Song Position Pointer” (SPP) is also sent.

Because not all devices can process SPP, the real-time message “Continue” is also sent. The exception to this is when you start at position 1/1/1. In this case, the real-time message “Start” is sent, instead of “Continue”.

MIDI Clock to all Ports (+)

Both MIDI Clock and MTC can be sent to all ports simultaneously: select “+”.

MIDI Clock can easily be sent via a bus, along with other normal MIDI events (notes, controllers). With multiport MIDI interfaces (like the Unitor or MTP), better timing is achieved by sending MIDI Clock to all ports, rather than to several individual ports.



If MIDI Clock is transmitted to all ports, the events are sent only once from the computer to the interface. If you address individual ports, one event has to be sent for each individual port, which worsens the timing for all ports.

Note for MIDI Time Piece 1 Users:

The old MTP I did not support the selection of specific outputs. For this reason, you should always use an “MO” or “PO” setting, as MIDI Clock Events will then always be available from every output, whichever setting you select.

Unfortunately, Song Position Pointer (SPP) does not note the port setting. If, for example, you select “M3”, SPP will only be transmitted on output 3, while MIDI Clock is available on all outputs. This would mean that a Drum machine on output 4 would start and stop without problem, but not be able to determine its current barposition from Logic, at any given time.

Allow Song Position Pointer while playing

According to the MIDI Standard, Song Position Pointer is normally only sent when you start. This option in Logic causes SPP to also be sent while the sequencer is running. The advantage is that external devices can also follow Logic in Cycle mode.

If your **external** devices cannot process SPP you should switch off this option. This improves the timing when cycling.

Transmit MTC (MIDI Time Code)

The switch to the right activates transmission of MIDI Time Code. In the display next to it, you can define the port to which MTC is transmitted.

It is not advisable to send MTC to all ports. Unlike MIDI Clock, if you want to send MTC (which is very data-intensive), use a MIDI port which is not being used for anything else.

Transmit MTC Delay

This parameter allows you to delay the transmitted MIDI Time Code. Negative values mean that the MTC is transmitted earlier. This enables you to compensate for any reaction delays in the external MTC slaves.

Send MMC (MIDI Machine Control)

This is where you can switch on transmission of MIDI Machine Control. These commands are then always sent whenever you operate Logic's transport functions (Start, Stop, Rewind, etc.).



MMC is normally used when Logic is running as a slave to an external master (e.g. ADAT), where you want to control the external master's transport functions from within Logic. Logic therefore acts as MMC master, and MTC slave simultaneously.

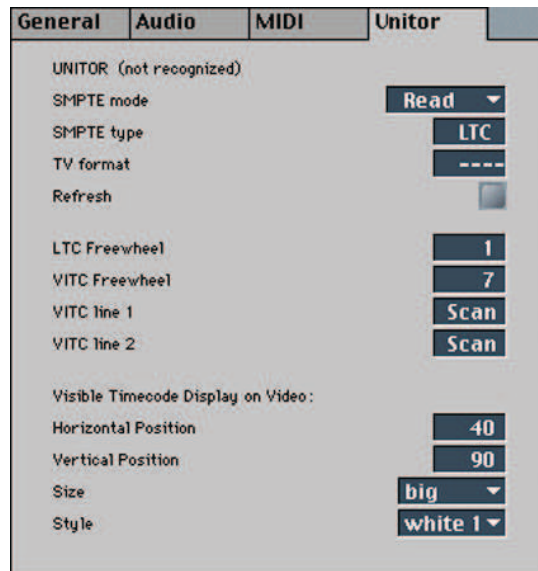
If you want to use the external master's transport controls, you don't need MMC. In this case Logic as the slave, will follow the MTC master

You can also use MMC to put tracks from the MMC slave device into record-enabled status. For details, please refer to the section *MIDI Machine Control (MMC)* on page 16-15.

Unitor

This page is where you set the major synchronization parameters for the Unitor 8.

Please note that you can comprehensively edit the Unitor 8 using the "Unitor 8 Control" software supplied with it, or with SoundDiver.



SMPTE Mode

Here, you can instruct the Unitor 8 to write SMPTE. You must switch from *Read* to *Generate*, and set the frame rate and start-time in the "General" page.

SMPTE Type

This is where you define what SMPTE format you want to use:

<i>LTC</i>	Longitudinal Time Code is written to a tape track.
<i>VITC</i>	Vertical Interval Time Code is written invisibly to a video tape.

TV Format

Allows definition of the television format for time code burn-in:

<i>PAL</i>	The video format used in Europe. If you are working with video in SECAM Norm, select PAL here.
<i>NTSC</i>	The video format used in the USA and Canada.

You don't just need this setting when writing and reading VITC. If you are operating with LTC, and you want to generate a video picture with burned-in SMPTE time, you have to set the correct format here.

Refresh

In Refresh mode, fresh time code is generated in sync with received time code. "Refresh" works with both VITC and LTC.

You should always use "Refresh" whenever you have to copy an LTC track, because you cannot directly copy LTC without a considerable loss of quality. When copying whole multitrack tapes, you should patch all the tracks directly, but refresh the TC track via the Unitor 8.

You should only use Refresh mode if you are copying time code.

LTC Freewheel, VITC Freewheel

Here, you can set the freewheel time in frames for LTC, and VITC individually.

The freewheel parameter affects the SMPTE reader, and specifies how long the synchronizer continues transferring MTC to the sequencer, after time code ceases to be read.

 Long freewheel times can maintain synchronization, even if there are drop outs in time code, but they also increase Logic's reaction-time after the time code master stops. In practice you should set the value as large as necessary (for sustained operation), and as small as possible (for short waiting times).

VITC Line 1, VITC Line 2

VITC is written into two lines of the video picture, which are normally invisible. The lines should not be adjacent, and are usually situated between 12 and 20.

If “Scan” is enabled, the VITC lines are automatically recognized. You should only enter the lines manually if there are problems with “Scan” mode.

Visible Time Code Display on Video

Here you can set the position, size, and color of the time code counter window, which is burned-in with the video picture.

16.2 Special Functions

Switching on External Sync

Switching on the sync button on the Transport causes Logic to synchronize to the sync source you’ve chosen.



You can use the Sync button to turn off the external sync at any time, without changing your selected sync source. This allows you to remove Logic temporarily from the sync master’s time axis. This could prove useful if, for example, you need to quickly edit a sequence while the external sync source (tape machine, VTR etc.) is still running.

Recording with External Synchronization

When “Record” is pressed during external sync, Logic goes into MIDI Record mode, but does not start until it detects external time code.

Incoming MIDI Time Code Display

The flashing dot on the Transport window’s sync button indicates that Logic is receiving error-free MIDI time code.

If the dot “sticks”, an error has occurred. Although Logic is capable of dealing with many MTC errors itself, you should nevertheless check the quality of your SMPTE signal, as well as other potential sources of error.

MTC Interpretation

Because the MIDI standard only supports four of the possible six time code formats (the 30 fps and 29.97 fps formats cannot be differentiated), Logic has to decide which format is “intended”, when it encounters incoming time code:

incoming MTC format	is interpreted as
24 fps	24 fps
25 fps	25 fps
30 drop fps	29.97 drop fps
30 fps	30 fps

In other words, the much more commonly used 29.97 fps and 30 drop fps formats are used rather than the uncommon 30 fps, and the virtually-unheard-of 29.97 drop fps formats.

However, you can manually set the format from the Tempo List editor to whatever you like: for example to convert material to 30 fps for black and white TV transmission in the NTSC format.

Measuring the Frame Rate

If you select **Options > Settings... > Compatibility**, and then select the New Phase Control Timer Model, when you use external synchronization via MTC/SMPTE, Logic measures the external frame rate. The result is shown in the Transport window, beneath the tempo.

Receiving MIDI Clock/SPP

Sync via MIDI Clock/SPP is the most accurate method, if Logic is being synchronized to a bar-referenced master.



MIDI Clock has a resolution of 24PPQN (pulses per quarter note), while Logic has an internal resolution of 960 PPQN (some 40 times more accurate). For this reason, Logic has to interpolate the 39 steps between two incoming clock impulses itself.

If you experience small variations in the timing between master and slave, you can tighten the sync when in external sync mode, by entering the expected tempo changes from the master into Logic's internal tempo list as well.

Even if you don't take this step, the sync should still be fairly tight, as long as you avoid large deviations, such as an internal setting of 200 bpm, with an external tempo of 40 bpm.

Continue Event

When a MIDI Continue Event is received, Logic doesn't leap to the last valid position received via MIDI Clock. Instead, playback recommences from the current song position. This allows you to change the song position manually while the sequencer is stopped, and then restart from the new position, with a MIDI Continue command.

MIDI Machine Control (MMC)

MMC is a set of MIDI commands, which Logic uses to control the transport functions of any MMC-capable tape machine. The recording process can also be controlled and automated from Logic via MMC. This tape machine, then provides the SMPTE signal to which Logic sync's to as a slave (see the section *General on* page 16-1).

You can control connected devices from Logic using the normal "transport" functions (including direct positioning and cycle jumps). Don't forget that Logic has to wait for the connected device to finish rewinding or forwarding.

Switching on MMC

Activate *MIDI Machine Control* from the sync button's pull-down menu. You can also temporarily switch the function off from here, to allow you to carry out any necessary quick edits.

This option can also be accessed by selecting **Options > Settings > MIDI Options**.

Record Functions

Logic supports up to 64 MMC tracks, so it can operate devices like the Alesis ADAT™ via MIDI machine control.

Each arrange track can be made to act as a tape track by selecting an instrument with the “tape deck” icon (#305). You need to create only one instrument with the “tape deck” icon, and you may then assign that same instrument to as many tracks as you need to control your external recorder. It is a good idea to group these tracks together in their own folder. These tracks must be placed at the top of the Track List, in your Arrange window. If you pack them into a folder, this must be the first track in the list.

 The “tape deck” (#305) icon is the only icon that actually affects the way an object behaves. All the other icons are purely graphic in nature.

If the tape track is the current record track, the following functions apply:

Selecting the tape track switches the corresponding track on the tape machine to “record ready”, and deactivates the “record ready” status of any other tracks. To select several tracks for recording, use .

The record button in the Transport window sends the “record strobe” command to the tape machine. This also puts Logic into MIDI record mode, and sends an MMC “Play” command to the tape machine (HDR or whatever). Logic doesn’t start, until it receives time code back from the multitrack.

If you use the autodrop function, the tape machine goes into record at the left locator position, and drops out of record at the right locator position.

If you click on any track—not just a tape track—while holding down the  and  keys, you can individually toggle

the record ready status for each track of the tape machine, i.e. selected tracks are switched on, and non-selected tracks are switched off. By clicking on a track while holding down the  key, you can simultaneously switch all other tracks out of record ready. If the current record track has been assigned a tape deck icon (see above), you must not use .

 (or the key command *Record Toggle*) is used to toggle the record status, if a tape track is the currently selected record track.

After finishing an MMC-operated recording, Logic automatically creates an empty sequence on the tape track. This is to let you know that a recording has taken place on the tape. This applies to all MMC recordings, including those controlled by the autodrop function. If you activate several tape tracks using , the corresponding number of sequences are created. If there is already a sequence with an identical start point on a tape track, no new sequence is created on the same track, to avoid overlapping objects.

You should finish all MMC-controlled recordings with STOP or . Some tape machines react differently to a series of "MMC Record" commands. Sometimes Logic shows a track to be recording, when the tape is actually playing back (or even worse, the opposite situation.). So, to be on the safe side, you should always finish a recording with STOP or .

MMC under OMS

You can specify a port for the transmission of MMC (MIDI Machine Control) commands when using OMS (Open Music System). If OMS is installed and in use, you can select the port in the dialog **Options > Settings > Synchronization Settings > MIDI** below the Option *Transmit MMC*.

Alesis BRC

For owners of the BRC control unit: set the BRC up so that MIDI time code is filtered out during fast-forward and rewind.

“Gen Sync” must be switched on, to allow MTC to be transferred.

Bar Ruler-Time Ruler



Use View > **Show SMPTE Time Ruler** to individually switch the bar ruler in each window to a time ruler. The number on the left of the vertical lines refers to minutes, and the one on the right refers to seconds. If the vertical height is sufficient, both the time ruler and the bar ruler are displayed.



Event Position and Length as Time

In the Event List editor, selecting **View > Position & length in SMPTE units** switches all the positions and lengths to SMPTE times.

Both functions have the same key command *Positions/Time Ruler in SMPTE units*.



If the *time* ruler is switched on when you move an object's position:

the position in the Info Line is also shown as a SMPTE position (e.g. in the Giant SMPTE Display/Giant Bar Display transport buttons),

holding down the  key switches the snap grid to frames, and holding down the  key as well, switches it to subframes.

Naturally, the time ruler takes account of any tempo changes.

Positioning Objects to Frames (Pickup Clock)

set the song position to the required SMPTE time, for example, by going to a specific frame of the film. Select the object which you want to place at this time position (it doesn't matter whether this is an individual event, tempo event or an arrange object). **Pickup Clock Position** then sets the start time of the object as required, **Pickup Clock and select next event** selects the next object after moving your current selection, so that after making alterations, you can easily continue.



When using Pickup Clock with audio regions, it is the region's anchor (not start-point) that is moved to the song position.

Fixing Objects to Frames

With film synchronization, you often want specific sound effects to play at a specific SMPTE time, rather than a particular bar position.

If the tempo of the piece needs to be altered later, then the SMPTE time location of events that have already been positioned will change. The **Lock SMPTE Position** function prevents this from happening.

You can SMPTE-lock single events in the Event List, or entire Arrange objects (with all the events they may contain).

Functions > Objects > Lock SMPTE Position

With this function, selected objects are “locked” to their current SMPTE position. If there are any subsequent tempo changes, the bar positions of the “locked” objects are adjusted accordingly. You can tell that an object or event is locked to a SMPTE position by the small padlock symbol in front of its name.

Copies made of locked sequences or folders do not preserve the locked setting, whether copied/pasted via the clipboard, or by dragging while the  key is held down.

Functions > Objects > Unlock SMPTE Position

All selected objects which were “locked” to their SMPTE position are now fixed to their current bar position, just like normal events. This means they may once again be shifted by any future tempo changes. The padlock symbol in front of the name disappears.

Moving the SPL by Frame

In addition to the usual position commands, you can also use the Key Commands *Rewind One Frame* and *Forward One Frame* to move one frame backwards or forwards.



Positioning Bars to Frames

If you want a particular bar in the song to coincide with a specific SMPTE time, you have to alter the tempo of the preceding passage. The following procedure will save you from having to do this by trial and error:

Open the Tempo List (page 15-3).

Create a tempo event at the desired bar position (page 15-3).

Set the desired time position for this tempo event in the ***SMPTE-Position*** column. The preceding tempo event is automatically adjusted to generate the correct bar and time position for the inserted tempo event.

You can then delete the tempo event you inserted, if you want to keep the same tempo for the following passage.

16.3 Synchronizing Film Files

Synchronizing QuickTime™ Movies



QuickTime™

QuickTime™ is a Macintosh standard for digitized, data-compressed films which

can run in individual windows. If you want to play these films, you need the System extension shown below in your System folder.

You also need a fast hard disk, and above all a fast Mac, so the movie may be played smoothly. If you want to record a film like this yourself from a video recorder, you need specialized hardware (an AV Mac at least), and the relevant software.

You can make QuickTime movies run in a Logic window, in sync with a song. Wherever you move the song position the film follows, and vice versa.

Opening a Movie

Select **Options > Movie...** in the Arrange window, to open a typical file selection box. You can then select a QuickTime movie from your hard disk.



If you select **Options > Movie as Float...** *you* can open a QuickTime movie in a float window, as the name implies.

Reopening a Movie

The key command, Open **Movie *Again***, reopens a movie window with the same film. This is particularly useful if you wish to open the film in a different screenset.



Working with Movies

In the QuickTime window, you can enter the SMPTE offset of the film at the bottom (the “Movie Start”). This is independent of the SMPTE offset for the song. By grabbing and dragging the position slider at the bottom of the picture, or by using the forward/rewind buttons to the right, you can move the film to any position, and Logic will jump to the corresponding song position. All the QuickTime transport functions interact with those of Logic.

Optimal Image Size

If you click once on the video image, the QuickTime window will expand to fill the whole screen in such a way that the entire video image, whatever its shape, just fits your monitor. The window borders will also disappear (*Best Size without Borders*).



This proportional enlargement does mean that strangely-shaped images may not fill the monitor. However, there are two advantages: the proportions of the image are retained, and, because the graphics card is not overloaded with conversion calculations for altering the proportions of the movie, a faster screen refresh rate can be obtained.

Clicking on the image restores it to its original format.

If you have an accelerated graphics card and a second monitor, movies are usually attached to the monitor that does not contain the main menu. You can check or change this setting from the Monitors and Sound control panel.

Since MacOS System 7.5, all images on connected monitors have had individually rounded edges. This can affect the performance of an accelerated graphics card. If yours appears to work more slowly under System 7.5 than under older MacOS versions, please contact the manufacturer of your graphics card.

Options

Clicking and holding on the movie image opens the option menu. From here, you can set various proportional image formats, from half-size to 8x size.

The following image sizes are also selectable, to allow a non-proportional enlargement of the image if it is oddly-shaped for the monitor you are using:

Synchronizing Film Files

Pixel 512 x 384

Changes image size to NTSC Underscan format.

Pixel 640 x 480

Changes image size to PAL Underscan, or NTSC Overscan format.

Pixel 768 x 576

Changes image size to PAL Overscan format.

Make Proportional

This selects the next-largest proportional setting.

Center

Choosing this places the video image in the middle of your screen.

Best Size without Borders

This option is equivalent to the click on the video screen, as described in section *Optimal Image Size* on page 16-22.

Synchronization Methods

The **Contiguous Sync** option allows you to switch between the following sync methods:

When **Contiguous Sync** is turned on, Logic controls the playback speed of the movie, so as to keep music and pictures in sync at all times. If the film has a soundtrack, the audio signal will be transposed slightly, if necessary, by means of sample rate conversion.

This sync method is possibly more precise, but requires greater processing power.

“Movie Drop Sync” (with **Contiguous Sync** switched off) means the QuickTime movie will be shown at its original speed. If any time corrections need to be made, this is done by

repeating frames or omitting them. If there is a soundtrack, a quiet click will be heard at this point. This method is also usually reasonably accurate.

Try out the different methods, to see which one gives the best results for your computer. When you switch sync methods, Logic may need a few seconds to establish the new sync method.

In Internal Sync mode, the Global Timer Adjustment parameter (Options > Settings... > Compatibility) allows you to alter the replay speed of QuickTime movies. By using extreme settings, you can investigate the interaction between the film, your graphics card, and the different sync methods. Don't forget to return these parameters to normal after experimenting with them.

16.4 The Basics

Summary

If you want to synchronize Logic to a second sequencer system (workstation, drum machine) MDI Clock/SPP is almost always the best choice.

For all other situations where you use external synchronization (tape machine, stand-alone hard disk recorder, video recorder), use MTC. To synchronize Logic to SMPTE time code, you need a “synchronizer”, which converts SMPTE into MTC. This can be done by nearly all interfaces with multiple MIDI ports.

Time Code and Clock Synchronization

Synchronization involves ensuring that several devices run absolutely in time with each other. This doesn't just mean that the devices start at the same point, and run at the same speed.

To set exactly the same speed would require infinite precision (even with digital devices). Instead, synchronization means that while the devices are running, *every point* along the time axis of all the devices must be linked.

This requires that the devices to be constantly “connected” to each other. This is only possible if one of the devices acts as the “master”, defining the current position, while the all other devices act as “slaves”, and constantly try to follow this position as closely as possible.

There are two different ways of doing this, depending on the type of devices being used:

1. With devices like sequencers or drum machines, the positional information shared is expressed in bar positions- in other words, they use “bar-referenced synchronization”.

2. Devices like tape machines or hard disk recorders usually use “time-referenced synchronization”- the positional information shared by these devices is time information, in what is known as time code. Devices that use time-referenced sync include tape machines for video, like VTRs (video tape recorders) and VCRs (video cassette recorders). With time-referenced sync, the tape speed or sampling rate is not dependent on the musical tempo of the recorded song.



Bar-referenced synchronization is only appropriate if you are using devices from the first category. As far as devices from the second category are concerned, time-referenced sync really ought to be used. A single tape machine could theoretically control several devices from the first category, by playing recorded bar-referenced time code. However, for various reasons, most professionals would work using time-referenced code. For one thing, this is the only way to sync additional machines up to the tape later.

A sequencer synchronized to tape has to calculate the bar position from the time position, using its tempo track.

Bar-Referenced Synchronization

First, a quick trip down memory lane..

Chapter 16

Synchronization

Pulse Clock and FSK (historical)

The original method of synchronizing analog sequencers, or drum machines to each other was by transferring electrical impulses (clock signals) at specific intervals. The norm was 24 ppqn (pulses per quarter note), which is equivalent to a resolution of 1/96 note. However, some companies used 48 (Korg, Linn), 64 (PPG), 96 (Oberheim) or 384 ppqn (Fairlight). By way of comparison, Logic has a resolution of 960 ppqn.

By encoding these impulses as the "shift between two pitches" (FSK code, frequency shift keying), it was possible to record this kind of code onto tape. It was mainly used to synchronize drum machines to tape machines. There were plenty of disadvantages to it, however

- FSK code laid down a fixed tempo.

- It was impossible to program intros/outros later.

Both Pulse Clock and FSK had two further disadvantages:

- If any impulses were lost through signal dropouts, the synchronized devices would run constantly out of sync from that point onwards.

- Because there was no position indicator, you always had to start the song from the beginning (FSK 2 or Smart FSK overcame this by encoding the song position as well).

These disadvantages eventually killed off clock and FSK, neither is now used professionally.

MIDI Clock/SPP

With the advent of the MIDI Standard, an equivalent to clock impulses was incorporated into the command protocol: MIDI Clock. MIDI Clock events are transmitted by the master at a rate of 24 times per quarter note. To avoid having to start songs from the beginning every time, an additional MIDI command is also transmitted: song position pointer. This transmits 1/16 notes from the beginning of the song. Because two data bytes are available for encoding (14 bit), it is possible to distinguish a maximum of 16384 different 1/16 notes or 1024 bars. The slave recognizes the current song position within this region, and synchronizes to it.

Modern devices always use a combination of MIDI Clock and song position pointer.

If a sequencing program has a higher timing resolution than 1/96 note, the positions between must be interpolated. The resolution of Logic is 1/3840 notes.

Time-referenced Synchronization

Time-referenced synchronization originates from the field of film synchronization, but nowadays it is also used for audio work. This is why it divides a second not into tenths and hundredths, but into frames. One frame was originally the time it took for a single frame of film (i.e. one image) to pass through a film camera, or projector.

Unfortunately, the number of frames used per second varies according to country, norm and usage. For film, the international norm is 24 frames per second (fps). American black and white television uses 30 fps. With the introduction of color television, the frame rate of the NTSC norm, used in America and Canada, had to be reduced to 29.97 fps for technical reasons. In Europe, a lower frame rate of 25 fps was used from the start, and with the introduction of color television this was adopted by the European PAL TV standard, as used in Europe today.

The original reason for the differing film rates, incidentally, derives from the different rates of alternating current used on the different continents (USA: 60Hz, Europe: 50 Hz), which corresponds to the number of half-frames of film passing through a camera/projector per second.

SMPTE/EBU

It was the American Society of Motion Picture and Television Engineers (SMPTE) which first laid down a norm for encoding the individual frames. This designates exactly 80 bits per frame for encoding the hour (0-23), minute, second, and frame (frame number within the second). Some of the surplus bits are used to indicate the frame rate, i.e. the number of frames per second. This encoded data stream of 80 bits per frame is known as SMPTE time code. Because the individual bits themselves have a definite time spacing, they are also used as a further subdivision of a frame, called a “subframe”.

Chapter 16

Synchronization

This code was adopted without alteration by the European Broadcasting Union (EBU), for use with the European frame rates, and renamed “EBU Time Code”. In practice this time code is usually referred to as SMPTE time code, or just SMPTE (pronounced: “simptee”).

Drop Frame (df)

There is a technical problem with the frame rate of 29.97 fps. Only whole frames can be counted, but rounding up to the 30th frame in every second would cause a deviation between the time code time, and the actual time (a difference of about 5.4 seconds for a typical feature film). So, the following trick was employed: in every minute 2 frames are left out (“dropped”), except in minutes that can be divided by 10.

This may sound complicated, but is actually simpler than “one leap-year every 4 years except in years divisible by 100, apart from those also divisible by 400”...

To describe this method, “drop” or “df” (drop frame) is added after the name of the frame rate. Because it is used so often with 30 fps, it is more common to indicate *non-usage* by adding “non drop”, “nd” or “ndf” to avoid confusion.

The 29.97 nd format is seldom used (just like the pointless 30 df), because the time code time “drifts” from “real” time.

Frame Rates

The SMPTE frame rate must be set the same for all connected devices; you can’t mix different frame rates in one system. The diagram below shows the various frame rates, with the respective duration of a frame, or subframe:

Frame Rate [fps]	Frame [ms]	Bit [μs]	Source
24	41.67	520.8	Film
25	40	500	PAL
29.97 drop	33.37	417.1	-
29.97	33.37	417.1	NTSC color

Frame Rate [fps]	Frame [ms]	Bit [μs]	Source
30 drop	33.33	416.7	-
30	33.33	416.7	NTSC S/W

In Europe 25fps is used both in the audio field and for synchronizing television or video productions.

American audio productions typically use 30 nd, but with video the frame rate is nearly always 29.97df.

American Audio Engineers do sometimes label a 29,97 df time code as “30 drop “!

International audio productions are sometimes recorded with 25 fps and 29.97 df on two different tracks of the tape.

LTC/VITC

The 80 bit frames of the SMPTE time code can be stored in two different forms:

As an audio impulse on an audio track. This is called longitudinal time code (LTC).

As a signal in the scanning gap of the video picture. (The complete picture for each frame is transmitted slightly faster than the duration of a frame. There is a short gap, while the electronic beam of the cathode ray tube travels from the end of the bottom line, back to the beginning of the top line. Because the beam is temporarily switched off, time code transferred during this gap does not interfere with the picture.) This is called vertical interval time code (VITC).

LTC is used for all audio productions, and often for video synchronization as well. Tape duplication plants can record LTC onto one of the audio tracks, usually track 2. The SMPTE time can also be written onto the picture. Because of the almost universal use of LTC for audio work, the term “SMPTE” is used synonymously with it.

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With video synchronization, LTC can only be output during playback at normal speed. VITC (pronounced “vitzi”) can be output while fast-forwarding, or rewinding. This is very useful when creating frame-synchronized sound effects, or musical phrases.

Emagic’s networkable 8x8 MIDI interface, the “Unitor8” has a synchronizer that can read and write both LTC and VITC.

MIDI Time Code (MTC)

MTC is a “translation” of the SMPTE bits into MIDI data, and contains the time and frame rate information. This requires one status byte, and eight data bytes. MTC defines only 24, 25, 30 df and 30 ndf.

Synchronizers

A professional synchronizer is a device which can write and read SMPTE signals. It is connected to a master, and one or more slave tape machines, and can also control their transport functions.

With computer MIDI interfaces, built-in synchronizers generally have just a SMPTE input and output. SMPTE signals arriving at the input are automatically passed on to the computer by the interface, in the form of MTC.

Which device should be the Master, and which the Slave?

As a general rule, the slowest machine should be used as the master, so that the slaves are always ready after rewinding or fast-forwarding. Since a sequencer naturally “winds” much faster than even the quickest tape machine, it always acts as the slave.

Synchronization Procedure

The following steps apply to external equipment, rather than Logic.

Recording SMPTE

If you want to synchronize a computer-based sequencer to an analog tape machine, the first thing to do is record a SMPTE signal to one of the outside tracks (the outside tracks on analog tape machines are of a marginally lower quality, because of the (very slight) “fluttering” of the tape, but they are perfectly adequate for time code). Connect the SMPTE out of the interface with the input of the tape machine (or the input of the mixing desk, if you want to route the time code, and set a level). It is customary to use the track with the highest number. Set the level of the time code to -10 VU. Avoid recording stationary time code by starting the time code generator before the recording. For several reasons—for example to avoid drop-outs which can lead to gaps or jumps in the time code, and to give you the option of extending a song later,—it is a good idea to record the time code for the length of the whole tape (this is known as “striping” the tape). It is also customary to set a SMPTE code start time of just under one hour (01:00:00:00). The first song on a tape then always begins at exactly one hour. Other tapes of the same project can be given SMPTE times with consecutive hours, which means the SMPTE time can be used to clearly identify a tape, if the tape boxes get mixed up.

Synchronizing Sequencers to Tape

Connect the output of the time code track to the SMPTE input of the computer. To minimize crosstalk, it is better to make a direct connection rather than routing the signal through the mixing desk. The computer does not have to begin bar 1 at a SMPTE time of 0 hours, 0 minutes, 0 seconds and 0 frames (00:00:00:00), you can set an “offset”, to make the sequencer wait for the correct SMPTE position, before it starts the song. If the second song on tape begins at (say) 01:04:50:00, you need to set the SMPTE offset to match. Make a note of the SMPTE offset on the track sheet for the song, or next to the song title on the tape box.

The bar position which is reached at a specific SMPTE time position depends on the tempo of the sequencer. If you have

begun to record onto tape, you won't be able to alter the tempo without disrupting the synchronization. You should, therefore, also make a note of the precise tempo on the track sheet.

The SMPTE offset and tempo settings are saved with the song, but it is not unheard of for the song file and the tape to be separated.

Synchronizing Several Sequencers

If you want to synchronize several sequencer programs with a tape machine acting as the master, you should try at all costs to synchronize just one sequencer via SMPTE or MTC. You can then synchronize the other sequencers to this one, using MIDI Clock/SPP! When synchronizing several sequencers (with no tape machine) you should Only use MIDI clock/SPP. This avoids deviations in the bar position, since otherwise, each sequencer has to calculate this independently, from the time information using its own tempo track.

If you cannot avoid synchronizing several sequencers using SMPTE, for example because the distances within the system are too great for MIDI signals, you can even out any deviations in the bar position using *Global Tempo Correction* in the **Options > Settings... > Compatibility** page (see page 17-26).

Click Track

It is common practice to record a click track, i.e. a metronome sound, to run throughout any song at the correct tempo, in addition to the time code. This is partly due to possible deviations in the positional calculation of different sequencers from the recorded SMPTE code. It also allows overdubs in studios without MIDI equipment, although nowadays, such studios are rare.

When recording a click track or other signal onto the track next to the SMPTE code, it is better to set a fairly low record level. Crosstalk from a high-level signal onto the SMPTE track can corrupt the SMPTE track, and interfere with smooth synchronization.

Audio Synchronization

The most common problem with external synchronization and digital audio is the problem of having two clock sources: the time code master, and audio wordclock. In every system with several time references, there is room for drift.

In the analog world this well-known problem is held in check because the tape speed is constantly monitored to minimize deviation between the slave machine time code and the master time code.

The same applies with a computer-providing the audio hardware supports continuous variation of the “tape speed” (sample rate), like the Audiowerk 8.

You can “misuse” the digital output on the Audiowerk8 to synchronize other audio hardware (e.g. AudioMedia or ProTools) via the other hardware’s digital input.

Otherwise, you can use an external synchronizer with a wordclock output such as the USD (Universal Slave Driver) for ProTools, or an MTP AV.

If your audio hardware has no external clock or variable sample rate, there are other ways of synchronizing to a time code master described in section *Sync Mode* on page 16-6.

Sync Problems and Solutions

Faulty Digital Synchronization

If Logic is synchronized to external wordclock (*Audio Sync Mode*: External/free), you must ensure that a valid digital signal is always available. If you are getting error messages like “Sample Rate 13,x kHz recognized” it may be that the DAT recorder (or whatever clock source you have connected to your audio hardware’s digital input) does not transmit wordclock in stop or pause mode (or has switched itself off).

Faulty Synchronization to an External Tape Machine

Create a new song, make a new recording and see if that does the trick. Why? If an old recording on tape was not properly synchronized to time code you will not be able to use it. One basic rule: the playback situation must be identical to the recording situation.

If everything is working fine with the new recording, this means the present setup is o.k. Next, check whether anything has changed in your global setup. Has the frame rate changed? Has the tape speed changed? If you have changed a 30 fps setting, try variations such as 30 drop or 29.97.

If MIDI and Audio are not synchronized

Go to the Audio page of the synchronization dialog window and under *Audio Sync Mode*, select *MTC continuous* or *MTC Trigger/ASD*.

If your audio hardware does not support this operating mode, you should cut extremely long regions into shorter sections.

If MIDI and SMPTE are not synchronized

Check all the frame rate settings. The frame rate of all connected devices must be identical, including the time code on the tape machine, the synchronizer, and in Logic itself.

Some synchronizers encode the wrong frame rate in MTC. In this case open the Tempo Editor, switch off the *Detect* option, and set the correct frame rate manually.

Special case: ProTools 4.0 as MTC Master

If you want to use Protools 4.0 as the time code master (sending MTC to Logic via the IAC driver), do the following:

Place the IAC driver file in the OMS folder, within the System folder.

Switch on IAC in the current OMS settings.

In the ProTools software: go to “Setups / Peripherals” and choose a “device” named “Generic MTC Reader”, with “00 IAC Bus #1 as the sync port”.

In the ProTools software select “Set active in Background” from the “Options” menu, and select “Run MIDI In Background” under “Setups / OMS MIDI Setups”.

In Logic, open **OMS Input Connections** under **File > Preferences...** and select “00 IAC Bus #1” as the input connection.

Chapter 18

File Transfer

18.1 Logic Songs to Other Platforms

Logic is currently available for the following types of computers:

- Apple Macintosh (68040 processors)
- Apple Power Macintosh (PowerPC and G3 processors)
- PC-compatible computers running Windows 98
- Atari ST, TT and Falcon (up to version 2.5)

Song files created in Logic on any of these computer platforms are fully cross-platform compatible. Make sure you are using the latest version of the program (see the section *Different Program Versions* on page 18 - 2). Songs created in version 3.5 or higher will not open in previous versions, due to updates/changes in the stereo audio object.

Disk Formats

Before you can exchange song files between different computer platforms, you have to choose a disk format which can be read by both. The ***MS-DOS format*** is compatible with all of the above-named computer platforms.

The “PC Exchange” control panel (Found in MacOS 7.5 and later) allows you to read, write, and format disks in the MS-DOS format. The “Access PC” control panel (made by Insignia Solutions) or the “DOS Mounter” system extension (from Dayna) can also be used. Less convenient (but just as effective) is the “File Conversion” program supplied with older versions of the MacOS. “Mac-PC Manager” (by Software Architects Inc.) can even mount PC formatted hard disks.

Transfer

The transfer process itself is very straightforward:

- Format a disk in the MS-DOS format,
- Save the song onto this disk from the source computer, and
- Load the song from the disk to the destination computer.

Special Features of the Atari ST

When transferring to (and from) the Atari ST, remember to use DD (Double Density) disks, because STs cannot read HD (High Density) disks.

Special Features of the Macintosh

When transferring from the Macintosh to other platforms, make sure that the file name does not exceed the MS-DOS character limit (maximum 8 characters). When transferring to PCs, add the extension “.LSO” to the song file name. When transferring to the Atari, add the extension “.LOG” to the end of the song file name.



Windows 95/98 can easily handle file names of up to 256 characters, but all of the available Mac utilities impose the older Dos 8.3 character file name limitation when copying files to DOS formatted disks.

Different Program Versions

When transferring song files between versions of Logic, the rules are as follows:

Newer program versions can always read song files created in older versions.

Older program versions can only read song files from newer versions if the data has not been altered. This is not guaranteed to work in all cases, due to the constant changes made in new updates.

Version 3.5

For technical reasons, version 3.5 uses a different song format than earlier versions. Songs created in version 3.5 cannot be used by earlier versions.

Songs from earlier versions will be automatically converted when loaded into version 3.5. On the first save command, the “Save as.....” dialog appears, which allows you to save the converted song under a different name. This way, the original pre-version 3.5 song remains unchanged.

Version 2.5

If you have created songs in version 2.0 which contain cable switchers in the Environment, you may find that these switchers turn into meta event faders in version 2.5, and no longer function correctly. This problem is easily fixed by:

- Selecting all the cable switchers in your Environment,
- Giving them the value 48 as the output definition under -1 (the channel number is irrelevant).
- Verifying this change has fixed the problem, by checking to see if the switchers that are set to AutoStyle appear as switchers again in the display.

18.2 Files from Other Programs

To help you to make the transition to Logic from other sequencing programs, and allow you to keep using your old songs, Logic is able to read data from other programs.

If you select **File > Open**, only Logic songs appear in the file selection box. However, if you select **File > Import**, all the files in the current directory are automatically searched to see if they can be read. Readable files are recognized, even if you haven't assigned the DOS extension to the Macintosh document type in “PC Exchange”. In other words, you don't need to worry about specifying any file type.

MicroLogic Songs

This is an easy task, because Micrologic song files use the same file format as Logic. They can be loaded just like any other Logic song file.

Notator SL Songs

Before you import Notator songs, you should make the following preparations in Notator SL:

- Some SL playback parameters (like mutes), cannot be converted by Logic. You should therefore delete any sequences that you don't need.
- Logic cannot directly read Notator SL's tempo events. If your Notator song contains tempo changes, you must first create a sync reference in Notator SL's SMPTE window.

For more information, consult your Notator SL manual.

Transfer

- Format a DD (Double Density) disk in MS-DOS format.
- Save a prepared Notator SL song onto the disk.

Load the SL song into Logic, by selecting **File > Import**.

Depending on the structure of the imported song, Logic sometimes has to re-organize the storage structure, and begin the loading operation from scratch. This is completely normal, and is done to avoid wasting memory.

When the Notator SL song is loaded, it is transformed into a Logic song. The arrange levels "a" to "d" are turned into four folder tracks. Each of the individual folders represents a pattern with 16 tracks. The order of the folders corresponds to the order of the patterns in the Arrange List.

The ports are converted as follows:

A	Modem port 1
E	Modem port 2

F	Modem port 3
G/B	Modem port 4
H/C	Modem port 5
I/D	Modem port 6

18.3 MIDI Files

Standard MIDI Files are not specific to a particular sequencer program, or type of computer. They contain the following information:

- MIDI events with their time positions and channel assignments,
- Names of the individual tracks,
- Names and positions of markers,
- Tempo changes,
- Copyright marks.

Logic supports the importing and exporting of Standard MIDI File formats 0 and 1:

- Format 0 can contain one track,
- Format 1 can contain multiple tracks.

Neither format recognizes any division of a track (e.g. into several sequences).

Loading Standard MIDI Files

To load a Standard MIDI File, select **File > Import**.

If there is already a song in the memory, you will be asked “**Create new environment or copy current environment for MIDI File? New/Copy**”. If you click **Copy**, the existing environment is copied. The tracks of the Standard MIDI File are automatically assigned to suitable instruments.

If there is no song in the memory, or you answer the above question by clicking New, the default environment for new songs is used instead.

Copyright

The copyright mark is read as marker text.

Bank Select

When you open Standard MIDI Files, program change and controller events occurring at the same position are moved by one tick, so that they will remain in their intended order. This prevents Logic from reversing the transmission order of the events. The reason for this is that certain MIDI devices will not respond properly to program change and bank select events that do not occur in the correct order.

This guarantees that there will be no timing problems, because the transmission of a MIDI event always lasts longer than 1 tick.

Saving Standard MIDI Files

If you want to play a Logic song on another sequencer, you can do so by saving it as a Standard MIDI File. Consult the other sequencer's instruction manual to see which Standard MIDI File formats it can read. Any sequencer should be able to interpret at least the type 0 file format.

Preparing the Song

Because of the limitations of the Standard MIDI File format, you should prepare your Logic song in the following way:

- Neutralize all playback parameters with the normalize function (select them all by pressing  , then select **Functions > Sequence Parameter > Normalize Sequence Parameters**),
- Convert all playback quantization with the fix quantize function ( , **Functions > Sequence Parameter > Fix Quantize**,

Convert all aliases into real copies ( , **Functions > Alias > Turn to Real Copy**,

Convert all loops into real copies ( , **Functions > Sequence Parameter > Turn loops to Real Copies**),

Convert all sequences on each track into a continuous sequence ( , **Functions > Merge > Objects per Tracks**).

Answer the “**Convert multiple Event MIDI Channels to Instrument MIDI Channel? No/Convert**” dialog box with  or **Conuert**.

Additional preparations for file format 0

Since Standard MIDI File type 0 format files can only save one sequence, you must also merge all sequences into one ( , **Functions > Merge > Objects**).

Saving a Song as a Standard MIDI File

Select all of the required sequences, and select **File > Export MIDI File...** you can now enter the destination directory.

Remember that most hardware sequencers can only read MS-DOS formatted disks.

In File Format 0

See whether under **Options > Settings > Global...** the parameter ‘Export MIDI File...’ save single Sequences as File Format 0 is checked. If it is, selecting **File > Export MIDI File...** when only one sequence is selected means that file format 0 will be automatically used.

Glossary

32-bit addressing When running Logic this must be switched on in the “Memory” control panel.

Adat Abbreviation for Alesis Digital Audio Tape. The Adat (supported by Alesis, Studer and Fostex) is a digital multitrack cassette recorder with eight audio tracks using an S-VHS video tape, with 16 or 20 Bit quantization. The optical port allows the parallel transmission of all eight audio channels.

Aftertouch Also known as pressure. MIDI data-type generated by pressure on keys after they have been played. There are two types; channel aftertouch, whose value is measured by a sensor stretching along the whole keyboard and whose data apply to the whole MIDI channel, and polyphonic aftertouch (rare) which is individually measured and transmitted for every key.

AIFF Abbreviation for Audio Interchange File Format. Data format for audio files in the Macintosh operating system.

Alias Double of an object which does not contain any data itself but just refers to the data of the original.

Autodrop, Auto Punch Automatic drop-in recording with adjustable drop-in/out positions.

Bar Ruler Ruler at the top edge of the screen, divided into bar units.

BBS Abbreviation for Bulletin Board System. An electronic mailbox.

Cable A virtual cable represents the MIDI connection between two Environment objects.

Catch Function for making the section of the song currently displayed in the window reflect the current song position.

Cha Abbreviation for channel or MIDI channel

Check box A small box. Placing a check in it (by clicking on it) activates an option.

Glossary

Click Metronome, or metronome sound.

Clock Electrical synchronization impulse, transmitted every 1/96 note. Was used in older drum machines before the advent of MIDI. (i.e. MIDI Clock).

Controller, control change MIDI data type, e.g. for sliders, pedals, switches or standard parameters like volume and panning. The type of command is encoded in the first databyte, the value in the second databyte.

Count-in Beats which sound prior to the start of a recording.

C-Press Channel pressure or *aftertouch*.

Cueing Monitoring while fast-forwarding or rewinding.

Cycle Function which constantly repeats the passage between the *Locator-positions*.

Data bytes These define the content of a MIDI message. The first data byte represents the note, or controller number; and the second the velocity, or controller value.

Default, default value The preset parameter value.

Delay In the Environment window, an object that can create a series of repeats. In the Arrange window, a playback parameter which can delay or advance a selected Track by a given number of milliseconds.

Dialog, Dialog box A window containing a query or message. It must be cancelled or replied to before it will disappear and allow you to continue.

DIMM Abbreviation for Double Inline Memory Module. Type of RAM chip.

Display Format Value Adjustable note value for the grid used in displays and operations. Third number in the position indicator.

Drag & Drop Grabbing objects with the mouse, moving them, and releasing them.

Driver In Logic: specialized support for audio hardware. The part of Logic which allows you to address and use this hardware.

Drop, drop-in, drop-out (Punch in/out) Going into and out of record to record over a section of an existing recording.

Edit Local menu with clipboard functions.

Editor Window for editing MIDI Events.

EQ, Equalizer Used to boost or cut frequencies within a sound spectrum. There are several types available in Logic.

Erase Delete

Event definition Parameter for defining the display of a line in the *Hyper Editor*.

Windowclass Status of the window as a *float window* or a normal window. Float windows are not hidden by normal windows in the same program.

File In Logic the two most important types of file are: 1. Song files containing all the MIDI events and parameter settings in a song (including mixer automation data) plus information about which audio files are to be played; 2. Audio files containing the actual recordings of guitar, vocals, etc. (not actually stored in the song files).

Flip menu *See Pull-down menu.*

Float window Window with special status which always “floats” on the surface above all the other windows, but can only be operated with the mouse.

Folder A “song within a song”. This can contain either a complete arrangement or just parts of it: sequences or other folders.

Font Character printing style.

Frame Unit of time. A second in the *SMPTE* standard is divided into frames, corresponding to the frames in a film or video.

GM (General MIDI) Standard for MIDI sound modules, including standardized instrument sounds on the 128 program numbers, a standardized key assignment for drum and percussion sounds on MIDI channel 10, 16-part multi-timbral and at least 24-voice polyphony.

Grab (an object) Placing the mouse pointer on the object, pressing the mouse button and keeping it held down.

GS Extended *GM* standard developed by Roland Inc.

Glossary

Hierarchical menus Structured menus where highlighting an individual entry opens yet another menu.

Hyper set All simultaneously displayed event definitions in the Hyper Editor.

Icon Small graphic symbol. In Logic an icon may be assigned to Environment objects.

Info Line Display at the top of the window which tells you the position of objects when using the mouse tools.

Insert Point on a mixer where you can patch in an effect (Plug In). The audio channels and bus objects on Logic's adaptive and audio mixers have inserts for effects.

Instrument Logic's virtual counterpart to a real sound source or synth module.

Key command Function which can be executed by pressing a specific key.

Legato Method of musical performance that smoothly connects one note to the next.

Local menu Menu in a window containing functions which are relevant to only that particular window.

Local Off Operating mode on a MIDI synthesizer with a keyboard where the keyboard (local) does not directly play its own integrated sound generator. This is useful when using it as a master keyboard in a MIDI setup with a sequencer.

Locators (left and right) Two programmable song positions which set the cycle limits in the Transport window. The Locators also can be used to define the area to be edited for certain functions.

Loop Constant repetition of an object, up to the next object in the same track, or to the end of the current folder, or song (whichever comes first).

Merge Mix, combine together.

Meta event Type of event in the Event List. Contains non-MIDI events that can control special Logic features, like text or screenset configurations.

MIDI (Musical Instrument Digital Interface) Standardized, asynchronized, serial and event-oriented interface for electronic musical instruments.

MIDI Clock Short MIDI message for clock signals. See also: SPP.

MIDI Event Individual MIDI command, such as a note on command. Continuous controller movements (e.g. modulation wheel) produce a quick succession of individual events with absolute values.

MIDI Message A message transmitted via MIDI consisting of 1 status byte and none, one, two or (with system exclusive commands) many data bytes. See MIDI Event.

MIDI Multi Mode Multi-timbral operating mode on a MIDI sound module in which different sounds can be controlled polyphonically on different MIDI channels. A multi mode sound module behaves like several polyphonic sound modules. General MIDI prescribes a 16-part multimode, (i.e. the ability to control 16 different parts individually). Most modern sound generators support multi mode. In Logic, multi mode sound modules are addressed via multi instruments.

MIDI System Exclusive Message **Also SysEx** System exclusive data forms the top tier in the hierarchy of MIDI commands. These messages are tagged with an identification number for each manufacturer (the Sys Ex Manufacturer's ID number). The actual contents of these MIDI commands is up to the manufacturer. They are used for transferring whole sound programs and/or system settings, and for addressing individual parameters used in sound generation or signal processing. Editor software (such as Emagic's SoundDiver) transmits and receives MIDI system exclusive messages from MIDI devices, allowing you to program these MIDI devices on your computer.

Modifier (Special keys) Computer keyboard keys used together with "normal" keys to change their function: .

Modulation Generally, a slight, continuously varying pitch change. The MIDI standard is controller no. 1, transmitted on keyboards by the non-centered control wheel (or by moving the joystick vertically).

Glossary

Moving *Grabbing* an object, moving it with the mouse button held down and releasing it at the target position.

MTC (MIDI Time Code) “Translation” of a *SMPTE* signal into the MIDI Standard.

Mute Switch off (a sound or track).

Normalize 1) This function applies the settings of the current *playback parameters* to the selected events (by altering the actual events themselves), and clears *the playback parameter* settings.

2) (Audio): This function raises the volume of a recorded audio file to the maximum digital level without altering the dynamic content.

Note number (Note #) Pitch of a MIDI note, controlled by the first data byte of a MIDI note event.

Object In the Arrange window, a general term for sequences or folders. In the Environment, it also applies to instruments, faders, mixers, keyboards, monitors, Audio Objects, etc.

Option Alternative function, often in the form of a *checkbox*, sometimes also as a menu entry to be ticked.

Parameter box Field on the left side of the screen where you can adjust the parameters of the selected object.

Paste Add. The command   adds the contents of the clipboard-i.e. whatever you copied with   or cut with  —to the position of the cursor, or song position line.

Pitch Bend Message MIDI message transmitted by a keyboard’s pitch bend wheel.

Pitch Bend Wheel Hand wheel for generating pitch bend messages.

Pixel One matrix dot on the computer screen. Short for “picture element.”

Play parameters The parameters for quantization, transposition, velocity, compression and delay which do not alter the stored data but merely affect how the events are played back.

Position indicator Units: bars, beats, sub-divisions (often written simply as *divisions* in this manual) and ticks.

Post Fader Positioned after the fader in the signal flow, i.e. the level of a signal routed post fader to an auxiliary changes when

the fader is moved. Post fader aux sends are normally used for effects (such as reverb) so that the reverb level changes with the channel level, and the ratio of original and effect signal remains constant

P-Press Polyphonic key pressure, also known as polyphonic aftertouch. Rare and very data-intensive type of command, measuring the pressure exerted on each individual key. Very few keyboards have sensors for poly pressure.

Pre Fader Positioned before the fader in the signal flow, i.e. the level of a signal routed pre fader to an auxiliary does not change when the fader is moved. Pre fader aux sends are normally used for monitor mixes, so that the mix on stage or in the studio headphones does not change when the levels are altered in the control room.

Preferences Storable settings for a program.

Pull-down menu Selection menu which opens when you grab the parameter input box. Occasionally referred to in this manual as a flip menu.

Punch, punch-in, punch-out (Drop in/out) Going into and out of record to record over a section of an existing recording.

Quantization Time-correction of note positions by moving them to the nearest point on a selectable grid.

Region Chosen area of an audio file which is registered in the Audio window for use in the song and, can be placed in the Arrange window on the audio tracks, just like a sequence can be placed on the MIDI tracks. The region could be the whole length of the audio file, or any portion thereof. A region is created automatically after every audio recording.

Replace Operating mode where a new recording deletes the previous one (like a tape machine).

Rubber band Technique for selecting items by click-holding the mouse, and lassoing it over a group of adjacent items.

Scroll bars and scroll box Gray beam at the edge of a window with a movable box inside it for adjusting the section of a song displayed in the window.

Glossary

Screenset Storage location for the layout of the various Logic windows. Each Logic song may store up to 90 Screensets.

Scrubbing On tape machines: manually moving the tape across the tape head. In a sequencer: manually moving the song position line through the song, triggering playback of the MIDI events.

Selecting Choosing. Selected objects are displayed in inverted colors.

Sends Abbreviation for auxiliary sends (used for effect or monitor sends).

Sequence A collection of MIDI events which is shown in the Arrange window as a horizontal beam with a name on it.

SMPTE Standard for time coding in a specialized sound signal used for synchronizing different devices. There are six formats. Acronym for the Society of Motion Picture and Television Engineers. Specifies the “counter position” of a “SMPTE clock”. Composed of hours, minutes, seconds and frames. The MIDI variation of SMPTE is MIDI Time Code (MTC).

Song Position Line Vertical line in the Arrange, Matrix and Score windows which indicates the current song position. It can be grabbed with the mouse and moved (see scrubbing).

Sound source General term for MIDI-controllable synthesizer, sampler, drum machine, digital piano, etc.

SPP, song position pointer A specific type of MIDI message which gives the distance from the song start in 1/16 notes, and is transmitted along with MIDI clock.

SMF (Standard MIDI File) Standard file format for exchanging songs between different sequencers or MIDI file players.

Status byte First byte in a MIDI message, which determines the type of message.

Subframe A sub-division of a SMPTE *frame*, corresponding to the individual bits of a SMPTE frame. One frame consists of 80 bits.

Swing Parameter which alters the rigid timing of a quantization grid by delaying every other note of a specified *sub-division* by a definable amount.

Glossary

Synchronization Method for keeping several recording/play-back devices locked together timing-wise.

Synchronizer Unit for centrally controlling the synchronization of several devices.

Tick, plural = Ticks The smallest unit of timing resolution in a sequencer. In Logic this is 1/3480 note.

Timing Measure of the ability to play notes at the right time.

Toggle Switches backwards and forwards between two states (windows, parameter values, etc.)

Track column Situated to the left of the working area of the Arrange window. Displays the instruments assigned to various tracks.

Track Mixer Adaptive Mixer which automatically configures itself to show every audio and MIDI track, in the order that they appear in the Arrange window or in an open folder. If you move the controls on the adaptive mixer while recording, automation data is stored in the relevant tracks in the form of normal MIDI Controllers.

Track object Environment object defined in the track column of the Arrange window to which the data of the track are routed.

Undo Function which reverses the previous operation.

Update 1. New version of a program. Often arrives a month late.
2. Mixer automation mode in Logic, where the old automation data is replaced by new recordings of existing controllers.

Velocity Force with which a MIDI note is struck; controlled by the second data byte of a note event.

View Local menu with display options.

Virtual memory Area of the hard disk which can be used by the Mac as an extension of the RAM memory. The disadvantage in using it is its very slow access time. When running Logic it must be switched off; this is done from the "Memory" control panel.

WAVFile Audio file format in the Windows 92/98 operating system. Uses the file extension, ".WAV".

Work area The area of a window in which you edit objects (folders, sequences, events, environment objects).

Glossary

WYSIWYG “What You See Is What You Get”. The ability of a program to accurately display the forthcoming printout on the screen, e.g. Page View mode in Logic’s Score window.

XG Extended General MIDI standard from Yamaha, compatible with Roland GS.

Zero crossing A point in an audio file where the waveform crosses the zero amplitude axis. If you cut an audio file at a zero crossing there will be no click at the cut point.

Zoom function Shrinking or enlarging the display in Logic windows.

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