Team TnD Presents:

The Logic Audio Platinum v5.0 Manual

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Team TnD has cleaned up, fixed and resized the original scans to make this manual more easy to read.

We've also converted it to an easy to access PDF format.

The original scans were over 70mb in size. The TnD version is much, much less.

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Team TND

emagic

logic|5

Music Production Software

>> Version 5.0, January 2002

>> English Edition

Reference Addendum

Mac

Microsoft Windows Compatible

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Chapter 1

Welcome

Thank you, and congratulations on, the purchase of Logic 5.0. Version "Five" is the culmination of our greatest efforts in Logic's successful history. It involved a great deal of work which, from a musicians point of view, should prove to be very satisfying and useful. This manual introduces all of the new functions and many improvements made since the release of Logic 4.0. As a Logic 4.x user, you may already be familiar with many of the features introduced in recent updates. That said, a little "study" of these features and enhancements will aid in accelerating your work and in saving you time. Many of the new facilities are found in context-sensitive, hierarchical menus which you may have overlooked in the releases between versions 4.0 and 5.

The most sensational innovation; the ability to remotely control Logic with the Logic Control and Logic Control XT expansion module, is documented in the dedicated Logic Control/Control XT manual. The workflow and "feel" of mixing music productions is so seamlessly supported, that the "desire"—in terms of physical size—for a big studio console, will walk out the door. As mentioned, this addendum doesn't deal with the features directly associated with the Logic Control. We wish you countless hours of inspiration, fun and success with Logic 5.0!

Your Emagic Team

Main Features of Version 5.0

Before getting into the details, the following outlines the primary new features of Logic Platinum 5. In addition to these, there are countless other small improvements.

New Track Automation System

Independent of Logic's record-ready, record or play-status, every track can be fully automated. The new *Track Automation* system inherits all of the advantages of Logic's former automation concepts and features many new enhancements:

- 32 Bit resolution of Track Automation data;
- Sample-accurate processing of Track Automation data;
- Names of automated parameters are shown in clear text;
- Several automation modes, allowing intelligent management of overdubs and additional mix automation recording(s).

Improved HyperDraw

HyperDraw has been significantly improved as a legacy of the new Track Automation system.

- All, or parts of, the automation data can be moved and copied.
- Automation data nodes can be moved and copied. The shape of the curve between the nodes can be interpolated from linear to concave, convex or S-shaped. This flexibility eliminates the need for a large number of nodes.

Three Built in Synthesizers

- Logic features three different subtractive virtual analog synthesizers: The monophonic **ES M** (**ES Mono**) bass synthesizer, and polyphonic **ES E** (**ES Ensemble**) and **ES P** (**ES Poly**) synthesizers.

New Plug-In-Effects

- *SubBass*—generates synthetic spectral components in the sub bass range, most similar to an extensively-featured "boom box".
- *Exciter*—generates synthetic spectral components in the treble range.
- *De-Esser*—eliminates hiss noises from vocal takes. Given its extensive range of parameter controls, it can be used for frequency-specific compression.
- *Phase Distortion*—generates unpredictable spectral components.
- *Clip Distortion*—introduces drastic distortions.
- *Denoiser*—eliminates noise (FFT-based single-ended noise reduction).
- *Tremolo*—vintage-style amplitude modulation.
- *Stereo Spread*—enhances the stereo spectrum.
- *Limiter*—universal tool for limiting levels and for increasing volume.
- *Adaptive Limiter*—delivers maximum volume increase for single voice recordings.
- *Multiband Compressor*—the most powerful tool for increasing volume and improving the sound of complete mixes, especially in a mastering context.

Audio

- All Logic-versions support 24 Bit recordings and sample rates of up to 96kHz: MicroLogic AV, Logic, Logic Gold and Logic Platinum.
- The Audio Mixer features side chains, allowing the use of a compressor as a ducker (Gain reduction dependent on the level of another track), for example. Ducking instrumental

playback by sidechaining the lead vocals is a common practice in pop music mixdowns.

- REX2 file support. Logic can exchange data with the *ReCycle* software by *Propellerhead Software*.
- Logic allows audio scrubbing with any audio hardware.
- OMFI File Compatibility (Open Media Framework Interchange) allows for the interchange of AVID, ProTools and Logic sessions.
- OpenTL (Open Track List) format compatibility for data exchange with Tascam hard disk recorders.
- POW-r Dithering during "bounce".

Score

- Multi Page View for simultaneous display of multiple pages with realtime adaption to zoom level.
- Step Time Input window for extremely fast note input via computer keyboard ("note typewriter").
- Notation of aliases and loops.
- Colored notes.

Logic Control—Hardware Controller Support

- Seamless integration of the Emagic Logic Control/Logic Control XT, jointly developed with Mackie Designs. The features of the Logic Control units are documented in a separate manual. The Logic Control features motorized faders and gives you much more than the physical feel of a hardware mixing desk and hardware autolocator. Convenience and working flow are dramatically improved by this unique controller.

New Interface Languages

- New interface languages: Italian, French, Dutch, Spanish, Norwegian and Japanese—the latter should *only* be used on a Japanese-localized operating system, with corresponding fonts. To switch beween languages, select **Options > Settings > Display Preferences**, choose the language, quit and restart Logic. Default Language is the language of your computer's keyboard.

Chapter 2

Operating Systems and Processors

2.1 CPU Optimization

Audio > Audio Preferences > Processor Type ensures that Logic's Audio Engine always uses code optimizations specific to the processor running on your computer. It should always be switched *on*. The optimization can be switched off, but was only included for (extremely rare) compability reasons.

Logic features special optimizations for these processor types:

- Pentium III
- Pentium4
- Athlon
- AltiVec (Apple Macintosh G4)—not switchable

2.2 Mac OS X: Development in Progress

Mac

Logic does not currently run on Mac OS X. We strongly advise that Logic Macintosh users do not attempt to run Logic on Mac OS X.

Emagic are currently developing a Mac OS X-native version of Logic, and will advise users of availability and further details on our website.

Logic 5 is fully compatible with Mac OS 9.1.

2.3 USB MIDI on Apple Macintosh Computers

Mac

The Logic 5 updater/installation program automatically adds the latest "USB Unitor Family Driver" version. This new driver is essential for the Unitor-USB fix to work, and must be located inside the Extension Folder within your System Folder.

2.4 Apple Macintosh G4

Mac

Logic makes efficient use of the processing power of the Apple Macintosh G4 through utilization of the *Velocity Engine*. Logic's internal, native audio engine is able to take advantage of multi processor Apple Macs. (Mac AV, Audiowerk, 1212I/O, DirectI/O, ASIO, StudI/O, EASI). To enable multi processor Macintoshs, check the **Multi Processor Support** check-box in the **Audio > Audio Drivers** menu option.

Chapter 3

Automation

3.1 Automation Concept

"Automation" is the ability of a mixing desk to record, edit and play-back the movements of its volume faders. Fully automated consoles not only record the motion of the volume faders, but the motion of *all* knobs and switches, including pan, EQ and aux send controls.

All of Logic's mix functions can be automated, without restriction. This also applies to all Logic effect plug-ins—i.e. the parameters of all effects and Audio Instruments. This is also true for VST plug-ins which support automation according to the VST standard.

As Logic 5.0 supports the Logic Control hardware, the dream of simultaneous control over multiple parameters becomes a reality. Needless to say, adjusting multiple mix or effect parameters concurrently is impossible with a mouse.

If you've worked with versions prior to version 5.0, please note that the automation concepts have changed.

From version 5.0, the concept behind Logic's automation system has changed. The former automation strategies explained in the reference manual are retained mainly to ensure complete compatibility with existing songs, and to maintain the option to use this older working style, if desired.

The new Track Automation system combines all of the advantages of the former strategies and features many additional "pros". From the release of version 5, we encourage you to adopt the working methods of this new Track Automation system.

Please note that due to the changes introduced by the new Track Automation system, a new Song file format has been introduced with Version 5. Please consult the *New Song File Format* section, on page 45, for further information.

In Place of MIDI Controllers in Sequences and Regions…

If you liked to draw level and pan controller streams with control change messages 7 (MIDI volume) and 10 (pan) in HyperDraw (or the Hyper Editor), you'll be glad to hear that the new automation system is based on a much improved version of HyperDraw. The automation data generated by the new system, however, is stored in a completely new area which does not interfere (and has nothing to do) with the MIDI data. The new Track Automation data is assigned to *tracks*, not to individual sequences or regions. This is visibly evidenced by the gray shaded area (which you draw the automation data curves on, using the track based HyperDraw) that covers the whole track, rather than individual sequences or regions. This makes it easier to place volume fader information right *before* a sequence or region, for example. The familiar object based HyperDraw techniques remain, but now utilize a highly improved HyperDraw interface.

In Place of Channel Splitter Tracks…

Nothing. It's time to say "goodbye" to the channel splitter objects in the Environment. Formerly, these were defined, and used, as track objects, carrying the MIDI controller data for up to 16 MIDI channels (i.e. 16 tracks). They were usually called "A-Playback", "B-Playback" or "GM Mixer Modem" in former default songs.

The Automation of the Adaptive Track Mixer

The adaptive Track Mixer's automation system has always allowed the recording of MIDI sequences (containing MIDI continuous controller automation data) by engaging Logic's record mode and moving the onscreen knobs and faders. This

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technique remains—in all channel strips set to MIDI. This also applies to the Audio Mixer as well.

When—and *only* when—a channel strip is set to *MIDI* and Logic is in record mode, are the Track Mixer's knob movements recorded as new MIDI sequences directly on the selected track. In this scenario, no new 32 Bit Track Automation data is generated. You should only use this technique if you wish to have, and later use, such data in dedicated sequences, which can easily be moved, copied or merged with other sequences.

The new Track Automation

When it comes to mixing down, the new Track Automation system is far superior to the three techniques mentioned above. Each of these automation techniques has its specific benefits, and each technique allows the editing of automation data in any suitable MIDI editor window. The catch-*22* is that the variety of automation techniques on offer wasn't consistent or self-evident in use.

With version 5.0, the automation system has been simplified and standardized. The new Track Automation combines all advantages of the former techniques. You can record Mixer movements and edit them with HyperDraw. HyperDraw is now simpler to use and much more flexible. HyperDraw allows for convex, concave and S-shaped curves between the nodes, eliminating the need for a huge number of nodes. The editors you are familiar with are still available for the editing of automation data.

Track Automation

The new Track Automation system, introduced with version 5.0, is no longer attached to sequences and regions, but to tracks. The automation data is stored in a—normally invisible—folder which is independent of all MIDI data. The resolution of Track Automation data is 32 Bit, which is far more precise than that of MIDI.

This is an important difference between the former, and new, automation concepts: The Track Automation system is based on a new type of data, which resembles MIDI control change messages, but is completelely unrelated.

Recording Track Automation Data

You can record Track Automation data in real time by moving the Mixer's controls (the faders or the pan pots, for example), while their objects (channel strips) are set to the Touch, Latch or Write modes. In real world usage, you'll rarely (if ever) use the rather destructive Write mode, which erases all automation data. The normal "writing" modes are Touch and Latch. These modes, explained in detail from page *26* onwards, are set for every track object (and related channel strip) individually.

Important: The recording of Track Automation data takes places no matter whether Logic's transport is, or is not, in record or playback mode! In addition, the Arrange window track selection or audio record ready status is irrelevant. Whatever you touch *will* be recorded when a channel strip is set to one of the write modes. The movement of Mixer controls (when in a write mode) can be used to overwrite and/or edit existing automation data in realtime.

Behavior in Stop Mode

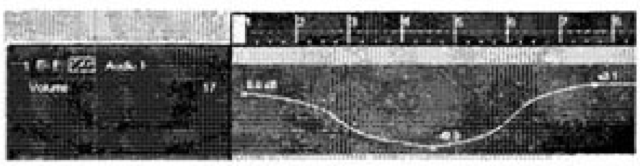
In Stop mode (Playback halted by the Stop button in the Transport), Logic ignores the automation modes and won't write any automation data, even if you move any of the Mixer controls. There is, however, one exception: As long as no automation data of the selected controller type (volume, pan etc.) has been recorded, the corresponding control setting (fader, pan pot etc.) will affect the whole song, just as with a mixing desk. This is the default behavior for every mix parameter when you start a new song.

In other words: In a scenario where you're working on a new song, which contains no automation data and the playback is stopped; The setting of any control moved in the Mixer will affect the whole song. This setting will be overwritten as soon as you start creating automation data (during playback in Touch or Latch mode, or graphically, in

Stop mode, with track based HyperDraw)—This is, in effect, what the automation is for: moving the controls automatically during the song's playback. Touching a control in Stop mode won't change existing automation data, not even in Touch or Latch mode.

Graphical Editing of Track Automation Data

You can write and edit automation data graphically and numerically. The graphical editing of automation data is performed by utilizing *Track Hyper Draw*, introduced in Logic 5.0. As mentioned earlier, it's also possible to draw MIDI data into sequences and regions with the—highly improved—HyperDraw. You could still automate your mix this way, but this is not recommended in most situations.



*Yellow Track Hyper Draw Curve of Audio-Object 1's "***Volume***" parameter. Alongside the nodes, numerical expressions in dB can be seen. Although no region exists in the displayed area, the volume fader of the Audio 1 channel strip will move when playing bars 1 to 8. That is, as long as its automation mode isn't set to "Off" or "Write". The graphic is a screenshot of the PC-version. The S-shape has been achieved by pulling the line to the right with the Automation Tool (Curve setting).*

To display Track Automation data in the Arrange window, select **View > Track Automation**. This is also available as a Key Command. As long as this function is unchecked, sequences and regions appear as per usual. The gray, shaded Track Automation area is only displayed when the vertical zoom level is set to a sufficient height, as is the case with conventional object based HyperDraw. As opposed to the conventional HyperDraw function, engaging the Track Automation data display will automatically set a suitable zoom level. The Track Automation data is displayed on a transparent grey shadowed area, allowing you to see the audio waveform in regions and notes in sequences at a reduced contrast level.

You can edit the Track Automation data directly in this display.

Each track can be displayed at an individual vertical zoom level: Click-hold on the very left lower edge (to the left of the track number) of the desired track in the Track List and drag downwards.

Zooming is performed with the telescope (Mac OS), or the zoom bars (see picture) of the PC version. Alternatively, you can also use the magnifying glass of the tool box.

Moving Automation Data When Moving Regions And Sequences

It's possible to move sequences and regions with, or independent from, the Track Automation data. Switch the modes by selecting **Options > Track Automation > Automation Settings > Move Automation with Objects**. You can choose between Never, Always and Ask, which should be self-explanatory. Ask is the default setting.

It is also possible to move a node, along with all following nodes, by holding ⌥ (Mac) or ctrl (PC) while dragging.

Choosing the Parameter To Be Displayed

In the flip menu of the *top* panel which appears in the Arrange window Track List, you can select the parameter which you wish to display and edit. This can be the volume fader, the pan pot, or any other parameter of the corresponding Mixer channel strip. The parameters are represented by different preset colors when displayed as a curve in the automation track. When you edit a parameter in the Track Mixer, a plug-in-window, or via a Logic Control unit, the most recently edited parameter will be displayed.

Tip: As a quick note on the handling of the "Pan" parameter in Logic:
1. Mono Tracks: If the track is mono, the Pan parameter really is a panorama which allows the panning of a mono track from left to right.
2. Stereo Tracks: If the track is stereo, the Pan parameter works as a balance control between the left and right channels. It reduces the

volume of the left/right channel as it is turned further to the other channel.

Viewing Multiple Automation Parameters

You will commonly automate Volume, Pan and Mute parameters for most tracks. As mentioned above, you can select the desired automation parameter via the top flip menu panel of each track.

To the lower left of each track in the Track List, you will see a small arrowhead, pointing to the right. If this is clicked once, an additional sub-track will become visible.

Logic automatically sets the automation parameter type to a type already recorded, but not currently shown, if applicable.

You will note that this sub-track doesn't feature the track name or the automation *Mode* panel. This is because the "parent" track is globally responsible for these sub-tracks.

Each sub-track can have a different automation parameter set in the flip menu, which allows you to view multiple automation data types for the parent audio or MIDI track.

In the lower left of the sub-track, a further right-pointing arrow is available. This allows you to create additional sub-tracks for other automation types. A maximum of 40 such sub-tracks, associated with a particular parent track, can be created in this fashion.

⌥ (Mac) (ctrl on Windows) clicking on the closed triangle will add as many tracks as needed to show all types of previously recorded automation data. To avoid the unnecessary creation of too many tracks, up to ten (existing) parameter tracks are created with each click. You can, of course, click a second time to add more automation tracks.

By pressing ⌥⇧ (Mac) (ctrl⇧ on Windows) and clicking, up to 40 tracks will be created with a single click.

To close individual sub-tracks, simply click on the arrowhead (which now points downwards) on the desired sub-track or parent track.

ℹ [⌥] (Mac) ([ctrl] on Windows) clicking on the open triangle will close all sub-tracks created for this multi-automation view. Please note that *only the view* is closed, *not* the automation data deleted. The automation data remains active and can be viewed again by [⌥] (Mac) ([ctrl] on Windows) clicking on the closed triangle a second time.

Note that when all sub-tracks have been "collapsed", their automation data is still visible in the Automation Track of the parent track, provided that the Zoom level is sufficient. These appear in a different color, and at a reduced contrast level.

Platinum only: To increase the visibility of the automation data of these sub-tracks, open the **Options > Settings > Display Preferences.** In this window, use the mouse as a slider on the panel alongside the **Other Data Transparency in Arrange** entry (default: 25%). A value of around 50 or 60% will provide great contrast for the sub-track automation data. You can also change the **Object Transparency in Arrange** option to further enhance visibility (default: 33%).

Summary: To Create And Edit Track Automation Data

Follow these instructions to get a "feel" for what it's all about:

- Press an appropriate numerical key in order to open a Screenset with the environments Audio Mixer, or adaptive Track Mixeropened. Alternately, just open one of these Mixers by selecting **Audio > Audio Mixer** or **Windows > Track Mixer**.
- Just above the pan pot of each track you'll see a small parameter window with the word Off in it. Click-hold in this panel and select Write here, in one (or more) of the channel strips. Note: This must be set independently for every channel strip (for every track).

- Start Playback. Remember: Recording Track Automation data does not require Logic to be in record mode!
- Move some of the channel strip controls ... say, the volume fader and pan pot.
- Press the appropriate numerical key in order to open a Screenset which contains an Arrange window.
- Engage **View > Track Automation**. Logic automatically zooms in vertically, allowing the data to be seen. To zoom in further, increase the vertical zoom with the telescope (Mac OS) or zoom bars (Windows version) at the lower right edge of the Arrange window. Alternately, drag down the left-most lower edge of the Track List (to the left of the track's number) entry with the mouse, in order to zoom in the track vertically.
- Beneath the name of the track in the Track List, a parameter panel appears, showing the displayed parameter's name—or Display Off. Click-hold on this panel and select Volume. The word appears in bold letters, in cases where such data (volume) already exists on the track. Given that you just messed around with the volume fader during playback, and that the track was set to Write mode, volume data should indeed exist.
- The recorded volume automation data stream appears as a yellow curve. Pan data is displayed in green.
- You may want to increase the horizontal zoom as well. Touch the nodes of the automation data curve and move them. See how touching a line between two nodes feels (when moving it), and see what happens when you click on the gray background of the automation track (i.e. not on a line or node).
- Change the curve between two nodes with the Automation Tool using its curve setting by touching the line between them. Move the mouse vertically and horizontically to get a feel for how you define convex, concave and both types of S-shaped curves. This ability saves on the creation and editing

of a lot of nodes (alternatively: use the standard Arrow Tool with ctrl alt held).

Please note that all parameters of the Mixer's channel strip (of the audio object …) are fully automated—including equalizer and bus send controls. Data which you create and edit here is independent from the sequences and regions on the track. In other words, this automation data will remain, even if you delete the regions and sequences!

Please don't confuse the Track Automation data display with that of the *object based HyperDraw*, which offers the same new interface, including the variety of curve types between the nodes. *object based HyperDraw* offers the editing of Control Change Messages 7 (MIDI Volume) and 10 (Panorama). It can be used to edit recordings made with a channel strip in MIDI mode.
Tip: Avoid the use of this functionality and try to use the Track Automation instead, particularly if you're not a "power user".

Context Sensitive Display

In order to reduce the number of parameters displayed, the display is context sensitive, which greatly simplifies operation. What this means is that only the parameters which actually control something in the selected Mixer channel strip (or in its plug-ins), will be displayed. The display of the parameter names is also context sensitive: this is a major plus, because rather than seeing an anonymous number, you can see the name of the function you're dealing with in plain text. Parameters which already exist in the track are displayed in bold lettering in the context-sensitive menu.

3.2 Automation Modes

In every channel strip of the Track Mixer or Audio Mixer, you can select the automation modes individually. At higher vertical zoom levels, the automation mode parameter is also visible in the Track List of the Arrange window. Given that Track Automation data can be recorded during playback mode, the default setting is Off, as any mix automation may prove disconcerting while arranging. You can choose between the following automation modes in any MIDI or audio (including Audio Instrument, bus and output) object:

Off

Off will disable the current Track Automation data without deleting it. No automation data will be written, nor read and played back. If the current automation mode is Off, any edits to Track Automation data in the Arrange window will automatically switch the automation mode to Read. This ensures that the data, as currently edited, will be played.

Read

Read will automate the current track using the existing automation data. The data cannot be changed in realtime by touching the fader.

Touch

Touch will automate the current track in the same fashion as Read. Should the fader be touched, the existing Track Automation data of the current fader type will be replaced by new movements—for as long as the fader is pressed. Touch is the most appropriate, "standard" mode, used for designing the mix. With automation engaged, it provides you with the option of correcting and improving the mix at any time. The time required by a parameter to return to its previously recorded setting, is selected via **Options > Track Automation > Automation Settings... > Automation Ramp Time (ms)** (see page 28).

Latch

Latch basically works like Touch, but after releasing the fader the current value will replace any existing automation data—for as long as the sequencer is in playback (or record) mode. Press stop to finish or to define the end of your parameter editing by stopping playback (or recording).

Write

In Write Mode, existing Track Automation data is erased according to the **Track Automation Settings** as the Song Position

Line passes it: **Options > Track Automation > Track Automation Settings > "Write" mode erases.** If you move any of the Mixer's controls, this movement will be recorded—if you don't, according to the **Track Automation Settings** existing data passed by the SPL is simply deleted: **Options > Track Automation > Track Automation Settings > "Write" mode erases.** So be careful to ensure that you don't erase your pan, bus, and EQ automation by mistake, if your intention was to only erase some volume fader information! After stopping the transport, all tracks set to Write will automatically be reset according to the **Track Automation Settings: Options > Track Automation > Track Automation Settings > "Write" mode changes to.** This is to protect your remaining data from accidental erasure.

The Write mode of traditional mix automation will rarely be needed when working with Logic's Track Automation. It's mainly there to complete the selection of automation modes. It's easier to erase the automation data by selecting **Options > Track Automation > Delete All Automation Data of Current Track** (or **... All Tracks**, respectively). In earlier analog mix automation systems, the Write mode was the only way to completely erase automation data.

MIDI

MIDI disconnects the Mixer controls from the Track Automation. Conventional MIDI control change message data will be recorded into sequences—as per the automation used by Logics Track Mixer in versions prior to 5.0. The fader will act like a standard external MIDI source and will be recorded and played back as normal MIDI data in MIDI sequences.

Automation Ramp Time (ms)

Under **Options > Track Automation > Automation Settings...** you'll find the **Automation Ramp Time (ms)** parameter. This is only used by the Touch automation mode. In this mode, the value of an edited parameter will return to it's former setting—during the time span (in milliseconds) defined by this option.

Switching Automation Modes For All Channels Simultaneously

Hold [⌥] (Mac OS) or [ctrl] (PC) when switching the mode of a channel strip, and all channel strips which had previously been set to the same automation mode, will switch as well.

Switching Track Automation Modes By Key Command

Any track's automation mode can be set by a Key Command defined via the Key Commands window: *Toggle Current Track Automation Touch/Read, Latch/Read, Write/Read, MIDI/Read.* When the desired mode is set, the Key Command toggles between the set mode and Read. There are also Key Commands to set all tracks to one automation mode: *Set All Tracks to Automation* … (Off, Read, Touch, Latch, Write MIDI).

3.3 Conversion of MIDI/Track Automation Data

There are some situations where conventional (the old style) automation data in sequences is more appropriate than Track Automation data. As an example: When mixing down, this method should be the exception. We ask that you take into account the possibilities of performing fade-ins, fade-outs and cross-fades with audio regions non-destructively with the cross-fade tool.

To convert data recorded in MIDI automation mode, drawn in object based HyperDraw, or data from songs saved by older Logic versions into Track Automation data, two convert functions are available. Access these via **Options > Track Automation >**.

With these functions you can convert the mix information of songs (saved in pre-Logic 5 versions) into the new Track Automation format. Select all sequences and regions in the Arrange window ([⌘][A] Mac OS (Windows [ctrl][A]), and choose **Options > Track Automation > Move All Object Control Data To Track Automation**.

Please note that all following functions require that at least one object be selected *before* their use.

Move Current Object Data to Track Automation

This function converts the MIDI control change message type currently displayed by object based HyperDraw into Track Automation data.

Move Object Control Data To Track Automation

This function converts all MIDI control change message types into Track Automation data, including those which aren't currently displayed by HyperDraw.

The "move" options also include the ability to move Track Automation data to objects. This is the reverse of the functions above.

Move Current Track Automation Data to Object

This function converts the currently displayed Track Automation type, and moves it into a selected object.

Move All Track Automation Data To Object

This function converts all Track Automation types, including those which aren't currently displayed, and moves them into a selected object.

3.4 Copying Automation Events as MIDI Data

This function allows you to conveniently copy Automation data in a number of ways. This may be useful if you wish to quickly duplicate sections of your automation track repetitively.

- Select an Automation Track. Ensure that no *objects* (MIDI sequences) are selected (otherwise, their data will be copied).
- Select **Functions > Copy MIDI Events...**
- Adjust the settings as desired in the dialog which opens

- Click the "Do It" button.

3.5 Deleting Track Automation Data

The following, self-explanatory options are available for the erasure of automation data. Choose **Options > Track Automation >**

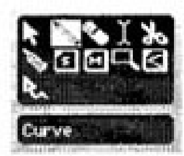
- **Delete currently visible Automation Data of Current Track;**
- **Delete All Automation Data of Current Track;**
- **Delete All Automation Data of All Tracks.**

ℹ A double click, while holding [⌥] (Mac) or [ctrl] (PC) selects all current automation data on the track. Pressing [⌫] will erase the data.

There is a further "delete" option available.

- **Delete Orphan Automation Data of Current Track**
  "Orphan" automation data is data which is not connected to any parameter anymore or has lost its destination parameter. This can happen when copying automation data between tracks for example. This function will delete the "orphaned" information.

3.6 Automation Tools



The tools in the *Toolbox* perform in the following way when dealing with Track Automation/HyperDraw editing tasks (refer to the *Automation/HyperDraw Editing* section, on page 33):

Standard Arrow Tool

The standard arrow tool acts as a multi function automation tool when used with various modifier keys. Please refer to the following section for specific details on the use of the various modifiers with this tool.

Pen Tool

This tool allows the freehand drawing of automation events, including curves.

Eraser Tool

This tool erases automation events.

Automation Arrow Tool—("Broken Arrow")

This tool has various functions which are dependent on the setting shown in the flip menu just below the tool. The two flip menu options are *Curve* and *Select*.

Curve

You can bend the line between two nodes or any selection. There are 4 differnt curve types available: convex, concave, and two different types of S-curves.

Note: this function is also available when using the standard Arrow Tool if the [ctrl][alt] modifiers are pressed simultaneously.

Select

You can rubber-band select any lines and/or nodes in the Automation track. If *the object* is clicked once, all currently visible automation events that fall within the object borders will be selected. Once a selection has been made, you can freely adjust the Automation data levels, copy, move them etc.

Note: the rubber-band function is also available when using the standard Arrow Tool, by holding down [⇧].

Hint: if you want to change the volume (or other controller) of an object, relative to all existing automation, you can do the following:

- Choose the "Select" mode from the Automation Tool flip menu
- Select the desired parameter—Volume, for example

- Click on the object. This will select the data within the object borders.
- Move the selection up or down. This is relative if you click outside the line, and absolute if you click on the line directly.

3.7 Automation/HyperDraw Editing

You will have noticed that HyperDraw has been improved a lot. HyperDraw is not only used for the editing of MIDI data (as it was before), but also for the editing of Track Automation data. MIDI data is drawn in the *object based HyperDraw mode*, accessed by **View > HyperDraw**, and Track Automation edits are performed by *track based HyperDraw*, accessed via **View > Track Automation**.

ℹ Please note that in order to see HyperDraw data, a sufficient vertical zoom level is required. Switching on the Track Automation display automatically zooms vertically, if the previous zoom level was insufficient.

HyperDraw's redesign makes it much more ergonomic. It interprets gestures and clicks in a perfect fashion. You will quickly and intuitively come to grips with its great new "feel", and will find that many different actions can be performed with a minimum of clicks and movements. These are the improvements (using the standard Arrow Tool):

- Nodes are easier to click because the clicking area has been increased.
- Click and move feels better now: Nodes don't "jump" when clicked inexactly because all changes are relative.
- Lines can be clicked.

LogicLogic "intelligently" analyses the direction of mouse movements, making the control over nodes in the Track Automation data much easier.

In cases where the mouse is initially moved in one direction—vertically or horizontally—the move is "limited" to this direction. This makes it easy to retain the node(s) value(s) while changing the position(s), or to keep the position(s) while changing the value(s). If the mouse is moved on the other plane in the X-Y axis (horizontally or vertically), the directional "limit" will be overridden and you can freely position the node(s). You can, of course, limit the movement of nodes to one direction. If you press the ⇧ key while changing the value horizontally or vertically the movement will be locked in that direction.

- A short click on a node will delete it.
- A short click on, or just outside, a line (not a node) will add a new node on the line.
- A long click on a line allows you to move the line, along with its two endpoints—i.e. the nodes that encompass the line.
- Click and hold outside a line will create and select a new node, allowing you to move that node immediately.
- A long click on a node, line or selection, while holding ⌥ (Mac OS) or ctrl (PC), allows the copying of the node, line or selection.
- A click, without previously making a selection, while holding ⌥ (Mac OS) or ctrl (PC), will allow you to drag all data, beginning from the current mouse position.
- A double click while holding ⌥ (Mac OS) or ctrl (PC), selects all automation data.
- A long click, while holding ctrl ⌥ (Mac OS), or ctrl alt (PC) on a node, a line or a selection allows various curves to be set on a line or across the entire current selection. Four types of shapes are available. Curves can also be editied using the *Automation Tool* set to Curve. These can be set by moving the mouse in one of four possible directions:
  - Horizontal S-Curve: Move the mouse right.
  - Vertical S-Curve: Move the mouse left.

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- Concave: Move the mouse down.
- Convex: Move the mouse up.

Probably the most useful shape when changing level is the horizontal S-Curve. Click a line while holding ctrl ⌥ (Mac OS), or ctrl alt (PC) and move the mouse to the right. Alternately, make use of the Pencil or Automation Arrow tools.
As a tip: When designing curves, always trust your ears—not your eyes!

- The very first click into an empty HyperDraw track creates a new node at that position, and another node at the beginning. This ensures that you won't create parameter controls with "gaps" in the middle of sequence, but have full control over the parameter from the beginning of the automation track. This applies to both Region/Sequence HyperDraw and Track Automation.

Multiple Selection of Nodes using the *Automation Tool* set to Select

- A a short click on a node will select the node.
- A short click on a line will toggle the selection of the line.
- Right underneath the parameter name in the Track List, a small numerical display together with a graphical value meter shows the current parameter value. A short click with the right mouse button (Win) or ⌘ (Mac) on the numerical display (or value meter) will select all automation data of the current parameter on that track.
- A click outside of the Track Automation data in the Arrange will deselect all.
- Clicking and holding the mouse button allows a rubber-band selection.

Additional functions: Clicking and holding the (Mac) or ctrl (Win) keys after making the rubber-band selection, will create one new node on each side of the selected area once the mouse button is released.

When clicking and holding the ctrl ⌥ (Mac) or ctrl alt (Win) keys two new nodes will be created on each side of the selected area.

- Non-contiguous selections are possible by holding ⇧.
- A long click in a selected area allows all selected nodes to be moved.
- While clicking and holding a selection can be moved. The selected area automatically "erases" any data which exists in the area it is been moved over. Immediately moving back the selection will restore the "erased" nodes. This works intelligently in cases where non-contiguous selections have been made.
- While clicking and holding a selection with ⌥ (Mac) or ctrl (Win) pressed a selection can be moved as a copy. While moving the selection existing automation date gets automatically "restored" after the copied selection has passed over them. This works intelligently in cases where non-contiguous selections have been made.

Relative and Absolute Value Changes of Selections using the *Automation Tool* set to Select

There are two choices available when changing the values of a selection of nodes:

- Clicking on a line or node, you can change all values by the same absolute amount.
- Clicking outside of a line within the selected area (on a node or outside a node), all values will be changed proportionally by a percentage value.
- Right underneath the parameter name in the Track List, a small numerical display together with a graphical value meter shows the current parameter value. A short click with the right mouse button (Win) or ⌘ (Mac) on the numerical display (or value meter) will select all automation data of the current parameter on that track. Right-click (Win) or ⌘ (Mac) and drag vertically on the numerical display (or value

meter) to scale the current automation data on that track in real time.

Value Display in HyperDraw

Numerical values are automatically displayed at HyperDraw nodes if there is sufficient space. Numerical values are context-sensitive—i.e. the centered pan position is displayed as 0 (not 64), and volume is displayed in dB.

If no event exists for volume and pan, Logic draws a horizontal line of the current value (if available). If clicked, a single event with that value will appear. Further clicks work as per usual.

HyperDraw Mode Velocity

New Hyperdraw of Note Velocity feature. Once active—by selecting **View > Hyperdraw > Note Velocity**, click-holding at any point in a HyperDraw window will activate the note velocity line tool, much like that found in the HyperEdit window. Releasing the mouse button will then change the tool to a line, which you can visually place onscreen. The end of the line is inserted by clicking a second time. This will automatically scale all note velocities, aligned to the inserted line. Please note that this only makes sense when the sequence area covered by the line already contains notes.

The Note Velocity Line Tool works in different modes:

1. **Absolute:** With no modifier key pressed, the velocity of notes will be changed to the value of the line.
2. **Relative:** with the ⌘ (Mac) key held, or the right mouse button (PC), the original and new note velocity will be analyzed. The resulting velocity is the average of both.
3. **Just Selected:** with ⌥ (Mac) or ctrl (PC) pressed, only previously selected notes will be affected.

Both Relative and Just Selected modes can be combined (⌘⌥ (Mac) or ctrl (PC) and right mouse button).

Chapter 4

Windows and Editors

4.1 Transport Window

New Look of Transport Buttons

You'll recognize Logic 5.0 at first glance by the new design of its backlit LED-style transport buttons.

Extended Functionality of Fast Forward and Fast Rewind keys

Fast Forward and Fast Rewind (Transport): A short mouse click will jump to the next or previous marker. If no markers exist, a short click will jump 1 bar forward or backward. Long clicks will rewind or fast forward as in earlier versions. Moving the mouse left or right will increase or decrease the rewind/forward speed—and can even change the winding direction.

Shuttle—Winding Like a Reel-to-Reel Tape Recorder

There are two Key Commands for "winding the reels"—i.e. moving the Song Position Line—called *Shuttle Rewind* and *Shuttle Forward*. You will find them in the Key Commands window by searching for the *Shuttle* character string. Repeated hits of the key will increase the winding speed. Repeated hits of the opposite key will slow down the shuttle speed, and eventually change the direction. Shuttle disables Cycle mode. Shuttling is halted by the stop command.

Metronome Click

The metronome speaker click sound has been changed in the Mac OS version. It's a non-pitched, neutral click sound with two different levels for the nominator and denominator.

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4.2 Arrange Window

Improved and More Flexible Look

You'll already have noticed Logic 5's improved, high-contrast new look. The background design is more flexible and user-definable. This aids the handling of all Track Automation tasks in the Arrange window, as it offers more contrast for objects, the Song Position Line, rubber-band selection outlines and the guide lines when dragging objects. In situations where multiple automation data selections are made, the area turns black.

At higher zoom levels, objects are completely colored, including the waveform or MIDI content display. Selected objects are indicated by a white separation line between text and graphics. This makes it easier to recognize selected objects, particularly when surrounding objects use **dark colors**, which can be easily confused with the black used to indicate selected objects.

Arrange Window Background

In place of the former **View > White Background**, you'll find **View > Plain Background**, which is a light yellow by default.

Under **Options > Settings > Display Preferences**, you'll find two parameters which control the Arrange window appearance:

- **Background Pattern In Arrange** allows you to choose a background pattern, but only if **View > Plain Background** is unchecked.
- **Plain Background In Arrange** allows you to choose any RGB color, but only if **View > Plain Background** is checked. Click the desired color in the palette to define a new color.

Muted Sequences, Audio Regions and Folders appear pale or textured

In the Arrange window muted sequences, audio regions or folders appear in a pale (muted) color or textured: **Options >**

Settings > Display Preferences > Muted Objects are textured. Muted sequences, audio regions and folders are also identified by a dot which precedes the object name, as per earlier versions of Logic.

Arrange Window Features

Drag&Drop of Audio Files

Audio files can now be dragged directly from the Finder/ Windows Explorer/Desktop/Folders into the Arrange window (just like MIDI files). You do not need to drag them into the Audio window first.

Selecting Looped Sequences

Looped sequences can be selected by clicking on loop-repetitions. A short click will select the looped object. Click-holding behaves like a click on the background. All objects are deselected and rubber-band selection becomes active, allowing multiple objects to be selected.

New Key Command: Record Enable Track

New Key Command: *Record Enable Track*. This command toggles between armed/unarmed modes on the selected track in the Arrange window.

New Key Commands for Note Lengths

The Key Commands window offers the following new functions in the "Various Sequence Editors" section:

- *Note Overlap Correction (selected|selected)*
- *Note Force Legato (selected|selected)*
- *Note Overlap Correction for repeated notes.*

Editing Start Point of Sequences and Regions

The left corner change behavior has been improved. Now dragging the left corner of arrange objects, or notes, is suppressed in cases where the objects are very small. This should prevent accidential changes to the left corner.

Higher Vertical Zoom Resolution

The vertical zoom has more, and finer steps.

Scrolling

Touching the background in the Arrange window, while holding the ⇧ ctrl (Mac OS) or ⇧ alt (PC) keys, allows you to scroll the window both horizontally and vertically.

4.3 Matrix Editor

Change Note Start Point

In the Matrix window, the head of a note (startpoint) can be changed while retaining the original endpoint. This is done by grabbing and dragging a note at the bottom left corner. This function is similar to adjusting the startpoint of a sequence in the Arrange window.

4.4 Hyper Editor

Hyper > Create Hyper Set For Current Events creates a new Hyper Set with event types that match those of the selected events. This is especially nice if you want to quickly create a Hyper Set for all event types created in the conventional object based Hyper Draw. You can open the Event Editor, select all, deselect the note events, open the Hyper Editor and use **Hyper > Create Hyper Set For Current Events**. All used event types can then be edited in the Hyper Editor.

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4.5 Environment

Cross-Platform Compatibility of Chord Memorizer and SysEx Mapper

The "Chord Memorizer" and "SysEx Mapper" Environment objects are now fully cross-platform compatible.

4.6 Quicktime Movie Support

Version 5 introduces a number of changes to the handling of Quicktime movie files—Mac OS only.

- A long click on the Movie window while holding COMMAND allows the opening of the menu containing the various movie options.
- A short click on the Movie window while holding COMMAND toggles between a borderless big view and the normal view.
- Clicks in the upper right area of the Movie window works as above without the need for the COMMAND modifier.

In cases where movies have active areas ("Buttons", for example), they can now be used. In previous Logic versions, such buttons did not work because all clicks within the movie area accessed the flip menu or toggled between a borderless big view and the normal view.

Logic|5

Chapter 5

Functions

5.1 New Song File Format

All old songs will automatically be converted to the Version 5 format when opened. Please note that Version 5 songs can *not* be opened by any 4.x version. Should you wish to use an older Logic version, or visit another studio that uses an older version, you may retain a copy of the song (particularly your Autoload) in Version 4 format.

Following the initial conversion, all further song save operations will automatically open the file selector, allowing you to save the song with a different name.

As a convenience, there is also an "Export 4.8 format song" option. Choose **File > Export > Export Song as Logic 4.8 Song...**, type in a file name and click on the Save button.

Please note that not all data can be saved in Version 4.x song format. Generally, this applies to data connected with new version 5 features, such as Track Automation.

5.2 Unified Virtual and Classic MIDI Engine

This new functionality integrates Logic's MIDI and Audio engines. This allows a range of new options, which are of particular use for Audio Instruments.

In the Environment window, Audio Instruments can be processed via Logic's realtime objects, such as arpeggiators, delays etc. Cabling of these objects must run *in* to the Audio Object which has been defind as an Instrument. This new functionality allows a number of creative options. An in depth description of all Environment functions and instructions on how to use them can be found in the Logic Reference Manual.

In the Arrange window, the unification of the two engines allows sequences on Audio Instrument tracks to respond to changes made to; the quantise, loop, transpose, delay, velocity, dynamic and gate time parameters in the sequence parameters box.

The new MIDI engine is capable of dealing with sample-accurate time stamp information, which ensures that sample-accurate playback timing is retained for Audio Instrument channels when Environment objects are used for processing.

If you encounter any problems or unusual behaviour:

Unchecking the "Use Unified Virtual and Classic MIDI Engine" checkbox will disable this new functionality. This is accessible via **Options > Settings > MIDI Interface Communication.**

We would appreciate an e-mail message from you indicating the nature of the problem, if any, and as many system details as you can provide. Please send messages regarding issues with this new functionality to **support@emagic.de** In the Subject line, please add the following: "UME".

5.3 Global MIDI Delay

Logic's audio capabilities have changed in a way that now allows many users to work without an external mixing desk—be it analog or digital. When working with one external MIDI instrument (or a MIDI sound module with integrated audio input mixing function), you can connect it's analog audio outputs directly to the audio inputs of the audio hardware used by Logic. As all audio (and MIDI) processing involves latency of one sort or another—which is extremely short with Logic and most recent hardware—you can compensate for this lag time by setting a negative value with the new parameter **Delay all MIDI Output**. The amount of latency depends on the audio interface used and its buffer settings.

You can find and set the parameter under **Settings > Synchronisation > MIDI**, in milliseconds.

To find the right value for your system, record a one bar, 4/4 kick drum sequence from your sound module as an audio file. Once the audio is recorded, ensure that it sits exactly on every beat—use the loop function for the audio region. Loop the MIDI sequence (playing the MIDI module) in the same way, and run both the recorded sound and the "live" sound of the MIDI module, at the same volume. If necessary, adjust the **Global MIDI Delay** parameter while listening to both sounds. If you hear any "phasing", tweak this setting until both sounds run in perfect synchronicity.

5.4 New Tool Functionality

In addition to the new automation tools discussed in the *Automation Tools* section, on page 31, the Scissors tool now operates in the following way.

When pressing OPTION (CTRL on Windows), while the Scissors tool is selected, and clicking on an object: the object will be divided equally. The Information Line will display "Divide Multiple".

As an example, if this method were employed at position 3 1 1 1 of a 12 bar long sequence object (starting at position 1 1 1 1), the sequence would be divided into 6 equal parts. If divided at position 5 1 1 1, it would be divided into 3 equal parts.

5.5 Key Commands

Nudge Commands

The arsenal of nudge Key Commands has been expanded. These apply to notes (in the Matrix, Event or Score Editor), as well as to regions and sequences in the Arrange Window. The names of the Key Commands are self-explanatory:

- *Nudge Event Length by Beat +1*
- *Nudge Event Length by Beat –1*
- *Nudge Event Length by Bar +1*
- *Nudge Event Length by Bar –1.*

Setting Note, Sequence and Region Starts and Ends to fit Song Position

You can move the start or end of a selected object (sequence, region or notes in any editor) to the Song Position Line, by use of a Key Command. Unlike the *Pickup Clock* command, audio data is not moved, when using *Set Object Start to SPL Position* on an audio region. The relevant Key Commands are:

- *Set Object Start to SPL Position*
- *Set Object End to SPL Position.*

Although these Key Commands work in any open editor window, their use only makes sense in the Arrange Window and the Matrix Editor, and perhaps in the Score Editor. This is because these editors are more "visual".

The use of *Set Object Start to SPL Position* is only possible when the Song Position Line (SPL) is placed *before* the start of the event. Similarly *Set Object End to SPL Position* only makes sense if the SPL is placed *after* the end of the event. If not, the length of the object will be shortened to one format (denominator) value.

Key Commands toggle between Tools

Key Commands which switch to a specific tool, now toggle between this tool and the previously selected tool.

Key Commands for Tools

New Key Commands for choosing a specific tool.

Video to Song Adjustment

Option > Settings > Video to Audio adjustment has been renamed to **Video to Song Adjustment**.

5.6 Recalling Screensets, Windows and Editors

Toggling Windows and Editors On and Off

Logic 5.0 features Key Commands that switch windows and editors of a given type on and off. In cases where a window is already open, but not topped, it will first be topped. With the next hit of the defined key, it will close. Open the Key Commands window and search for the *toggle* character string. Any of the editor windows can be opened, brought to the top and closed this way (Arrange, Event, Matrix, Hyper, Environment, Track Mixer, Transport…).

Recalling Screensets 1 to 9

Screensets 1 to 9 can now be recalled by freely defined key and MIDI remote commands, not only the numerical keys. This way, you can redefine the numerical keys for other puposes, such as toggling windows on and off, as described above. The commands are called *Recall Screenset 1* (… *9*, respectively).

5.7 Hierarchical Menus

A hierarchical menuing structure has been implemented for audio and MIDI objects defined in the Environment, and for native, VST 1 and 2, DirectX and Audio Instrument plug-ins. These flip menus are accessible from the Arrange window's Track List, audio channels and from parameter boxes throughout Logic. Use of these menus greatly accelerates access to different objects and is useful for organizational purposes, particularly on systems with extensive environments or numerous plug-ins.

To access the hierarchical flip menus, simply click and hold on a MIDI or audio track name in the Arrange window's Track List, an audio channel's insert point or on the **Cha** parameter value in any audio object parameter box.

The hierarchical menu structure reflects Environment layers. Entries in the main and sub-menus are listed in alphabetical and/or numerical order. Audio objects are sub-grouped by class—Audio Tracks, Instruments, Bus, Master, Inputs and Outputs, plus stereo in/outputs.

Multi Instruments are hierarchical and have 16 MIDI channels plus "All" as a sub-menu. The current program name for each channel is shown as well.

The "Sort by Layer" setting is also valid for the hierarchical instrument menu: If enabled, the Layer names will be used in the hierarchical path. If disabled, some Environment objects are hierarchical regarding their types (e.g. "Instruments" or "Macros"). Empty hierarchical paths will not be displayed.

In "Display Preferences" you can disable the hierarchical instrument menu. If disabled, the old style, long instrument menus will be displayed.

On Mac OS computers, the font size of the hierarchical menus depends on the global "Large Local Window Menus" display setting under **Apple Menu > Control Panels > Appearance > Fonts**.

Mac

Hierarchical Menus for Plug-ins

Mono and stereo plug-ins are separately grouped by API—Logic, VST or DirectX.

Logic plug-ins are further sub-grouped by type—Helper, Delay, Modulation, EQ etc.

VST plug-ins can be manually grouped by creating a folder—e.g. "modulation" within the "VSTPlugIns" folder inside the Logic program folder. Simply drag and drop the desired VST plug-ins into this newly created folder and their menu structure will be reflected when clicking on the Logic Mixer plug-in fields.

5.8 Extended Zoom Functionality

Logic enhances its zoom functionality with new navigation management options. These commands allow selected objects, or a region defined by the locators, to be zoomed to fit the screen. In addition, the last thirty zoom levels and window scroll-bar positions can be freely defined and recalled for each window. There are no default Key Commands for these navigation options, so you will need to assign them in the Key Commands window. Open **Options > Settings > Key Commands** and in the Search field, type in "nav". This will display the entries described below. Assign Key Commands as per the instructions found in the Key Commands section of the manual.

- *Store Navigation Snapshot*—the current zoom and window position scroll bars settings are saved as a "step" in the navigation path.
- *Navigate Back*—recalls the previous step in the navigation path
- *Navigate Forward*—advances to the next step in the navigation path

New Zoom Key Commands

- *Zoom to fit Selection vertically & horizontally, store Navigation Snapshot*
- *Zoom to fit Selection horizontally, store Navigation Snapshot*
- *Zoom to fit Locators, store Navigation Snapshot*

All zoom commands create a new step in the navigation path. The "Navigate Back" Key Command allows you to recall the previous zoom settings.

Remember: Using the magnifying glass allows you to zoom in several times. A brief click with the magnifying glass will return to the previous zoom and window position scroll bars settings.

Auto Track Zoom

This feature automatically toggles between a "zoomed", and standard viewing mode for the selected track. As new tracks are selected, they will automatically be zoomed to a level set by you.

To use this function, simply select View > Auto Track Zoom, and resize any track, by grabbing its lower left hand corner and moving the mouse vertically. Release the mouse button when done.

Now, as each track is selected, it will automatically zoom to the track size (zoom level) that you just set up.

MMC Input and "Full Frame Messages"

Logic can be controlled by MMC (MIDI Machine Control) and also by "Full Frame Messages" (**Listen to MMC Input**). "Transmit MMC" has been available in Logic for quite some time.

By clicking the sync button on the Transport bar, you can open the synchronization settings. Select the MIDI page and switch on **Listen to MMC Input**.

Logic then recognizes these commands:

- *Play*,
- *Deferred Play* and
- *Stop*.

Deferred Play is a special command for mechanically slow synchronisation slaves like reel-to reel tape recorders. Rather than making the machine play immediately, the machine is asked to reach the desired SMPTE position first, before playback is started. You'll find no difference in Logic's response to the "play" and "deferred play" commands, as Logic can locate as quickly as any hard disk recorder.

Logic ignores these message, when incoming external MTC (MIDI Time Code) commands are being detected.

Logic also obeys so-called "full frame messages", setting Logic's SPL to a new position, without starting playback. Once again, incoming MTC data receives higher priority if conflicting information is received.

Some synchronizers send "Full Frame Messages" instead of MTC to locate the slave device (Logic in this case) to a new position, without implicitly starting the playback. This is useful when in slow shuttle or single frame advance with video machines, because the slave device is perfectly located without being in playback mode.

MMC Locate also with internal Sync

If MMC is enabled, the "MMC Locate" command is now also sent in internal sync mode.

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Chapter 6

Plug-Ins

6.1 Plug-Ins, in General

Open plug-in window on insertion

Inserting a plug-in on any channel's insert point will automatically launch the plug-in window. This functionality can be disabled by unchecking **Open plug-in window on insertion** under **Audio > Audio Preferences**.

All Plug-Ins Totally Automated

Like Logic's own plug-ins, VST plug-ins can also be automated. Automation is not possible with DirectShow plug-ins (Windows) due to technical reasons based on the principles of the DirectShow interface.

Effect Plug-In Delay Compensation

The preference **Plug-in delay compensation,** which you'll find under **Audio > Audio Preferences** compensates for delays of audio data which may be introduced by some effect plug-ins. The compensation works with Logic internal effect plug-ins: Ad-Limiter, Denoiser, Enveloper, EnVerb, Limiter, Multipressor, Noise Gate, Spectral Gate as well as with some VST plug-ins.

For the **Plug-in delay compensation** to function the plug-in must be capable of passing on information about its processing delay to the host application (Logic). Check with the plug-in manufacturer to find out if your effect plug-ins provide this facility.

Plug-in delay compensation is effective on effect plug-ins inserted in Audio Tracks and Audio Instruments. To ensure that this compensation works with insert effects on Audio

Instruments, please ensure that the **UME (Unified Virtual and Classic MIDI Engine)** is engaged in **Options > Settings > MIDI Interface Communication**.

6.2 VST2 Support

Logic supports the VST2 format for signal processing and virtual instrument plug-ins.

"VstPlugIns" Folder

In order to work, VST plug-in files must be copied to the *VstPlugIns* folder, which resides in the Logic program directory/folder. Simply create a folder with this name if it does not yet exist.

VST Plug-ins Consider MIDI Clock

VST 2 instruments can receive VST clock information, making it possible to perfectly synchronize delays, LFO's and other effects to the song tempo. Note that the plug-in must *request* this information from Logic, and that *not all* plug-ins do this. Check with the plug-in manufacturer to see if this functionality applies to the particular plug-in used.

VST-Plug-Ins respond to Program Change Messages

Logic supports programs for VST/VST2 plug-ins, in cases where a plug-in provides program memory. You will find a menu in the head of the plug-in window that displays the currently selected program. Double click on a program name to rename it.

With VST2 instruments, internal settings (sound programs) can be changed via standard program change commands. After pressing stop, an "All notes off" message will be sent to the used Audio Instrument channel(s).

Loading Original "Effect" and "Bank" Settings

Mac

Load Settings allows the loading of Cubase™-format "Effect" and "Bank" plug-in settings into plug-in program memory. These particular settings are shipped with many plug-ins and can be found in the *VstPlugIns* folder.

Hint: Copy or move these settings into the "Plug-In Settings" folder, and from there into the folder of the corresponding plug-in to directly access these settings without having to use the file selector.

Switching VST Plug-Ins with the Plug-In Enabler

You can use the Plug-In Enabler to activate and deactivate DirectX plug-ins. VST plug-ins can be disabled by moving the associated DLL file into another folder.

For Users of VST DSP Cards

Should you use one of the VST DSP audio hardware cards, such as the *TC PowerCore*, *Universal Audio UAD-1* or *Creamware XTC*, please use the **Plug-in delay compensation** preference, which you'll find under **Audio > Audio Preferences.** It compensates for delays of audio data which may be introduced when using plug-ins with these cards.

This compensation is effective on effect plug-ins inserted in Audio Tracks and Audio Instruments. To ensure that this compensation works with insert effects on Audio Instruments, please ensure that the **UME (Unified Virtual and Classic MIDI Engine)** is engaged in **Options > Settings > MIDI Interface Communication.**

6.3 Proprietary Logic Plug-ins

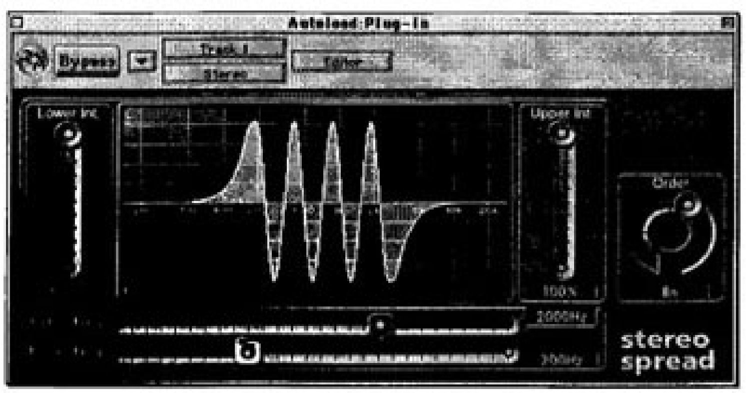
"Proprietary" Logic plug-ins are the name we give to the plug-ins included in Logic's feature set. All signal processes used in recording, mastering and broadcast studios, plus those found in sound reinforcement systems are included: dynamics of all kinds, delay and modulation effects, reverberation and room simulation, equalizers, non-harmonic distortion processes, exciter, sub-bass generator, stereo-basis enhancement and single-ended noise reductions.

The following section covers the many plug-ins that have been added since version 4.0:

Stereo Spread

The "StereoSpread" plug-in is a standard effect for mastering. There are several ways to extend the stereo base, including "Spatial Sound" algorithms of the Ghettoblaster type or tampering with the signal's phase. They all sound OK initially, but most tend to nullify the transient response and mono-compatibility. *Stereo Spread*, however, works differently.

The (**Helper** >) *Direction Mixer* is one such process. Its **Basis** parameter allows stereo spreading, when set to values higher than **1.0**. With non-transient, non-percussive material, for productions that won't be played back in mono, and for single tracks (like string pads in a complex arrangement), this effect can be of great benefit.



The *Stereo Spread* follows a pure and simple method: it extends the stereo base by alternately distributing a (selectable) number of frequency bands in the middle of the frequency range to the left and right channels. This greatly increases the perception of the stereo effect, especially when "giving" a stereo effect to monaural recordings—without making them sound totally "alien".

Order

The Order knob determines how many frequency bands the signal is to be divided into.

Upper int.

This parameter controls the intensity of the base extension of the upper frequency bands.

Lower int.

This parameter controls the intensity of the base extension of the lower frequency bands.

Note that 1. the stereo effect shows up mainly in the middle and higher frequencies, and 2., by distributing very low frequencies to the left and right speakers, you will reduce the energy distribution of any speaker. Therefore, you should always tend towards a lower intensity value for the lower frequency bands, and set the "Lower Freq." (see below) to around 300Hz (at least).

Upper Freq.

This sets the upper limit of the highest frequency band to be distributed in the stereo image.

Lower Freq.

This sets the lower limit of the lowest frequency band to be distributed in the stereo image (see also the note in the "Lower Int." section above).

The Graphic Display

The graphic displays the selected order (i.e. how many bands the signal is divided into), and and how strong the effect of the stereo spread will be in the upper and lower frequency bands.

The upper section represents the left channel, the lower section the right channel, and the frequency scale (as with the Fat EQ) shows the frequencies in ascending order from left to right.

Spreader

This plug-in is designed to enhance the stereo spectrum, just like the "StereoSpread" plug-in, but its functional principle is totally different. The *Spreader*, in a way, works like a chorus effect, but utilizes a very sophisticated frequency smoothing technique on a sample-by-sample calculation basis. This offers a sonic purity which is unknown in conventional chorus effects. The modulation of delay time with a triangular wave results in periodic non-linear frequency shifts, which means that the pitch is affected by the effect. The modulation is phase-inverted on both channels.

The stereo version works in "true stereo" (two channel operation), whilst the mono-stereo version converts a mono input signal into a stereo signal.

LFO Intensity and **Speed** can be set freely. High values lead to detuning effects. **Mix** defines the ratio of effect signal and dry original signal output; you'll set it to 100% if you insert the effect into a bus.

Channel Delay adds an extra simple delay by the displayed number of sample words. By delaying the right channel, the stereo effect becomes even broader.

De-Esser

A De-Esser is a signal processor used for the rejection of hissing, or sibilant noises. This is why it is called a "De-Esser", and occasionally an "S"-Suppressor. You can, of course, reject sizzling frequencies with an equalizer; But a De-Esser rejects the frequency band for only as long as there is a level threshold which is being overidden in a specific frequency band. This "dynamic" ability is why the sound won't get darker when no sizzling consonant is sounding. A De-Esser is a frequency-specific compressor which only compresses a particular frequency band in a complex full band signal. It features extremely fast attack and release times.

With the Emagic *De-Esser*, the dynamic rejection does not necessarily take place in the same frequency range that's being analyzed. Rather, the *De-Esser* performs a gain reduction in the frequency band displayed in the lower window, for as long as the level exceeds a threshold which falls within the frequency range displayed in the upper window.

Please don't confuse a De-Esser with an effect known as a "Vocal Stressor". The latter reduces the gain of the entire range when the level exceeds a threshold defined in a given frequency range. This type of processing can be achieved with any compressor with a high pass filter or EQ inserted in its side chain.

The Emagic *De-Esser* does not make use of a frequency dividing network (crossover utilizing low and high pass filters). Rather, it is based on a subtraction of the isolated frequency band, leaving the phase-curve untouched.

The De-Esser is important in FM radio stations, because sharp "S"-type consonants can cause harsh intermodulation distortion noises.

These are the parameters in detail:

Team TND

Detector Frequency

Here's where you define the frequency band that the *De-Esser* reacts to. It's not necessarily the same band that will be reduced.

Detector Sensitivity

Here's where you define the threshold which has to be exceeded around the **Detector Frequency**, in order to make the *De-Esser* reduce the level around the **Suppressor Frequency**.

Monitor

With this switch, you can monitor the side chain signal the *De-Esser* acts upon. If you want to reduce sibilant noises, listen to the input signal here, and set the **Detector Frequency** so that the sizzling frequency range becomes most audible.

Suppressor Frequency

Here's where you define the frequency band which will be reduced, when it falls within the frequency band defined by the **Detector Frequency, and** the threshold level, defined by **Detector Sensitivity** is exceeded.

Strength

Strength sets the amout of gain reduction around the **Suppressor Frequency**.

Smoothing

Smoothing controls the reaction speed of the gain reduction start and end phases. It's a combination of attack and release time parameters, as known from compressors.

Limiter

The limiter is also standard effect for processing a summed stereo signal, i.e. for mastering. A limiter performs a dynamic compression at a ratio of ∞:1, at a threshold set close to maximum—at least with analog limiter applications.

Gain

Most analog limiters would have a "Threshold" control, like that on a regular compressor, rather than a "Gain" control. This sets the level at which the limiter will begin to work.

As Logic's *Limiter* is digital, and that limiting is normally applied before the signal is routed to another digital process (which cannot handle more than 0dBFS), we can presuppose the following:

1) the input signal sometimes reaches 0dB, but does not exceed this ceiling, and

2) that the limiter is being used to raise the signal's overall volume. This is the reason why you find a **Gain** control here—i.e. to set the desired level of gain for the signal.

[i] The Limiter is designed in such a way that if set to 0dB Gain and 0dB Output Level, it doesn't work at all, on normalized regions.

Lookahead

Lookahead determines how far the processor "looks into the future", in order to react earlier (thus better) to incoming volume peaks, and therefore avoiding signal overloads. During playback from hard disk, the tracks are (internally) delayed slightly, allowing the "lookahead" process to react before signal peaks occur. This is not the case with realtime-processing, for instance, when the plug-in is inserted into an input object in stop mode.

Set **Lookahead** to higher values if you want the limiting effect to take place before the maximum level is reached.

Release

Here, you can set the time the limiter will need after limiting to "release"the effect and to restore the gain to its original level.

Output Level

Using this simple volume control, you can set the desired maximum level of the *Limiter's* output signal.

Soft Knee

The *Softknee* parameter produces a softer transition from "no limiting" to full limiting. If switched off, the compression curve—the output level displayed as a function of the input level—has a hard knee. If switched on, the transition to full limiting is non-linear, meaning softer. The limiting of the signal will start before reaching 0dB. This will avoid distortion artefacts occurring when strong limiting is used without a soft knee.

If you want to experiment with the effect of the **knee** parameter: The compressor (and multipressor) algorithm displays the **knee** parameter graphically.

Gain Reduction Display

The graphic display shows the reduction of level (starting from 0dB downwards).

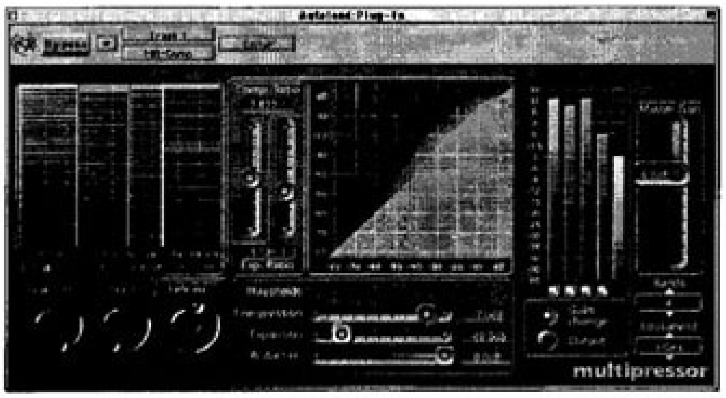
Multipressor

The "Multipressor" (an abbreviation for *multi*band com*pressor*) is the epitome of an audio mastering tool. Like no other signal processor, it can raise the perceived volume and thicken the sound of a complex arrangement without exceeding the available headroom. That said, a "good" setting is a matter of practice and experience with mastering on a wide range of material.

The Emagic *Multipressor* features a graphical user interface that's both easy to understand and easy to use. Its crossover frequencies and compression parameters can be edited directly by grabbing the graphic display with the mouse. The compression curves in the middle display, and the gain reduction (or level meters, respectively) for each band, make the *Multipressor* easier to use than any dedicated, sophisticated hardware mastering compressor.

It offers a significant advantage over analog and digital hardware multiband compressors, because of its lookahead function, which allows it to "see" a little into the future. The process pre-reads levels that will be played back from hard disk in a sophisticated buffer. The audio is then sent from this buffer through the process and played back—with no delay occuring in its output. What's more, the phase response of the process won't be distorted. This is because the *Multipressor* utilizes a phase neutral subtraction techology, instead of a conventional freqency dividing crossover network. In conjunction with the other mastering tools—the Fat EQ and Adaptive Limiter plug-in—this process is superior to the most sophisticated mastering processors, especially when you consider its 24Bit/96kHz support within the digital domain.

The Multipressor requires a lot of processing power. In cases where your computer won't allow for another multipressor plug-in, it's advisable to bounce a mixdown file first. This bounced file can be played back on a single stereo track and treated with mastering tools, while the other plug-ins are disabled.



Function Principle

The multi-band compressor splits the incoming signal into two to four different frequency bands before applying compression. These frequency bands are then compressed (and downward-expanded, to suppress noise) independently. After compression, the frequency bands are mixed back together.

The aim of independent compression of different frequency bands is to reach high compression levels on the bands that need it, without the "pumping" effect that is normally heard at high compression levels.

Here's an example: If the bass drum produces level peaks, a simple compressor would reduce the level following every beat of the bass drum. If the signal is a mix which includes the voice of a singer, you will hear the singer's voice becoming softer on each bass drum beat — i.e. the signal will "pump". Peak levels produced by the singer would also reduce the level of the bass drum. If you compress the two primary frequency bands for these elements independently, they cannot affect each other.

Much higher **Ratios**, and therefore a much higher average volume is possible before the unwanted artefacts of compression will be heard.

Hint: always ensure that the incoming signal is not digitally clipped; the compressor works best when the signal is below 0dB.

Downward Expansion

Strong multi-band compression allows you to raise the overall volume level—resulting in a dramatic increase of the existing noise floor and ambient room reflections. "Downward expansion" allows you to reduce or suppress this noise. Each frequency band features a downward expander. This works as the exact counterpart to the compressor: while the compressor compresses the dynamic range of the higher volume levels, the downward expander expands the dynamic range of the lower volume levels. With downward expansion, the signal will be reduced in level when it is lower than the defined **Threshold** level. The effect can be compared to a noise gate, but rather than simply cutting off the sound, it proportionally fades the volume using an adjustable **Ratio**. The effect is more pronounced as the level falls further below the threshold.

Bands

This parameter (on the right side) determines the number of independently compressable frequency bands, and has a crucial impact on the amount of computing power needed for the effect. Classic multi-band compressors use 3 Bands.

Lookahead

Lookahead (just below the Bands parameter) determines how far the processor "looks into the future", in order to react earlier (thus better) to volume peaks. In playback mode (when mastering and bouncing), all tracks are internally delayed in order to compensate for the lookahead time span. This is not the case with realtime input signals.

Set **Lookahead** to higher values, when the **Peak/RMS** control (see below) is set further towards RMS.

Peak/RMS

Adjusting the control between *Peak* (full left) and *RMS* (Root Mean Square; full right) is dependent on the type of signal you wish to compress: An extremely short *Peak* detection setting is suitable for compressing low-powered, short and high peaks, which do not typically occur in music. The *RMS* detection

Team TND

method measures the power of the audio material over time and thus works much more musically. This is because human hearing is more responsive to the overall power of the signal than to single peaks. As a basic setting for most applications, the center position of the control is recommended.

Attack

Allows you to define the time (in milliseconds) required before compression is faded in. At high values for **Attack**, fast pulses can exceed the threshold, but the sound remains more vivid. To avoid exceeding the threshold, you can make use of the Adaptive Limiter.

The attack of the Multipressor is extremely fast. If the Multipressor produces clicks and pops while changing parameters (ie.; during automation), you can reduce these clicks by setting the attack parameter to values lager than 5ms for the desired band. This will smooth the compression curve and all resulting volume changes.

Release

Defines the time required by the compressor to "release"the effect, following compression. Just as with the other values, the best setting for this is greatly dependent on the material to be compressed.

As an example, with fast drum&bass passages you would need to set the shortest possible *Release* time, if you wanted all individual impulses to be compressed. On speech recordings, it often makes sense to set values greater than 1 second (1000 ms). The aim is to avoid "pumping" of the signal, once this parameter is set.

Multiband Graphic

The graphic editor to the left displays several settings, both graphically and numerically.

Crossover Frequencies

The crossover frequencies (vertical borders) between the bands are variable. To change a band's crossover point, grab the border(s) directly within the graphic and move them to the left or right. The frequency is displayed numerically at the bottom of the graphic.

Absolute Volume

The horizontal line in the middle of each band displays its current level (default: 0dB). By grabbing the area below this line and moving it up and down, you can set the absolute volume level of the corresponding band. The level is displayed numerically at the bottom of the graphic.

This ability allows the *Multipressor* to serve as a basic equalization tool, dependent on how the crossover frequencies are set.

Threshold Display

The horizontal lines (up to three) in the lower area of the window represent the *Threshold* values for *Compression* (upper line), *Expansion* (middle line), and *Reduction* (bottom line). You can set these values by using the controls of the same name (see below).

Selecting the Band to Edit

You can select which frequency band you wish to edit, by clicking in the lower section of each band. Set the *Ratio* and *Threshold* values for each band as desired.

Comp. Ratio

This is the central parameter for compression, in conjunction with *Compression Threshold*. The *Comp Ratio* determines the strength or rate of reduction of the range to be compressed. In most cases the most useful combinations of these two settings are either 1) low *Threshold* and low *Ratio* or 2) high *Threshold* and high *Ratio*.

The first set of settings is suitable for speech recordings which contain passages of differing volume, that need to be levelled out.

The latter is suitable for any material containing percussive instruments. As short transients have a high level, the instrument cannot be mixed at the desired volume, so the overall signal becomes too low. In this scenario, you would select a high compression rate (Ratio), which comes into effect at higher volume levels.

Exp. Ratio

This is the central parameter for controlling downward expansion, in conjunction with *Expansion Threshold*. It determines the strength of expansion, applicable to the range to be expanded.

The general idea behind "Downward Expansion" is to get "between" the level of the noise (raised by the compressor itself) and the level of the *Compression Threshold*, in order to reduce it. Ideally, this will result in only the "noise" portion of the signal being affected by the expansion process. Use the Expander with care: It can cause a loss of signal energy that you had intended to "push" with the Multipressor.

- Normally, on signals with a relatively low noise floor, the *Expansion Threshold* can be set low. This way, a greater range of the middle volume levels remain unaffected, and only very low passages will be expanded (i.e. reduced in level). At extreme settings, you can almost achieve the functionality of a Noise Gate (cutting off the noise floor down to almost 0dB).
- In signals with a strong "hiss", expansion should start slightly below the *Compression Threshold*, at a low *Ratio*, to prevent noisy passages from being reduced too much in level.

Graphic curve

The graphic area in the middle of the editor shows the ratio between the input level (horizontal scale) and output level (vertical scale) of all bands. The colors correspond to the colors of the frequency bands in the left graphic area. By adjusting the

Ratio and *Threshold* controls, you can change the curvature of the selected frequency band.

Thresholds: Compression

Here, you can set the minimum level at which the compressor will begin to work. If the control is set all the way to the right (0dB), the entire compressor section of the *Multipressor* is "off-line". The further the control is moved to the left, the lower the level above which the compressor will work.

Thresholds: Expansion

Here, you can set the maximum level at which the expander should work. If the control is set all the way to the left (–50dB), expansion will only occur on signals that fall below this level. (The *Exp. Ratio* can be set to a minimum of 1.2:1; below –50dB the expansion always takes place at this low ratio.) The further the control is moved to the right, the higher the level below which the expander will work.

Reduction

Allows you to define how much the noise level should be reduced (this is not a threshold value). If you move the control all the way to the left, the reduction will be maximized (by -50dB). If the control is set all the way to the right, no reduction will occur.

Level Meter

In the level meter to the right, you may monitor either the change of level caused by the compression, or the output volume of each band, depending on whether you have selected *Gain change* or *Output* (see below). You can individually switch the bands on and off, in order to listen to single bands, by using the switches below the meters. If the switch below a band's meter is lit (light green), then the band will be audible. If it is dark, the band will be muted.

Gain Change/Output

The *Gain change* and *Output* buttons can be used to switch the operation of the meters.

- If *Gain change* is selected, the meters indicate the strength of reduction to the level of the audio material by the compressor.
- If *Output* is selected, the meters show the absolute output level of the corresponding frequency band.

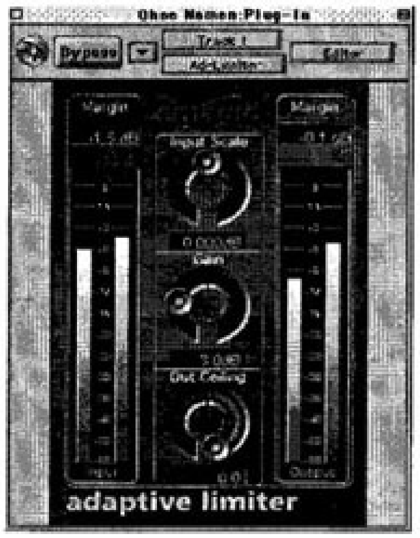
Master Gain

Allows you to reduce the overall gain increase (or increase the overall gain reduction), resulting from the *Multipressors* settings.

Ad-Limiter

With its *Compressor*, *Multipressor* and *Limiter*—not to forget the destructive process of normalizing—Logic features several extremely versatile options for increasing perceived volumes. A further tool which can be used to increase the perceived level of signals is the *Adaptive Limiter* plug-in. In the world of analog processors, it could be more closely compared to a "clipper", rounding and smoothing only harsh level peaks, rather than to a VCA-type limiter. It allows you to achieve maximum gain, without having to fear exceeding 0dBFS. The *Ad-Limiter* may slightly color the sound. An effect most similar to an amplifier when driven hard.

Team TND

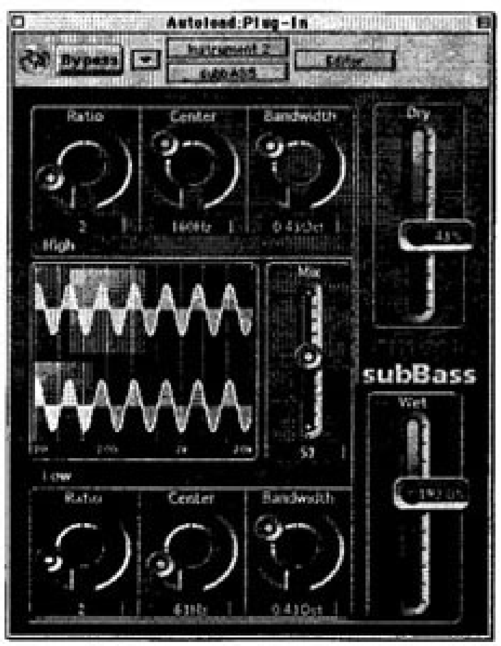


Following the Adaptive Limiter process, tracks normalized to 0dB appear to sound about 2dB louder, depending on the source signal. As with other Logic dynamic processes, this plug-in also features a lookahead facility (set to a fixed time span), allowing it to "look into the future". The Adaptive Limiter reacts to level peaks in signals streaming from the hard disk before they are played back, and delaying the monitored signal. The typical use of the Adaptive Limiter is in the summed mix. It is placed after the *Multipressor* and before *Volume*, in order to produce a CD of maximum loudness. As limiters compress signals, it can produce results which sound louder than those resulting from normalizing in the Sample Editor.

Start the process by adjusting the **Input Scale**, just as you would set a mixing desk's Gain parameter, or a digital recorder's recording level. The parameter behaves much like a Gain control, but its purpose is to adjust the input level, which must never exceed 0dBFS. Adjust the **Gain** parameter to "musically" control the internal process of peak smoothing and gain increase. **Out Ceiling** reduces the output level of the process in very fine steps within a range of only 2dB. This is no threshold control, just a simple output gain. The Margin-Display shows the maximum measured level. In order to reset the Margin display, click the "**Over**" lamps.

SubBass

This plug-in *generates* harmonic signal portions in a frequency band which is lower than the input signal it is derived from—i.e. an artificial bass. This may remind you of a distortion or overdrive effect: These generate harmonic contens, which means that the new frequency portions have "n" times the frequency of the input signal. As opposed to these effects, the *subBass* is a synthesizer which generates sub-harmonics. It is most similar to units often found in the power amplifier racks of discotheques. By comparison, the subBass plug-in is far more flexible. Caution: The process is known as a loudspeaker killer!



Choose moderate monitoring levels, and never try to playback sub-bass frequencies with loudspeakers which aren't capable of doing so.

Team TND

Never try to force a loudspeaker to output these frequency bands with an EQ.

The simplest use of the subbass plug-in is as a simple "octaver", just like the effects pedals available for electric bass guitars. A simple frequency division circuit in such octaver pedals requires a monophonic input sound with a clearly defined pitch. Octavers are usualyy only capable of producing an output signal which sounds one or two octaves lower than the input signal. An octave is a frequency division by 2. A ratio of 4 means two octaves, and a ratio of 8 equals three octaves.

All whole fractions between 2 and 8 are possible as well, but don't necessarily suit the key. A ratio of 3 results in a transposition of a duodecima (1 octave and a fifth, about 19 semitones) downward. This transposes a c1 into an F.

As opposed to a pitch shifter, the waveform of the signal synthesized by the *subBass* has nothing to do with the waveform of the input signal. It's shape is sinusoidal—but a pure sine wave is rarely achievable (or heard) in complex arrangements. It is mixed with the original signal. The mix ratio is defined by the **Mix** parameter.

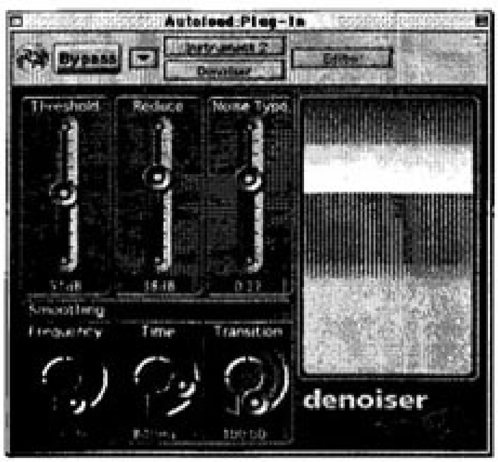
The *subBass* creates two bass signals, derived from two freely definable portions of the incoming audio. This means that processing is not limited to monophonic signals with a defined pitch, but also complex summed signals. **Center High** and **Center Low** define the center frequency of the band that the transposed signal will be derived from. The **Bandwidth** of the analyzed band(s) is/are set by dedicated parameters. Within these frequency bands, the filtered signal should have a reasonably stable pitch, in order to be analyzed correctly. The graphic shows the frequency bands of a typical boom box, which transposes two frequency bands—with the width of a fifth—by one octave each. If the analyzed frequencies chosen are a little higher, the *subBass* plug-in plays these frequencies, much like a bass player doubling the lower notes of a guitar player.

In real life, narrow bandwidths lead to the best results as they avoid unwanted intermodulations. Set **Center High** a fifth higher than **Center Low**, which means a factor of 1.5 for the **Center Frequency**. Derive the sub-bass to be synthesized from the existing bass portion (only) of the signal, and transpose by only one octave in both bands (**Ratio** = 2). Do not overdrive the process, to avoid distortion. If you recognize (hear) frequency gaps, move the **Center Frequencies**, or widen the **Bandwidth** a little.

Be prudent when using the *subBass*, and compare your super-low end signal portions with other productions.

Denoiser

The *Denoiser* eliminates or reduces noise in recordings. Being a single-ended noise reduction process, the effect causes audible artefacts.



Threshold

This value allows you to adapt the process to match the noise floor level of the source material.

Tip: Find a passage where (almost) only noise can be heard, and set the Threshold so that only signals of this volume will be filtered out.

Reduce

Here, you determine the amount of reduction in dB.

Noise Type

The noise floor may mainly contain high frequency or mainly low frequency portions:

- The value 0 means white noise (neutral, equal frequency distribution in each bandwith measured in Hz),
- With positive values (above 0), you'll mainly reduce pink noise, which contains more bass (warmer, darker sound, or equal frequency distribution in each bandwith measured in musical interval ratios).
- With negative values (below 0), you'll mainly reduce blue noise, which contains more hiss (sharp sound).

Smoothing Parameters

The *Denoiser* works on the basis of FFT analysis, to recognize frequency bands of a lower level and lesser harmonicity. It then reduces these bands to the desired dB value. In principle, this method is never exact, as neighboring frequencies will also be affected. If you use the *Denoiser* too agressively, the algorithm will produce the so-called "glass noise" effect, which is truly interesting, but in most cases is even less desirable than the existing noise.

There are three parameters for reducing this effect in three dimensions:

Time Smoothing

This is the simplest form of smoothing. Here, you set the time the Denoiser requires to reach or release maximum reduction.

Frequency Smoothing

Here, you define the width of a frequency band which overlaps neighboring frequencies. As mentioned, these neighbors will also be reduced by the noise reduction. More precisely: If the Denoiser recognizes that only noise is present in a certain frequency band, the higher the Frequency Smoothing parameter is set, the more it will also change the neighboring frequency bands, in order to avoid the unpleasant "glass noise".

Transition: Level Smoothing

Here, you can set a factor for a smoother transition of the denoising process to the "neighboring" volume levels. More precisely, if the *Denoiser* recognizes that only noise is present in a certain volume range, the higher the Transition Smoothing parameter is set, the more it will also change similar level values, in order to to avoid glass noise.

The Graphic Display

The graphic displays the **Threshold**, **Reduce**, **Noise Type** and **Transition** parameters.

Exciter

An exciter *generates* high frequency portions that are not present in the input signal. It is somewhat akin to a non-linear distortion process, resembling overdrive and distortion effects. As opposed to these types of effects, however, the distorted signal is mixed with the dry signal.

The harmonics generator is fed by a high-pass-filtered version of the input signal. This means that the artificial harmonics have frequencies at least one octave above the lowest frequency allowed to pass by the high pass filter (at least the double frequency). Humans can barely distinguish the original treble signal from artificial harmonics in very high frequency ranges.

Team TND

The Cutoff Frequency parameter of the high pass filter is called **Frequency**. The graphic displays the frequency range that is used as the source signal for the process. **Harmonics** controls the level of the effect signal mixed with the original signal. If you disable **Input**, the original signal won't be fed through. This should be disabled when using the plug-in in a bus return object, being fed by auxiliary bus sends from several channel strips simultaneously, or when you wish to listen to the effect signal in isolation.

The lower you tune the filter; the more harmonics the effect will produce, the lower the frequencies of the artificial harmonics will be—and the less natural they will sound.

Color

You can choose between two sound characteristics of the distortion. **Color 1** is less dense in spectrum than **Color 2**. The effect of **Color 2** is more intense, but it also introduces more (un)wanted intermodulation distortion.

Basically, higher settings for **Frequency** and higher **Harmonics** (Mix) settings are preferable, because we can't distinguish between artificial and original treble frequencies very well. Exciters can give "life" to digital recordings. They are especially well-suited for audio tracks where the treble somehow got lost. Exciters are very commonly used on guitar tracks.

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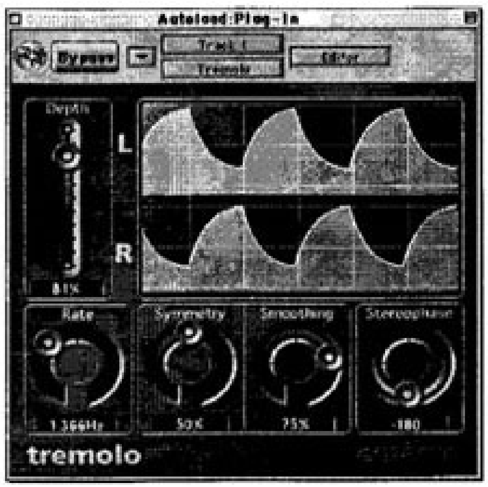
Tremolo

The tremolo effect is a cyclic modulation of the amplitude, resulting in periodic changes of volume. As opposed to the vibrato effect, the amplitude (not the frequency) is the modulated parameter. You'll know this effect from vintage guitar combo amplifiers (sometimes erroneously called "Vibrato" on some amp combos).

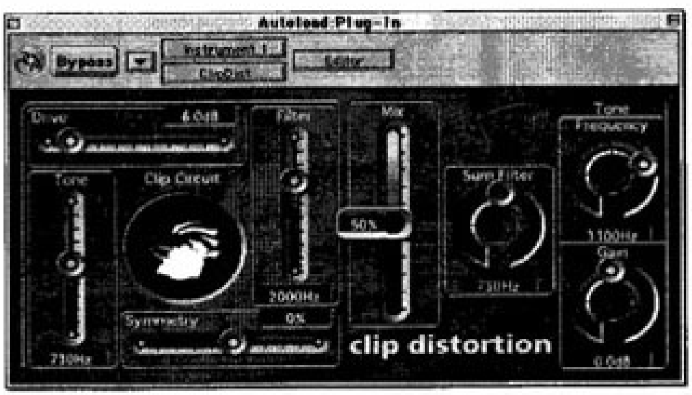


The intensity of modulation is set with the **Depth** parameter. **Rate** defines the speed (frequency) of the modulation. If **Symmetry** is set to 50% and **Smoothing** to 0%, the modulation has a rectangular shape. This means that the time of the full volume signal is equal to that of the low volume signal, and switching between both states occurs abruptly. Define the loud/quiet time ratio with the **Symmetry** parameter, and make it fade gently in/out with the **Smoothing** parameter. **Stereophase** defines whether the modulation takes place in phase or out of phase, when in stereo mode. It can be set to any phase angle. Set to "out of phase" (–180°), the balance cyclically "wanders" from left to right and back (Stereo Panning). When set to 0°, left and right channels are altered in volume simultaneously (in phase).

The graphic display is self-explanatory: All parameters, except modulation speed (**Rate**), are displayed.

Clip Distortion

The *Clip Distortion* plug-in is a non-linear distortion process which produces unpredictable spectra. Beyond drastic distortions, it's well suited for the simulation of warm tube overdrive sounds.



You'll probably check out the effect of each parameter as a matter of course for any given input signal. At any rate, this is what the controls actually do in the signal flow of this algorithm:

The input signal is first amplified by the value defined with the **Drive** parameter, which is a simple Gain control. The signal is then sent through a high pass filter. The filter cutoff frequency is defined by the **Tone** fader. The actual non-linear distortion process follows this.

Symmetry controls the sound characteristics of the distortion effect, from transistor (extrem left or right settings) to valve type sounds (around the 0% mark).

Imagine a system diagram, displayed as an X-Y graph. The output signal is shown as a function of the input signal, with the output representing the x-axis and the input representing the y-axis: The non-linear distortion circuit curve looks like an S. (No treatment at all would result in a straight line with a 45° angle. Amplification would raise the angle.) The **Symmetry** parameter allows you to stretch and bend the *S*-curve, so that for any value other than 0%, it is asymmetrical.

Once distorted in this way, the signal then passes a low-pass filter, the cutoff frequency of which is set by by the **Filter** fader. The **Mix** parameter mixes the effect signal with the dry signal. This signal then passes yet another low-pass filter, the cutoff frequency of which is set by by the **Sum Filter** control. All filters have a low rejection of 6dB/octave. The last stage of signal processing is a tunable high shelving filter. If you set its **Frequency** to about 12kHz, it will behave like a normal treble control, as found in any Mixer channel strip or hi-fi amplifier. As opposed to such treble controls, the *Sum Filter* allows signal boosts and rejections of up to ±30dB. This somewhat unorthodox combination of serially-linked filters allows many different frequency spectra to be defined and created. With particular "gaps" in the signal, this sort of non-linear distortion can sound very nice indeed. The clip circuit graphic displays every parameter symbolically, with the exception of the shelving filter controls.

Check out filters and EQs, including the simple parametric EQ, inserted *before* the Clip Distortion. Also try combining the Clip Distortion with other distortion plug-ins to create interesting results.

Phase Distortion

The *Phase Distortion* plug-in is based on a modulated delay line, much like the well-known chorus or flanger effect. As opposed to these, the delay time is not modulated by a low frequency oscillator (LFO), but rather by a low-pass-filtered version of the audio input signal itself. This is how the signal modulates its own phase position.

In the signal flow of this effect, the parameters do the following:

The input signal only passes the phase delay line and is not affected by any other process. With **Mix**, you can mix the effected signal with the original signal. The phase is modulated by a sidechain signal—i.e. the input signal. The input signal passes through a resonating low pass filter, the **Cutoff** Frequency and **Resonance** of which can be set with dedicated controls. You also can listen to the filtered sidechain (instead of the Mix signal), if you engage **Monitor**. The maximum phase shift is set with **Max Modulation**. The depth of the modulation itself is controlled with the **Intensity** parameter which will not exceed the maximum phase shift, defined by the **Max Modulation** setting.

In Controls View, there is one more parameter which can't be seen in the plug-in window. It is only valid for the stereophonic version of the effect. Normally, a positive input value results in a longer delay time. If you engage **Phase Reverse** (On), positive input values result in a reduction of the delay time on the right channel only.

ℹ You can change the phase of the signal before and after the process with the (**Helper** >) *Gainer* plug-in, thus reversing the stereo effect.

6.4 Plug-in Improvements

Reverbs Optimized for Athlon AMD

The performance of reverb plug-ins has been improved for computers utilizing the Athlon processor.

Fat EQ slope characteristic adjustable

The shelving filter's slope characteristics for bands 2 and 4 are now adjustable via the **Q** parameter.

New Plug-in Parameters

New plug-in parameters have been added to some plug-ins. Please note that all new parameters are accessible when the plug-in view is switched to the Controls View. To see the new parameters in the Editor view, please activate the **001011** switch in the gray header of the plug-in window. This button is only visible when a plug-in actually has additional parameters available in the Controls view.

Compressor: Output Clip

Output Clip clips the output at 0dB. You have the choice of two knee curves when approaching the limit. Settings are: OFF/Soft/Hard.

Oscillator: Sine Wave

Osc Mode switches the output of the oscillator between Normal and Sine. When set to Sine, the volume sliders for all waveforms except Sine are disabled, but ring modulation still works. Select Sine for the smoothest and purest ring modulation effects.

Autofilter: LFO Syncable to Logic's Tempo

The LFO is is syncable to bars & beats. Switch **Beat Sync.** to ON, use **BPM Rate** to adjust the ratio between LFO speed and the sequencer tempo. You can shift the phase relationship between the LFO and the sequencer with **Sync Phase**.

Tape Delay: Delay Modulation, Flutter Simulation and Freeze

The *Tape Delay* features an LFO hidden in the Editor View. The LFO modulates the delay time, enabling you to achieve vibrato and chorus effects with it—even when using long delay times. The LFO (with adjustable speed and modulation intensity) produces a triangular wave, which can be smoothed with the **Smooth** parameter. This also smoothes out any "flutter". **Flutter** simulates the irregularities in tape speed of the original tape transports used in analogue tape delays. It is also adjustable in speed and intensity. **Freeze** preserves the delay feedback at the time of its activation: **Freeze** (ON). The delay feedbacks are repeated with constant volume until **Freeze** is deactivated: **Freeze** (OFF).

The Tape Delay plug-in is available in mono and stereo versions.

6.5 Enverb and Automation

The main parameters of the Enverb Plug-In can't be automated. This is due to the fact that the algorithm requires constant recalculation of base values over the time axis.

This is a small price to pay for the Enverbs unprecedented ability to apply a constant reflection cluster to the input signal, continuously and in realtime. This approach is in sharp contrast to the shaping of a reverberation signal through the use of a post-signal volume envelope, as would be the case with a conventional "gated reverb". Realtime parameter movements will mute the Enverbs output. This restriction on the automation of the Enverb plug-in is unique in Logic. All other plug-ins can be fully automated.

In a real-world scenario, you can achieve dynamic "morphs" between two Enverb reflection clusters in the following way: Insert the Enverb into two return objects and feed them with two independent aux sends. You can easily automate the send levels (even independently for each track, if you wish) and thus create reverb cluster crossfades between two (or more) Enverb reverberation effects.

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Chapter 7

Audio Instruments

7.1 Audio Instrument Objects

Logic supports the use of virtual instruments such as Emagic's **ES1**, **ES2**, **EVP88**, **EXS24** and the included **ES M**, **ES E** and **ES P** synths, plus VST2-compatible sound generators. To facilitate the use of these software instruments, Logic features a category of audio objects called *Audio Instruments*. Virtual instrument plug-ins are inserted into the *top* insert slot of these Audio Instrument channels. The default song—the song that opens if you move your Autoload away from the Logic folder—features a number of ready-to-go Audio Instrument channels.

Audio Instruments integrate seamlessly into Logic's internal digital Mixer; all plug-in effects can be used, and all parameters can be automated. The direct connection with Logic's sequencing engine guarantees unsurpassed precision through sample-accurate playback timing, superior to that of any external MIDI synthesizer. This playback precision is due to the fact that no conversion into MIDI data is required—despite the Audio Instruments being played by the same MIDI events and MIDI sequences that external MIDI instruments use.

An Audio Instrument is an audio object with the **Cha** parameter switched to one of the Instrument channels. Each audio object can behave as an Audio Instrument by changing the *Cha* parameter in its object parameter box. You can only insert the **ES1 (or other instrument)** plug-in into an audio object—created by selecting **New > Audio Object**—after its **Cha** parameter is set to an Instrument channel.

The maximum number of Audio Instruments is 32 with Logic Platinum. These include: **ES1**, **ES2**, **EVP88**, **EXS24** or VST2 instruments.

Global Instrument Tune

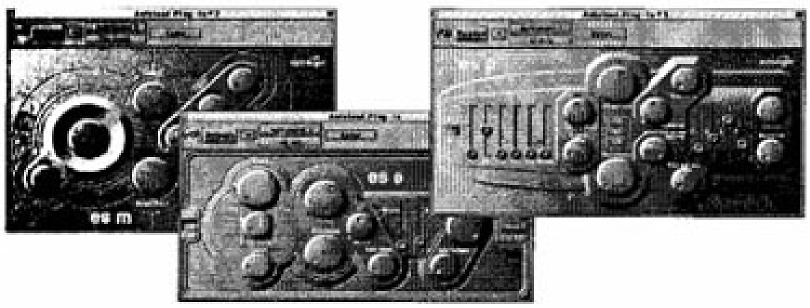
This new parameter, found under **Audio > Audio Settings,** remotely controls the main tuning parameter for all Audio Instruments (plug-in synthesizers, like the ES1 or EXS24 sampler and others) by ±50 Cents. This parameter also affects VST2 plug-in instruments. Some instruments from other manufacturers may not recognize this remote command.

7.2 Three Integrated Synthesizers

Logic features three integrated synthesizers, which are used in the same fashion as the optional **EVP88, EXS24, ES 1, ES2** or VST2 instrument plug-ins. In order to open and use the included synthesizers, click-hold on the upper-most insert slot of an Audio Instrument channel strip (in the Audio or Track Mixer). In the ensuing flip menu, select **Stereo > Logic > ES E** or **ES P**, or **Mono > ES M,** from the hierarchical menu. If you haven't already activated the **Preferences > Audio > Open Plug-In Window automatically**, double-click on the (now blue) slot, and the plug-in window will open. Select a track which corresponds to the Audio Instrument in the Arrange window, and check that it is in record-ready mode (red **R** switch).

The Audio Instrument track must be in record-ready mode, when you want to play the plug-in-synthesizer in realtime. Only the plug-in instrument of the currently selected track can be played.

All three synthesizers are subtractive virtual analog synthesizers with easy-to-understand user interfaces—a joy not only for experienced synthesizer fans, but also ideally-suited for beginners. If you're new to synthesis, you'll find that your first steps in synthesizer playing and programming are easy. The es synths are modelled on the marvellous analog synthesizers of the seventies and eighties.



*The three synthesizers—***ES M**, **ES E** *and* **ES P**.

The **ES M**, **ES E** and **ES P** synthesizer plug-ins are designed for specific musical purposes:

- The **ES M** (**ES Mono**) is a monophonic bass synthesizer. Its design is so simple that you'll find it hard to create a "bad" sound. It resembles a well-known bass synthesizer which is very popular in techno and dance music styles.
- The **ES E** (**ES Ensemble**) is designed to deliver pad sounds—classic string synthesizer sounds, for example.
- The **ES P** (**ES Poly**) is a sophisticated synthesizer. Its ADSR envelope generators are well-suited for percussive and classic synthetic brass sounds, plus hundreds of other well-known synthesizer sounds.

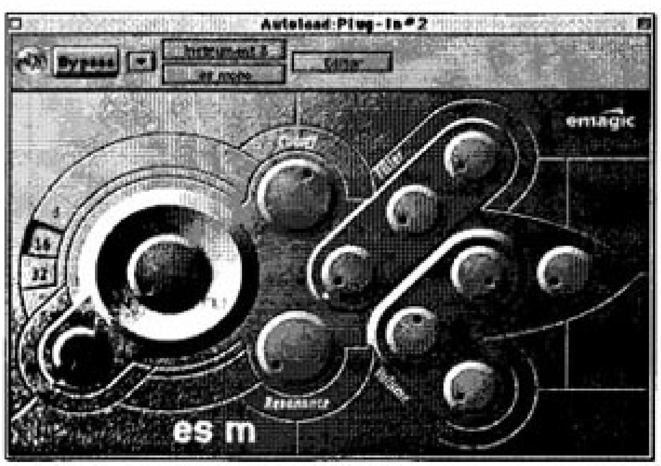
The **ES E** and **ES P** feature eight-voice polyphony. The **ES M** is monophonic. All synthesizers feature one oscillator per voice. The polyphonic synthesizers feature integrated chorus effects. All three synthesizers are designed to make the most efficient use of available processing resources. You can open several identical plug-ins with different settings in different Audio Instrument channels.

For more complex subtractive virtual analog synthesizer sounds, the optional Emagic ES 1 and ES 2 synthesizers are recommended. These instruments also benefit from the same seamless integration into Logic.

The description of parameters is brief in this manual. We have made the presumption that, as a Logic user, you will have had at least some basic experience with the world of subtractive synthesis. If this is not the case, simply check out some of the included *Settings* to see how things work. As a reminder: Alongside the *Bypass* switch in the plug-in window, there is a flip menu accessible via the downwards-pointing arrow. Click-hold on this arrow, and select **Load Setting**. In the ensuing dialog box, select the desired Setting. Feel free to edit the parameters while playing and listening. In the following pages, you'll find a brief description of the various synthesizer parameters.

ES M

The **ES M (ES Mono)** is designed for use as a bass synthesizer. That said, you may want to use it for other, higher-pitched solo voices.



Emagic **ES M** *panel*

Octave Select

Choose the octave transposition here. You can also transpose the Audio Instrument using the track playback parameters in the Arrange window, as with MIDI tracks.

Glide

The **ES M** is permanently in a "Fingered Portamento" Mode, with notes played in a legato style resulting in a glide (portamento) from pitch to pitch. The speed of the glide is set with the **Glide** parameter. At a value of 0, there is no glide effect at all.

Mix

Mix cross-fades between a sawtooth wave and a 50%-rectangular wave, which sounds one octave lower.

Cutoff

Cutoff Frequency of the resonance-capable dynamic low pass filter. Its slope is 24dB per octave.

Resonance

Resonance parameter of the dynamic low pass filter. Surrounding the cutoff frequency is a band of resonant frequencies, the intensity of which is set by this parameter. With this filter, the loss of bass caused by *Resonance* can be compensated for by a bass boost.

Int

The **ES M** features two very simple envelope generators with a single decay parameter. **Int** sets the modulation of the cutoff frequency by the filter envelope.

Decay (Filter)

Decay Time of the filter envelope. This parameter is only effective if **Int** is not set to 0.

Velo (Filter)

Velocity sensitivity of the filter envelope. This parameter is only effective if **Int** is not set to 0.

Decay (Volume)

Decay Time of the dynamic stage. Attack time and Release time, as well as the Sustain level of this synthesizer always equals zero.

Velo (Volume)

Velocity sensitivity of the dynamic stage. The parameter defines how sensitive the level of an **ES M** voice is to harder key strikes.

Vol

Master volume of the **ES M**.

Overdrive

Overdive/distortion for the **ES M** output. Caution: The overdrive effect significantly increases the output level.

ES E

The eight-voice polyphonic **ES E** (**ES Ensemble**) was designed for "pad" and ensemble sounds, but please feel free to design totally different sounds.

Emagic **ES E** *panel*

Octave Select

Choose the octave transposition here. You can also transpose the Audio Instrument using the track playback parameters in the Arrange window, as with MIDI tracks.

Wave

At the left-most setting of the **Wave** parameter, the oscillators output sawtooth signals, which can be modulated in frequency by the integrated LFO. Across the remaining range, the oscillators output rectangular (or pulse waves), with the average pulse width being defined by the **Wave** parameter. The pulse width can be modulated—that's what "PWM" (Pulse With Modulation) stands for.

Vib/PWM

If **Wave** is set to sawtooth, this parameter defines the amount of frequency modulation, resulting in a vibrato or siren effect, depending on LFO speed and intensity. If **Wave** has been set to a pulse wave, this parameter controls the amount of pulse width modulation (PWM). When the pulse width becomes very narrow, the sound sounds like it is being "interrupted". As such, set the PWM intensity with care, and select the **Wave** parameter's 12-o'clock-position (50% rectangular) for the pulse width, if you want to achieve the maximum modulation range.

Speed

Speed controls the frequency of the pitch (sawtooth) or pulse width modulation.

Cutoff

Cutoff frequency of the resonance-capable dynamic low pass filter.

Resonance

Resonance parameter of the dynamic low pass filter. Surrounding the cutoff frequency is a band of resonant frequencies, the intensity of which is set by this parameter.

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AR Int

The **ES M** features one simple envelope generator per voice. It features an **Attack Time** and a **Release Time** parameter. **AR Int**, defines the amount of cutoff frequency modulation applied by the envelope generator. Negative modulations are possible.

Velo (Filter)

Velocity sensitivity of the cutoff frequency modulation applied by the envelope generator. This parameter is only effective if **AR Int** is not set to 0.

Attack

The **Attack Time** is the time required for each note to "fade in".

Release

The **Release Time** is the time required for each note to fade out, following the release of the key.

Velo Volume

Velocity sensitivity of each notes level.

Volume

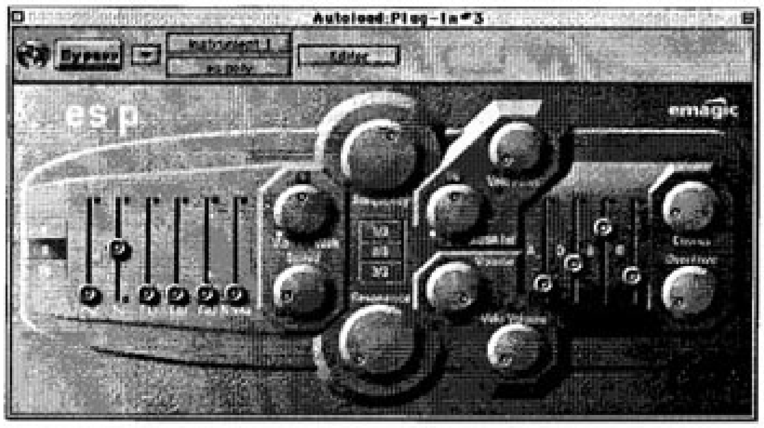
Master Volume of the **ES E**.

Chorus/Ensemble

The **ES E** features a chorus/ensemble effect, switchable in three values (plus off).

ES P

The eight-voice polyphonic **ES P** (**ES Poly**) is quite a flexible synthesizer. Functionally, (despite its velocity sensitivity) is somewhat reminiscent of the affordable polyphonic synthesizers of the leading Japanese manufacturers: Its design is easy to understand, it is capable of producing lots of useful musical sounds, and you may be hard-pressed to make sounds with it that can't be arranged in at least some musical style. Classic analog synthesizer brass sounds are just one of its many goodies.



Emagic **ES P** *panel*

Octave Select

Choose the octave transposition here. You can also transpose the Audio Instrument using the track playback parameters in the Arrange window, as with MIDI tracks.

Wave

The faders on the left side of the panel allow you to mix several waveforms, output by the oscillators of the **ES P**. In addition to triangular, sawtooth and rectangular waves, the rectangular waves of two sub-oscillators are also available. One of these sounds one octave lower than the main oscillators, and the other, two octaves lower. The pulse width of all rectangular waves is 50%. The right-most fader adds white noise to the mix. This is the raw material for classic synthesizer sound effects, such as "ocean waves", "wind" and "helicopters".

Vib/Wah

The **ES P** features an LFO which can either: modulate the frequency of the oscillators, resulting in a vibrato, or the cutoff frequency of the dynamic low pass filter, resulting in a wah wah effect. Turn the control to the left in order to set a vibrato, or to the right to cyclically modulate the filter.

Speed

Speed controls the rate of the oscillator frequency or cutoff frequency modulation.

Cutoff

Cutoff frequency of the resonance-capable dynamic low pass filter.

Resonance

Resonance parameter of the dynamic low pass filter. Surrounding the cutoff frequency is a band of resonant frequencies, the intensity of which is set by this parameter.

1/3, 2/3, 3/3

The cutoff frequency can be modulated by MIDI note number (keyboard position); You may know this parameter as "Keyboard Follow" on other synthesizers. You have the choice of no modulation, one third, two thirds or full keyboard follow (3/3, or 100%). When set to 3/3, the relative harmonic content of each note is the same, independent of its pitch.

ADSR Int

The **ES P** features one ADSR envelope generator per voice. **ADSR Int** sets the amount of cutoff frequency modulation by the ADSR envelope generator. The **ES E** allows for negative filter modulations as well.

Velo Filter

The cutoff frequency modulation by the ADSR envelope generator, controlled by **ADSR Int**, is velocity sensitive. The amount of velocitiy sensitivity is set by this parameter.

Volume

Master volume of the **ES P**.

Velo Volume

Velocity sensitivity of each notes level.

A (Attack Time)

Attack time of the envelope generator.

D (Decay Time)
The time the envelope needs to fall from maximum level to the sustain level.

S (Sustain Level)
Level that is sustained after the completion of the attack and release phases. It is only valid as long the key is still held.

R (Release Time)
Fade out time of each note, following the release of the key, or, to be more precise, after the *Note Off Event*.

Chorus
Intensity of the integrated chorus effect.

Overdrive
Overdive/distortion for **ES P** output. Caution: The overdrive effect significantly increases the output level.

7.3 ReWire

Logic supports the *ReWire* software interface from software manufacturer *Propellerhead*. ReWire is an interface between sequencer and audio applications. ReWire works with following cards and drivers using Logic's internal audio engine:

… under Mac OS:

Mac

- Mac AV
- Audiowerk
- 1212 I/O
- Direct I/O (Digidesign cards)
- ASIO
- StudI/O

… under Windows:

- PC AV (MME, EASI, DirectSound, WDM)
- ASIO
- Audiowerk

- Direct I/O (Digidesign cards).

Please start Logic first, and then start your ReWire application(s) Propellerhead's *ReBirth*, for example. Please also refer to any notes regarding this in ReWire's "read me" files.

Logic will display the audio returns of ReWire-connected applications as audio objects: You're able to insert plug-ins and create sends to busses on these returns, as per a normal audio track.

In order to use the audio returns of ReWire-connected applications, select an audio object and choose the desired ReWire audio-return in the flip menu of the **Cha** parameter of the audio object. In this flip menu you will find ReWire channels listed with their names along with the audio tracks, inputs, busses and outputs. ReWire or the application to which the connection belongs generates the names for these channels. The naming of a ReWire channel is not relevant to Logic; For example, if a channel is called "Mix" it has no specific function related to that name as far as Logic is concerned. It is up to decide what to do with whatever channel. If there is no audio object for a ReWire return created in the Environment, no audio is transferred between a ReWired application and Logic. How does this work in practice with ReBirth, for example? In this case, you will find two channels called "Mix L" and "Mix R" within Logic. The two Mix channels return the summed mix of ReBirth's outputs. You will also find channels for every individual instrument. When creating an audio object for another channel such as the bass drum, for example, you will only hear the bass drum through that channel, and it is removed from the mix returns.

In comparison to VST2, ReWire has a relatively high MIDI latency due to technical reasons. This is not an issue when using software like "Rebirth" with Logic, as both applications are audio sequencers that are sample-accurately synced via the audio clock. Logic is always Sync master. As Logic is also the Cycle master, "Skip Cycle" works, too.

These functions are shared between the applications:

- Start/Stop,
- Repositioning,
- Cycle Start and End,
- Tempo Change.

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Chapter 8

Mixers

8.1 Improved Design

The adaptive Track Mixer can be reconfigured to only show desired tracks and objects. The design has been changed to save display space and to be more ergonomic. The Audio Mixer, which has, in fact, been a layer of the Environment ever since the first version of Logic, benefits from most of these improvements as well. Given that the Track Mixer is so easily configured, and the Track Automation has been standardized, you'll probably find that you prefer using the Track Mixer, even if you previously used the Audio Mixer to perform your mixes.

8.2 Track Mixer

Open the Track Mixer by selecting **Windows > Track Mixer** and the Audio Mixer by selecting **Audio > Audio Mixer**, or via the appropriate Key Commands. Both Mixers can also be launched via default screensets.

L-Shape: Displaying Long Instrument and Program Names

All channels have a rectangular shape by default. **View > L-Shape** engages an "L" shaped display of instrument and program names.

Track Filters: Selecting which Kind of Objects to Display

Normally, the Track Mixer is "adaptive". This means that it only shows the channel strips which correspond to the instruments and audio objects, as they appear in in the Track List of the Arrange window. Under **View > Parameters**, you can hide/ show several button which define the type of channel strips and objects to be displayed or hidden. These are called the *Track Filters*. As an alternative to these Track Filters, you can choose the appropriate entries in the Track menu, in order to make the desired objects visible.

Global View and Local View

With version 5.0, the Track Mixer allows a **Global View**. Choose **Tracks > Global View** or press the **Global View** switch (if you've engaged **View > Parameter**). This allows you to see the entire "Mixing Desk", including MIDI tracks, not only the audio portion, as is the case with the Environment's Audio Mixer. The Global View shows all Instruments used in the song:

- All MIDI Instruments (Multi Instrument, Mapped Instrument) where the "Visible in Flipmenu" box is checked;
- All valid Audio Channels which exist in the environment;
- All valid virtual Audio Instruments which exist in the environment;
- All valid Audio Busses which exist in the environment;
- All valid Audio Outputs which exist in the environment;
- All valid Audio Masters which exist in the environment;
- All valid Audio Inputs which exist in the environment;
- All valid Audio Auxiliaries (TDM) which exist in the environment.

If set to **Global View**, the Track Mixer allows easy access to all available busses, outputs, tracks etc.

If **Global View** is *not* engaged (Local View), the Track Mixer only shows channels which correspond to those shown in the Track List of the Arrange window, or its current folder.

The Global View of the Track Mixer also shows all tracks which appear in the Automation Folder. Logic automatically creates, deletes and sorts all tracks in the Automation Folder.

Track Filter Switching Techniques

You can independently set the Track Filters for both Global and Local View modes. With ⌘ (Mac OS) held/right mouse button (PC), the Track Filter setting remains unchanged when switching **Global Mode** on and off.

- Clicking without a modifier on a disabled type will enable the view of this type and disable all other types.
- Clicking without a modifier on an already enabled type will enable all types. A further click goes back to a single enabled type.
- ⇧-Clicking allows you to individually enable and disable types, in any combination.
- Clicking with ⌘ (Mac OS)/right mouse button (PC) if **Global View** is disabled will activate the **Global View**, and enable the view of the current type.
- Clicking with ⌘ (Mac OS)/right mouse button (PC) if **Global View** is enabled simply disables **Global View**.
- Clicking with ⌥ (Mac OS)/ctrl (PC) locates the window to the first track of the current type. Filter and Global view modes remain unchanged—with one exception: If the clicked type is currently disabled, it will be enabled to ensure that the command was successful.

The Global View can display channels which do not exist in the current arrangement as tracks. If in Global View such a track has it's automation armed (Write, Touch, Latch mode), and data is recorded to it, a corresponding track will automatically be created, and selected in the Arrange window.

Navigating Specific Channel Strips

The **Track** menu offers the following entries:

- Goto MIDI Instruments
- Goto Audio Instruments
- Goto Audio Tracks
- Goto Audio Busses
- Goto Audio Outputs
- Goto Audio Aux
- Goto Audio Input

The window will scroll to the position of the desired track type. In cases where a selected track type doesn't exist in the current folder, Global View will automatically be enabled and the track type will be searched for.

The Track Mixer's MIDI Tracks

The rotary controls of the MIDI Instrument Tracks feature name fields for their parameter names, eliminating the need for legends. These knobs usually default to control over parameters like the **Chorus** and **Reverb** levels, or the **Freq** (Cutoff Frequency deviation) of the MIDI expander's low pass filters. Click-holding on these names will open a flip menu which allows you to assign any MIDI controller to the corresponding knob. This method of parameter definition is the same for all MIDI instruments. Under the **View** menu, you can add more controls, allowing the display of further parameters.

The program name fields of the MIDI tracks are blue, as with effect or Audio Instrument plug-ins. Clicking on them, with ⌥ (Mac OS)/ctrl (PC) held, allows the enabling/disabling of program change transmission. This parameter is also displayed as a checkbox in the **PRG** parameter field of the object parameter box (in the Arrange and Environment windows, along with the **MIDI Port** and **Cha** settings).

Removed Parameters

As Logic 5.0 features the new Track Automation system, certain automation parameters for the Track Mixer have been removed from the record button menu, and the Track Mixer and Environment **Options** menu(s). In addition, the legend previously found to the left has been removed, as these parameters have moved to dedicated knobs on the MIDI channels themselves.

Improved Navigation in the Mixer

Double clicking on a send or out assignment panel on any audio channel will "jump" to the allocated Bus/Output channel fader, making navigation in the Environment's audio layer faster. This functionality also works in the Track Mixer. Here, a new Environment window will launch and the destination object will be selected in the Environment's audio layer. Should an Environment window already be open, (with link mode active) this will be used and will update as described.

Peak/Clip Detector Display

Above the level meter each audio channel features a new peak/clip detector display. It shows signal peaks in dB—up to a maximum of +6dB. Above 0dB the highest peaks are hold until reset by a mouse click on the dB value. To achieve an optimal signal level lower the volume fader by exactly the displayed clip amount.

8.3 Audio Input Objects

Audio input objects are available for these audio hardware types:

- Mac AV (Mac OS)
- PC AV/EASI (Windows)
- Audiowerk
- Direct I/O
- StudI/O (Mac OS)

- Korg 1212 (Mac OS)
- EASI (Mac OS)
- ASIO

This new class of audio object represents the audio inputs of your audio hardware. You can open an audio input object this way:

1. Choose the Audio layer in the Environment, or alternately select **Audio > Audio Mixer**.
2. Select **New > Audio Object**.
3. Double-click on the newly-created audio object.
4. In the parameter box of this new audio object set the **Cha** parameter to **Input 1**.

The design of the audio input object is virtually identical to the other audio objects you are already familiar with. This makes the functionality of the audio input object self-explanatory. Effect plug-ins, such as Logic's reverbs and delays, also work while Logic is stopped. This allows the appealing prospect of inserting compressors into Logic's inputs, without the need for external compressors. It becomes even more enticing when using 24 Bit hardware. The plug-ins behave just like external hardware processors. Bus sends can be configured as either pre or post-fader.

The audio input object functionality could be used in place of your old mixing desk, or can save time while bouncing.

Please note that at the moment while recording (not while bouncing), the effects of audio input objects are not active and are thus not recorded. This will be changed in upcoming versions.

Audio input objects simplify bouncing

The audio input objects make the bouncing procedure much easier, as there is no longer any need to record the audio signals of your MIDI devices before their output can be bounced (mixed down) with other audio recordings. Thanks to the live audio input objects, you can mix down the audio signal from your external hardware MIDI synthesizers and sound modules into a stereo mix (by bouncing an output object).

8.4 Surround

Eight Channel Surround

Surround is available for these audio hardware types:

- TDM (Mac OS)
- PC AV (including EASI and EASI MME)
- Audiowerk
- Direct I/O
- StudI/O (Mac OS)
- Korg 1212 (Mac OS)
- EASI (Mac OS)
- ASIO.

Logic Platinum allows for mixdowns in surround format, with up to eight loudspeaker output channels. Surround formats supported include all common formats, including the most demanding, like the 7.1 format. To effectively use surround, the audio hardware and driver must support (and be equipped with) more than two outputs.

Encoding or decoding does not take place. The use of surround does not imply any manipulation of frequency response characteristics or delay treatment. The processing only affects the relative volume settings for the output pattern selected for that channel. It's up to you to insert plug-ins, such as delays or EQs, high or low pass filters etc., into the output objects serving as the surround channels.

All channels and busses can be set to different surround formats independently. Click-hold on the output parameter field of the bus objects (above the panorama control), and set the output to Surround in the flip menu. The pan control is replaced by a twodimensional surround control. The loudspeakers are represented by colored dots, and the pan position is indicated by a white dot that can be grabbed and moved.

Changing the surround settings of multiple channels simultaneously

You can change the surround settings of all selected channels simultaneously. Select all the objects you want to set to the same assignment, and hold [⌥] (Mac OS) or [ctrl] (Windows). If you do not hold the modifier key, Logic will ask you if this is your intention.

The Pan Window

Double-click the surround control which has replaced the pan control. The pan window will launch, giving you a closer look at the surround control. Grab the parameter field at the top of the window, and select the desired surround format—format 7.1, for example, which is designed for big cinema theaters. The seven loudspeaker symbols appear around the surround control. The routing of the signal to the speaker outputs is intuitively controlled by the position of the blue dot in the surround field, which can be dragged and dropped.

The flip menu above the surround panner allows the selection of the following surround types:

Stereo, LCR, Center only, Quadro, Pro Logic, Pro Logic w/o C., 5.1, 5.1 w/o Center, 7.1, 7.1 w/o Center, EX, EX w/o Center.

Remember that this mode can be set independently for each audio channel.

The routing of the signal to the loudspeaker channels takes place most intuitively by dragging the light blue dot in the direction of the dedicated speaker symbols. The movement of the surround position on a given axis and diversity is supported by these functions:

- Hold ⌘ (Mac OS)/Right Mouse Button (PC) to lock the angle.
- Hold ctrl (Mac OS)/alt (PC) to lock the diversity.
- Hold ⌥ (Mac OS)/ctrl (PC) to reset angle and diversity to the middle.

ℹ Move the light blue panorama dot and check out the modifier keys.

LFE: Subwoofer Effect Channel

The abbreviation stands for "Low Frequency Enhancement" or "Low Frequency Effects", as the LFE output most commonly leads to a subwoofer amp. The use of a subwoofer type speaker is not a must. Technically speaking, the LFE output is nothing more than an additional post fader auxiliary send, with the LFE control as its volume fader. If you only want low frequency signals to reach the output, insert a low pass filter plug-in into the output object defined as the LFE "subwoofer" output. A cutoff frequency of 120Hz is standard for surround applications. The LFE fader determines the amount of signal from the source audio to be routed to the LFE output. In the surround settings of the *Audio Hardware & Drivers* menu, you can define which output of your audio hardware will serve as the LFE output (see below: Assigning surround channels to audio outputs).

ℹ The LFE channel isn't meant to play back all low bass portions of a cinema soundtrack. It's not appropriate to route the kick drum from a film score to the LFE channel. These bass portions are played back by the front speakers: The PA system might utilize active crossovers and subwoofers to play back the bass, but it won't be adressed by a separate channel of the audio you produce with Logic. The LFE channel serves as an effect channel for particular low frequency (bass) events which occur in a handful of (action-) scenes in a movie, e.g. when the alien spaceship is landing.

Surround Automation

As with all of Logic's mixing functions, you can record, edit and play back every surround movement as per pan or volume. The Track Automation, described in the*Automation* section, on page 17, is recommended for the automation of surround movements. The following controller numbers are assigned to the surround parameters:

25 – Surr. Angle

26 – Surr. Diversity

27 – Surr. LFE

Using the Track Automation system, you won't ever have to deal with these Controller numbers.

Assigning Surround Channels to Audio Outputs

The assignment of surround channels to the outputs of your audio hardware is accomplished via **Audio > Audio Preferences > Audio > Surround**. The standard settings for the Pro Logic, 5.1, 7.1, and EX formats (each of which is available with or without center speaker), as well as Quadro, Center Only and LCR (Left-Central-Right) are pre-set, but can be edited.

Bouncing Surround Audio Files

In the **Audio > Audio Preferences > Audio > Surround** settings, you can see which file name extensions will be applied to the file names resulting from the *Bounce* procedure. Bouncing a surround mix will create more audio files than your usual stereophonic bounce/mixdown. These extensions are used to identify the files. By clicking on them, you can edit the extensions.

Select **Surround Bounce** in the normal bounce dialog window, which is launched when the bounce button at the bottom right of an output audio object is clicked. No matter which *Bnce* button you may have pressed, all outputs selected in the surround settings will be bounced simultaneously.

TDM users please note that a Surround Bounce requires a number of unused audio tracks equivalent to the number of channels of the Surround format (e.g. eight tracks for a 7.1 Surround Bounce).

Master Volume Fader

The master fader changes all output gains. It acts as a separate attenuator stage; The level relationships between output objects are not affected. This is very helpful as a proportional output volume control and is particularly useful for surround mode in Logic Platinum (fades of the complete surround mix).

The master fader is an audio object with the **Cha** parameter set to Master. Moving the master fader does not affect the position(s) of the output object sliders, but directly affects the signal levels. These level changes are reflected in each output channel's level meter.

8.5 TDM Systems

Mac

TDM systems are only supported by Logic Platinum for Mac OS.

More TDM Aux Objects

The maximum number of Aux objects has been increased from 32 to 64 (16 to 32 stereo) in TDM systems.

64 Buses

The maximum number of Buses has been increased to 64, which can be adressed by a maximum of 8 send slots per channel.

I/O Inserts in TDM systems

In Logic Platinum, TDM hardware users can insert external effect processors on audio tracks, inputs, auxiliary tracks, busses and outputs via Logic Platinum's I/O track insert objects.

Mono or stereo I/O inserts can be used on both mono and stereo track objects. To insert a stereo plug-in on a mono track hold ⌥.

The send level of an I/O insert can be adjusted on the main output objects in Logic, while the return input level has to be set on the external signal processing device.

As per internal ("native") plug-ins, TDM I/O inserts can now be bypassed by ⌥-clicking the TDM insert point on a channel. To avoid internal feedback loops: I/O inserts can not be placed on the same "Output" or "Input" object.

Send Levels in TDM Mixer raised

The send levels on track, input and aux channels are raised by 6dB.

This might change the mix levels of older songs recorded with Logic Platinum versions prior to 4.7.

AudioSuite-Plug-Ins

AudioSuite plug-ins can be used with DAE 5.0.

WAV Files under TDM/DAE

TDM/DAE is able to play back WAV format files. Please note that DAE 5.1 or higher is required. Also note that the following DAE limitations still apply: You can use either 24 or 16 bit files, and no stereo interleaved files.

Parallel Use of OMS and Logic's Proprietary MIDI Drivers

TDM plug-ins (for TDM hardware by Digidesign) and several virtual samplers and synthesizers (but not the EXS24 or the ES1) must be controlled via OMS (Open Music System). OMS and Logic's proprietary MIDI driver can be used in parallel. This provides the advantage of being able to control external MIDI Instruments with the timing precision of AMT (Active MIDI Transmission, only available with Emagic's AMT8 and

Unitor8 MkII MIDI Interfaces), which would not be possible in pure OMS mode.

Select **Options > Settings > MIDI Interface Communication > Use OMS in addition to the built-in MIDI driver**. If enabled, Logic will use the built-in MIDI driver and OMS simultaneously. Everything works very much like using just the built-in MIDI driver. However, the difference is that you have a choice for each Environment instrument:

1. If an OMS instrument is *not* set ("OMS unassigned"), the object sends MIDI information "normally" to the output port.
2. If an OMS instrument *is* set, the object sends the MIDI information to the OMS instrument.

Please note:

- The OMS output mapping is obviously not active because it is not neccessary: the object port setting works normally.
- The OMS inputs work in addition to the Logic MIDI inputs.
- Logic initializes the built-in MIDI driver first, and then OMS. This may be significant in cases where both drivers want to make use of the same resources—like a UNITOR USB driver. Whichever is first will win, so in this case the built-in MIDI driver will get access to the Unitor. Please remove the OMS Unitor driver from the OMS driver folder should you see a "Could not use the Unitor driver" message from OMS.

Re: Emagic TDM Plug-ins

Due to technical reasons, the **Auto Filter** and **Tape Delay** plug-ins, available to users of TDM hardware in Logic Audio Platinum 4.x versions, can *not* be used with Logic Platinum 5.

Chapter 9

Audio in General

9.1 New Audio File Selector Window

In the Windows version, the file selector which is opened via **Audio Window > Audio File > Add Audio File** now features a **Play** button which plays back the selected audio file. The **Auto Play** function automatically plays back every audio file as it is clicked. A Recent Directories flip menu allows for accelerated access to your audio files. There are two different versions of the File Selector. One is launched when [⇧]-right-clicking with the pencil tool alongside an audio track in the Arrange Window (or with the menu command: **Audio > Import Audio File**), and the other when the **Add File menu** is accessed while in the Audio window. The version opened while in the Audio window is more advanced and allows multiple files to be imported or removed. The **Show/Hide Path** switch allows the drive and folder that contains the audio files to be visible/invisible.

9.2 Audio File Format Selectable

Mac

Under **Audio > Audio Preferences**, the **Recording File Type** can be defined. SDII is the default; Other choices are AIFF and WAV.

9.3 Update Arrange Position

When **Edit > Update Arrange Postion** is engaged, a drag of the region start in the Sample Editor also affects the position(s) of the region in the Arrange window, provided that the anchor is dragged as well. This ensures that the position of a given audio region stays at the same place in the arrangement. When moving the region startpoint back, the remaining portion of the region will start playback at a later position.

Such edits have always been possible with Logic—but you needed to grab the start of the region in the Arrange window. With **Update Arrange Postion** switched on, you can make these edits more precisely in the Sample Editor, without changing the position of audio in the Arrange window. This is provided that the region start and anchor are set to the beginning of the audio file (so that a dragging of the region startpoint also drags the anchor).

This is the most common scenario: Anchor and region start are always at the beginning of the audio file, whenever you record audio or add any audio file. Moving the region startpoint also moves the anchor, as the anchor can't be placed before the region startpoint.

- If **Update Arrange Postion** is engaged, moving the anchor won't change the audible result. The movement of the anchor in the Sample Editor and the new position in the Arrange auto-compensate. Moving the region start in the Sample Editor will lead to the same result as dragging the region startpoint in the Arrange window. This is the default setting.
- If **Update Arrange Postion** is disabled, a movement of the anchor changes the position of audio regions in the Arrange window. This is how Logic always behaved prior to version 5.

9.4 ReCycle Import

ReCycle is the name of a software application from Propellerhead Software. It mainly serves as an editing and production tool for loops (repeatedly looped audio samples). ReCycle uses specific file formats which can be imported by Logic. ReCycle/REX files also can be converted into EXS sampler instruments!

This is described under *Converting ReCycle/REX Files into EXS24 Instruments* section, on page 145.

ReCycle/REX file import requires an installed REX Shared Library system extension—version 1.2.1 (build 45) or higher (Mac) or REX Shared Library DLL version 1.2.1.43 or higher (PC). This system extension is installed by the Logic 5 installer. To download the latest version of the library visit the support section of **www.propellerheads.se**

Recognised ReCycle file formats

Logic can import the following file types:

- Old ReCycle File
  - File Suffix: *.rcy*
  - Abbreviation: RCSO
- Old ReCycle export file
  - File Suffix: *.rex*
  - Abbreviation: REX
- New (2.0) ReCycle file
  - File Suffix: *.rx2*
  - Abbreviation: REX2

Importing ReCycle/REX files into the Arrange Window

To import a ReCycle/REX file to an audio track select **Audio > Import Audio File** and choose a ReCycle/REX file instead of a regular audio file.

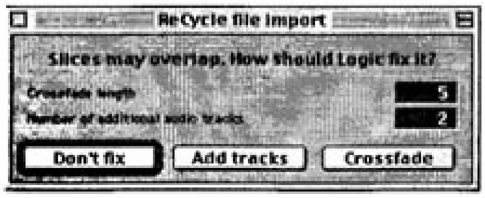
With the Pencil Tool ⇧-right-click (Windows) or Mac ⌘⇧-click (Mac OS, respectively) on an audio track and select a ReCycle/REX file.

In the info line of the open dialog window the properties of the selected file are displayed. You will get information about: file type, mono/stereo, sample rate, number of slices, beat/bar divisions, preview tempo in ReCycle, original tempo of the ReCycle/REX file.

Alternatively you can drag a ReCycle/REX file from the Windows Explorer/Macintosh Finder onto an audio track.

Import Overlap Function

On occasion, you may encounter the following dialog, when importing ReCycle/REX files.



The launching of this window results from:

- The song tempo used in Logic being higher than that of the imported ReCycle/REX file. The audio regions generated during import overlap once aligned in the bar/beat grid of Logic.
- Use of the Stretch function on audio slices in ReCycle without adjusting the corresponding song tempo in Logic.

You may handle the imported file in the following ways:

Don't fix: Imports the ReCycle/Rex file as is. All audio slices are placed as individual audio regions on one audio track. "Slices" of the file may overlap, to match Logic's tempo. In this case playback of an audio region is interrupted in favor of the following overlapping region.

Crossfade: All audio slices are placed as individual audio regions on one audio track. In areas where the regions overlap Logic 5 performs a non-destructive audio crossfade. The crossfade is equal powered. The length of the crossfade is determined by the value (shown in milliseconds) displayed in the **Crossfade length** panel.

Add tracks: will distribute the "slices" as audio regions across multiple audio tracks. Each overlapping audio region is placed on an additional audio track in Logic 5. This ensures playback of all audio regions in the overlapping area without interruption as it were the case using a single track (see above). Logic 5 automatically calculates the necessary additional track and displays them in the field: **Number of additional audio tracks**. You can also enter a value manually. For minor tempo deviations one additional audio track is usually enough. For major deviations up to 16 audio tracks can be added.

The adjustment of the numerical values is via use of the mouse as a slider, or double-clicking on the panel(s), and direct numerical entry.

Once all settings are made, Logic will create a folder containing one audio track (using the same Audio Mixer object as the track it is created on), which in turn contains the slices of the ReCycle/REX file at the appropriate positions. Each of these "slices" will reference the same audio file, which can be found in the Audio window, as usual.

Copying Audio Data Into the Clipboard

You can copy audio data into the clipboard for further use in ReCycle by selecting **Audio > Copy ReCycle Loop** (on the Macintosh: in the Arrange window's local *Audio* menu). This option will only be active if you have selected *one* audio part in the Arrange window, and will copy the selected region of the referenced audio file into the clipboard. You can then use the "*Paste as new document*" function in ReCycle to create a new ReCycle document using this audio data. In conjunction with the "*Paste ReCycle Loop*" function (see below), you can perform a complete data transfer from Logic to ReCycle, and back, without using any file selection dialogs!

The size of the transferred audio region is limited to 10 MB, due to the fact that the clipboard resides in the computers RAM. 10 MB, however, should be sufficient for most ReCycle loops (e.g., an 8 bar stereo loop in 4/4 at 70 bpm uses approximately 5 MB).

Pasting ReCycle loops into the Arrange Window

If you use the "*Copy Loop*" option in ReCycle to copy the loop currently being edited into your computers clipboard, you can paste this loop into the Arrange window by selecting **Audio >**

Paste ReCycle Loop. The result will be the same as if a ReCycle/REX file had been imported.

Make sure to deactivate the ReCyle option: **Transmit as one sample.** Note: Should after importing a ReCycle loop show up as one single audio region in Logic is this an indication that the ReCyle option: **Transmit as one sample** is still active.

9.5 OMF and Open TL Import and Export

Logic Platinum has the ability to export and import song contents using the OMF (Open Media Framework, also known as OMFI—Open Media Framework Interchange) and OpenTL (Open Track List) file formats. Typically, the OMF format is used for data exchange with Digidesign ProTools software, and the OpenTL format for data exchange with Tascam hard disk recorders, such as the MX2424.

Both of these file formats only support the exchange of audio data (audio media and the usage of this audio media in a song)—MIDI and automation data will simply be ignored when using the export functions.

OMF Export

To export the current song as an OMF file, select **File > Export > Export Song as OMF file.** This will launch a dialog with the following options:

OMF File Version

This allows you to export the file in OMF Version 1 or Version 2 format. You will usually choose Version 2 here. Version 1 is only useful for backwards compatibility with older OMF capable software.

Include Audio

If this box is checked, all exported audio files will be integrated into the OMF file itself (Note: This can result in a very large file, and can make file export times very long). If unchecked, only file references are written to the OMF file. If this method is used and you wish to copy the OMF file to another hard disk, you need to ensure that all referenced audio files are also copied.

Convert interleaved to split stereo

As some applications don't support interleaved stereo files, Logic can convert interleaved stereo files to split stereo. (e.g. when exporting an OMF file for a ProTools session) Check this box if this is the desired behavior. Note that this only works if the **Include Audio** option is checked.

Pan tracks

If this box is checked, Logic adds information to the OMF file which sets the panning of the exported tracks as they appear in Logic itself. (i.e., Stereo tracks in **Universal Track Mode** are exported as two mono tracks. Assuming this option is checked, the left mono track is panned to the left and the right one, to the right). Caution: This does not work with the conversion Software OMF Tool by Digidesign. Don't check this box if you are using OMF Tool to convert the OMF file into a ProTools session as it is not supported by OMF Tool.

Executing Export…

After pressing the **OK** button, a file dialog allows you to specify the destination folder of the OMF file to be generated. Choose a location and you're done. If you have checked the **Include Audio** option, Logic will display a progress bar for each exported audio file.

OMF Import

The **Import OMF/OpenTL file to song...** menu option will open a file selector, which allows you to choose OMF files (on the PC, files with the ".OMF" extension, on the Mac, "OMFT" file types are also supported). After file selection, a further dialog allows you to choose the location for audio data extracted from the OMF file. After confirming this dialog, the OMF file is imported into the currently open song (i.e. new audio tracks are generated for every track contained in the OMF file and the audio parts are placed appropriately on these tracks).

OpenTL Export

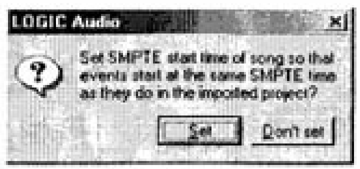
To export the current song's audio in the OpenTL format, select **File > Export > Export Song as OpenTL File**. This will launch a file selection dialog which allows you to specify the destination folder, and file name, of the resulting OpenTL file. After confirming this dialog, Logic will ask you if the song's SMPTE start time should be added to event start positions. In this scenario, audio parts in the resulting OpenTL file will have the same SMPTE start time as in the Logic Song. If you choose Don't add, the part positions are calculated relatively to the song start (i.e. an audio part which starts at Bar 1 will have a start time of 00:00:00:00:00).

At the next step, Logic will ask if a copy of all audio files should be made. You should use this option if you want to copy the resulting OpenTL file, including all audio data, to another hard disk. Logic will then create two sub-folders in the folder which contains the OpenTL file: namely "Audio Files" and "Track Files" (which is the suggested structure for OpenTL exports). If you choose that no copy is to be made, the file references in the OpenTL file will point to the original audio files.

Alternatively, you can have all exported files placed in one folder, which is named after the exported file. This is the convention used by the Tascam MX recorders, so it is recommended that *Create* be selected in response to the "Create a dedicated folder for OpenTL export files?" question.

OpenTL Import

To import an OpenTL file, the **Import OMF/OpenTL file to song...** menu option is used. Files with the extension ".TL" will be shown in the file selector. Following file selection, Logic may launch the following dialog (only if the first event in the imported song starts at a time greater than 1 hour):



Confirm by pressing **Set** if you want the SMPTE start time of the song to be changed accordingly, or **Don't set** if the song's SMPTE start time should not be changed.

Should the first audio part found by the OpenTL import have a start time greater than zero (less than zero is impossible), Logic will ask if you wish to: "Set the start time of first imported event to start of song?" If you answer with **Set**, the first imported audio part will be placed precisely at the song start point.

This solves the problem that may occur if the song on the Tascam machine starts at a large time value, but less than 1 hour (such as 59 minutes). In this situation, the user can't immediately see the imported audio parts because they have been imported beyond the song's end point (possibly even beyond the largest possible song end point, depending on the song tempo).

9.6 POW-r Dithering

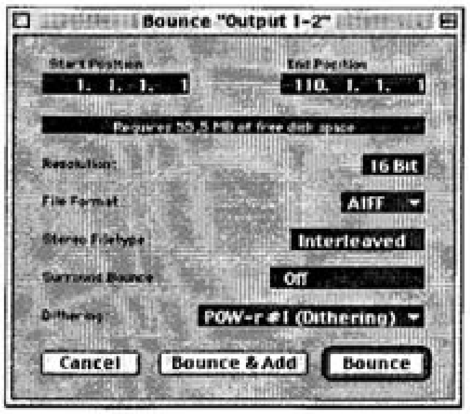
Logic 5.0 operates internally at 32-bit floating point resolution. However, in the Bounce window the **Resolution** of files generated by the bounce process can be set to 16 Bit. In this case Logic 5 needs to recalculate its high internal resolution to create 16-bit bounce files. This reduction in bit resolution can produce unwanted digital artefacts such as quantization noise, which is perceived as a "graininess" in the signal. This type of artefact is often seen in conjunction with audio material featuring a wide dynamic range.

To counteract these types of side effects, Logic 5 features **POW-r** (Psychoacoustically Optimized Wordlength Reduction) dithering during the **Bounce** process. POW-r is licensed from the development team of the 'POW-R Consortium LLC'. (www.mil-media.com/pow-r.html). POW-r offers three distinct dithering modes which can be accessed via the **Bounce** window. The selected POW-r mode is written into the audio files.

Generally speaking, dithering the same audio file multiple times should be avoided. Dithering should only be applied once—as the last step in the final mix. Therefor it should be turned off—**Dithering**: None—if you plan to use bounced files in Logic as it would be the case with the function **Bounce&Add** for example.

An in-depth describtion of the Bounce function can be found in the Logic Reference Manual.

The Bounce funktion is accessed via the Environment Audio Mixer or via the Track Mixer (in Global view) by pressing the Bounce (**Bnce**) button of an Output Object.



There are four menu options:

- **None**: No dithering is applied.
- **POW-r #1**: uses a special dithering curve to minimize quantization noise.
- **POW-r #2 (Noise Shaping)**: uses additional noise shaping over a wide frequency range which can extend the dynamic range by 5 to 10dB.
- **POW-r #3 (Noise Shaping)**: uses additional, optimized noise shaping which can extend the dynamic range by 20dB within the 2 to 4kHz range—the range the human ear is most sensitive to.

Note: Noise Shaping minimizes side effects caused by bit reduction by spectrally displacing the quantization noise above 10kHz—the range the human ear is least sensitive to.

The mode which will sound the best to you is primarily dependent on the program material and on your personal taste. The "right" setting is best determined by auditioning the audio material with each of the dithering modes.

9.7 Audio Drivers

The Audio Driver is the software which facilitates communication between Logic and the audio hardware.

New Audio Engine (Windows)

All Windows versions of Logic (starting with version 4.5) now feature a new, significantly improved audio engine that has much more in common with the Macintosh version. It offers shorter latency times, faster fader response, improved VST support and better overall performance. Monitoring, that is, listening to the input signal at the audio output, is now available for all driver types in Logic for Windows. There are six driver types, in total, available for use; you will find these on the **Audio Driver** and **Audio Driver 2** panels, found via the **Audio > Audio Hardware & Drivers** menu.

Changes from Former Logic Windows Versions

In former versions, there were four categories of hardware types: *PC AV*, *Roland VS*, *DS 2416* and *Audiowerk*. *PC AV* was used to support *EASI*, *ASIO*, *MME* and *Direct Sound*. The integration of the *DS2416* took place via an *MME* driver in *PC AV*.

Now we have *PC AV*, *Roland VS*, *Audiowerk*, *Direct I/O*, *DS 2416* and *ASIO*. *PC AV* now uses *EASI* (Emagic Audio Streaming Interface), with support for MME (WDM) and Direct Sound devices via *EASI MME* and *EASI DS*, which replace the former *MME* and *DirectSound* support. These are accessible in the **Driver** parameter field in the *PC AV* area of the **Audio > Audio Hardware and Drivers** menu page.

There are separate drivers for Audiowerk, Direct I/O and ASIO, and also a dedicated driver for the integration and remote control of Roland VS-machines (VS-880, VS-880 EX and VS-1680). The direct connection to Digidesign's I/O interface is new. This affects the Audiomedia III and Digi 001 hardware.

ASIO drivers are now a separate category. The Yamaha DSP factory is now supported via the ASIO driver. Assign it in the ASIO driver field, but do not activate the ASIO selection box. You should only activate the DS 2416 checkbox, if using this card. Adjustment to its parameters are made in the ASIO fields. Former versions used the PC AV driver for the DSP factory, which is no longer possible. With the ASIO driver, you can use plug-in synthesizers and other virtual instruments with the DSP Factory (DS2416).

New Audio Engine Parameters for Windows Versions

These parameters appear in the Mac OS versions too, but are already well-known there.

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 this zipper noise.

Max. Number of Audiotracks

The Audio Engine requires free system memory. 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 audio hardware types simultaneously.

Larger Disk Buffer

This option influences the amount of audio data that is read from the disk in advance. This option is switched off by default, matching the demands of fast hard drives and powerful computers. If you receive frequent error messages while running Logic in this mode, you should switch this setting on, allowing the playback of more tracks and achieving higher reliability. More RAM is needed in this case, however.

Larger Process Buffer

This parameter determines the size of the native buffer used to compute mixes and effects. Do not activate this option if you own a fast computer. This reduces response times to operations such as volume changes or soloing. Experiment to find the setting that coaxes the best performance from your system.

Hints Concerning Using Various Hardware Types with the New Windows Audio Engine

DirectShow Plug-in Integration

Due to several technical changes, some Direct Show plug-ins may not appear. If so, exit Logic, and start the latest version of the Plug-In Enabler (PIE) appplet in order to re-activate the deactivated plug-ins—assuming that these worked correctly in earlier versions.

VST Plug-in Integration

VST plug-ins are not handled by the PIE in the current version of Logic. The PIE is only responsible for Direct Show plug-ins. In order to use VST plug-ins in Logic, these plug-ins must be placed in the "Vstplugins" folder, which must be located in the Logic folder. In order to prevent particular VST plug-ins from appearing in Logic's plug-in list, move the plug-in file to another folder, which could be named "Vst unused", for example.

EASI MME

Sound cards that only have an MME driver, can be used through utilization of the EASI MME driver supplied by Logic. In order to configure it, please start the "EASIMME.exe" applet, found in the Logic folder. More information about its use is supplied by the EASI MME HELP file. In order to use the desired configuration with Logic, select the EASI MME driver in the flip menu of the PC AV section in the **Audio > Audio Hardware & Drivers** page.

EASI DirectSound

If you want to use a DirectSound (stereo) driver for your sound card in Logic, please select the "EASI DirectSound" driver in Logic's **Audio > Audio Hardware & Drivers** page. In order to configure it, please start the "EASIMME.exe" applet and create a virtual device containing the appropriate DirectSound driver. More information about this is supplied by the EASI MME HELP file.

Yamaha DSP Factory

In order to directly access Yamaha's DS2416, activate the DS2416 option on Logic's second audio hardware configuration page. The I/O handling is no longer controlled by MME, but by ASIO. Settings like the clock source need to be set in the ASIO section, directly under the DS 2416 option.

Yamaha ASIO drivers (version 2.4 or higher) are available in 16 Bit and 32 Bit versions. The 32 Bit driver allows for the configuration of latency values between 500 and 3072 samples. In order to use the DS2416 in combination with an SW 1000 sound card, the support of the SW 1000 in the ASIO driver of the DSP Factory must be switched OFF! In addition, the clock source parameter in the ASIO section must be set to "SI" (serial in).

Direct I/O

In order to directly access Digidesign hardware via the *Direct I/O* driver interface, select the Direct I/O checkbox in the second audio hardware configuration page (**Audio > Audio Hardware & Drivers**). A Direct I/O driver must be installed on the system in order to do so (Wave Driver version 1.6 or higher). For further information, please contact Digidesign.

Automatic Conversion of Compatible Song Audio Configurations

If you load a song that was created with, say, PC AV or Mac AV, running on a machine currently using an ASIO device (for instance), Logic will ask if you wish to switch from PC AV or Mac AV to ASIO. This allows the song to be played back immediately, without the need to manually reassign all used audio objects (tracks, busses, Audio Instruments etc.). Any plug-in effects used will be recognized, and will retain all of their settings.

Software Monitoring

The **Monitoring** switch found in **Audio > Audio Hardware & Drivers > DRIVER** (DRIVER denotes the audio hardware/driver you use) has been renamed to **Software Monitoring, indicating** its increased functionality. With **Software Monitoring** engaged, audio monitoring is performed by Logic. With **Software Monitoring** disengaged, the audio card and its driver is responsible for monitoring. E.g., if your audio hardware supports ASIO2.0 *zero latency monitoring* and you want to use this feature, disable **Software Monitoring**. Please note that Logic's Audio Input Objects are only fully functional (e.g. use their inserts and bus sends for live effects) when **Software Monitoring** is enabled. With **Software Monitoring** disabled, they will only deliver the monitor signal coming from the audio hardware/driver.

EASI MME Expanded

The EASI MME Extension 1.0.0.3 now supports two further formats:

- A sample rate of 96kHz (provided the driver and the hardware offer this feature);
- Support of WDM drivers when using Windows Me, Windows 2000 or Windows XP.

EASI for the Mac OS Versions

Mac

Emagic's standard for the integration of audio hardware, EASI, is now also available for Logic Mac versions. Soundcard manufacturers now have a driver interface that enables their audio hardware (e.g. PCI soundcards) to be easily adapted for the Macintosh.

Improved Latency with Audiowerk

Mac

The Audiowerk's **I/O Buffer Size** driver parameter can be set to smaller values. This reduces the latency for monitoring and for the use of virtual instruments, such as the ES1 synthesizer or EXS24 sampler plug-ins. Note that reducing this value will place a higher strain on the CPU, and can lead to audio glitches and/or a reduced playback track count.

Yamaha DSP Factory

Yamaha DSP Factory support has been improved in both the Macintosh and Windows versions of Logic. Logic Gold/Platinum for Macintosh (and Windows) now allow the use of virtual instruments with the DSP Factory. 16 Audio Instrument channels are available. It should be noted that only native plug-ins can be used on instrument channels. Sends to the DS2416 are not possible. The "Output" of an instrument channel is an audio track pair. All instrument outputs are set to "Track 15-16" by default. It is still possible to have audio files on these tracks, but the signal is mixed with the output of the instruments.

Please note that Yamaha drivers for the DS2416 card are available for download from their website. Their newer ASIO drivers allow the **I/O Buffer Size** to be reduced to 512 samples. The default is 2048, as with older ASIO drivers.

Roland VS Support

Logic supports the Roland VS-880, VS-880EX and VS-1680. The combination of Logic and a Roland VS can't be compared with conventional audio hardware. One advantage of the Roland VS, however, is that it can be used as a standalone system.

Please don't assume that the Roland VS-Series represent an alternative to other audio hardware for Logic—this wouldn't be fair on the successful VS series, or Logic. Logics VS implementation is intended to facilitate the cooperation between musicians working with both systems, and for transferring home-recorded pre productions into professional studio environments.

MIDI Cabling

For Logic to communicate with the VS, both MIDI in and out must be connected.

- Connect the MIDI out of your VS unit to the MIDI in of your MIDI interface.
- Connect the MIDI out of your interface to the MIDI in of your VS unit.

MIDI Settings on Your VS

SysEx Communication

Power up your VS, and after it launches and initializes, make sure that:

- SysEx Transmit is set to ON (*SysEx.Tx = On*) and
- SysEx Receive is set to ON (*SysEx.Rx = On*).

You can access these parameters via "*System*" then "*MIDI*" on your Roland VS. Use the "*Parameter*" keys to select the settings, and the dial to adjust them. Refer to the users manual of your VS, if you can't find these parameters.

The VS stores these settings with each song. This means that when changing to a different song in the VS, you may need to check the above again.

MTC & MMC Parameters of the VS

The VS and Logic are synchronized via MTC (MIDI Time Code).

- Set the VS to be MMC slave. (*MMC = Slave*)
- Set the VS to transmit MTC (*Gen. = MTC*).

This parameter is accessed under "*Sync/Tempo*".

Logic Settings

Start Logic. Select **Audio > Audio Hardware and Drivers.** Choose **VS** and select your Roland VS unit from the list.

If Logic can't find a VS unit you will see a warning message. In this situation, please re-verify the MIDI settings on the VS unit.

MIDI Machine Control (MMC)

In the main menu, select **File > New.** Set Logic to transmit MMC commands. In order to do so, click-hold the *Sync* button on the Transport window, and check MIDI Machine Control (MMC).

If you wish, you can save this newly created song, along with any other desired settings. If saved with the name "Autoload" in the Logic program folder, this song will automatically open when Logic is launched.

Tempo Changes on the VS

Never change the song tempo in Logic, as this will probably cause the regions to be played out of sync. Do not use Logic's *Tempo List* or other tempo functions! Any desired tempo changes should be made on the VS unit. Logic always reads the tempo exclusively from the VS.

Playlist

When an existing song is loaded, the VS audio regions found in this Logic song are deleted, and the Arrange will display nothing on the audio tracks. This is normal, as the playlist is always stored in the VS unit itself.

Retrieving the Current VS Playlist

Define a Key Command for the retrieve-playlist function. This only needs to be done once, and Logic will remember it:

- Open the Key Commands window from the global **Windows** menu.
- Click *Find* and type "**VS**" (without the "").
- You should see the *Get complete playlist from VS* Key Command.
- Click on it.
- Enable **Learn Key**.
- Define whatever key you want by pressing it once. (A Function key may be a good choice here).
- Close the Key Commands window.

This Key Command works when the Arrange window is the top window. When you press the defined key, the VS will be asked to send its playlist to Logic.

When you press your defined key, Logic will request the following from the VS:

- The tempo or tempo list of the currently selected song in the VS;
- The frame rate of the currently selected song in the VS;
- The whole playlist of the currently selected song in the VS. Logic will read the playlist of all 8 (or 16 with a VS-1680) tracks of the currently set V-Tracks.

This will take a while. After this is done, you should see all VS tracks in the Arrange window.

Waveform Overview

You will now notice a small window labeled "Creating overview". It may take a while to finish this for all audio tracks, depending on the length of your VS song. Creating the overviews for an "average" song takes around 10 minutes.

You may continue working while Logic is still requesting the overviews. Logic automatically stops reading them when you start playback, and continues reading when you stop playback.

The rate at which the overview data can be transferred is determined by factors outside of Logic's control.

The overviews are saved on your computer's hard drive after they have been completely read. This means that when you load this VS song into Logic in the future, the overviews do not have to be recreated. Logic creates a directory called "(VS-Files)" on the partition of your hard drive where the Operating System folder is located. A folder within this directory for each VS song is created. Logic manages this folder automatically.

Refresh Overviews

If you later find that overview data stored on the hard drive does not match what you hear when playing:

- In the main menu, select **Windows > Open Audio**.
- Select the track files you wish recreate the overview(s) for.
- In the Audio window, select **File > Refresh overview(s)**.

The overviews may be displayed incorrectly if you work on the VS without Logic, and later work with Logic. There is no way for Logic to know what has changed in such cases, unless you subsequently refresh the overviews.

Editing in the Arrange Window

Editing works on a Cut/Copy/Paste principle.

The Following is possible:

- Moving a region (simply click-hold and move),
- Copying a region—click-hold and move while holding ⌥ (Mac OS) or ctrl (PC) respectively.

Performing any of the above edits will automatically send a command via MIDI to the VS, which will then perform the operation. Please note that editing multiple regions simultaneously will take some time to complete in the VS. A small window will show the status of the edit with a progress bar.

If you initiate too many commands at once you may hit the *Get complete playlist* Key Command to abort. The commands not yet completed are skipped, and those already completed are reflected after the playlist is loaded.

The following operations are not possible:

- Changing the length of a region;
- Grabbing the left or the right corner of a region to change its start or end point.

If you move a region in the Arrange window so that it overlaps another region, the overlapping data from the end of the earlier region will be removed. This cannot be subsequently undone. If you accidentally do this, and wish to recover, the only recourse would be to reload the playlist from the VS. This would, of course, discard any other changes made in Logic since the song was last saved on the VS.

The same applies if you copy a region so that it overlaps another region. The VS does not offer realtime non-destructive file/region based editing: What is gone is gone, so be careful when editing. The following cannot be undone in Logic:

It is not possible to delete a region and to undo this. The only way to get it back is to re-load the song in the VS and perform the *Get complete Playlist* function in Logic again.

It's highly recommended that you undo audio editing mistakes directly on the Roland VS unit, rather than in Logic. Use the *Get complete Playlist* function after the undo operation.

Saving Changes in the VS

Unless you save the song in the VS, your changes are not stored. You can go back to the previous status by reloading the song in the VS and selecting *Get complete Playlist from VS* in Logic again.

The song in the VS can not be saved remotely by Logic. Saving can only be done from the VS' front panel.

Saving Changes to the Overview Files

When you select *Get the current playlist* and you have made edits in the Arrange window, Logic will ask you to save the overview data which also has changed. You should respond with:

Yes, when you are also saving the song in the VS.

No, when you have reloaded a previously saved version of your VS song into the VS.

No, when you have loaded a different song into the VS and you have NOT previously saved the changed song in the VS.

Changing a Song in the VS

After changing a song in the VS, you will need to *Get the complete Playlist from the VS* before making any kind of changes to the arrangement inside Logic.

Effects Control from Logic

In the Track Mixer, accessed via **Windows > Open Mixer**, you'll find two return objects, called *Effect 1* and *Effect 2*. Below the **Inserts text on the channel strips, you will see** a blue field. This field allows you to select one of the effects available in the VS effect processors. Click-hold to select the effect from a list. Once the desired effect is selected, double-click the blue insert panel to open an edit window for the effect's parameters. Each audio channel strip has two Sends (*Eff1* and *Eff 2*, both of them 'post fader'), which define the amount of effect added to each of the channels.

V-Tracks

Logic always reads the playlist from the currently active V-Track in the VS. When you want to see, hear and edit a different V-Track, set this in the VS and *get the playlist* again.

The Regions are named in this manner: "TrackA1-1" for Bank A, Track 1 and V-Track 1; "TrackA4-7" for Bank A, Track 4 and V-Track 7.

V-Track Banks (VS-880EX only)

Currently, only Bank A is read by Logic.

Digidesign Digi001

Emagic has optimized its support of Direct I/O for the Digi001. To make use of the Digi001 support, *DigiSystem INIT 5.0* is required.

Logic Platinum allows the playback of up to 96 tracks with the Digi001, dependent on the performance of your computer and your hard-drives. All inputs and outputs of the Digi001 can be addressed. It is possible to use Split Stereo, Interleaved and Mono files simultaneously. Logic's Audio Engine allows the simultaneous playback of 16 and 24 bit files.

In Logic you can utilize the internal effects, external plug-ins in VST1 and VST2 formats and all virtual instruments for playback via the Digi001.

In order to use all 18 in and outputs of the Digi001, check **Audio > Audio hardware and drivers > Direct I/O > Use 16 (or more) ins and outs.**

When first launched after a new installation, Logic checks all supported hardware drivers. Due to technical reasons it looks for the presence of a Digidesign TDM system first. Starting "Launch DAE" will lead to conflicts with Digi001. Please ensure that DAE is not active, and that **Audio > Audio Hardware & Drivers > Direct I/O** is engaged, instead.

General Remarks on Digidesign Direct I/O

On Mac OS with *DigiSystem INIT(TM) 5.0* and Windows with the *Digidesign Wavedriver*, you can set the **I/O Buffer Size** to .125, which reduces the latency. This makes the Digidesign device suitable for input monitoring and for real-time playing of virtual instruments. The downside of this is that the CPU load is increased, so it may not be possible to play back as many audio tracks as with larger buffer sizes.

For the Audiomedia III card, 0.5 is still the smallest possible value for the I/O buffer size.

Audio > Audio Hardware & Drivers > Direct I/O > Hardware Setup allows you to set various parameters available for the different Digidesign cards, such as **Clock Source** and **Input Gain.** The global *Audio Hardware Setup* Key Command directly opens this window.

DigiSystem (TM) INIT 5.0 cannot be used with older DAE versions; If you also address your Digidesign Hardware via DAE, please retain the corresponding *DigiSystem INIT.*

Mac

ASIO Clock Source

Some ASIO Drivers allow you to switch the synchronization source for the audio sample rate. If a driver offers this choice, you can select it via **Audio > Audio-Hardware and Drivers > ASIO Clock Source.** In cases where there is no choice, you'll see a "----" or a read-only text entry in the flip menu field.

Chapter 10

Improvements

This chapter deals with additional improvements, changes and miscellaneous information on the Logic product range.

10.1 Audio-related Changes and Enhancements

- Windows: Song-specific configurations pertaining to Direct-Show plug-ins are now handled better.
- Platinum PC: PC AV: ReWire labels now show correctly.
- Platinum Mac OS: Third party TDM plug-ins: Due to the nature of some third party TDM plug-ins, their settings would not be restored after opening a song. This has been fixed.
- ES1: Further increases in ES1 performance. It is now available as a mono or stereo Audio Instrument and features a new chorus/ensemble effect. The chorus/ensemble effect can be switched into the ES1 output via three presets. All parameters are now visible on the slightly modified graphic user interface.
- When Creating a new Audio object, the default for "Val as" is now "dB".
- Windows: Noises that occurred (on some systems) before Audio Regions processed via DirectShow plug-ins have been removed.
- In the Sample editor, **Edit > Insert Loop** has been removed. **Edit > Selection to Loop** offers the same functionality.

- The Chase Events feature, which allows Logic to react to sustain pedal, other controller and pitch bender messages which lie before the start of playback, also works with Audio Instruments.

10.2 General

- Disconnection and reconnection of a USB Unitor Family interface while Logic is running is improved.
- The synchronization of video playback to sequencer clock has been improved.
- The transmission of MIDI Time Code (MTC) under heavy performance load has been improved.
- The Spanish localization has been further enhanced.
- DirectShow plug-ins (Windows) now support 48kHz audio files in Logic 5.
- The combination of Logic and the Logitech MouseWare Extension has been improved.
- The OMS connection for MIDI-driven TDM plug-ins (e.g. the Access Virus TDM) has been improved.
- Further refinements of the OMS implementation.
- The MIDI burst behavior with USB MIDI interfaces has been vastly improved.
- ESB TDM: A specific synchronization problem between DAE/TDM and Direct TDM has been resolved.

Improved Handling of Windows MIDI Drivers

If a MIDI driver crashes when sending SysEx or opening the port, Logic will notify you that the driver has been disabled in the next session. You can re-activate it in Logic after you have updated the driver. Please contact the driver's manufacturer for an updated version, if required.

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If a MIDI driver cannot be opened, the message now contains a button to disable it permanently. To reactivate a deactivated driver, open the "win.ini" file with a suitable editor. In the [Logic] section you will find a list of all MIDI drivers recognized by Logic. Following the driver name you will find a number. Set this number to "1" (activated) or "0" (deactived), according to your needs (Windows).

The default MIDI input buffer size for non-Emagic drivers has changed from 32kB to 60kB. This increase improves the handling of large sysex dumps (Windows).

10.3 Preferences

The **Global Preferences** window has been slightly reworked. The parameters are grouped more logically. All Automation settings are saved in the preference file.

"Warning" Sound Levels

Global Preferences > Disable Notification Sounds allows for the disabling of alert sounds, such as those heard during error messages or the "Autolink Success" flop sound. If enabled, all Sound Labels are much lower in volume than in previous versions.

Compatibility Settings to Open Logic 4.x Songs in Logic 5

Switch Sample Accurate Automation

Audio > Audio Preferences > Sample accurate automation allows to fully or partially switch the new sample accurate automation system for audio channels. In the OFF setting the automation responses exactly as it did in Logic 4.x ensuring compatibility with songs utilizing audio channel automation created and saved in the Logic 4.x song format. The setting Volume, Pan, Sends offers sample accurate automation for volum, pan and send controls, the plug-in automation however, is executed as it were in Logic 4.x. This reduces the CPU load when sample accurate plug-in parameter automation is not crucial.

Compatibility of Software Instrument Automation

The flip-menu setting **Options > Settings > MIDI Options > Software Instruments use MIDI Controller 65-127 for plug-in parameters (Version 4.x behaviour)** ensures compatibility when opening songs utilizing audioplug-in automation created and saved in Logic 4.x.

Logic 4.x used MIDI controller messages starting with CC 65 to automate software instruments. However, third party software instruments may use other MIDI controller numbers to control their parameters. In this case use the setting: **Software Instruments use MIDI Controller as standard MIDI controls.**

10.4 MIDI File Import

The way Logic interpretes multiple Note On and Note Off messages on the same channel and track (which overlaps) has changed. As a result, the shortening of note lengths for identical overlapped notes does not happen any more.

10.5 EXS24

An addendum to the EXS24 manual—available as an electronic document on the CD-ROM—describes these features:

- Improved Sample Instrument and file management
- Simple to use Search facility
- SoundFont 2 format Sample Import
- SampleCell II format Sample Import
- ReCycle/REX format Sample Import
- Akai format Sample Import improvements
- Additional sample import options
- New menu options
- Enhanced and additional editing options

Further to the facilities outlined in the electronic addendum, Version 5.0 of Logic introduces the following EXS24 features.

VSM (Virtual Sample Memory)

Logic 5 supports Virtual Sample Memory (VSM) for the EXS24. Thanks to VSM, the EXS24 can play back audio files of practically any size. The in-depth documentation and the necessary system file can be found in the download section of www.emagic.de. If your computer is connected to the internet during installation, these files will automatically be installed.

Converting ReCycle/REX Files into EXS24 Instruments

This function only is available with the EXS24 installed.

The conversion of ReCycle/REX files requires an installed REX Shared Library system extension—version 1.2.1 (build 45) or higher (Mac) or REX Shared Library DLL version 1.2.1.43 or higher (PC). This system extension is installed by the Logic 5 installer. To download the latest version of the library visit the support section of **www.propellerheads.se**

Mac

Another possible use of ReCycle/REX files in Logic is to convert them into EXS24 sampler instruments. For this purpose, the **Instrument** menu in the EXS24 Instrument Editor offers a **ReCycle convert** submenu. Here's how it works:

Extract sequence and make new instrument

Using this menu option will create a new EXS24 instrument with the name of the ReCycle loop.

If an EXS24 instrument with this name already exists, a # sign and a number will be appended to it, making the filename unique, in the Sampler Instruments folder).

After selecting a ReCycle/REX file, you will be prompted to enter a **Velocity Factor** (see below). Usually, you will leave this set to zero (all MIDI notes set to velocity 127). Press the OK button after entering the desired value.

You will now see that Logic has generated a number of Zones (one for each slice of the ReCycle/REX file you have imported) and one Group that all of these Zones are assigned to.

In addition, a MIDI sequence has been generated on the currently selected track, at the current song position (rounded to whole bars). This MIDI sequence is used to play the imported slices at the timing given by the ReCycle/REX file. You can generate new MIDI sequences at any time from the imported EXS24 instrument (see below), so feel free to modify or delete this sequence.

Extract sequence and add samples to current instrument

With this menu option, you have the option of adding a ReCycle loop to the EXS24 instrument currently opened in the Instrument Editor. This allows you to assign several different loops to one EXS instrument, enabling their use on the same Audio Instrument track.

If there is no instrument open in the Instrument Editor, this function works just like **Extract sequence and make new instrument**.

Extract sequence(s) from ReCycle Instrument

Use this option to generate MIDI sequences from an EXS24 instrument created from a ReCycle/REX file import. The sequences are created on the currently selected track, at the current song position, and rounded to bars. One MIDI sequence is generated for every imported ReCycle loop in the currently open instrument. If there is no imported ReCycle loop in the currently open instrument, this menu option is disabled.

You will be asked for a velocity factor (see below), as well.

You'll find the same function in the **Options** Menu of the EXS24 (beneath the instrument selection box). Here, it is easier to see which Audio Instrument (and EXS24 instance) you're using—in cases where several instances are active.

So, to **Extract sequence(s) from ReCycle Instrument**:

- Insert the EXS24 into an Audio Instrument track.
- Load a sampler instrument created by the ReCycle Import function.
- Choose **Options > Extract sequence(s) from ReCycle Instrument**. The MIDI sequence will appear on the desired track.

Paste loop from clipboard as new instrument

This works just like **Extract sequence and make new instrument**, except that the content of the clipboard is used (assuming you've copied a loop into it by using ReCycle's *Copy loop* function), rather than an audio file.

Paste loop from clipboard to current instrument

This works just like **Extract sequence and add samples to current instrument**, except that the content of the clipboard is used (if you have copied a loop into it by using the *Copy loop* feature in ReCycle).

Velocity Factor

The *velocity factor* (in %) allows you to control how the loudness of each slice in the imported ReCycle/REX file affects the velocity values of the MIDI note generated to trigger it. If you enter a positive value, higher velocity values will be generated. The highest value you can enter is 100. Negative values result in lower velocities. **Velocity Factor** 0 sets all velocity values of the corresponding MIDI sequence to 127.

Intelligent Deletion of Sampler Instruments

EXS24 instruments can now be deleted intelligently via the new "Delete" option in the Instrument menu of the EXS24 Edit window. A built-in security function prevents the accidental deletion of audio files that are used by other Sampler Instruments.

Load Multiple Samples

In the EXS instrument editor, it's possible to **Load Multiple Samples**.

Prelistening to Samples

There is a **PreListen** function within the new file selector on Windows.

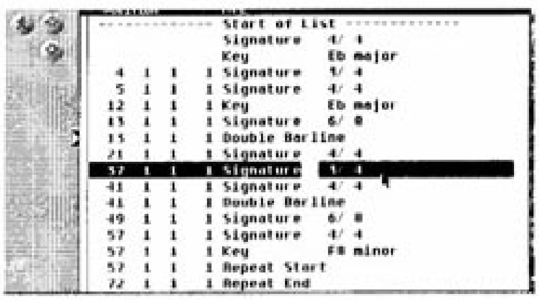
Sine as Surrogate Signal

The EXS24 plays a sine wave in place of the assigned wave file, should it not be found.

Chapter 11

Time and Key Signature Editor

In the Time and Key Signature Editor window it is possible to copy, move and delete time signature changes, key signature changes and global score symbols (repeat signs, special barlines etc.). To open this window, choose **Signature/Key Change List Editor** from the (global) **Options** menu.



This editor looks and works very similar to the Event List editor, except that new list entries can only be created by copying existing ones. The following score symbols are displayed in this window as list entries, if they are present in the score of the song:

- Time signature changes
- Key signatures and key changes, as displayed in the Score Edit window
- Repeat signs (repeat start, repeat end, repeat end+start, brackets for first and second endings)
- Special barlines: double barlines, final double barlines ("end of score"), half/short barlines, hidden barlines (displayed as

dotted lines on the screen) and manually inserted regular barlines.

Note: the initial time and key signature of the song is always displayed here, without bar position indications at the top of the list.

Please note: None of these symbols affects Logic's MIDI and audio playback. However, time signatures not only affect the score display, but also the bar position display in all windows and therefore all functions which somehow depend on the chosen time signature and on bar positions, like the MIDI metronome click or the functions in the Transform Edit window, if bar positions are used to restrict its chosen function to a certain part of the song.

11.1 Copying Time and Key Signatures

In this window, time signatures, key changes, repeat signs and special barlines can be copied exactly as in the Event List window:

Select all list entries which you want to copy by clicking on them ([⇧]+click for multiple selection) and use menu **Edit > Copy** or the key command [⌘][C] (Mac OS) or [ctrl][C] (Windows) to copy them to the clipboard.

Now insert the copied entries by choosing Edit > Paste or the key command [⌘][V] (Mac OS) or [ctrl][V] (Windows). The copied list entries are now being inserted into the list, an entry box opens for the bar position of the first inserted list event. Type the desired bar position in the usual way (like 37,1,1,1) and close that box by pressing the [Return] or [Enter] key. Subsequent list entries are inserted at bar positions that correspond to their original distance in relation to the first copied list entry.

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11.2 Moving and Deleting List Entries

All list entries in this window (except the initial time and key signatures at the top of the list) can be moved or deleted. Again, this corresponds exactly to the way the Event Edit window works:

- To *delete* a list entry, select it and press the [Backspace] key.
- To *move* an entry, you can either click on its bar position with the mouse, keep the mouse button pressed and move the mouse up or down; or you double-click on the bar position of the list entry and type the desired bar position into the entry box which opens after the double-click.

11.3 Changing Time and Key Signatures in the List Editor

Existing time signatures and key changes can be altered directly in this list:

- Their *position* can be changed as described in the previous section.
- *Time signatures* can be changed by positioning the mouse pointer at the nominator or denominator in the list and dragging the mouse up or down to increase or decrease that value.
- To alter a *key signature*, either click on the word major or minor to switch between parallel major and minor keys, or click and hold on the root of the key: This opens a pop-up menu with the root notes of all available keys. Select the desired key by moving the mouse to the corresponding note in that pop-up menu.

All these changes will immediately affect the display in the Score Edit Window and in all other windows.

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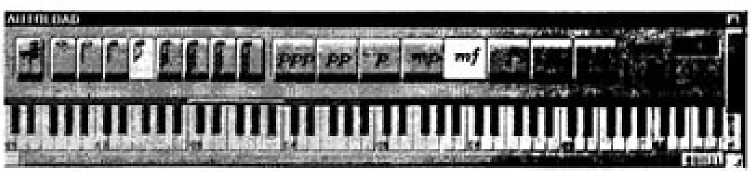
Chapter 12

Step Input

The step input functionality has been reworked completely in Logic 5.0 and has been expanded significantly. In addition to the already existing **MIDI Step Input**—which has been adapted to correspond with the new step input features—you can also use *key commands* (using the computer keyboard as a kind of music typewriter) or **mouse clicks** in the new **Keyboard Window**, which can be opened via the (global) menu **Windows > Open Keyboard Window**:



This new window is the central element of the graphical user interface for all types of step input. However, it is only the *visible* part of the new functions. Much more important is a whole group of *key commands* that allows the user to insert notes via the computer keyboard. Many of those key commands correspond to the buttons in the Keyboard Window, like those which determine length or velocity of the notes which are to be inserted. Before we go into detail about those key commands, here is an outline of the basic principles:

12.1 Preparations for Step Input

In order to use the step input functions, there has to be a MIDI sequence into which notes are going to be inserted. Usually you will create a new sequence for that purpose, using the Pencil Tool on the desired track of the Arrange Window.

The next step will be to open (or activate) an editor window that displays the contents of that sequence. This can be either a Score Edit Window, a Matrix Edit Window or an Event List Window, depending on personal preference. The **Input Mode** has to be activated in that window (Click the MIDI IN button or use the key command *MIDI IN* to toggle Input Mode on and off). Now you can open the Keyboard Window if it's not already open anyway (menu **Windows > Open Keyboard Window**).

If you decide to work in the Score Edit Window, you have to adjust the display settings for that sequence accordingly: **Interpretation** has to be **off**, the display quantize setting (**Qua**) should be set to the smallest note value you intend to insert.

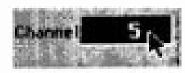
12.2 Input

Before you start, set the Song Position Line (SPL) to the spot where you want to insert the first note, for example by adjusting the song position in the Transport Window.

Single Notes

For getting to know things it's easier to use the mouse in the beginning, clicking the key for the desired pitch in the Keyboard Window, and also the buttons for note length, velocity etc. This inserts a note with that pitch at the current song position into the selected sequence, with the length and velocity chosen in the Keyboard Window before. This can also be done with key commands—we will talk about that in a minute ...

The **pitch** of an inserted note is determined by the key that you click in the Keyboard Window. **Length** and **velocity** are determined by the corresponding buttons in that window. If the buttons look like shown in the illustrations to the right, a sixteenth note with velocity setting 96 would be inserted—this corresponds to the forte (*f*) indication. The eight available velocity values are represented by the tradition volume indications *ppp*, *pp*, *p*, *mp*, *mf*, *f*, *ff* and *fff*, which correspond to the velocity values 16, 32, 48, 64, 80, 96, 112 and 127.

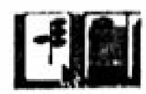


The MIDI channel of an inserted note is determined by the **Channel** setting in the Keyboard Window. Remember: The channel settings of individual notes usually has no effect on the playback channel and therefore the sound of that sequence—this is determined by the track instrument settings in the Arrange Window. The channel settings of individual notes are mainly used for polyphonic score display.

Proceeding

After a note is inserted, the song position line will always jump to the end of the inserted note. The following note can be inserted right away, following the same principles again.

Chords



To insert several notes at the same bar position, the **Chord Mode** has to be activated by clicking on the button on the far left side of the window or using the key command *Chord Mode*. This will keep the song position line from moving after inserting a note, allowing the user to insert as many notes as desired at the same bar position, one after the other. Another click on the chord button (or the same key command again) will deactivate the Chord Mode (this is best done before inserting the last chord tone); the song position line will now move forward again after inserting notes.

12.3 Step Input Using the Computer Keyboard

Note input with the computer keyboard works according to the same principle as just described. However, instead of choosing pitch, velocity, note length etc. with mouse clicks, key commands are used for all functions. To have better control over what you are doing it is recommended to still open the Keyboard Window when you start using these functions, however, it actually isn't really necessary to open that window when using key commands.

All key commands for step input are user definable. In the Key Command Window (open via [global] menu **Options > Settings > Key Commands**) they can be found at the very bottom of the list, under the header *Keyboard Input.*

The following key commands are available:

Note 'C'
Note 'C#'
Note 'D'
Note 'D#'
Note 'E'
Note 'F'
Note 'F#'
Note 'G'
Note 'G#'
Note 'A'
Note 'A#'
Note 'H'
Pause
Next note will be sharp
Next note will be flat
Chord Mode
Erase
Step backwards
Step forward
Octave 0
Octave 1
Octave 2
Octave 3
Octave –2
Octave –1
Octave +1
Octave +2
1/1 note
1/2 note
1/4 note
1/8 note
1/16 note
1/32 note
1/64 note
1/128 note
Next three notes are triplets
Next two notes are a dotted group
Velocity 16 (ppp)
Velocity 32 (pp)
Velocity 48 (p)
Velocity 64 (mp)
Velocity 80 (mf)
Velocity 96 (f)
Velocity 112 (ff)
Velocity 127 (fff)

Team TND

Octave 4
Octave 5
Octave 6

12.4 Key Commands—More Details

Pitch

The twelve key commands *Note C* to *Note H* will insert that note with the currently set length and velocity at the current position of the song position line (**SPL**) into the currently selected sequence. To insert chords, use the *Chord Mode* command. Switch off chord mode using the same command once again.

Instead of assigning key commands to all twelve pitches there is also the option to only define key commands for the seven pitches of the C major scale and use these in combination with *Next note will be sharp* and *Next note will be flat* to insert the other pitches.

When creating a system for these key command assignments two approaches seem obvious: either to arrange these commands on the computer keyboard similar to a piano keyboard (C D E F G A B in a row next to each other and C# D# F# G# A# above these at corresponding positions), or—especially when the second system mentioned above is used—to assign keys to pitches of the C major scale according to their note names.

Octave Ranges

The octave where a note will be inserted depends on the current position of the insert range bar. This can be seen in the Keyboard Window: There is a small horizontal bar above the piano key image that spans exactly one octave. The chosen note will always be inserted inside the range marked by that bar.

This bar changes its position all the time according to the last inserted note: The center of the insert range bar will always be aligned to the last inserted note. The next note will be inserted in that range unless the range bar is moved using key commands or the mouse.

An example: If a G is inserted right after inserting a C, the G will be put below the C: Inserting the C places the range beam between G (left end) and F# (right end), with the C in the center. Regardless which note is chosen now, it will always be inserted inside that range. However, the beam will move on as further notes are inserted, its center always aligned to the last inserted note.

Since larger intervals would be impossible that way, there are key commands for placing the range bar in different octaves: Octave 0 to Octave 6, and also Octave +1/–1/+2/–2, which move the range bar up or down one or two octaves. Apart from that the range bar can also be moved with the mouse.

Note Length and Velocity

Note Length and Velocity are determined *before* one of the pitch commands is used to actually insert a note. The currently selected values can be seen in the Keyboard Window.

If you are working in the Score Edit Window, don't forget to adjust the display parameters for the sequence accordingly (see above).

Rests, Correction, Moving Forward and Back

The key command *Rest* moves the song position line forward according to the currently set note value without inserting a note, thereby creating a rest.

The *Step backwards* and *Step forward* command also move the SPL forward or back by the length of the selected note value.

The *Erase* command will delete the last inserted note and move the SPL to the position where that note began. This allows quick corrections—another note can be inserted right away replacing the erased one.

Triplets, Dotted Note Groups

The key command *Next three notes are triplets* is self-explaining. The rhythmic value of the triplets corresponds to the currently selected note value in the Keyboard Window. If an eighth note is selected there, an eighth note triplet will be created. After three notes have been inserted, binary values will be inserted again, unless the command is applied again.

Next two notes are a dotted group works in a similar way. The special thing about it is, that a pair of notes will be inserted which complement each other, like a dotted eighth and a sixteenth note.

12.5 Interaction with Other Key Commands

If the input mode is activated in an editor window (Score, Matrix or Event List), these key commands have priority over all other key commands.

This means that you may also assign keys or key combinations to these functions which already are assigned to other functions. As long as the input mode is active, the key assignments for step input functions will apply; if the input mode is switched off, the other key assignments (for the same keys) will have priority.

So you can use key assignments that are already in use for other functions, you should just be careful not to use keys that are assigned to functions which you also might want to use when you work with step input.

A reminder: There are lots of other key commands for editing notes and for moving the SPL which can also be very useful when working in step input mode: Transport functions like *forward* and *rewind*; various *Nudge* commands for moving notes or changing their length; commands like *select next/previous event* or *transpose event +1/–1*, and all the *Note Attributes* in the Score Edit Window.
Consider that the last inserted note is always selected, so it can immediately be moved, changed in pitch or made longer or shorter using the mentioned key commands. Therefore the key assignments for these commands should not be used for step input commands—that way all these functions are still available in step input mode.

12.6 MIDI Step Input

MIDI Step Input is very similar to key command step input as described above:

First there has to be a (often empty) MIDI sequence, which has to be selected in an editor window. **Input Mode** has to be active (MIDI IN button or key command).

Selecting the desired note values is done using the same key commands as described above. The same applies to the functions *Rest*, *Step forward*, *Step backwards* and *Erase*.

However, **pitch** and **velocity** are simply determined by whatever keys are pressed on the MIDI keyboard. Also, there is no need for the **Chord Input** function: Simply play the desired chord on the MIDI keyboard, it will be inserted at the current song position.

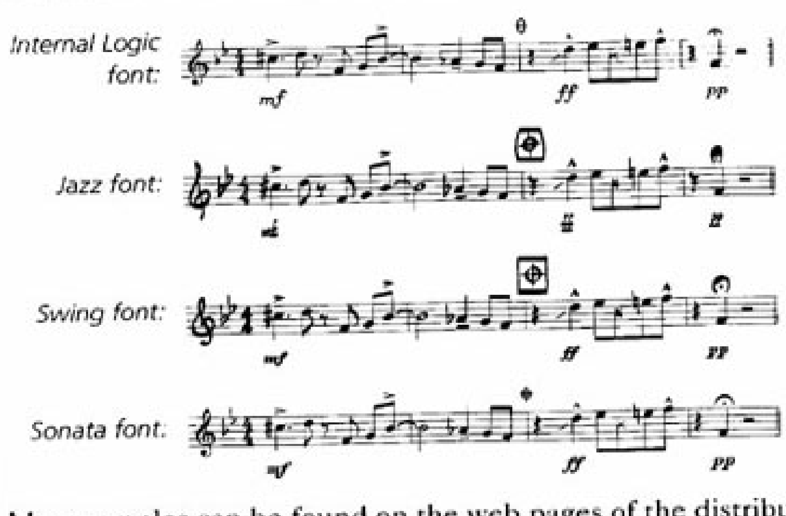
Chapter 13

Score Edit Window

13.1 Score Display Using External Symbol Fonts

The use of external symbol fonts for the score display can change the appearance of the score completely. Apart from Logic's own font, which is directly integrated in the program itself, it is possible to use *JazzFont*, *SwingFont* or the *Sonata* font. These fonts are distributed by other companies (see below) and are not included with the Logic program package.

Sonata has a rather traditional character. *Jazz* and **Swing** have a hand-written appearance, as it can be seen in various fake-books. For comparison, here is the same short phrase printed four times with the different fonts:



More samples can be found on the web pages of the distributors. The *Jazz* and *Swing* font also come in a bundle with matching text fonts which can be used in Logic for text input of all kinds.

These fonts can either be acquired in music and computer stores or at the following internet addresses:

- *JazzFont* and *SwingFont* can be purchased directly from Richard Sigler, their creator, at http://www.jazzfont.com/
- Sonata is distributed by *Adobe*. The following URL takes you directly to the page where Sonata is offered: http://www.adobe.com/type/browser/P/P_021.html

Requirements and Installation

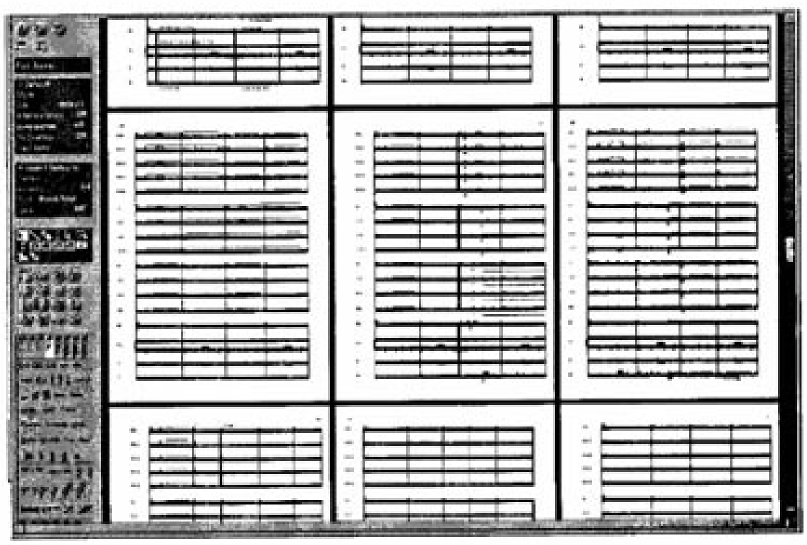
To use these fonts with Logic, they have to be properly installed in the system, just like regular text fonts. If you use a Macintosh, copy or move the font files into the *Fonts* folder, which is inside the system folder. If you are a Windows user, just start the installation program that you get when acquiring the fonts and follow the instructions on the screen.

Then choose menu **Options > Score Preferences**. In the dialog window that opens, activate the option **Use External Symbol Font (if available)**. Since especially with Sonata the screen display in small zoom levels is rather unclear, there is also an option *Only for Printout*, which causes the chosen font to be used only for printing, but uses the internal Logic font for the screen display.

13.2 Screen Display—New Functions

Multiple Page View

In Page View mode, Logic will now automatically display as many pages next to each other as possible, according to the size of the Score Edit window and the current zoom level.



This is especially useful when working on the overall layout of an extended piece.

More Zoom Levels—Finer Resolution

There are a lot more zoom levels in the Score Edit Window now, the difference between them has become much smaller. This enables users to choose the score display in each Score Edit Window size exactly according to their individual needs.

A reminder: For selective zooming of only a small part of the window that is to be displayed as large as possible, use the magnifying glass tool or better the **zooming shortcut**: While keeping the alt-key (Windows)/ctrl-key (Mac OS) pressed, use the rubber-band selection technique to select the desired part of the score display. A click on the background while holding the same key again will bring you back to the original zoom level.

13.3 General Score Display Functions

Alia in the Score Display

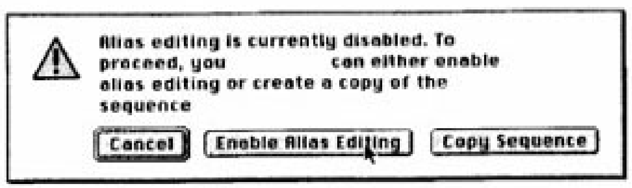
A reminder: *Alias* (pl. **Alia**) is a term used for objects in the Arrange Window which are not independent sequences, but only "mirror" other sequences at different positions and/or tracks. Wherever an Alias exists, its original sequence is played back.

Alia can now also be displayed in the Score Edit Window and also be printed. The display of Alia has to be activated in the **Global Format** dialog window (menu **Layout > Global Format**), by checking the option **Show Alias** there.

This option is a *Song Setting*. If you always want to see Alia in the score display, you should activate this option in your **Autoload** song.

An Alias can be assigned a different Score Style than its original sequence. For example, a melody played in unison by several instruments can be handled as a regular MIDI sequence and several Alia of that sequence. Different Score Styles can be used for the different instruments, also transposing Score Styles. If you decide to change something in the melody, you don't have to edit all the copies of that melody, but just make the change in one sequence. That change will automatically be reflected in all Alia of that sequence. The same applies to repeated sections within a piece.

By default, Logic will not let you edit the contents of an alias (only the settings in the Display Parameters Box). If you try to edit the contents of an Alias (for example move a note), the following dialog will appear:



There are three choices now:

- **Cancel** reverts the last edit.
- **Copy Sequence** converts the Alias into a separate, independent sequence, which can now be edited without affecting the original sequence.
- **Enable Alias Editing** activates a mode which allows you to edit the contents of an original by editing its Alias. In this case, the original and all its Alia will reflect the changes you apply to any of them.

Enable Alias Editing is a song setting which can also be activated in the **Global Format** dialog window (menu **Layout > Global Format**).

Apart from that, there is also an option to convert an Alias to an independent sequence by selecting it in the Arrange Window and choosing **Functions > Alias > Convert to Real Copy** or the corresponding key command. (This option already existed in former versions of Logic.)

Alternating Page Margins

The option **Alternating Margins** can be found in the *Global Format* window (menu **Layout > Global Format**). If activated, the left and right page margin settings will be swapped on every second page. This can be useful if a score is going to be bound as a book: Usually the inner margin has to be a bit bigger than the outer one. But also the opposite is possible imaginable in order to leave space for remarks added later to the score.

13.4 Score Layout Details

Individual Alteration of Stem Length

The stem length of each note can be changed individually. This can be used to change the stem length of notes which are not beamed to any other notes, to alter the inclination of a beam that connects several notes by changing the stem length of the first and/or last note under that beam, or to move a beam up or down as a whole in parallel motion, which is especially useful for cross-staff beaming situations (see below).

There are three new menu commands in the **Attributes > Stem > ...** submenu and also corresponding key commands, which can be found in the Score section of the Key Command window:

stem end: move up
stem end: move down
stem end: default length

As the name implies, *stem end: move up* works independently from the actual stem direction—the end of the stem is always moved *up*, which results in an increased stem length if the stem points up, and in a shorter stem length if the stem of the selected note points down. The same principle (with opposite effect) applies to *stem end: move down*.

Although this method might seem odd at first glance, it is very useful when cross-staff beaming is used and the beam is positioned between the staves: In this case you can simply select all beamed notes and move the whole beam up or down in parallel motion.

Reminder: The general default setting for the stem length can be changed in the **Extended Layout Parameters** window (open via menu **Layout**). This setting applies to all notes in the song.

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Cross-staff Beaming

Music for keyboard instruments or harp, which is notated in two staves, sometimes contains passages where notes in the upper and lower staff (played by left and right hand) are connected with a beam to emphasize the continuing musical phrase, as shown in the following illustration:

Please note: If you want to use this function, you should already have a good knowledge of all functions associated with Logic's Score Styles. Extensive information about this topic can be found in the Logic reference manual.

Required Settings:

Cross-staff beaming is only possible between staves which belong to the same Score Style and therefore to the same MIDI sequence. Apart from that, the **Voices** in this Score Style have to have to different MIDI channel assignments—it is not possible to achieve cross-staff beaming with a Score Style that only uses a split point pitch for Voice separation.

The Procedure in Detail

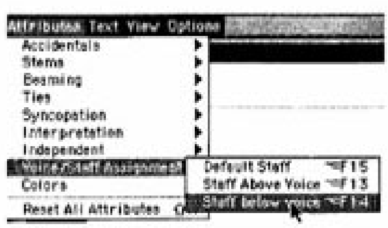
Since notes can only be connected with beams, if they belong to the same **Voice** (in the sense of Logic's own terminology), it is not possible to simply select notes in both staves and apply the usual beaming attribute functions to them.

As an example, let's imagine a two-staff Score Style with one Voice per staff. Before beaming the notes, it is necessary to assign all notes which are to be connected by a beam to the same Voice, let's choose the Voice of the upper staff.

To change the Voice assignments for the notes in the lower staff, you can either use the *Voice Separation Tool* and draw a line below those notes or select all these notes ([⇧]-click or rubber-band selection) and change their MIDI channel in the Event Parameter Box to the channel of the upper Voice. In the example shown above all notes would have to be assigned to the upper Voice first. Now all those notes will be displayed in the upper staff, probably with a lot of ledger lines as in the following illustration:

The next step is to define the beaming connections as desired. This can be done with the regular note attribute functions (menu **Attributes > Beaming > Beam selected** or corresponding key command).

Now select those notes once more which should actually be displayed in the bottom staff and choose the menu **Attributes > Voice/Staff Assignment > staff below Voice** or the corresponding key command.



[ℹ] In Logic Windows the menu bar looks a bit different, but the *Attribute* menu itself contains exactly the same functions.

This moves all selected notes down to the lower staff, although they still remain part of the upper **Voice**.

You could also choose the opposite approach—first assigning all notes to the lower Voice, then moving part of the notes to the upper staff with **Attributes > Voice/Staff Assignment > staff above Voice**.

To display all selected notes in their original staff, use **Attributes > Voice/Staff Assignment > default staff.**

Stem Direction

The default stem direction for notes that beamed across staves is towards the center. I.e. the beam is displayed between the two staves, stems in the lower staff point up, stems in the upper staff point down.

The vertical position and the inclination of that beam can be altered by applying the stem length change functions described above: To move the whole beam in parallel motion, select all beamed notes at once and choose **Attributes > Stems > Up** or **... Down**, to change the inclination of the beam, select the first or last note of that group and apply those functions.

However, you can also define the same stem direction for all notes with the regular note attribute functions, thereby placing the beam above the top staff or below the bottom staff. If you decide that you like the in-between beam better, you'll have to select all notes again and choose **Attributes > Stems > Default**.

Rest Display

As always, rests are displayed automatically in Logic, according to the **Rest** settings in the used Score Style. However, since in the cross-staff beaming situation described above, most or even all notes belong to the Voice of the top staff, the lower staff will contain rests—some even directly at the position of notes. To avoid this, you should use a Score Style where the automatic rest display is deactivated for the Voice of the bottom staff. Wherever rest display is desired, **User Rests** can be inserted with the mouse from the Partbox.

Colored Notes

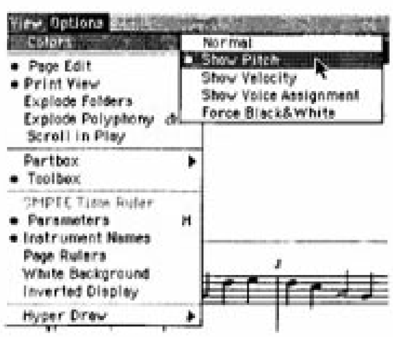
Notes can be displayed and printed in color. This could serve several purposes:

- For teaching: Colored display of pitches as additional help for beginners, especially small children.
- To distinguish between different **Voices** in the same staff when using polyphonic Score Styles, which also can be helpful in the Voice separation process.
- To distinguish between different MIDI note velocities.
- Using custom colored notes for any conceivable purpose.

The use of colors for notes can be employed on several levels: For individual notes, for all notes of a particular Voice within a Score Style or for all notes displayed in a Score Edit Window.

Color Settings for all Notes Displayed in a Window

The **View** menu of the Score Edit Window contains the entry **Color** which has a submenu with five options. These settings here always have priority over all other color settings. However, they only apply to this particular window.

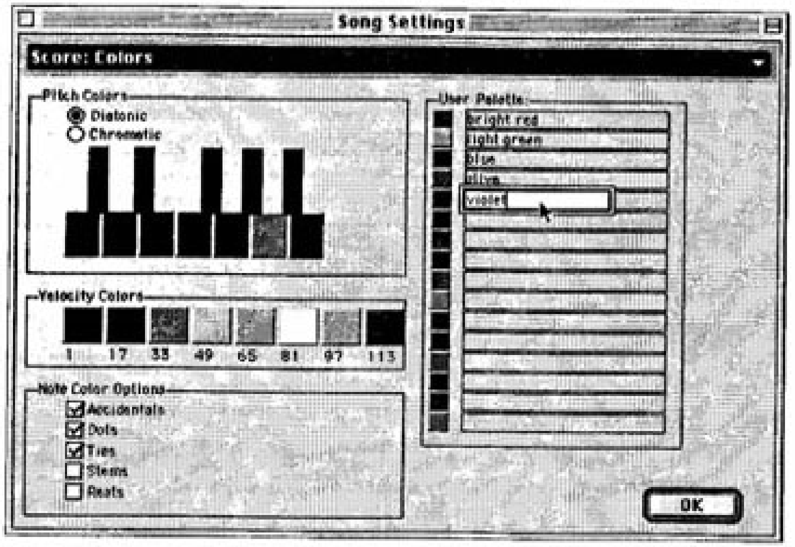


- **Normal** or **Individual** means that colors are assigned according to the color settings in Score Styles and Note Attributes. As long as those haven't been changed in new songs, this will result in a regular black & white printout and black notes on yellow background on the screen.

- **Show Pitch** applies colors according to the pitch of notes. The colors for that can be edited in a special window that open via the menu **Layout > Colors** (see below). There you can also determine whether notes with accidentals will have the same color as without accidentals, or if each note in the chromatic scale will have its own color.
- **Show Velocity** applies eight different colors according to MIDI velocity. Also these colors can be edited in **Layout > Colors.**
- **Show Voice Assignment** assigns different colors to different **Voices**. This only makes sense in Score Styles that contain more than one Voice. This mode employs the colors of the user palette, which can also be found and edited in **Layout > Colors.**
- **Force Black&White** does exactly that. This mode is useful when color options have been used in Score Styles or colors have been assigned to notes using Note Attributes, but one wants to print a regular black & white score and then return to the colored display.

Color Palette

The menu **Layout > Colors** opens the color palette for the currently active song:



This is the window in the Mac version. However, apart from the window layout itself, there is no difference to the Windows version.

This is where all the color palettes can be found which the different coloring modes refer to. Clicking on any color opens the standard color editing window, where this particular color can be changed.

These colors are saved with the song and can be different in each song. They can be imported from any other song as part of the score song settings using menu **Options > Import Settings > Score Settings**.

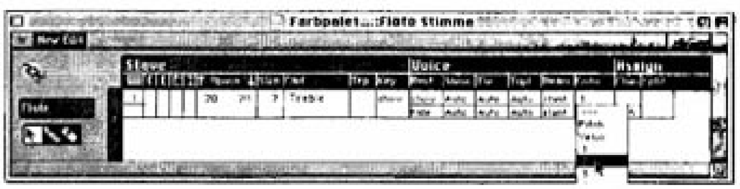
There are three palettes and some additional options for details in this window:

- **Pitch Colors**: Each note of the octave gets a color, they are arranged like piano keys. **Diatonic/Chromatic** determines whether there are twelve or seven different colors.

- **Velocity Colors:** These eight colors are applied according to the MIDI velocity of notes, from left (minimum) to right (maximum).
- **User Palette:** These colors not only can be edited freely, but also named, using the text entry boxes next to them. These names are displayed as color options in different color selection menus (Score Styles, Note Attributes).
- **Note Color Options:** These options determine whether accidentals, dots, stems and beams are displayed colored (according to the notes they belong to) or black. **Rests** refers to **user rests**, which are mainly used in polyphonic Score Styles.

Color Settings in Score Styles

In Score Styles, each Voice can be displayed with colored notes. This has to be set in the Color column of the Score Style Window for the desired Style.



The color options are the same as listed above: Black ("---"), **Pitch, Velocity** and the 16 colors of the **user palette.**

These settings only affect sequences which use an according Score Style. In addition, the active color mode in the Score Edit Window has to be **Normal (Individual)**, which is the default setting.

Color Settings for Individual Notes

Single notes can also be assigned colors individually, again according to the three color palettes. The note has to be selected, then menu **Attributes > Colors >...** is used to assign the desired color (user palette 1 to 16) or color mode (pitch or velocity).

Being Note Attributes, these settings have priority over the Score Style settings. However, they only apply if the color mode in the Score Edit Window is set to **Normal (Individual)**.

Oversize Staves

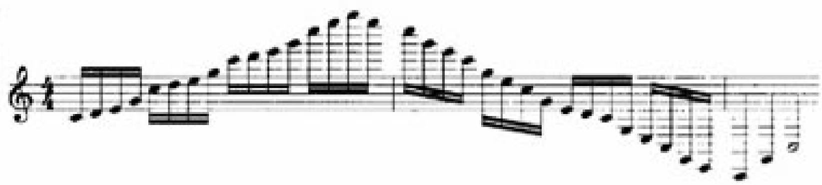
The size range in Logic's Score Styles (until now 0 to 10) has been extended to go to 15. This makes it possible to print staves with a height of up to app. 14.5 millimeters (= app. 0.57 inches), as its often done in beginner's books for small children.

Consider the note color options in combination with this feature!

A hint: If you want to create even bigger staves, you can additionally use the **Scaling** function in the Instrument Set Window. Create an Instrument Set containing the desired instrument/s and set the Scaling parameter to 200%. This can produce staves which are almost 3 centimeters high (= app. 1.14 inches). You might want to use landscape format in this case.

Octave Symbols

The Partbox group that contains clefs also contains octave signs which automatically affect the note display when inserted: If an *8va* sign is inserted above a staff, all notes between the bar position of the *8va* symbol and the end of its bracket will be displayed an octave lower, which in conjunction with the *8va* sign means the same result as the original notation. The MIDI playback remains unaltered. The same applies to the other available octave signs (one octave lower, two octaves higher or lower). Below you see the same MIDI sequence before and after octave signs are inserted:



example: original sequence

the same sequence after inserting octave symbols

The end of the bracket can be dragged with the mouse to the right or left as desired, also beyond line breaks.

The actual appearance of these symbols can be determined in **Layout > Clefs & Signatures > Octave Symbols**. Here it's possible to define any desired text according to personal preference for each of these symbols by inserting it in the corresponding text boxes. This makes it possible for example to use 8vb instead of 8va bassa. (Theoretically, *any* text would be possible to represent each of these symbols.) Also the text font, size and style for octave symbols can be changed here.

The octave symbols displayed in the partbox are only icons which don't change when the default definitions are changed in the Clefs & Signatures dialog.

Distance between Notes and Barlines

The default setting for the relative distance between the first and last note of a bar and the preceding or subsequent barline can now be affected by changing the following two parameters in the **Extended Layout Parameters...** (menu **Layout**):

Bar Start Spacing
Bar End Spacing

Please note: To change the default settings of these parameters only makes sense if rather extreme settings have been chosen for the general Spacing parameters in the **Global Format** window.

Accidental Size in Chord Symbols

The size of accidentals in chord symbols (to display the root note, like Ab or F#) in general changes with the font size used by the **Chord Root** Text Style. However, the **Chord Symbol Accidental Scale** parameter in the **Extended Layout Parameters...** window (menu **Layout**) allows to change the accidental size in relation to that font size setting: Positive values result in an increased size, negative values in a diminished size of chord symbol accidentals. The default setting is zero.

Additional Bar Number Layout Options

Centered Position for Bar Numbers

There is a new option for the display of bar numbers, which can be found in **Layout > Numbers and Names... > Bar Numbers > Hor. Position > at Bar Center**: Bar numbers can now be displayed not only above or below *barlines*, but also above or below the *center* of the bars.

In the example shown above, a negative **Vert. Position** value and a circled text format have been used for the bar numbers.

Hide Bar Numbers in Linear View

This option can also be found in **Layout > Numbers and Names... > Bar Numbers** and will hide the bar numbers in linear view (where usually the bar numbers are displayed in the bar ruler anyway).

Key Commands for Moving Symbols

The following key commands can be used to move any score symbol that can be positioned independent of notes or staves, as well as text events and chord symbols:

Nudge Position Up
Nudge Position Down
Nudge Position Left
Nudge Position Right

These functions do not alter the bar positions of the affected symbols and text events, but their **hor.pos.** and **vert.pos.** parameters by ±1.

Reminder: These parameters define the exact graphical deviation of an object from its bar position (**hor.pos.**) and from the top line of the staff (**vert.pos.**). The measurement unit for these settings is one sixteenth of the distance between two lines of a staff. Therefore these commands only cause very small movements, so it is recommended to work in a very high zoom level when using these functions for layout work.

Team TND