EXS24
User Manual

Version 4.7.2
October 2001

English

emagic
Soft- und Hardware GmbH

r2.01 – 09.10.2001

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Dear EXS24 User,

The manual is based on the EXS24 version of Logic Audio/ MicroLogic AV 4.7.2. Depending on the version of Logic Audio/MicroLogic AV used, the functionality of your EXS24 may be different when compared to the description in the manual. For this reason, and to remain current with the EXS24 development, we recommend using the latest version of Logic Audio/MicroLogic AV posted on our web site www.emagic.de. EXS24 features added after this manual was printed are discussed in the ReadMe files of the corresponding Logic updates.

Your Emagic Team

1 Welcome …

… and thank you for your purchase of the Emagic Xtreme Sampler—EXS24. We are proud of this very powerful and flexible software-based sampler for your Logic production environment, and are confident that you will find it an ideal addition to your studio.

The EXS24 offers all of the facilities that you would expect to find in a hardware sampler, without the cost and bulk of this type of device. As a purely software-based instrument, the EXS24 is perfectly integrated into Logic Audio, and makes use of your computer's RAM and hard disk(s). This integration within the computer environment offers instant access to all audio data and Sampler Instruments used in a Logic song file. These files are stored on your computer's hard disk(s). This integration simplifies sample library management and eliminates the need for separate physical devices and the cables required to connect them.

You can make use of the editing features of Logic, or another audio editor for your samples. This is far more convenient, not to mention faster, on a computer monitor than on the cramped display found on most hardware samplers. As the samples are stored within the computer, the slow and often unreliable transmission of sample data back and forth between your sampler and PC or Macintosh is eliminated.

The EXS24 is compatible with the EXS 24, AKAI S1000 and S3000, SampleCell, WAV, AIF(F) and SoundFont2 sample formats (compatibility with other formats will follow), allowing access to large and comprehensive sampler libraries.

The EXS24 offers numerous sample processing options, enabling you to tailor sounds to meet your needs.

The response of the EXS24 to the note messages of a MIDI sequence track is a little faster than that of a MIDI sound generator connected via a MIDI cable. This ensures that your sample playback timing is rock-solid.

Whatever you play on the EXS24 can be recorded by simply pressing Logic Audio's record button. In addition, any adjustments to the EXS24's controls can be recorded and edited using any of Logic Audio's MIDI editing windows. All EXS24 parameter automation plus mix automation—volume, panorama and effects changes over time—can be freely recorded, edited and replayed.

Last, but not least: as a highly optimized Logic Audio instrument, the EXS24 offers great performance, even on slower machines. The EXS24's performance is scalable, so you can look forward to enhanced functionality and increased polyphony on future computer technology. The number of possible Sampler Instruments available for simultaneous playback is directly related to the computer's processing and RAM resources. The more RAM you have, and the faster your CPU, the more Sampler Instruments can be loaded and played.

And what of the sound?
As the EXS24 uses high-end algorithms with 32 Bit internal processing, is completely digital, and seamlessly integrates into Logic, you are guaranteed pristine, clear sample playback—up to 24 Bit and 96 kHz, if you wish (and your audio hardware is appropriate). With the EXS24, there's no need to concern yourself over sound quality or compatibility issues with future audio formats.

This manual will introduce you to the concepts and functions of the EXS24. Please read it thoroughly to make the most of your new sampler, and to ensure that you don't overlook any important information.

We wish you a fun and productive time with the EXS24!

Your Emagic Team

2 What the Package Includes

Your EXS24 package contains the following components:

- This manual
- The EXS24 CD
- A registration card
- Demo CDs

Please complete the registration card as soon as possible and send it to the Emagic distributor in your country or territory. In return you will get a free EXS-Factory-Library with over 500 MB of samples.

Once registered, you will have access to:

- A regular update and support service via the Internet:
  http://www.emagic.de

- Support via our Hotline:
  In the USA: e-mail: support@emagicusa.com
  phone 1-530-477 1050
  In Germany: e-mail: support@emagic.de
  phone +49-(0)4101-495-110

In other countries: please consult the Emagic distributor in your country or territory.

3 Installation

Installation

In order to install the EXS24, you require a copy of Logic Audio Silver, Gold, Platinum or MicroLogic AV 4.0 or higher. The EXS24 installer will determine if a copy of Version 4.x is already installed. As the EXS24 requires Logic Audio 4.3 or higher to run, the EXS24 installer will perform an update if necessary.

Insert the EXS24 CD ROM and start the installation program (installer) that appears automatically onscreen. If the installer does not appear automatically, browse to the CD ROM using your operating system's file management utilities.

The following items will be installed in, or added to, the main Logic program folder:

- Readme files containing additional and updated information which was not known at the time this manual was printed.
- The *Sampler Instruments* folder which contains all of the Sampler Instruments received with the EXS24. This folder will also be used for the storage of all Sampler Instruments added or created in future. A Sampler Instrument contains all sample mapping information plus the modulation, filter, volume and pan settings needed for a fine Grand Piano multisample, as an example.
- The final folder added during installation is the *EXSamples* folder, which contains all of the raw samples (audio files) that the Sampler Instruments make use of.

Please note: During installation you will have the option to perform a *Custom* (Mac) installation. A Custom installation will install everything *but* the *EXSamples* folder. This option is useful if you do not have enough hard disk space. In this scenario, you will be required to insert the EXS24 CD ROM every time you wish to load one of these Sampler Instruments into your EXS24. For your convenience, we strongly recommend that you have more than enough hard disk space for your sample libraries. As of September 2001, there are over 40 EXS24-format CD ROM's commercially available, not to mention the many thousands of AKAI, Sample Cell, SoundFont2 and other sample format files which can be found online or purchased—so your sample library is sure to grow quite quickly.

Copy Protection

The EXS24 is copy protected via its original CD ROM and is authorized independently of the copy protection systems of other Emagic software. When activating the EXS24 for the first time, and later at infrequent, irregular intervals, you will be asked to insert the EXS24 CD. Please take care not to lose the CD, and to always keep it at hand. These security precautions are the prerequisite for any future software development. We would like to thank you in advance for your understanding.

- When the EXS24 is first used, it will be authorized. This may take a little while. Please be patient. Should you have ejected the EXS24 CD ROM, you will be prompted to reinsert it.
- At periodic, irregular intervals, you will be asked to insert the EXS24 CD in order to renew the authorization. Once again: Please take care not to lose the CD, and to always keep it at hand.

Optimization Programs

The EXS24's copy protection is not affected by disk optimization and defragmenting programs. You may use programs such as Norton Speed Disk or DiskExpress II as often as you wish.

Formatting the Hard Disk

If you format or partition a hard disk which contains the authorization for the EXS24, you will need to reinstall and reauthorize the EXS24.

4 Quick Start

The “Audio Instrument” Object Type

In Logic Audio’s Mixer (or Environment window Audio layer in Logic Audio Silver, Gold and Platinum), there is an audio object type called an “Audio Instrument”. Audio Instrument objects appear as channel strips in the Environment’s Audio layer and Track Mixer window. These objects allow the insertion of software instrument plug-ins into their top insert slot. The default song—the song that opens automatically if you move your Autoload away from the Logic program folder—features several pre-configured Audio Instruments.

An Audio Instrument is an audio object (or an Audio Track in MicroLogic AV) with the *Cha* parameter switched to one of the *Instruments*. Any audio object can be switched to operate as an Audio Instrument by changing this parameter in the Object Parameters box. The maximum number of audio instruments is determined by your version of Logic Audio. You can only insert the EXS24 plug-in into an audio object—created by selecting **New > Audio Object**—after its *Cha* parameter is set to an *Instrument* channel.

To create a new Audio Instrument channel in MicroLogic AV, simply select **Track > Create Audio Instrument**.

Loading and Playing an EXS24 Instrument

The EXS24 ships with a ready-to-play Sampler Instrument library. Following the installation of the EXS24, these Instruments can be found within the *Sampler Instruments* subfolder of the *Logic* program folder. Please follow these steps in order to use the Sampler Instruments:

- Start Logic.
- Create a new Audio Instrument object (see above) and select the EXS24 from the list of plug-ins in the first plug-in slot.

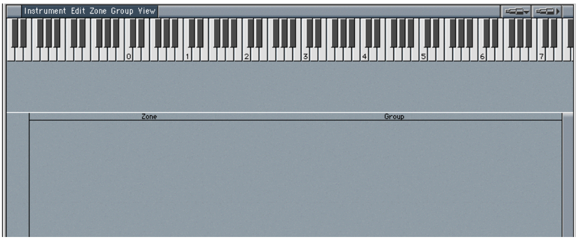
You can also use an existing Audio Instrument object for the EXS24, of course.

- Release the mouse button, once the EXS24 is selected from the list.
- Double-click on the blue EXS24 label in the Audio Instrument object in order to open the plug-in window.
- Once the EXS24's graphical interface is opened, you can select one of the Sampler Instruments by clicking on the flip menu above the silver panel area (directly above the cutoff knob). The menu is "sticky" and will remain open, even without the mouse button being depressed.
- Scroll to the Sampler Instrument name that you wish to load and click once. The selected Sampler Instrument will then load.

- In Logic's Arrange window, select the track which corresponds to the Audio Instrument channel in which the EXS24 has been inserted, and start playing your MIDI keyboard. Have fun, change the sound by twisting the knobs, pressing switches and moving sliders (don't worry—you can't destroy the original Sample Instrument). Feel free to insert effect plug-ins on the channel or busses to further enhance and manipulate the sound of the EXS24. And if you are one of the many users which run Logic Audio Platinum within a TDM™ system, the optional Emagic System Bridge (ESB) will add the DSP power of TDM to the already vast potential of Logic Audio's native audio engine.

Creating and Editing Instruments

Now that you know how to insert an EXS24 instance, and load a Sampler Instrument, it would be a good time to briefly introduce you to the Instrument Editor. Please open the Instrument Editor window via the Audio > EXS24 Instrument Editor menu. The parameters and functions of the Instrument Editor window are described in this section.



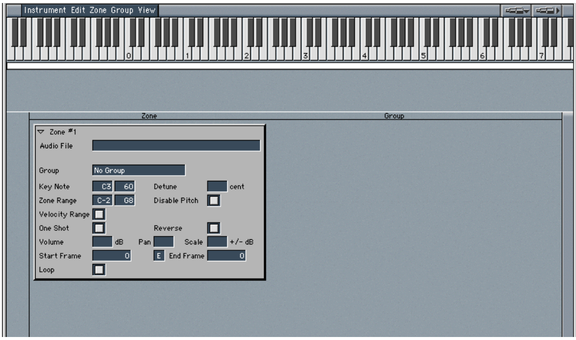
The Instrument Editor shown above is empty as no Instrument has been loaded or created. The keyboard in the upper window area can be used to trigger notes for the EXS24 in the currently selected track. Below the keyboard, a number of Zones are shown.

Creating a New Instrument and a Zone

- To create a new Instrument, select Instrument > New from the Editor window's menu. The Instrument > Open function allows you to load an existing Instrument for editing.

i In order to hear your edits, please ensure that the correct Instrument is loaded into the EXS24 instance assigned to the currently selected track, and is selected in the editor.

- Create a new Zone for the Instrument via the Zone > New Zone menu. A small window will appear to the left of the editor window. Clicking on the small triangle, left of the Zone's name, will minimize/maximize the window.



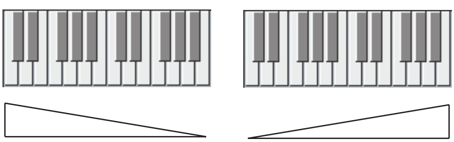
- Click on the empty field alongside the *Audio File* label. A file selection dialog box will launch, allowing you to select a sample from the hard disk or CD ROM. Selection of a sample will load it into the Zone.

Adjusting the Zone Parameters

- Set up a key range for the sample with the two *Zone Range* parameters; *Key Note* allows you to determine the note used to trigger the sample at its original pitch.

Reverse plays the sample from its end to the beginning. This option works non-destructively, and doesn't change the audio data.

- Adjust volume and pan position for the sample with the corresponding parameters. Negative *Scale* values make notes lower than the note position defined by the *Key Note* parameter sound louder than higher ones; positive values have the opposite effect. Use this parameter for balancing the volume of a sample across the selected key range.

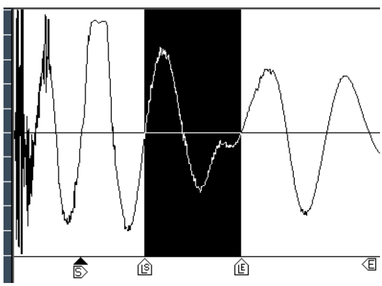


Negative values for Scale increase the volume of lower notes (left of the Key Note), positive values increase the volume of higher notes (right of the Key Note).

- If necessary, adjust the playback start and end points for the sample with the *Start* and *End* parameters.
- Activate *loop* if desired; the loop parameters are hidden when the loop parameter is deactivated. You can set a start and end point for the loop and, if needed, fine-tune the loop with the *Tune* parameter.

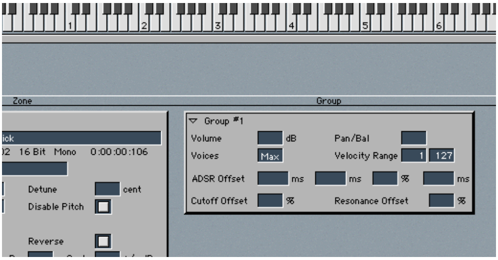
Editing Samples

You may have noticed the small *E* buttons next to the start, end and loop point parameters. Clicking on these will launch the selected sample in Logic's Sample Editor, allowing you to edit the sample borders graphically, and to make use of all of the Sample Editor window's functionality. When loop is activated, you can also edit the loop points graphically: the *LS* marker indicates the loop start point and *LE*, the loop end point.

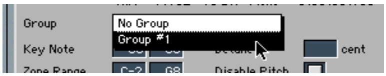


Creating a Group

- Create a new Group by selecting **Group > New Group** in the editor's menu. A Group window will appear on the right-hand side of the editor.



- Select the new Group as a target in the Zone's *Group* flip menu. The Group parameters will now affect the sample in the Zone.



Multiple Zones and Groups

You may create as many Zones and Groups as you wish, and can assign as many Zones to a Group as desired. The Groups offer several parameters for simultaneous control over all assigned Zones:

- The *Voices* parameter allows you to determine the maximum number of voices for a Group. A practical use of this would be to set up a classic "hi-hat mode" within a full drum kit, mapped across the keyboard. In this scenario, you could assign both an open and closed hi-hat sample to a Group, and set the *Voices* parameter of the Group to *1*. In this example, the most recently triggered of the two hi-hat samples will mute the other, as only one voice is allowed for the Group. This mirrors the real-world behavior of hi-hats. When samples in Zones are assigned to another Group, the other sounds of the drum kit can still be played polyphonically.
- The Group *Volume* and *Pan* parameters simultaneously affect the settings of all Zones assigned to the Group. This works much like a sub group on a mixing console.

- The two *Velocity Range* parameters are used to set up a velocity window for the Group. Use these parameters for sounds where you wish to mix, or switch between, samples dynamically by playing your MIDI keyboard harder or softer—e. g. with layered sounds, or when switching between different percussion samples, for example.
- Each Group offers separate *ADSR* parameters for offsetting the ADSR volume envelope settings made in the plug-in window: The *Attack*, *Decay*, and *Release* time parameters can be adjusted by ±9999 ms, the *Sustain* level by ±50%.
- Similarly, the *Cutoff* and *Resonance* settings of the plug-in window can be offset by ±50% for each Group.

It is possible to play all Zones without defining and assigning a Group—in such cases, the parameters defined in the plug-in window work in an absolute (i. e., identical) manner for all Zones.

Detailed descriptions of all Zone and Group parameters can be found in *The Parameters of the EXS24* section, from page 43 onwards.

5 The EXS24—Concepts and Functions

Overview and Integration

Multiple EXS24 instances can be opened simultaneously. Each instance requires its own Audio Instrument channel. Up to 24 EXS24 instances can be run in parallel, dependent on the version of Logic Audio used. Each instance of the EXS24 offers up to 64 mono or stereo voices (total number of voices is CPU-dependent).

Each EXS24 instance allows you to select, load, and play a *Sampler Instrument*. Such Sampler Instruments can consist of a single sample or multiple samples. As examples; a piano multi-sample, a drum kit, or a collection of loops spread across the keyboard range. The Sampler Instrument carries information about velocity layers, sample note position, panning and more. To use the EXS24 with different multisamples responding on different MIDI channels, you will need to open the desired number of EXS24 instances on different Audio Instrument objects.

Dependent on the mode in which you use the EXS24, each instance has a mono or stereo output which is fed into Logic's mixer, where it can be processed by insert plug-ins and/or sent to busses. Given a fast enough computer, you could conceivably arrange and mix a complete song using several EXS24 instances—entirely within your PC or Mac. As the EXS24 responds to MIDI note input, you can freely edit the parts of the song using Logic's MIDI editing windows, right up to the final mix stage.

The *Bounce* function of the Master audio object(s) allows you to "print" submixes of EXS24 tracks to disk at any time. The resultant audio files generated by this process can then be used as normal audio tracks within your arrangement. This type of functionality may prove of use when a song requires more processing power than your CPU is capable of delivering. All parameters of each EXS24 instance, and all associated mixer parameters can be fully automated during the "bounce" procedure.

The Plug-in Window

Hands-on operation of the EXS24 is performed in two windows: the plug-in window and the Instrument Editor window.

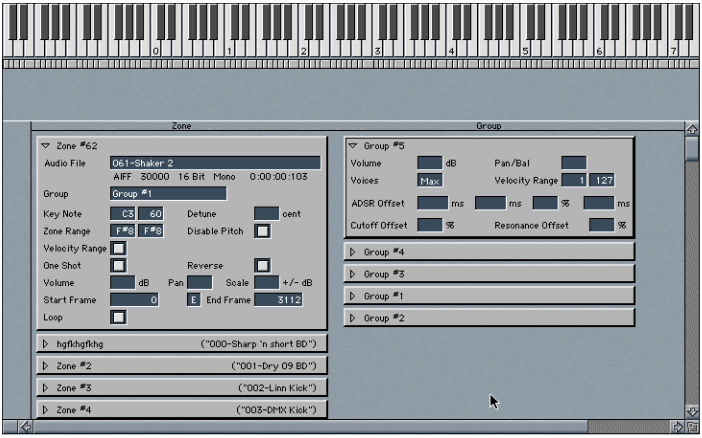
The plug-in window can be accessed by double-clicking on the blue EXS24 insert panel on an Audio Instrument object. When launched, the plug-in window allows access to all of the EXS24's parameters; e. g. filter and envelope settings. Each instance of the EXS24 has a discrete plug-in window, allowing each to have unique parameter settings, even if the same Instrument is loaded.

The plug-in window with no Sampler Instrument loaded.

The Instrument Editor

The EXS24 Instrument Editor window, on the other hand, is shared between EXS24 instances. The EXS24 Instrument Editor can be opened from Logic's **Audio > EXS24 Instrument Editor** menu.

It is used to organize samples, to construct or modify Instruments, and to convert foreign sample formats (AKAI S1000/3000 etc.). You can also assign samples to keys or key ranges and set start, end, and loop points plus all of the normally tedious tasks associated with sampling in the EXS24 Instrument Editor. Fortunately, the EXS24 Instrument Editor is much easier and more pleasant to work with than that of a hardware sampler. This is illustrated by the transparent architecture of the EXS24: samples are assigned to *Zones*, Zones are assigned to *Groups*. The end product of these assignments is a *Sampler Instrument*.



Several Zones and Groups in the Instrument Editor.

Zones

A *Zone* is a location into which a single sample (or audio file, if you prefer this term) can be loaded from hard disk or CD ROM. The sample loaded into the Zone is memory resident i. e., it uses the RAM of your computer. A Zone offers various parameters for controlling the playback of the sample. Each Zone allows you to determine the range of notes over which the sample should be heard (Key Range), and the “root key” (Key Note) i. e., the note at which the sample sounds at its original pitch. In addition, sample start, end, and loop points plus volume and several other parameters can be adjusted within the Zone. You can define as many Zones as you wish. Each Zone requires at least one EXS 24 voice when played.

Groups

Imagine a drum kit has been created, with a number of different samples being used in several Zones, mapped across the keyboard. In many musical circumstances, it would be great to be able to treat each of the samples independently with the EXS 24’s sound editing parameters—to alter the decay of the snare, or to use a different cutoff setting for the hi-hat samples, for example.

This scenario is where the *Groups* come in—they allow for the very flexible organization of samples. You can define as many Groups as desired, and can assign each Zone to one of these Groups. In a drum set, for example, you could assign all kick drums to Group 1, all snares to Group 2, all hi-hats to Group 3 and so on.

Why might you want to do this?
A Group makes it possible, for example, to define a velocity range for all assigned Zones, allowing you to choose a specific velocity window in which the grouped Zones should sound. Each Group also features offset parameters for the amplitude envelope and filter settings made in the plug-in window.

It's also possible to play all Zones without defining and assigning even a single Group—in this case, the parameters in the plug-in window work in an absolute manner for all Zones. To clarify, all samples in all Zones will be affected equally by the parameter adjustments made in the plug-in window.

Given that up to 24 EXS24 instruments (dependent on your version of Logic Audio) can be used simultaneously, opening several instances of the EXS24 provides the advantage of a dedicated channel strip for each and every sound you use. This allows full control over the sound (via EXS and effects parameters) during composition and mixdown.

File Types and File Organization

The EXS24 uses the following file types and hierarchical structures:

Audio File

A single sample on your hard disk. The EXS24 is compatible with all audio file formats supported by Logic. Audio files are handled in the EXS24 Instrument Editor, where they can be edited and organized into Sampler Instruments.

Sampler Instrument

A Sampler Instrument points to one or more audio files, and organizes them as multi samples or drum maps, respectively. Within the Sampler Instrument you may assign different samples to different key and velocity ranges, set loop points, and adjust other playback parameters. You can also work with Zones and Groups (see the *Multiple Zones and Groups* section, from page 17 onwards), which always belong to a Sampler Instrument, and are not stored or loaded separately.

Please note: Audio files are ***not*** contained in a Sampler Instrument. The Sampler Instrument simply stores information about an Audio File's name, its parameter settings, and its location on the hard disk. When you delete or rename an audio file, the Sampler Instrument that makes use of it will be unable to find it, so take care when handling audio files.

A Sampler Instrument is the file type that is loaded into the EXS 24 for playing. When you select a Sampler Instrument in the EXS 24's flip menu, the associated audio files are automatically located on the hard disk, and are subsequently loaded into your computer's RAM.

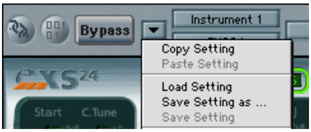
In order to be visible within the EXS 24's Sampler Instrument flip menu, Instruments must be stored in the *Sampler Instruments* sub-folder of the main Logic program folder.

Please note: You can store your Sampler Instruments in any folder on any of your computer's hard drives. To do so, you must create an alias pointing to this folder within the ***Sampler Instruments*** folder located in the ***Logic*** program folder. Please refer to the *File Organization* section, from page 26 onwards.

You can manually load Sampler Instruments from other locations into the EXS Instrument Editor at any time. Such Instruments also appear in the EXS 24's Sampler Instrument load flip menu.

Settings

Settings are used to store all parameter adjustments made in the plug-in window. Every Logic plug-in allows you to store and recall Settings, and the EXS24 is no exception. The Settings for the EXS24 are stored in the *EXS24* folder, which itself is located in the *Plug-In Settings* folder within the main *Logic* program folder.



Important: the Settings that can be stored and recalled in the plug-in window are *not* part of the Sampler Instrument being loaded.

Settings reside above the *Sampler Instruments* in the hierarchy: A Setting contains a pointer to a Sampler Instrument, and when a new Setting is selected, the Sampler Instrument it points to is automatically loaded. As such, Settings are convenient for organizing and accessing your favorite Sampler Instruments. Settings also recall any changes made to parameters within the plug-in window.

Automation

As with every Logic plug-in, the EXS24 can be fully automated. To do so, simply select the desired EXS24 track in the Arrange window's Track List, activate Record, and move the faders, knobs and switches in the EXS24's plug-in window. The Audio Instrument object into which the EXS24 is inserted will send the controller data to Logic. This data will be recorded on the selected track, and will automate the EXS24's faders and switches when played back. You can record automation data in one or multiple takes, and on one or more tracks.

The MIDI controllers used for automation can be edited or created in any of Logic's suitable editors. For a list of the controllers see *MIDI Controller List* section, from page 84 onwards.

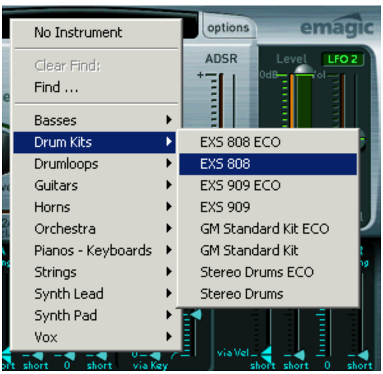
6 File Organization

As your sample library grows, the list of Sampler Instruments will also expand. To aid you in keeping the list of Sampler Instruments manageable, the EXS24 features a sophisticated, but easy to use method of file management.

The Sampler Instrument flip menu directly reflects the folder structure within the *Sampler Instruments* folder. You can choose to sort your Sampler Instruments in groups i. e., basses and guitars—by sound type, alphabetically, or by song.

To organize your Sampler Instruments into a preferred hierarchy:

- Simply create a folder—e. g. "Basses" within the *Sampler Instruments* folder, with your operating system's file management utilities.
- Drag and drop the desired EXS24 Sampler Instruments into this newly created folder and their menu structure will be reflected when clicking on the EXS24 Sampler Instruments flip menu.



Note that you will need to relaunch Logic Audio after changes are made to the folder hierarchy in the ***Sampler Instruments*** folder.

The menu is designed to limit its display to sub-menus of folders that contain EXS instrument files. Other folders are not added to the menu. Aliases pointing to folders which contain EXS instrument files outside the *Sampler Instruments* folder can also be added to the menu. Even the *Sampler Instruments* folder itself can be an alias to a folder on a different drive or location.

When selecting a Sampler Instrument from a sub menu, a bold entry at the top of the root menu is added to indicate the current selection. The sub menu that contains the selected Sampler Instrument is also shown in bold type, as are further sub menus. This makes it easy to trace the file path of the currently loaded Sampler Instrument.

Note that on Mac OS, the hierarchical structure is limited to four (4) levels. There is no limit on the number of files that the menu can display, but the maximum number of sub-menus is 100. This limitation of the hierarchical menu functionality may change with future Mac OS updates.

Saving of Song-related EXS24 Instruments

This feature allows all EXS24 Instruments associated with a song to be saved/loaded into/from a single folder location, which also contains the song file. These Sampler Instruments will then be exclusively associated with this song.

This is useful for two reasons:

- It makes the archiving and handling of songs, including the associated Sampler Instruments, easier.
- It makes it simpler to deal with a particular set of samples that will not be used in another song i. e., vocals, modified drum kits etc.

It works as follows: When opening a Logic song, the EXS24 initially looks for a sub-folder named “Sampler Instruments” in the folder which contains the song file. If such a sub-folder exists, all Sampler Instruments found in this folder are added to the Sampler Instrument flip menu in the EXS24 GUI. This new entry in the Sampler Instrument flip menu will appear as a sub-menu item that matches the song file name. This behavior ensures that the EXS24 will always locate any song-related Sampler Instrument files *before* searching in the global Sampler Instrument folder, found in the Logic program directory.

To make use of this feature:

- Create a new folder for a song and name it.
- Save the song file itself into this song folder.
- Within this song folder, create a folder named “Sampler Instruments”.
- Simply copy/move the Sampler Instrument files required into this folder. Note that only the Sampler Instrument files, *not* the raw samples used by these Sampler Instruments should be copied, except when archiving (or unique samples are used), as discussed below.
- When Logic is booted, the song is loaded, and an EXS24 instance is opened; a new hierarchical menu item will appear within the EXS24 Sampler Instrument flip menu when clicked. This new menu item will retain the song’s name and contains all of the Sampler Instrument entries copied to this folder earlier.
- When saving any newly created or modified Sampler Instruments, ensure that you use the “Save as” function and browse to the “Sampler Instruments” folder inside the new song folder.

When saving on a per-song basis, you should observe the following folder hierarchy:

- The song folder contains the song file and the “Sampler Instruments” folder.
- The “Sampler Instruments” folder contains all Sampler Instruments that are used in this song exclusively.

As the EXS24 automatically locates the audio files associated with Sampler Instruments, it generally does not matter where these audio files are stored. One circumstance, however, where the storage location of the audio files *does* matter is as follows: Should you need to archive the song with all related data, or wish to deal with a particular set of samples that will not be used in another song, you will want to store the *audio files* inside the song folder as well.

This will change the folder hierarchy as follows:

- The song folder contains the song file and the “Sampler Instruments” folder.
- The “Sampler Instruments” folder contains all Sampler Instruments that are used in this song exclusively—e. g. vocals etc.
- For *each* Sampler Instrument used, a separate folder containing the audio files associated with the respective Sampler Instrument.

To assist you in doing this, the EXS24 Instrument Editor provides the following functions:

Instrument > Copy Audiofiles

Copies the audio files of any Sampler Instrument edited in the EXS24 Instrument Editor to the target directory of your choice. A folder for the audio files associated with this Sampler Instrument is created in the target location. The Sampler Instrument file itself is also copied.

Instrument > Move Audiofiles

Moves the audio files of any Sampler Instrument edited in the EXS24 Instrument Editor to the target directory of your choice. A folder for the audio files associated with this Sampler Instrument is created in the target location.

Functions available as Key Commands

The *Backup audiofiles of all USED and ACTIVE instruments of current song* key command copies the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location. All used Sampler Instrument files are also copied.

The *Move audiofiles of all USED and ACTIVE instruments of current song* key command moves the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location.

Searching for Sampler Instruments

As a further navigational enhancement, the EXS24 features a built-in *Find* function, which works in conjunction with the hierarchical menu structure discussed earlier.

In order to minimize the number of Sampler Instruments displayed in the Sampler Instrument flip menu, you can make use of the *Find* function. This will limit the Sampler Instrument flip menu to only display Sampler Instrument names that contain the word “piano” or “bass”, as an example. This will also hide any sub-menus that don’t contain the search word. Simply select *Find* in the Sampler Instrument flip menu and, in the ensuing dialog box, type in the character string (search term) to search for.

The *Clear Find* option in the Sampler Instrument flip menu will display the full menu but does not clear the actual search term typed into the search dialog. To return to the limited menu, simply select *Enable Find*. The selection of Enable/Clear Find allows you to toggle between the two without re-typing the search word.

If you wish to use a different character string, select the *Find* option a second time and type in the desired search term.

7 Sample File Import

The EXS24 is compatible with the popular AKAI S1000 and S3000, SoundFont2, SampleCell and, of course, the EXS 24 native sample formats.

Using EXS24 Files

We strongly recommend that you copy any EXS24 Sampler Instruments shipped on CD ROM to your hard drive(s)—for two reasons: firstly, to always have direct, immediate access to your Sampler Instruments without searching for and inserting CD ROM's, and secondly, to be able to sort your Sampler Instruments according to your needs.

To copy an EXS24-format Sampler Instrument, along with its associated audio files, from CD ROM to your hard drive(s):

- Copy the Sampler Instrument files from the CD into the *Sampler Instruments* folder within the *Logic* folder.
- Copy the associated samples from the CD into the *EXSamples* folder within the *Logic* folder.
- Please note that you can sort your Sampler Instruments to suit your own needs (see the *File Organization* section, from page 26 onwards).
- For advanced users: The EXSP24 file system is able to work with aliases for Sampler Instrument folders. Furthermore, a Sampler Instrument searches for, and finds, all samples it uses on all active hard drive(s)—as long as you do not delete or rename the samples.

Using EXS24 Instruments directly from CD ROM

Normally, the Sampler Instrument and associated samples (audio files) will be stored on your hard disk(s), but on occasion, you may wish, or need, to load an EXS Sampler Instrument from CD ROM.

To use an EXS Sampler Instrument stored on CD ROM:

- Copy the Sampler Instrument file (*not* the associated samples) from the EXS format CD ROM into the *Sampler Instruments* folder.
- When the Sampler Instrument is loaded, ensure that the appropriate CD ROM is in the computer's CD ROM drive. If the appropriate CD ROM (i. e., the one which contains the desired Sampler Instrument, and its associated audio files) is in the drive, the EXS24 will automatically search for the associated samples on all local media. It will locate the CD ROM and will load the Sampler Instrument.
- If the CD ROM is not present, you will be required to insert the appropriate disc and re-load the Sampler Instrument.

Note that aliases/shortcuts may only be used for files stored on hard disk, not on CD ROM.

Importing SoundFont2 Files

To make use of this functionality, simply copy or move your SoundFont2 files into the *Sampler Instruments* folder.

Select the file name in the EXS24 Sampler Instrument load flip-menu and the file will automatically be converted. An EXS Instrument file will be created in the *Sampler Instruments* folder which contains the original SoundFont2 file. The raw samples associated with the Sampler Instrument will be placed in a *SoundFont Samples* folder within the *Logic* program folder.

Should a SoundFont2 Bank file (a Bank contains multiple sounds—a General MIDI bank, for example) be loaded, it will create a *Bank* folder and also a *Samples* folder. These new folders will have the same name as the SoundFont2 Bank file, with the word "Bank" or "Samples" appended.

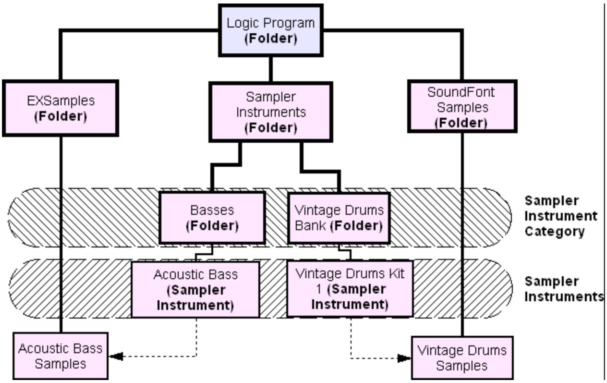
All sounds contained in the bank will automatically have an EXS Sampler Instrument file created and placed into the newly created *Bank* folder. The EXS24 Sampler Instrument flip menu will automatically be updated to reflect the new folder hierarchy. All samples associated with the *Bank* will automatically have a *Samples* folder created inside the *SoundFont Samples* folder which resides in the *Logic* program folder.

As an example, a SoundFont2 bank file named “Vintage Drums” is imported by the EXS24. It contains over 50 individual drum kits from several different vintage drum machines. A new folder named *Vintage Drums.Bank* will be created in the *Sampler Instruments* folder. A second folder named *Vintage Drums.Samples* will be created in the *SoundFont Samples* folder. Both of these folders are found in the main *Logic* program folder.

The Sampler Instrument flip menu hierarchy is updated and the original “Vintage Drums” entry is replaced with a “Vintage Drums.Bank” entry. This new entry is a folder which contains the individual Sampler Instruments, which can be selected and loaded as per usual.

Once conversion is complete, the original SoundFont2 source file(s) can be freely deleted from the hard disk(s).

Please note: You can store your imported Sampler Instruments in any folder on any of your computer’s hard drives. To do so, you must create an alias pointing to this folder within the ***Sampler Instruments*** folder located in the main ***Logic*** **program** folder. Care should be taken when importing samples to ensure that when a song is loaded, the associated Sampler Instruments will be found. Sampler Instruments are only searched for in the ***Sampler Instruments*** folder (or an alias to it). Any Sampler Instruments stored in other locations will not be located and must be loaded manually.



The folder hierarchy of the EXS24.

Importing SampleCell Files

The importation of SampleCell format files is as per that of SoundFont2 files. Simply copy or move your SampleCell files into the *Sampler Instruments* folder.

Select the file name in the EXS24 Sampler Instrument load flip-menu and the file will automatically be converted. An EXS Instrument file will be created in the *Sampler Instruments* folder which contains the original SampleCell file. The raw samples associated with the Sampler Instrument will be placed in a *SampleCell Samples* folder within the main *Logic* program folder.

Once conversion is complete, the original SampleCell source file/s can be freely deleted from the hard disk(s).

Should you import SampleCell or AKAI format Samples, they will appear as a *SampleCell Samples* or *AKAI Samples* folder on the same level as the EXSamples, Sampler Instruments and SoundFont Samples folders. Please refer to the EXS24 folder hierarchy diagram above.

AKAI Import

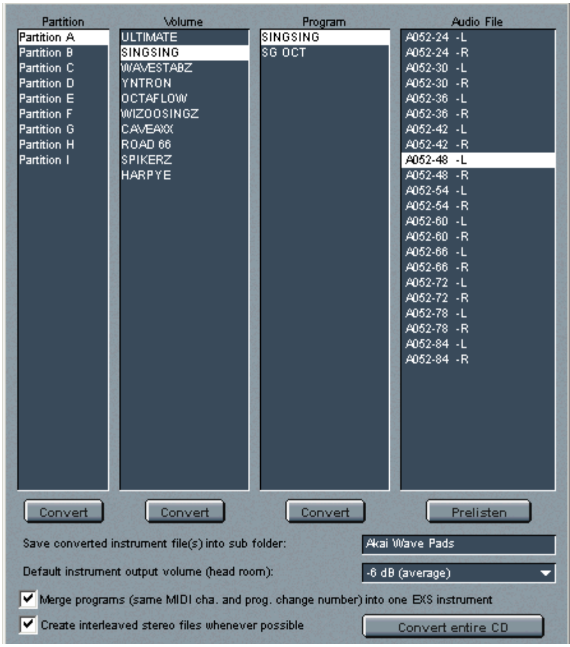
This section discusses the AKAI import procedure. The EXS24 can import samples saved in the AKAI S1000 and S3000 sample formats. The AKAI Import function can be used to import:

- an entire AKAI format CD ROM
- an AKAI Partition
- an AKAI Volume
- an AKAI Program
- an Individual Audio File (sample)

These options have been provided to give you the most flexible and efficient method of dealing with your sample library. There may be a sample or two, or perhaps a particular drum kit which you would like to import from an AKAI CD ROM.

Similarly, you may wish to import the contents of an entire CD ROM in one simple operation, rather than spend the time dealing with individual Partitions, Volumes, Programs and Audio Files.

This way, you can load and audition all of an AKAI CD ROM's programs and files within Logic Audio. Later, at your convenience, you can make use of your operating system's file management utilities to remove or reorganize your imported AKAI sounds, as discussed in the *File Organization* section, from page 26 onwards.



The AKAI Import Window

Using the AKAI Import Function

- Select **Options > AKAI Import**. This will launch a window similar to that shown above, with the text “Waiting for AKAI disk” spread across the four columns.
- Insert an AKAI format sample disc into your CD ROM drive and the AKAI Import window will commence reading the data. Following the reading of the CD ROM, the display will update to show the contents of the CD ROM. The *Partition* column will display information, with Partition A, Partition B a. s. o. entries listed.

Note that the reading of a CD ROM may take some time, dependent on the amount of sample data and file structure of the disc. In addition, the speed of the CD ROM mechanism, bus speed, memory and other factors—such as the CD ROM driver—can affect performance.

- To view the contents of the Partitions, click once on the appropriate entry with the mouse button. This will display the *Volume* information contained within the *Partition*.
- To continue through the architecture of the CD ROM, click on the *Volume* entries to view any *Programs* contained therein, and on the *Program* entries, to view the raw *audio files* (samples).
- Once you have made your selection of Partition, Volume or Program, click on the Convert button beneath the appropriate column. The selected Partition, Volume or Program will be imported along with all associated audio files.
- Any audio files imported will be stored within a folder which matches the name of the *Volume*. This folder is created within the *Logic* > *AKAI Samples* folder. The Sampler Instrument(s) created by the import procedure matches the Program name(s). It is placed inside the Sampler Instruments folder, or a sub folder as determined by the *Save converted instrument file(s) into sub folder* parameter discussed in the *AKAI File Organization* section, from page 37 onwards.
- Should you wish to convert an entire AKAI CD ROM, click on the "Convert entire CD" button found to the lower right of the AKAI Sample Import window.

ℹ Sub-folders with the name of the volume are created when converting a partition. If a volume only contains one program, no sub-folder is created.

ℹ Sub-folders with the name of the partition are created when converting more than one partition.

AKAI File Organization

In the following graphic, the "AKAI Wave Pads" folder contains several "Volumes", which contain "Programs".

The "SINGSING" folder contains two patches—SG OCT and SINGSING. The SG OCT and SINGSING Sampler Instruments are stored in the *Sampler Instruments > AKAI Wave Pads > SINGSING* folder.

The audio files associated with the SG OCT and SINGSING Sampler Instruments appear in the *AKAI Samples > SINGSING* folder.

To illustrate what occurs with the file structure when a Program is imported, rather than a Volume, the TABLOIDZ and TB SWEEP Programs were converted. These programs appear as TABLOIDZ.EXS and TB SWEEP.EXS in the Sampler Instruments folder.

The audio files associated with the TABLOIDZ Sampler Instrument appears in the *AKAI Samples > TABLOIDZ* folder.

Sampler Instrument management works with AKAI samples imported from CD ROM, in the same fashion as with other sample formats. Given the different file structures used by many AKAI format discs, however, you should take care to follow these guidelines.

- Create a shortcut to any folder on your hard disk/s which contains your AKAI sample library (or where you wish to store it). Name the shortcut "AKAI Samples" and all converted AKAI CDs/samples will automatically be saved in this destination folder. The “AKAI Samples” shortcut must be placed within the Sampler Instruments folder.

- If converting an entire CD ROM, you can create a shortcut with the sample CD’s name—“Dance MegaSynth” for example. This can be placed in the Sampler Instruments folder directly, or as a sub-folder within the AKAI Samples folder. The advantage with the second method is that all imported AKAI Instruments will be placed under the AKAI Samples sub-menu within the EXS 24’s load window flip-menu.
- N.B.—Assuming that an entire CD has been converted, you will find an “AKAI Samples” folder (which actually contains the raw sample data) and several “Partition” folders within the destination folder. The Partitions may contain several folders which bear the name of the imported instrument(s). The “.EXS” files (i. e., the EXS Instruments) may be contained in either the Instrument folders or in the Partition folders.

Additional AKAI Import Parameters

Within the AKAI Import window, you will find additional parameters listed below the four gray column areas. We will discuss these in their order of appearance.

Save converted instrument file(s) into sub folder.

Entering a name into this parameter field is achieved by clicking once with mouse and typing in the desired name, followed by pressing ⌅. In the example shown within the *AKAI File Organization* section, from page 37 onwards, an “AKAI Wave Pads” folder was created.

All imported Volumes and Programs will automatically be added to this menu, and folder structure, until the name is changed. This facility may be useful, particularly when importing an entire CD, to create a folder name which reflects the CD ROM’s name. Alternately, you may wish to use a category name, such as “Strings”. This way, any imported Programs or Volumes will be added to the “Strings” category.

If an existing category name is used, the imported Sampler Instrument will be *added* to the folder/menu. It will *not* create a new menu entry/folder of that name.

Default instrument output volume (head room)
This parameter is extremely useful for many AKAI CD ROM’s. Please select this option before converting a CD ROM.

- For drum CDs, select a headroom value of –3 up to zero dB.
- For piano/string/pad CDs, a headroom value of –9 is recommended, or the sound may/will clip with polyphonic use of these types of instruments.
- In cases where you’re not sure of which headroom value to select, choose –6 dB (average).

The “Output Volume” can also be changed later via **Options** > **Additional Parameter** > **Output Volume**.

Merge programs (same MIDI cha. and prog. change number) into one EXS instrument
This parameter is set to *OFF* by default. Its use is dependent on the structure of program material on the CD ROM being imported.

To explain, many CD ROM’s created for AKAI samplers may feature several programs which contain single velocity layers for an instrument. AKAI samplers require the loading of an entire volume, or all necessary single programs, to be able to hear/play all velocity layers. All of these single programs are automatically assigned to the same MIDI channel and also react to the same MIDI program change number.

The EXS24 AKAI Conversion intelligently checks for these settings and will build a single EXS Sampler Instrument out of multiple single programs. In general, this type of behavior is desirable with these types of CDs. When importing samples of this type, this option should be set to ON.

The same is true for drum CD ROM's where single programs contain one instrument from a complete drum kit (kick/snare/hi-hat/etc. as separate entities) You'll probably want these single AKAI programs to be merged into a single EXS Sampler Instrument as a full drum kit.

There are, however, a number of AKAI CD ROM's where a single program of an AKAI Volume contains the entire instrument, and where other programs in the same Volume have the same MIDI channel and MIDI prog. change number preset. On this type of CD ROM, use of the merge programs parameter is not desirable and the option should be set to *OFF*.

Create interleaved stereo files whenever possible
This option should always be left enabled, as interleaved files offer better performance within the EXS24. When executing an AKAI conversion, some audio files are created as split stereo *and* as interleaved stereo files.

The detection of when it is possible to build an interleaved file is based on information stored with both the AKAI Program and audio files. Both the left and right files must have the same settings; otherwise they can not be used to create an interleaved file/multiple interleaved files.

Prelisten Function

The AKAI Import window features a "Prelisten" button, which is found below the Audio Files column. This facility allows you to individually audition AKAI audio files before deciding whether or not to import them.

Following selection of an individual file (sample) within the Audio Files column:

- Press *Prelisten*.
- This will commence playback of the selected audio file and the Prelisten button will update, with the word "Stop" appearing on the face of the button.

- The selected Audio File will loop continuously until the “Stop” button is pressed.

8 The Parameters of the EXS24

Plug-in Window Parameters

The parameters described in this section are most easily manipulated from within the *Editor* view of the plug-in window. If you see a number of horizontal sliders on a blue background, please switch from the *Controls* view to the *Editor* view, by clicking on the flip menu in the gray area of the plug-in window.

Common Parameters

Flip Menu for Selecting a Sampler Instrument

This menu allows the selection and loading of a Sampler Instrument into your computer's RAM. In order to appear within this list, a Sampler Instrument must reside in the *Sampler Instruments* subfolder of Logic's program folder.

Edit Button

This button opens the currently loaded Sampler Instrument in the EXS24 Instrument Editor.

Options Button

Clicking the *Options* button launches a menu in which the following options appear:

- *Recall default EXS 24 settings* recalls a “neutral” setting for all parameters in the plug-in window.
- *Recall settings from Instrument* command manually recalls the original parameter settings of the loaded Sampler Instrument. This parameter is extremely useful if you’ve been over zealous with your “tweaking”.
- *Save settings to Instrument* parameter “burns” the current settings of the plug-in window into the Instrument file. When the Instrument is reloaded, these settings are restored in the plug-in window.
- *Delete Settings from Instrument* removes the stored settings from the Instrument.
- Additional Parameter sub-menu. Please refer to the *Additional Parameters* section, from page 59 onwards.
- *Rename Instrument…* allows the renaming of the currently opened Sampler Instrument. When invoked, a file dialog box will open. This will overwrite the existing Instrument name.
- *Save Instrument as…* allows the storage of the currently opened Sampler Instrument under a different name. When invoked, a file dialog box will open.
- *Delete Instrument* will delete the opened Sampler Instrument.
- *AKAI Convert* launches the AKAI Convert window (see the *Using the AKAI Import Function* section, from page 36 onwards). This menu option accelerates working with AKAI samples, as you do not need to open the EXS 24 Instrument Editor.
- *Soundfont* and *SampleCell Convert* will launch a dialog with instructions on performing these conversions.

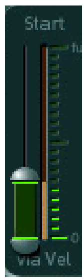
Legato/Mono/Poly Switches

These switches determine the number of voices used by the EXS24:

- When *Poly* is selected, the maximum number of voices is set via the numeric field alongside the *Poly* button. To change the value, click and hold with your mouse, and drag up or down to increase/decrease polyphony.
- When *Mono* or *Legato* is selected, the EXS24 is monophonic, and uses only one voice.
- In *Legato* mode, Glide is only active on tied notes. Envelopes are not retriggered when tied notes are played (single trigger).

In *Mono* mode, Glide is always active and the envelopes are retriggered by every note played (multi trigger).

Start



This slider adjusts the start point for the sample(s); higher values move the sample start point towards the end of the sample. The maximum value is where the loop start point begins. The start point can be modulated by velocity: the upper half of the slider determines the start point for minimum velocity, the lower half for maximum velocity. By clicking and dragging the area between the two slider segments, you can adjust both simultaneously.

Time

The *Time* knob allows Filter and the Amplitude envelope times to be reduced (shortened), dependent on the incoming note pitch. This is especially useful when simulating acoustic instruments. On many acoustic instruments, higher notes have a shorter decay phase than lower notes. The higher the *Time* parameter's value, the more pronounced the effect, with envelope times getting shorter as you play higher pitched notes on the keyboard.

Curve

Curve allows the attack slope characteristics of the filter and amplitude envelopes to be varied. When centered, the envelope slopes are linear. Increasing the *Curve* value leads to falling exponential slopes, and reducing its value yields rising exponential slopes.

Pitch Parameters

Coarse Tune

Offsets the pitch of the sample(s) in semitones by up to ±2 octaves. The middle position of the slider (which can be set by clicking the small *0* button) leaves the pitch unaltered.

Glide

The effect of this slider depends on the setting of the *Pitcher* slider: When *Pitcher* is centered, *Glide* determines the time it takes for the pitch to slide from one note to another (portamento). When the *Pitcher* parameter is set to a value above its centered value, *Glide* determines the time it takes for the pitch to glide down from this higher value back to its normal value. When *Pitcher* is set to a value below the centered value, the pitch glides from this lower setting back up to the normal value.

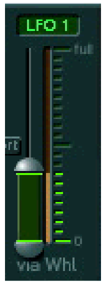
Pitcher

The *Pitcher* slider works in conjunction with the *Glide* slider: When the *Pitcher* is centered (which can be set by clicking the small *Port* button), *Glide* determines the portamento time. When *Pitcher* is set to a higher or lower value, a pitch envelope is activated. In this scenario, *Glide* determines the time it takes for the pitch to glide from the higher/lower *Pitcher* setting back to the original value. The *Pitcher* parameter can be modulated by velocity: the upper half of the slider determines the setting for maximum velocity, the lower half for minimum velocity. By clicking and dragging in the area between the two slider segments you can move both simultaneously.

Please note that the upper half of the *Pitcher* slider can be set above the center position, and the lower half below the center position. When the *Pitcher* sliders are set in this fashion, lower velocity values cause the pitch to rise from the lower setting back to the original note pitch, while higher values cause it to fall from the higher setting back to the original value. In other words: the polarity of the pitch envelope can be changed according to velocity values.

When both halves of the pitcher slider are set below or above the centered position, either a low or high velocity will slide up/down to the original pitch. Dependent on the position of the upper/lower halves of the slider in relation to the center position, the time required for the slide up/down to the original note pitch can be adjusted independently for both soft/hard velocities.

LFO Pitch Modulation



The *LFO1/2 via Whl* slider defines the intensity (or depth) of pitch modulation from LFO 1 or LFO 2. The desired LFO can be selected by clicking on the label above the slider. The intensity of LFO pitch modulation can be controlled by the Modulation wheel on an attached MIDI keyboard. The upper half of the slider determines the intensity when the Modulation wheel is set to its maximum value, and the lower half when set to its minimum value. By clicking and dragging in the area between the two slider segments, you can simultaneously move both.

Additional Pitch Parameters in the Controls View

The additional parameters described here can be found in the *Controls* view of the plug-in window; marked with a dot to the left of their names. You can change to this view by selecting *Controls* from the flip menu in the upper window area. As an alternative press the Control View button to the left of the Bypass button. A blue panel with all additional parameters will open below the EXS24 (as shown). These parameters can also be found in the **Options > Additional Parameter** menu.

Pitch Bend Up

The amount of pitch bend (in semitones) that can be introduced by moving the pitch bend wheel to its maximum position.

Pitch Bend Down

The amount of pitch bend (in semitones) that can be introduced by moving the pitch bend wheel to its minimum position. When *Linked* is selected, the *Pitch Bend Up* value is used.

Transpose

This parameter allows you to transpose the EXS24. In contrast to the *Coarse Tune* parameter, *Transpose* not only affects the pitch, but also moves the Zones according to the *Transpose* setting.

Fine Tune

Allows the EXS24 to be fine-tuned.

Filter Parameters

Filter On/Off Switch

This button switches the filter section on or off. Please note that the knobs and buttons in the silver panel area and the Filter Envelope are active only when the filter is turned on. When the filter section is turned off, the EXS 24 is far less CPU-intensive.

Filter Type Switches

The lowpass filter of the EXS 24 offers different modes which can be distinguished by the steepness of their slope: the higher the steepness (expressed in dB/octave), the more pronounced the filtering effect. You can choose between 12, 18, and 24 dB/octave slopes.

The 24 dB mode offers two variations:

- *24 dB classic*—the sound loses bass and gets thinner when high *Resonance* values are used.
- *24 dB fat*—compensates for this side-effect and delivers strong bass frequencies, even at high resonance settings

Drive

This knob allows the filter input to be overdriven. Turning *Drive* up leads to a more dense and saturated signal, with additional harmonics being introduced/becoming audible.

Cutoff

The cutoff frequency of the lowpass filter. As you turn this knob to the left, an increasing number of high frequencies are filtered from the signal. The *Cutoff* value also serves as the starting point for any modulation involving the filter.

Resonance

Turning up *Resonance* leads to an emphasis of the frequency area surrounding the frequency defined by the *Cutoff* parameter. Very high *Resonance* values introduce self oscillation, and cause the filter to produce a sound (a sine wave) on its own.

Simultaneous Control of Cutoff and Resonance

By clicking and dragging on the chain symbol located between the *Cutoff* and the *Resonance* knobs, you can control both parameters simultaneously: vertical mouse movements alter *Cutoff*, and horizontal mouse movements affect *Resonance* values.

Key

This knob defines the amount of filter cutoff frequency as determined by note number. When *Key* is fully turned to the left, the cutoff frequency is not affected by the note number, and is identical for all notes played. When *Key* is set fully right, the cutoff frequency follows the note number 1:1—if you play one octave higher, *Cutoff* is also shifted by one octave. This parameter is very useful in avoiding overly filtered high notes.

LFO Cutoff Modulation

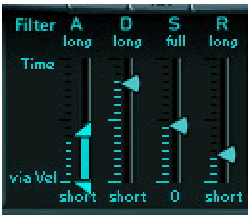
The *LFO1/2 via Whl* slider determines the intensity of cutoff modulation from LFO 1 or LFO 2. The desired LFO can be selected by clicking on the label above the slider. The intensity of LFO cutoff modulation can be controlled by the Modulation wheel: the upper half of the slider determines the intensity when the Modulation wheel is set to its maximum value, the lower half when it is set to its minimum value. By clicking and dragging in the area between the two slider segments, both can be moved simultaneously.

ADSR via Vel

This slider determines the intensity of cutoff modulation by the Filter ADSR envelope. It works bi-directionally, and shifts *Cutoff* either up (+ area) or down (- area) via the envelope. In a centered position (which can be set by clicking on the small *0* button), *Cutoff* is not affected by the envelope. The *ADSR* parameter can be modulated by velocity: the upper half of the slider determines the modulation intensity for maximum velocity, the lower half for minimum velocity. By clicking and dragging in the area between the two slider segments, you can move both simultaneously.

Filter Envelope

This is an ADSR envelope generator for modulating the cutoff frequency of the filter. This envelope does not have any effect when the filter section is turned off. The envelope offers *Attack*, *Decay*, *Sustain*, and *Release* parameters.



The attack time can be reduced (shortened) by velocity: the upper half of the slider determines the time for minimum velocity, the lower half for maximum velocity. By clicking and dragging in the area between the two slider segments, both can be moved simultaneously.

Additional Filter Parameters in the Controls View

The additional parameters described here can be found in the *Controls* view of the plug-in window; marked with a dot to the left of their names. You can change to this view by selecting *Controls* from the flip menu in the upper window area. As an alternative press the Control View button to the left of the

Bypass button. A blue panel with all additional parameters will open below the EXS24 (as shown). These parameters can also be found in the **Options > Additional Parameter** menu.

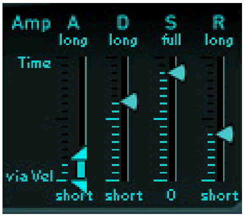
Filter via Vel

Determines the amount of *Cutoff* modulation by velocity. Any incoming velocity values are added to the current *Cutoff* value.

Volume and Pan Parameters

Amp Envelope

This is an ADSR envelope generator for controlling the sound's level over time. It offers *Attack*, *Decay*, *Sustain*, and *Release* parameters.



The attack time can be reduced by velocity: the upper half of the slider determines the time for minimum velocity, the lower half for maximum velocity. By clicking and dragging in-between the two slider segments, both can be moved simultaneously.

Level via Vel

Controls the volume of the sound. The *Level* parameter can be modulated by velocity: the upper half of the slider determines the volume for maximum velocity, the lower half for minimum velocity. By clicking and dragging in the area between the two slider segments, you can move both simultaneously.

LFO Level and Pan Modulation

The *LFO1/2 via Whl* slider determines the intensity of level and/or pan modulation of LFO 1 or LFO 2. The desired LFO can be selected by clicking on the label above the slider. Modulation intensity can be controlled by the Modulation wheel. As both level and pan modulation are controlled by this slider, the following should be taken into account:

- When both segments of the slider are set above the centered position, the lower segment determines the intensity of level modulation when the modulation wheel is set to its minimum value; the upper segment sets the intensity when the wheel is set to its maximum value.
- When both segments of the slider are set below the centered position, the lower segment determines the intensity of level modulation when the modulation wheel is set to its maximum value; the upper segment sets the intensity when the wheel is set to its minimum value.

By clicking and dragging in the area between the two slider segments, you can move both simultaneously. Please note that pan modulation only works when the EXS24 is used in stereo mode.

Additional Volume Parameters in the Controls View

The additional parameters described here can be found in the *Controls* view of the plug-in window; marked with a dot to the left of their names. You can change to this view by selecting *Controls* from the flip menu in the upper window area. As an alternative press the Control View button to the left of the Bypass button. A blue panel with all additional parameters will open below the EXS24 (as shown). These parameters can also be found in the **Options > Additional Parameter** menu.

Output Volume

The main volume parameter for the EXS 24. Move this slider to find the right balance between avoiding distortion and getting the best (highest) resolution in the channel fader and the *Level via Vel* slider.

Key Scale +/–

This parameter modulates the sound's level by note number (i. e., position on the keyboard). Negative values increase the level of lower notes. Positive values increase the level of higher notes.

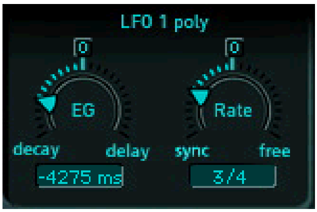
Velocity Offset

This parameter adds or subtracts a user-definable and constant value to or from the velocity. This parameter is useful for the adjustment of a Sampler Instrument to match your playing style which can then be saved as an Instrument or Setting. This parameter is independent of any similar velocity setting parameters in Logic's MIDI track parameters.

LFO Parameters

LFO 1 EG

This knob allows LFO 1 to be faded out (*Decay* area) or faded in (*Delay* area). In the centered position (which can be set by clicking on the small *0* button), the LFO intensity is constant.



LFO 1 Rate

This is the frequency of LFO 1. It can be set in note values (left area), or in Hertz (right area). In the centered position (which can be set by clicking on the small *0* button), the LFO is halted and generates a constant modulation value at full level (*DC* = "Direct Current").

This allows you to perform a nice trick: Set up an LFO to modulate, say, cutoff (see the *LFO Cutoff Modulation* section, from page 51 onwards), with the modulation wheel controlling the LFO's intensity. Then set the LFO's rate to DC. As the LFO's modulation intensity is controlled via the modulation wheel, you can now make use of the modulation wheel to manually open the filter.

Waveform for LFO 1 and LFO 2

These two switches allow the selection of the waveform type used by LFO 1 and LFO 2. A selection of; Triangle, falling and rising Sawtooth, Square up and Square down, a random stepped waveform, and a smoothed random waveform is available for each LFO.

LFO 1 is a polyphonic LFO with key sync. This means that when LFO 1 is used, each voice of the EXS24 has its own discrete LFO. When a note is played, the LFO corresponding to that voice starts its cycle. This scheme means that the LFO cycles of each voice played are not in sync and operate independently of each other. This opens up a range of modulation possibilities. As an example—the LFO of one voice could generate the maximum modulation value, while the LFO assigned to another voice could output its minimum value. This extremely flexible approach can result in some very lively modulations.

In contrast, LFO 2 is a monophonic LFO without key sync. This means that LFO 2 runs continuously, and is not restarted by the triggering of a new note. All voices are modulated by the sole LFO, so the degree of modulation at any given time is the same for all voices. This results in a rather synthetic-sounding modulation.

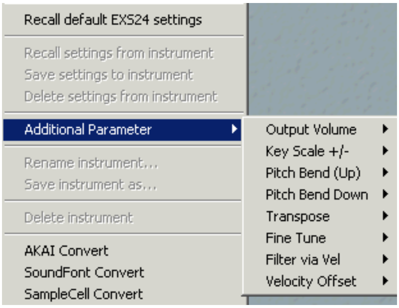
Use these different characteristics to tailor the sound to your needs.

LFO 2 Rate

The frequency of LFO 2. It can be set in note values (left area), or in Hertz (right area). In the centered position (which can be set by clicking on the small *0* button), the LFO is halted, and generates a constant modulation value with full level (DC = "Direct Current"). Again, don't overlook this feature if you want to control an LFO-modulated parameter directly via the modulation wheel (see the *LFO 1 Rate* section, from page 57 onwards).

Additional Parameters

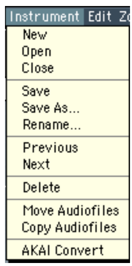
Many of the parameters accessible from the Controls view of the EXS24 can now also be found in the **Options > Additional Parameter** menu.



The inclusion of these options allows faster and simpler access to these functions without the need to switch between the Controls and Editor views.

Instrument Editor Parameters

Instrument Menu



New

Creates a new, empty Sampler Instrument. In order to load a sample, you will need to create a new Zone (see below).

Open

Allows an existing Sampler Instrument to be loaded into the Instrument Editor.

Close

Closes the currently opened Sampler Instrument. If you have made any changes to the Instrument, you will be asked if you would like to save them.

Save

Saves the currently loaded/edited Sampler Instrument. When you create a new Instrument and save it for the first time, you will be asked to give it a name. If you have edited an existing Sampler Instrument and save it via this command, the existing file name is used and the old version is overwritten.

Save As…

This command also saves the currently loaded/edited Sampler Instrument. When Save As… is used, you will be prompted to give the file a name. Use this command when you want to save a copy of an edited Sampler Instrument, rather than overwriting the original version.

Rename

This command allows you to rename the loaded Sampler Instrument. The renamed version replaces the previous version on the hard disk.

Previous/Next

Selects the previous or following Sampler Instrument allowing you to quickly switch between several Sampler Instruments opened for editing.

Delete

This deletes the currently opened Sampler Instrument.

Move Audiofiles

Moves the audio files of the selected Sampler Instrument into a desired folder location. Use of this option will launch a standard operating system file navigation/browse utility. You may browse to an existing folder or enter a new name, as desired. If no folder name is entered, a new folder will be created which matches the Instrument name, and all audiofiles will be moved into this folder.

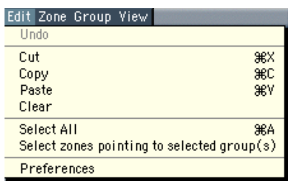
Copy Audiofiles

Operation is as per the *Move Audiofiles* function, but files are duplicated, rather than moved, to the nominated folder. This facility should be used as part of your working methods when creating Logic songs, as discussed in the *Saving of Song-related EXS24 Instruments* section, from page 27 onwards.

AKAI Convert

Opens a window in which you can convert files from an AKAI format CD ROM (see the *Using the AKAI Import Function* section, from page 36 onwards).

Edit Menu



Undo

Allows the most recent change to the Sampler Instrument to be undone.

Cut, Copy, Paste

The standard commands for cutting, copying and pasting values. In addition to values you may also cut, copy, and paste selected Zones and Groups.

When multiple Zones and Groups are cut, copied, or pasted simultaneously, the Group assignments of the Zones are retained.

Clear

Deletes the currently selected Zone or Group. Clear can be undone with the **Undo** command.

Select All

Selects all Zones and Groups of the loaded Sampler Instrument.

Select Zones pointing to selected group(s)

This command automatically selects all Zones that point to one or multiple selected Groups.

Preferences

This parameter opens the Preferences window where you can:

Choose the interpolation quality used by the EXS24. When *SR Conversion* is set to *Best*, the highest possible sound quality is maintained when transposing samples. It should be noted that this option requires additional CPU cycles over the *Original* setting, which will be adequate in most cases.

Select the format in which the EXS24 handles the loaded sample data via the *Sample Storage* parameter. When set to *Original*, the samples are loaded into RAM at their original bit depth, and are converted to Logic's internal 32 Bit floating format on playback. When *32 Bit Float* is selected, the samples are stored and loaded in this format. This eliminates the need for any realtime conversion, meaning that the EXS24 can handle the sample data more efficiently and can play back more voices simultaneously. It should be noted that this requires twice as much RAM for 16 bit samples, and a third more RAM for 24 Bit samples.

Use the *Velocity Curve* parameter, which determines the EXS24's responsiveness to velocity values received from your

MIDI keyboard. Negative values increase the response to soft key strikes, and positive values decrease it.

Use the *Search Samples On* parameter to nominate the drive from which AKAI CDs should be read. You may either choose the CD drive normally used by the operating system or—if your computer has a SCSI-Interface—the SCSI ID of a connected SCSI CD ROM drive.

Drives can be selected individually, or grouped as follows:

- *Local Volumes* internal storage media (hard disks and CD ROM mechanisms) attached to or installed in the computer directly.
- *External Volumes* storage media accessible over a network.
- *All Volumes* both internal and network media are scanned for appropriate data.

It should be noted that selecting External or All Volumes may result in a dramatic increase in the time required by the EXS24 to find and load Sampler Instruments and files.

The following preferences are particularly useful when used in conjunction with the function discussed in the *Load Multiple Samples* section, from page 65 onwards.

Read key note from:

- *file/filename*—initially reads information about the root key from the file itself (in the header of AIFF or WAVE) when loading an audio file into a zone. If no information of this type exists in the file header, a smart analysis of the filename may detect a root key. If this second method doesn't provide any useful results, C3 will be used as the default root key in the zone.
- *filename/file*—as above, but vice versa, with the filename read first, and the header read second.
- *file only*—reads from the file header only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.

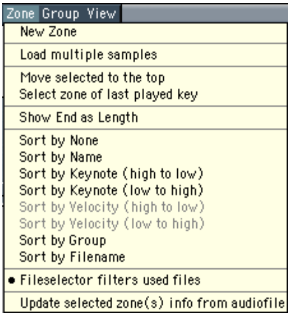
- *filename only*—reads from the filename only. If no root key information exists, C3 will automatically be assigned to the zone as the root key.

Key note at filename pos.

When loading an audio file into a zone this option is used for the analysis of a root key in the filename. Possible options in the flip menu are “Auto”, or numerical values from 1 to 30.

- “Auto” is the recommended value. It provides a smart analysis of numbers and keys in the filename. A number in the filename can be recognized regardless of its format i. e., “60” or “060” are both valid. Other valid numbers can range between 21 and 127. Numerical values outside of these are generally just version numbers etc. A key number is also a valid possibility for this use—“C3”, “C 3”, “C_3”, “A-1”, “A -1” or “#C3”, “C#3”, for example. The possible range is “C-2” up to “G8”.
- There may be cases where a sound designer has used multiple numbers in a filename, which is common with loops, with one value being used to indicate tempo. e. g. “loop60-100.wav”. In this situation, it isn’t clear which, if either of the numbers, indicates a root key or something else: 60 or 100 could indicate the file number in a collection, tempo, root key etc. You can set a value of “8” to read the root key at a position (letter/character) “*8*” of the filename—namely the 100 (E6). Alternately, setting a value of “*5*” will select the 60 (C3) as the root key position.

Zone Menu

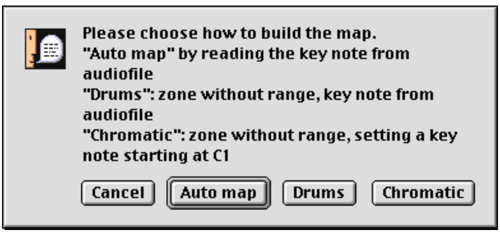


New Zone

Creates a new Zone in the currently loaded Sampler Instrument.

Load Multiple Samples

Allows several samples to be loaded in one operation. The Instrument Editor reads the key note from samples, and places the samples into new Zones. These Zones are automatically created. The key note is located in the middle of the Zone, with Zone borders being determined by the Zone borders of neighboring Zones. Zones with identical key notes will be layered. A root key is automatically determined by use of the settings described in the *Preferences* section, from page 62 onwards.



Three automatic mapping functions are available when loading multiple samples:

- *"Auto map"* uses the key note information (root key) stored with the audio files and places the samples (as zones) over the keyboard range. The number of keys that constitute a zone is intelligently determined by the placement of neighboring zones.
- *"Drums"* uses the key note information (root key) stored with the audio files. Each zone contains a single (i. e., one) key on the keyboard determined by the key note information.
- *"Chromatic"* ignores all key note information (root key) of the audio files and places the samples on the keyboard in chromatic order, starting at C1.

Move selected to the Top

When this option is activated, the parameter window of a selected Zone is moved to the top of the onscreen Zone listing, and is automatically opened.

Select zone of last played key

When active, this menu option allows you to switch between Zones by pressing a key on the onscreen keyboard, or via a connected MIDI keyboard.

Show End as Length

When this option is activated, the sample length is shown in the Zone's parameter window, rather than the end point value.

Sort by…

These commands determine the order in which the Zone windows are displayed:

Sort by None: The Zones are shown in the order in which they were created.

Sort by Name: The Zones are sorted alphabetically.

Sort by Keynote (high to low): The Zones are sorted according to their *Key Note* settings; the higher key notes are displayed at the top of the list.

Sort by Keynote (low to high): The Zones are sorted according to their *Key Note* settings; the lower key notes are displayed at the top of the list.

Sort by Velocity (high to low): The Zones are sorted according to their *Velocity Range* settings; the higher velocity ranges are displayed at the top of the list.

Sort by Velocity (low to high): The Zones are sorted according to their *Velocity Range* settings; the lower velocity ranges are displayed at the top of the list.

Sort by Group: The Zones are shown in the order of their Group assignment.

Sort by Filename: The Zones are sorted alphabetically according to the names of their audio files.

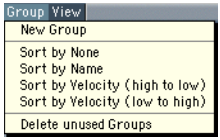
Fileselector filters used Files

Is this option active only audio files which have not already been loaded (ie: are not in RAM) are displayed in the file selection dialog.

Update selected zone(s) info from audiofile

This option is used when editing loops in an external sample editor. It reads loop settings from the audio file and updates the settings of the Zone accordingly. In addition, this feature also validates the loop's length and start position.

Group Menu



New Group

Creates a new Group.

Sort by…

These commands determine the order in which the Group windows are displayed:

Sort by None: The Groups are shown in the order in which they were created.

Sort by Name: The Groups are sorted alphabetically.

Sort by Velocity (high to low): The Groups are sorted according to their *Velocity Range* settings; the higher velocity ranges are displayed at the top of the list.

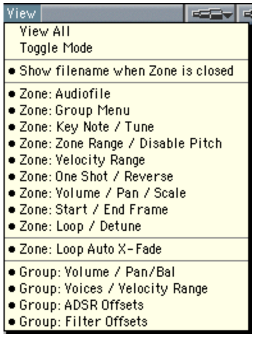
Sort by Velocity (low to high): The Groups are sorted according to their *Velocity Range* settings; the lower velocity ranges are displayed at the top of the list.

Delete unused Groups

This command deletes all Groups that do not have a Zone assignment. This deletion can be undone with the **Undo** command.

View Menu

This menu offers display options for the Zone and Group windows:



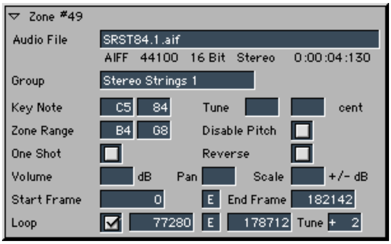
- **View All** shows all available parameters. When this option is activated, you can manually prevent certain parameters from being displayed by deselecting the corresponding option from the menu.
- When **Toggle Mode** is on, only one parameter will be displayed at a time. You can select the desired parameter by selecting the corresponding option from the menu.
- When **Show filename when Zone is closed** is activated, the names of loaded audio files are shown in the title bars of closed Zone windows.

The following menu entries allow you to select the parameters you wish to be displayed.

ℹ Hint: When working with the Zone and Group windows, it can be useful to close or open all windows at once. To do so, click on one of the triangles while pressing ⌥.

ℹ To set a switch parameter for a number of (selected) Zones simultaneously, hold ⌘ when clicking the switch.

Zone Parameters



Zone Name

New Zones are numbered consecutively. Double-clicking on a Zone's number allows you to enter a name instead.

Audio File

A file selection dialog can be opened by clicking on the gray field next to the "Audio File" label. This will allow you to select and load a sample into a Zone. When loaded, the sample's name is displayed in the gray field. Additional information about the sample is displayed beneath its name (sample format, sampling frequency, bit depth, mono/stereo status and sample length).

Group Menu

Group allows you to assign a Zone to an existing Group. When *No Group* is selected, the Zone is only affected by the parameters of the plug-in window.

Key Note/Tune

Key Note allows you to determine the note at which the sample will sound with its original pitch. The *Key Note* is displayed as both a note name and a numerical value. The *Tune* and *cent* fields allow fine-tuning of the sample in semitone and cent increments.

Zone Range/Disable Pitch

The two *Zone Range* parameters allow you to define a key range for the Zone. When *Disable Pitch* is activated, the sample is always played at its original pitch, regardless of the note number.

Velocity Range

Activation of the *Velocity Range* option, in conjunction with the use of the two *Velocity Range* parameters allows you to define a velocity range for the Zone.

One Shot/Reverse

Activating *One Shot* causes the Zone to ignore the length of notes used to trigger the sample—the sample is always played to the end. This option is useful for drum samples, where you often don't want the MIDI note length to affect sample playback. *Reverse* plays the sample from its end to its beginning. This option works non-destructively, leaving the audio data in the sample unchanged.

Volume/Pan/Scale

Volume adjusts the volume of the Zone.

Pan adjusts the pan position of the Zone. This parameter only works when the EXS24 is used in stereo.
Scale—Negative *Scale* values make notes lower than the note position defined by the *Key Note* parameter sound louder than higher ones; positive values have the opposite effect. Use this parameter for balancing the volume of a sample across the selected key range.

Start/End Frame

The *Start* and *End* parameters set the sample's start and end points, respectively. Clicking on the small *E* button(s) between the two values will launch Logic's Sample Editor, allowing you to set the start and end points graphically.

Loop

The loop parameters become visible when this option is activated, and the sample will loop when sustained MIDI notes are received.

Loop Start, *Loop End*—You can define discrete loop start and end points in these fields, allowing you to cycle (loop) a portion of the audio file. Data entry is via the mouse as slider, or by double-clicking and directly typing in a value. Clicking on the small E button(s) between the two values will launch Logic's Sample Editor, allowing you to set the loop start and end points graphically: Loop Start is represented by the LS marker and Loop End, by the LE marker.

Tune—This parameter allows the tuning of the looped portion of the audio file to be different to that of the non-looped portion. It is adjustable in cent increments (±50 cents).

Loop Auto X-Fade

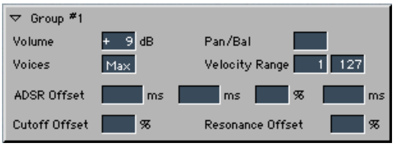
Auto Crossfade—The EXS 24 is the first sampler that allows for non-destructive crossfade-looping in realtime. In a crossfaded loop, there is no hard "cut" between the loop end and loop start points. Rather, the loop end and start points are crossfaded for a smooth transition. This is especially convenient with samples that are hard to loop, and would normally produce clicks at the transition point i. e., the "join" in the loop.

The *Auto Crossfade* field allows a pre-determined value (in milliseconds) to be used as a default when the auto crossfade option is enabled. The higher the value, the longer the crossfade, and the smoother the transition between the loop end and start points.

EqPower—EqPower allows you to enable an exponential crossfade curve that causes a volume boost of 3 dB in the middle of the crossfade range. This will fade out/fade in the joined portions of a loop at an equal volume level.

The “perfect” settings for the crossfade parameters depend on the sample material. A loop which cycles reasonably smoothly is the best starting point for a perfect crossfade loop, but a crossfaded loop does not always sound better. Just experiment a little with the parameters, and you’ll soon find out how, when and where they work best.

Group Parameters



Volume/Pan/Bal

Volume—Adjusts the volume of the Group, and therefore the volume of all assigned Zones, simultaneously.

Pan/Bal—Adjusts the pan position of the Group (stereo balance for stereo samples), and the pan position of all assigned Zones simultaneously.

Voices/Velocity Range

Voices—Determines how many voices the Group is allowed to use. This parameter is discussed in the *Multiple Zones and Groups* section, from page 17 onwards.

Velocity Range—The two *Velocity Range* parameters allow you to set up a velocity range for the Group. The settings made here override the settings in the Zones, if necessary: When a Zone’s velocity range is larger than that allowed by the Group setting, the Zone’s velocity range is limited by the Group setting.

ADSR Offsets

The amp(litude) envelope settings from the plug-in window can be offset separately for each Group by these parameters.

Each time parameter has an offset range of ±9999 ms, the sustain level can be varied by ±50%.

Filter Offsets

The *Cutoff* setting of the plug-in window can be offset separately for each Group (±50%).

The *Resonance* setting of the plug-in window can be offset separately for each Group (±50%).

9 EXS24 Key Commands

A number of Key Commands are available for the EXS24 which accelerate editing in Logic Audio, and provide additional functionality. They are found in the Key Commands window.

These Key Commands have no default keyboard assignments, so you will need to create them, should you wish to take advantage of these shortcuts and facilities. Please consult your Logic Audio reference manual for information on accessing the Key Commands window and on the assignment of keyboard shortcuts to functions.

Previous Instrument
Selects the previous Instrument (when multiple Instruments are opened for editing) allowing you to quickly switch between several Instruments. An Instrument must be selected for this function to work.

Next Instrument
Selects the following Instrument (when multiple Instruments are opened for editing) allowing you to quickly switch between several Instruments.

Select zones pointing to selected group(s)
Following the selection of a Group, use of this function will select all associated Zones, ensuring that any edits made are only performed on Zones associated with the selected Group.

Previous Zone/Group
Selects the previous Zone/Group (when multiple Groups/ Zones exist) allowing you to quickly switch between them. A Zone or Group must be selected for this function to work.

Next Zone/Group
Selects the following Zone/Group (when multiple Groups/ Zones exist) allowing you to quickly switch between them. A Zone or Group must be selected for this function to work.

New Zone
Creates a new Zone.

New Group
Creates a new Group.

View: All/Toggle Mode
Toggles between viewing all parameters in Zones and Groups and a limited view which displays the Audio File name in Zones and the Volume/Pan parameters in Groups.

View: Next Zone Parameter
This key command is designed to aid in the adjustment of the same parameters in each Zone. It limits the Zone display(s) to individual parameters (or rows of parameters) and steps through them from top to bottom. This allows you to adjust the appropriate parameters of all Zones in a Group more easily as working on a reduced set of parameters is simpler. It should be noted that if all Zone parameters are visible before the Next Zone Parameter Key Command is invoked, use of the function will switch to a reduced view mode.

View: Next Group Parameter
Operation and functionality is as per the Next Zone Parameter, for Group Parameters.

Move Audiofiles
Moves the audio files of the selected Instrument into a desired folder location. Use of this option will launch a standard operating system file navigation/browse utility. You may browse to an existing folder or enter a new name, as desired. If no folder name is entered, a new folder will be created which matches the Instrument name, and all audiofiles will be moved into this folder.

Copy Audiofiles
Operation is as per the *Move Audiofiles* function, but files are duplicated, rather than moved, to the nominated folder. This facility should be used as part of your working methods when creating Logic songs, as discussed in the *Saving of Song-related EXS24 Instruments* section, from page 27 onwards.

Move audiofiles of all instruments…
Moves the audio files of all Sampler Instruments in the Sampler Instruments folder to the target directory of your choice. In the target location, folders for the audio files associated with these Sampler Instruments are created.

Backup/Copy audiofiles of all instruments…
Copies the audio files of all Sampler Instruments in the Sampler Instruments folder to the target directory of your choice. In the target location, folders for the audio files associated with these Sampler Instruments are created. In addition, the Sampler Instrument files themselves are also copied.

Backup audiofiles of all USED and ACTIVE instruments of current song…
Copies the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location. All used Sampler Instrument files are also copied.

Move audiofiles of all USED and ACTIVE instruments of current song…
Moves the audio files of all (active) Sampler Instruments used by the current song to the target directory of your choice. Folders for the audio files associated with these Sampler Instruments are created in the target location.

10 Basics

A Brief History of Sampling

The idea of an instrument that could change its sound at any time, and that could imitate any other instrument, dates back centuries. By the 15th century, organ builders had managed to simulate violins, flutes, trumpets, and even human-like sounds with their instruments. Some years later, organs were perfected that could imitate birdsong.

Following the inception of film sound, several instruments were built that used film for the storage and playback of sound. Motion picture sound was based on the concept of recording sound onto the film itself as a separate track. Changes in brightness were read via an opto-electrical mechanism, and sound was replayed. This meant that sound was transferred to light and graphics in the widest sense. Creative musicians of the time began to scratch these films manually, to draw waves on them, and to film gearwheels and other things in order to produce interesting sounds from these images.

The immediate next of kin to today's samplers, however, was the Mellotron. This was a very bulky keyboard instrument that used a separate tape recording of an acoustic instrument for each and every key. Pressing a key started the playback of the corresponding tape; after releasing the key, the tape was drawn back by a spring. Due to the very complicated electro-mechanical mechanism used by the Mellotron, it was a very heavy and frequently unreliable keyboard instrument.

Compared to this, the first digital samplers at the beginning of the eighties seemed ultra-modern, but from today's point of view they did not offer much for their 5 or 6 digit price tag: a few seconds of sampling time, and sound quality that is surpassed by today's speaking toys. Nevertheless, early samplers like the Fairlight CMI and E-mu's Emulator are considered legendary. They had a great impact on music and on the development of electronic musical instruments in the following years.

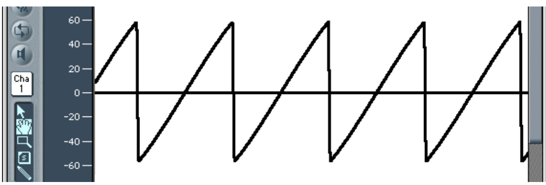
Nowadays, hardware samplers all sound good and are comparatively affordable. However, this is not the end of development for samplers. With computers getting faster and faster, it is now possible to build a fully-fledged sampler entirely in software, making hardware samplers unnecessary. Your EXS24 is proof of this…

Subtractive Synthesis

Subtractive synthesis is synthesis using filters. All analog and virtual analog synthesizers use subtractive synthesis to generate sound. Samplers and sample players also employ subtractive synthesis techniques, but use digital recordings (Samples) rather than oscillators, to supply the raw "waveforms". The signal of an oscillator or a sample contains a varying number of harmonics. The fundamental (or root) frequency of the signal primarily determines the perceived pitch, and the amplitude (level) determines the perceived volume.

Cutoff and Resonance Explained

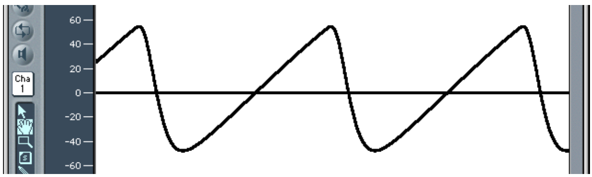
This picture shows an overview of a sawtooth wave (a = 220 Hz); The filter is open, with *Cutoff* set to its maximum, and with no resonance applied.



The screenshot shows the output signal of Emagic's ES1, displayed in Logic's Sample Editor at a high zoom setting.

When Michelangelo was asked how he would manage to cut a lion out of a block of stone, he answered, "I just cut away everything that doesn't look like a lion". This, in essence, is how subtractive synthesis works: Just filter (cut away) those components of sound which should not sound i. e., you subtract parts of the signal's spectrum. After being filtered, a brilliant sounding sawtooth wave becomes a smooth, warm sound without sharp treble.

The picture below shows a sawtooth wave with the filter half-closed (24 dB/Fat). The effect of the filter is somewhat like a graphic equalizer with a fader set to a given cutoff frequency (the highest frequency being fed through) pulled all the way down (full rejection), so that the highs are damped. With this setting, the edges of the sawtooth wave are rounded, making it resemble a sine wave.

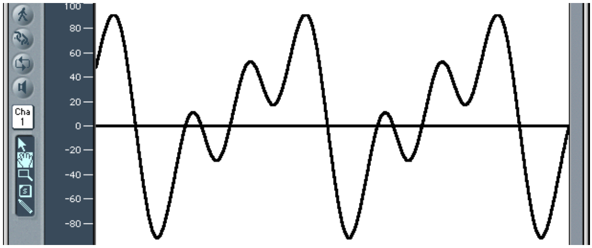


The waveform length shown here is not really longer than the image shown earlier, but the zoom setting is higher.

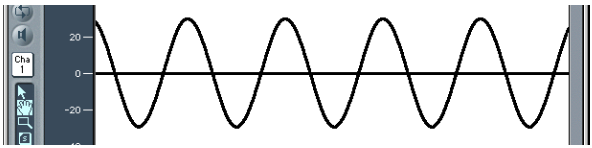
Fourier Theorem and Harmonics

"Every periodic wave can be seen as the sum of sine waves with certain wave lengths and amplitudes, the wave lengths of which have harmonic relations (ratios of small numbers)". This is known as the Fourier theorem. Roughly translated into more musical terms, this means that any tone with a certain pitch can be regarded as a mix of sine partial tones. This is comprised of the basic fundamental tone and its harmonics (overtones). The basic oscillation (the first partial tone) is an "A" at 220 Hz. The second partial has double the frequency (440 Hz), the third one oscillates three times as fast (660 Hz), the next ones 4 and 5 times as fast, and so on.

You can emphasize the partials around the cutoff frequency using high values for *resonance*. The picture below shows a sawtooth wave with a high resonance setting, and the cutoff frequency set to the frequency of the third partial (660 Hz). This tone sounds a duodecima (an octave and a fifth) higher than the basic tone. It's apparent that exactly three cycles of the strongly emphasized overtone fit into one cycle of the basic wave:

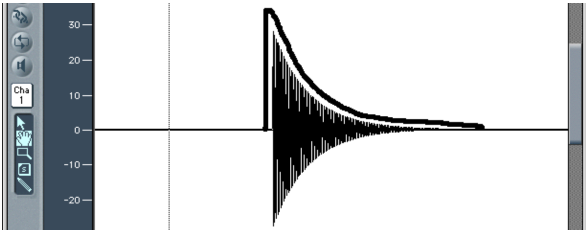


The effect of the resonating filter is comparable to a graphic equalizer with all faders higher than 660 Hz pulled all the way down, but with only 660 Hz (*cutoff frequency*) pushed to its maximum position (*resonance*). The faders for frequencies below 660 Hz remain in the middle (0 dB). A maximum resonance setting results in the self-oscillation of the filter. It will then generate a sine wave.

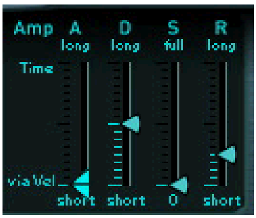


Envelopes

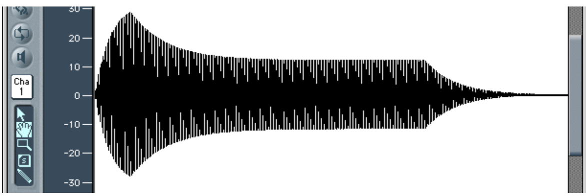
What does the term "envelope" mean in this context? In the image, you can see the waveform of a percussive tone. It's easy to see how the level rises immediately to the top of its range, and how it decays. If you draw a line surrounding the upper half of the waveform, you can call it "the envelope" of the sound—a graphic displaying the level as a function of time. It's the job of the ADSR to set the shape of the envelope. That's why the ADSR is called an "envelope generator".



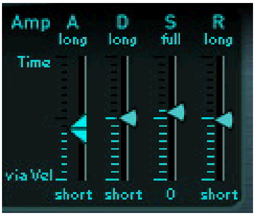
The screenshot shows a recording of an EXS 24 sound created with this setting of the ADSR parameters (*attack*, *decay*, *sustain* and *release*).



When you strike a key, the envelope travels from zero to maximum level in the *attack time*, falls from this maximum level to the *sustain level* in the *decay time*, and maintains the *sustain level* as long as you hold the key. When the key is released, the envelope falls from its *sustain level* to zero in the *release time*. The brass or string-like envelope of the following sound—the envelope itself is not shown in this graphic—has longer *attack* and *release times* and a higher *sustain level*.



The ADSR has been set this way:



A second ADSR can control the rise and fall of the cutoff frequency. The intensity of this frequency modulation is controlled by velocity. The sensitivity of velocity modulation is set in the *ADSR via Vel* controls. Modulation can be thought of as a remote control for a given parameter. There are multiple *sources* that can serve as a modulation source: e. g.—the pitch (note number), the velocity sensitivity, or the modulation wheel.

11 MIDI Controller List

Common	Mono Mode	71
	Voices	72
	Start Fixed	73
	Start via Vel	74
	Time via Key	11
	Attack Curve	112
Pitch	• Pitch Bend (up)	9
	• Pitch Bend (down)	70
	• Transpose	5
	Coarse Tune	76
	• Fine Tune	77
	Glide	78
	Pitcher	79
	Pitcher via Vel	80
	Modulation LFO	81
	Mod. Depth Fixed	82
	Mod. Depth Wheel	83
Filter	Filter (on/off)	84
	Filter Type	85
	Filter LFO	86
	Filter LFO Fixed	87
	Filter LFO Wheel	88
	Filter Cutoff	89
	Filter Resonance	90
	Filter Drive	91
	Filter via Key	92
	• Filter via Vel	93
	Filter ADSR Fixed	94
	Filter ADSR via Vel	95

	Filter Attack	106
	Filter Att. via Vel	107
	Filter Decay	108
	Filter Sustain	109
	Filter Release	110
Volume	• Output Volume	67
	• Key Scale +/-	68
	Level Fixed	96
	Level via Vel	97
	Tremolo/Pan LFO	98
	Pan Modulation	99
	Tremolo	100
	Amp Attack	113
	Amp Att. via Vel	114
	Amp Decay	115
	Amp Sustain	116
	Amp Release	117
LFO's	LFO 1 Dec./Delay	101
	LFO 1 Rate	102
	LFO 1 Waveform	103
	LFO 2 Waveform	104
	LFO 2 Rate	105