NATIVE INSTRUMENTS
SOFTWARE SYNTHESIS

FM7

Operation Manual

FM7
English

Operation Manual written by Craig Anderton

Special thanks to Yamaha for their support and for creating the milestone DX synthesizers.

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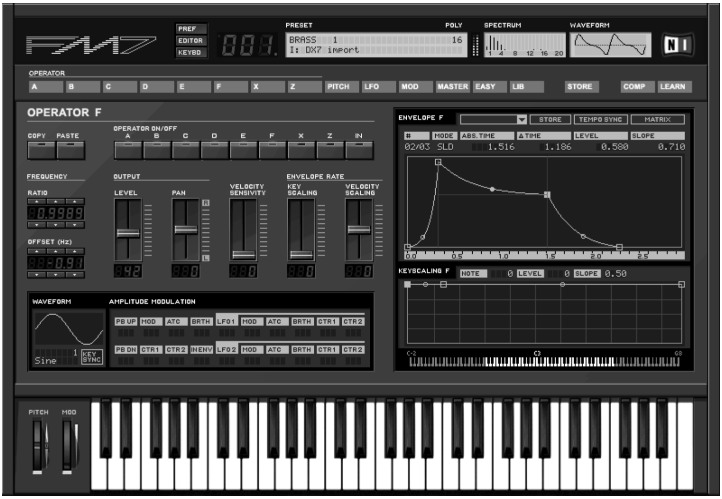
Introduction



Congratulations on your purchase of the Native Instruments FM7 software synthesizer. You'll be glad you did.

The FM7 is based on FM synthesis, which made its mass market debut in 1983 with the Yamaha DX7. Boasting aftertouch, velocity sensitivity, a new type of synthesis that was very different from analog subtractive synthesizers, a new protocol called "MIDI," and a shockingly low list price, the DX7 was an instant hit and went on to become the best-selling synthesizer of its time. It spawned several offspring, including the famous TX81Z (still one of the best synths for MIDI guitar) and the TX802, probably the finest hardware implementation of FM synthesis that Yamaha ever produced. The latest in the line is the DX200 Desktop Control Synthesizer which is also recommended as a MIDI controller hardware for FM7.

Now FM synthesis is back, and better than ever. Designed as a convenient plug-in, the FM7 has 32-bit resolution for superb sound quality, an innovative algorithm programming matrix whose flexibility is light years ahead of older FM synthesizers, extensive modulation, effects, a filter module/operator, and multiple waveforms.

Although FM synths had a reputation of being difficult to program - and they were! - the FM7 offers a unique “easy” page that makes it simple to customize sounds to perfection, without having to learn programming.

All the great features of FM synthesis are included, such as an “analog” control to add slight amounts of randomness, and a microtuning page for alternate tunings. But it also has modern essentials, such as envelope and delay sync to MIDI clock, 64-voice polyphony, variable resolution for lo-fi sounds, total MIDI control, and much more to bring FM synthesis into the 21st century.

The FM7 integrates perfectly into the computer-based virtual studio, whether Macintosh or Windows. It can serve as a stand-alone module, turning your computer into a synthesizer. It also works as a sequencer plug-in with Steinberg’s popular VST 2.0™ and ASIO™ interfaces, as well as Cakewalk’s DXi standard, Digidesign’s DirectConnect, and Mark of the Unicorn’s MAS.

If you were around for the first wave of FM synthesis, you’ll be pleasantly surprised at just how good it can sound with up-to-date technology. If you’re new to FM synthesis, you’re in for a great time. You can coax sounds out of the FM7 that range from angelically clean to hellishly nasty...from sweet to sour...and from traditional to innovative.

Have fun! I sure am.

--Craig Anderton

Craig Anderton is an internationally-recognized authority in the field of musical electronics, as well as a professional musician whose music can be heard on over 15 major label releases. Somehow, he convinced us to get him a beta copy of the FM7, and somehow, we convinced him to help us with the manual.

Audio Interfaces

Audio interfaces allow Native Instruments software to communicate with the audio hardware of your computer and other programs that you may have installed. This chapter contains detailed information on the various audio interfaces and how to use them. The features of the various interfaces are described together with their suitable applications.

Basically, there are two ways of using NI software: as a “stand-alone” or as a “PlugIn”. In following, the two versions are described together with their corresponding interfaces.

Stand-alone Application

This method allows you to use NI software as stand-alone programs with any of the following interfaces (drivers): ASIO, MME, DirectSound, and Core Audio. In this case, your computer acts as a stand-alone instrument, similar to a hardware synthesizer with a MIDI port and analog inputs and outputs. The table contains an overview of which interfaces are suitable for stand-alone operation on the various computer platforms:

Interface/driver	Windows	MacOS X
ASIO 2.0	●	
DirectSound	●	
MME	●	
Core Audio		●

PlugIn

Used as PlugIns, NI software are not a stand-alone programs but rather program “modules” that can be integrated into a “host” program such as a sequencer. PlugIn mode allows you to integrate it seamlessly with the sequencer. Furthermore, it has many other uses as a PlugIn:

- MIDI sequencing and audio mix-down of the MIDI tracks within a single program
- Comfortable automation of parameters in the sequencer
- Further processing of signals using additional PlugIns
- Sample-accurate timing with MIDI controllers (when used as VST 2.0 PlugIn)
- Restoring of all PlugIn settings when the host document (such as a song file of the sequencer) is loaded
- Integration with other instruments into a "virtual studio"

This table provides you with an overview of which interfaces are supported by which host programs:

Interface/driver	Host Programs	Windows	Mac
VST 2.0 PlugIn	**Cubase, Nuendo, Logic 5.x**	●	●
Cakewalk DXi	**Sonar**	●	
Audio Units	**Logic 6.x, 5.x**		●
RTAS	**Pro Tools 6.x, LE, Free**	●	●

Overview of Operating Systems and PlugIns

The interfaces described below are effectively different ways in which NI software can communicate with your sound card. The interfaces that are available on your computer depend on the sound card you are using as well as your computer platform (Windows or MacOS).

ASIO ("Audio Streaming Input Output") is a sound card driver architecture developed by Steinberg. ASIO is available for MacOS and Windows computers. It offers low latency and supports multi-channel audio cards. With its high performance and low latency, the ASIO driver interface is highly recommendable.

DirectSound is an interface developed by Microsoft and is a component of DirectX 5.0 or higher for Windows 98/ME/2000/XP. Whether or not DirectX works well depends on the sound card you are using. If the audio buffer size that you set is too small with DirectSound, glitches and clicks may occur in the audio.

MME is the standard “Wave” driver in Windows. Most sound cards support this interface and work with it quite well. However, MME is even less suitable than DirectSound for real-time applications. This is noticeable by a comparatively high latency.

Core Audio is a new audio interface available with MacOS X that allows you to use external audio hardware as well as the integrated audio output of the Mac.

RTAS is based on an interface protocol from DigiDesign that allows you to use PlugIns with ProTools (or other software that is compatible with DigiDesign). RTAS PlugIns function independently from additional TDM hardware and are nonetheless able to offer the widest range of features. In this case, the host processor alone performs all of the computations for the PlugIn.

Audio Units is the OS X PlugIn format devloped by Apple. They may be used in similar fashion to VST PlugIns.

DXI 2 is a PlugIn interface for software synthesizers and instruments based on Microsoft DXi technology. Sonar from Cakewalk and Fruity Loops are the most well known host sequencers that support DXi.

VSTi is the PlugIn format developed by Steinberg. It is cross platform and can be used in a variety of hosts.

NI software as PlugIn

VST 2.0 PlugIn

In addition to the stand-alone version, NI software can also be used as a VST PlugIn. The advantages of the VST 2.0 format allow us to provide you with a powerful PlugIn.

For more information on the VST 2.0 format, refer to the user guide provided with your VST host program.

Using NI software in Cubase SX 2

- Launch Cubase, go to the **Devices** menu option and select the **VST Instruments** menu option.
- A window showing the instrument rack appears. Click on an empty slot and choose FM7 from the available list of instrument PlugIns.

- The PlugIn will now appear in your list and automatically be turned on. It will also create a set of audio channels in your VST mixer that will be used for mixdown within your project. This will allow you to mix, pan, and process FM7 's output just like any other existing audio track in your Cubase song.
- Click on the **Edit** button to call up the FM7 interface. Here you can control and edit all the features and functions that FM7 has to offer.
- Now go to the "Project" page and add a MIDI track (if you do not have one already created).

- Go to the **Output** parameter section for this MIDI Track and click on the field. This will create a list of available MIDI out ports to assign to this MIDI track. Choose **FM7 VST** from the list.

Note: If FM7 does not appear in the list of available VST instruments inside your VST 2 host application, then it is not installed correctly.

After having loaded an Instrument from the library you should be able to trigger it via MIDI using a keyboard controller. FM7's sound will generate through the VST mixer and directly to your sound card. If the PlugIn does not receive MIDI or generate audio, then make sure to check the following areas:

- Make sure "MIDI thru" is enabled in Cubase.
- The MIDI channel of your MIDI track must correspond to the receive channel of the loaded instrument.
- Make sure that you have properly configured your sound card for use with Cubase.

(please refer to your Cubase manual for more information)

Using NI software in Nuendo 2.0

- Launch an empty or current project in Nuendo.
- Click on the **Devices** menu and choose **VST instruments** from the menu options (or press F11 on your keyboard).
- A window showing the instrument rack appears. Click on an empty slot and choose **FM7 VST** from the available list of installed PlugIns.

- The PlugIn will now appear in your list and automatically be turned on. It will also create a set of audio channels in your VST mixer that will be used for mixdown within your project. This will allow you to mix, pan, and process FM7's output just like any other existing audio track in your Nuendo project.
- Click on the **Edit** button to call up the FM7 interface. Here you can control and edit all the features and functions that FM7 has to offer.
- Now go to the "Project Editor" page and create a MIDI track (if you do not have one already created).

- Go to the **Output** parameter section for this MIDI Track and click on the field. This will create a list of available MIDI out ports to assign to this MIDI track. Choose **FM7 VST** from the list. Also make sure you assign the MIDI input port to correspond to whatever MIDI controller your are using.

- Record enable the MIDI track.

Note: If the NI software does not appear in the list of available VST instruments inside your VST 2 host application, then it is not installed correctly.

After having loaded an Instrument from the library you should be able to trigger it via MIDI using a keyboard controller. FM7's sound will generate through the VST mixer and directly to your sound card. If the PlugIn does not receive MIDI or generate audio, then make sure to check the following two areas:

- Make sure "MIDI thru" is enabled in Nuendo.
- The MIDI channel of your MIDI track must correspond to the receive channel of the loaded instrument.
- Make sure that you have properly configured your sound card for use with Nuendo

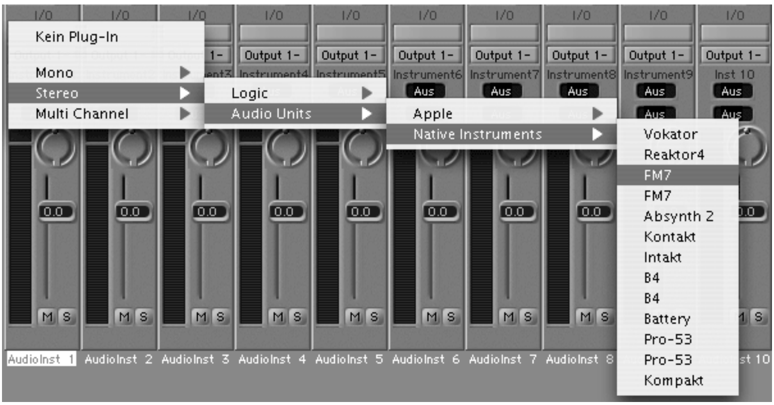
(please refer to your Nuendo manual for more information).

Audio Units PlugIn

Using NI software in Logic 6.x and 5.x (please note the setup is the same for Audio Units, Mac and VST, Windows)

- Launch Logic and create an audio instrument track or set an existing audio or MIDI track to an audio instrument track by clicking on it, holding down the mouse button and choose **Audio ⇨ Audio Instrument ⇨ AudioInst 1**.

- Double click the audio instrument track to open the environment window. Logic scrolls automatically to the first instrument bus in the Logic mixer.
- Choose the FM7 Audio Unit/VST PlugIn in the appropriate insert slot of the instrument mixer bus, either in the arrange or mixer window. Then click onto the insert slot, hold down the mouse button and choose **Stereo ⇨ Audio Units/VST ⇨ FM7**. (FM7 is also available as a multi-channel insert)



- The PlugIn now appears in the instrument slot and is ready to use. The instrument mixer channel will allow you to mix, pan, and process FM7’s output just like any other existing audio track in Logic.

- If the FM7 interface is not already open, double click on the mixer's FM7 slot to call up the FM7 interface. Here you can control and edit all the features and functions that FM7 has to offer.

Note: If FM7 does not appear in the list of available VST instruments inside your VST 2 host application, then it is not installed correctly.

After having loaded an Instrument from the library you should be able to trigger it via MIDI using a keyboard controller. FM7's sound will generate through the VST mixer and directly to your sound card. If the PlugIn does not receive MIDI or generate audio, then make sure to check the following two areas:

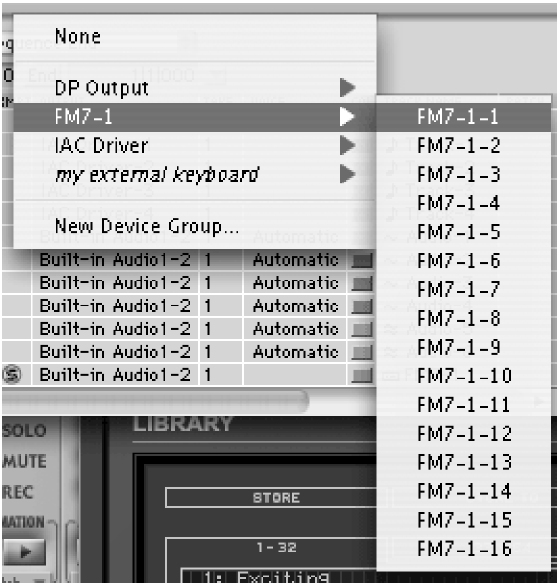
- Make sure "MIDI thru" is enabled in Logic.
- The MIDI channel of your MIDI track must correspond to the receive channel of the loaded instrument.
- Make sure that you have properly configured your sound card for use with Logic.

(please refer to your Logic manual for more information).

Using NI software in Digital Performer 4.1 (or higher)

- Launch Digital Performer and create an instrument track by selecting **Project ⇨ Add Track ⇨ Instrument Track⇨ FM7.**

- Create a MIDI track by selecting **Project** ⇨ **Add Track** ⇨ **Midi Track.** In Digital Performer's track overview window (or in the sequence editor window) assign the ouput of this MIDI track to "FM7-1" and a MIDI channel. If you instantiate further FM7 PlugIns they will be named "FM7-2", "FM7-3" etc.



- The PlugIn is now ready to use. The mixer of Digital Performer will allow you to mix, pan, and process FM7's output just like any other existing audio track.
- To play FM7 with your keyboard, record enable the MIDI track which you have routed to FM7 and make sure **Midi Patch Through** is enabled in the Studio menu of Digital Performer.
- Double click on the FM7 slot in Digital Performers mixing board to call up the FM7 interface. Here you can control and edit all the features and functions that FM7 has to offer.

Note: If FM7 does not appear in the list of available Audio Unit PlugIns inside your Audio Units host application, then it is not installed correctly.

After having loaded an Instrument from the library you should be able to trigger it via MIDI using a keyboard controller. FM7's sound will generate through Digital Performers mixer and directly to your sound card. If the PlugIn does not receive MIDI or generate audio, then make sure to check the following two areas:

- Make sure **Midi Patch Through** is enabled in the Studio menu of Digital Performer.
- The MIDI channel of your MIDI track must correspond to the receive channel of the loaded instrument.
- Make sure that the instruments track output is correctly set.
- Make sure that you have properly configured your sound card for use with Digital Performer.

(please refer to your Digital Performer manual for more information).

DXi 2 PlugIn

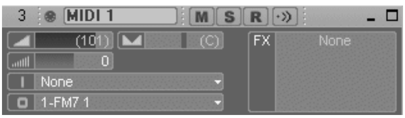
DXi 2 is a PlugIn interface for software synthesizers and instruments based on Microsoft's DirectX technology.

Using NI software in Sonar

- Launch Sonar
- In the synth rack choose **FM7 DXi 2**.

Loading the FM7 DXi 2 PlugIn in the synth rack

- Route a MIDI track to the DXi 2-PlugIn by selecting **FM7** in the Out drop down list.



Assign a MIDI track to the FM7 2-DXi-PlugIn

After having loaded an Instrument from the library you should be able to trigger it via MIDI using a keyboard controller. FM7's sound will generate through the Sonar mixer and directly to your sound card. If the PlugIn does not receive MIDI or generate audio, then make sure to check the following two areas:

- Make sure "MIDI thru" is enabled in Sonar.
- The MIDI channel of your MIDI track must correspond to the receive channel of the loaded instrument.
- Make sure that you have properly configured your sound card for use with Sonar.

(please refer to your Sonar manual for more information).

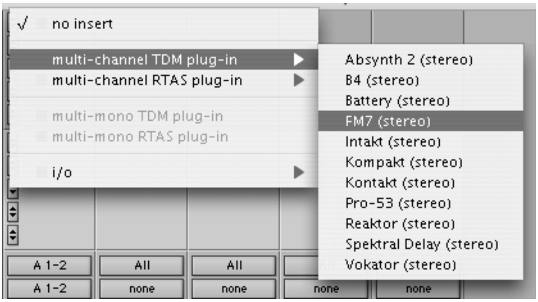
RTAS PlugIn / HTDM PlugIn

Using NI software with Pro Tools 6.x under Mac and Windows

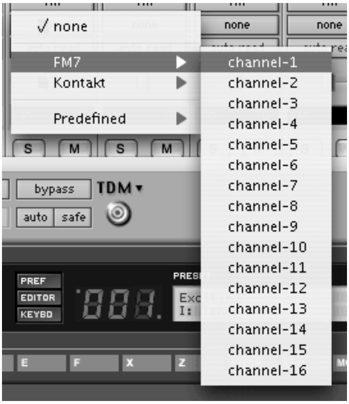
The RTAS format is an interface protocol for Mac OS and Windows that allows you to use PlugIns with ProTools independently from additional TDM hardware, while nonetheless offering the widest range of features. In this case, the host processor alone performs all of the computations for the PlugIn.

- Launch Pro Tools
- Create a new **AUX** track **File** ⇨ **New Track**
- Create a new **MIDI** track the same way
- Locate the channel mixer **Windows** ⇨ **Show mix**
- The dark grey box at the topmost section of the AUX channel is the RTAS insert section. Click on the first empty slot to show all available RTAS PlugIns.

- Choose **FM7** from the menu



- Now locate the **MIDI** channel you just created
- In the output slot, choose FM7 and the appropriate channel



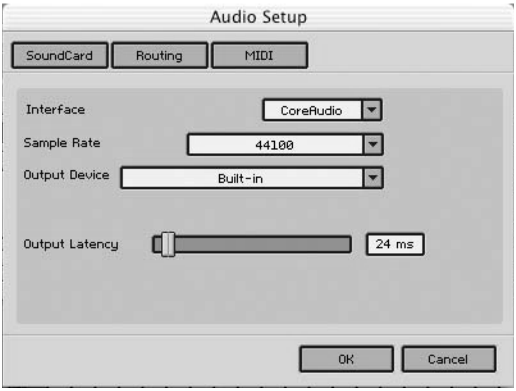
After record enabling the midi track, you will be able to play FM7 with your midi keyboard.

(Please refer to your Pro Tools manual for more information on how to record FM7's output).

Note that on Pro Tools TDM systems stereo RTAS PlugIns can only be instantiated on stereo audio tracks.

Mac Standalone

The stand-alone version allows you to use the application independently from other programs. In order to use the Standalone version you have to do the audio and MIDI settings first. You can call up the **Audio + MIDI Settings** setup dialog from the File menu. For setting the standalone interfaces please choose **Setup...** from the **File** -menu.



Audio + MIDI Setting dialog

Soundcard tab

Interface

All of the supported (and installed) audio interfaces are available in this drop-down list. Select the desired audio driver (MME, DirectSound, ASIO, SoundManager, Core Audio) from this list.

Sample Rate

Depending on the sound card and driver you are using, various sample rates are available. Set the desired sample rate here.

Output Device

Here you can define which of the installed audio interfaces should be used for the audio outputs based on the driver selected under **Interface**.

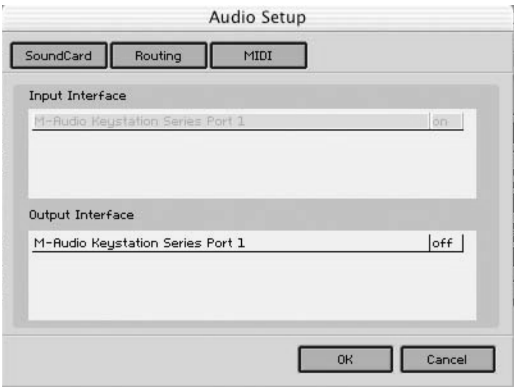
Output Latency

This box displays the output latency. With some drivers you also get a latency slider for setting an individual latency.

Routing tab

If you are using a multi-channel sound card, FM7 also allows you to freely select which channels to use for the output signals.

MIDI tab



These two boxes (MIDI inputs and MIDI outputs) display all of the MIDI inputs and outputs that are correctly installed on your system. Click in the right column to “off” or “on” to activate or deactivate the corresponding MIDI input or output. From this point on, FM7 sends and receives MIDI on the activated inputs and outputs.

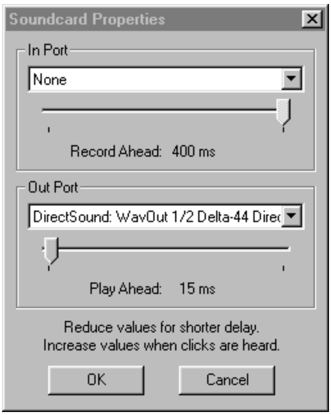
PC Standalone

Soundcard Settings

The FM7 software needs a soundcard for playing the sounds it produces. No extra drivers are needed when using a standard sound card as the software's audio interface. The FM7 software makes use of the standard MME, DirectX or ASIO drivers installed with the cards.

DirectSound and MME

Some adjustment of the FM7 software's parameters is necessary to tune it to the hardware and achieve optimum performance. Choose **Soundcard** in the **System** ⇨ **Audio Port** menu and open the appropriate settings dialog window by selecting **System** ⇨ **Audio Settings....**



Audio Settings dialog window for standard Windows soundcards

If you have an ASIO driver for your audio card you should try these out since ASIO drivers mostly allow lower latency settings than MME or DirectSound drivers. Many ASIO drivers can have latencies of 10 ms and less.

Devices

If more than one soundcard (or driver port) is installed, the selection box **Out Port** chooses which soundcard (or which driver port) will be used by the FM7 software.

In the list of available ports for **Out Port,** the available devices are marked **MME:** or **DirectSound:**. The latest DirectSound drivers for your soundcard are probably better optimized with regard to latency (delay) in the audio output than the earlier MME (WaveOut) drivers and normally perform better. We recommend that you try out all the available drivers, and use the one that gives the most responsive, delay-free performance.

Do not use emulated DirectSound drivers (which are usually marked “emulated”) because these are actually MME drivers made to look like DirectSound and will perform poorly.

A precondition for using DirectSound is that the Windows extension DirectX 5.0 (or later) is installed on your PC.

Note: The **low latency** technology employed in the FM7 software makes high demands on **soundcard drivers**. Please make sure you have the very **latest drivers** available for your soundcard.

Latency

The delay (latency) that occurs during audio output depends on the size of the audio buffer the software passes to the soundcard. For smooth operation this buffer must have a minimum length which depends mainly on the type of soundcard and driver used.

If you are installing the FM7 software for the first time, you can skip the settings described below and first get to know the system. Return here later to optimize the latency for the best possible performance.

With the Out Port’s **Play ahead** slider you can adjust the audio buffer size. The smaller the buffer, the faster the FM7 response. However, if the buffer is set too small, clicks will appear in the audio output (MME) or extreme latency will result (DirectSound).

Important: To get good **performance** from the FM7 software you must optimize the **Play ahead** by hand every time you change your soundcard or update the soundcard driver.

To optimize the buffer length for your system, select any preset and play it while at the same time moving the slider for **Play ahead** in the **Soundcard Properties** dialog window. Move the slider to the left to reduce **Play ahead** until you start to hear clicks in the sound output. Now move it back to the right a bit to find the point where the clicks disappear. This is the optimum output buffer size for your card.

When using MME drivers, the sound will break up when **Play ahead** is too small. With DirectSound drivers on the other hand, there will only be one glitch after which the effective latency suddenly becomes very large (about 1 second).

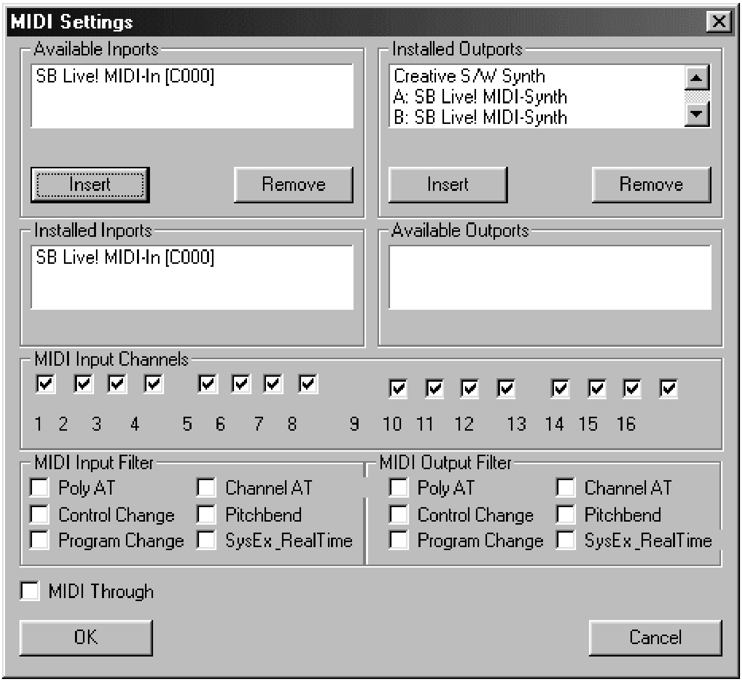
By the way, the number of voices used inside the software doesn't have any affect on latency. It does, of course, affect overall performance because it increases the load on the CPU.

Please note that the soundcard output levels depend on the card's mixer settings. You can control this device using the Windows accessory **Volume Control**, the **Multimedia Properties** of the **Control Panel**, or a mixer program delivered with the soundcard.

MIDI Interfaces

The MIDI Ports through which the FM7 software communicates with the rest of the world are selected in the **MIDI Port** dialog window, which is opened via the menu entry **System** ⇨ **MIDI Settings...**. All the existing MIDI ports in your PC that are installed under Windows appear here, and can be chosen for use with the FM7.

If a MIDI input port has already been opened by another program, the software will not be able to use it and it will not appear under **Available Inports**. In this case free up the port in the other program or make sure that the FM7 software starts up first. Conversely, an inport has to be removed from the list of **Installed Inports** before another program has access to it.



MIDI Port dialog window

MIDI Input Channels

You can select MIDI channels for FM7. The program receives MIDI only on channels you have activated here. MIDI data received on other channels will be ignored.

MIDI Input and Output Filters

You can set MIDI filters for the MIDI input and output seperately. If you want the FM7 to ignore specific MIDI controllers, select the appropriate checkboxes.

MIDI Thru

Activate this checkbox if you want FM7 to pass incoming MIDI data directly to the MIDI output. MIDI thru is useful if you want to set your MIDI interface as MIDI input for FM7, but also record the MIDI data in another program. If you control FM7 from a sequencer, do not activate MIDI thru for FM7, since in certain configurations you might get a MIDI loop.

Interface Conventions

Mac and Windows Conventions

Operating the FM7 on Mac or Windows machines is identical, except for slight differences caused by differences in their operating systems.

Key commands are given for Windows. For the Mac, when the text says "right-click," use Ctrl-click..

Sliders

To change a slider setting, click on it with the mouse. Drag the mouse up or down to change the value. For "fine tuning," press the Shift key prior to moving the slider, and keep holding the Shift key while moving the slider.

Numeric Values

For parameter tweaking, click on the numeric readout below the slider. Drag the mouse up or down to change the value. When there is both a slider and a numeric value, you get finer resolution by dragging the numeric value. Hold additionally the Shift-key to get an even finer resoluion.

For setting up the parameters **ratio** and **offset** drag on the 1s digit of the numeric readout, then the value will change in 1s. If you drag on the 10s digit, then the value will change in 10s.

If the numeric value has buttons above and below with small arrows, clicking on the upward arrow raises the value by one, and clicking on the downward arrow lowers the value by one. Clicking and holding on these buttons scrolls through the values at a moderate rate.

Buttons and Switches

Click once on a button or switch to enable, click again to de-select or disable. A selected switch will be either outlined in red, or change color.

Graphic Interfaces

Some parameters, such as envelopes, are shown graphically as curves and lines with nodes.

- To change an envelope's shape, click on a node with the mouse, and drag the slope point into the new position.
- To create a new node, right-click where you want the node to appear.
- To delete a node, right-click on the node.

Test Driving the Standalone FM7

Now that the FM7 software is installed, let's check it out.

- Select the FM7 from the Windows start menu or double-click on the FM7 icon on the desktop.
- After the FM7 screen has appeared click on the Lib button in the selector strip to go to the libary page. This page is where you manage (select, save, import, etc.) your FM7 presets and banks of presets.

When you call up the FM7, one of the presets in a group of 32 presets will be outlined. There are three ways to play this preset.

- The FM7 has a "virtual keyboard" at the bottom of the screen. Click on notes to trigger them. You can also click on the mod and bend wheels and move them. To "retract" or "extend" the keyboard, click on the KEYBD button next to the FM7 logo.
- Use a MIDI controller. Make sure your MIDI controller's output plugs into your computer interface's MIDI input, which should be assigned as the Inport under the *System* ⇨ *MIDI settings* menu. If you move your controller's mod and pitch bend wheels, the virtual wheels will move also.
- In standalone mode only, the computer keyboard can trigger notes. The letters are chosen to emulate a musical keyboard. For example, Z-M plays notes C-B, S plays note C#, D plays note D#, etc.

Page Select Area

In the Page Select area you can switch between the different FM7 panels. The first eight panels are for the eight FM7 operators. The next six buttons open the panels for the pitch, LFO and modulation parameters as well as the Master page, the Easy Edit page and the Preset Library.

If you press the same Page Select button several times, you can switch between two pages, the one which was opened before you pressed the latest page select button and the current page. This allows you to compare the parameter settings of two pages with each other.

The Store button overwrites the selected preset immidiately with the current edit buffer. With the Comp button you can compare two sound editing states with each other, the Learn button is for easily assigning MIDI controllers to a parameter (See "Preferences" on page 82.).

Auditioning Presets

Click on additional presets to hear more cool sounds. For more information on playing back and storing presets, see the Library chapter.

Common Parameters - All Pages

The upper portion of the FM7 remains constant as you switch among pages. Available parameters include:

FM7 Logo. A klick on the logo opens the FM7 About window where you find the FM7's program version number and other information.

Preferences, Editor, Keyboard switches. Click on **Preferences** to bring up the Preferences menu (See "Preferences" on page 82.), **Editor** to show/hide the editor, and **Keybd** to show/hide the keyboard and wheels.

Preset readout. Click and drag up or down to change presets. An LED toward the upper left of the readout lights in the presence of MIDI activity. An LED in the lower right of the readout lights if the preset has been edited.

Preset display. Shows the information stored with the preset. The lower row is a general purpose display. You can click onto the letter in front of the row to change the information being displayed:

- **I:** Preset info
- **D:** Preset save date
- **C:** Preset categories
- **P:** Selected parameter
- **A:** Preset author

In some FM7 modes (e.g. MIDI Learn) the display shows important additional information.

Poly readout. (inside the Preset display) Click and drag up or down to change the maximum number of voices (polyphony).

Output meter. Realtime stereo level meter for the overall amplitude of a sound.

Spectrum display. This non-editable display shows the harmonic spectrum of the selected sound.

Waveform display. This non-editable display shows the waveform of the selected sound.

NI Logo. A klick on the NI logo launches your web browser and opens the start site of the Native Instruments homepage if your computer is connected to the internet.

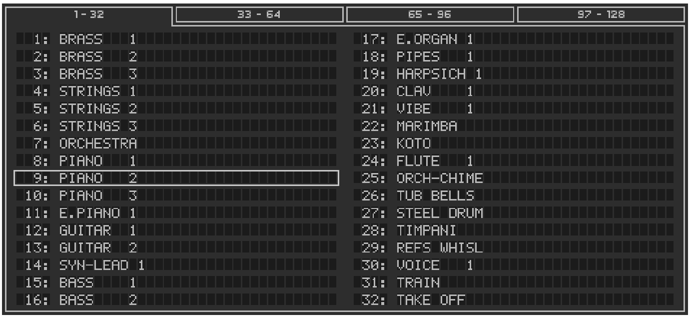
The Library

The library page is where you move, copy, store, save, import, and manage presets and banks. Access the Library page at any time by clicking the **LIB** button. When selected, the button turns red.

There are four main Library components.

Bank/Preset Listing

The Bank/Preset listing is toward the lower left.



- A bank is divided in four blocks (1-32, 33-64, 65-96, 97-128). Select the desired block by clicking on its tab.
- There are 32 Presets within each block. Select a preset by clicking on it. An outline appears around the name.

Master Control Strip

The master control strip contains the preset/bank management tools. Select a function by clicking on its associated box.

Store: Any edits you make to a preset are kept in an edit buffer. You must store any changes if you want the preset to retain them. If you don't store, the preset will revert to its previous settings once you switch to a different preset.

Store To: Click and this box flashes. Select the target Block and Preset number where you want the Preset stored.

Load: Loads an individual FM7 preset (*.f7p suffix), a block of 32 presets (*.f7b suffix) or a FM7 bank of 128 presets (*.f7a suffix) from storage media (hard drive, floppy, etc.). Navigate to the desired file, select, and click on Open.

Import Sysex: The FM7 can accept System Exclusive Data (preset parameters) from Yamaha's DX7, DX7II and DX200 synthesizers and convert the sound to its own parameter format.

In standalone mode FM7 automatically receives any MIDI SysEx data in the recognised formats (do not click on Import SysEx; this process is automatic). Single Presets go into the edit buffer (and need to be stored manually), while Banks go into the currently selected bank of 32 Presets.

Note that receiving SysEx data via MIDI is not possible when using FM7 as a plug-in because VST and the other plug-in standards do not currently support SysEx.

Another way (that always works) is to load the SysEx data from a file. Click on the Import SysEx button, then navigate to a compatible SysEx file (it will usually have a .syx suffix under Windows), and open it. You can find lots of DX7 SysEx data files on the internet. You can also capture MIDI SysEx data and save it to disk as a file using programs like Midi-Ox (PC freeware for private use, http://www.midiox.com), as well as sequencer programs and even some keyboards

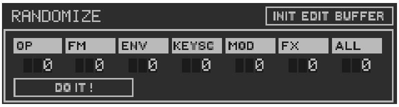
DX7 Operator 1 maps to FM7 operator F, operator 2 to operator E, operator 3 to operator D, etc.

Save Preset: Saves the selected preset as a *.f7p file to your storage media. Navigate to where you want to save the file, then click on Save.

Save 32: Saves the currently visible block of 32 presets as a *.f7b file. Navigate to where you want to save the file, then click on Save.

Save All: Saves all 128 Presets as a *.f7a file. Navigate to where you want to save the file, then click on Save.

Randomize



This function changes preset parameters randomly, according to the parameter “mask.” The higher a number entered for a parameter, the wider the possible range of random values. *Example:* To randomize just the Modulation parameters, set a non-zero value for MOD, then click on Do It!

The All parameter changes all values simultaneously to that of the All value.

Init Edit Buffer

Click on this to restore a preset to its most basic default connection - a single operator feeding the output, and panned to center. Remember, you need to store the edit buffer in a preset to retain the patch.

Details

This shows Preset information.

Name: If the Preset has been edited, this will start with EDIT, otherwise it will start with the Preset number. You can rename by selecting the name (drag or double-click) and typing in the new name.

Category: You can classify a Preset’s sound by up to three categories. Choose the best descriptions from the pull-down menus.

Author: Rename by selecting (drag or double-click) and typing in the new name. You can enter a default name in the FM7 preferences to avoid filling it out everytime.

Date: Shows the date on which the preset was saved, like the filestamping found in the Windows or Mac operating systems.

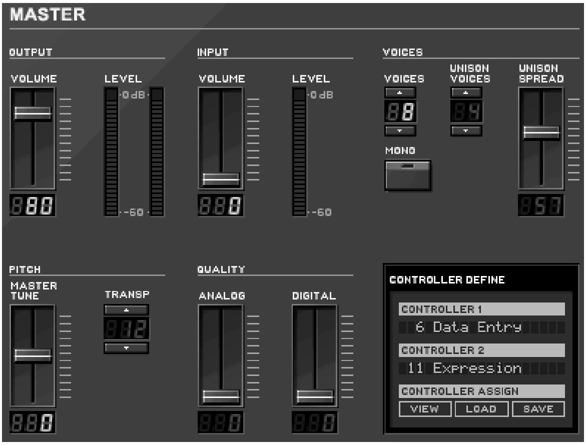
Info: Use this as a mini-note pad to type information about the Preset.

Master Page

This page contains parameters that affect the instrument as a whole. Access it by clicking the **Master** button. When the page is on top, the **Master** button will be red.

Here is what the various parameters do.

Master



Output

Volume changes the level of the entire instrument. Try to keep this as high as possible, consistent with not overloading the device (e.g., mixer or soundcard) being fed by the FM7 (as indicated by the associated meter hitting 0 dB). This parameter is not stored or recalled as part of a sound preset, which is why the digits are green.

Input

Volume This controls the level of signals being fed into the FM7 when using it as a processor. As with output, keep the control as high as possible without causing the associated level meter to hit 0 dB. This parameter is not stored or recalled as part of a sound preset, as indicated by the digits being green).

Poly

Poly sets the available polyphony. Only voices that are being played consume CPU power, so this parameter acts as a governor to set a limit on the number of voices. This parameter is mirrored by the Poly parameter in the main (top) panel display. This parameter is not stored or recalled as part of a sound preset (again, the digits are green).

Mono limits playback to one note at a time, like the old monophonic analog synthesizers. Like in the DX7 this also switches to single trigger envelopes (legato playing). However, you can still stack multiple voices on this one note using Unison (see next).

Unison Voices determines how many voices can be stacked on a single key in unison mode. If many keys are pressed and not enough voices are available, the FM7 assigns fewer voices to each key, so individual notes “thin out” rather than disappear.

Unison Detune detunes the unison voices for a fatter, chorusing-like effect. Higher values give increased detuning.

Pitch

Master Tune offsets the pitch from -99 to +100 cents for precise pitch matching to other instruments. This parameter is not stored or recalled as part of a sound preset, as indicated by green digits.

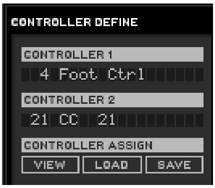
Tranposition can transpose pitch up or down two octaves, in semitone increments.

Quality

Analog This introduces random variations between voices, as used to happen with analog gear - for example, component values would change as temperatures drifted, and so on. Higher values increase the degree of randomness, and is very effective in Unison mode.

Digital changes the bit resolution, and therefore the sound quality. The original DX7 was a 12-bit device, whereas later models used 16 bits. This control lets you simulate, as well as go even “lower-fi” than, the “vintage” sound.

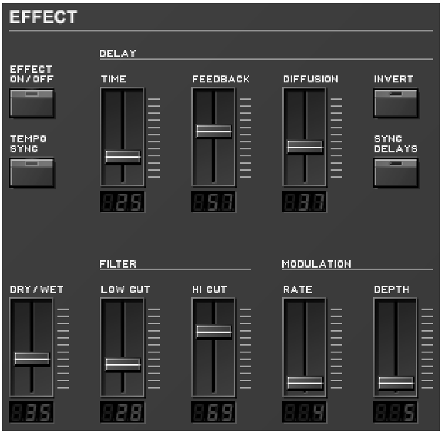
Controller Define



Controller 1 selects the MIDI controller number for Controller 1 in the Modulation Matrix. Click on the controller number and drag vertically to select. Controller 2 similarly selects the MIDI controller number for Controller 2.

FM7 can be edited or automated from remote devices; the three lower fields allow viewing, loading, and storing the controller assignments for all parameter automation. Normally these are set by loading a configuration for a particular fader/control box, or using the MIDI learn function to assign individual parameters to individual controllers.

Effect



This consists of a 4-tap stereo delay line with four independent modulation LFOs, and whose output can be filtered. Note that several presets for popular effects are available from the Easy page.

Effect On/Off enables or disables the Effect section.

Tempo Sync synchronizes the delay time to any incoming MIDI clock tempo. If Sync Delays is activated, the modulation LFO is also synchronized to the tempo.

Note: Not all VST host software supplies tempo information to plug-in instruments, so this feature may not work with your host software.

Dry/Wet determines the mix of straight and processed sounds. 0 gives dry sound only, 100 gives effect sound only. 50 is generally optimum for chorusing and flanging effects.

Delay

Time sets the time between repeats.

Feedback causes the delay output to feedback to the input, thus creating multiple echoes. The Delay Time control sets the spacing between echoes.

Diffusion spreads the 4 echo taps in time. Higher values give more diffusion. With Time = 55, Feedback = 70, Diffusion = 100, Dry/Wet = 85, Low Cut = 100, Hi Cut = 40 and Modulation Depth = 20, you'll hear a reverb-like effect.

Invert alters the phase of the repeats. Inversion gives a somewhat more diffused sound with echoes, and with flanging, changes the effect's tonality.

Sync Delays phase locks the modulation LFOs and diffused echoes.

Filter

Low Cut determines the delayed signal's low end frequency response. 0 gives full low frequency response. 100 cuts out the low frequencies.

Hi Cut edits the delayed signal's high end frequency response. 0 gives minimum high frequency response. 100 gives maximum high frequency response.

Modulation

This adds a periodic, cyclic shift to the delayed signal time. At low delay times, adding modulation by turning up the Depth control creates flanging effects. At slightly longer delay times, it's possible to obtain chorusing. At high values, you can get some really warped sounds by turning up the Depth.

Rate sets the period of the oscillator controlling the delay time. 0 = slowest rate, 100 = fastest rate.

Depth determines how much the modulation varies the delay. 0 = no modulation, 100 = maximum modulation.

Easy Edit Page

FM synthesizers were never easy instruments to program. Familiar synth parameters were suddenly replaced by a confusing array of operators, modulators, rate/level envelopes, and other terms that had no precedent with analog synthesis. No wonder so many people bought third-party sounds!

In designing FM7, Native Instruments has created several algorithms that provide common synth programming functions, and know which FM parameters to manipulate to create these effects. For most sounds, the Easy Edit page will be sufficient to customize a sound for your needs. Let’s look at the available parameters; call up a preset and experiment with the settings to hear how each one affects the sound.

All the Easy Edit sliders go positive and negative, to let you increase or decrease the relevant properties of the sound. When the sliders are centered (zero), you hear the original sound.

Main Settings

Apply

If you like the edits you’ve made to the sound, click on Apply to store the edited sound in the Edit buffer. This means that the Easy Page settings revert to zero, but the relevant sound settings on the other pages are changed so that you get the same sound. You can then continue Easy-Editing from there.

Reset

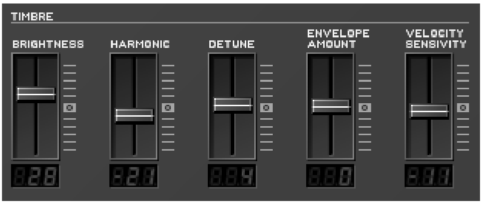
Use this to return to the original, pre-edited sound.

Effect Select

This lets you choose from several preset effects settings, as well as Off (no effect at all) and Normal (the existing effect settings take priority).

The FX Strength parameter varies the intensity of the effect. 0 = original intensity, -99= effect disabled, 100 = maximum effect.

Timbre



Brightness increases the sound's treble content.

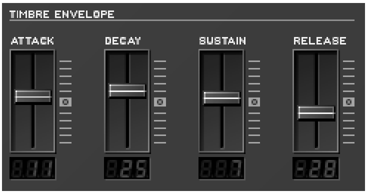
Harmonic changes the harmonic content (by adjusting the operator Ratios). Depending on the patch, the changed harmonics may be somewhat atonal.

Detune changes the pitch slightly of different oscillators in a multi-oscillator preset, which creates a "fatter" sound.

Envelope Amount sets how much the timbre envelope influences the sound.

Velocity Sensitivity determines the extent to which changes in timbre correlate to the dynamics of your playing.

Timbre Envelope



The FM7 multi-stage envelopes allow for extremely precise control, but are time-consuming to set up. The Timbre Envelope brings all these parameters together into the familiar ADSR (Attack Decay Sustain Release) envelope format. Changing any Timbre Envelope parameter causes multiple changes in the individual operator envelopes to accomplish the desired effect.

Attack Positive values increase the existing attack time, negative values decrease the attack time.

Decay Positive values increase the existing decay time, negative values decrease the decay time.

Sustain Positive values increase the existing sustain level, negative values decrease the sustain level.

Release Positive values increase the existing release time, negative values decrease the release time.

LFO

As with the envelopes, the LFO parameters tie in with multiple parameters. Here’s what each control does.

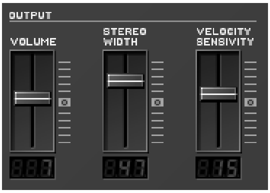
Rate changes the LFO speed. Positive values increase the speed, negative values decrease it.

Vibrato alters how much the LFO modulates pitch. Positive values increase the amount of pitch modulation, negative values reduce it.

Timbre determines how much the LFO modulates the frequency response. In an analog synth, this is equivalent to altering the amount of filter modulation from an LFO. Positive values increase the amount of timbre modulation, negative values reduce it.

Tremolo sets the extent to which the LFO modulates amplitude. Positive values increase the amount of amplitude modulation, negative values decrease it.

Output



Volume changes the overall preset (not instrument) level. Positive values raise the volume, negative values decrease it.

Stereo Width affects the spread of operators in the stereo field. Positive values increase the width to create a wider stereo spread, negative values collapse the spread to center (mono).

Velocity Sensitivity sets how velocity affects the overall output level. Positive values increase velocity response, while negative values decrease the response to velocity.

Amplitude Envelope

This allows tweaking amplitude (level) characteristics in a manner similar to using the Timbre Envelope to tweak timbral characteristics. Changing any Amplitude Envelope parameter causes multiple changes in the individual operator envelopes to accomplish the desired effect.

Attack Positive values increase the existing attack time, negative values decrease the attack time.

Decay Positive values increase the existing decay time, negative values decrease the decay time.

Sustain Positive values increase the existing sustain level, negative values decrease the sustain level.

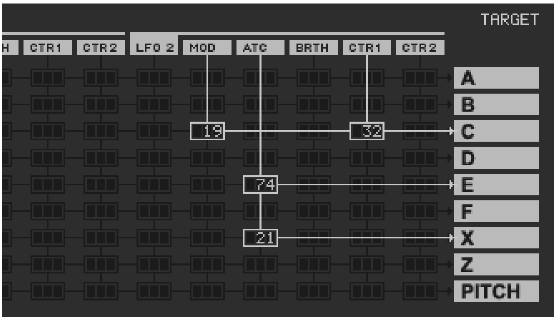
Release Positive values increase the existing release time, negative values decrease the release time.

Modulation Page

Although each operator page shows modulation information for that individual operator, this page shows the modulation parameters for *all* operators, in one convenient screen.

Modulation Basics

The amplitude of each operator has a major effect on the overall sound. Therefore, many modulation sources are available to vary the operator amplitude in real time.



Modulation is set up as a matrix, with columns of modulation sources and rows of modulation targets (destinations). At each junction, you can vary the amount of modulation going to the destination.

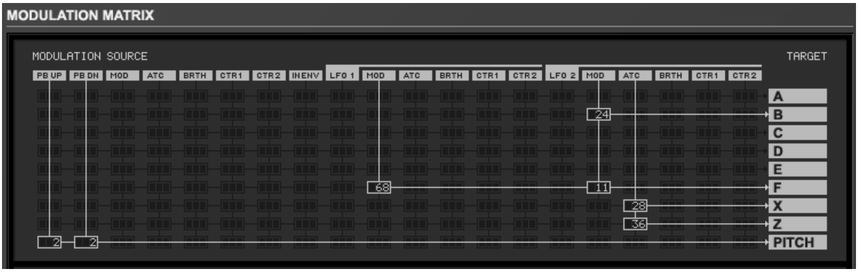
Adding a Modulation Control

- Imagine a line going downward from the modulation source. Imagine a second line going across to the modulation target.
- Click and hold at the junction represented by a black field in the background.
- Drag up. The imaginary lines are now drawn in, and a box appears with a numerical. Drag until the numerical shows the desired modulation level.

Removing a Modulation Control

Click on the numerical box, and drag down. Upon reaching 0, the box and lines disappear.

The Modulation Matrix Router



This shows all modulation sources and targets. Available modulation sources (left to right) are:

PB UP Pitch bend up range

PB DN Pitch bend down range, applied in negative direction

MOD Modulation wheel (MIDI controller #1)

ATC Aftertouch (channel pressure)

BRTH Breath controller (MIDI controller #2)

CTR1 1st MIDI controller assigned on Master Page

CTR2 2nd MIDI controller assigned on Master Page

IN ENV Envelope derived from input signal amplitude

The parameters for LFO1 and LFO2 are identical, so we will list those only for LFO1.

LFO1 Main LFO output

LFO1 MOD LFO1 output controlled by mod wheel

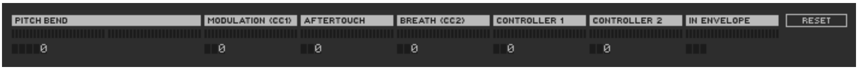
LFO1 ATC LFO1 output controlled by aftertouch

LFO1 BRTH LFO1 output controlled by breath controller

LFO1 CTRL1 LFO1 output controlled by the 1st MIDI controller assigned on the Master page

LFO1 CTRL2 LFO1 output controlled by the 2nd MIDI controller assigned on the Master page

Modulation Monitor and Controls



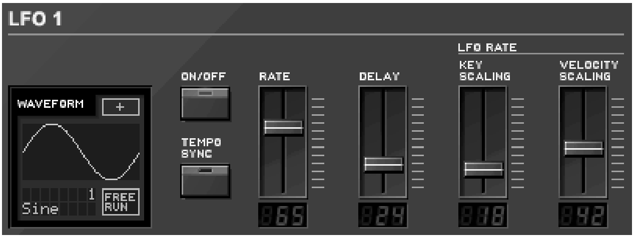
The lower strip of controls provides two independent functions: modulation input monitoring, and modulation input control by mouse. These are available for the following modulation sources: **Pitch Bend, Modulation, Breath, Controller 1** (assigned on Master page), and **Controller 2** (assigned on Master page). **Input Envelope** has only the meter, and does not include the control function.

- The meters display the incoming control signals.
- The numerical not only works as a display but also allows to enter the value by mouse action (click on it and drag up or down to change).
- The Reset button restores all displays to zero.

The LFO Page

This page contains all LFO-related parameters for the two LFOs. It has three main sections: LFO1 parameters, LFO 2 parameters, and an LFO-specific excerpt from the Modulation matrix.

LFO Parameters



The two LFOs are identical, so we will describe only LFO1.

Waveform chooses among the various modulation waveforms. The waveforms are the same as available for the FM operators. Click in the numerical and drag up or down to select the desired waveform.

Waveform Sign sets the polarity of the waveform. **+** produces positive values, **-** produces negative values.

Free Run/Key Sync has two options. With *Free Run,* the LFO waveform runs continuously. When you press a key, the modulation picks up whatever part of the LFO curve is happening at that moment. With *Key Sync,* pressing a key resets the LFO to the beginning of the waveform. *Example:* Suppose Sine is selected as the waveform. With Key Sync, upon pressing a key, the waveform starts at 0 and goes positive. With Free Run, the waveform could start anywhere - 0, the peak, the lowest value, somewhere in between, etc.

On/Off enables/disables the LFO.

Tempo Sync matches the LFO frequency to the song tempo and syncs the LFO to the beat. When disabled, the LFO ignores the song tempo.

Note: Not all VST host software supplies tempo information to plug-in instruments, so this feature may not work with your host software. In standalone-mode, the FM7 does respond to MIDI clock signals.

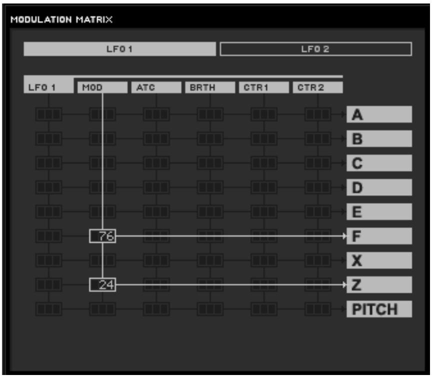
Rate sets the LFO's base frequency.

Key Scaling changes the LFO's rate as you play over the keyboard. Higher values cause the rate to speed up as you play progressively higher on the keyboard.

Velocity Scaling changes the LFO's rate according to velocity. With higher values, higher velocities increase the LFO rate.

LFO Invert flips the waveform upside down. This is useful in Tempo Sync mode, to make the modulation go down instead of up, on the beat.

LFO Page Modulation Matrix



This is a subset of the Modulation Matrix page. Please refer to that chapter for information on assigning controllers; the only difference is that to conserve space, only one LFO is visible at a time. With the two buttons at the buttom you select which LFO you want to see.

To summarize, LFO1 represents the LFO output. MOD, ATC, BRTH, CTR1, and CTR2 modulate the LFO output. Any of these modulation sources can feed any of the target destinations via use of a matrix. Clicking at the junction of a source and target creates a box with a numerical; click on the number and drag up or down to change the modulation amount.

Pitch Page

The Pitch Page has four main sections:

- An envelope that varies the note pitch over time
- A microtuning pane for creating non-standard tunings
- Pitch bend parameters and a pitch-related excerpt of the modulation matrix
- Several pitch parameter control sliders

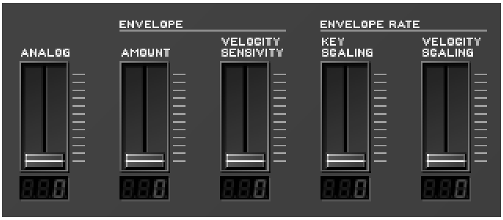
Portamento Controls

Portamento On/Off enables or disables the portamento function.

Portamento Auto allows portamento to occur only when playing legato, where there is no gap between notes. If you release a key before playing the next one, there will be no portamento.

Portmento Time sets how long it takes for the pitch to glide from one note to another. 0 = shortest time, 100 = longest time.

Envelope Controls



Analog isn't really an envelope control, but duplicates the function found on the master page that introduces random changes in the pitch to more closely simulate analog oscillators.

Envelope Amount determines how much the envelope affects pitch.

Velocity Sensitivity sets how the overall envelope amplitude responds to velocity. At lower values, velocity has less effect. At higher values, velocity kicks the envelope up higher.

Key Scaling edits how keyboard pitch affects the envelope times. At higher values, playing higher up on the keyboard shortens all time values. This emulates many plucked instruments, whose notes attack and decay more quickly at higher pitches.

Velocity Scaling edits how velocity affects the envelope times. At higher values, hitting the keys harder shortens all time values.

Pitch Bend Controls

Pitch Bend Mode offers four modes.

- *Normal* affects all notes equally.
- *Highest* affects only the highest note if several notes are held. This is designed to provide guitar-like pitch bending effects.

- *Lowest* affects only the lowest note if several notes are held.
- *Keyon* allows pitch bend to work only as long as a key is held down. There is no bend during the sound's release phase.

Tune adds a fixed amount of detuning from -99 to +100 cents.

Transpose adds a fixed amount of detuning in semitone steps, from -24 to +24 semitones.

Pitch Modulation

This is a subset of the Modulation Matrix page. Please refer to that chapter for information on assigning controllers; the only difference is that to conserve space, the pitch modulation sources are arranged in two lines stacked above each other, instead of one long line.

Microtuning Controls

Even-tempered tuning, which divides an octave into 12 equal intervals, was developed to simplify modulation. However there are many other scales, such as just intonation, microtunings, stretch tunings so that pianos sound more in tune, etc. Several of these are available in the FM7 as presets, or you can create your own tunings using the microtuning controls below the pitch envelope.

Each of the twelve semitones that make up an octave (C, C#, D, etc.) has an offset control that can change the note's tuning by -99 cents to +100 cents. Zero offset gives a standard even-tempered scale. Any offsets are preserved throughout all octaves *(example:* if D is offset by +3 cents, all D notes will be offset by +3 cents).

Stretch Octaves

The Octave stretch parameter leaves middle C unaffected, but positive offset values create progressively sharper note tunings when playing higher up on the keyboard, and progressively flatter note tunnings when playing lower on the keyboard. A negative offset value does the reverse - higher notes become progressively flatter, while lower notes become progressively sharper.

An offset of 50 compresses the scale into two semitones per key, or two octaves into one octave. An offset of -50 does the opposite, expanding the normal scale into a quarter tone scale (in other words, there are 24 notes per octave).

Pianos use *stretch tuning,* which can be simulated in the FM7 by a *small* positive offset. This is because the harmonics of lower strings are slightly sharp, as the strings cannot be as long as the laws of physics would like, because of a piano's size constraints. Therefore, tuning them slightly flat allows the harmonics to fall into tune with the rest of the piano. A related phenomenon occurs at the high end, where notes have to be tuned slightly sharp to sound in tune with harmonics from lower strings.

With small offset values (near 0), the number is an octave's detuning in cents. *Example:* a reading of +5 means that each octave is 5 cents sharp.

Selecting and Saving Microtuning Presets

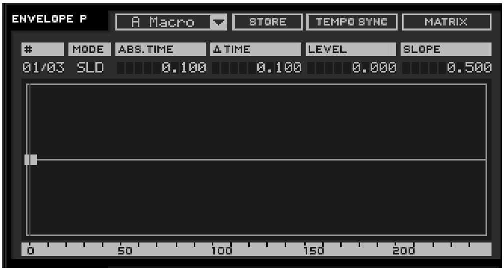
To select a Microtuning preset click on the arrow in the right edge of the name field and choose a preset from the drop down list.

To save a tuning preset you created, enter a name in the name field and click on Store.

Pitch Envelope

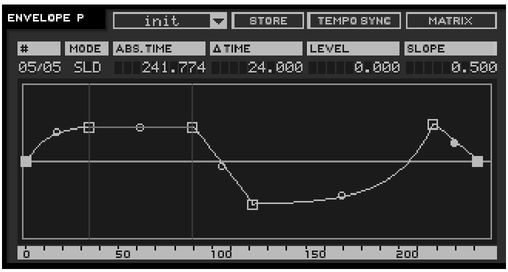
The pitch envelope varies pitch over time. Envelopes are time/level types, where you specify levels, and the times that it takes to get from one level to the next. Levels are specified by inserting *breakpoints* along a line, then moving the breakpoints up or down to change levels, and left or right to change times.

Here is the initialized pitch envelope:



Right-click (Mac: ctrl-click) wherever you want a breakpoint. The first and last breakpoints move up and down together, because the envelope starts from where it ended.

The following diagram shows an envelope with multiple break points.



The left-most breakpoint is where the envelope starts. The next two breakpoints fall on two vertical red lines. These indicate the segment where the envelope reaches the "sustain" pitch, marked by the horizontal red line. On this pitch the envelope stays as long as the key is held down.

If you insert one or more breakpoints between the red markers the "sustain" segment becomes a "loop". This means that this part will be repeated as long as the note is sustained. You can create novel vibrato effects by changing the positions of these breakpoints.

If there are more than 3 breakpoints the sustain or loop segment can be shifted by moving the vertical red lines with the mouse. They snap to the next breakpoint, except the start and the end point.

Upon releasing the key, the envelope continues with the segment after the second red marker. In our example there are two more breakpoints after the sustain, one below and one above the center line, then the final end point.

Dragging the small circle between breakpoints can change the line’s shape between the breakpoints from concave, to straight, to convex.

Envelope Parameter Strip

Now let’s investigate the envelope parameter strip above the envelope. Except as noted, clicking on a numerical and dragging up or down edits the value; they also update automatically if you move the breakpoint.

shows two numbers. The first field is the index number of the breakpoint or stage being edited. It can be changed by mouse move to select another stage. The second field is the total number of envelope breakpoints. This is for display only and cannot be edited.

Mode has two options, SLD and FIX. In SLiDe mode, if you move a breakpoint left or right, the envelope to the right of the breakpoint moves as well to maintain the same times and levels past the breakpoint being edited.

In Fix mode, the total envelope time doesn't change. Moving a breakpoint to the right not only lengthens its distance compared to the breakpoint to its left, but shortens the distance compared to the breakpoint at its right.

Abs. Time shows the amount of time in seconds from the start of the envelope to the breakpoint being edited.

Delta Time shows the amount of time in seconds from the breakpoint being edited to the breakpoint at its immediate left.

Level shows the breakpoint level referenced to the center line.

Slope shows the state of the line between breakpoints. 0.5 indicates a straight line. 0.999 indicates a maximally convex curve. 0.001 indicates a maximally concave curve.

The numericals for Delta time, Level and Slope can be edited by mouse (click and move up or down).

Envelope Ruler and Zoom

The ruler below the envelope is calibrated in seconds. If the envelope extends past the envelope's visible range, click on the ruler and drag to the left or right to see a different range of the ruler. Double-click on the ruler to fit the envelope exactly within the visible part of the ruler.

To change the ruler range (zoom function), right-click on the ruler. Drag right to zoom out, left to zoom in. This not only changes the display, but also sets the quantization time for the Tempo Sync function (described later).

Note that when zooming, the point of the envelope where you right click is fixed. When zooming out this is only the case until the beginning of the envelope is visible at the left end of the ruler. Upon further zooming out, the zero point becomes fixed.

Envelope Function Strip

The Envelope Function Strip offers several useful functions.

Envelope Preset offers a drop-down menu that comes with several common envelope types.

Store lets you save a particular envelope in the preset list. To save an envelope you created, enter a name in the name field and click on Store.

Tempo Sync superimposes a grid on the envelope display that corresponds to rhythmic values, such as quarter notes, eighth notes, 16th notes, etc. Breakpoints can be snapped to these points.

The quantization value depends on the current zoom factor, as set by right-clicking on the envelope ruler and dragging. (When Tempo Sync is enabled, the ruler markings will be calibrated in rhythmic values instead of seconds.) If the incoming tempo information changes, the envelope times are re-computed to match the current song tempo. *Example:* if the envelope loop is set to a 1 beat duration, it will remain 1 beat long as the tempo changes.

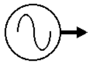
Enabling Tempo Sync does not change breakpoint positions in existing envelopes. However, you can edit an existing envelope by moving the breakpoints so that they snap to the nearest quantization value.

Note: Not all VST host software supplies tempo information to plug-ininstruments, so this feature may not work with your host software. In standalone-mode, the FM7 does respond to MIDI clock signals.

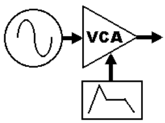
Matrix is a shortcut to the algorithm creation matrix screen.

FM Synthesis Programming Basics

The principle behind FM synthesis is remarkable in its simplicity. The basis of all FM synthesis is a sine wave oscillator, represented below as a circle and an output.

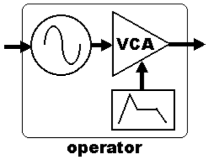


A keyboard or other controller sets the oscillator's pitch. However, we now need a way to gate this oscillator on and off. Analog synthesizers used a circuit called a *VCA* (voltage-controlled amplifier) to alter the oscillator level. The level depended on a control signal (called a *control voltage)* fed into the VCA. Many modern digital synths and programs still use this terminology, even though the level changes are all generated digital, by altering numbers within the program. Some digital synths refer to VCA as *DCA* (for Digitally Controlled Amplifier"), while others just refer to it as *amp.*.



With analog synths, the control signal that changes the level is generated by a circuit called an *envelope generator* (EG). It causes the level to change in a predictable way over time. For example, to create a plucked sound, the envelope might start at a very high value and then drop over several milliseconds to a much lower value, or even turn off completely. In digital gear, the program generates a data stream that changes level the way an envelope generator would. However, the concept is still usually referred to as envelope generation. This grouping of a sine wave oscillator, VCA/DCA, and EG is called an *Operator,* which is the basic building block of FM synthesis.

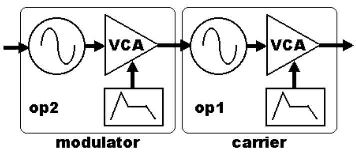
As a pure sine wave is pretty boring from a musical standpoint, this leads us to the extremely clever aspect of FM synthesis. Let's add a control input to the operator to modulate its overall amplitude.



Feed a sine wave into that control input. A low frequency wave produces tremolo by slowly changing the level over time. But an audio range signal produces one of two results. Signals that are not harmonically-related to the main oscillator create "clangorous" sounds. Injecting a harmonically-related signal generates harmonics that sound more "in-tune." (Both types of effects can be useful.)

The amount of harmonics depends on the *level* of signal injected into the main oscillator, and the harmonic structure depends on the modulating oscillator's *frequency.* As timbre is primarily affected by the injected signal's amplitude, adding a VCA after the modulating oscillator (along with an envelope generator to control the VCA/DCA) allows predictable control over the signal, hence the overall timbre. Our FM synthesis block diagram now looks like this.

Note how the same operator structure can provide an audio signal

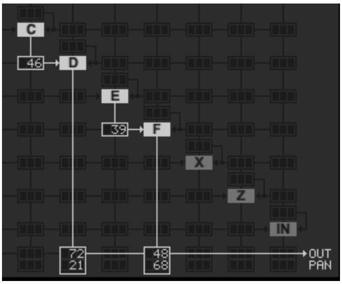


(Operator 1) or modulate that audio signal (Operator 2), so we need to differentiate between the two functions. The operator we hear is called the *carrier.* An operator is called a *modulator* if it modulates the carrier.

This two-operator structure can actually make some very sweet brass timbres. Increasing the output of Op 2 creates a sound somewhat like opening up a lowpass filter; decreasing the output is like closing the filter down.

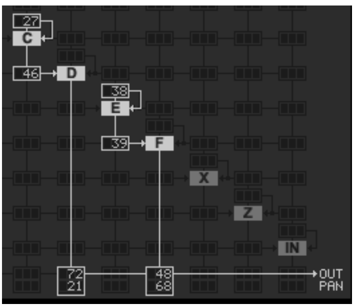
Meet The Algorithm

You can combine operators in various ways to create a variety of Algorithms. The FM7 has several preset algorithms, accessible from a drop-down menu on the FM Matrix page. They use up to six conventional operators, and you can see how they are combined in the various macro algorithms. For example, the following algorithm has two carriers. Each has its own modulator.



Feedback

The next diagram shows the previous algorithm, but with feedback added to the modulators - in effect, an operator becomes its own modulator. Increasing feedback can add bite and grittiness.



The Envelope Generator

The envelopes in vintage synths worked on the rate/level principle, which specified the *rate* at which the envelope goes from one *level* to another. Unfortunately this was quite confusing, as going from zero to a high level would take longer than going from zero to a low level, given the same rate. The FM7 solves this problem by letting you set a specific time for one level to transition to another, and does all the necessary calculations internally to convert this into the correct rate.

The envelope level typically starts at zero. To create an attack, you specify the level to be attained, and the time it takes for the envelope to reach that level. Other stages of the envelope are set similarly, according to times and levels. Going from a higher level to a lower one produces a decay; going from lower to higher produces an attack.

About the Next Chapter

Now it's your turn to experiment. The next two chapters explain how to program algorithms, and the operators inside the algorithms. This is fairly heavy stuff, so it’s very helpful to call up some of the FM7 presets, and analyze how the parameters are set. Turn off each operator, one by one, to see how that operator affects the sound. As you vary parameters, note how each parameter changes the overall sound. Studying how the presets are constructed will teach you how to tweak them for your own particular application. For example, if you want a mellower sound, you'll know to turn down the modulator output levels.

And don't forget the many expressive possibilities: the operator amplitude can be tied to velocity, and adding some LFO in the right places can do wonders. Keyboard and level scaling are also very helpful when trying to achieve sounds that resemble acoustic instruments, as they allow the sound to change dynamically as your playing moves between higher and lower pitches.

Furthermore, the FM7 adds two new elements not used in traditional FM synthesizers: a saturator module and a filter. Both of these are extremely useful to add another dimension to the basic operator-generated sound.

Algorithm Programming

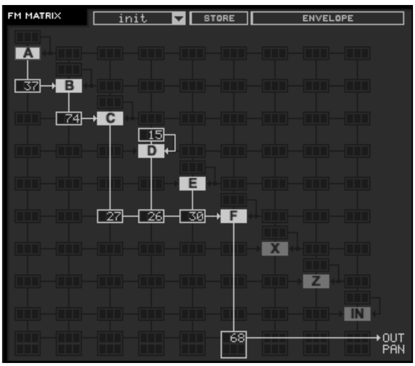
Although the FM7 comes with numerous presets and can load in presets from FM synths, learning how to program the FM7 will reward you with unique sounds that have your own sonic signature. The depth and range of FM synthesis is astonishing, from complex, acoustic-like timbres to wild sound effects.

Programming starts with an algorithm, then continues with programming the various operators that make up the algorithm. The final programming step involves adding effects and tweaking.

Accessing the FM (Algorithm) Matrix

Unlike FM synthesizers, which had a fixed selection of algorithms, you can create a virtually infinite number of algorithms via the FM7's programming matrix. You can access the matrix from any Operator programming page (A-F, X, or Z) or the Pitch page. If the matrix isn't visible, click on the Matrix label in the upper right corner of the envelope pane.

The matrix lets you route modulators to carriers, and send carriers to the audio output. Any operator can be a modulator, a carrier, or both simultaneously, and any number of modulators can modulate any number of carriers. Let's analyze the algorithm for the Preset Piano 2 included in the DX7 bank delivered with the FM7.



Operator A modulates Operator B, which is a modulator for Operator C. Operators C, D, and E all modulate Operator F, which serves as a carrier, and feeds the output. Note that Operator D has feedback.

Inactive operators are „greyed out“. In the X and Y operators the letters are greyed out if Bypass is activated.

The following mouse operations and key commands are available in the Matrix:

- Click on operator selects the appropriate operator page
- Right click (Mac: Ctrl click) on operator switches it on and off.
- Shift+Right Click (Mac: Shift+Ctrl click)on operator X and Z switches Bypass on and off (if operator X/Z is active)
- Double click on operator jumps to its envelope.

Constructing an Algorithm

- To send an operator output to another operator input, imagine a line going downward from the first operator. Imagine a second line going across to the right, to the target operator.
- Click and hold at the junction of these two imaginary lines, as represented by a dark field in the background where you click.
- Drag up. The imaginary lines are now drawn in, and a box appears with a numerical. This controls the level of the modulator feeding the carrier. Drag until the numerical shows the desired level.
- Any operator can feed back not just to itself, but to any other operator. To create feedback from one operator to another, imagine a line going upward from the first operator and a second line going across to the left, to the target operator.
- Click and hold at the junction of these two imaginary lines.
- Drag up. The imaginary lines are now drawn in, and a box appears with a numerical that acts like a volume control. Drag until the numerical shows the desired level.

Removing an Algorithm Connection

Click on the numerical box, and drag down. Upon reaching 0, the box and lines disappear.

Matrix Function Strip

Three functions are available.

Drop-Down Menu presents a list of preset algorithms, including all those found in the original DX7.

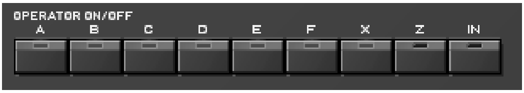
Store lets you save an algorithm to the list of presets. To save an algorithm you created, enter a name in the name field and click on Store.

Envelope is a shortcut to the selected operator's envelope page.

Programming Operators

Each of the 6 operators (labeled A-F) has its own page where you may program its parameters. You select the page via the operator selector strip. The selected operator's label turns red. *Note:* You can also select an operator by clicking on its associated letter in the algorithm creation matrix.

Operator On/Off Selector

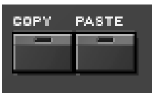


While programming, it's convenient to be able to turn off extraneous operators so you can focus on the sound created by a particular operator or group of operators. The Operator On/Off buttons are available on every operator page. Click on a button to enable the operator (button LED lights red), click again to disable (button LED goes out).

A right click (Mac: Ctrl click) on an active X or Z operator switches it to bypass mode (button LED lights yellow).

Operator A-E

Copy and Paste

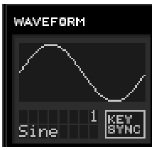


This is a shortcut for copying one operator's settings to another. *Example:* Suppose one operator provides a signal to the audio output, and you want to double this with an operator exactly one octave higher. With the operator page showing for the operator you want to copy, press Copy. The Paste indicator turns red to show that there are operator parameters in the copy buffer and ready to paste.

Now select the page for the destination operator, and click on Paste. The opeator parameters are transferred. Now you can tweak the destination operator.

Waveform Selection

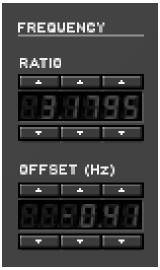
An operator can have any of 32 different waveforms (the original DX7 had only sine waves). Clicking and dragging up or down on the numerical selects the waveform.



Key Sync vs. Free Run

Clicking on this box toggles between these two selections. *Key Sync* mode resets the oscillator phase at the beginning of each note. When any group of operators needs to have a fixed phase relationship to preserve a particular tone quality, activate Key Sync for all operators in the group.With *Free Run* mode, there is no phase reset.

Frequency Ratio and Offset



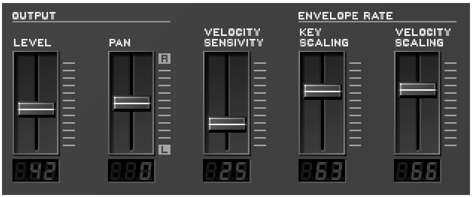
Frequency Ratio defines the mathematical relationship of the operator frequency compared to the fundamental frequency of the note being played. 1.0000 means the operator is the same pitch as the fundamental, 2.000 sets the operator to the second harmonic (1 octave higher), 3.000 is the third harmonic (octave + fifth), etc. 0.5000 is the subharmonic one octave below the fundamental.

These numericals are edited by clicking on one of the digits and moving it up or down. The arrows above and below each numerical allow incrementing (up arrow) or decrementing (lower arrow) one value at a time.

When the Ratio is not an exact integer (e.g. 1.0030) then the operator is detuned and will "beat" against other operators. The beat frequency rises if the note pitch increases, and falls if the note pitch decreases.

Frequency Offset applies a constant frequency offset (in Hz) to the operator to cause detuning and beating against other operators. This is constant regardless of pitch. *Example:* a 5 Hz Offset causes beating at exactly 5 Hz between the selected operator and another operator with no offset.

Output, Velocity, and Envelope Rate Controls



Level serves as a master level control for all numericals in the matrix column associated with the selected operator, except for any feedback to itself. *Example:* If operator E modulates operator F and sends its output to the main output, there are two numericals in the column, one connecting to operator E and one to the output. Lowering or raising Level will lower or raise both numericals. Similarly, raising or lowering the highest-value numerical will raise or lower the slider.

Note: the Level slider will always assume the value of the column numerical with the highest value. It also works ratiometrically. In other words, if one numerical is at 50 and another is at 100, reducing the second numerical from 100 to 50 doesn't reduce the other other numerical to 0, but to half of its value, as 50 is half the value of 100.

Pan mirrors an operator's pan control in the algorithm creation matrix, and edits the audio output's placement in the stereo field, from left to right. Changing this control changes the corresponding numerical in the matrix; changing the numerical similarly changes the slider.

Velocity Sensitivity sets how the overall envelope amplitude responds to velocity. At lower values, velocity has less effect. At higher values, velocity kicks the envelope up higher. If the operator is a carrier, higher-amplitude envelopes increase the volume. If the operator is a modulator, higher-amplitude envelopes change the timbre by increasing the brightness.

Key Scaling edits how keyboard pitch affects the envelope times. At higher values, playing higher up on the keyboard shortens all time values. This emulates many plucked instruments, whose notes attack and decay more quickly at higher pitches.

Velocity Scaling edits how velocity affects the envelope times. At positive values, hitting the keys harder shortens all time values. At negative values the enevlope times become longer with higher key velocity.

Amplitude Modulation

This is a subset of the Modulation Matrix page. Please refer to that chapter for information on assigning controllers; the only difference is that to conserve space, the pitch modulation sources are arranged in two lines stacked above each other, instead of one long line.

Matrix/Envelope Pane

The lower right-hand pane displays either the algorithm programming FM matrix, or the amplitude envelope and keyscaling graph. If the matrix is visible, click on the Envelope label in the upper right-hand corner of the pane to see the envelope/keyscaling graphs. If the envelope/keyscaling graphs are visible, click on the Matrix label in the upper right-hand corner of the pane to see the matrix.

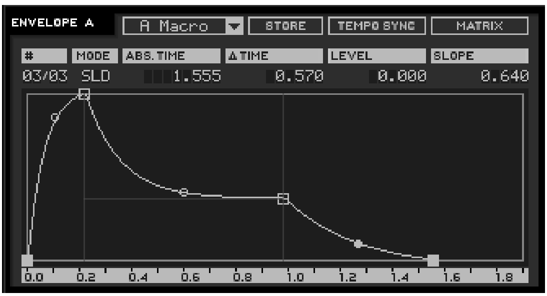
Operator Amplitude Envelope

The amplitude envelope varies the operator amplitude over time. Envelopes are time/level types, where you specify levels, and the times that it takes to get from one level to the next. Levels are specified by inserting *breakpoints* along a line, then moving the breakpoints up or down to change levels, and left or right to change times.

Here is the initialized amplitude envelope:

Right-click (Mac: ctrl-click) wherever you want a breakpoint. The first and last breakpoints move up and down together, because it is assumed you will eventually want the envelope to end up where it started.

The following diagram shows an envelope with multiple break points:



The left-most breakpoint is where the envelope starts. The next breakpoint to the right sets the envelope's maximum level; the first two breakpoints therefore create the envelopes's attack. The next breakpoint to the right specifies the level to which the envelope will fall after passing through the attack phase (decay).

This and the maximum breakpoint fall on two vertical red lines. These indicate the segment where the envelope reaches the "sustain" level, marked by the horizontal red line. On this level the envelope stays as long as the key is held down.

If you insert one or more breakpoints between the red markers the "sustain" segment becomes a "loop". This means that this part will be repeated as long as the note is sustained. You can create novel tremolo effects by changing the positions of these breakpoints.

If there are more than 3 breakpoints the sustain or loop segment can be shifted by moving the vertical red lines with the mouse. They snap to the next breakpoint, except the start and the end point.

Upon releasing the key, the envelope continues with the segment after the second red marker. In our example there is one more breakpoint after the sustain, the final end point.

Dragging the small circle between breakpoints can change the line’s shape between the breakpoints from concave, to straight, to convex.

Envelope Parameter Strip

Now let’s investigate the envelope parameter strip above the envelope. Except as noted, clicking on a numerical and dragging up or down edits the value; they also update automatically if you move the breakpoint.

shows two numbers. The first field is the index number of the breakpoint or stage being edited. The second field is the total number of envelope breakpoints. This is for display only and cannot be edited.

Mode has two options, SLD and FIX. In SLiDe mode, if you move a breakpoint left or right, the envelope to the right of the breakpoint moves as well to maintain the same times and levels past the breakpoint being edited.

In Fix mode, the total envelope time doesn't change. Moving a breakpoint to the right not only lengthens its distance compared to the breakpoint to its left, but shortens the distance compared to the breakpoint at its right.

Abs. Time shows shows the amount of time in seconds from the start of the envelope to the breakpoint being edited.

Delta Time shows shows the amount of time in seconds from the breakpoint being edited to the breakpoint at its immediate left.

Level shows the breakpoint level referenced to the center line.

Slope shows the state of the line between breakpoints. 0.5 indicates a straight line. 0.999 indicates a maximally convex curve. 0.001 indicates a maximally concave curve.

Envelope Ruler and Zoom

The ruler below the envelope is calibrated in seconds. If the envelope extends past the envelope's visible range, click on the ruler and drag to the left or right to see a different range of the ruler. Double-click on the ruler to have the envelope fit exactly within the visible part of the ruler.

To change the ruler range (zoom function), right-click on the ruler. Drag right to zoom out, left to zoom in. This not only changes the display, but also sets the quantization time for the Tempo Sync function (described later).

Note that when zooming, the point of the envelope where you right click is fixed. When zooming out this is only the case until the beginning of the envelope is visible at the left end of the ruler. Upon further zooming out, the zero point becomes fixed.

Envelope Function Strip

This strip, located above the Envelope Parameter Strip, offers several useful functions.

Envelope Preset offers a drop-down menu that comes with several common envelope types.

Store lets you save a particular envelope in the preset list. To save an envelope you created, enter a name in the name field and click on Store.

Tempo Sync superimposes a grid on the envelope display that corresponds to rhythmic values, such as quarter notes, eighth notes, 16th notes, etc. Breakpoints can be snapped to these points.

The quantization value depends on the current zoom factor, as set by right-clicking on the envelope ruler and dragging. (When Tempo Sync is enabled, the ruler markings will be calibrated in rhythmic values instead of seconds.) If the incoming tempo information changes, the envelope times are re-computed to match the current song tempo. *Example:* If the envelope loop is set to a 1 beat duration, it will remain 1 beat long as the tempo changes.

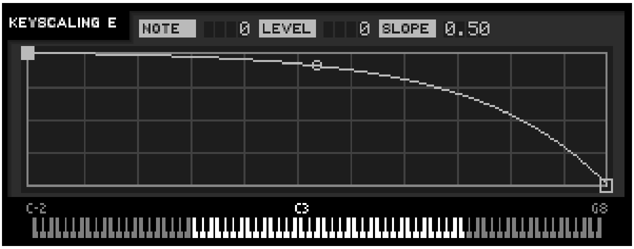
If an envelope has already been created, enabling Tempo Sync will not change the existing breakpoints. However, you can edit an existing envelope by moving the breakpoints so that they snap to the nearest quantization value.

Note: Not all VST host software supplies tempo information to plug-ininstruments, so this feature may not work with your host software. In standalone-mode, the FM7 does respond to MIDI clock signals.

Matrix is a shortcut to the algorithm creation matrix screen.

Keyscaling Graph

Keyscaling sets how the operator amplitude changes across the keyboard range. Like the envelopes, you can create/delete breakpoints to create a particular curve shape, and dragging the small circle between the breakpoints can change the line's shape from concave, to straight, to convex..



A small keyboard graphic below the curve serves as a reference for breakpoints. The white area corresponds to the range of the FM7 keyboard, or a standard 5-octave MIDI keyboard controller.

The default keyscale line is at 0 (maximum level), as indicated by a horizontal line going across the top of the graph. You reduce the level for a particular area of the keyboard by lowering the line over a particular keyboard range. In the screen shot above, amplitude falls off starting at around middle C, and the slope gets progressively steeper the higher you play on the keyboard.

Here's what the various parameters indicate.

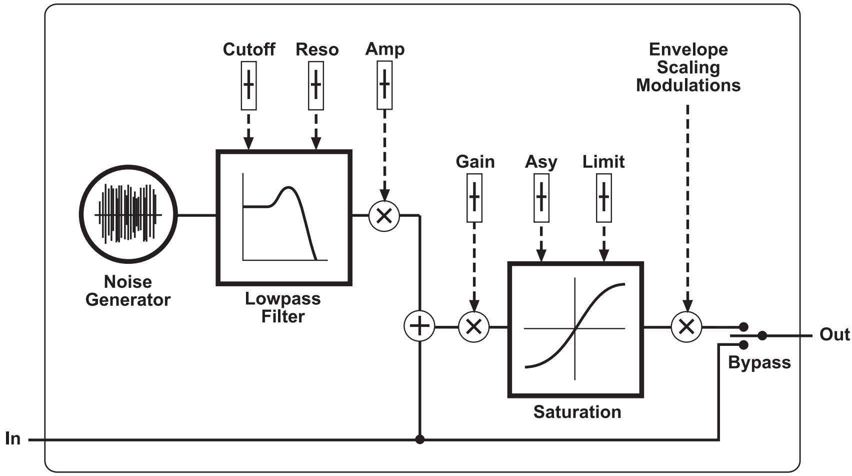
Note shows the MIDI note number of any breakpoint on which you click, or when you create a breakpoint. Moving the breakpoint updates the MIDI note number.

Level shows the level of any breakpoint on which you click, or when you create a breakpoint. Moving the breakpoint updates the level. Levels are referenced to 0, so the more negative the number, the lower the level.

Slope shows the state of the line between breakpoints. 0.5 indicates a straight line. 1.000 indicates a maximally convex curve. 0.000 indicates a maximally concave curve.

Operator X

This special-case operator resembles a conventional operator in many ways: its amplitude can be envelope-controlled, keyscaled, amplitude modulated, and fed back to other operators. However, it both generates noise and can process an input signal with distortion. The noise is mixed with the input signal, processed by a saturation stage and multiplied with the envelope. Unlike the other operators, it has has two pages of parameters.



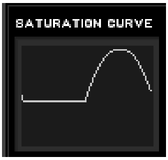
Operator X block diagram

Page and Bypass Selection

Page 2 selects the second page of parameters. When selected, its light turns red to indicate that page 2 parameters are being displayed.

Bypass offers a convenient way to turn off operator X's effects. When bypassed, operator X essentially becomes transparent.

Saturation Curve Display

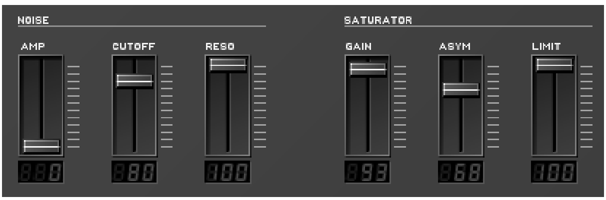


This display shows the effects on the waveform of manipulating the various saturation controls. It has no adjustable parameters.

Amplitude Modulation

This is a subset of the Modulation Matrix page. Please refer to that chapter for information on assigning controllers; the only difference is that to conserve space, the pitch modulation sources are arranged in two lines stacked above each other, instead of one long line.

Page 1 Controls



There are two main sections: Noise and Saturator. Noise adds digitally-generated random noise (this is very useful for modulating carriers, but also provides a useful audio output). Saturator adds limiting/distortion.

Noise Amp sets the noise level.

Noise Cutoff sets the noise generator's lowpass filter cutoff. With very low cutoff values, the noise source can serve as a low frequency random modulator.

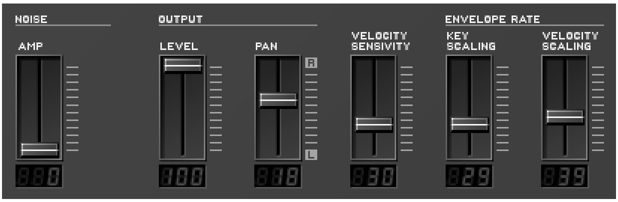
Noise Reso sets the noise generator's lowpass filter resonance. It adds a sense of pitch to the noise.

Saturator Gain sets the level going through the Saturator. The higher the level, the more pronounced the effects of the Asyn and Limit controls.

Saturator Asy offsets the symmetry of the saturation. The higher the value, the greater the saturation of negative amplitudes.

Saturator Limit clips both the positive- and negative-going portions of the waveform. Lower values produce more clipping; a value of 100 introduces no clipping. *Note:* Extreme clipping will also lower the output level.

Page 2 Controls



Noise Amp duplicates the same control on Page 1 for convenience.

All other controls are identical to the same controls in the standard operators, as described in this chapter under **Output, Velocity, and Envelope Rate Controls.**

Envelope

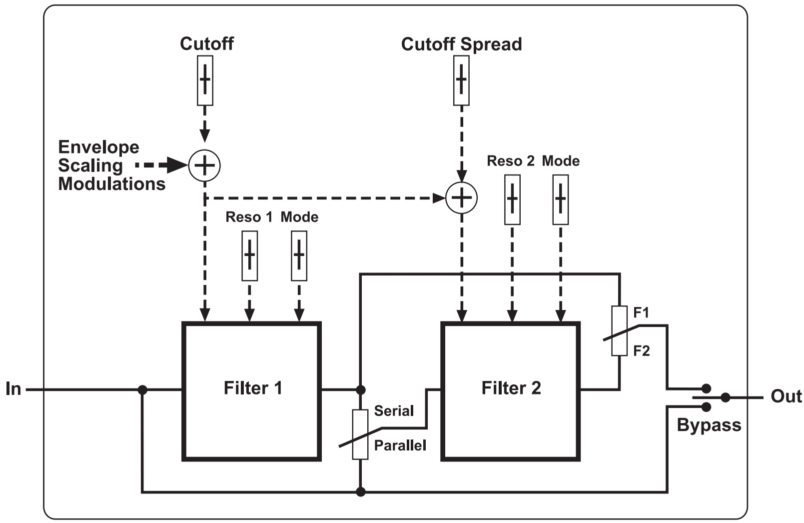
This is identical to the envelope function in the other operators, as described in this chapter under **Operator Amplitude Envelope.**

Keyscaling

This is identical to the keyscaling function in the other operators, as described in this chapter under **Keyscaling Graph.**

Operator Z

This special-case operator is a signal processor that contains two separate filters (each a 2-pole 24dB/oct multimode filter), which can be combined in almost any imaginable configuration. Yes, FM synthesis finally has a great-sounding, analog-style multimode filter.



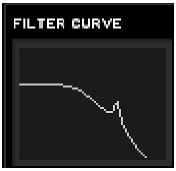
Operator Z block diagram

Page and Bypass Selection

Page 2 selects the second page of parameters. When selected, its light turns red to indicate that page 2 parameters are being displayed.

Bypass offers a convenient way to turn off operator Z's effects. When bypassed, operator Z essentially becomes transparent.

Filter Curve Display

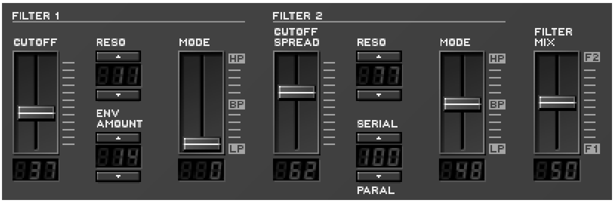


This display shows the response curve produced by the action of the two filters. It has no adjustable parameters.

Cutoff Modulation

This is a subset of the Modulation Matrix page. Please refer to that chapter for information on assigning controllers; the only difference is that to conserve space, the cutoff modulation sources are arranged in two lines stacked above each other, instead of one long line. Also note that unlike the other operators, this modulation affects the filter cutoff frequency, not an amplitude parameter.

Page 1 Controls



Page 1 has controls that relate to the two filters.

Cutoff controls the initial cutoff frequency of both filters.

Reso performs the same function for each filter, and sets the degree of resonance (boost at the cutoff frequency).

Env Amount determines how much the envelope affects the cutoff frequency. Positive values kick the cutoff higher, while negative values lower the cutoff.

Mode performs the same function for each filter. It allows setting the response mode in a continuously variable fashion from low-pass, to bandpass, to highpass.

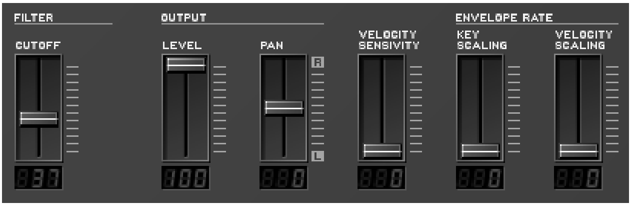
Cutoff Spread applies to Filter 2 only. It offsets Filter 2's frequency compared to Filter 1. A value of 0 sets identical cutoffs for both filters. Increasing the value increases Filter 2's cutoff compared to Filter 1.

Serial/Parallel adjusts the configuration of the two filters from series (Filter 2 follows Filter 1) to parallel (the same input signal feeds both filters, and their outputs are mixed together), to anywhere in between.

Filter Mix changes the balance of the output of the two filters, from only Filter 1 output, to both outputs, to only Filter 2 output.

Note that the display of the filter response is invaluable in figuring out how these parameters affect the filter response.

Page 2 Controls



Filter Cutoff duplicates the same control on Page 1 for convenience.

All other controls are identical to the same controls in the standard operators, as described in this chapter under **Output, Velocity, and Envelope Rate Controls.** Remember that the envelope affects filter cutoff rather than an amplitude parameter.

Envelope

This functions similarly to the envelope function in the other operators, as described in this chapter under **Operator Amplitude Envelope**, but affects filter cutoff rather than amplitude. Also note that the 1:1 option causes the filter to track the envelope in such a way as to preserve the sound's harmonic structure as you play up and down the keyboard.

Keyscaling

This is identical to the keyscaling function in the other operators, as described in this chapter under **Keyscaling Graph** except for the additional 1:1 button:

1:1 applies only to the Operator Z, the filter operator. Clicking on it creates a curve where the filter cutoff frequency tracks the keyboard in a linear fashion (i.e., the cutoff changes to retain the same harmonic structure regardless of which key you play).

Application Notes

A typical use of operator Z would be to put the filters in series and select low pass mode for both. This provides the typical 24dB/octave low pass filter response as found in traditional analog synthesizer filters, such as the Minimoog. To emulate the Oberheim filter sound, use only one filter, as these filters had a 12 dB/octave response. These configurations are particularly effective when sweeping the filter with the envelope.

Another option is to put the filters in parallel, choose bandpass mode with relatively high resonance, and spread them a bit to simulate formants, such as vocal formants (this usually doens't require cutoff frequency modulation).

Typically, the signal produced by the waveform operators feeds through operator Z to make it softer and warmer (possibly just after passing through the operator X).

Of course, with the FM Matrix, operator Z can connect anywhere between the other operators.

Preferences

The **PREF** button opens a dialogue where you can set some options for your preferred FM7 operating modes.

Memory Protect

This switches into a safer mode of storing presets. If it is enabled you will be prompted, when you click on the Store button, before overwriting a preset in memory. The default setting is *on*.

Compare Mode

You can choose from two Compare operation modes:

When *Compare with Preset* is on, Compare discards the changes in the edit buffer and restores the settings of the latest recalled preset.

In the *Back to Last Compare* mode, pressing Compare restores the settings before the last Compare action. This allows to develop a sound step-by-step, with a one-step undo.

In either mode, Compare can be used after recalling a preset to return to your latest editings before the recall.

Velocity

The original DX7 has a note velocity range from 0 to 100, while all other keyboards and sequencers generate notes with velocities between 0 and 127. If you control the FM7 from a DX7, you can use the *DX7 Keyboard* option to adapt the velocity range. With a standard keyboard connected in this mode, velocities higher than 100 let the presets sound brighter than they have been programmed. In the *Standard Keyboard* mode a DX7 would not use the full dynamic range, so the sounds would be softer than intended.

MIDI Controller Range

Many FM7 parameters have a range from 0 to 100. If you remote control the FM7 by MIDI control changes it can be advantageous to read the identical values between 0 and 100 from the remote device. This can be attained with the *0...100* option; values higher than 100 will be clipped to 100.

Note: this does not apply to parameters that can be negative (like Pan). In the standard setting, *0...127,* the values are mapped to the full range of an FM7 parameter.

The second pair of switches let you decide how the buttons react to MIDI controllers. The standard mode is *0...63: Off / 64..127: On,* where values smaller than 64 set the button to the off position, and values greater than 63 set it on. Some MIDI instruments, especially with multi-step selector switches, send with each change a value from the sequence 0, 1, 2, 3, ... Here the *Even: Off / Odd: On* option is appropriate, so that the FM7 switches toggle on/off with each action of the remote control switches.

MIDI Assign

When *Data Entry* is activated, selected parameters are outlined and can be controlled by external hardware via MIDI control changes. *Data Entry Controller* sets the number of the MIDI controller to be used for this.

The *Use Op A Controllers for Selected Op* function allows a page-oriented assignment of MIDI controllers. The controllers assigned to operator A's parameters will control the parameters of another operator, when you switch to its page. The switching between operators is assigned to a MIDI controller with the number set by *Op Select Controller*.

Some MIDI hardware units allow the incremental control, and sometimes the display, of the current state of the values. This has the advantage that knob movements do not cause any jumps. If you use such a device, enable *Send Controllers When Changed* in order to provide the hardware with the latest values set in the software. *CC Send MIDI Channel* allows choosing a separate channel for these controller events.

MIDI Learn

Remember to disable MIDI Learn as soon as a MIDI controller has been assigned to an FM7 control; this ensures that the assignment will not be overwritten by later events. But you may want to keep MIDI Learn enabled should you want to assign several controls in a row. In this case, disable *Switch Off After Assignment.* Learn is then finished only by a click on the Learn button.

Automation

If you run FM7 as a plug-in in Logic and get a problem with the automation, enable the *Workaround for Emagic Logic* function. It works as follows:

Logic filters out most controllers for VST instruments, as the controllers are used to automate the channel busses (exceptions are the controllers 0, 1, 2, 4, 5, 6, 11, 12, 13, 14 and 15). This workaround opens a new way to send controllers to the FM7; FM7 presents 64 parameters to Logic. If you automate these parameters (using the conventional way in Logic - either by choosing MIDI controllers between 65 and 119 or the parameter list of the plug-in), MIDI controllers between 65 and 119 are generated in FM7 on MIDI channel 1.

The MIDI controller number in Logic depends on the position of the plug-in in the mixer bus. If FM7 is the upmost plug-in (what normally is the case when you use FM7as VST instrument) then the Logic MIDI controllers are equal to FM7 (so if you send contoller 90 to the FM7 plug-in, FM7 also receives this as controller 90). If you use FM7 as a VST effect plug-in and another insert effect is used before the FM7plug-in, the controller numbers change in the amount of 16 at a time per effect inserted before FM7. So if you use one insert effect before FM7, then Logic controller number 90 becomes 74 in FM7. If there is any plug-in behind the FM7 plug-in, then FM7 has only 16 controllers at its disposal. If you need more, the signal should be routed to an audio bus, where you can insert your other plug-ins without affecting the number of available controllers for FM7.

Envelope Slope

This Default Value is used for the slope of every new segment created in the Envelope editor.

Randomize

The Randomize function on the Lib page applies random changes to groups of parameters. In several cases a slight change can cause a dramatic change in the sound. Therefore in the *Cautious* mode not all parameters are affected by the Randomize function; e. g. in the FM Matrix no new connections will be created or in the Envelopes no breakpoints will be deleted or created. In the *Aggressive* mode there are no such restrictions.

Author

The name that you enter here is stored with all your presets, unless you edit the Author in the LIB page.

Dump Controllers

Clicking on this button sends the settings of all parameters that are assigned to a MIDI controller. This can be useful to initialize an external MIDI device.

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