Reference Manual

ASR

SAMPLER RESAMPLER

ENSONIQ™

LEADING THE WORLD IN SOUND INNOVATION

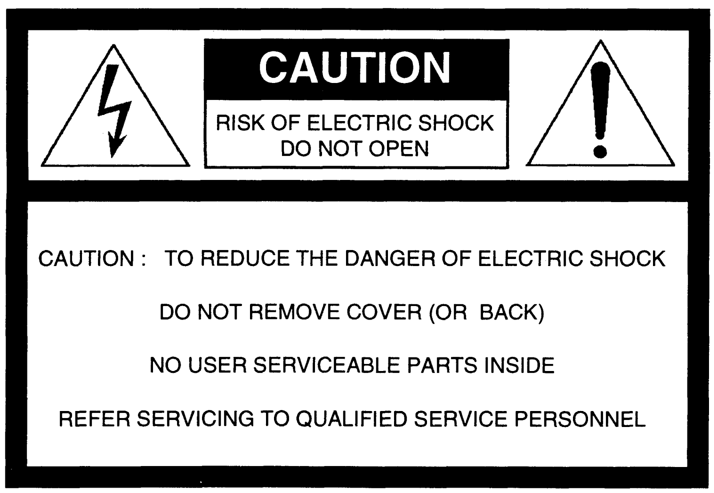
READ THIS FIRST!

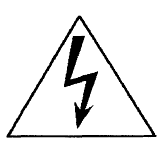
WARNING!!

Grounding Instructions

This product must be grounded. If it should malfunction or break down, grounding provides a path of least resistance for electric current to reduce the risk of electric shock. This product is equipped with a cord having an equipment-grounding conductor and a grounding plug. The plug must be plugged into an appropriate outlet that is properly installed and grounded in accordance with all local codes and ordinances.

DANGER: Improper connection of the equipment-grounding conductor can result in the risk of electric shock. Check with a qualified electrician or service personnel if you are in doubt as to whether the product is properly grounded. Do not modify the plug provided with this product — if it will not fit the outlet, have a proper outlet installed by a qualified electrician.





This symbol is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electronic shock to persons.

This symbol is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

SEE IMPORTANT SAFETY INSTRUCTIONS ON BACK COVER!

ENSONIQ™

ASR X

Sampler Resampler

Reference Manual

Version 1.12

ASR-X Reference Manual

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Printed in U.S.A.

Please record the following information:

Your Authorized ENSONIQ Dealer:__________________________ Phone:______________

Your Dealer Sales Representative:____

Serial Number of Unit:_________________________ Date of Purchase:_______________

Your Authorized ENSONIQ Dealer is your primary source for service and support. The above information will be helpful in communicating with your Authorized ENSONIQ Dealer, and provide necessary information should you need to contact ENSONIQ Customer Service. If you have any questions concerning the use of this unit, please contact your Authorized ENSONIQ Dealer first. For additional technical support, or to find the name of the nearest Authorized ENSONIQ Repair Station, call ENSONIQ Customer Service at (610) 647-3930 Monday through Friday 9:30 AM to 12:15 PM and 1:15 PM to 6:30 PM Eastern Time. Between 1:15 PM and 5:00 PM we experience our heaviest call load. During these times, there may be delays in answering your call.

Although every effort has been made to ensure the accuracy of the text and illustrations in this manual, no guarantee is made or implied in this regard.

IMPORTANT:
Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

* Reorient or relocate the receiving antenna.
* Increase the separation between the equipment and receiver.
* Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
* Consult the dealer or an experienced radio/TV technician for help.

Changes or modifications to the product not expressly approved by ENSONIQ could void the user's FCC authority to operate the equipment.

In order to fulfill warranty requirements, your ASR-X should be serviced only by an Authorized ENSONIQ Repair Station. The ENSONIQ serial number label must appear on the outside of the unit, or the ENSONIQ warranty is void.

Part Number 931 0 0215 01-C **Model Number MM-156**

Temperature Guidelines

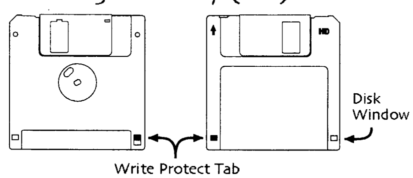
The ASR-X contains a substantial amount of computerized and electronic circuitry that can be susceptible to damage when exposed to extreme temperature changes. When the ASR-X is brought inside after sitting in a cold climate (i.e., the back seat of your car), condensation builds up on the internal circuitry in much the same way a pair of glasses fogs up when you come inside on a cold day. If the unit is powered up as this condensation occurs, components can short out or be damaged. Excessively high temperatures also pose a threat to the unit, stressing both the internal circuits as well as the case. With this in mind, it is highly advisable to follow these precautions when storing and setting up your ASR-X:

- Avoid leaving the ASR-X in temperatures of less than 50 degrees Fahrenheit or more than 100 degrees Fahrenheit.
- When bringing the ASR-X indoors after travel, allow the unit at least 20 minutes to reach room temperature before powering up. In the case of excessive outdoor temperatures (below 50 degrees Fahrenheit or above 100 degrees Fahrenheit), allow an hour or more before power up.
- Avoid leaving the ASR-X inside a vehicle exposed to direct sunlight.

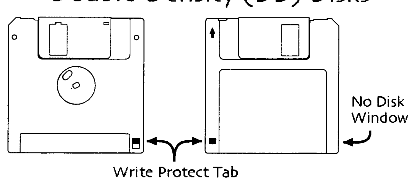
Care and Feeding of the Disk Drive

The ASR-X's disk drive is used to store sounds, rhythms, and sequencer data. This quad-density disk drive will store your data on a high-density (HD) 3.5" micro floppy disk. You can also store data on a DOS-formatted double-density (DD) 3.5" micro floppy disk.

High-Density (HD) Disks



Double-Density (DD) Disks



Disks have a sliding write-protection tab so that you can protect your data against accidental erasure. When the write-protection tab covers the protect window, you can store information on the disk. Sliding the tab so that the window is open will protect the disk against being accidentally reformatted or having files deleted. High density disks can be easily identified because they have an additional disk window located on the lower right corner of the disk.

Floppy disks are a magnetic storage medium, and should be treated with the same care you'd give important audio tapes. Just as you would use high quality audio tapes for your important recording needs, we recommend using high quality floppy disks for your ASR-X. Here are a few Do's and Don't's concerning disks and the disk drive.

Do's:

- Use either high-density (HD) or double-density (DD) 3.5" disks. Both types are available from most computer stores.
- Keep your disks and the disk drive clean and free of dust, dirt, liquids, etc.
- Label your disks and keep a record of what is saved on each.

Don't's:

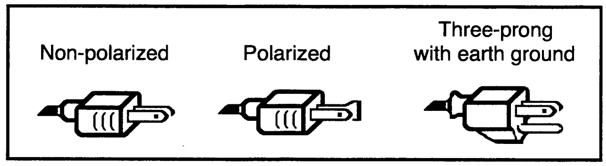
- Don't use single-sided (SD) disks. These disks have not passed testing on both sides. While a single-sided disk might work with the ASR-X, it is possible that you will eventually lose important data to a disk error if you try using single-sided disks.
- Don't put anything other than a disk into the disk drive.
- Don't transport the unit with a disk in the drive.
- Don't expose disks to temperature extremes. Temperatures below 50° F and above 140° F can damage the plastic outer shell.
- Don't expose your disks to moisture.
- Don't dry your disks in a microwave oven.
- Don't subject disks to strong magnetic fields. Exposure to magnetic energy can permanently damage the information on the disk. Keep disks away from speaker cabinets, tape decks, power cables, airline x-ray equipment, power amplifiers, TV sets, and any other sources of magnetic energy.
- Don't eject the disk while the drive is operating (i.e., when the disk drive light is on).

Clean Up and Maintenance

Clean the exterior of your ASR-X with a soft, lint-free, dry (or slightly damp) cloth. You can use a slightly dampened cloth (with a mild neutral detergent) to remove stubborn dirt, but make sure that the ASR-X is thoroughly dry before turning on the power. Never use alcohol, benzene, volatile cleaners, solvents, abrasives, polish or rubbing compounds.

Polarization and Grounding

Like many modern electrical devices, your ENSONIQ product has a three-prong power cord with earth ground to ensure safe operation. Some products have power cords with only two prongs and no earth ground. To ensure safe operation, modern products with two-prong power cords have polarized plugs which can only be inserted into an outlet the proper way.



Some products, such as older guitar amplifiers, do not have polarized plugs and can be connected to an outlet incorrectly. This may result in dangerous high voltages on the audio connections, which could cause you physical harm or damage any properly grounded equipment to which they are connected, such as your ENSONIQ product.

To avoid shock hazards or equipment damage, we recommend the following precautions:

- If you own equipment with two-pronged power cords, check to see if they are polarized or non-polarized. You might consider having an authorized repair station change any non-polarized plugs on your equipment to polarized plugs to avoid future problems.
- Exercise caution when using extension cords or plug adapters. Proper polarization should always be maintained from the outlet to the plug. The use of polarized extension cords and adapters is the easiest way to maintain proper polarity.
- Whenever possible, connect all products with grounded power cords to the same outlet ground. This will ensure a common ground level to prevent equipment damage and minimize hum in the audio output.

AC outlet testers are available from many electronic supply and hardware stores. These can be used to check for proper polarity of outlets and cords.

AC Line Conditioning

As with any computer device, the ASR-X is sensitive to sharp peaks and drops in the AC line voltage. Lightning strikes, power drops, or sudden and erratic surges in the AC line voltage can scramble the internal memory, and in some cases, damage the unit's hardware. Here are a few suggestions to help guard against such occurrences:

- A surge/spike suppressor. A surge/spike suppresser absorbs surges and protects your gear from all but the most severe over-voltage conditions. You can get multi-outlet power strips with built-in surge/spike suppressers for little more than the cost of unprotected power strips, so using one is a good investment for all your electronic equipment.
- A line conditioner. This is the best, but by far the more expensive way to protect your gear. In addition to protecting against surges and spikes, a line conditioner guards the equipment against excessively high or low line voltages. If you use the ASR-X in lots of different locations with varying or unknown AC line conditions, you might consider investing in a line conditioner.

Table of Contents

1 Welcome

The ASR-X Experience Begins

Congratulations on your purchase of the ENSONIQ ASR-X Advanced Sampler/Resampler. The ASR-X is the ultimate groove machine—ideal for deejays and musicians who enjoy grabbing sounds out of the air and turning them into mind-boggling loops.

This book—the ASR-X Reference Manual—contains detailed information on all of the ASR-X's many features. If you'd like to start getting to know your ASR-X through a hands-on tour of its hot spots, take a look at the ASR-X User's Guide, which contains step-by-step quick-starts for the major features of the ASR-X.

For the latest information on the ASR-X and other ENSONIQ products, visit ENSONIQ's World Wide Web site at http://www.ensoniq.com.

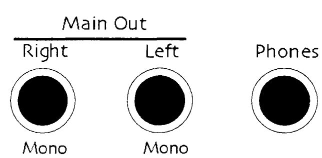
What Else is in the ASR-X Box?

The following items are included with every ASR-X shipped from the ENSONIQ factory:

- ENSONIQ X-Audio Sampling CD Volume 1—Producers' Mix
- FDX-100 sound and demo floppy disk
- AC power cable
- ENSONIQ ASR-X User's Guide
- ENSONIQ ASR-X Reference Manual
- hex wrench

Setting Up the ASR-X

Using the Audio Outputs



As shipped from ENSONIQ, the ASR-X provides two ways to listen to the sounds it makes:

- You can listen to the ASR-X using headphones by plugging your headphones into the 1/4" Phones jack on the ASR-X rear panel.
- Using 1/4" audio cables, you can connect the Left and Right Main Outs to a mixer or amplifier. The ASR-X outputs produce a great-sounding stereo image. If you'd prefer to use the ASR-X in mono, connect only the Left or Right Main Out jack to your mixer or amplifier, and make sure nothing is plugged into the other Main Out jack.

Warning: You can use 1/4" to RCA-type adapters to connect the ASR-X outputs to a home stereo, but do so with care, since the dynamic range of the ASR-X is much greater than that of a CD or record, and could damage your speakers. See "Setting the Output volume" below.

Aux Out 1, Aux Out 2, Aux Out 3, Aux Out 4

These four pairs of stereo outputs become available for use with the purchase and installation of an ENSONIQ X-8 output expander board. You can connect them to a mixer, amplifier or stereo system.

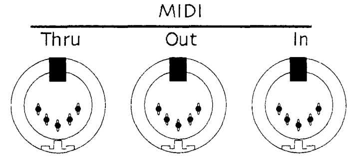
Setting the Output Volume

The ASR-X, like all digital equipment, produces its best fidelity when its front panel Volume knob is turned all the way up.—when using the ASR-X with a mixer or amplifier, use the input level controls on the mixer or amplifier to find a usable volume setting for the ASR-X. When the ASR-X, is connected to a home stereo, turn the ASR-X Volume knob all the way down, power up (see below), and, while playing its pads with maximum force, slowly turn up the ASR-X Volume knob to find a level that sounds good but doesn't cause the stereo's inputs to distort.

The Audio Inputs

The two Audio Input jacks on the rear panel of the ASR-X allow you to sample audio from a microphone or line-level audio source—such as a CD player or phonograph—connected to your ASR-X. The use of the Audio Inputs is described in Chapter 5.

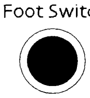
Making MIDI Connections



The ASR-X is a very MIDI-savvy machine. The rear panel MIDI jacks are:

- MIDI Thru—This jack is used when the ASR-X is part of a MIDI daisy-chain, with multiple MIDI devices connected in a row. Any MIDI data that the ASR-X receives will be passed along from this jack to the MIDI In of the next device in the series.
- MIDI Out—The ASR-X sends MIDI data out of this jack. Since the ASR-X can play external MIDI sounds from its pads or sequencer tracks, or provide a timing reference for an external sequencer, you can connect this jack to the MIDI In of a MIDI sound module, sequencer, or MIDI patchbay.
- MIDI In—The ASR-X responds to MIDI data sent through a MIDI cable connected to this jack. connect the cable's other end to the MIDI Out of an external MIDI controller, sequencer, storage device or MIDI patchbay. When the ASR-X is receiving MIDI data, its front-panel MIDI LED flashes.

Using a Foot Switch with the ASR-X



Connecting a foot switch to the ASR-X's rear-panel Dual Foot Switch jack allows you to use a foot switch for conventional purposes—as a sustain pedal, for example—or for performing certain operations hands-free. Chapter 7 describes the many possible uses of a foot switch with the ASR-X. The ASR-X can accommodate a dual foot switch—such as ENSONIQ's SW-10—or a single foot switch, such as ENSONIQ's SW-2 or SW-6.

Powering the ASR-X

Connect one end of the supplied AC cable to the ASR-X's line jack—located next to the On/Off switch on the rear panel—and the other end to a grounded AC outlet. The ASR-X works with all standard voltages.

Turning On the ASR-X

When powering up the ASR-X, as with any audio gear, turn down your monitoring system to avoid any unwanted level spikes. To turn on the ASR-X, press in the top of the rear-panel On switch.

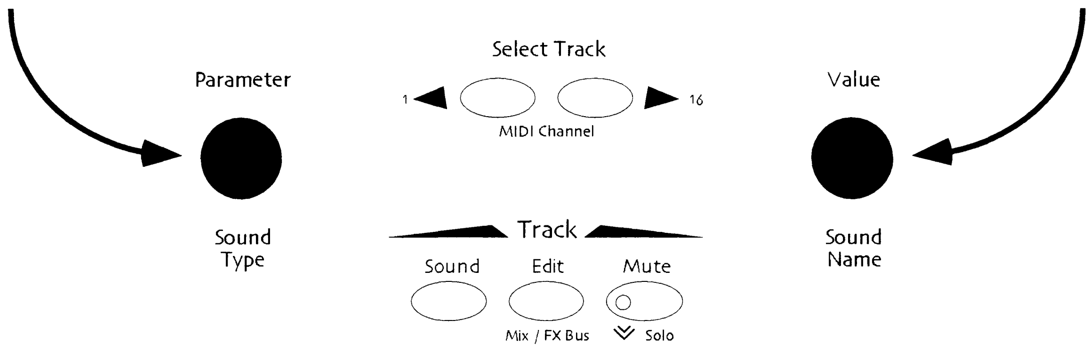
The ASR-X Controls

Each section of the ASR-X provides controls for performing tasks pertaining to that section. In addition, there are a set of common controls and indicators that you'll find yourself using again and again.

The ASR-X Display

The display located in the center of the ASR-X front panel is your doorway to all of the ASR-X's workings. Information relating to everything you do is presented on this display. Each chapter in this manual describes what you'll see while using your ASR-X—and what it all means.

The Knobs



In the center of the front panel, below the display are two knobs central to most every ASR-X activity. These knobs each have two names, since they operate in two wide-ranging contexts.

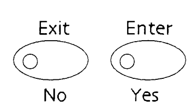
- When you're selecting sounds for tracks or for pads in the ASR-X, the central knobs are referred to as the Sound Type and Sound Name knobs. These names are printed underneath the knobs, as shown above. The Sound Type and Sound Name knobs are the key to unlocking the power of the ASR-X's SoundFinder feature. Each knob does just what its name suggests: the Sound Type knob selects a kind of sound, and the Sound Name selects an individual sound.

Tip: To learn more about selecting sounds and SoundFinder, see Chapter 2.

- Virtually every other ASR-X activity also uses the two central knobs. In these contexts, they're referred to as the Parameter knob and the Value knob. These names are printed above the knobs, as shown in the above illustration.
  - A *parameter* is a characteristic of the ASR-X software that can be changed.
  - A *value* is the setting of a parameter.

The Parameter knob is used for the selection of a parameter for editing, and the Value knob changes its value. There are times that the knobs are used for selecting procedures to be performed—in these cases, the knobs are referred to as the Parameter and Value knobs.

The Exit/No and Enter/Yes Buttons and Their LEDs



The two buttons marked "Exit/No" and "Enter/Yes" are central to the performance of many ASR-X procedures, and are used for navigating the ASR-X displays and parameters.

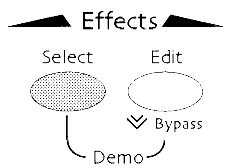
Most of the operations performed on the ASR-X are posed as questions on the ASR-X display—at such a time, think of these buttons as No and Yes buttons. When the ASR-X asks you a question, the LEDs in the buttons flash as a reminder that the ASR-X requires a response from you in order to proceed with the selected operation. To answer "No" or "Yes," press the appropriate button.

Some of the ASR-X's features offer sets of parameters and procedures. In such cases, you'll begin by answering "Yes" to a top-level question, which will have the effect of beginning the procedure. From there you'll encounter parameters related to the top-level question presented on sub-displays. To exit back out to the top level of the ASR-X, you can press the Exit/No button.

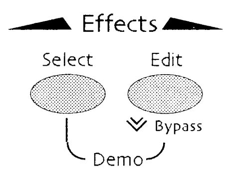
Playing the ASR-X Demo

The ASR-X contains a short demonstration piece that can give you an idea of the kind of music it can produce. This demo is based on the wave data built into the ASR-X. To play the demo:

1. Locate the Effects section on the ASR-X front panel.
2. Hold down the Select Effect button.



3. While still holding the button down, press the Edit Effect button.



4. Release both buttons.
   The display will show...

```
Start demo playback?
MAINDEMO:  Gizmo Bop
```

5. Press the Enter/Yes button to hear the demo.
6. To stop the demo, press any button on the ASR-X front panel.

Important ASR-X Concepts

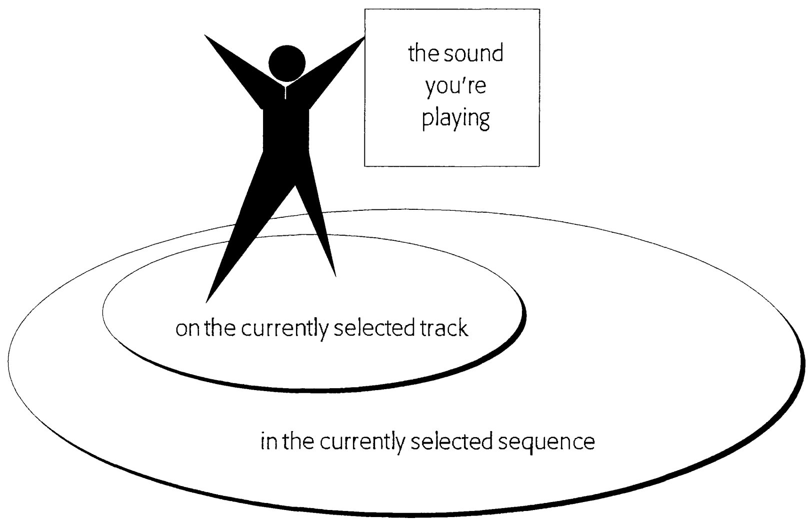
Architecture Overview

The ASR-X is a groovestation containing a variety of elements, each represented by an area (or two, in the case of the pads) of the ASR-X front panel:

- tracks
- pads (and pad editing)
- effects
- sampling/resampling
- sequencer
- disk functions and global settings

While each area has its own chapter in this manual describing it in detail, it's important that you always know where you are and what you're doing.

You Are Here



- In the ASR-X, there is always a sequence selected, even if you haven't recorded anything.
- In the ASR-X, there is always a track selected, even if you haven't recorded anything.

Therefore, the important thing to remember about the architecture of the ASR-X is this simple rule:

> You're always on the currently selected track in the currently selected sequence.

This means that:

- When you press the Track Sound button and pick a new sound, you're choosing a new sound for the currently selected track (described in Chapter 2).
- When you play the pads, you're playing the sound on the currently selected track (see Chapter 3).
- When you convert a standard sound into a RAM kit, the newly created RAM kit is assigned to the currently selected track (described in Chapter 3).
- When you select new sounds or otherwise edit or process what's on a pad, you're editing one of the pads in the RAM kit on the currently selected track (described in Chapter 3).
- When you sample or resample and send your wave(s) to one or more pads, you're sending them to a RAM kit on the currently selected track (described in Chapter 5).
- When you play the pads and record in the sequencer, you're recording on the currently selected track (described in Chapter 6).

What's Where

The ASR-X contains essentially two types of memory:

1. ROM (for "Read-Only Memory")—This is a permanent and unchangeable area of memory that contains the wave data used by the sounds shipped with your ASR-X; it also contains those sounds.
2. RAM (for "Random Access Memory")—This area of memory that holds:
   - the contents of the Scratch Pad
   - the sounds that play your samples
   - sequences
   - waves you've sampled and sent to pads
   - RAM kits you've created and edited
   - System / MIDI settings

Note: RAM memory is fast, efficient memory; it's also volatile, which means that everything you do will be stored in RAM only until you turn off the ASR-X, at which time RAM is cleared. While this offers you a clean slate each time you turn on the ASR-X, it also means that it's important to remember to save your work to disk before powering down.

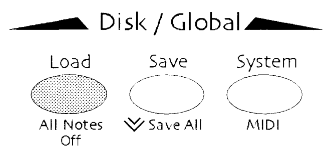
The "Allocating memory" Message

From time to time you may see "Allocating memory" briefly appear on the ASR-X display. This is completely normal—it means that the ASR-X is defragmenting its RAM memory to make most efficient use of available space. Defragmenting takes bits and pieces of free RAM joins them into uninterrupted, larger chunks of memory. This helps the data that you store there play back quickly and smoothly.

How Many Notes Can be Played at Once

The ASR-X supports 32-voice polyphony, which means that 32 sound layers can be playing at any given moment. Different sounds use different numbers of layers—sounds based on the samples you create use one or two per note, while ROM sounds may use up to 16 per note—so the number of notes that can be played simultaneously depends very much on the sounds being used. To learn more about sound layers, see Chapter 3.

The All Notes Off Button



It's not uncommon for MIDI devices to get momentarily confused, given the amount of MIDI data that courses through the cables in the average MIDI studio, and the ASR-X is no exception. The Disk/Global Load button doubles as a handy All Notes Off button. If notes in the ASR-X continue playing when you feel they should stop, double-click this button to turn off all of the currently sounding notes.

About Note, Tips and Warnings in the ASR-X Documentation

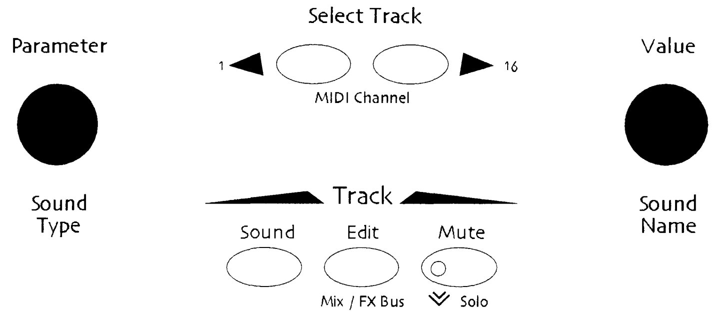
Throughout the ASR-X Reference Manual and User's Guide, you'll encounter notes, tips and warnings, offset from the rest of the text by borders, and always beginning with the word "Note," "Tip" or "Warning" in bold type. Each of these has its own meaning:

- Note—A note is information regarding the topic being discussed that should not be overlooked.
- Tip—A tip points out a use for the feature being discussed that might not be obvious, but that's worth being aware of.
- Warning—A warning states information that's vital in preventing damage to the ASR-X, other equipment, or you.

Optional ENSONIQ Accessories for the ASR-X

- X-8 output expander—The X-8 output expander provides additional outputs for your ASR-X, usable as four pairs of stereo outputs, or as eight mono outputs.
- SP-5 SCSI interface—The SP-5 allows you to save and load files to and from a SCSI disk drive.
- SW-10 foot switch—The SW-10 provides two foot pedals mounted on a single base that let you take full advantage of the ASR-X many hands-free possibilities.
- SW-2 foot switch—The SW-2 synthesizer-style single foot switch can access the many possibilities available to a foot switch on the ASR-X.
- SW-6 foot switch—The SW-2 piano-style single foot switch can access the many possibilities available to a foot switch on the ASR-X.
- X-Audio audio CDs—Each X-Audio series CD contains a huge assortment of materials that can be sampled into the ASR-X.
- EXP Expansion BoardsEXP Series Wave Expansion Boards—These exciting boards from ENSONIQ provide the ASR-X with new sounds and ROM wave data.

2 Tracks



Introduction to Tracks

Whatever you do in the ASR-X—and whatever kind of sound you play from its pads or via MIDI—you're always on a track in a sequence, even when you haven't yet recorded any notes. When you choose and listen to the sounds built-in to your ASR-X, or that you've sampled/resampled yourself, you're actually choosing sounds for the currently selected track. Tracks are absolutely central to life with an ASR-X—that's why the Track buttons are where they are.

Each track has:

- a sound that can be played using the ASR-X pads or from an external controller via MIDI.
- a set of parameters that determine how the sound will behave while it's assigned to the track.
- a mute/solo capability that can silence the track or isolate by turning all other tracks off.
- its own MIDI channel for receiving and transmitting MIDI data. Each track's MIDI channel is the same as its track number.

There are 16 tracks in each sequence that can be used for recording or as a 16-part multi-timbral MIDI sound module. The ASR-X transmits MIDI data whenever its pads or sequencer are played.

These topics are discussed in detail in this chapter.

To Select One of the Tracks in the Currently Selected Sequence

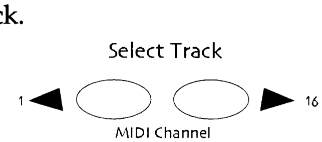
The track display tells you which track is currently selected. To view this display, press the Track Sound button.

The currently selected track

```
T01          ROM08:000
PERCSOLO: Thump Kick
```

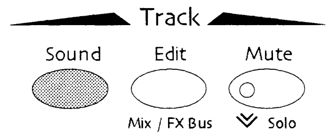
To Select a Track

1. Press the Select Track right arrow button select a higher-numbered track, or the left arrow button to select a lower-numbered track.



Tip: Hold down either button to scroll through the tracks.

Selecting a Sound for a Track



There are four ways to place a sound on the currently selected track, all of which begin with pressing the Track Sound button:

- You can select a track's sound using the front-panel Sound Type and Sound Name knobs, taking advantage of the ASR-X powerful SoundFinder feature—see "Selecting a Track's Sound Using the Sound Type and Sound Name Knobs" below.
- You can select a track's sound by sending MIDI Bank Select and Program Change messages to the ASR-X on the track's MIDI channel—see "Selecting a Track's Sound Via MIDI" later in this section.
- You can transform any ROM sound on the track into a new, editable RAM drum kit by sampling or resampling and sending your sample to a pad (see Chapter 5).
- You can transform any ROM sound on the track into a new, editable RAM drum kit by editing the sound using the Pad editing controls (see Chapter 3).

When a new sound is selected for a track, the ASR-X will automatically reset certain track parameters if the System/MIDI Track ParamReset parameter is set to "On" (see Chapter 7). For a list of these parameters, see "Track ParamReset Behavior" in Chapter 9.

Banks and Sounds

Sounds are stored in the ASR-X in groups called *banks.* A bank can contain up to 127 sounds. Each bank has a corresponding MIDI Bank Select number that allows it to be selected via MIDI, and within each bank, each sound has a program number corresponding to a MIDI Program Change value so that it, too, can be selected via MIDI (see "Selecting and Playing a Track's Sound Via MIDI" later in this chapter).

Selecting a Track's Sound Using the Sound Type and Sound Name Knobs

Sound selection using the ASR-X front-panel knobs is simple. The ASR-X utilizes ENSONIQ's acclaimed SoundFinder™ technology to make the location and selection of sounds logical and easy.

SoundFinder

SoundFinder is a database of all the sounds in your ASR-X. The power of a database lies in its ability to let you to view information in a manner of your choosing. SoundFinder keeps a list of all the sounds available in your ASR-X, and shows them to you in musically convenient categories called *sound types.*

SoundFinder sound types show you sounds by instrument family—vocals or bells, for example—or by a number of other useful criteria, including the location in the ASR-X's memory where they reside. The ALL-SND category is especially useful,: it shows all of the ASR-X sounds arranged in alphabetical order.

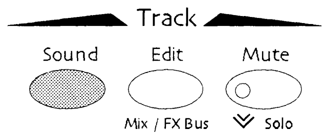
Tip: When you select a sound from a SoundFinder category, the ASR-X remembers the sound you've selected, and offers it to you as a first choice when you return to the category.

While most of SoundFinder's categories describe types of musical instruments, there are three additional categories that allow you to select sounds based on the location in memory in which they reside:

- EXP-SND—This category contains sounds located on an EXP Series Wave Expansion Board.
- ROM-SND—This category contains sounds stored in permanent ROM.
- RAM-SND—This category contains sounds stored in temporary RAM. There are two banks' worth of RAM sound memory (to learn about banks, see "Banks and Sounds" above).

To Select a Sound Using the Sound Type and Sound Name Knobs

1. Press the Track Sound button.



The display shows you information that's helpful when picking sounds:

The track being edited   Where the sound it resides   The sound's MIDI bank and program numbers

```
T:01         ROM08:000
PERCSOLO: Thump Kick
```

The currently selected SoundFinder category   The currently selected sound

2. Turn the Sound Type knob to select a different SoundFinder category, if desired.

Tip: To quickly locate all RAM kits, turn the Sound Type knob all the way to the left (to the USER-SND category). To find the sounds that play your waves, turn it all the way right (*CUSTOM).

---

3. Turn the Sound Name knob to select a new sound.

Note: When you select a new sound for a track, corresponding Bank Select and Program Change values are transmitted via MIDI on the track's MIDI channel.

Selecting and Playing a Track's Sound Via MIDI

You can select sounds for tracks by sending the ASR-X MIDI Bank Select LSB and Program Change values on each track's MIDI channel (the MIDI channel corresponding to its track number). Sounds can also be played from MIDI on each track's MIDI channel.

Note: Each track *always* receives and responds to MIDI data received on its MIDI channel, regardless of which track is currently selected.

Note: In order for the ASR-X to respond to Bank Select and Program Change messages, its System/MIDI Bank&ProgChgRecv parameter must be set to "On" (see Chapter 7 for more information). In addition, the target track's ProgramChngeRecv and Bank Select Recv parameters must also be set to "On" (these two parameters are described later in this chapter).

The track sound selection display shows you the Bank Select LSB and Program Change values for the displayed sound. You can program these Bank Select and Program Change values into an external MIDI device in order to select the sounds via MIDI later on.

The sound's Bank Select LSB value   The sound's Program Change value

```
T:01         ROM09:012
BASS-SYN:Barkin'Bass
```

To Select and Play a Track's Sound Via MIDI

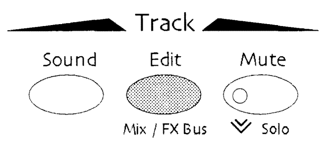
1. Set a MIDI device to transmit on the channel of the track whose sound you want to select or play.
2. Send the appropriate Bank Select and Program Change values to the ASR-X.
3. Send note and controller data from your external device to play the track's sound.

Editing Track Parameters

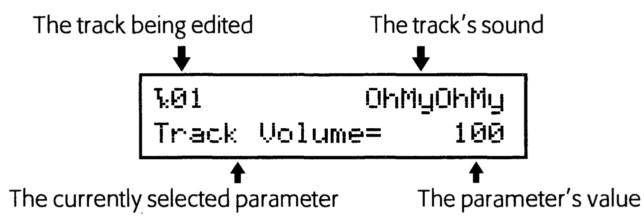
Editing a track's parameters—and therefore determining the behavior of its sound—involves the same technique regardless of the parameter being edited.

To Edit a Track Parameter

1. Press the Track Edit/Mix/FX Bus button in the Track section of the ASR-X front panel.



2. Turn the Parameter knob to select the track parameter you'd like to edit.
   All of the track parameter displays show the track number and currently selected sound on the top line, and the selected parameter and its value on the bottom line:



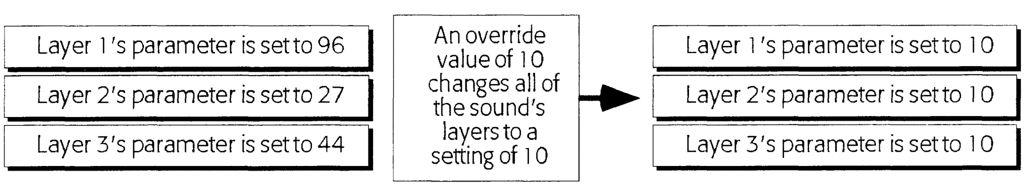
3. Turn the Value knob to change the setting of the selected parameter .

Overrides and Offsets

ASR-X sounds are made up of layers of waves. Track parameters allow you to easily change the settings in all of a sound's layers at once by altering them in one of two ways. Each track parameter is either:

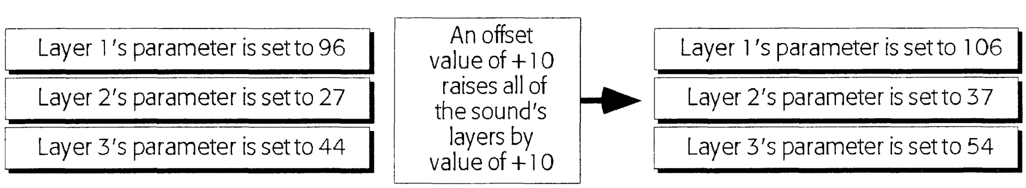
- an override, which sets all of the layers in the currently selected sound to the absolute value you set.
- an offset, which raises or lower the programmed values by the amount you set.

Overrides set all of a sound's layers to the same absolute value for the selected parameter.



When an override parameter is set to "Prog," the originally programmed setting for each layer is retained.

Offsets are typically used to affect all of a sound's layers at once, retaining their different settings for the selected parameter in relation to one other. Offset parameters offer values that have positive/negative aspects (shown with a "+" or "-"). When an offset is set to "0," the originally programmed value for each layer is in effect.



Note: Offsets adjust layer parameters only within the parameters' legitimate ranges—they can't force them beyond those limits. If a track offset parameter appears to be having no effect, it's likely that the setting for the parameter in the sound's layers has already reached its maximum or minimum setting.

Editing Track Parameters Via MIDI

Track parameters can be edited via MIDI in two ways. Some of the parameters, such as Track Volume, Mix (Expression) and Pan correspond to standard MIDI sound controllers, and can be adjusted by sending the ASR-X values for the relevant controllers. In addition, most of the track parameters can be edited using special registered and non-registered MIDI parameters (RPNs and NRPNs). To learn more about RPNs and NRPNs, see Chapter 9.

What Each Track Parameter Does

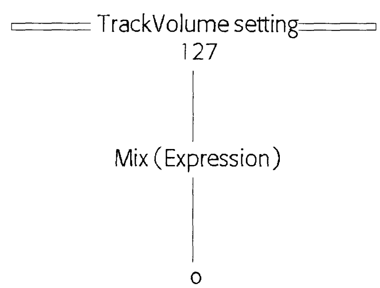
Track Volume

The Track Volume parameter allows you to override the loudness ceiling programmed into the selected track's sound. A Volume setting of 127 will leave the sound's level set as it was programmed. Lower values will reduce the sound's loudness—down by 96dB at a value of 0.

Track Volume can also be edited via MIDI with controller #7 (Volume) messages.

Mix (Expression)

The Mix (Expression) parameter can raise or lower the level of the sound on the selected track, but only as high as the maximum set by the Track Volume parameter.



You can set an acceptable loudness ceiling for a sound with the Track Volume parameter, and use the Mix (Expression) parameter to adjust its level without worrying that it will ever become too loud.

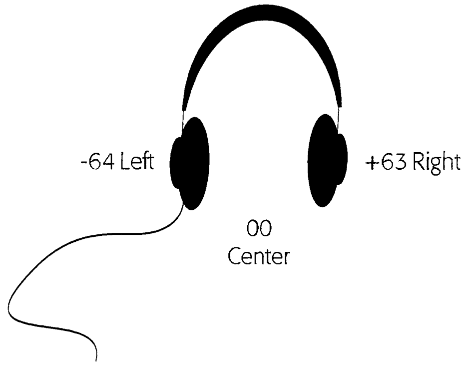
Mix (Expression) can also be edited via MIDI with controller #11 (Expression) messages.

Vol/Mix Polarity

The Vol/MixPolarity parameter reverses the manner in which the selected track's sound will respond to Volume and Mix (Expression) messages received via MIDI. When set to +Pos, the sound will respond normally: higher Volume and Mix (Expression) values will result in greater loudness. When it's set to -Neg, higher Volume and Mix (Expression) values will lower the level of the sound.

Track Pan

ASR-X sounds are programmed to be heard in specific places in the left/right stereo field. By adjusting the Track Pan setting, you can offset the stereo placement of the currently selected track's sound. A value of Center 00 will leave the sound panned as it was programmed. Lower values will shift it to the left, and higher values will move it to the right.



If components within the sound are panned differently, their relative positions will be maintained as the pan value shifts the sound in either direction.

Track Pan can also be edited via MIDI with controller #10 (Pan) messages.

FX Bus

The FX Bus parameter allows you to assign the selected track—and its sound— to the current sequence's insert or global reverb effects, or left un-effected, or "dry." This is accomplished by assigning the track to an FX (for "effect") bus. You can select:

- Prog—to have the sound of each pad use its own FX Bus setting in a kit sound, or to have a standard sound use its Alt Bus setting.
- Insert—to send the selected track's sound to the sequence's insert effect.
- LightReverb—to add a minimal amount of global reverb to the selected track's sound.
- MediumReverb—to add a greater amount of global reverb to the selected track's sound.
- WetReverb—to add the maximum amount of global reverb to the selected track's sound.
- Dry—to leave the selected track's sound un-effected, or "dry."

Note: When an X-8 output expansion board has been installed, an additional four busses become available. These stereo FX busses, AuxOut1, AuxOut2, AuxOut3 and AuxOut4 allow you to send a track's sound directly to the auxiliary outputs. To use the auxiliary busses as separate mono busses, pan the tracks routed to them hard left or hard right (see "Track Pan" above).

The ASR-X Effects are described fully in Chapter 4.

MIDI controller #91 can be used to select one of the reverb busses or the dry bus for any track other than the insert control track. This is accomplished by sending the ASR-X a controller #91 value on the track's MIDI channel. When the track receives a controller #91 value of:

- 0, it will be assigned to the Dry FX bus.
- 1-40, it will be assigned to the LightReverb FX bus.
- 41-80, it will be assigned to the MediumReverb FX bus.
- 81-127, it will be assigned to the WetReverb FX bus.

TrackMIDIOut

Each track in the ASR-X can transmit MIDI data on its corresponding MIDI channel. When a track's TrackMIDIOut parameter is set to "Disable," no MIDI data for the track will be transmitted from the pads or sequencer. When the parameter is set to "Enable," the track's Bank Select, Program Change, note and controller data will be transmitted.

Note: The ASR-X is intelligent about the transmission of Bank Selects and Program Changes—it sends them only if they're different from the last ones that were transmitted from a given track.

Pitch Bend Up and Pitch Bend Down

The Pitch Bend Up and Pitch Bend Down parameters allow you to separately set how you want the selected track's sound to respond to up and down Pitch Bend messages received via MIDI.

Pitch Bend Up and Pitch Bend Down can be set to:

- 1-12dn or 1-12up—to lower or raise the pitch of the selected track's sound by 1 to 12 equal-temper semitones when Pitch Bend up or down messages are received via MIDI.
- Prog—to respond to received up or down Pitch Bend messages according to the programming in the track's sound.
- Sys—to use the global system Pitch Bend Up or Down values (see Chapter 7 for details).
- Off—to ignore received up or down Pitch Bend messages.

Tip: Each track provides a filter—the Pitch Bend Recv parameter—that you can use to disable or enable its response to Pitch Bend messages received via MIDI. This parameter is described later in this chapter.

Octave Shift

The Octave Shift parameter allows you to shift, by octaves, the pitch at which the selected track will play its sound. A setting of 0oct means the sound will play at its programmed octave tuning value. You can tune the sound up or down by a maximum of four equal-temper octaves.

Semitone Shift

The Semitone Shift parameter allows you to shift, in semitone steps, the pitch at which the selected track will play its sound. A setting of 0st means the sound will play at its programmed semitone tuning value. You can shift the sound upward by as much as 63 equal-temper semitones or downward by 64 semitones.

Fine Tuning

The Fine Tuning parameter allows you to re-tune the sound on the selected track by cents. A setting of 0cents means the sound will use its programmed fine tuning value. You can lower or raise the sound's fine tuning by -50 to +49 cents. 100 cents equals one semitone.

PitchTbl

ASR-X contains a variety of non-standard tunings, or pitch tables. The PitchTbl parameter allows the selected track's sound to use one of these special tunings.

Tip: Each track in the ASR-X has its own PitchTbl parameter that determines the pitch table to be used by the sound on the track. By setting each track to a different pitch table, you can program the ASR-X's tracks to produce 16 different tunings at once!

The PitchTbl parameter can be set to:

- Prog—to use the pitch table the sound was originally programmed to use.
- Sys—to use the global system pitch table. (See Chapter 7 to learn about designating a system-wide pitch table.)
- One of the pitch tables built into the ASR-X.

Chapter 9 provides a list of the built-in ASR-X pitch tables.

Tip: With the proper software, you can also design your own pitch table on a computer, and transmit it to the ASR-X via MIDI. "About RAM Pitch Tables" in Chapter 9 provides detailed information on creating your own pitch tables.

Glide Mode

The Glide Mode parameter allows you to set the glide characteristics of the selected track's sound. The parameter can be set to:

- Prog—so that gliding from note to note will occur according to the sound's programming.
- Off—so that no gliding will occur.
- On—so that all of the layers in the sound will glide from note to note.

Note: When this parameter is set to "On"—enabling gliding in the selected track's sound—adjust the Glide Time parameter (described below) to set the speed at which the track's sound will glide from note to note.

If the Glide Mode parameter is set to "Prog" or "Off," the parameter can also be toggled on or off via MIDI by sending MIDI controller #65 (Portamento) values to the ASR-X on the selected track's MIDI channel. Values of 64 or above will turn glide on; values of 63 or lower will turn it off (there is no way to select the Prog setting via MIDI). When controller #65 is used for this purpose, the ASR-X display will not show that the parameter has been reset—it will simply happen. In order to return control of the Glide Mode parameter to the ASR-X's front-panel, a controller #65 value of 63 or less must be sent to the ASR-X on the selected track's MIDI channel.

Glide Time

When a track's sound is programmed to glide from note to note, the Glide Time parameter allows you to adjust the speed at which its notes will glide from one to the next. The parameter can be set anywhere from -64 to +63. A value of 0 means that the sound will glide at its programmed speed. Higher values will slow the sound's glide, and lower values will cause it to speed up.

Delay Offset

The Delay Offset parameter can be used to increase the amount of time it will take for a track's sound to be heard after it receives a key down message, either from a pad or via MIDI. If a sound has been programmed with a delay time, the delay offset will lengthen that delay time by up to 2500 milliseconds (ms). If a sound has no programmed delay time, the Delay Offset parameter can delay it up to 2500ms. If the parameter is set to 0ms, no delay time will be added to the sound.

SyncLFO&Noise

The SyncLFO&Noise parameter allows you to alter the behavior of any LFOs and noise generators in the selected track's sound that are programmed to be synchronized to the ASR-X's sequencer or to incoming MIDI clocks. The parameter can be set to:

- Prog—to allow the synchronized LFOs and noise in the track's sound to behave as programmed.
- Normal—to de-synchronize any synchronized LFOs and noise in the track's sound.
- 1/1 to 1/32T—to set the rhythmic relationship of any synchronized LFOs and noise in the track's sound to the ASR-X's system tempo, or to incoming MIDI clocks. A "T" following a number signifies a triplet value.

Tip: The System/MIDI ClockSource parameter determines whether the ASR-X sequencer or MIDI clocks will control synchronized LFOs or noise. See Chapter 7.

Normal LFO Rates

The Normal LFO Rates parameter allows you to raise or lower the programmed speed of any unsynchronized LFO's in the selected track's sound. The parameter can be set from -64 to +63. A value of 0 means the track's sound will retain its programmed LFO rate. A value other than 0 will be added to or subtracted from the sound's originally programmed rate.

LFO Depth

The LFO Depth parameter allows you to increase or decrease the programmed depth of the LFO's in the selected track's sound. The parameter can be set from -64 to +63. A value of 0 means the track's sound will retain its programmed LFO depth. A value higher than 0 will increase the depth of the sound's LFOs, while values below zero will reduce it.

LFO Delay Time

The LFO Delay Time parameter allows you to lengthen or shorten the delay programmed for any of the LFOs in the selected track's sound. The parameter can be set from -64 to +63. A value of 0 means the track's sound will retain its programmed LFO delay setting. Any value above 0 will lengthen the sound's LFO delay times, while any values below 0 will shorten them.

Amp Env Attack

The Amp Env Attack parameter allows you to lengthen or shorten the attack times of amplitude envelopes within the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the attack times of amplitude envelopes within the track's sound behaving as programmed. Values above 0 will lengthen the attack times, while values below 0 will shorten them.

Amp Env Decay

The Amp Env Decay parameter allows you to lengthen or shorten the decay times of amplitude envelopes within the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the decay times of amplitude envelopes within the track's sound behaving as programmed. Values above 0 will lengthen the decay times, while values below 0 will shorten them.

Amp Env Release

The Amp Env Release parameter allows you to lengthen or shorten the release times of amplitude envelopes within the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the release times of amplitude envelopes within the track's sound behaving as programmed. Values above 0 will lengthen the release times, while values below 0 will shorten them.

Filter Cutoff

The Filter Cutoff parameter allows you to raise or lower the filter cutoff settings programmed into the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the cutoff settings in the track's sound unchanged. Values above 0 will raise the cutoff settings, while values below 0 will lower them.

Filter Resonance

The Filter Resonance parameter allows you to raise or lower the resonance settings programmed into the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the resonance settings in the track's sound unchanged. Values above 0 will increase the resonance settings, while values below 0 will lower them.

Filt Env Attack

The Filt Env Attack parameter allows you to lengthen or shorten the attack times of filter envelopes within the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the attack times of filter envelopes within the track's sound behaving as programmed. Values above 0 will lengthen their attack times, while values below 0 will shorten them.

Filt Env Decay

The Filt Env Decay parameter allows you to lengthen or shorten the decay times of filter envelopes within the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the decay times of filter envelopes within the track's sound behaving as programmed. Values above 0 will lengthen the decay times, while values below 0 will shorten them.

Filt Env Release

The Filt Env Release parameter allows you to lengthen or shorten the release times of filter envelopes within the selected track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will leave the release times of filter envelopes within the track's sound behaving as programmed. Values above 0 will lengthen the release times, while values below 0 will shorten them.

Amp&Filt Env Vel

The Amp&Filt Env Vel parameter allows you to increase or decrease the velocity sensitivity of the amplitude and filter envelopes within the select track's sound. The parameter can be set anywhere from -64 to +63. A value of 0 will not change the responsiveness of the amplitude and filter envelopes in the track's sound. Values above 0 will increase the effect of velocity upon the sound's envelopes, while lower values will decrease its impact.

Key Range Lo, Key Range Hi

The Key Range Lo and Key Range Hi parameters allow you to limit the pitches that the sound on the selected track will play. The Key Rànge Lo parameter sets the lowest note that will play, while the Key Range Hi parameter sets the highest. Either parameter can be set anywhere from A0 to C8. Middle C is C4. (Some MIDI controller manufacturers refer to Middle C as C3—if you're playing the ASR-X from an external MIDI device, check the device's manual.)

Note: A sound's Key Range Lo value should not be set above its Key Range Hi setting, nor should its Key Range Hi value be set below its Key Range Lo setting.

VelocityRange Lo, VelocityRange Hi

The VelocityRange Lo and VelocityRange Hi parameters allow you to set an allowable velocity range for the selected track. When the track receives velocity values from the pads or via MIDI that fall outside of that range, the track's sound won't play. The VelocityRange Lo parameter sets the lowest allowable velocity; the VelocityRange Hi parameter sets the highest. Either parameter can be set from 0 to 127.

Note: A sound's VelocityRange Lo value should not be set above its VelocityRange Hi setting, nor should its VelocityRange Hi value be set below its VelocityRange Lo setting.

VelocityMode

It's not uncommon for different components of ASR-X sounds to be heard only when the pads are struck, or MIDI notes are received, with particular velocities. The Velocity Mode parameter provides a way to alter sounds so you can reliably produce these values and, therefore, the sounds those velocities produce. When the Velocity Mode parameter is set to any value other than Normal, velocities that fall within the sound's velocity window (see the VelocityRange Lo, VelocityRange Hi parameter description above) are automatically converted to the velocity set with the Velocity Mode parameter. The possible settings for the parameter are Normal, and Fix 001 through Fix 127.

PressureMode

The ASR-X responds to channel and polyphonic pressure messages that it receives via MIDI. The PressureMode parameter allows you to determine how the track's sound will respond to MIDI pressure messages. You can set this parameter to:

- Off—so that the track's sound will not respond to keyboard pressure. If pressure has been assigned as an insert effect modulation source, that response to pressure is also disabled.

- Auto—so that the track's sound will respond to whichever type of pressure the ASR-X receives via MIDI.
- Channel—so that the track's sound will only respond to channel pressure.
- Key—so that the track's sound will only respond to key pressure.

ProgramChngeRecv

The ProgramChngeRecv parameter enables or disables the selected track's response to received MIDI Program Change messages. The parameter can be set to "On" or "Off."

Bank Select Recv

The Bank Select Recv parameter enables or disables the selected track's response to received MIDI Bank Select messages. The parameter can be set to "On" or "Off."

Data Entry Recv

The Data Entry Recv parameter enables or disables the selected track's response to received Data Entry (controller #6) messages. The parameter can be set to "On" or "Off."

Pitch Bend Recv

The Pitch Bend Recv parameter enables or disables the selected track's response to received Pitch Bend messages. The parameter can be set to "On" or "Off."

Mod Wheel(1) Recv

The Mod Wheel(1) Recv parameter enables or disables the selected track's response to received Mod Wheel (controller #1) messages. The parameter can be set to "On" or "Off."

FootPedal(4) Recv

The FootPedal(4) Recv parameter enables or disables the selected track's response to received Foot Pedal (controller #4) messages. The parameter can be set to "On" or "Off."

Volume(7) Recv

The Volume(7) Recv parameter enables or disables the selected track's response to received Volume (controller #7) messages. The parameter can be set to "On" or "Off."

Pan(10) Recv

The Pan(10) Recv parameter enables or disables the selected track's response to received Pan (controller #10) messages. The parameter can be set to "On" or "Off."

Expressn(11) Recv

The Expressn(11) Recv parameter enables or disables the selected track's response to received Expression (controller #11) messages. The parameter can be set to "On" or "Off."

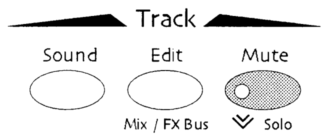
Sustain/SostRecv

The Sustain/SostRecv parameter enables or disables the selected track's response to received Sustain or Sostenuto (controllers #64 and #66, respectively) messages. The parameter can be set to "On" or "Off."

SysCTRL1 Recv, SysCTRL2 Recv, SysCTRL3 Recv, SysCTRL4 Recv,

The SysCtrl1 Recv, SysCtrl2 Recv, SysCtrl3 Recv and SysCtrl4 Recv parameters enable or disable the selected track's response to received MIDI messages for any of the special user-assignable SysCTRLs (see Chapter 7 to learn more about these definable controllers). The parameters can be set to "On" or "Off."

Muting and Soloing a Track



The Mute/Solo button provides an easy way to enable or disable the playback of the tracks in a sequence. You can silence, or *mute*, the selected track—or you can *solo* the track by silencing all of the tracks except the selected track.

Tip: The sequencer will automatically record track mutings and un-mutings if they're performed while the track being muted or un-muted is being recorded.

Muting and Soloing from the Front Panel

- To mute the currently selected track, press the Mute button once—the Mute LED will light, and the word "mute" will appear in the display to show that the selected track has been silenced.

```
T01 mute   ROM08:000
PERCSOLO: Thump Kick
```

- To unmute the currently selected track, press the Mute button once—the Mute LED will turn off and the track will once again be audible.
- To solo the currently selected track, double-click the Mute button—the Mute LED will flash, and the word "solo" will flash in the display.

```
T01 solo   ROM08:000
PERCSOLO: Thump Kick
```

- To un-solo the currently selected track, press the Mute button—the Mute LED will turn off and any tracks that were audible prior to soloing the track will once again be audible.
- To solo groups of tracks—this is called a *group-solo*—select each of the tracks in turn and double-click the Mute button for each track.
- To remove the currently selected track from a group-solo, double-click the Mute button.

The ASR-X solo is intelligent in that it remembers if any tracks in the sequence were already muted prior to soloing, and restores them to that state when the solo is disengaged. When a track is soloed, and any track other than the soloed track is selected, the display will flash the word "mute." Tracks that were already silenced before the solo was engaged will show a non-flashing "mute."

Muting Tracks via MIDI

Tracks can be muted via MIDI by sending a controller #119 message on the channel whose number corresponds to the track you want to mute. The track will respond to a controller #119 value of:

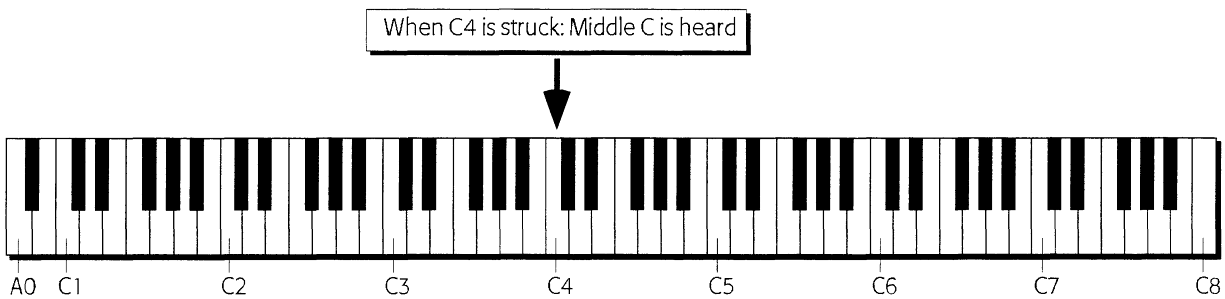
- 127 by muting the track.
- 000 by un-muting the track.
- 064 causes a track that's part of a group-solo to be removed from the solo group.

3 Pads

The Pads: Overview

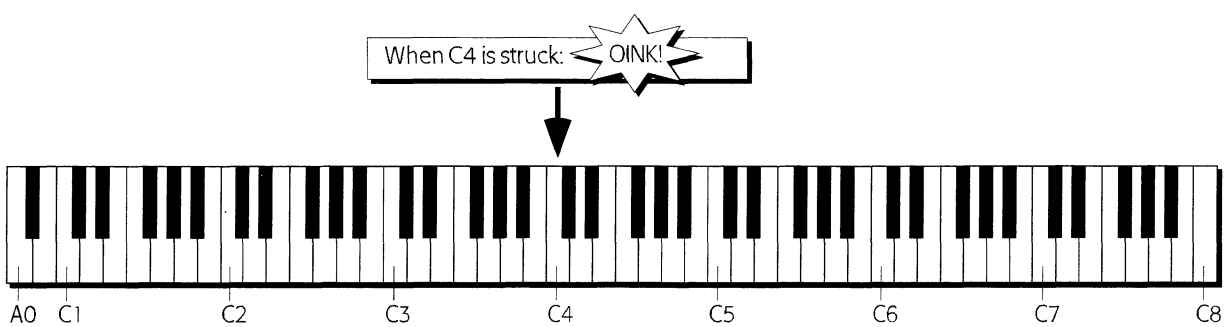
What are the Pads?

All MIDI samplers and MIDI synthesizers—the ASR-X, of course, belongs in both categories—share two fundamental elements: sounds and a way to play them. The most common device used to play sounds is the conventional white-and-black-keys keyboard. Typically, a key on a keyboard will play the note that would be produced by striking the same key on a traditional instrument, such as a piano.



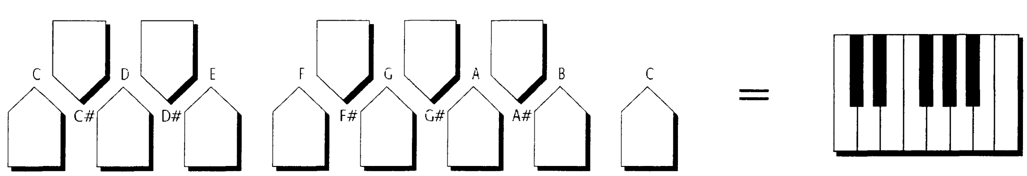
Each semitone is represented by a MIDI note name. The octaves—which begin at each C natural—are numbered, as shown above. The ASR-X can address MIDI notes from A0 to C8.

In the flexible realm of the sampler, however, any sound can be assigned to any MIDI note.



A key on a keyboard connected to a sampler, therefore, is really nothing more than a switch that plays whatever sound is assigned to the corresponding MIDI note. The ASR-X provides pads instead of a piano-style keyboard for this purpose—the ASR-X is a groove machine, and grooves are most fun when banged into being. (You can also play ASR-X sounds via MIDI from any MIDI controller; see Chapter 2.)

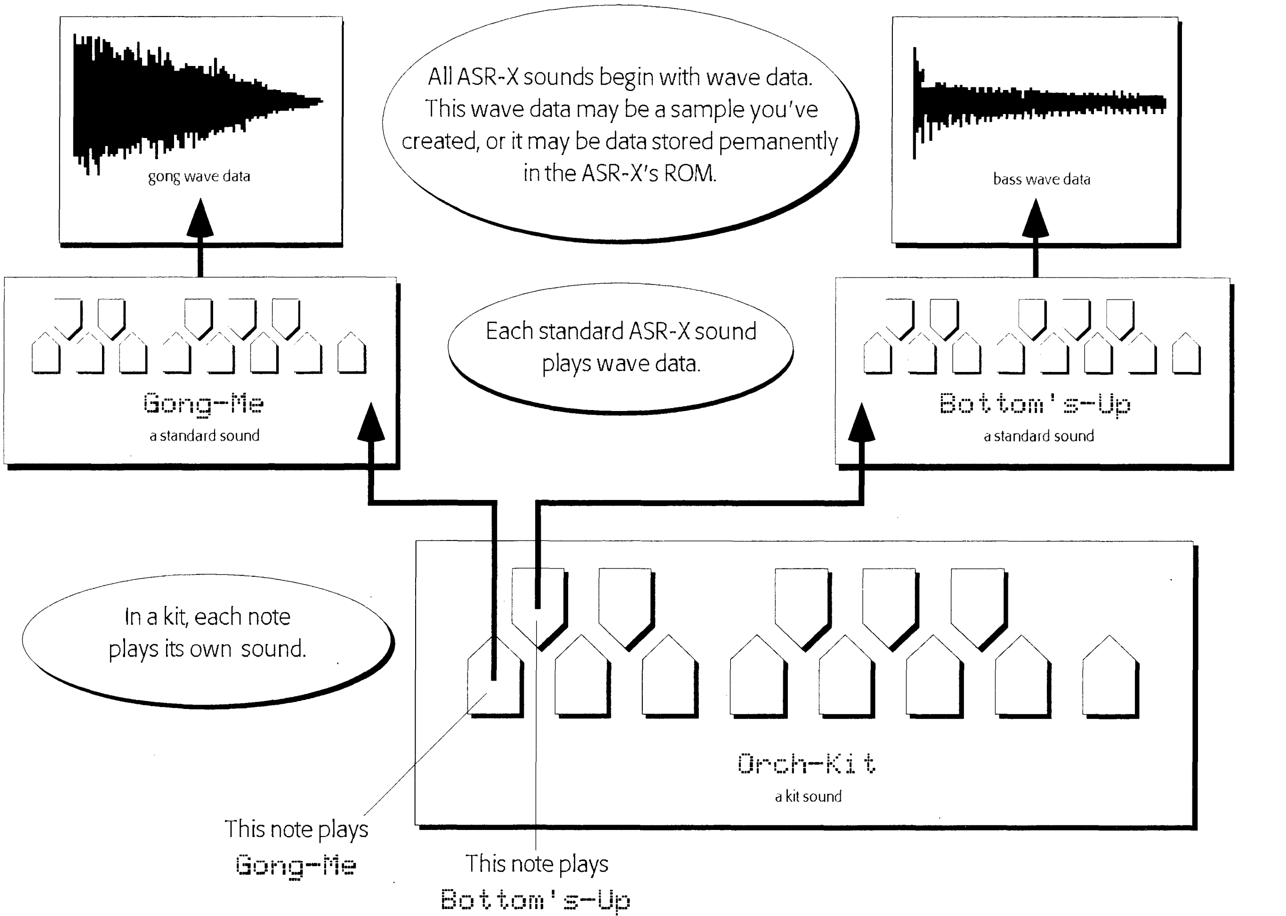
The 13 ASR-X pads trigger 13 adjacent MIDI note numbers, the equivalent of 13 adjacent keys on a piano-style keyboard (unless the Kit Mapper, described later in this chapter, is on). You can use the pads to play single notes or chords.



The pads default to playing the octave beginning at C2, though they can be re-directed up or down to trigger the MIDI note numbers in any octave (see “Octave Transpose Buttons” later in this chapter).

What the Pads Play

The ASR-X provides two major types of sound structures—standard sounds and kit sounds. Precisely what the pads play depends on the structure of the sound assigned to the currently selected track.



Standard Sounds

Standard sounds play digital recordings of audio called *waves*. This can be:

- waves built into your ASR-X ROM.
- waves you've loaded into your ASR-X.
- waves that you've created in the ASR-X.

The waves in standard sounds are arranged in layers comprised of wave data and parameters that shape the data. Some of the ROM standard sounds in your ASR-X are comprised of multiple layers, which may contain groups of related waves in order to accurately reproduce a real-world or synthesized sound. Sounds that play the waves you create on the ASR-X are organized in layers, as well—stereo waves are played by sounds with two layers, mono waves are played by sounds using one layer.

When a standard sound is selected, each pad will play the sound at a different pitch, determined by the setting of the selected track's PitchTbl parameter (see Chapter 2), and whether or not the Kit Mapper is turned on (the Kit Mapper is described later in this chapter).

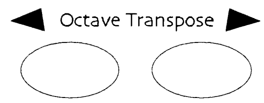
Kit Sounds

Kit sounds utilize a powerful structure first introduced in ENSONIQ's MR synthesizer series. In a kit sound, each note from B1 to D7 actually plays its own complete sound—either a standard sound or another kit sound. Therefore, what the pad plays depends on the sound you've assigned to it.

If you've assigned the same standard sound to more than one pad, they play the same sound. Since each pad has its own set of PAD parameters (described later in this chapter), you can program the pads to play different variations of the same sound, perhaps setting them to play at different pitches. You can also program each pad in a kit to play a sound that's unrelated to what the other pads are playing—in this case each pad triggers something completely unique.

Note: Each pad in a kit defaults to playing its sound at the pitch that would be heard at C4. The Tuning Shift parameter described later in this chapter can change the pitch of the pad's sound.

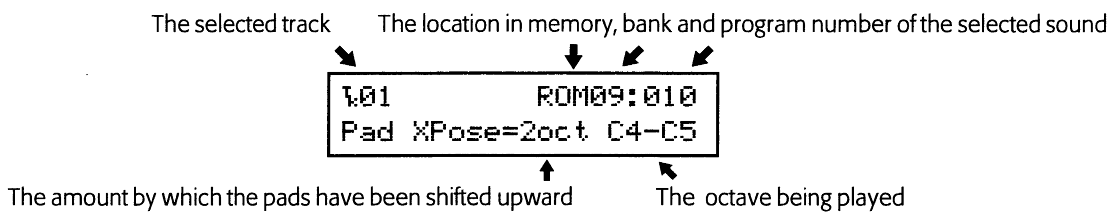
Octave Transpose Buttons



The ASR-X pads default to playing the octave-plus-one-note beginning at the C natural two octaves below Middle C—C2. The Octave Transpose buttons provide a means of changing which of five octaves in the selected sound will be addressed by the 13 pads. You can:

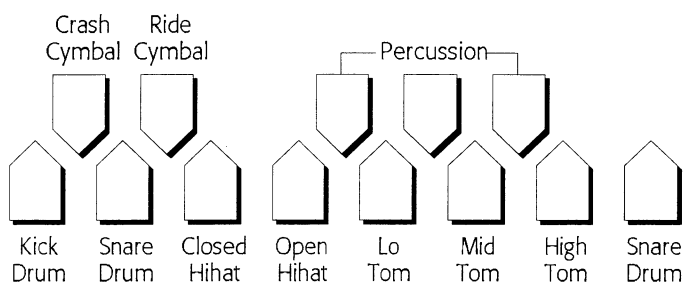
- repeatedly press the either Octave Transpose button to redirect the pads upward or downward.
- press either Octave Transpose button once, and turn the Value knob to select the desired octave.

The Pad Xpose (short for "pad transpose") display shows you the octave in the currently selected sound that's being played by the pads:



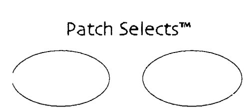
The Kit Mapper

Typically, the pads in the ASR-X play 13 adjacent notes. When you're using a kit sound tha conforms to the ENSONIQ drum or percussion maps (described in Chapter 9), these 13 notes may be variations of the same kit component. The Kit Mapper re-assigns the pitches played by the pads so that the important elements of a typical kit—which are mapped to different octaves within the kit—are available at once.



- To turn on Kit Mapper , tap the left Octave Transpose button until the the display shows "PadXpose=Kit Mapper." To turn it off, press the right-hand Octave Transpose button.

Patch Select Buttons



The Patch Select™ buttons provide access to variations of the ASR-X ROM sounds. The layers in these sounds are programmed to supply up to four different versions of the basic sound, or sometimes completely different sounds that complement the basic sound. The Patch Select buttons are used for turning on and off these different sets of layers.

Note: All ENSONIQ samplers since the original EPS have offered the expressive power of Patch Selects. Well-programmed sounds created on those instruments take advantage of this feature.

To hear the effect of the Patch Select buttons, press one or both as you play an ASR-X ROM sound. The four possible Patch Select states are:

- Right—when only the right button is pressed.
- Left—when only the left button is pressed.
- Both—when both buttons are depressed.
- Off—when no Patch Select button is pressed.

The default behavior of the Patch Select buttons is that they are active only when they're being held down. This can be changed by resetting the System/MIDI Patch Selects parameter (see Chapter 7).

Patch Selects and MIDI

The Patch Select states listed above can be invoked via MIDI by sending MIDI controller 70 values on the MIDI channel of the track containing the sound you wish to manipulate. Send the ASR-X a value of:

- 32 to "press" the left Patch Select button.
- 127 to "press" both Patch Select buttons.
- 64 to "press" the right Patch Select button.
- 0 to "press" neither Patch Select button.

Programming the Pads

Overview

The ASR-X allows you to edit the behavior of the pads in any kit sound. You can:

- select a new sound to be played by the pad.
- adjust the manner in which the pad will play its sound by setting volume, panning, effect routing and tuning parameters.

When a pad is playing a sound that uses waves you've created on your ASR-X by sampling or resampling, you can also:

- set the manner in which the pad's sound will play back its wave(s).
- program the sound using an extensive suite of sound-sculpting parameters.
- perform various permanent operations upon the sound's wave data.

Note: Attempting to perform wave operations by pressing the Pad Process button when the selected sound is not playing an ASR-X-created wave will cause the display to momentarily show "Can't process sound! Try resampling."

Any ASR-X sound can be converted into a kit so that it can be edited. The sound will function essentially as it always did—however, you'll be able to re-program the sound pad-by-pad.

In order to program a sound's pads, two conditions must be met:

1. The sound must be a kit, or converted into a kit for editing.
2. The sound must be in RAM, so that it can be altered (sounds in ROM are unalterable).

The ASR-X has a name for a sound that meets both of these requirements: it's called a *RAM kit*.

To Prepare the Selected Track's Sound for Pad Editing

The ASR-X knows when a sound is ready to be edited. If the selected sound is a RAM kit, it's already editable. When the selected sound is not a RAM kit—if, for example, it's a ROM sound or a non-kit RAM sound—the ASR-X will ask the following question when you press the Pad Sound or Edit buttons:

```
Make a RAM kit from
T01:Sweep Bass?
```

The selected track   The selected sound

When you press the "Yes" button in response to this question, the ASR-X creates a copy of the selected sound as a kit in RAM, and assigns it to the selected track. The newly created kit will add an underscore and a two-digit number to the end of the sound's original name—abbreviating the original name if necessary— to show that it's based on the original sound. The new kit can be found in the USER-SND and DRUM-KIT SoundFinder categories.

Tip: You can rename a RAM kit at any time using the MemoryManager. See Chapter 7.

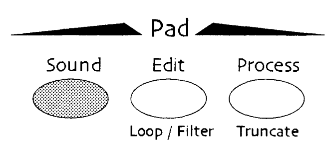
If the selected sound is a ROM kit sound—so that it already has the desired kit structure for editing, but is a permanent, uneditable ROM sound—you can press the Pad Sound or Edit buttons and press any pad to view the name of the sound it's playing and the settings of its parameters. If you attempt to change the sound played by a pad, the above display will appear, asking if you want to make a RAM copy of the kit.

Selecting a Pad for Editing

To edit a pad, it must first be selected. This is accomplished by simply pressing the desired pad. The displays that relate to the various pad-editing functions all show, in their upper-left corners, the pad that's currently selected. If you'd like to select a pad outside of the current pad octave range, use the Octave Transpose buttons to select the octave in which the pad can be found—then press the desired pad to select it for editing,

Choosing a Pad's Sound

When the selected track contains a RAM kit sound, pressing the Pad Sound button allows you to choose a new sound for any of its pads.



The pad sound-selection display is similar to the track sound-selection display:

The pad being edited   The selected track   The sound's memory residency, bank and program number

```
padD2  T01 ROM08:000
PERCSOLO: Thump Kick
```

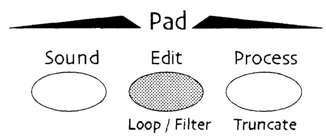
The currently selected SoundFinder category   The currently selected sound

You can choose a new sound for the selected pad by turning the Sound Type knob to pick the type of sound you want, and the Sound Name knob to select the individual sound.

Tip: If the selected track contains a ROM kit sound, you can press the Pad sound button and then press each pad button to view the name of the sound being played by the pad; however, you can only change a pad's sound if you've copied the ROM kit into RAM for editing.

Overview of the Pad Edit Parameters

The Pad Edit parameters allow you to determine the behavior of the sound played by each pad in a RAM kit. This includes ROM or RAM sounds that play the ASR-X's built-in sound waves, as well as the waves that you create yourself and have sent to pads. All of these parameters are accessed by pressing the Pad Edit button.



To make navigating among these many parameters simpler, the Pad Edit parameters are divided into 12 sub-groups.

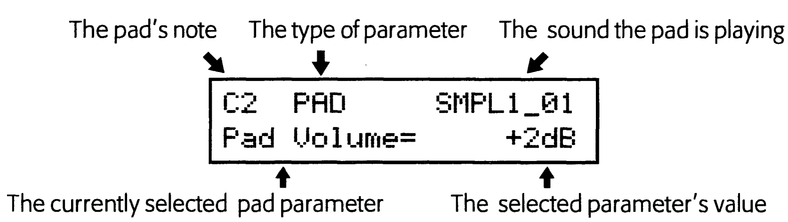
The PAD parameters are always available, regardless of the type of sound being played by the selected pad. They allow you to determine the manner in which the pad will play its sound, and are described below in "Determining a Pad's Behavior." The PAD parameter settings become part of the selected track's RAM kit.

Tip: You can view the PAD parameter settings for a ROM sound by pressing the Pad Edit button, pressing any pad, and turning the Parameter knob to view the pad's settings. In the case of non-kit sounds, all of the pad's will show the same parameter values.

The ASR-X also provides the following groups of sound parameters when the selected pad is playing a sound based on an ASR-X-created wave stored in RAM memory. The settings for these parameters become part of the pad's sound. The full sample-programming power of the ASR-X is unleashed through the use of these parameters, described later in this chapter in "Editing a Pad's Sound."

- WAVE
- PTCH
- ENV1
- FILT
- FLT1
- FLT2
- ENV2
- AMP
- ENV3
- MOD
- MISC

All of the Pad Edit parameters share a common display layout that tells you the note corresponding to the pad being edited, the type of parameter selected, the name of the sound the pad is playing, and the selected parameter's current value:



Tip: You can jump among the parameter groups by repeatedly pressing the Pad Edit button.

Note: When editing Pad parameters, it's important to remember that each track can play its sound—or sounds, in the case of kits—in its own way. If editing Pad parameters produces unexpected results, check the track parameters for the currently selected track to see if they're influencing the sound you're attempting to edit.

Determining a Pad's Behavior

PAD Parameters

The PAD parameters allow you to determine the manner in which each pad in the currently selected RAM kit will play its sound. All of the PAD parameters settings are permanently stored in the RAM kit when you save it. When a pad's sound contains multiple layers, all of its layers are affected simultaneously by PAD parameter edits.

Pad Volume

The Pad Volume parameter allows you to raise or lower the level of the sound being played by the selected pad. The parameter can be set anywhere from -50dB to +14dB. When the Pad Volume parameter is set to 0dB, the pad's sound will play at its originally programmed volume.

Pad Pan

The Pad Pan parameter allows you to shift the stereo image of the selected pad's sound leftward or rightward in the stereo field. The parameter can be set anywhere from Left -64 to Right +63. A value of Center 00 will leave the sound's original stereo placement intact.

Note:: This parameter shifts the entire sound being played by the selected pad left or right, so that the sound's internal stereo imaging is preserved.

FX Bus

The FX bus parameter allows you to assign the selected pad's sound to one of the ASR-X's FX busses. The parameter can be set to:

- Prog—so that if the pad is playing a standard sound, the sound's Alt Bus will be used, or if the pad is playing a kit sound, the sound played by each note in the kit will use its own FX Bus setting.
- Insert—to route the pad's sound to the currently selected sequence's insert effect.
- LightReverb—to apply a minimal amount of reverb to the pad's sound.
- MediumReverb—to apply an average amount of reverb to the pad's sound.
- WetReverb—to apply a large amount of reverb to the pad's sound.
- Dry—to leave the pad's sound un-effected.
- AuxOut1, AuxOut2, AuxOut3 or AuxOut4—to send the pad's sound directly to one of the four auxiliary outputs. These values are only available when an X-8 output expansion board is installed.

Note: These values are used whenever the selected track's FX Bus parameter (see Chapter 2) is set to "Prog."

Tuning Shift

The Tuning Shift parameter allows you to raise or lower the note to be played by the pad. In many cases, this parameter will have the effect of raising or lowering the pitch at which the pad's sound will be heard. When the pad is playing a sound that contains more than a single wave—examples of this would be drum kits, or sounds with multiple-sample layers—the parameter will have the effect of pointing the pad to a different note—and therefore, possibly different wave data— within the pad's sound. The parameter can be set anywhere from -64st ("steps") to +63st. When the Tuning Shift parameter is set to 0st, the pad's sound will play at the pitch equivalent to striking a Middle C (C4) . When the pad is playing a wave you've created in the ASR-X, the wave will be heard at its original pitch.

Note: The Tuning Shift parameter raises or lowers the note to be played by the pad in semitone steps when the sound employs an equal-temperament tuning table. However, some ASR-X sounds use special tunings. For example, the tuning of drum sounds often varies only by small increments as you move from key to key, in order to simulate the subtle pitch shifts of real-world drums. The effect of the Tuning Shift parameter depends, therefore, on the tuning table used by the pad's sound.

Editing a Pad's Sound

The following groups of parameters allow you to program sounds based on ASR-X waves.

The ASR-X Modulators

Some of the parameters in this section can be changed—or *modulated*—in real time by an external mechanism called a *modulator*. These parameters can be set to:

Off	for no modulation
Full Amt	The maximum amount of modulation is applied to the modulation destination
LFO	the selected wave's LFO
Stepped	a significant amount of random noise modulation at a rate determined by the NoiseSource Rate parameter (see later in this section)
Smoothed	a subtle amount of random noise modulation at a rate determined by the NoiseSource Rate parameter (see later in this section)
Env1	the selected wave's Envelope 1
Env2	the selected wave's Envelope 2
Env3	the selected wave's Envelope 3
Velocity	MIDI velocity: higher values cause greater modulation; lower values cause less modulation
Vel+Press	a combination modulator, with MIDI velocity and pressure messages together achieving maximum modulation amounts
MIDI Key#	MIDI note numbers set the modulation destination parameter to absolute corresponding values
Keyboard	MIDI note numbers above C4 raise the modulation destination's value from its setting; lower note numbers reduce it
Pressure	MIDI channel or polyphonic (ENSONIQ PolyKey™) pressure; higher values cause greater modulation, lower values cause less modulation
PitchWhl	MIDI pitch bend raises or lowers modulation destination value; a pitch bend wheel at rest transmits a central modulation value of 64
ModWheel	MIDI modulation wheel (controller #1); maximum values are attained when the mod wheel is pushed all the way forward
Whl+Press	A combination modulator, with MIDI mod wheel and pressure messages together achieving maximum modulation amounts
FootPedal	MIDI foot pedal (controller #4); maximum values are attained when the foot pedal is pushed all the way forward
Sustain	MIDI sustain pedal (controller #64) operating as a modulation switch: down produces maximum modulation; up produces no modulation
Sostenuto	MIDI sostenuto pedal (controller #66) operating as a modulation switch: down produces maximum modulation; up produces no modulation
SysCTRL1	the first of the ASR-X's assignable MIDI controllers (see Chapter 7)
SysCTRL2	the second of the ASR-X's assignable MIDI controllers (see Chapter 7)
SysCTRL3	the third of the ASR-X's assignable MIDI controllers (see Chapter 7)
SysCTRL4	the fourth of the ASR-X's assignable MIDI controllers (see Chapter 7)
PatchSel	the Patch Select buttons: the left button produces a modulation value of 32; the right button 64; both buttons 127; neither button 0

WAVE Parameters

The Playback of Waves

The waves you create on your ASR-X are digital recordings of a sound. Digital recording captures audio by taking snapshots of the sound many times per second—44,100 times per second in the ASR-X. Therefore, instead of recording continually, it actually samples the sound many times per second. On playback, the ear perceives these snapshots, or "samples," as a single sonic entity—in the ASR-X, this single entity is called a "wave." The ASR-X can play the list of samples that make up a wave forward or backward, play specified sections of samples, or play sections of them over and over for as long as you hold down a pad or key on an external MIDI keyboard. The WAVE parameters control these features.

Parameter	Range	Description
PlayMode	OnceForward, OnceBkwrd, LoopForward, LoopFwd&Bwd	Determines the direction and manner in which the wave will play: OnceForward—the wave will play from beginning to end once and stop. OnceBkwrd—the wave will play from back to front once and stop. LoopForward—the wave will play from the beginning to its loop end point, at which time it will start again from the loop start point and play to the loop end point repeatedly until the pad or key is lifted. LoopFwd&Bwd—the wave will play from the beginning to its loop end point, at which time it will play backwards to the loop start point and then forwards to the loop end repeatedly until the pad or key is lifted.
Start/Loop	00 to 99% for sample start, loop start and loop end points	Provides three editable fields that allow you to set the wave playback start point, loop start point and loop end point as percentages of the wave's samples. This can be viewed as a coarse adjustment for these three points. Optimal loop points are automatically offered when the System/MIDI AutoZero Corss parameter is set to "On" (see Chapter 7).
Sample Start	0 to the number of samples that comprise the entire wave.	Determines the point from which the wave will play on key-down, expressed as individual samples. This is a fine-adjust for the wave playback start point.
Loop Start	0 to the number of samples that comprise the entire wave.	Determines the point from which the wave will loop when PlayMode is set to LoopForward or LoopFwd&Bwd, expressed as individual samples. This is a fine-adjust for the wave playback loop start point.
Loop End	0 to the number of samples that comprise the entire wave.	Determines the point to which the wave will play, whether the wave is set to loop or not, expressed as individual samples. This is a fine-adjust for the wave playback loop end point.
StartToEndIndex	0 to 127	Allows you to choose one of 128 locations between the Sample Start and Loop End points from which to begin wave playback. A setting of 0 causes the wave to start playback from the Sample Start point.
IndxModSrc	(see modulator list)	Selects a modulator for the StartToEndIndex. See "The ASR-X Modulators" earlier in this section for a list of the available StartToEndIndex modulators.
Index ModAmt	-127 to +127	Determines the degree to which the IndxModSrc will affect the StartToEndIndex.

A Couple of WAVE Ideas

- You can set Sample Start to a higher value than Loop Start. When your wave is a beat loop, this lets you play a few beats from the end of the wave before the loop starts playing.
- By modulating the StartToEnd Index, you can start playback of a wave from a different place within the wave every time you strike its pad. When Envelope 3 is set to Repeat (see later in this chapter), the wave will restart playback from the StartToEnd Index point each time the envelope repeats.

PTCH Parameters

The PTCH parameters—for "pitch parameters"—allow control of the selected sound's pitch bend, tuning, glide and modulation.

Parameter	Range	Description
Pitch Bend Up	12 down to 12 up, Off	Determines the maximum number of steps by which the pad's sound will be raised or lowered when the ASR-X receives pitch bend messages from a MIDI pitch bend wheel pushed all the way up (forward).
Pitch Bend Down	12 down to 12 up, Off	Determines the maximum number of semitone steps by which the pad's sound will be lowered or raised when the ASR-X receives pitch bend messages from a MIDI pitch bend wheel pulled all the way down (back).
PitchBendMode	Normal, Held	Determines whether or not the sound will pitch-bend normally or in held mode. Normally, when MIDI pitch bend messages are received, all notes sounding are affected by the pitch bend messages. In held mode, only notes physically being held down—notes which have not yet received a key-up message—are affected when pitch bend messages are received. The held option is useful for a number of musical situations, including the simulation of pedal steel guitars or solo string lines played against a chordal background.
SemitoneTuning	-64st to 64st	Lowers or raises the pitch of the pad's sound by semitones.
Fine Tuning	-127 to +127	Fine tunes the pitch of the pad's sound by steps of one cent (1/100 of a semitone).
KeybdTrack	various	Determines the pitch response of the pad's sound to MIDI note numbers. The default setting is Western equal temperament; other options include ratio relationships to received note numbers, inverted equal temperament or assignment to the sound's pitch table, determined by the PitchTbl parameter (see below).
PitchTbl	various, RAM	Selects a pitch table which may be accessed by the sound (see "List of ROM System Pitch Tables" in Chapter 9 for a list of pitch tables). The ASR-X supports the MIDI Tuning Change Standard—pitch tables may be transmitted via MIDI SysEx to the ASR-X's RAM pitch table (see "ASR-X MIDI Implementation" in Chapter 9 for more details).
Glide Mode	Off, On	Enables/disables glide (portamento) in the pad's sound. The exact nature of the sound's glide is determined by the Voice Mode parameter (see below).
Glide Time	0 to 127	Determines the amount of time it takes for the pitch to glide from one note to another when glide is enabled: 0 represents the shortest glide time, 127 the longest. When Voice Mode=Mono (see below), glide in the ASR-X is constant-time portamento: the time it takes to glide from note to note is the same regardless of how far way from each other the notes are.
Voice Mode	Poly, Mono	Determines whether the pad's sound will be polyphonic or monophonic. When Voice Mode=Poly, notes glide from a random selection of pitches.
PtchModSrc	(see modulator list)	Selects a pitch modulator for the pad's sound. See "The ASR-X Modulators" earlier in this section for a list of the available pitch modulators.
Pitch ModAmt	-127 to +127	Determines the amount and polarity of pitch modulation caused by the Pitch Mod within the overall limit designated by the Mod Range parameter (see below).
Pitch ModRange	0st to 64st	Determines the maximum amount of pitch shifting the Pitch Mod may cause, in keyboard steps. The amount of pitch change invoked by each step is dependent on the sound's pitch table.
LFO Pitch ModAmt	0 to 127	Determines the degree to which the LFO will affect the pitch of the pad's sound.

Env1PitchModAmt	-127 to +127	Env1PitchModAmt provides a special routing that endows Envelope 1 with unique capabilities in the modulation of the sound's pitch. When applied to the sound's pitch via the Env1PitchModAmt parameter, Envelope 1 automatically sustains at the pre-enveloping pitch, regardless of its Sustain Level (4) setting. Instead, its Sustain Level (4) setting serves to determine which Envelope 1 level values will cause the pitch to rise above the un-enveloped pitch and which level values will drive it below. Envelope 1 level values equal to the Sustain Level (4) value will cause the sound to play at the un-enveloped pitch. Higher level values will shift the pitch upward, and lower values will shift the pitch downward. This feature allows for the creation of bi-directional pitch envelope shapes, while conveniently ensuring that the pad's sound will always sustain at the un-enveloped pitch.

ENV1 Parameters

The following parameters pertain to the first of the selected sound's three envelopes. Envelope 1 is typically applied to pitch, though it may be used as a modulator for any modulatable parameter. When Envelope 1 is applied to a sound's pitch through the Env1PitchModAmt pitch parameter, it's endowed with some special attributes, described above.

Parameter	Range	Description
Envelope Mode	Normal, Finish, Repeat	Envelope 1 may function in one of three ways: • Normal—Envelope 1 plays through normally. When the key is released, the envelope takes the Release Time (5) to go from the current level down to zero. • Finish—Envelope 1 finishes playing through all its stages, ignoring the key-up event. The envelope spends no time at the Sustain Level (4) stage. When the Decay Time (4) interval is finished, instead of stopping at the Sustain Level (4) stage, the envelope immediately goes into the Release Time (5) stage. This is good for percussive-type sounds where you want the envelope to be the same for every note, no matter how long the key is held down. • Repeat—At the end of the Ramp Time (3) stage, instead of sustaining, Envelope 1 goes immediately back to the beginning and repeats, starting with the Attack Time (1) stage. When the key is released, the envelope stops repeating and moves into the release stage, taking the Release Time (5) interval to go from the current level down to zero. This type of envelope can be used to create complex LFO-type effects.
Attack Time (1)	0 to 99	Determines the time it takes for the envelope's level to travel from zero (when a note-on is received) to Attack Level (1). The higher the value, the longer the time.
Attack Level (1)	0 to 127	Determines the level the envelope will reach at the end of the time defined by Attack Time (1).
Ramp Time (2)	0 to 99	Determines the time it takes the envelope to go from Attack Level (1) to Ramp Level (2).
Ramp Level (2)	0 to 127	Determines the level the envelope will reach at the end of Ramp Time (2).
Ramp Time (3)	0 to 99	Determines the time it takes the envelope to go from Ramp Level (2) to Ramp Level (3).
Ramp Level (3)	0 to 127	Determines the level the envelope will reach at the end of Ramp Time (3).
Decay Time (4)	0 to 99	Determines the time it takes the envelope to go from Ramp Level (3) to the Sustain Level (4) stage. At the end of DecayTime (4,) the envelope will remain at Sustain Level (4) until the key is released.

Sustain Level (4)	0 to 127	Determines the level the envelope will reach at the end of Decay Time (4) and that it will retain until a note-off or sustain-off message is received. When Envelope 1 is used to modulate pitch through the Env 1 Amt parameter, this parameter functions differently—see "Env1PitchModAmt" above.
Release Time (5)	0 to 99	Determines the time it takes the envelope to return to zero after the key has been released.
Keybd TimeScaling	0 to 99	Makes the envelope times longer or shorter, depending on the key played. The scaling effect of this parameter is based on a center break point of F4+. Higher values will make all envelope 1 times (except Release Time [5}) shorter for keys above F4+, and longer for keys below F4+. Envelope times for F4+ itself are not affected by this parameter.
VelAtckTimeModAmt	0 to 99	Determines the degree to which higher velocities will shorten Envelope 1's Attack Time (1). This parameter will have no effect if Attack Time (1)=0.
VelRelTimModAmt	-127 to +127	Determines the degree to which higher release velocities will make Envelope 1's Release Time (5) shorter or longer. When the value is positive, a higher release velocity value will result in a shorter Release Time (5). When the value is negative, a higher release velocity value will result in a longer Release Time (5). This parameter will have no effect if Release Time (5)=0.
Vel Levels ModAmt	-127 to +127	Determines to what degree velocity will affect envelope levels. Values above 0 increase the amount of velocity required to reach the Envelope 1 values determined by its level settings. Vel Curv gives you further control over the velocity response of the envelope.
Vel Curve	Quickrise, Convex1, Convex2, Convex3, Linear, Concave1, Concave2, Concave3, Concave4, LateRise	Selects which of the velocity response curves the envelope will use if the velocity level control (Vel Levels ModAmt) is set to some value other than zero.

FILT Parameters

Each sound in an ASR-X sound has a pair of independently configurable multi-mode dynamic digital filters. The following FILT—for "filter"—parameters determine the overall behavior of the sound's two filters.

Parameter	Range	Description
Mode	3PoleLP/1PoleLP, Resonant2LP/2LP, Resonant2BP/2BP, FilterBypass	Determines the filter configuration for the sound: LP=low-pass filter, which allows frequencies lower than the filter cutoff frequency (Fc) to be heard; HP=high-pass filter, which allows frequencies above the Fc to be heard. Each sound has two filters: the first is always LP, while the second may be LP or HP. The steepness of each filter is determined by its pole setting; the higher the pole value, the more extreme the filter's slope becomes. A 1-pole filter rolls off frequencies at 6 dB per octave, a 2-pole filter at 12 dB, and a 3-pole at 18 dB per octave. The Resonant2LP/2LP value makes both filters resonant; Resonant2BP/2BP creates a combined dual resonant band pass filter.
Link	Independent, FLT2 uses FLT1	When set to On, Filter 2 uses Filter 1's settings; when Off, Filter 2 uses its own settings.
Resonance (Q)	0-50	When Filter Mode=Resonant2LP/2LP, this sets the loudness of the frequencies at the cutoff points of both filters. When Filter Mode=Resonant2BP/2BP, this sets the width of both of the bands, and the cutoff frequency levels.

FLT1 and FILT2 Parameters

The following parameters are available for both of the selected sound's two filters.

Parameter	Range	Description
Filter Cutoff	0 to 127	Determines the selected filter's cutoff frequency. Filter 1 is always a low-pass filter: frequencies within the selected wave that are lower than the FLT1 Filter Cutoff setting will pass, or be heard. Frequencies above it will be filtered out. Lowering the FLT1 Filter Cutoff value is similar to turning down the treble on a home stereo. The effect of the cutoff frequency in FILT2 will depend on the setting of the FILT Mode parameter.
Keybd Track	Off, various	Determines how the selected filter's cutoff frequency will change as various pitches are played, expressed in ratios. Positive values raise the cutoff as higher notes are played.
TrackBreakpoint	C-1 to G9	Determines which note will be treated as the nominal center of the key track range, and produce neither negative or positive cutoff modulation.
Cut ModSrc	(see modulator list)	Selects a modulator for the selected filter's cutoff frequency. See "The ASR-X Modulators" earlier in this section for a list of the available modulators.
Cutoff ModAmt	-127 to +127	Determines the amount by which the Cut ModSrc will lower or raise the selected filter's cutoff frequency.
Env2CutoffModAmt	0 to 127	Determines the degree to which Envelope 2 will affect the selected filter's cutoff frequency.

ENV2 Parameters

The following parameters pertain to the second of the selected sound's three envelopes. Envelope 2 is typically applied to filter cutoff settings, though it may be used as a modulator for any modulatable parameter. The parameters available for Envelope 2 are identical to those associated with Envelope 1 (see "ENV1 Parameters" earlier in this chapter).

AMP Parameters

The AMP—for "amplifier"—parameters provide control of the selected sound's keyboard rolloff characteristics, volume modulation and stereo panning modulation.

Parameter	Range	Description
Rolloff Mode	Off, Below, Above	Enables/disables a progressive volume reduction for the sound, either above or below the Roll Breakpoint (see below).
Roll Slope	0-127	Determines the extremity of the rolloff when Rolloff Mode is set to "Above" or "Below.".
Roll Breakpoint	C-1 to G9	Determines the note above or below which the rolloff occurs when Rolloff Mode is not set to "Off."
Vol ModSrc	(see modulator list)	Selects a modulator for the sound's volume. See "The ASR-X Modulators" earlier in this section for a list of the available modulators. Note that Envelope 3 always affects the sound's volume.
Volume ModAmt	-127 to +127	Determines the degree to which the Vol ModSrc will lower or raise the volume of the sound.
Pan ModSrc	(see modulator list)	Selects a modulation source for the sound's position in the stereo field. See "The ASR-X Modulators" earlier in this section for a list of the available modulators.
Pan ModAmt	-127 to +127	Determines the degree to which the modulator will move the sound's stereo position to the left (negative values) or right (positive values).

Alt Bus	Default, LightReverb, MediumReverb, WetReverb, Dry	Determines the effect bus to which the sound will be routed when it's selected for a track if the System/MIDI AutoSelect FXBus parameter is set to "On."

ENV3 Parameters

The following parameters pertain to the third of the selected sound's three envelopes. Envelope 3 is typically applied to the sound's volume settings, though it may be used as a modulator for any modulatable parameter. The parameters available for Envelope 3 are identical to those associated with Envelope 1 (see "ENV1 Parameters" earlier in this chapter).

MOD Parameters

The MOD parameters—or "modulation parameters"—control the behavior of the sound's LFO and noise generator.

Parameter	Range	Description
LFO Shape	Triangle, Sine+Tri, Sine, Pos-Tri, Pos-Sine, Sawtooth, Square	Determines the wave shape of the sound's LFO: Triangle—commonly used to modulate pitch to produce vibrato Sine+Tri—mixture of a sine and triangle wave, a somewhat pointy sine wave Sine—pure fundamental frequency, more rounded in its peaks and valleys than the triangle wave Pos-Tri—a positive-only triangle wave useful for simulating vibrato on instruments like the guitar where a player can only bend notes up Pos-Sine—positive-only sine wave useful for simulating vibrato on instruments like the guitar where a player can only bend notes up Saw—sawtooth wave commonly used for special effects Square—positive-only square wave useful for producing in-tune trill effects
LFO Start Phase	0 to 127	Determines the starting phase of the LFO, when Retrigger=On. With a setting of 0, the LFO will always restart at the beginning of its cycle. Tip: When LFO Start Phase=0, this parameter determines what part of the LFO wave will be applied as a fixed modulator upon key-down.
LFO Rate	0 to 99	Determines the speed of the LFO. Tip: When this parameter is set to 0, the LFO will produce modulation only upon new note-ons, and will not further modulate already-sounding notes.
Rate ModSrc	(see modulator list)	Selects a modulator for the LFO rate. See "The ASR-X Modulators" earlier in this section for a list of the available LFO Rate Mod modulators.
LFO Rate ModAmt	-127 to +127	Determines the degree to which the Rate ModSrc will slow down or speed up the LFO Rate.
LFO Depth	0 to 127	Determines the amplitude of the LFO.
DpthModSrc	(see modulator list)	Selects a modulator for the LFO depth. See "The ASR-X Modulators" earlier in this section for a list of the available LFO Depth Mod modulators.
LFODepth ModAmt	-127 to +127	Determines the degree to which the modulator will decrease or increase the LFO depth.
LFO Delay Time	0 to 99	Determines the time it takes for the LFO to go from zero to the amount determined by the LFO Depth parameter. Values above 0 will cause the LFO to take longer to achieve its full depth.

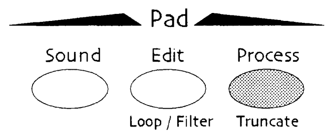
LFO Key Restart	Off, On	Determines whether the LFO will restart with each note-on. When set to "Off," the LFO will cycle continuously without resetting, whether a note is being played or not. When set to "On," the LFO waveform will always commence at its starting location, as determined by the LFO Start Phase parameter, when a note-on is received.
LFO Sync	Normal, various rhythmic divisions of the current sequence tempo or received MIDI clocks	Enables/disables synchronization of the LFO to the currently selected sequence, by providing rhythmic divisions of its pulse. The LFO may be also be synchronized to received MIDI clocks when the System/MIDI ClockSource parameter is set to "MIDI."
NoiseSourceRate	0 to 127	Determines the speed of the stepped and smooth modulators (see "The ASR-X Modulators" earlier in this section). Tip: When this parameter is set to 0, the noise modulators will choose new random values only upon new note-ons, and will not further modulate already-sounding notes.
Noise Sync	Normal, various rhythmic divisions of the current sequence tempo or received MIDI clocks	Enables/disables synchronization of the stepped and smooth noise modulators to the currently selected sequence, by providing rhythmic divisions of its pulse. The LFO may be also be synchronized to received MIDI clocks when the System/MIDI ClockSource parameter is set to "MIDI."

MISC Parameters

The MISC—for "miscellaneous"—parameters are a small assortment of parameters and a sound-re-naming facility.

Parameter	Range	Description
Sustain Pedal	Off, On	Enables or disables the sound's response to sustain pedal presses.
Key Group Assign	Off, 1 to 16	Allows assignment of the sound to one of 16 monophonic key groups. Key groups are used when you'd like two or more sounds to cut each other off, particularly helpful when emulating real-world situations where two sounds would be mutually exclusive. For example, when programming hi-hat sounds, you can assign your open hi-hat sound and your closed hi-hat sound to the same key group. When these two sounds are played as part of a RAM kit, the last one played will silence the other, as it would in a real hi-hat.
SoundFinder	all SoundFinder categories	Determines the SoundFinder category for the sound.
FinderPref	None, DEMO-SND, USER-SND, USER&DEMO	Enables inclusion of the sound in the DEMO-SND and USER-SND SoundFinder sound type categories. The USER-SND category provides easy access to sounds you've created yourself.
Rename Sound?	(see description)	When this display is visible, pressing the Yes button will cause the sound naming page to appear. The top line of the display shows the sound's current name. You can re-name the selected sound by turning the Parameter knob or pressing the Select Track buttons to choose any of the 11 character positions, and turning the Value knob to dial in the desired character for each position.

Processing a Sound's Wave



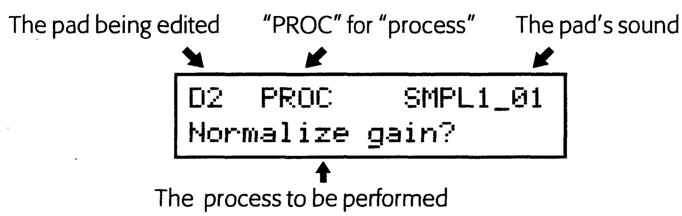
When a pad contains a sound based on an ASR-X-created wavesample, or "wave," the Pad Process button provides access to a number of tools for processing the sound's wave. Since these tools actually modify the wave data, each time you perform one of the pad processes:

- the ASR-X makes a copy of the wave
- it performs the selected operation
- it places the processed copy on the Scratch Pad. You can then play the scratch pad to audition the results of the process you've performed.

If you're pleased with the results, you can send the contents of the Scratch Pad to a pad in your kit (the procedure for sending to pads is described in Chapter 5).

The Pad Process Display

The processes accessed by pressing the Pad Process button share a common display:



This display asks you if you'd like to perform the process shown. For some of these questions—Normalize gain?; Invert Sample data?; Truncate length?—a press of the Yes button initiates the displayed procedure. For the others, pressing the Yes button leads you to further settings that you may want to adjust before performing the procedure. You can cancel the selected process whenever the red / green No / Yes LEDs are flashing by pressing the No button.

As each process takes place, the ASR-X display informs you of its progress.

The Pad Processes

Normalize gain?

The ASR-X can normalize the selected wave to digitally boost its volume to its loudest level short of clipping. This allows the wave to take the fullest possible advantage of the 16 bits available for its reproduction, and helps ensure that you won't have to over-boost its volume for it to be heard. Normalization seeks out the wave's loudest sample, multiplies it to the highest acceptable level, and then uses the same multiplication value on the rest of the wave's samples.

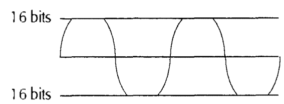
Since the process requires no user input, pressing the Yes button in response to "Normalize gain?" executes the normalization operation.

Scale loudness?

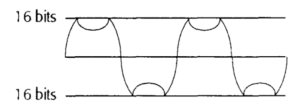
The ASR-X lets you lower or raise the overall volume of a wave by percentage you set through the use of its scaling facility. When you press the Yes button in response to "Scale loudness?" two settings become available that allow you to set the manner in which the wave will be scaled. To view these two settings, turn the Parameter knob; to adjust them, turn the Value knob.

- Scale factor—lets you set the percentage by which your wave's volume will be raised or lowered, from 1% to 200%. A setting of 100% will leave the wave at its present volume. Values lower than 100% will reduce its volume, and values over 100% will increase it.
- Clip Method—If the volume of a wave is scaled to a level that requires more than the available 16 bits, the sound will clip. The Clip Method provides two settings—Normal or Warp— that allow you to determine what will happen to such waves:

The Normal clip method squares off excess volume at 16 bits, resulting in standard clipping.



Warp takes the amount by which the wave would exceed 16 bits and applies it as a volume reduction.



Tip: The Warp setting can lead to some interesting distortion effects.

When you've set the two scaling parameters to your liking, press the Yes button to scale the wave.

Reduce sample bits?

The ASR-X samples audio at a resolution of 16 bits. While this resolution produces excellent sound, 16-bit data can use up significant amounts of the ASR-X's RAM. If you lower the resolution of a selected wave, you can free up RAM for more sampling. In addition, there may be times when you'd like a rougher-sounding sample. Reducing sample bits is an excellent way to deliberately "trash" a wave. When you press the Yes button in response to "Reduce sample bits?" the ASR-X presents a display that allows you to set the desired bit resolution of your wave.

```
Reduce         SMPL1_01
Number of bits=     12
```

Turn the Value knob to change this value

When you've selected the desired resolution, press the Yes button to reduce the wave's resolution.

Invert sample data?

The ASR-X can invert a wave's data, essentially turning it upside-down, in order to make it easier to loop. Inverting a wave does not change its sound.

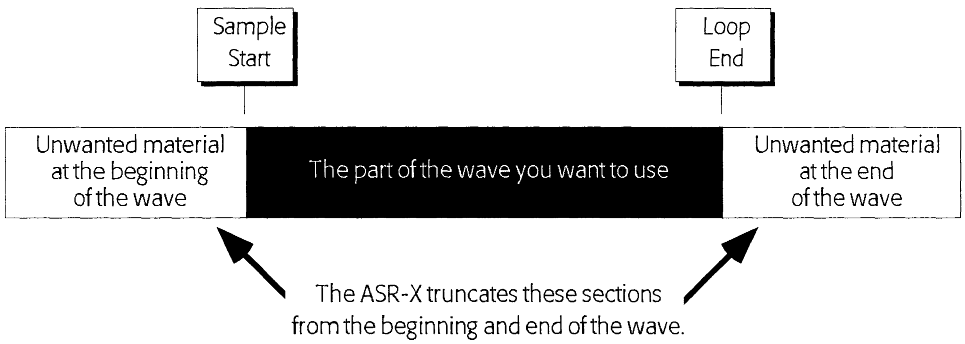
When this wave is inverted, it starts off looking like this...

...and ends up looking like this:

Answering the "Invert sample data?" by pressing the Yes button initiates the inversion operation.

Truncate length?

In order to make most efficient use of you ASR-X's memory, you should trim and discard those portions of your wave's data that you don't intend to use, freeing up the memory space they occupy. When you press the Yes button in response to "Truncate length?" the ASR-X deletes all data in your wave that occurs before the Sample Start point and after the Loop End point.



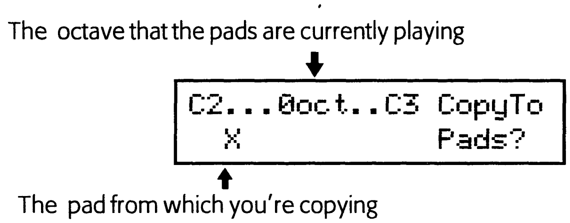
Copy sound?

The ASR-X allows you to copy the selected wave to other pads in the currently selected RAM kit. When you press the Yes button in response to "Copy sound?" the CopyMode display appears, where you can turn the Value knob to select one of two copy modes:

- Params—This copy mode will only copy the selected wave's parameter values without copying the wave itself.
- Params+Data—This copy mode will copy both the wave and its parameters.

Note: You can use this process to create multiple copies of a wave, each with its own loop settings. The original wave's Start/Loop, Sample Start, Loop Start and Loop End parameters are not duplicated along with its other parameter settings so that copies are created ready for re-looping.

When you've selected the desired copy mode, press the Yes button to perform the copy procedure. The ASR-X will show:

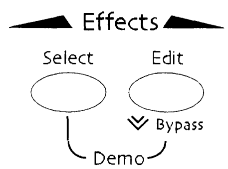


The display top line shows the octave currently selected for playing by the pads. You can press the Octave Transpose buttons to select a different octave's worth of pads to which to copy the selected wave and/or its parameters. The "X" shows you the pad that's currently selected—the pad that contains the data you're about to copy. When the desired octave is displayed, press the pad or pads to which you want to copy your data—a corresponding pad emblem will appear in the display for each pad you press. When you've selected your destination(s), press the Yes button to complete the copy procedure.

4 Effects

Overview of the ASR-X Effects

The ASR-X contains an ENSONIQ ESP2 digital signal processing chip that simultaneously provides two effects for each sequence: an insert effect and a global reverb. One track in each sequence—called the insert control track—can be endowed with some special abilities relating to the sequence's insert effect. These concepts are explained in this overview. Each of the 16 tracks in a sequence can be routed to either of these effects, left un-processed—or *dry*—or sent to one of the auxiliary outputs that are available when an X-8 output expansion board has been installed in the ASR-X . The procedures for taking advantage of these features appear elsewhere in this chapter, unless otherwise noted.



Insert Effects

Insert effects are powerful, highly programmable effects. There are 40 insert effects in the ASR-X, any one of which can be selected for use with any sequence:

01 Parametric EQ
02 Hall Reverb
03 Large Room
04 Small Room
05 Large Plate
06 Small Plate
07 NonLinReverb1
08 NonLinReverb2
09 Gated Reverb
10 Stereo Chorus
11 8-VoiceChorus
12 Rev→Chorus
13 Rev→Flanger
14 Rev→Phaser
15 Chorus→Rev
16 Flanger→Rev
17 Phaser→Rev
18 EQ→Reverb
19 Spinner→Rev
20 DDL→Chorus
21 DDL→Flanger
22 DDL→Phaser
23 DDL→EQ
24 Multi-Tap DDL
25 Dist→Chorus
26 Dist→Flanger
27 Dist→Phaser
28 Dist→Auto Wah
29 ResVCF→DDL
30 Dist→VCF→DDL
31 Pitch Detuner
32 Chatter Box
33 Formant Morph
34 RotarySpeaker
35 Tunable Spkr
36 Guitar Amp
37 Dist→DDL→Trem
38 Comp→Dist→DDL
39 EQ→Comp→Gate
40 EQ→Chorus→DDL

Insert effects can be manipulated in real time, allowing exceptionally musical control of their behavior (see "Insert Effect Real-Time Modulation Parameters" later in this chapter).

The Insert Control Track

In each sequence, one track can be designated as the insert control track. The insert control track has some special properties:

- Any sound that has been programmed to use an insert effect will automatically install that insert effect into the sequence when the sound is selected for use by the insert control track.
- The insert control track can be used to manipulate the sequence's insert effect in real time. In addition, when an external MIDI device is being used with the ASR-X, MIDI messages received on the insert control track's MIDI channel can also manipulate the insert effect.

Global Reverb

Global reverbs are top-quality programmable reverb effects. There are eight global reverbs, any of which can be selected for use in any sequence:

01 SmoothPlate	05 Small Room
02 Large Hall	06 Reflections
03 Small Hall	07 Bright
04 Big Room	08 Huge Place

FX Busses: How Sounds are Sent to the Effects

Each track and each pad in the ASR-X has its own FX Bus parameter for assigning its sound to one of the FX busses (for "effect busses"). The FX busses are the means by which sounds travel to an effect. There are five FX busses in an ASR-X as it's shipped from the factory:

- the insert FX bus
- the light reverb FX bus
- the medium reverb FX bus
- the wet reverb FX bus
- the dry FX bus

The insert FX bus directs a sound to the sequence's insert effect.

The light reverb, medium reverb and wet reverb busses all direct a sound to the global reverb.

Three busses are provided for this purpose so that each can be set to send a different amount of sound into the global reverb, resulting in three different degrees of reverb available for each sound.

A sound assigned to the dry FX bus will remain un-processed.

There is a sixth option available when assigning a sound to an FX bus, though it's not an FX bus. A "Prog" value is provided that allows the different keys in an ASR-X drum kit to retain their individual FX bus routings.

Chapter 2 describes the method for editing track parameters such as the FX Bus parameter. Chapter 3 describes how to edit the FX Bus parameter for a pad.

Note: When an X-8 output expansion board has been installed, an additional four stereo busses—AuxOut1, AuxOut2, AuxOut3 and AuxOut4—are available. These busses send a track and its sound directly to one of the auxiliary outputs. You can also use these stereo busses as eight mono busses by panning tracks hard left or right (see Chapter 2).

Tip: The ASR-X can automatically select FX busses for sounds as they assigned to tracks. See Chapter 7.

The Alt Bus

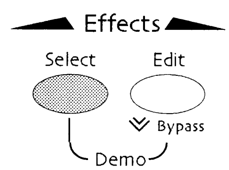
The ASR-X can automatically select an appropriate effect for a sound when it's selected for a track by reading the setting of the sound's Alt Bus parameter—see Chapter 7 to learn about the AutoSelect FXBus parameter, which enables this feature. See Chapter 3 to learn about setting the Alt Bus parameter. The Alt Bus is also used by standard sounds on tracks or pads whose FX Bus parameter is set to "Prog."

Selecting and Editing a Sequence's Effects

Selecting and editing a sequence's insert effect or global reverb involves the same simple pair of techniques, regardless of the effect being edited.

To Select an Effect

1. Press the Select button in the Effects section of the ASR-X front panel.



2. To select an insert effect, turn the Parameter knob so that the Insert Effect display appears:

This shows that you're on the insert effect selection display

```
Insert Effect=
    33 Formant Morph
```

The currently selected insert effect

To select a global reverb, turn the Parameter knob so that the Global Reverb display appears:

This shows that you're on the global reverb selection display

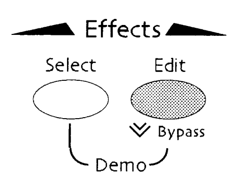
```
Global Reverb=
    01 SmoothPlate
```

The currently selected global reverb

3. Turn the Value knob to select the insert effect or global reverb you desire.

To Edit an Effect

1. Press the Edit button in the Effects section of the ASR-X front panel.



2. To edit the insert effect, turn the Parameter knob until the display shows:

```
Effects:
Edit insert effect?
```

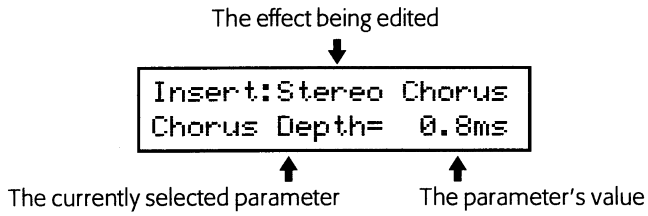
To edit the global reverb, turn the Parameter knob until the display shows:

```
Effects:
Edit global reverb?
```

3. When you've selected the type of effect you'd like to edit, press the Yes button.

4. Turn the Parameter knob to select any of the parameters available for the effect you're editing.

   The left-most parameters provide settings that determine the context in which the effect operates (see "Insert Effect and Global Reverb Context Parameters" below). These are followed by parameters relating to the effect itself—the top line of each of these displays shows the name of the effect being edited:



   At the end of the parameter list for each insert effect is a set of parameters that enable and control real-time modulation for the effect. See "Insert Effect Real-Time Modulation Parameters" below.

5. Turn the Value knob to change the setting of any parameter .

Tip: For a listing of all of the effect parameters available in the ASR-X, see later in this chapter.

Insert Effect and Global Reverb Context Parameters

In addition to the parameters provided for sound sculpting, each insert effect and global reverb also contains parameters that allow you to determine how the effect will fit into the sequence in which it's being used.

Each insert effect contains these context parameters:

- Insert FX Bus: Input Mix—This parameter allows you to establish the amount of insert effect you want to apply to any sounds routed to the insert FX bus. This is expressed as a wet/dry balance, with "dry" describing sounds prior to being processed by the insert effect, and "wet" describing the output of the insert effect. This parameter can be set anywhere from "Full Dry" to "Full Wet."
- Insert FX Bus: GlobalReverb Amt—The insert FX bus mix (described above) can be fed into the global reverb, so that reverb can be added to sounds processed by the insert effect. This parameter determines the amount of insert FX bus signal sent to the global reverb, and can be set anywhere from 0 to 127.

Each global reverb contains these context parameters:

- LightReverb FX Bus: Global Reverb Amt'—This parameter sets the amount of signal sent to the global reverb from the light reverb FX bus. This parameter may be set anywhere from 0 to 63.
- MediumReverb FX Bus: Global Reverb Amt'—This parameter sets the amount of signal sent to the global reverb from the medium reverb FX bus. This parameter may be set anywhere from 32 to 95.
- WetReverb FX Bus: Global Reverb Amt'—This parameter sets the amount of signal sent to the global reverb from the wet reverb FX bus. This parameter may be set anywhere from 64 to 127.
- Reverb: (selected reverb's name) Return Level—This parameter sets the level of the global reverb output. This can be used as an overall global reverb control that simultaneously raises or lowers the reverb volume for all of the reverb FX busses. Settings from 0 to 127 are available.

Insert Effect Real-Time Modulation Parameters

The ASR-X insert effects can be manipulated in real time, providing the opportunity for animated, expressive effect processing. This manipulation is achieved through the modulation of insert effect parameter settings, using a control device of your choosing on the selected sequence's insert control track. Each insert effect provides a set of parameters that allow you to set up real-time control of the effect.

Note: While a single track in each sequence—the insert control track—controls the real-time modulation of an insert effect, the changes made to the insert effect will be applied to any sounds routed to the insert FX bus.

- Insert: Mod Src—This parameter allows you to select a device with which the insert effect will be controlled. A wide range of devices is supported:

Off	There will be no effect modulation.
FullModAmt	The parameter being modulated will be set to its maximum amount.
Velocity	The parameter being modulated will respond to the quickness, or hardness, of keystrikes from the pads or received via MIDI.
Vel+Pressure	The parameter being modulated will respond to the quickness, or hardness, of keystrikes from the pads or received via MIDI, combined with received MIDI poly or channel pressure messages.
+PosMIDIkey#	The parameter being modulated will use the MIDI note number from the most recently struck pad or most recently received MIDI keystrike as its value setting, with 0 being interpreted as the lowest note of the MIDI range and 127 as the highest .
-NegMIDIkey#	The parameter being modulated will use the MIDI note number from the most recently struck pad or most recently received MIDI keystrike as its value setting, with 127 being interpreted as the lowest note of the MIDI range and 0 as the highest.
Pressure	The parameter being modulated will respond to received MIDI poly or channel pressure values.
PitchWheel	The parameter being modulated will respond to received MIDI pitch bend values.
ModWheel	The parameter being modulated will respond to received MIDI mod wheel (MIDI controller #1) values.
Wheel+Press	The parameter being modulated will respond to a combination of received MIDI mod wheel and poly or channel pressure values
FootPedal	The parameter being modulated will respond to received MIDI foot pedal (MIDI controller #4) values.
Sustain	The parameter being modulated will respond to sustain pedal presses produced by a foot switch connected to the ASR-X or received via MIDI.
Sostenuto	The parameter being modulated will respond to sostenuto pedal presses produced by a foot switch connected to the ASR-X or received via MIDI.
SysCTRL1-4	System Controllers 1-4 are system-wide user-designated real-time modulators (see Chapter 7 for further information).

- Insert: Mod Src Min and Insert: Mod Src Max—These two parameters allow you to establish a range of values from the control device to which the insert effect will respond. Each of these may be set anywhere from 000% to 100%.
- Insert: Mod Dest—This parameter allows you to choose the insert effect parameter you'd like to manipulate.
- Insert: Mod Dest Min and Insert: Mod Dest Max—These two parameters allow you to set limits for the amount of change that can be made to the setting of the parameter being modulated. If the Mod Dest Min value is set above than the Mod Dest Max value, response to the control device will be inverted: lower control device values will raise the parameter's setting, and vice versa.

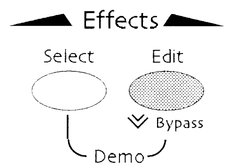
Note: To control an insert effect in real time using an external MIDI device, make sure to set your external control device to the same MIDI channel as the sequence's insert control track.

Setting a Sequence's Insert Control Track

You can designate any of a sequence's 16 tracks as the Insert Control Track, or you can turn the Insert Control Track feature off for the selected sequence.

To Set the Insert Control Track

1. Press the Edit button in the Effects section of the ASR-X front panel.



2. Turn the Parameter knob until the display shows:

```
Effects:
InsertCntrlTrack= 01
```

The number you see here may be different

3. Turn the Value knob to select an insert control track for the currently selected sequence, or select "off" to disable the feature.

Bypassing a Sequence's Effects

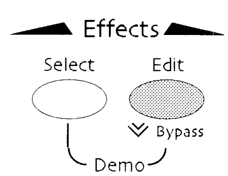
There may be times at which you'll find it useful to temporarily silence a sequence's insert effect or global reverb in order to hear a sound—or sounds—without the effect with which they're associated. This is accomplished by *bypassing* the effects. The ASR-X provides three methods of achieving this:

- You can quickly bypass both the insert effect and global reverb by rapidly double-clicking on the Effect section's Edit button.
- If either "Edit insert effect?" or "Edit global reverb?" are displayed, you can press the Edit Effect button a second time to bypass the displayed effect.
- You can use the Bypass parameter to bypass the insert effect, the global reverb, or both.

When an effect is bypassed, *BYPD* appears on all of the displays with which the effect is associated. If both effects are bypassed "*ALL-BYPASS*" is shown.

To Use the Bypass Parameter for Bypassing Effects

1. Press the Edit button in the Effects section of the ASR-X front panel.



2. Turn the Parameter knob until the display shows:

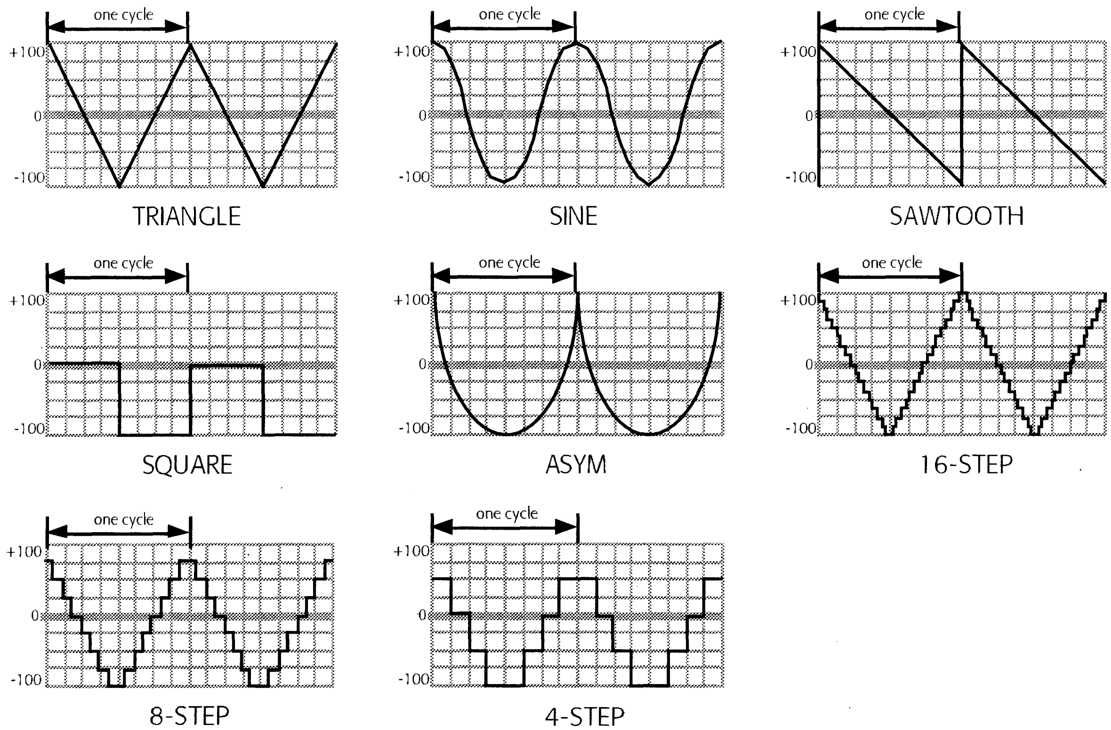
```
Effects:
Bypass=          None
```

The setting you see here may be different

3. Turn the Value knob to select the effect, or effects, you'd like to silence.

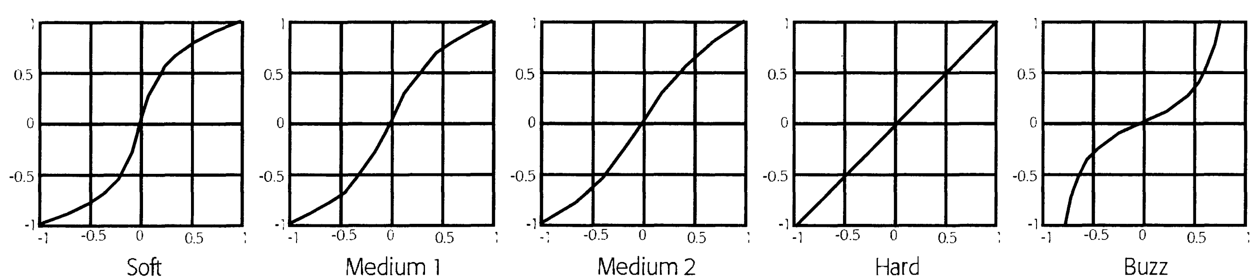
LFO Wave Shapes

Many insert effects have an LFO Shape parameter that determines how the LFO signal will rise or fall. There are eight possible values:



Distortion Curves

Many distortion-based insert effects have a Dist Curve parameter that determines the type of clipping produced by the distortion. There are five possible distortion curves:



Insert Effect Parameters

01 Parametric EQ

The Parametric EQ insert effect offers a minimum phase, four-band parametric EQ.

Parameter	Range	Description
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	Bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency parametric.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, used to control different bandwidths within the mid range.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.

02 Hall Reverb

03 Large Room

04 Small Room

Hall Reverb is a large acoustic space, and provides a high density reverb. Large Room reverb provides ambience, and Small Room reverb simulates the ambience and shorter decay times of a small space.

Parameter	Range	Description
Decay	0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay to a very low level after the input signal stops. Higher values are recommended for the hall reverb.
LF Decay	-99% to +99%	Functions as a tone control and boosts (when set to a positive value) or cuts (when set to a negative value) the rate at which low frequencies will decay.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.

HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass.
Primary Send	-99% to +99%	Controls the level of the diffused input signal into the reverb definition.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.
Detune Rate	0.00Hz to 1.54Hz	Controls the LFO rate of detuning introduced into the reverberation decay. Detuning creates a slight oscillating pitch shift into the decay, giving it a more natural sound by breaking up resonant nodes.
Detune Depth	0% to 100%	Controls the depth of the detuning, that is, how much the pitch will change. Low values yield a metallic sound. Some sounds may require very low values, while others sound more natural with higher values.
PreDelay	0 to 36ms	Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.
ER 1 Time	0 to 112ms	Controls the delay time for the first pre-echo. Pre-echoes are the first sounds reflected back from the walls or reflective "live" surfaces. Higher values delay the diffused signal more.
ER 1 Send	-99% to +99%	Controls the level of the first pre-echo, with the echo routed directly to the output. The sign denotes the phase of the echo.
ER 1 Level	-99% to +99%	Controls the level of the first pre-echo. This pre-level controls the echo send to the Definition.
ER 2 Time	0 to 112ms	Controls the delay time for the second pre-echo.
ER 2 Send	-99% to +99%	Controls the level of the second pre-echo, with the echo routed directly to the output.
ER 2 Level	-99% to +99%	Controls the level of the second pre-echo. As a signal continues to bounce off the different reflective surfaces (walls), it decreases in volume. Set this parameter to a lower value than Ref 1 Level, in order to create a natural sounding echo.
Position 1 Position 2 Position 3	-99% to +99% -99% to +99% -99% to +99%	These parameters simulate the depth of the hall. Think of them as three different microphones placing at various distances within the hall (Position 1 is closest to the front, and Position 3 is farthest from the front). When the range (volume) is higher for Position 1, the sound appears closer to the front, whereas a higher setting for Position 3 appears farther from the front, suggesting a deeper (wetter) hall. The sign denotes the phase of the echo.
Output Bal	Full <L to Full >R	Controls the left/right stereo balance of the reverb signal.

05 Large Plate

06 Small Plate

A plate reverb takes the vibrations from a metal plate and uses them to create a metallic sounding reverb. Large plate reverbs are often used to enhance a vocalist's performance, and small plate reverbs are often used in the studio for drums and percussion.

Parameter	Range	Description
Decay	0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops. High values of decay sound good on plate reverbs.
HF Damping	100Hz to 21.2kHz	Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy. Higher values yield an abrupt decay. This parameter controls the cut off of a low pass filter in series with the decay within the definition.
HF Bandwidth	100Hz to 21.2kHz	This parameter acts as a low pass filter on the output of the plate reverbs, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.
Diffusion 1	0 to 100	Smears the input signal to create a smoother sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear, making the echoes less apparent.
Diffusion 2	0 to 100	This diffuser, similar to and in series with the previous one, offers control over lower frequency ranges. Plate reverbs tend to sound metallic, and the diffusers help to smear the signal, eliminating the metallic sound.
Definition	0 to 100	Controls the rate at which echo density increases with time. Higher values can cause the echo density to build at a rate that exceeds the decay rate. For the best results, try to select the highest value that works with your sound source.
PreDelay	0 to 36ms	Controls the amount of time it takes for the input signal to be presented to the plate reverb. A value of 0 would offer no delay.
ER 1 Level ER 2 Level ER 3 Level ER 4 Level	-99% to +99% -99% to +99% -99% to +99% -99% to +99%	Control four early reflection levels. Setting these levels to lower values will produce a wetter sound. These four reflection levels are close to the input of the Definition
Output Bal	Full <L to Full >R	Controls the left/right stereo balance of the plate reverb signal.

07 NonLinReverb1

08 NonLinReverb2

Non linear reverbs can be used to obtain blooming reverb, gated reverb, reverse reverb and early reflections. In general, they do not produce an exponentially decaying reverb. Unlike the hall, room and plate reverbs, NonLinReverb1 and 2 pass the input signal through the reverb diffusers only once. For this reason the reverb diffusers are called density, to distinguish them from the other reverb diffusers (called definition). Density controls the amount of echo density, as opposed to the rate of increase of echo density. The NonLin Reverbs purposely impose a coloration on the resulting sound.

Parameter	Range	Description
Env 1 Level Env 2 Level Env 3 Level Env 4 Level Env 5 Level Env 6 Level Env 7 Level Env 8 Level Env 9 Level	-99% to +99% -99% to +99% -99% to +99% -99% to +99% -99% to +99% -99% to +99% -99% to +99% -99% to +99% -99% to +99%	These parameters control the output tap levels sequenced in time across the density from input to output. Envelope Level 1 is tapped right after the diffusers and before the echoes. If this is undesirable, set Envelope Level 1 to 0%. Envelope Levels 8 and 9 are positioned at the very end of the Density setting these too high can cause excessive ringing. Envelope Levels 8 and 9 are also very dry. Set all nine tap levels to find the envelope for your application. We recommend the average Envelope Level not to exceed a value of ±45% to prevent overdriving these reverbs.
HF Damping	100Hz to 21.2kHz	The HF Damping is located within the density. This parameter selects the amount of high frequency energy to be filtered out.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth parameter acts as a low pass filter on the output signal, controlling the amount of high frequencies that will be heard. The higher the setting, the more high frequencies are heard.
Primary Send	-99% to +99%	Controls the level of the diffused input signal which is nearly instantaneous with respect to the input. This signal is injected directly into the Density at the specified level.
Diffusion 1	0 to 100	This parameter smears the input signal transients of higher frequency ranges. Higher values are recommended for smoother decay. Very low values will give a highly repetitive echo-like sound. Diffusion 1 and 2 exist within each diffuser block.
Diffusion 2	0 to 100	Diffusion 2 is similar to Diffusion 1, but offers control of lower frequencies. In general a setting of 50 can be considered an equal mix of dry/diffused sound. This setting is a good starting point.
Density 1	0 to 100	Density 1 controls the number of echoes.
Density 2	0 to 100	Density 2 controls the number of echoes in a lower frequency range. In general, to get the smoothest sound, Density 2 is usually less than the value of Density 1.
ER 1 Time	0 to 112ms	Controls the amount of time it takes for the first pre-echo to be injected into the density. Pre-echoes are the sounds which have been reflected back from the walls or other reflective surfaces.
ER 1 Send	-99% to +99%	This parameter controls the level of the first pre-echo.
ER 2 Time	0 to 112ms	This controls the amount of time it takes for the second pre-echo to be injected into the density.
ER 2 Send	-99% to +99%	This parameter controls the level of the second pre-echo. Experiment with both positive and negative on all echoes to change the tonal character of the results.
Output Bal	Full <L to Full >R	Controls the left/right stereo balance of the reverb signal.

09 Gated Reverb

When the output of a reverb is muted partway through its decay, it creates a gated sound. To achieve this gated effect, the gated reverb must gate a number of internal parameters, not just the output amplitude envelope. It is, however, the output amplitude over which you have control. The ASR-X offers a highly controllable gated reverb, optimized for percussive instruments, but useful for any sound. The gated reverb triggers whenever the input signal exceeds a user programmable threshold. This trigger threshold should be set as low as possible, so that none of the input signal is missed. The gate will stay open as long as the input signal remains above the threshold, and all the input signals will be accumulated under this gate until the total input signal level falls below the hysteresis level. When this happens, the hold time will begin. The reason for hysteresis is to eliminate false retriggering and to ensure precise hold time durations. If you desire a separate gate on each and every note, use the Non Lin reverbs.

Parameter	Range	Description
Gate Thresh	-96.0dB to 0.0dB	Sets the signal level that triggers the gated reverb. When the incoming signal reaches this value, it triggers (starts) the gated reverb. Higher values would require a stronger incoming signal. Set this parameter as low as possible to work with your particular source, but not so low as to cause false triggering.
Gate Hysteresis	0dB to 48dB	Sets the lower threshold level relative to Gate Thresh below which the Gate Hold Time begins. If the difference between Gate Thresh and Gate Hysteresis is lower than the level of the incoming signal, the gated reverb will continue to retrigger. With a high decay rate, this adds a cavernous quality to percussion instruments.
Gate Attack	50us to 10.0s	Sets the attack time of the gated reverb once the incoming signal has reached the trigger level. Generally the attack should be short and not set longer than the Gate Hold time.
Gate Release	50us to 10.0s	Sets the amount of time after the Gate Hold time has elapsed for the gated reverb to shut down. Generally these times are very short.
Gate Hold	50us to 10.0s	Sets the amount of time that the reverb will hold after the retrigger and before the release. The Gate Hold time will begin again if retriggered.
Decay	0sec to 10.0sec	Sets the decay rate. In general, the decay rate is set very high.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverb. Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy.
Diffusion 1	0 to 100	Smears the transients, so as to diffuse and smooth the sound. Lower values will cause impulsive sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding). Recommended setting is approximately 50.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Recommended setting is approximately 50.
Definition	0 to 100	Controls the rate of echo density build up in the reverb decay. If set too high, the echo density will build at a rate that exceeds the decay rate. A general rule of thumb: Definition should not exceed the Decay Rate. We recommend settings between 25 and 50.
Slap Time	0ms to 108ms	Controls the delay time of an internal dry stereo signal to create a slapback. In general, the slapback is greater or equal to the Gate Hold time to achieve a reverse effect.
Slap Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the slapback (internal dry) signal.
ER 1 Level ER 1 Level ER 1 Level ER 1 Level	-99% to +99% -99% to +99% -99% to +99% -99% to +99%	These parameters control four early reflection levels. Setting these levels to lower values will produce a wetter sound. A setting of 0% turns the early reflections off.
Output Bal	Full <L to Full >R	Controls the left/right stereo balance of the gated reverb signal.

10 Stereo Chorus

This stereo chorus uses delays to produce pitch and amplitude modulation.

Parameter	Range	Description
LFO Rate	0.0Hz to 20.0Hz	Controls the rate of pitch modulation applied to the delays.
Chorus Depth	0.0ms to 25.0ms	Controls the excursion of modulation. As this parameter increases, the amount of detuning also increases.
ChorusCenter	0.0ms to 50.0ms	Controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the chorus.
Spread	(wide stereo to mono, to reversed image)	Offers control of the synthesized stereo field. The farthest counterclockwise setting of the Value knob offers true stereo, the middle setting forces the left & right into the center (mono), and turning the Value knob fully clockwise inverts the left & right signal.
Chorus Phase	0deg or -180deg	Controls the relative phase between left and right LFOs.

11 8-VoiceChorus

8-Voice Chorus offers a symphonic chorused sound having eight different voices and using eight separately randomized LFOs. This effect is good for creating an ensemble of instruments from single sources (there is no internal filtering applied to any of the chorused voices).

Parameter	Range	Description
EQ Input	Off, -49.5dB to +24dB	Adjusts the input volume of the EQs to eliminate the possibility of clipping boosted signals.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency band.
Mid 1 Q	1.0 to 40.0	Bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. Raising the value will produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
Dry Blend	Full Dry to Full Wet	Controls the dry to wet mix of the chorus .
HPF Cutoff	10Hz to 10.9kHz	Controls the cutoff frequency of the high pass filter frequency applied to the input signal.
LFO Rate	0.0Hz to 7.0Hz	Controls the rate of pitch modulation applied to the delays.
Chorus Depth	0.0ms to 300ms	Controls the excursion of modulation.
ChorusCenter	0.0ms to 300.0ms	Controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the chorus.
Center Offset	0% to 100%	Controls the relative spacing in nominal delay time among the eight voices. 100% is the maximum setting.
Chorus Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Chorus Feedback	-99% to +99%	Controls the amount of feedback applied to the chorus. Positive settings are in-phase, negative values are out-of-phase, and impart a different tonality to the chorus.

12 Rev→Chorus

Combines a plate reverb with a stereo chorus.

Parameter	Range	Description
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.
Chorus Mix	Full Dry to Full Wet	Controls the dry/wet mix of the chorus.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of pitch modulation to the chorus.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left & right LFOs.
Chorus Depth	0.0ms to 25.0ms	Controls the amount of modulation.
Chorus Center	0.0ms to 50.0ms	Controls the delay times within the chorus. Adjusting this parameter will change the tonal character of the chorus.
System Feedback	-99% to +99%	Controls the amount of feedback applied from the output of the chorus to the input of the reverb.

13 Rev→Flanger

This insert effect features a plate reverb with a flanger effect.

Parameter	Range	Description
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverb to decay after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass.

Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.
FlangerMix	Full Dry to Full Wet	Controls the dry/wet mix of the flanger.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of modulation applied to the flanger.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Flanger Depth	0.0ms to 25.0ms	Controls the range of the high-to-low frequency sweep in the flanger effect.
FlangerCenter	0.0ms to 50.0ms	Sets the sweep mid-point of the flanger effect.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the flanger.
Feedback	-99% to +99%	Controls the amount of feedback applied to the flanger. Positive or negative values will impart a different tonality to the flange effect, either accenting the peaks or the notches.
System Feedback	-99% to +99%	Controls the amount of feedback applied from the output of the flanger to the input of the reverb.

14 Rev→Phaser

Combines a plate reverb with a 12-pole phase shifter.

Parameter	Range	Description
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.
Phaser Mix	Full Dry to Full Wet	Controls the dry/wet mix of the phaser.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the phaser.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
Phaser Depth	0 to 100	Controls the amount of modulation applied to the phaser.
Phaser Center	0 to 100	This parameter controls the mid-point of the phaser.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the phaser. This parameter should normally be set to 100%.
Feedback	-99% to +99%	Controls the amount of feedback applied to the phaser. Positive or negative values will impart a different tonality to the phaser effect, either accenting the peaks or the notches.

15 Chorus→Rev

Chorus→Rev combines a rich sounding chorus with the standard reverb.

Parameter	Range	Description
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the delay time of the chorus.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Chorus Depth	0.0ms to 25.0ms	Controls the amount of modulation.
Chorus Center	0.0ms to 50.0ms	Controls the four delay times within the chorus. Adjusting this parameter will change the tonal character of the chorus.
Rev Mix	Full Dry to Full Wet	Controls the dry/wet mix of the reverb.
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.

16 Flanger→Rev

This insert effect features a flanger combined with a plate reverb.

Parameter	Range	Description
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the flange effect.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Flanger Depth	0.0ms to 25.0ms	Controls the range of the high-to-low frequency sweep in the flanger effect.
FlangerCenter	0.0ms to 50.0ms	Sets the sweep mid-point of the flanger effect.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the flanger. This parameter should be set to 100% for maximum effect.
Feedback	-99% to +99%	Controls the amount of feedback applied to the flanger. Positive or negative values will impart a different tonality to the flange effect, either accenting the peaks or the notches.
Rev Mix	Full Dry to Full Wet	Controls the dry/wet mix of the reverb.
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay to a very low level after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.

17 Phaser→Rev

A 12-pole phase shifter with reverb.

Parameter	Range	Description
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the phaser.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
Phaser Depth	0 to 100	Controls the amount of modulation applied to the phaser.
Phaser Center	0 to 100	This parameter controls the mid-point of the phaser.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the phaser. This parameter should normally be set to 100%.
Feedback	-99% to +99%	Controls the amount of feedback applied to the phaser. Positive or negative values will impart a different tonality to the phaser effect, either accenting the peaks or the notches.
Rev Mix	Full Dry to Full Wet	Controls the dry/wet mix of the reverb.
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverberation to decay to a very low level after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.

18 EQ→Reverb

A parametric EQ with reverb.

Parameter	Range	Description
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.

Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
Rev Mix	Full Dry to Full Wet	Controls the reverb mix.
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverb to decay after the input signal stops.
HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverb. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	Acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass. The higher the setting, the more high frequencies are allowed to pass.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, controls lower frequency ranges.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this too high can cause the echo density to build at a rate which exceeds the decay rate.

19 Spinner→Rev

Combines a pseudo-three dimensional panner with the standard reverb.

Parameter	Range	Description
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of modulation applied to the spinner.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between the left and right and front and back LFOs. Set this to ±90 deg for circular motion.
DDL Mod Depth	0.0ms to 10.0ms	Controls the left to right mod depth of delay time. Try setting this to 0.3 ms for average head size.
DDL ModCenter	0.0ms to 50.0ms	Fixed delay time.
Level Mod	0% to 100%	Left to right LFO mod depth to level.
L-to-R Mod	0% to 100%	Left to right LFO mod depth to filter.
F-to-B Mod	0% to 100%	Front to back LFO mod depth to filter. If the sum of the L-to-R Mod and F-to-B Mod is greater than 100%, the filter can "thump" as it closes down.
Cancellation	-99% to +99%	Sets the depth and phase of the opposite speaker cancellation signal.
Rev Mix	Full Dry to Full Wet	Controls the dry/wet mix of the reverb.
Decay	0.0sec to 10.0sec	Controls the amount of time it takes for the reverb to decay after the input signal stops.

HF Damping	100Hz to 21.2kHz	Controls the rate of attenuation of high frequencies in the decay of the reverberation. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.
HF Bandwidth	100Hz to 21.2kHz	Acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass. The higher the setting, the more high frequencies are allowed to pass.
Diffusion 1	0 to 100	Smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.
Diffusion 2	0 to 100	This parameter, similar to and in series with Diffusion 1, controls lower frequency ranges.
Definition	0 to 100	Controls the rate at which echo density is increased with time. Setting this too high can cause the echo density to build at a rate which exceeds the decay rate.

20 DDL→Chorus

DDL→Chorus combines four independent, controllable digital delays with a chorus.

Parameter	Range	Description
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly3 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the third independent delay.
Dly3 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly3 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly4 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the fourth independent delay.
Dly4 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.

Dly4 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the four rates of the modulation applied to the delay time of the chorus.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Chorus Depth	0.0ms to 25.0ms	Controls the amount of modulation.
ChorusCenter	0.0ms to 50.0ms	Controls the delay time within the chorus, and changes the tonal character.
Spread	(wide stereo to mono)	This parameter offers control of the synthesized stereo field. The farthest counterclockwise setting of the Value knob offers true stereo, the middle setting forces the left and the right into the center (mono), and turning the Value knob fully clockwise inverts the left and right signal.

21 DDL→Flanger

Combines four independent controllable digital delays with a flanger.

Parameter	Range	Description
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly3 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the third independent delay.
Dly3 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly3 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly4 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the fourth independent delay.
Dly4 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly4 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the flange effect.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Flanger Depth	0.0ms to 25.0ms	Controls the range of the high-to-low frequency sweep in the flanger effect.
FlangerCenter	0.0ms to 50.0ms	Sets the sweep mid-point of the flanger effect.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the flanger. This parameter should be set to 100% for maximum effect.
Feedback	-99% to +99%	Controls the amount of feedback applied to the flanger. Positive or negative values will impart a different tonality to the flange effect, either accenting the peaks or the notches.

22 DDL→Phaser

Combines a digital delay with a phase shifter.

Parameter	Range	Description
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly3 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the third independent delay.
Dly3 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly3 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly4 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the fourth independent delay.
Dly4 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly4 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the phaser.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
Phaser Depth	0 to 100	Controls the amount of modulation applied to the phaser.
Phaser Center	0 to 100	This parameter controls the mid-point of the phaser.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the phaser. This parameter should normally be set to 100%.
Feedback	-99% to +99%	Controls the amount of feedback applied to the phaser. Positive or negative values will impart a different tonality to the phaser effect, either accenting the peaks or the notches.

23 DDL→EQ

Combines a digital delay with a parametric EQ.

Parameter	Range	Description
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly3 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the third independent delay.
Dly3 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly3 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly4 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the delay time for the fourth independent delay.
Dly4 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly4 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.

LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.

24 Multi-Tap DDL

Multi-Tap DDL offers four diffusers in series feeding a nine-tap digital delay.

Parameter	Range	Description
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQ to eliminate the possibility of clipping boosted signals.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Diffusion 1	-99% to +99%	Sets the amount and phase of the first diffuser.
Diffus Time 1	0ms to 62ms	Sets the delay time of the first diffuser.
Diffusion 2	-99% to +99%	Sets the amount and phase of the second diffuser.
Diffus Time 2	0ms to 62ms	Sets the delay time of the second diffuser.
Diffusion 3	-99% to +99%	Sets the amount and phase of the third diffuser.
Diffus Time 3	0ms to 62ms	Sets the delay time of the third diffuser.
Diffusion 4	-99% to +99%	Sets the amount and phase of the fourth diffuser.
Diffus Time 4	0ms to 62ms	Sets the delay time of the fourth diffuser.
Dly Interval	Uniform, Linear+, Linear-, Expon.+, Expon.-, Random	Controls the spacing of the taps within the DDL.
MaxDlyTime	1/1 Sys to 1/32 Sys, 0ms to 500ms	Controls the maximum delay time.
Dly Smoothing	0ms to 500ms	Controls the amount of time it takes to change from one Dly Max Time setting to another. Low values result in more clicking but less detuning. High values result in less clicking but more detuning.
Feedback Tap	1 to 9	Selects one of the nine taps to be fed back into the input of the effect.
Dly Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly Damping	10Hz to 20.0kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.

Dly Levels	Uniform, Linear+, Linear-, Expon.+, Expon.-, Random	Controls the relative levels of the taps.
Dly Max Level	0 to 100	Controls the maximum level that any one tap can attain.
Dly Pan	Centered, Alternating, L->R, R->L, Center->Out, Out->Center, Random	Controls the panning of the taps in the stereo field.
Dly Spread	0 to 100	Controls the width of the stereo field. A setting of 0 is the narrowest (mono)—a setting of 100 is the widest (full stereo).

25 Dist→Chorus

Dist→Chorus combines a distortion with a chorus.

Parameter	Range	Description
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn Dist Volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if Distortion Gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cutoff frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency band.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.

HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the four rates of the modulation applied to the delay time of the chorus.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Chorus Depth	0.0ms to 25.0ms	Controls the amount of modulation.
ChorusCenter	0.0ms to 50.0ms	Controls the delay times within the chorus. Adjusting this parameter will change the tonal character of the chorus.
Spread	(wide stereo to mono)	This parameter offers control of the synthesized stereo field. The farthest counterclockwise setting of the Value knob offers true stereo, the middle setting forces the left and the right into the center (mono), and turning the Value knob fully clockwise inverts the left and right signal.

26 Dist→Flanger

Dist→Flanger combines a distortion with a flanger.

Parameter	Range	Description
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.

Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the flange effect.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Flanger Depth	0.0ms to 25.0ms	Controls the range of the high-to-low frequency sweep in the flanger effect.
FlangerCenter	0.0ms to 50.0ms	Sets the sweep mid-point of the flanger effect.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the flanger. This parameter should be set to 100% for maximum effect.
Feedback	-99% to +99%	Controls the amount of feedback applied to the flanger. Positive or negative values will impart a different tonality to the flange effect, either accenting the peaks or the notches.

27 Dist→Phaser

This insert effect combines a raspy distortion with a phase shifter.

Parameter	Range	Description
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.

LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the phaser.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
Phaser Depth	0 to 100	Controls the amount of modulation applied to the phaser.
Phaser Center	0 to 100	This parameter controls the mid-point of the phaser.
Notch Depth	0% to 100%	Controls the depth of the peaks and notches produced by the phaser. This parameter should normally be set to 100%.
Feedback	-99% to +99%	Controls the amount of feedback applied to the phaser. Positive or negative values will impart a different tonality to the phaser effect, either accenting the peaks or the notches.

28 Dist→AutoWah

Dist→AutoWah combines a voltage control filter and a raspy distortion, and a second voltage controlled filter. Three effects can be obtained: distortion, wah-wah, and auto-wah. The last two functions use the same VCF. These filters can be disabled or used as EQ if desired. There is a second VCF that exists after the distortion that can be set to act like a simple speaker simulator.

Parameter	Range	Description
Pre HPF Fc	10Hz to 1.50kHz	Filters out the low frequencies before the EQ. The higher the value, the less low frequencies will pass through.
Pre VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cutoff frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Pre VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.

PreVCF EnvAmt	-99% to +99%	Determines how much the amplitude of the incoming signal will modify the distortion filter cutoff frequency. When set to 0, no modification will occur. When set to mid positive values, the Pre-VCF Fc will go high, but then come down to its nominal setting. When set to negative mid values, the Pre-VCF Fc will go low, and then go back up to its nominal setting. How quickly it does so is determined by the Attack and Release parameters. This sound is the auto-wah. Positive values will boost the high frequencies, and negative values will cut the high frequencies.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Distortion	Off, On	Chooses between distorted and clean signals.
Post VCF Fc	10Hz to 7.10kHz	Determines the second distortion filter cutoff frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, this parameter controls the sharpness of the peak.
PostVCF EnvAmt	-99% to +99%	Determines how much the amplitude of the incoming signal will modify the distortion filter cutoff frequency. When set to 0, no modification will occur. When set to mid positive values, the Pre-VCF Fc will go high, but then come down to its nominal setting. When set to negative mid values, the Pre-VCF Fc will go low, and then go back up to its nominal setting. How quickly it does so is determined by the Attack and Release parameters.
VCF Attack	50us to 10.0s	Sets the attack of the envelope follower (i.e., determines how closely the attack is followed) once the incoming signal has been detected. Generally the attack should be short.
VCF Release	50usto 10.0s	Sets the amount of time after the incoming signal has ceased for the envelope follower to shut down. Generally these times are longer than the attack times.
Post HPF Fc	10Hz to 1.50kHz	Filters out the low frequencies after the distortion.

29 ResVCF→DDL

ResVCF→DDL combines a voltage control filter and a digital delay.

Parameter	Range	Description
VCF Input	Off, -49.5dB to 0.0dB	Acts as a trim control at the input of the VCF.
VCF Fc	10Hz to 7.10kHz	Determines the VCF cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah wah pedal effect.
VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
ADSR Attack	50us to 10.0s	Sets the attack time for the ADSR envelope shape.
ADSR Decay	50us to 10.0s	Sets the decay time for the ADSR envelope shape.

ADSR Sustain	Off, -49.5dB to 0.0dB	Sets the sustain level for the ADSR envelope shape.
ADSR Release	50us to 10.0s	Sets the release time for the ADSR envelope shape.
ADSR Env Amt	-99% to +99%	Determines the degree to which the envelope modifies the cutoff frequency of the VCF.
ADSR TrigMode	Single or Multi	Determines whether the envelope which controls the VCF will retrigger with each key-event (Multi) or not (Single).
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

30 Dist→VCF→DDL

Dist→VCF→DDL combines a distortion, a voltage control filter and a digital delay.

Parameter	Range	Description
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.

Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
VCF Input	Off, -49.5dB to 0.0dB	Acts as a trim control at the input of the VCF.
VCF Fc	10Hz to 7.10kHz	Determines the VCF cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
ADSR Attack	50us to 10.0s	Sets the attack time for the ADSR envelope shape.
ADSR Decay	50us to 10.0s	Sets the decay time for the ADSR envelope shape.
ADSR Sustain	Off, -49.5dB to 0.0dB	Sets the sustain level for the ADSR envelope shape.
ADSR Release	50us to 10.0s	Sets the release time for the ADSR envelope shape.
ADSR Env Amt	-99% to +99%	Determines the degree to which the envelope modifies the cutoff frequency of the VCF.
ADSR TrigMode	Single or Multi	Determines whether the envelope which controls the VCF will retrigger with each key-event (Multi) or not (Single).
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

31 Pitch Detuner

Pitch Detuner allows you to change the pitch of a sound to any pitch within a range of two octaves in either direction. We recommend using this insert effect as an LFO-controlled detuner.

Parameter	Range	Description
Voice1 Semi	-24 semi to +24 semi	Allows you to adjust the pitch of voice 1 up to two octaves above or below the original pitch in semi-tones (half steps).
Voice1 Fine	-100cent to +100cent	This parameter allows you to fine tune the pitch of voice 1.
Voice1 Level	Off, -49.5dB to 0.0dB	Adjusts the volume of voice 1.
Voice1 Regen	-99% to +99%	Controls the amount of feedback from the output of the pitch detuner back into the input. This allows you to create special effects with ascending/descending delays.
Voice1 Width	1ms to 185ms	Controls the splice width of voice 1. Select the width that sounds best to you. Shorter values result in a grainier sound, while longer values sound smoother.
Voice1 Mod	0% to 100%	Controls the amount of modulation applied to voice 1.
Voice2 Semi	-24 semi to +24 semi	Allows you to adjust the pitch of voice 2 up to two octaves above or below the original pitch in semi-tones (half steps).
Voice2 Fine	-100cent to +100cent	This parameter allows you to fine tune the pitch of voice 2.
Voice2 Level	Off, -49.5dB to 0.0dB	Adjusts the volume of voice 2.
Voice2 Regen	-99% to +99%	Controls the amount of feedback from the output of the pitch detuner back into the input. This allows you to create special effects with ascending/descending delays.
Voice2 Width	1ms to 185ms	Controls the splice width of voice 2. Select the width that sounds best to you. Shorter values result in a grainier sound, while longer values sound smoother.
Voice2 Mod	0% to 100%	Controls the amount of modulation applied to voice 2.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	This parameter controls the rate of pitch modulation which creates a chorusing effect. To achieve chorusing, this rate must be very low.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Regen Time	1/1 Sys to 1/32 Sys, 0ms to 185ms	Controls the amount of delay in the feedback path.

32 Chatter Box

This insert effect uses a formant filter with a time-varying spectrum to impart a dynamic vocal-like quality to almost any sound. Two LFOs are combined such that the filter morphs between four vowel shapes that you select. The first LFO is also tied to an auto-panner, which can bounce the vocalized signal through stereo space. Finally, a digital delay can be used to create highly unusual talking echo effects.

Parameter	Range	Description
VCF Input	Off, -49.5dB to 0.0dB	Trims the input to the formant filter so that clipping does not occur.
VCF Dry Amt	Off, -49.5dB to 0.0dB	Controls the level of the DDL signal to be mixed with the output of the formant filter.
Shape 1	A, E, I, O, U, AA, AE, AH, AO, EH, ER, IH, IY, UH, UW, B, D, F, G, J, K, L, M, N, P, R, S, T, V, Z	Select the shape of the first formant filter.

Shape 2	A, E, I, O, U, AA, AE, AH, AO, EH, ER, IH, IY, UH, UW, B, D, F, G, J, K, L, M, N, P, R, S, T, V, Z	Select the shape of the second formant filter.
Shape 3	A, E, I, O, U, AA, AE, AH, AO, EH, ER, IH, IY, UH, UW, B, D, F, G, J, K, L, M, N, P, R, S, T, V, Z	Select the shape of the third formant filter.
Shape 4	A, E, I, O, U, AA, AE, AH, AO, EH, ER, IH, IY, UH, UW, B, D, F, G, J, K, L, M, N, P, R, S, T, V, Z	Select the shape of the fourth formant filter.
FormantWarp	-12 to +12 semi	Shifts all formant frequencies up or down, warping the "size" of the formant filter.
AutoPan Depth	0% to 100%	Controls the depth of the auto-panning function after the formant filter.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	This parameter controls the rate of pitch modulation which creates a chorusing effect. To achieve chorusing, this rate must be very low.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO 2 Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	This parameter controls the rate of the second LFO.
LFO 2 Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the second LFO will use for pitch modulation.
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.

33 Formant Morph

This effect is similar to the Chatter Box, except that it has a distorter for increased harmonic content, and it uses a single LFO to morph between two vowel shapes that you select.

Parameter	Range	Description
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.

Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Post VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
VCF Input	Off, -49.5dB to 0.0dB	Trims the input to the formant filter so that clipping does not occur.
VCF Dry Amt	Off, -49.5dB to +24dB	Controls the level of the distortion/DDL signal to be mixed with the output of the formant filter.
Shape 1	A, E, I, O, U, AA, AE, AH, AO, EH, ER, IH, IY, UH, UW, B, D, F, G, J, K, L, M, N, P, R, S, T, V, Z	Selects the shape of the first formant filter.
Shape 2	A, E, I, O, U, AA, AE, AH, AO, EH, ER, IH, IY, UH, UW, B, D, F, G, J, K, L, M, N, P, R, S, T, V, Z	Selects the shape of the second formant filter.
FormantWarp	-12 to +12 semi	Shifts all formant frequencies up or down, warping the "size" of the formant filter.
AutoPan Depth	0% to 100%	Controls the depth of the auto-panning function after the formant filter.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	This parameter controls the rate of pitch modulation which creates a chorusing effect. To achieve chorusing, this rate must be very low.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for pitch modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.

Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.

34 RotarySpeaker

This insert effect adds the famous, classic rotating speaker effect to any sound. A tunable distortion is added to the input signal and is also passed through the rotors.

Parameter	Range	Description
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the distortion filter cut off frequency. Higher values have a brighter sound. This parameter can be modulated by any continuous MIDI controller for a wah-wah pedal effect.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency parametric.
Speed	Slow or Fast	Selects one of the two available rotor speeds, whose rates are determined by the Hi Slow, Hi Fast, Lo Slow, and Lo Fast parameters. The behavior of this switch accurately reflects an actual rotary speaker, taking time to speed up or slow down, based on the values of the inertia parameters. By assigning a modulation controller to this parameter, you can change between the slow and fast speeds in real time.
Spread	Stereo or Mono	Selects either a stereo or mono rotary speaker effect.
Crossover Fc	10Hz to 20.0kHz	Sets the crossover frequency between the low and high rotors.
Lo Hi Bal	Full <Lo to Full >Hi	Controls the volume balance between the low and the high rotor.
Rotor Mix	Full Dry to Full Wet	Controls the balance between the leakage (dry) signal and the rotor (wet) signal. We recommend settings near 70.0% wet.

Hi Inertia	100ms to 10.0s	Determines how long it will take for the horn portion of the rotor effect to speed up to the fast setting after switching from slow or vice versa. Adjust this parameter to simulate the effect of the rotary speaker gradually picking up speed.
Hi Slow	0.0Hz to 10.0Hz	Sets the speed of the horn rotor simulator when Speed=Slow. A real organ speaker cabinet has two sets of speakers (horns & woofer). This parameter is used to set the horn's rate.
Hi Fast	0.0Hz to 10.0Hz	Sets the speed of the horn rotor simulator when Speed=Fast.
Hi FM Min	0 to 100	Sets the amount of detuning for the horn rotor simulator when the Speed parameter is set to "Slow."
Hi FM Max	0 to 100	Sets the amount of detuning for the horn rotor simulator when the Speed parameter is set to "Fast." TheHi FM Min and Hi FM Max parameters can create "Doppler" effect.
Hi AM Min	0 to 100	Sets the amount that the horn rotor simulator volume will change as the speaker rotates when the Speed parameter is set to "Slow."
Hi AM Max	0 to 100	Sets the amount that the horn rotor simulator volume will change as the speaker rotates when the Speed parameter is set to "Fast." Broader ranges between Hi AM Max and Hi AM Min will create a deeper rotating speaker effect.
Lo Inertia	100ms to 10.0s	Determines how long it will take for the woofer simulator rotor Speed effect to slow down to the low setting after switching from Fast or vice versa. Adjust this parameter to simulate the effect of the rotary speaker gradually slowing down.
Lo Slow	0.0Hz to 10.0Hz	Sets the speed of the bass woofer's rotor simulator when Speed=Slow. A real organ speaker cabinet has two sets of speakers (horns & woofer). This parameter is used to set the woofer's rate.
Lo Fast	0.0Hz to 10.0Hz	Sets the speed of the bass woofer's rotor simulator when Speed=Fast.
Lo FM Min	0 to 100	Sets the amount of detuning for the woofer rotor simulator when the Speed parameter is set to "Slow."
Lo FM Max	0 to 100	Sets the amount of detuning for the woofer rotor simulator when the Speed parameter is set to "Fast." The Lo FM Min and Lo FM Max parameters can create a "Doppler" effect.
Lo AM Min	0 to 100	Sets the amount that the volume of the woofer rotor simulator will change as the speaker rotates when the Speed parameter is set to "Slow."
Lo AM Max	0 to 100	Sets the amount that the volume of the woofer rotor simulator will change as the speaker rotates when the Speed parameter is set to "Fast." Broader ranges between these two parameters will create a deeper rotating speaker effect.

Speed Control	Normal or Toggle	Allows you to select a modulator and define what type of modulation you want to use to affect the rotor speed. The two modulation modes are: • Normal — The modulation source continuously switches between the Speed slow and fast setting, based on the mod source position and/or movement. Try this setting with a Mod Wheel — you'll hear the rotary speaker change speed based on the position of the wheel (and the speed settings). • Toggle — The modulation source toggles the rotor speed between the Speed parameter's slow and fast setting. Every time the modulation source moves from zero in a positive direction, the rotating speaker effect changes speeds from slow to fast or vice versa. Try this setting with a Sustain pedal. With both types of modulation, the rotary speaker always takes the inertia time to get to the rotor speed slow and fast settings.

35 Tunable Spkr

This insert effect offers an EQ controllable speaker sound. By tuning three parametric filters, you can simulate many different speaker cabinet sounds that are used in all styles of music.

Parameter	Range	Description
Pre HP Fc	10Hz to 1.50kHz	Controls the boost or cut of the high pass filter frequency applied to the input signal.
EQ Input	Off, -49.5dB to +24dB	This parameter allows you to adjust the input level before the EQs to eliminate the possibility of clipping boosted signals.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid-frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
Mid 3 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 3 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 3 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
EQ Output	Off, -49.5dB to +24dB	Since speaker cabinets are "lossy," output gain is required to compensate losses in perceived volume. Setting this gain too high will cause clipping of the output signal.
HPF Cutoff	10Hz to 20.0kHz	Filters out the low frequencies. The higher the value, the less low frequencies pass through. This parameter is used to increase brightness.
LPF Cutoff	10Hz to 20.0kHz	Controls the boost or cut of the low pass filter frequency applied to the input signal.

36 Guitar Amp

This insert effect recreates the warm sound of a tube guitar amplifier. It does this by emulating tube distortion characteristics.

Parameter	Range	Description
Pre HP Fc	10Hz to 1.50kHz	Filters out the low frequencies before the preamp. The higher the value, the less low frequencies pass through.
Pre EQ Trim	Off, -49.5dB to +24dB	Controls the input level to the pre-amp EQ to eliminate the possibility of clipping boosted signals.
Pre EQ Fc	10Hz to 20.0kHz	Determines the center frequency of the parametric filter before the preamp. Higher values have a brighter sound.
Pre EQ Q	1.0 to 40.0	Determines the width of the resonant peak at the parametric filter center frequency. While the filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the sharpness of the peak.
Pre EQ Gain	Off, -49.5dB to +24dB	Adjusts the amount of boost or cut applied to the parametric filter in front of the preamp.
Preamp Gain	Off, -49.5dB to +24dB	Adjusts the amount of boost or cut applied to the incoming signal. This parameter can be thought of as the primary distortion stage (clipping). We recommend a setting of 0 dB, since these emulations were optimized for distortion there. Lower preamp gains will result in less distortion, while higher preamp gains will yield clipping distortion. For low preamp gain, it may be desirable to use low tube bias values.
Master Level	Off, -99dB to 0.0dB	This parameter controls the output level of the main amp.
Tube Bias	0 to 100	For preamp gains approximately 0 dB, this parameter controls the emphasis of even to odd harmonics which determines the tone of the amp. Mid values emphasize even harmonics and offer a warmer "glowing tube" sound, while the highest values may sound like tubes going bad. Tube bias and preamp gain are independent parameters. For low preamp gain, it may be desirable to use low tube bias values, because this more closely imitates the operation of a real amplifier.
Bias Attack	50us to 10.0s	Controls the time it takes for the incoming signal to get to the tube bias. Generally the attack should be short.
Bias Release	50us to 10.0s	Sets the amount of time after the incoming signal has ceased for the amp level to shut down. Generally these times are longer than the attack times.
Post HP Fc	10Hz to 1.50kHz	This parameter filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.
Amp BassGain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to the low shelving filter.
Amp Mid1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Amp Mid1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Amp Mid1Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency parametric.
Amp Mid2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Amp Mid2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Amp Mid2Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
Amp TrebGain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to the high shelving filter.

PostEQ Level	Off, -49.5dB to +24dB	This parameter controls the output level of the main amp before the output EQ.
Speaker LPF	10Hz to 20.0kHz	Attenuates the high frequency content of the signal driving the distortion at a rate of 6dB per octave starting at the corner frequency set by this parameter. The high-frequency bandwidth acts as a low pass filter on the signal going into the distortion, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.
Gate Thresh	-96.0dB to 0.0dB	Sets the upper threshold level at which the noise gate passes the audio.
Gate Hysteresis	0dB to 48dB	Sets the lower threshold level relative to Gate Thresh, below which the noise gate shuts off the audio.
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

37 Dist→DDL→Trem

A guitar-effect chain that includes voltage-controlled distortion, parametric EQ, digital delay, and LFO modulation.

Parameter	Range	Description
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the filter cut off-frequency after the distortion. Higher values have a brighter sound. This parameter can be used to emulate a speaker cabinet.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Post VCF Fc parameter determines where (at what-frequency) this peak will occur, this parameter controls the sharpness of the peak.

Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to the low frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to the high frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency shelf.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
LFO Rate	1/1 Sys to 1/32 Sys, 0.0Hz to 20.0Hz	Controls the rate of the modulation applied to the tremolo.
LFO Shape	Triangle, Sine, Sawtooth, Square, Asym, 16-Step, 8-Step, 4-Step	Determines the shape that the LFO will use for amplitude modulation.
LFO Phase	-180deg to +180deg	Controls the relative phase between left and right LFOs.
LFO Depth	Full Dry to Full Wet	Controls the amount of tremolo.

38 Comp→Dist→DDL

A bright guitar-effects chain that features compression, gate, voltage-controlled distortion, parametric EQ, and a digital delay.

Parameter	Range	Description
Comp Ratio	1.0:1 to INF:1	Sets the amount of compression. The range is based on decibels (dB) above the threshold. If set to 4:1 for example, it will allow 1 dB increase in output level for every 4 dB increase in input level. When set to infinity, it acts as a limiter.
Comp Attack	50us to 10.0s	Determines the time after the initial signal has been detected and before the compression takes affect.
Comp Release	50us to 10.0s	Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally set longer than the attack time.
Comp Thresh	-96.0dB to 0.0dB	Sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected. To turn off the compressor, set the level to +00 dB.
Comp Output	Off, -49.5dB to +48dB	This parameter boosts or cuts the compressed signal level.
Gate Thresh	-96.0dB to 0.0dB	Sets the upper threshold level at which the noise gate passes the audio.
Gate Hysteresis	0dB to 48dB	Sets the lower threshold level relative to Gate Thresh, below which the noise gate shuts off the audio.
Dist LPF Fc	10Hz to 20.0kHz	Filters out high frequencies prior to the distortion.
Dist Offset	-99% to +99%	Adjusts the balance of even-to-odd-generated harmonics.
Dist Gain	Off, -49.5dB to +48dB	Controls the gain going into the distortion effect. This will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the distortion volume down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.
Dist Curve	Soft, Medium 1, Medium 2, Hard, Buzz	Selects the type of clipping produced by the distortion. The curves range from tube-like distortion (Soft) to nasty distortion (Buzz).
Dist Volume	Off, -99dB to 0.0dB	Controls the volume of the distortion effect. Generally, if the distortion gain is set high, set this parameter lower.
Post VCF Fc	10Hz to 7.10kHz	Determines the filter cut off-frequency after the distortion. Higher values have a brighter sound. This parameter can be used to emulate a speaker cabinet.
Post VCF Q	1.0 to 40.0	Determines the level and width of the resonant peak at the filter cutoff point. While the Post VCF Fc parameter determines where (at what-frequency) this peak will occur, this parameter controls the sharpness of the peak.
Dist Dry Lev	Off, -49.5dB to 0.0dB	Controls the amount of dry signal to be mixed with the distorted signal.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to the low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.

Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to the high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

39 EQ→Comp→Gate

EQ→Comp→Gate combines an EQ with a full feature stereo compressor. When using high compressor ratios, this insert effect functions as a limiter. This effect operates by compressing (attenuating) signals above the threshold and passing the signals below the threshold. With higher ratios and lower thresholds, this effect can be used to create sustain.

Parameter	Range	Description
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
Lo Shelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.

Comp PreDelay	0ms to 100ms	Determines how long it takes before the compressor is activated.
Comp Ratio	1.0:1 to INF:1	Sets the amount of compression. The range is based on decibels (dB) above the threshold. If set to 4:1 for example, it will allow 1 dB increase in output level for every 4 dB increase in input level. When set to infinity, it acts as a limiter.
Comp Attack	50us to 10.0s	Determines the time after the initial signal has been detected and before the compression takes affect.
Comp Release	50us to 10.0s	Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally set longer than the attack time.
Comp Thresh	-96.0dB to 0.0dB	Sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected. To turn off the compressor, set the level to +00 dB.
Comp Output	Off, -49.5dB to +48dB	This parameter boosts or cuts the compressed signal level.
Gate Thresh	-96.0dB to 0.0dB	Sets the upper threshold level at which the noise gate passes the audio.
Gate Hysteresis	0dB to 48dB	Sets the lower threshold level relative to Gate Thresh, below which the noise gate shuts off the audio.
Gate Attack	50us to 10.0s	Determines the time after the initial signal has been detected for the gate to occur.
Gate Release	50us to 10.0s	This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.
Gate Hold	50us to 10.0s	This is the detection sustain time in the ADSR—it determines how long the gate will last.

40 EQ→Chorus→DDL

An effect chain that features a four-band parametric EQ, chorus, and four discrete delays.

Parameter	Range	Description
EQ Input	Off, -49.5dB to +24dB	Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.
LoShelf Fc	10Hz to 20.0kHz	Sets the center of the low frequency EQ.
LoShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this low frequency shelf.
Mid 1 Fc	10Hz to 20.0kHz	Sets the center of the mid frequency parametric.
Mid 1 Q	1.0 to 40.0	This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.
Mid 1 Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this mid frequency band.
Mid 2 Fc	10Hz to 20.0kHz	Identical to the Mid 1 Fc parameter, and is used to control different bandwidths within the mid range.
Mid 2 Q	1.0 to 40.0	Identical to the Mid 1 Q parameter, and is used to control different bandwidths within the mid range.
Mid 2 Gain	Off, -49.5dB to +24dB	Identical to the Mid 1 Gain parameter, and is used to control different bandwidths within the mid range.
HiShelf Fc	10Hz to 20.0kHz	Sets the center frequency of the high frequency shelf.
HiShelf Gain	Off, -49.5dB to +24dB	Sets the amount of boost or cut applied to this high frequency shelf.
EQ Output	Off, -49.5dB to +24dB	Controls the gain coming out of the parametric EQ.
Dry Blend	Full Dry to Full Wet	Controls the amount of the dry signal.

LFO Rate	0.0Hz to 20.0Hz	Controls the four rates of the modulation applied to the delay time of the chorus.
Chorus Depth	0.0ms to 25.0ms	Controls the amount of modulation.
Chorus Center	0.0ms to 50.0ms	Controls the four delay times within the chorus. Adjusting this parameter will change the tonal character of the chorus.
Dly1 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the first independent delay.
Dly1 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly1 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly1 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly1 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly2 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the second independent delay.
Dly2 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly2 Feedback	-99% to +99%	Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.
Dly2 Damping	100Hz to 21.2kHz	Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The lower the number, the more the signals are damped.
Dly2 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly3 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the third independent delay.
Dly3 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly3 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.
Dly4 Time	1/1 Sys to 1/32 Sys, 0ms to 630ms	Sets the amount of delay time for the fourth independent delay.
Dly4 Level	Off, -49.5dB to +12.0dB	Adjusts the volume of the delayed signal against the original dry signal.
Dly4 Pan	Full <L to Full >R	Determines the location of the delay in the stereo spectrum.

5 Sampling/Resampling

Overview

What is Sampling?

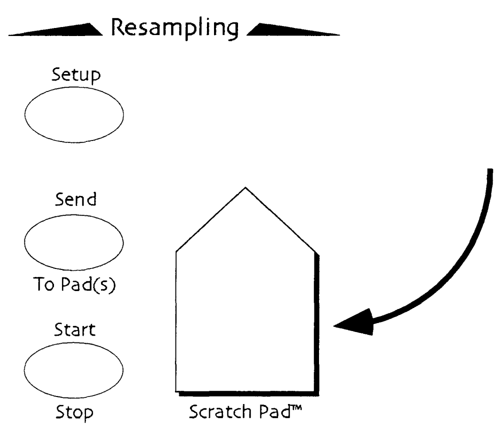
Sampling is the process of digitally recording sound. Digital recording captures sound by taking many brief snapshots of the sound—44,100 snapshots per second in the ASR-X—and storing each of these as numerical data. Each of these snapshots is each called a *sample*. When a sampler plays back the recording the spaces between such quickly occurring samples are imperceivable, and the original sound is faithfully reproduced. In the ASR-X, a digital recording is called a *wave*. In fact, "wave" refers to either a mono wave or a stereo wave, even though a mono wave is comprised of a single digital recording, while a stereo wave is actually made up of two such recordings panned left and right. Waves in the ASR-X are AIF (for "Apple Interchange Format") files.

What is Resampling?

Resampling is, as its name implies, simply sampling something again. The importance of resampling in the ASR-X should not be underestimated, though, since you can resample any sound the ASR-X produces and use the resampled material in your grooves, or as the basis for even more resampling. Used together with the ASR-X's built in effects and editing tools, resampling is the key to getting the most out of your ASR-X. It's for this reason that the sampling and resampling section and buttons on the ASR-X front panel are labeled "Resampling."

What Happens When You Create a Wave

When you sample audio on your ASR-X, the newly created wave is stored invisibly in RAM and becomes playable from the Scratch Pad.



This pad can be played in the same manner as any other pad. The Scratch Pad is unique, however, in that it's only a temporary means of playing a wave. To make fullest use of a wave, it must be assigned to one or more pads in a RAM kit. Sending to pads is described in detail later in this chapter.

Tip: You can save the contents of the Scratch Pad directly to disk along with a sound that will play them. The procedure is described in Chapter 7.

What can be Sampled in the ASR-X

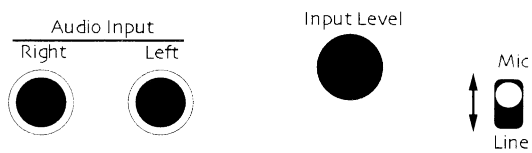
The ASR-X can create stereo or mono waves from:

- its own outputs, letting you easily resample new sounds from its sounds and sequences, taking full advantage of the ASR-X effects.
- the two Audio Input jacks on its rear panel that let you sample anything from a mic, turntable or CD player. You can make these samples with or without adding effects.
- the ASR-X outputs and the audio inputs at the same time.

"Selecting a Source," later in this chapter describes how to select your audio source.

Using the ASR-X Audio Inputs

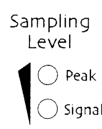
The rear panel of the ASR-X provides two 1/4" input jacks to which you can connect line-level audio sources—such as a turntable or CD player—or a low impedance microphone. You'll also find a Mic/Line switch and an Input Level adjustment knob whose uses are described below.



You can send audio into the ASR-X through either or both of the Audio Input jacks. If you're using a microphone, or microphone-level device, flip the Mic/Line toggle switch upward for the best results. When using a line-level device, flip the switch to its downward position.

Setting the Optimum Audio Input Volume

The volume of audio being sent into the ASR-X's Audio Inputs is shown in the front-panel Sampling Level LED array.

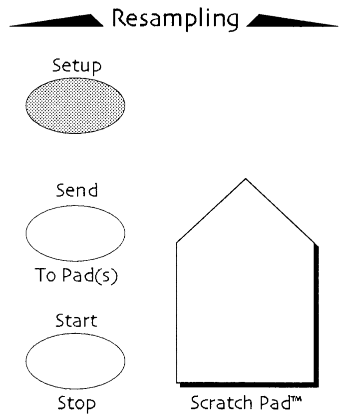


When the ASR-X detects incoming audio, the lower green LED flashes. When the audio is in danger of being too loud, causing clipping, the red LED flashes. In the ASR-X, the red LED does not necessarily mean that your input signal is too loud—it means only that you should listen to it carefully to make sure that it's not undesirably clipping or distorting. The red LED lights at -6dB.

To adjust the volume of the signal being sent into the Audio Inputs, slowly turn the rear-panel Input Level knob to achieve the best setting. You can also, of course, turn up or down the actual source of the audio.

Resampling Setup

This section describes the first steps of the sampling / resampling process: setting up. All of the features described in this section are accessed by pressing the Resampling Setup button.



Once you've pressed the Setup button ,you can turn the Parameter knob to select the parameter you'd like to adjust and turn the Value knob to set the selected parameter.

The Resampling Setup Display

All of the sampling / resampling setup parameters—with the exception of the Trig (for "trigger") parameter and meter—share a common display format in which the phrase "Resampling Setup" appears on the top line, and the parameter being adjusted appears on the bottom line:

```
Resampling Setup:
Source=       MainOut
```

The parameter selected for editing

Selecting a Source

The first item to determine when you want to sample or resample is the source of the audio to be sampled. When the Resampling Setup Source parameter is displayed, you can set it to:

- MainOut—to resample audio being produced by the ASR-X, including sounds or sequences.
- Input+MainOut—to capture audio being produced by the ASR-X, combined with audio being sent into its Audio Inputs. When this value is selected, an additional In Bus parameter is available that allows you to send the Audio Inputs' signal into the desired ASR-X effect (see "Selecting an FX Bus when Sampling a Mix of the Audio Inputs and the ASR-X Output" below).
- Input+Insert—to sample the Audio Inputs' signal after it's been processed through the currently selected insert effect.
- Input Dry—to sample the Audio Inputs' signal without any ASR-X effects added.

Selecting an FX Bus when Sampling a Mix of the Audio Inputs and the ASR-X Output

When Resampling Setup Source parameter is set to "Input+MainOut," you select the effect, if any, through which the Audio Inputs' signal will be sampled. The In Bus parameter can be set to:

- Off—to silence the Audio Inputs' signal.
- Insert, LightReverb, MediumReverb, WetReverb—to route the Audio Input's signal into the ASR-X effects (see Chapter 4 to learn more about ASR-X effects).
- Dry—to apply no effects to the Audio Input's signal.

Selecting a Stereo or Mono Recording Mode

The Rec Mode parameter allows you to determine whether you'll be recording a mono or stereo wave. You can set this parameter to:

- Stereo—so that a stereo wave (really a pair of waves panned left and right) will be created from audio produced by the selected source. When the Source parameter is set to "MainOut," the entire stereo image produced by playing the ASR-X 's pads or sequencer will be captured. When the Audio Inputs are being used, the incoming audio's stereo image is retained; if only the left or right Audio Input jack is being used, the signal will remain only on the left or right, respectively.
- Left Mono—so that a mono wave will be created from the left side of the selected audio source. The left side of the source will be panned to the center for monitoring purposes and for being sent into the effect and for being routed into the effects when they're being sampled (the left output of the effects will be captured in the wave).
- Right Mono—this functions in the same manner as Left mono, except that it uses the right side of the stereo.

Automatically Normalizing a Wave

The ASR-X can automatically normalize your wave when you create it. Normalizing digitally boosts the wave to its loudest volume short of clipping or distortion in order to achieve the best fidelity and signal-to-noise ratio. The Auto-Normalize parameter turns this automatic volume correction on or off.

Tip: You can normalize a wave after you've sampled it, if you prefer, by utilizing the Pad Process normalization feature (see Chapter 3).

Setting the Sampling/Resampling Time

Each wave occupies a portion of the ASR-X's sample memory for as long as the ASR-X is turned on, or until you erase the wave. The longer the duration of the wave, the more memory is required, and stereo waves, since they actually contain two mono waves, take up twice as much memory as mono waves do. If you create stereo waves, you'll consume the available memory twice as fast. The ASR-X's sample memory can be easily expanded to 34 megabytes through the installation of SIMM chips (see Chapter 7).

Tip: You can find out how much memory is available in your ASR-X using the Memory Manager. See Chapter 7.

The ASR-X provides the Record Time parameter to let you limit the amount of sample time you're willing to commit to a wave you're about to sample. You may choose to do this to hold a chunk of memory aside for later sampling, or simply to limit the length of the wave for musical reasons. Record Time can be set anywhere from 0.5 sec—for "seconds"—to the maximum amount of sampling time remaining in your ASR-X. The parameter shows you the amount of memory available for the type of sampling—stereo or mono—that you've selected with the Rec Mode parameter, described above.

Using the Pre-Trigger

The ASR-X will automatically begin sampling/resampling whenever it detects audio of a specified loudness when its Trig mode parameter (see below) is set to "Threshold" or "Note Event." Pre-triggering allows you to grab audio that occurs during a specified period of time just before your source reaches a volume loud enough to cause sampling to begin. This is possible since the ASR-X continually captures audio into its sample memory behind the scenes once sampling is enabled. Pre-triggering can help ensure that the front of whatever you're sampling isn't chopped off before it reaches the threshold volume (see "Setting the Trigger Threshold" below to learn about setting this threshold). You can set the length of pre-trigger time anywhere from 0ms (for "milliseconds') to 99ms.

Choosing a Trigger Mode

There are three ways that the ASR-X can begin sampling/resampling what it hears. Each of these choices is represented by a value that can be selected for the Trig Mode parameter:

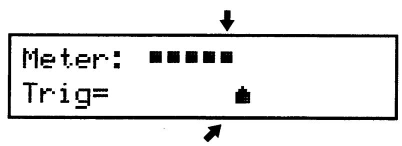
- Manual—With this setting, the ASR-X will only begin sampling when you press the Sample Start/Stop button.
- Threshold—With this setting the ASR-X will begin sampling when it detects audio from the selected source that reaches the threshold set with the Trig parameter, described below.
- Note Event—With this setting, the ASR-X will begin sampling when a pad is played or a MIDI note (note-on) message is received form an external MIDI device on any MIDI channel. This is especially handy when you're resampling sounds in the ASR-X.

Each of these modes is activated by pressing the Sample Start/Stop button, and de-activated by pressing the Start/Stop button a second time (see "How to Start and Stop Sampling a Wave" below).

Setting the Trigger Threshold

The Trig parameter allows you to set a volume threshold at which the ASR-X will begin sampling/resampling its source when the Trigger Mode parameter (described above) is set to "Threshold." This parameter is presented in a special display that makes it easy to select a useful volume:

The top line is a meter that shows the volume of notes as you play



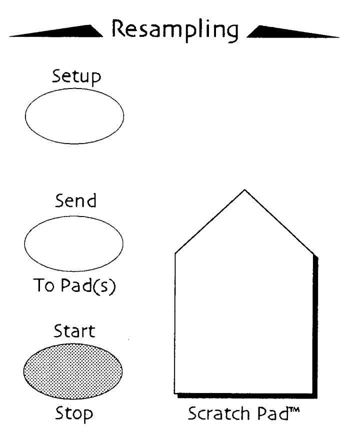
The pad symbol shows the current threshold setting

By playing some notes on a pad or via MIDI that represent what you intend to sample, you can see the volume of your audio on the display's top line. By turning the Value knob, you can move the pad symbol to match the level at which you expect to play the audio you'll be sampling.

Tip: Take a few moments to find the right Trig setting—if you set the threshold too low, sampling may begin too early; if you set it too high, sampling may not begin when you want it to if you play a pad or key too softly.

Sampling/Resampling a Wave

How to Start and Stop Sampling a Wave



The Start/Stop button is the device that turns sampling/resampling in the ASR-X on and off. When the ASR-X is not sampling, pressing the Start/Stop button engages the sampling function in a manner determined by the setting of the Trig Mode parameter (described above):

- When Trig Mode is set to "Manual," the ASR-X begins sampling at the moment you press the Start/Stop button.
- When Trig Mode is set to "Threshold," pressing the Start/Stop button causes the ASR-X to begin listening for a source signal loud enough to trigger the beginning of sampling.
- When Trig Mode is set to "Note Event," pressing the Start/Stop button causes the ASR-X to begin waiting for a pad to be played, or a MIDI Note message to trigger the beginning of sampling.

Note: After you press the Start/Stop button when Trig mode is set to either "Threshold" or "Note Event," the display will show waiting for "Waiting For Trigger." Pressing Start/Stop a second time will cause sampling to begin without triggering. To disable trigger sampling, press the Exit/No button.

Once sampling begins, the ASR-X display shows you its progress:

```
Sampling In Progress
======*
```

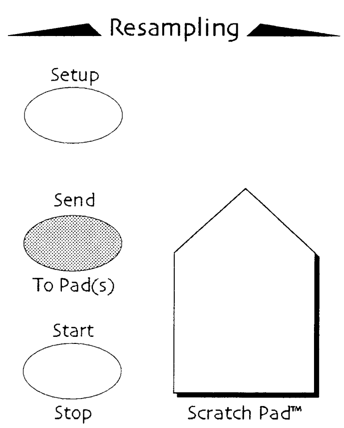
This graphically shows the amount of sampling time being used

The bottom line of the display becomes a meter that shows how much of the sampling time allotted with the Record Time parameter (see above) has been consumed by the wave you're creating.

Tip: You can view other areas in the ASR-X as sampling occurs—this can be handy when resampling the sequencer or when tweaking sounds or effects in real-time. "Sampling In Progress" flashes on the display's top line in alternation with the display pertaining to the non-sampling area of the ASR-X you've selected.

When sampling is in progress, pressing the Start/Stop button stops sampling. When sampling is complete, the "SendTo Pads?" display appears (see below to learn about sending a wave to pads). You can audition your new wave at this point by playing it on the Scratch Pad.

Sending a Wave to Pads



When you've finished sampling/resampling, the new wave is playable from the Scratch Pad. To make the wave truly usable, however, you'll want to send it to one or more pads in the currently selected RAM kit.

Note: If the currently selected track is not using a RAM kit when you begin sampling, the ASR-X will convert the track's sound into a RAM kit for you, so that you'll have somewhere to send your wave. The new kit—which will be named after the original sound with a number added to its end—can be found in the USER-SND and DRUM-KIT SoundFinder categories

Tip: If you'd like to sample into an otherwise empty kit—a "clean slate," in other words—select the ROM sound called "Silence" before you sample. This will be converted into a RAM kit that will play only your wave. You can assign any sounds you like to its pads after you've finished sampling and sending your wave to the desired pad or pads.

When you press the Resampling Start/Stop button to finish sampling, the "SendToPads?" display automatically appears:

The top line shows the octave to which the pads are currently pointing

```
C2...0oct..C3 SendTo
                Pads?
```

If you'd like to send the wave to a pad in an octave other than the one currently being played by the pads, you can select the desired octave using the Octave Transpose button (see Chapter 3 to learn more about using the Octave Transpose buttons). The display will always show you the octave currently being played by the pads.

After selecting the desired octave, press each pad to which you'd like to send your wave. (You can use the Octave Transpose buttons at any time during the SendTo Pads procedure, allowing you to send your wave to any pads in any octaves.) The display will graphically show which pads have been selected.

```
C2...0oct..C3 SendTo
                Pads?
```

This shows the second, fifth and sixth pads in the selected octave as having been pressed

If you've selected a pad, but would like to un-select it, press it again—and wave won't be sent to the pad.

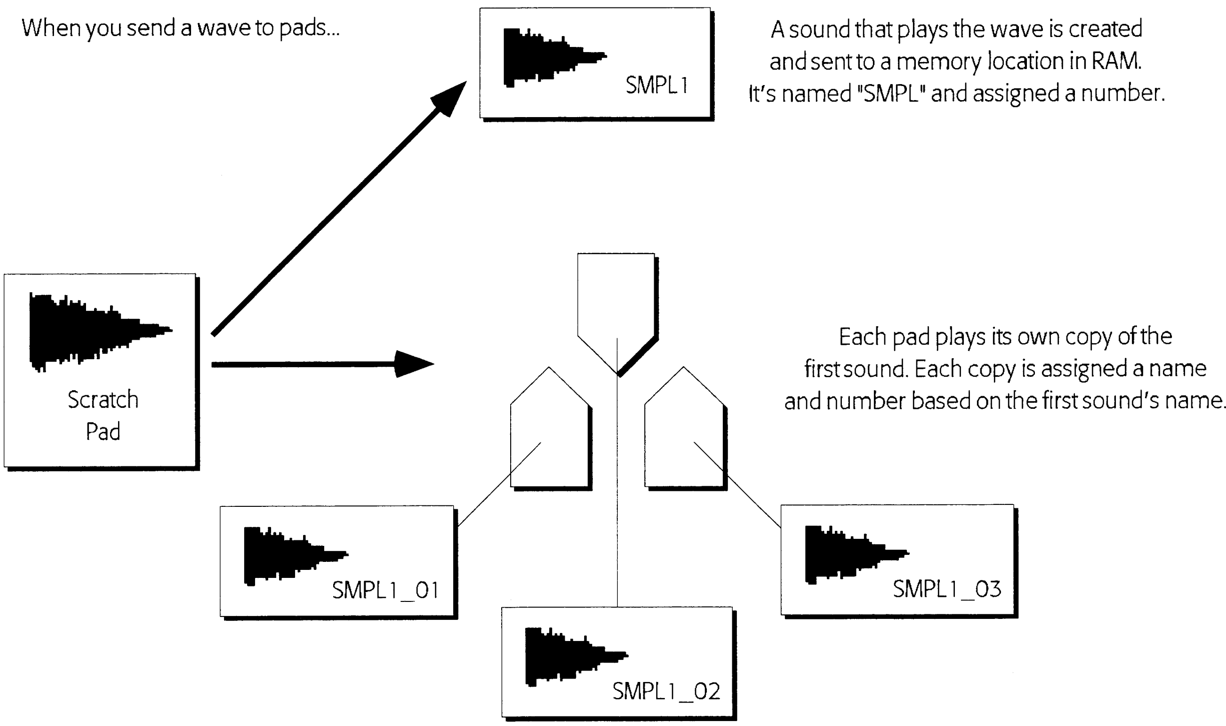
When you've selected all of the pads to which you want to send your wave, press the Yes button. If you'd like to cancel the procedure, press the No button.

Tip: After you've sent a wave to pads, the wave remains in the Scratch Pad until you sample something else or turn the ASR-X off. You can send the contents of the Scratch Pad to a pad in a RAM kit at any time by pressing the Resampling Send To Pad(s) button.

What Happens When You Send a Wave to a Pad or Pads?

When you send a wave to a pad in the selected track's RAM kit, the ASR-X creates a standard RAM sound that plays the wave (see Chapter 3 to learn more about ASR-X standard and kit sounds). The sound is named "SMPL" followed by a number—when you power up, the ASR-X starts back at SMPL1 and raises the SMPL number value each time you sample something new and send it to pads. The SMPL sound is not actually played by any of the pads—it's created as a safety copy of the sound that can be selected from SoundFinder and assigned to a track or pad at any time.

When you send your wave to pads, the ASR-X creates copies of the SMPL sound—one for each pad. Each of these will be named similarly to the sound, but will have an additional underscore and number following its name. For each pad you send the wave to, the number increases by one. This allows you to be able to tell which sound is played by each pad. You can change the name of any of these sounds using the Memory Manager if you like (see Chapter 7).



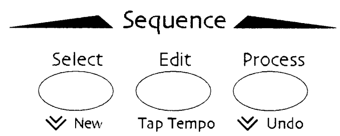
Having each pad play the wave using its own sound allows you to edit each pad's sound separately while keeping the wave on which it's based intact (to learn more about editing pad sounds, see Chapter 3). These sounds are normal ASR-X standard sounds and are stored in the lowest-numbered empty locations in RAM. They can be selected for use by a pad in any RAM kit or by any track in the sequencer.

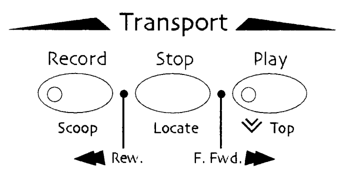
Automatically Spreading a Wave Across all of a Kit's Pads

There may be times when you'd like to spread your wave across all the pads, with each one playing it at a different pitch. To do this, press the Track Sound button, dial in the *CUSTOM SoundFinder category, and select the original SMPL sound (SMPL1, for example). As a standard sound, it will be played by all the pads. To hear the wave at its original pitch and speed, use the Octave Transpose button to aim one of the pads at Middle C (C4). Middle C is always the root key for a wave.

6 Sequencing

The ASR-X contains a potent 16-track sequencer for the construction of grooves—or any other kind of music. This chapter describes the concepts behind the ASR-X sequencer and how to harness its power. All of the sequence recording, playback and mixing controls are found grouped together on the ASR-X front panel.





There are two sets of sequencing buttons. They are:

- the Sequence buttons—which provide sequence settings as well as sequence and track tools. These are described in "The Sequence Select Button," "The Sequence Edit Button" and "The Sequence Process Button" later in this chapter.
- the Transport buttons—provide the controls for operating the sequencer. These are described in "Operating the Sequencer" later in this chapter.

Overview

How the ASR-X Sequencer Works

The ASR-X sequencer records the MIDI information generated by the ASR-X pads or by MIDI data received from an external MIDI device. When the sequencer plays this data back, it sends it to the areas within the ASR-X that produce its sounds and effects, and your music is faithfully reproduced.The sequencer can also be synchronized to an external MIDI timing source, such as a computer or stand-alone sequencer (see Chapter 7's "Edit MIDI Settings?").

Each musical event the sequencer records takes up space in the ASR-X's memory. Unlike conventional recording media such as tape, when there's no musical activity—during rests between notes, for example, or when you're holding a long note—no data is required and no memory is used.

What is a Sequence?

A sequence is a piece of music recorded by the ASR-X using MIDI technology. Each sequence can contain the separate, synchronized recordings of up to 16 performances, each using its own sound—each of these is called a *track*. Each sequence also has its own insert effect and global reverb to which its tracks can be routed (Chapter 4 explains the ASR-X effects in detail). ASR-X sequences are Standard MIDI Files that can be read from floppy by any Macintosh or PC-compatible computer—when you load an ASR-X sequence into your computer sequencer, you can still use the ASR-X's sounds by accessing them via MIDI.

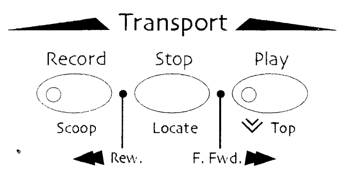
Note: The entire structure of the ASR-X is based on the 16 tracks of the currently active sequence (see Chapter 2 to learn about tracks, including how to select tracks, how to assign sounds to tracks and how to edit them).

There is always a sequence active in the ASR-X, even if you haven't recorded on any of its tracks yet. The ASR-X can hold up to 128 sequences, each of which can be selected in turn (see "The Sequence Select Button" later in this chapter to learn how to select sequences).

Each sequence can be renamed, copied or deleted from memory. Controls are provided that allow you to determine the behavior of a sequence, from its tempo to the nature of its pre-recording countoff, and so on. These sequence settings are described in "The Sequence Edit Button" later in this chapter. The performances contained on the tracks in a sequence can be edited and perfected through the use of various onboard processes, described later in this chapter in "The Sequence Process button."

Operating the Sequencer

The Transport buttons are the means by which most sequencer recording and playback operations are performed.



In general, the Transport controls function in a manner similarly to the controls on any cassette or CD player, tape recorder or VCR.

To accomplish this:	Do this:
Play a sequence	Press the Play button.
Stop playback of a sequence	Press the Stop button.
Jump back to the beginning of a sequence	Press the Play button twice.
Rewind to the top of the sequence	Hold down the Record button and press the Stop button.
Rewind bar by bar	Hold down the Record and Stop buttons.
Fast forward bar by bar	Hold down the Stop and Play buttons.
Record a track	Hold down the Record button and press Play.
To punch in on a track manually.	Press the Play button to start playback. Hold down the Record button and press the Play button at the location at which you want to start recording. Note: You can punch in using a foot switch—see Chapter 7. In addition, the sequencer Region feature provides automated punching-in—see "Using Regions" later in this chapter.
Start from any location within a sequence	Hold down the Stop button, and while continuing to hold it, turn the Parameter knob to select the desired measure, press the arrow buttons to select a beat, and turn the Value knob to select a clock. When you've selected a location, let go of the Stop button and press the Play button.
Scoop all instances of a pitch from a track	Select the track from which you'd like to scoop the note. Hold down the Record button and while holding down Record, press the pad, or key on an external MIDI keyboard, for the note you want to remove. The ASR-X will offer to remove all instances of the note. Press the Yes button to complete the procedure.
Scoop out notes as the track plays	In Add mode, select the desired track and hold down the Record button (see "RecordMode" later in this chapter). While continuing to hold down Record, press the Play button . Let go of both buttons. Re-press and hold the Record button, play the pad, or key on an external MIDI keyboard, for the note that you want to remove. As long as you hold down the Record button and the note's pad/key, all occurrences of the note will be removed from the track.

The Transport LEDs

The Play and Record button contain LEDs that provide information about what the sequencer is doing:

- The green LED in the Play button lights whenever the sequencer is playing.
- The red Record LED lights when recording is taking place in the currently selected track.
- The red Record LED flashes when the sequencer is waiting to record, or to scoop notes from a track.

The Transport Displays

When operating the sequencer Transport controls, one of two displays will always appear, except when you're adjusting the sequence setting accessed by pressing the Sequence Edit button. In all other cases:

- When the sequencer is playing, the sequencer track page is displayed. This display tells you what track is currently selected, where you are in the sequence, and the name and SoundFinder category of the track's sound. If a track has not yet been recorded, the display will show the word "Empty" in the special information area shown below. If the track is muted or soloed (see Chapter 2), this area of the display will show "mute" or "solo."

The currently selected track   Special information   The bar and beat of the current location in the sequence

```
1.01                1.01
KEYS      :LOOKIT THIS
```

The SoundFinder category and name of the track's sound

- When you are moving within a sequence in a non-play mode—such as when you're fast forwarding, rewinding or selecting a location within the sequence—the locate display appears.

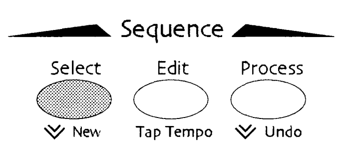
The current location of the sequence

```
Locate:             2.03
Go To=          2.03.001
```

The bar, beat and clock where playback will begin

Note: The location within a sequence is measured in bars—or measures—beats and clocks. A clock is 1/384th of a quarter note. This 384 ppqn (pulse-per-quarter-note) resolution means that the ASR-X sequencer can capture the most subtle of rhythmic nuances.

The Sequence Select Button



The Sequence Select button provides access to sequence selection and creation tools. Each time the Sequence Select button is pressed, one of two displays appears, allowing you to create a new sequence or select one that's currently in the ASR-X's memory.

Tip: To quickly select the sequence creation display, double-click the Sequence Select button.

Creating a New Sequence

When the sequence-creation display appears you can press the flashing Yes button to create a new sequence. The new sequence will be assigned a default name and number that tells you which empty sequence location it will occupy.

```
Create new sequence?
SEQ00002
```

The sequence's default name   This number shows the new sequence will be the second one in memory

Tip: See "The Sequence Process Button" later in this chapter to learn how to rename a sequence.

Selecting Sequences

When the ASR-X shows its sequence selection display, you can turn the Value knob to select any of the sequences currently in the ASR-X's memory.

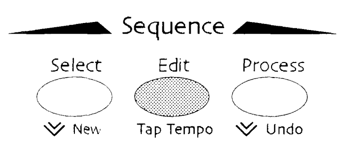
The selected sequence is the first one in memory   The number of sequences in memory

```
Sequence 1 of 4:
Grand Slam
```

The name of the selected sequence

When you've selected the desired sequence, press the Enter button to load it into the sequencer, where it can be played or edited.

The Sequence Edit Button



The Sequence Edit button provides access to sequence settings. When you press the Sequence Edit button, the ASR-X displays one of the settings for the currently selected sequence in the form of a parameter. Turn the Parameter knob to access each of these, and turn the Value knob to select the desired value for the displayed parameter.

Current Tempo

Each sequence has a basic tempo setting that determines how fast it will play, expressed in BPM ("beats per minute"). You can set the currently selected sequence's tempo manually, by selecting the Current Tempo parameter and dialing in the desired value, or by tapping out the beat you want at any time on the Sequence Edit button—when you do so, the ASR-X will jump to the Current Tempo display to let you see the tempo you're playing.

Tip: The Final Mix record mode (see "RecordMode" below) allows you to record tempo changes.

RecordMode

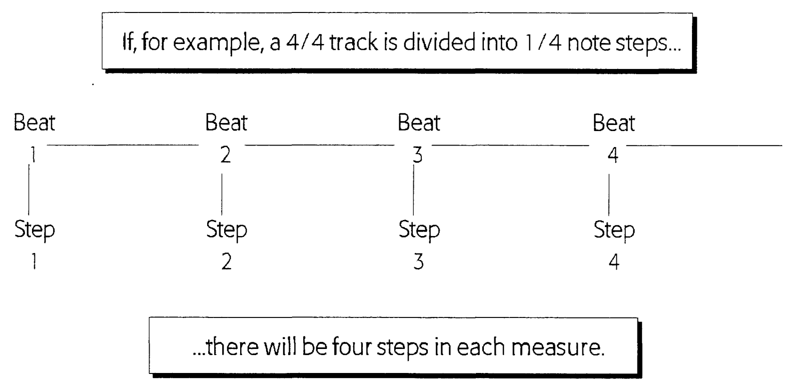
The ASR-X sequencer provides several different recording modes, each of which allows you to perform a different type of recording or mixing task:

- Replace—This is the most basic recording mode, where newly recorded material replaces anything that was previously recorded on the selected track. In Replace mode, the length of a sequence is defined by its longest track.
- Add—In Add mode, newly recorded material is combined with anything previously recorded on a track, so that both the new and old material is heard on playback. In Add mode, the length of a sequence is defined by the length of the first track.
- Step—The Step mode allows you to use the ASR-X sequencer as a non-real time recording device, where each note or chord is entered one at a time. See "Step Recording" below for details.
- Track Mix—Track mix mode allows you to record real-time track parameter changes onto a track. See "Recording Track Parameter Changes" later in this section.
- Final Mix—Final Mix mode allows you to record whole-sequence volume and tempo changes. See "Recording Sequence Volume and Tempo Changes" later in this section.

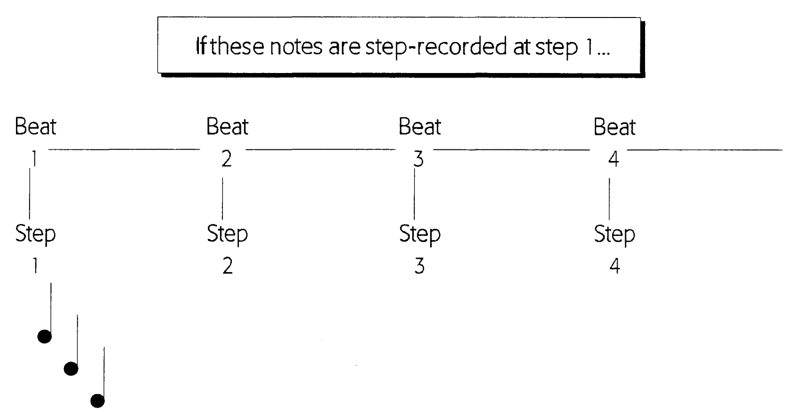
Step Recording

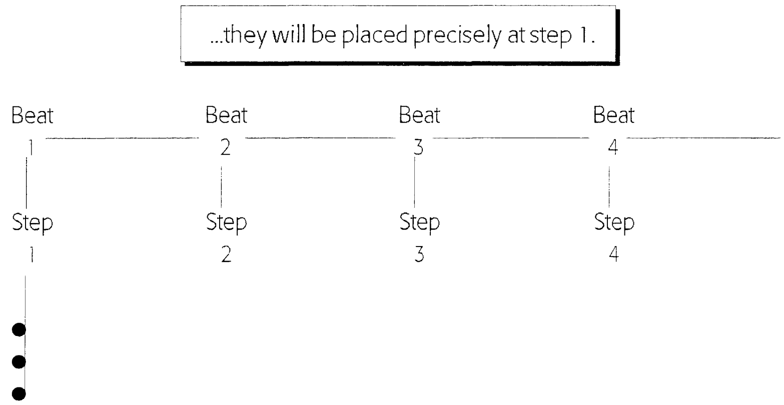
Step recording lets you record notes and chords on a track while the sequencer only moves forward when you instruct it to do so. During playback, a step-recorded track plays at the sequence's normal tempo, causing all the notes and pedal presses you've entered to sound as if they were performed normally. Step recording is ideal for impossible-to-play passages, or for times when a not-quite-human-sounding performance is desired.

In step recording, each track is divided up into divisions of a beat, called *steps*.



With the sequence at rest, you enter notes and sustain/sostenuto pedal presses at their desired locations.

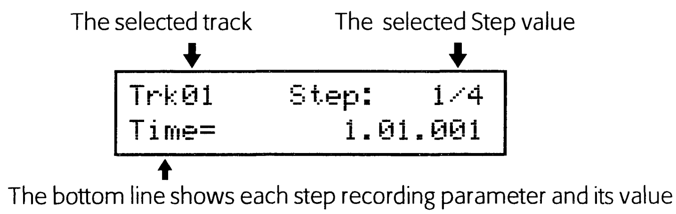




The sequence is then advanced, step-by-step, through the track, as you place the desired notes, chords or sustain/sostenuto pedal presses where you want them. Notes are recorded at the velocity with which they're played on the keyboard. By paying attention to the velocities at which you play your notes, you can help simulate a natural sound.

Tip: Chords can be recorded by playing the notes in the chord simultaneously or one at a time.

The ASR-X provides a suite of step-recording parameters. Turn the Parameter knob to select each parameter, and the Value knob to change the selected parameter's setting. It's best to set up all of the step parameters as you'd like before you play notes on the ASR-X pads or send notes to the ASR-X via MIDI. All of the step recording parameters share a common display format:



Step Record parameter	The parameter's purpose
Time	Shows your current location in the sequence. Notes played on the pads or received via MIDI will be places at the displayed location..
Step Size	Sets the division of the beat at which notes will be placed. This may be set to: 1/1, 1/1T, 1/2D, 1/2, 1/2T, 1/4D, 1/4, 1/4T, 1/8D, 1/8, 1/8T, 1/16D, 1/16, 1/16T, 1/32D, 1/32, 1/32T, 1/64D, 1/64 and 1/64T. Note: "D"="dotted value"; "T"= "triplet"
Gate Time	Sets the length of each recorded note. This can be set to: 1/1, 1/1T, 1/2D, 1/2, 1/2T, 1/4D, 1/4, 1/4T, 1/8D, 1/8, 1/8T, 1/16D, 1/16, 1/16T, 1/32D, 1/32, 1/32T, 1/64D, 1/64, 1/64T, Step and Held. Note: "D"="dotted value"; "T"= "triplet." When the parameter is set to "Step," the duration of each note will equal the Step size. When the parameter is set to "Held," the duration of each note is set by holding down the note's pad, or key on an external MIDI keyboard, and advancing the Time value, letting go of the pad or key where you want the note to end.

Gate Percentage	When Gate Time is set to any value other than "Held," the Gate Percentage parameter lets you to shorten the length of recorded notes by reducing the selected Gate Time value by a percentage. Tip: Try a Gate Percentage setting of 80% to approximate a real-time performance.
Auto-Step	Allows you to set the manner in which the sequencer will advance through the track. When Auto-Step is set to "On," each note played (see tip below about chords) will cause the sequencer to advance to the next step. When it's set to "Off," the track will advance to the next step each time you press the Enter button. Tip: You can move to next step using a foot switch, if you like. See Chapter 7's "Set system prefs?" Tip: The sequencer interprets notes played closely together—within 100 milliseconds of each other—as being a chord. When Auto-Step is on, only notes played further apart will advance the track to the next step. If you'd like to play the notes in a chord one-by-one, turn Auto-Step off.

To begin step recording, turn the Parameter knob counter-clockwise to return to the Time display, hold down the Record button and press the Play button. To end recording, press the Stop button.

Recording Track Parameter Changes

When you hold down the Record button and press the Play button to begin recording in Track Mix mode, the ASR-X shows the Track Mix display for one of the track's parameters (track parameters are described in Chapter 2).

```
Trk01 Mixdown    1.01
Mix (Expression)=127
```

Using Track Mix mode, you can record real-time changes for the following parameters, each of which can be altered via a standard MIDI controller:

- Mix (Expression)—controller #11
- Track Pan—controller #10
- Glide Time—controller #84
- Normal LFO Rates—controller #75
- Amp Env Attack—controller #73
- Amp Env Decay—controller #76
- Amp Env Release—controller #72
- Filter Cutoff—controller #74
- Filter Resonance—controller #77

To move among the available parameters, while recording in Track Mix mode, turn the Parameter knob or press either the Exit or Enter button. When the desired parameter is shown, you can turn the Value knob to change its setting—all changes you make will be recorded.

Recording Sequence Volume and Tempo Changes

When you hold down the Record button and press the Play button to begin recording in Final Mix mode, the ASR-X shows one of two Final Mix displays: one for the sequence's overall volume...

```
Trk01 Mixdown    1.01
Final Mix=       100%
```

...and one for its tempo.

```
Trk01 Mixdown    1.01
Final Tempo=   ♩:120
```

To move between the two displays, turn the Parameter knob or press either the Exit or Enter button. When the desired parameter is shown, you can turn the Value knob to increase the sequence's track mix settings—which rise and fall as a single entity— or its tempo setting. The rate of increase or decrease to the mix setting is expressed as a percentage of its original value.

Note: Use Final Mix for volume changes with care—it would be a good idea to save a safety copy of your work to disk first—since there is no undo available for Final Mix volume changes.

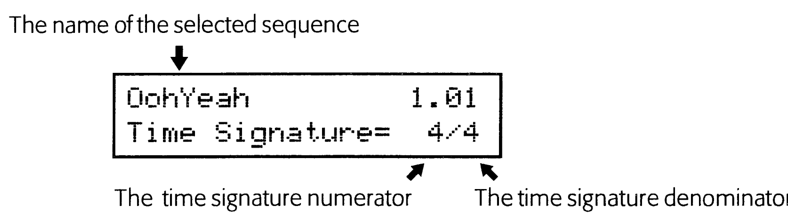
Tip: If you'd like to use the Final Mix mode on a selected group of tracks only, mute all of the sequence's other tracks. Final Mix will only affect the un-muted tracks.

Loop Playback

A sequence can be programmed to play through once to the end and stop, or to loop back to its beginning over and over again until you press the Stop button. The value selected for the Loop Playback parameter —either "No" or "Yes"—determines whether or not the sequence will loop.

Time Signature

The time signature of an ASR-X sequence can be changed whenever the sequence is not playing, either before or after recording has taken place. If you change the time signature after recording, your music will not change—it will merely be interpreted by the sequencer as being at the new time signature.



When Time signature is displayed, you can turn the Parameter knob to select the numerator or denominator—the selected item will flash to show it's selected—and turn the Value knob to dial in the desired value. The numerator can be set from 1 to 99; the denominator can be set to 1, 2, 4, 8, 16, 32 or 64.

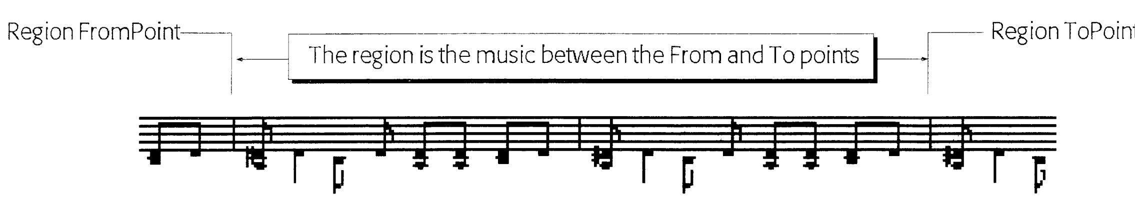
Tip: For you convenience, when you reset the time signature denominator, the denominator for the sequence's metronome click is set to match the new denominator (see "Click Timing" below).

Using Regions

The ASR-X sequencer allows you to define a section of the currently selected sequence as a *region*. A region can have several uses. It can be:

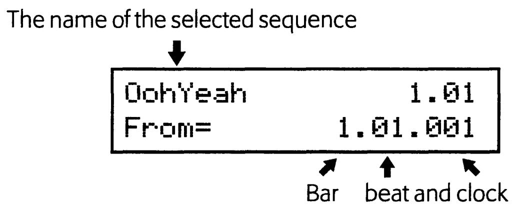
- the only section of the sequence that's heard when you play the sequence.
- the portion of the sequence that gets re-recorded during automated punching in.
- a section of the sequence or of a track upon which you perform one of the sequence processes (see "The Sequence Process Button" later in this chapter.)

A region is defined by setting and turning on the Region FromPoint and/or Region ToPoint. The FromPoint sets the beginning of the region, while the ToPoint sets its end.



Tip: You can turn on only the FromPoint or ToPoint if you'd like to use the sequence's original end or beginning , respectively, as the end or beginning of the region.

To turn on and set the Region FromPoint, set the Region FromPoint parameter to "On," and turn the Parameter knob to display the From= parameter:



Turn the Parameter knob to select the bar number, beat number and clock number you'd like to set, and turn the Value knob to dial in the desired value. The Region ToPoint is set in the same way, using the Region ToPoint and To= parameters.

Tip: You can set the From or To value to the nearest beat by pressing the Enter button.

To accomplish this:	Do this:
Play only a specific section of a sequence	Use the region parameters to define the section of the sequence you want to play. The standard play, rewind and fast forward functions will operate within the region you've defined.
Set up an automated punch-in	Use the region parameters to define the section of the sequence you want to record. Hold down the Stop button and turn the Parameter knob all the way counter-clockwise to set the Locate point to 1.01.001. Hold down the Record button and press the Play button. As the sequence plays, the Record LED will flash. When the sequence reaches the Region FromPoint, recording will begin.
Define a section of the sequence a portion of a track for processing.	Use the region parameters to define the section of the sequence you want to process. Each of the processes that can be performed on a sequence or track—except Undo—will offer a "within region" option that allows you to process only the defined region (see "The Sequence Process button" later in this chapter).

Edit Click/Countoff?

When "Edit Click/Countoff?" is displayed, responding by pressing the Yes button will call up a collection of parameters that allow you to determine the nature of the metronome click that can be used as a rhythmic reference while recording or listening to the selected sequence. Also included in this collection are parameters that allow you to customize the countoff, if any, that will be heard before the selected sequence plays during recording or playback..

Note: When a sequence is set to have a countoff, the countoff will be shown in the sequencer displays as negative values climbing upward to the first beat of the sequence.

Click

The Click parameter determines in what circumstances, if any, the sequence's reference metronome will be heard. It can be set to:

- Off—so that there will be no metronome heard during recording or playback.
- Record Only—so that the metronome will only be heard during recording.
- Play Only—so that the metronome will only be heard during playback.
- Record/Play—so that the metronome will be heard during recording and playback.

Click Sound

The Click Sound parameter determines the sound that will be used for the sequence's reference metronome click. It can be set to:

- Click—so that the metronome sound will be a mechanical click.
- Stick—so that the metronome sound will be two drumsticks hitting together.

Volume

The click Volume parameter sets the loudness of the sequence's metronome click.

Pan

The click Pan parameter value sets the stereo position of the metronome click.

FX Bus

The click FX Bus parameter allows you to send the metronome click through one of the ASR-X effects by routing it to:

- Insert—so that the metronome will be heard through the sequence's insert effect.
- LightReverb—so that the metronome will be heard with a small amount of reverb.
- MediumReverb—so that the metronome will be heard with an average amount of reverb.
- WetReverb—so that the metronome will be heard with a large amount of reverb.
- Dry—so that the metronome will not be routed through the ASR-X effects.
- AuxOut1, AuxOut2, AuxOut3, AuxOut4—so that the metronome will be routed to one of the ASR-X's auxiliary outputs and removed from the main mix.

Note: The AuxOut values are available when an X-8 output expander is installed in the ASR-X.

Click Timing

The Click Timing parameter sets the division of the beat to be played by the metronome, and may be set to 1/2, 1/4, 1/8, 1/16 and 1/32 notes, as well as their triplet values (shown with a "T.")

Countoff

The Countoff parameter determines in what circumstances, if ever, a countoff will be heard before the sequence begins. It can be set to:

- Off—so that there will be no countoff heard during recording or playback.
- Record Only—so that the countoff will be heard at the beginning of the sequence prior to recording.
- Play Only—so that the countoff will be heard at the beginning of the sequence during playback.
- Record/Play—so that the countoff will be heard before recording or playing back the sequence.

Countoff Sound

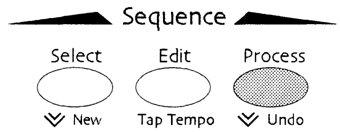
The Countoff Sound parameter sets the sound to be played as the countoff rhythmic reference. It can be set to:

- Quiet—so that there will be nothing heard during the countoff.
- Click—so that the countoff sound will be a mechanical click.
- Stick—so that the countoff sound will be two drumsticks hitting together.

Countoff Bars

The Countoff Bars parameter sets the length of the countoff in measures; it can be set to anywhere from 1 to 16 measures.

The Sequence Process Button



The Sequence Process button provides access to various track and sequence editing tools. Each of these performs a particular process on the selected track or sequence, and each is presented as a top-level question that can be answered by pressing the No or Yes button. Pressing No takes you to the sequence selection display described in "The Sequence Select Button" earlier in this chapter. Pressing the Yes button either executes the process or brings you to relevant parameters presented on sub-displays that allow you to determine exactly how the process will be performed. Once you've defined the process to be performed by adjusting the parameters offered on these sub-displays, pressing the Yes button will execute the selected process.

Note: The displayed questions relating to processes are not always shown on the display exactly as depicted in the following descriptions—on the display, they include the number of the selected track or sequence where appropriate.

Tip: When you're using the sequencer's region feature (see "Using Regions" earlier in this chapter), the sequences processes will conveniently offer "Within Region" values where applicable.

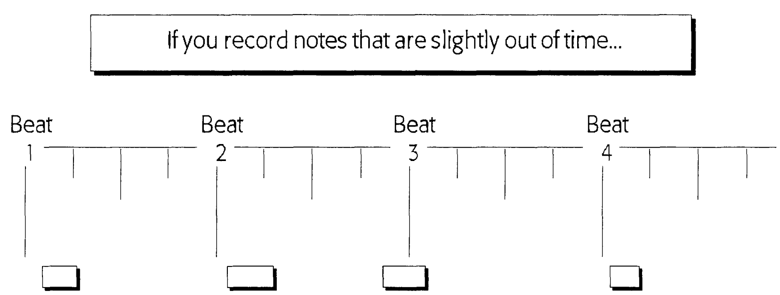
Undo track ?

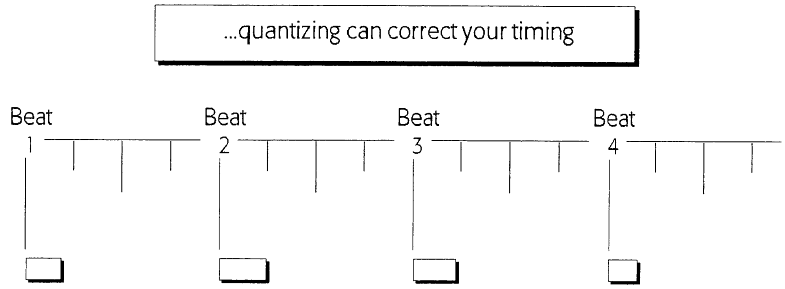
The ASR-X allows you to undo your last recording or process. Pressing the Yes button in response to this displayed question restores your track to the state it was in before you last recorded on it, or processed it.

Tip: You can quickly jump to the Undo question at any time by double-clicking the Sequence Process button.

Quantize track?

The ASR-X provides a powerful set of tools for correcting or altering the timing of recorded notes. These processes are described by the general term *quantizing*. When you quantize notes, you shift them in time to line up with specified divisions of the sequence's tempo, as in this simplified illustration:





The various ASR-X quantization tools are sophisticated, but simple to use. By using them in combination with each other, you can set up some quite elaborate quantizing processes. Each is presented as a parameter on a quantizing sub-display.

Tip: You can execute a quantize process by pressing the Yes button at any time that one of the quantize displays is visible—the current settings for all of the quantize parameters will be used.

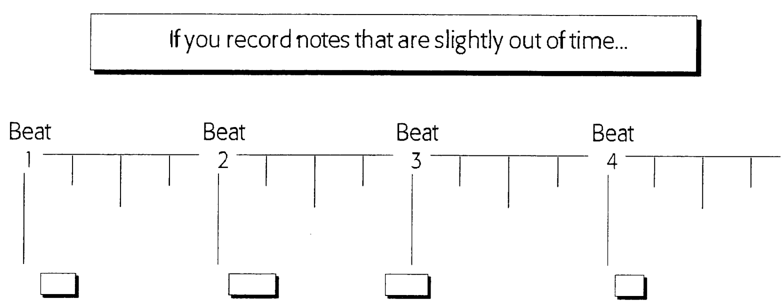
Template

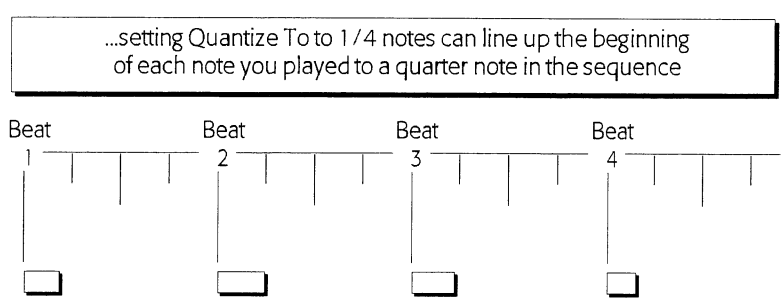
The ASR-X provides complete quantization setups as *templates.* These templates set all of the quantize parameters to sensible settings for the task after which they're named—for a complete list of the quantization templates, see Chapter 9. You can a template to use as is, or as a starting point for your quantization programming. When a template has been edited, this will show "**EDITED**."

Tip: You can create and save your own templates that contain quantization settings you'd like to re-use. The first four templates—USER TEMP 1 through 4—are memory locations in which you can store your favorite quantization settings. See "Save quantize as?" later in this section.

Quantize To

The Quantize To parameter sets the division of a beat to which notes will be aligned during the quantization process.





The parameter can be set to:

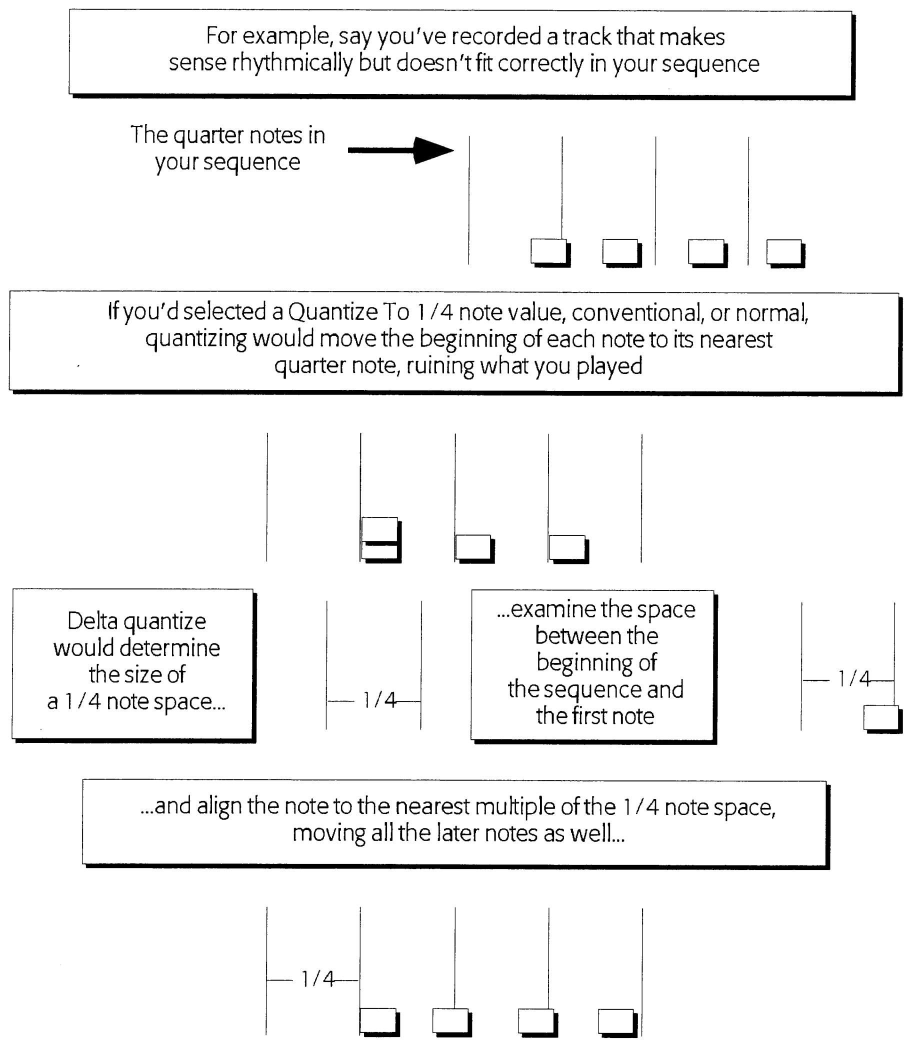
- 1/1—whole notes
- 1/1T—whole-note triplets
- 1/2—half notes
- 1/2T—half-note triplets
- 1/4—quarter notes
- 1/4T—quarter-note triplets
- 1/8—eight notes
- 1/8T—eight-note triplets
- 1/16—sixteenth notes
- 1/16T—sixteenth-note triplets
- 1/32—thirty-second notes
- 1/32T—thirty-second-note triplets
- 1/64—sixty-fourth notes
- 1/64T—sixty-fourth-note triplets

Method

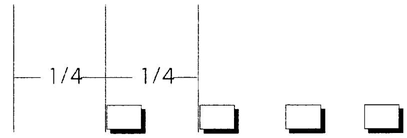
The ASR-X offers two ways to quantize your music. You can set the Method parameter to:

- Normal—to quantize the notes in the selected track using the traditional quantization method: moving the beginning of each note to the nearest occurrence of the Quantize To value.
- Delta—to use a revolutionary ENSONIQ method of quantization first introduced in the MR-61 and MR-76 that preserve's the player's musical intent in a way that normal quantization can't. With delta quantization, if what you've recorded drifts out of time with the sequence, as long as it makes rhythmic sense internally, you can quantize it to perfection.

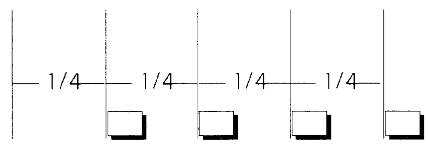
  Delta quantization examines the space—or *delta*—between the beginning of the sequence and the first note and resizes the space to match the nearest multiple of the quantize To value, shifting all of the later notes in the track so that they retain their original relationship to the first note. This process is then repeated for the space between the first and second note, and so on, until all of the notes in the track have been correctly quantized.



...and then do the same for the space between the first note and the second...



...and so on, until your quarter notes line up with the quarter notes in the sequence



Note: When Method=Delta, the only additional quantize parameter required—and available—is Quantize To.

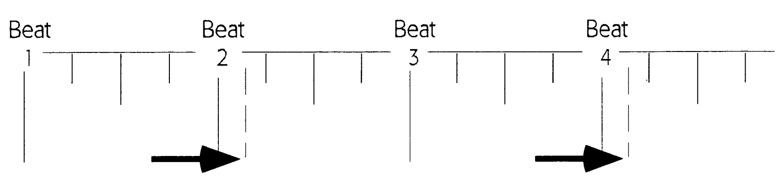
Strength

The Strength parameter determines to what degree the notes in the track will be aligned to the Quantize To value. This parameter allows you to correct the timing of the music on a track to the extent that you desire, without necessarily making it absolutely—some might say "unnaturally"—perfect. Sometimes, a little quantizing help is all that a performance needs. The Strength parameter is expressed in percentages. A value of 100% will line up the beginning of the notes in the track exactly to the division of the beat chosen with the Quantize To value. A Strength setting of 0% will leave the notes unaffected.

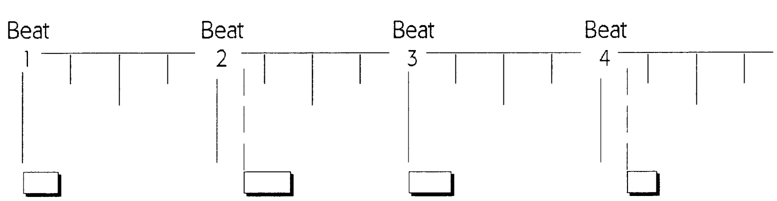
Swing

This parameter allows you to add a "swing" feel to your quantized tracks. Every other occurrence of the type of note set by the Quantize To parameter is altered to sit slightly behind the beat. When the notes in your track are aligned to the resulting combination of even and slightly lagging notes, a swing feel is achieved.

Swing makes every other appearance of the metric value you've selected slightly late



...and your notes are lined up with that altered rhythmic reference



The Swing parameter can be set from 50%—for no swing—where each of the Quantize To notes occurs precisely halfway between the note before it and the note after, to 74%, where every other note is pushed nearly halfway towards the following note.

Random

The Random parameter allows you to add aesthetically pleasing timing irregularities to a track as you quantize it. This can help simulate the small rhythmic fluctuations likely to be present in a naturally occurring performance. The irregularities provided by the ASR-X's randomizing function are intelligently created. They don't jump erratically ahead of or behind the beat note by note—instead, randomized notes occur in slightly rushed or lagging groups, as would be the case with a real musician playing around a rigid tempo. The Random parameter can be set from 0%—for no randomization—to 50%, where randomized notes may be as much as half of the Quantize To value ahead of or behind the beat.

Shift

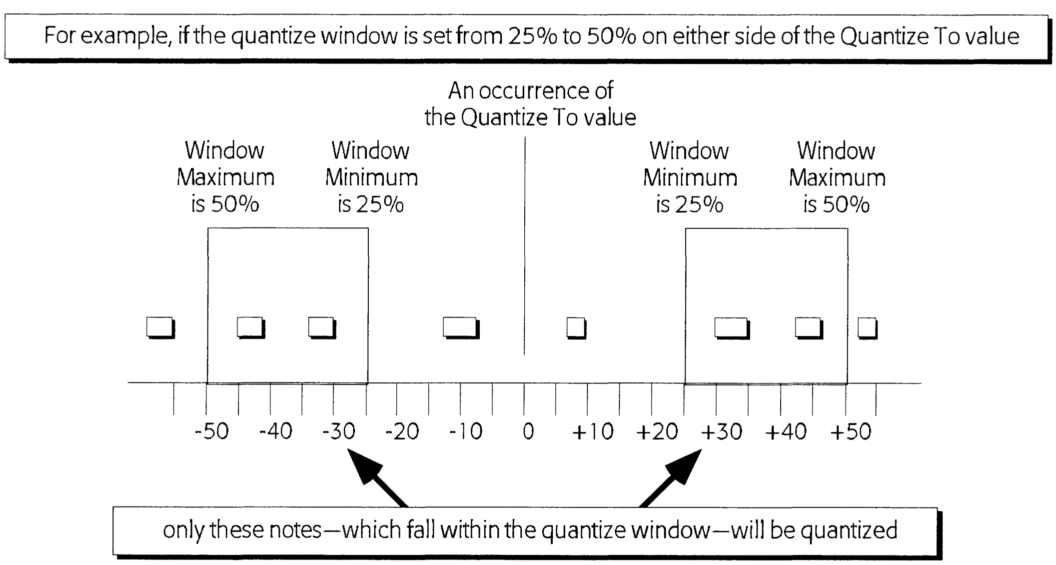
The Shift parameter allows you to move all of the music recorded on a track ahead in time, or back, by as much as the Quantize To value. Shift can be set anywhere from -100% to +100%. A Shift setting of 0% will not shift the music. A setting of -100% will make move it earlier in time by the amount set with the Quantize To parameter +100% will move it later by the same amount.

Low Key/High Key

The Low Key and High Key parameters allow you to select a note range to be quantized. All notes outside of this range will be left unaltered when you execute the quantize command. The Low Key parameter determines the lowest note that will be quantized, and the high Key parameter determines the highest.

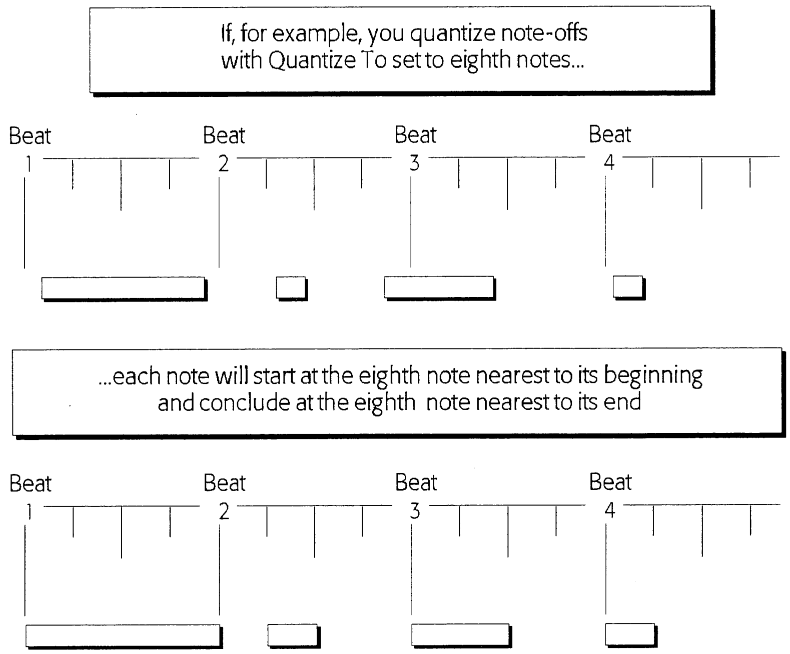
Window Minimum and Window Maximum

The Window Minimum and Window Maximum parameters allow you to determine by how much notes must deviate from the Quantize To value before they're subjected to quantization. This allows you to correct only the notes in a track that are clearly off, without affecting other unobjectionably placed notes. These parameters are expressed as percentages of deviation from the Quantize To value, and may be set from 0%—no deviation—to 50%, or halfway to the next occurrence of the value selected with the Quantize To parameter. The window created applies to notes that fall both ahead of and behind each occurrence of the value set with the Quantize To parameter. This is shown in the following illustration.



QuantizeNoteOffs

Quantizing typically affects the beginning of each note—the note-on. In the ASR-X, you can also quantize the ends of notes to the value set with the Quantize To parameter. This has the effect of changing the durations of the notes on the track to the length set with the Quantize To parameter. The QuantizeNoteOffs parameter may be switched on or off.



Move Note Offs

When you quantize the notes on a track, the beginning of each note is lined up to the Quantize To value. If the Move Note Offs parameter is set to On, the entire note will be moved according to the various quantizing parameters—and will remain the same length. If this parameter is switched off, only the beginning of the notes will be moved. The ends of the notes will be unchanged, and, therefore, the length of quantized notes will likely change as only their beginnings are moved to new positions.

Save quantize as?

The ASR-X allows you to save all of your quantization settings as one of four user templates that can be used throughout a work session (when you power down, the ASR-X RAM is cleared, including these templates). To save your current quantize settings, select USER TEMP 1, 2 3 or 4 and press the Yes button. To use one of your templates, return to the first quantizing sub-display—Template—and select the template you've saved.

Copy track?

The ASR-X offers the opportunity to copy a track, or elements of the track, with great specificity, and in a variety of ways to an intelligent assortment of destinations. The variety of tasks that can be accomplished using the track copy process utilize a shared set of parameters presented on this process's sub-displays. As you set each of the parameters, the ASR-X offers you additional options based on the values you've already selected—parameters are only available when they serve a sensible purpose.

Tip: The simplest rule of thumb to follow when using the track-copy process is to press the Yes button in response to the top-level question, set the first parameter displayed as you wish and then turn the Parameter knob clockwise to see if any other parameters are offered. At the point at which no additional parameters appear, press the Yes button to execute the process.

Scope

The Scope parameter is always provided when copying tracks. By setting its value, you determine what will be copied. You can select:

- Entire Track—to copy the whole track, including its track settings and all recorded data.
- TrkParams Only—to copy only the settings for the sequence's tracks, but no recorded data.
- TrkData Only—to copy the recorded data, but not the settings for the sequence's tracks.
- Within Region—to copy only the recorded data within the defined region. This value is available only when the Region RromPoint and/or ToPoint parameter is set to "On" (see "Using Regions" earlier in this chapter).

Paste

The Paste parameter determines how the copied material will be interact with the data already present in the location to which you'll be copying it. This parameter is not available when Scope=TrkParams Only. The Paste parameter can be set to:

- Append—to append the copied material to the end of the destination track.

When you append one track to another,

the track element you've chosen

is attached to the end of

any data on the destination track

resulting in a longer track:

The data on the destination track	The track element you've chosen

- Replace—to replace any data already present at the destination with the copied material.

When you copy using the Replace setting,
and there is

data in the destination location

it will be replaced by

the track element you've chosen

- Merge—to combine the copied material with anything already present at the destination.

When you merge one track with another,

the track element you've chosen

is combined with

any data on the destination track

resulting in a track that contains:

the data on the destination track and the track element you've chosen

(Destination) Seq

The Seq parameter allows you to choose a sequence other than the currently selected sequence as a destination for your copied material when Paste=Replace or Append. Turn the Value knob to select any of the sequences in the ASR-X's memory.

Destination Track

The Destination Track parameter allows you to select the track to which the track elements you've chosen will be copied. Turn the value knob to select the desired track in the sequence chosen with the (destination) Seq parameter above, or in the currently selected sequence if Paste=Merge.

DestTime

When Paste=Append, the DestTime parameter allows you to select a location within the destination track. The track you're copying will be appended to the destination track at the selected location. Turn the Parameter knob to select—and the Value knob to set—the desired bar, beat and clock value.

Erase track?

The ASR-X offers the opportunity to erase a track, or elements of the track, with great specificity. As you set each of the parameters provided on this process's sub-displays, the ASR-X offers you additional choices based on the values you've already selected.

To accomplish this:	After responding "Yes" to the top-level question, do this:
Erase all data on the track	Set Scope to "Entire Track" and press the Yes button.
Erase a specific MIDI controller's data	Set Scope to "Trk Data Only." Turn the Parameter knob clockwise and set Event to "Controller." Turn the Parameter knob clockwise and set Cntrl to the desired controller. Press the Yes button.
Erase a specific note range	Set Scope to "Trk Data Only." Turn the Parameter knob clockwise and set Event to "Note Range." Turn the Parameter knob clockwise and set Lo Key to the lowest note you want to erase. Turn the Parameter knob clockwise and set High Key to the highest note you want to erase. Press the Yes button.
Erase any other type of data	Set Scope to "Trk Data Only." Turn the Parameter knob clockwise and set Event to the type of data you want to erase. Press the Yes button.

Tip: When the Region FromPoint and/or ToPoint parameters are set to "On," the track-erasing process also offers a "Within Region" value for its Scope parameter. Use this value to perform all of the above-listed tasks—with the exception of erasing the whole track—on only the portion of the track that falls within the current region From and To points.

Tip: The simplest rule of thumb to follow when using this process is to press the Yes button in response to the top-level question, set the first parameter displayed as you wish and then turn the Parameter knob clockwise to see if any other parameters are offered. At the point at which no additional parameters appear, press the Yes button to execute the process.

Erase trk to end?

Answering "Yes" to this question provides a quick way to clear unwanted music from the end of the selected sequence. This process will erase everything recorded in the sequence that occurs after the currently selected location in the sequence. It's a good idea to hold down the Stop button and verify that you've got the desired location selected before answering Yes to this question. If it isn't, press the Stop button, and while holding it down, turn the Parameter knob to select the desired measure, press the arrow buttons to select a beat, and turn the Value knob to select the desired clock. When you let go of the Stop button, the sequence will go to the location you've dialed in.

Rename sequence?

Any sequence can be renamed at any time by pressing the Yes button in response to this question. Turn the value knob to select each character position and turn the Value knob to dial in the desired character.

Note: You can give a sequence a name of up to 20 characters, even though the sequencer displays on the ASR-X will not typically show all 20 of these characters. The extended name will be visible if you transport your ASR-X sequence—since it's a Standard MIDI File—via floppy to a computer. To view all 20 characters of a sequence's name on the ASR-X, hold down the Sequence Select button. You can turn the Value knob to view the long names of all onboard sequences.

Append sequence?

The ASR-X allows you to attach the currently selected sequence to the end of another sequence in memory. You can construct songs by appending the sections of the song to each other in the correct order. This process offers two sub-displays.

The number of the currently selected sequence

```
Append seq 2 to?
Seq=   1:    Shortie
```

The number and name of the target sequence

This display allows you to select the sequence to which the currently selected sequence will be attached.

The name of the selected sequence

```
Append seq 2 at?
DestTime=   1.01.001
```

Bar beat and clock

When this display is visible, you can at what point after the target sequence the currently selected sequence will be heard. This parameter defaults to attaching the selected sequence at the beginning of the measure following the target sequence's last recorded data. to change this setting, turn the Parameter knob to select the value you'd like to reset, and turn the Value knob to do so.

Tip: When two sequences with different time signatures and / or tempos are attached using the append process, both of their time signatures and / or tempos are retained.

Tip: When sequences are appended to each other, the insert effect and track settings from the destination sequence is used.

Copy this sequence?

The ASR-X allows you to copy the currently selected sequence to the next empty sequence location. You can choose to copy:

- Entire Seq—When this value is selected, everything in the sequence, including all of its tracks' settings and note data, will be copied.
- SeqParams Only—With this value, only the settings for the sequence's tracks will be copied.

Erase this sequence?

The "Erase this sequence?" process allows you to delete the currently selected sequence from memory. It also provides—when the Region FromPoint or Region ToPoint parameters are on (see "Using Regions" earlier in this chapter)—the opportunity to clear unwanted materials from the ends of the sequence. Two choices are offered when either of the above region parameters are set to "On":

- Entire Seq—With this value, the entire sequence will be erased when you press the Yes button.
- Outside Region—With this value selected, only recorded data that occurs earlier than the region From point (if the Region FromPoint parameter is set to "On") and/or after the region To point (if the Region ToPoint parameter is set to "On") will be erased.

Erase all sequences?

This question offer to clear your ASR-X sequencer memory. Press the Yes button to perform this process—a second display will ask you to confirm your decision. Press the Yes button again to erase all sequences currently in the ASR-X's memory.

The ASR-X Sequencer and MIDI

Recording into the Sequencer from an External MIDI Device

The ASR-X sequencer is always ready for recording from an external MIDI controller. Each of the 16 tracks in the sequencer receives data on the MIDI channel of the same number.

Incoming MIDI data received on:

MIDI channel 1	will be recorded on	track 1	MIDI channel 9	will be recorded on	track 9
MIDI channel 2	will be recorded on	track 2	MIDI channel 10	will be recorded on	track 10
MIDI channel 3	will be recorded on	track 3	MIDI channel 11	will be recorded on	track 11
MIDI channel 4	will be recorded on	track 4	MIDI channel 12	will be recorded on	track 12
MIDI channel 5	will be recorded on	track 5	MIDI channel 13	will be recorded on	track 13
MIDI channel 6	will be recorded on	track 6	MIDI channel 14	will be recorded on	track 14
MIDI channel 7	will be recorded on	track 7	MIDI channel 15	will be recorded on	track 15
MIDI channel 8	will be recorded on	track 8	MIDI channel 16	will be recorded on	track 16

To record MIDI data on one or more tracks, send the data on its like-numbered channel, and use the same techniques you'd use when recording from the ASR-X pads.

Note: Since recording always occurs on the track, or tracks, corresponding to the MIDI channel(s) on which data is received, keep in mind that the track that's currently selected on the ASR-X has nothing to do with which track(s) will be recorded. Set your incoming MIDI channels/tracks carefully so that you don't inadvertently end up recording over tracks you intend to keep.

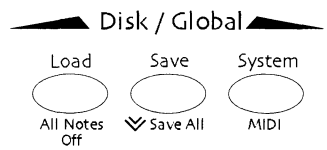
Note: The undo function is available after the recording of MIDI data only if a single track was recorded and the track was also selected on the ASR-X.

Transmitting MIDI Data from the Sequencer

MIDI transmission from the each of the ASR-X's tracks is enabled or disabled via the setting of the TrackMIDIOut track parameter. See Chapter 2 for a description of the TrackMIDIOut parameter.

7 Disk/Global

The Disk/Global Controls



The Disk/Global area of the ASR-X front panel contains two groups of controls that share the common goal of performing operations that affect the entire ASR-X.

The Disk-related buttons—Load and Save—are used for:

- loading files from disk into the ASR-X RAM.
- saving files from RAM to disk.

Operation of the Disk controls is described later in this chapter in "The Disk Buttons."

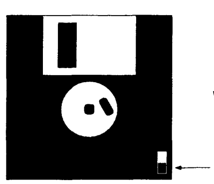
The System/MIDI button provides access to tools for:

- customizing the system-wide behavior of the ASR-X to suit the way you create music.
- setting up the overall MIDI functionality of the ASR-X.
- getting the most out of the ASR-X RAM.
- performing various disk-file management functions.

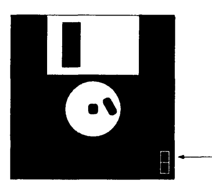
The System/MIDI tools are described later in this chapter in "The System/MIDI Button."

The Floppy Disk Drive

The ASR-X contains a built-in floppy disk drive. The ASR-X floppy drive can read or write to any 3.5-inch high-density or double-density floppy disk. Floppy disks can be write-protected so that the files they contain cannot be accidentally written over. If you plan to save ASR-X files to a floppy, make sure that it's write-protect feature is not engaged. You can tell if a disk is write-protected by flipping it over (so that it's label-side down) and examining the small window in its lower right-hand corner.

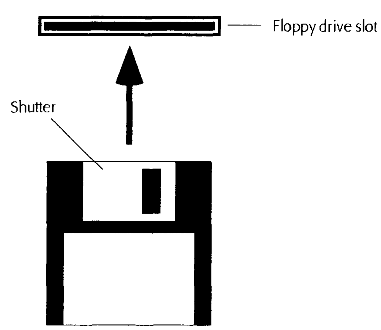


If the tab is in the down position, the write-protect window is open, and the disk is write-protected. It can only be read.

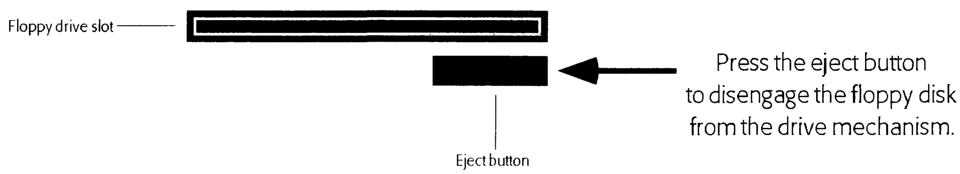


If the tab is in the up position, if the write-protect window is closed, and the disk is not write-protected. It can be both written to and read.

A disk is inserted into the drive—label-side up, with its shutter window to the right—by sliding the floppy into the drive's slot until the drive grabs the disk and seats it in the drive mechanism.



Disk are removed from the floppy drive by pressing on the button on the face of the drive—this causes the floppy to pop out far enough from the drive mechanism that it can be grasped and removed.



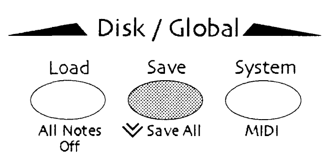
Warning: The floppy disk drive is a sensitive piece of equipment and, as such, should be approached with a measure of care. See "The Care and Feeding of the Floppy Disk Drive" at the front of this manual to learn the proper way to treat a floppy drive.

The Disk Buttons

Everything you do on the ASR-X can be stored easily to disk and loaded back into the ASR-X whenever you wish. ASR-X disks use a standard DOS format, so ASR-X sequence and wave files can be loaded into a Macintosh or PC-compatible computer for further work. Most disk operations are performed after pressing the Disk/Global Save or Load buttons. The System/MIDI button provides access to a collection of disk utilities—see "Access disk utils?" later in this chapter.

Note: Before you can save ASR-X files to disk, the disks must be properly formatted. This can be accomplished on a computer or on the ASR-X. To learn how to format a disk on the ASR-X, see "Access disk utils?" later in this chapter.

The Save Button



To save an ASR-X file to disk, you:

1. insert a DOS-formatted floppy into the ASR-X disk drive.
2. press the Disk/Global Save button.
3. select the type of file to be saved.
4. name the file.
5. press the Yes button to finish saving the file.

Note: If the files or files you're saving require more space than is available on a single disk, the ASR-X will ask if you're ready to proceed, and ask you to insert additional disks as needed.

File Types that can be Saved

When you press the Disk/Global Save button, the ASR-X reads the directory of the disk in the drive, and presents you with a list of the types of files that can be saved to disk.

The type of file to be saved

To choose a type of file to save, turn the Parameter knob to choose:

- ALL-SESSION—The ALL-SESSION file saves everything currently in RAM as files with a common name. An ALL-SESSION file saves:
  - an ALL-SOUNDS file that contains all of the sounds currently in RAM.
  - all of the waves currently in RAM as separate 1-AIF WAVE files.
  - an ALL-SEQS file that contains all of the sequences currently in RAM.
  - a SYSTEMSETUP file that stores your System/MIDI, Resampling Setup and Click settings.

Tip: Double-click the Save button to get to the ALL-SESSION saving display at any time.

- ALL-SEQS—An ALL-SEQS file saves all of the sequences currently in RAM as a single disk file.
- 1-SEQUENCE—A 1-SEQUENCE file saves the selected sequence to disk as a Standard MIDI File (SMF). Each track contains SysEx data that allows the track's parameter settings to be reloaded from disk or transmitted to the ASR-X via MIDI from an external sequencer.
- ALL-SOUNDS —An ALL-SOUNDS file saves all of the sounds currently in RAM. The ALL-SOUNDS file type also saves any waves currently in memory as 1-AIF WAVE files.
- 1-SOUND—The 1-SOUND file type saves the currently selected sound to disk. If the sound is playing a wave currently in RAM, the wave is saved to disk as well, as an 1-AIF WAVE file.

Tip: When you save a kit as a 1-SOUND file, all of the sounds and waves it uses are saved as well.

- SYSTEMSETUP—A SYSTEMSETUP file saves the current System/MIDI, Resampling Setup and sequencer Click settings.

Tip: You can save a SYSTEMSETUP to floppy that restores your settings automatically on power-up by naming the file "SYSSETUP"and turning on the ASR-X with the disk already in the drive.

Saving the Contents of the Scratch Pad to Disk

After you've pressed the Save button and the ASR-X has read your floppy's directory, you can save the contents of the Scratch Pad to disk by pressing the Scratch Pad—the ASR-X will create a sound that plays the wave(s) in the Scratch Pad and that you can then save to disk using the normal saving procedure.

Naming Disk Files

Each file you save to disk should be given a unique name. The ASR-X will not allow two files with the same name to exist on a single floppy—if you save a file that has the same name as a file already on the disk, the older file will be replaced by the new one. This allows you to easily update files by simply resaving them to floppy without re-naming them. But be careful—it also means that you can unintentionally erase a file you meant to keep.

To name a disk file you're saving, press the left and right Select Track buttons to select each of its eight character locations.

```
Save to disk?
ALL-SESSION:SESSION
```

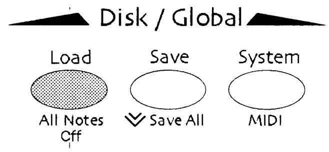
The character currently selected for editing is underlined

Turn the Value knob to choose the desired character for each location. If you're naming an ALL-SESSION file, each of its component files will share the name you designate.

When you've finished naming your file, press the Yes button to save the file to disk.

Note: When you name a disk file, you're not changing the name of the item it contains.

The Load Button



To load a file from disk into the ASR-X, you:

1. insert a DOS-formatted floppy into the ASR-X disk drive.
2. press the Disk/Global Load button.
3. select the type of file to be loaded.
4. select the specific file to be loaded.
5. if you're loading a 1-SOUND file, select the RAM location into which you want to load the sound.
6. press the Yes button to load the file.

Note: If files you're loading are on multiple disks, begin loading the files from the first of these disks—the ASR-X will ask for each disk as it needs it.

File Types that can be Loaded

When you press the Disk/Global Load button, the ASR-X reads the directory of the disk in the drive, and presents you with a list of the types of files that can be loaded from disk.

```
Load from disk?
ALL-SESSION:SESSION
```

The type of file to be loaded

To choose a type of file to save, turn the Parameter knob to choose:

- ALL-SESSION—The ALL-SESSION file type restores all of the items that were in RAM when the file was saved. It loads:
  - an ALL-SOUNDS bank.
  - the 1-AIF WAVE files played by sounds in the ALL-SOUNDS bank.
  - an ALL-SEQS file.
  - a SYSTEMSETUP file.
- ALL-SEQS—An ALL-SEQS file loads the sequences that were in RAM when the file was saved.
- 1-SEQUENCE—The 1-SEQUENCE file loads a Standard MIDI File (SMF) created on the ASR-X or any SMF-compliant sequencer.
- ALL-SOUNDS —An ALL-SOUNDS file restores all of the sounds that were in RAM when the file was saved, as well as any 1-AIF WAVE files required to produce the sounds.
- 1-SOUND—The 1-SOUND file type loads a single sound, as well as any 1-AIF WAVE files required to produce the sound.
- 1-WAV WAVE—A 1-WAV WAVE file loads a .wav-format wave file created on an external device and saved to the ASR-X floppy. 1-WAV WAVE files are loaded directly into the Scratch Pad, from where they can be sent to pads and incorporated into ASR-X sounds.

Note: The ASR-X converts .wav files to AIF format as it loads them into the ASR-X.

- 1-AIF WAVE— A 1-AIF WAVE file loads an AIF-format wave file created on the ASR-X or an external device and saved to the ASR-X floppy. 1-AIF WAVE files are loaded directly into the Scratch Pad, from where they can be sent to pads and incorporated into ASR-X sounds.

- SYSTEMSETUP—The SYSTEMSETUP loads the System/MIDI, Resampling Setup and sequencer Click parameter settings that were in place when the file was saved.
- ASR-SND—An ASR-SND file loads a sound saved to a high-density (HD) floppy from an ENSONIQ ASR-10 or ASR-88. Once imported, an ASR-SND becomes an ASR-X standard sound.

A Note About Imported ASR-10 and ASR-88 Sounds

Most ASR-10 and ASR-88 features have counterparts in the ASR-X voice architecture that are translated when a sound is imported. A few features lack such a counterpart, however:

- A-B FADE IN-TO, C-D FADEOUT-TO, and FADECURVE parameters settings are not imported.
- ASR-10/88 pitch tables are not imported.
- Only the START, LPSTRT-X and TRANSWAV loop modulators are translated.
- All ASR-10 and ASR-88 sounds are set to the MediumReverb FX Bus when they're imported.
- The ASR-X VelLevels Amount settings for Envelopes 1, 2 and 3 are derived by averaging the HARD VEL LEVELs 1 and 2 and the SOFT VEL LEVELS 1 and 2 for each envelope in the original ASR-10/88 sound.
- When a layer's LYR GLIDEMODE parameter is set to any value other than "OFF" in an ASR-10/88 sound, the layer's Glide Mode is set to "On" and its Voice Mode to "Mono" when it's imported.

Selecting an Individual File to be Loaded

Once you've selected the type of file to be loaded, turn the Value knob to select a specific file. Once you've chosen the file you want to load, press the Yes button, and the ASR-X will load the file.

Note: If a file you're loading was created on a Macintosh or Windows '95-or-later PC and is named using more than the eight characters supported by DOS, the file's name will be truncated according to the following rules: if the file was named on a Macintosh, an exclamation point will appear at the beginning of its name; if it was created on a PC-compatible, the last two characters displayed will be an arrow and a digit.

Selecting a Location into which a Sound will be Loaded

When loading a 1-SOUND file, an additional display appears when you press the Yes button after selecting the file to be loaded.

The currently selected RAM bank and program number

```
Load into  RAM00:000
RAM-SND :Hot 2 Touch
```

The name of the sound currently in the selected location

When this display appears, you can turn the Value knob to select any location in either of the ASR-X's two RAM sound banks—RAM 00 and RAM01. If you select a location that already contains a sound, the sound you're loading will replace the sound currently in the location. Unused locations show **EMPTY** instead of a sound name.

When you've selected a location, press the Yes button to finish loading the sound.

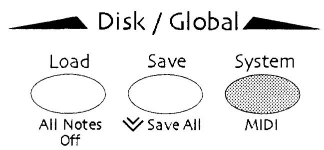
Folders/Directories and File Extensions

ASR-X users who load their ASR-X floppies into their computer disk drives will see the folder/directory structure in which the ASR-X saves its files. When files are saved to disk from the ASR-X, all of the folders/directories needed by the ASR-X are created on the disk. Each file type has its own 3-character DOS extension. As in any DOS-based system, the ASR-X identifies disk files by this extension.

Folder/Directory Name	What's Stored There	File Extensions
SESSION	ALL-SESSION files	.ssx
	SYSTEMSETUP files	.spb
BANKS	ALL-SOUNDS files	.sbx
SEQUENCE	sequence banks files associated with ALL-SESSION files	.mfb
	1-SEQUENCE files	.mid
SOUNDS	1-SOUND files	.sou
WAVES	1-AIF WAVE files	.aif
	alias files that prevent duplicate saving of 1-AIF WAVE files	.als

Note: Though it's desirable that files be stored in these folders/directories, the ASR-X can "see" and load files from anywhere on a disk.

The System/MIDI Button



The System/MIDI button provides access to parameters and tools for setting up your overall ASR-X environment and for memory and disk file management. These various items are grouped into several broader categories, each of which is accessed by pressing the System/MIDI button, turning the Parameter knob to view and then pressing the Yes button to respond to a displayed question. The questions are:

- Set system prefs?—Pressing the Yes button in response to this question causes the ASR-X to display parameters that determine what happens when you select new track sounds, the velocity sensitivity of the pads, and the behavior of the Patch Select buttons and any connected foot switches.
- Alter system pitch?—Pressing the Yes button in response to this question causes the ASR-X to display parameters that determine the ASR-X's response to received MIDI pitch bend messages, that allow you to fine-tune the overall pitch of the ASR-X, and set the ASR-X's tuning table.
- Edit MIDI settings?—Pressing the Yes button in response to this question causes the ASR-X to display parameters that determine the base MIDI channel for the ASR-X, its response to and transmission of sequencer synchronization data, its response to several types of received MIDI messages, its System Exclusive ID number and allows you to define four special system-wide real-time MIDI controllers.
- Access disk utils?—Pressing the Yes button in response to this question causes the ASR-X to display an assortment of utilities for formatting floppy disks, erasing disk files, renaming disk files, determining how files will be displayed, and also provides a read-out of the free space available on the disk currently inserted in the floppy drive.
- Enter MemoryManager?—Pressing the Yes button in response to this question causes the ASR-X to display informational displays showing how much free memory is currently available and the name of an installed expansion board, as well as a set of tools for onboard memory management that provide the ability to clear the onboard memory banks, erase or rename a sound and change a sound's SoundFinder category.

Some of these features are accessed by responding "Yes" to questions posed on sub-displays under the top-level question. To exit from a sub-display or from the System/MIDI displays altogether, press the Exit button each time you want to move up a level back out to the ASR-X front panel.

Each of these features is described in detail in the following section.

Set system prefs?

Touch Curve

The ASR-X pads are velocity-sensitive, responding with tremendous accuracy to how hard or soft you play. The Touch Curve parameter allows you to adjust the velocity response of the pads to match your playing style and technique. There are six available Touch Curve settings:

- Table-1—With this setting, the pads offer an easily controllable, compressed dynamic response. Table-1 is optimized for players with a light touch.
- Table-2—This setting is similar to Table-1, but designed for players who play hard.
- Table-3—With this setting, the pads offer a full dynamic range. This will feel good to musicians who have a high degree of control over the force with which they play. Table-3 is optimized for players with a light touch.
- Table-4—This setting is similar to Table-3, but designed for players who play hard.
- Fixed 64—This setting causes the pads to always respond as if you've hit them precisely half as hard as they can be hit. This can be useful in simulating vintage synthesizers with no velocity control.
- Fixed127—This causes the pads to always respond as if you've hit them as absolutely hard as they can be hit. This is good for drum/percussion parts in which you don't want dynamic changes.

Patch Selects

The ASR-X Patch Select buttons can be set to operate in one of two modes, each of which is invoked by one of two values for the Patch Selects parameter:

- Live—With this setting, the Patch Select buttons are momentary switches. This means that the sound changes caused by pressing either, or both, of the Patch Select buttons lasts for only as long as the button is physically held down.
- Held—With this setting, playing a note from the pads or via MIDI locks in the Patch Select button or buttons being held when the note is played. To release the button(s), tap either of the Patch Select button; subsequent notes will sound as they should when no Patch Select buttons are being pressed.

FtSw L and FtSw R ("Foot Switch Left and Foot Switch Right")

The ASR-X can accommodate either a dual foot switch with two pedals—such as the ENSONIQ SW-10—or a single foot switch with one pedal—such as the ENSONIQ SW-2 or SW-6. The FtSw L and FtSw R parameters allow you to assign a broad range of functions to any pedals you're using. When a dual foot switch is connect, both the FtSw L and FtSw R parameters are active, controlling the behavior of the left and right pedals, respectively. When a single foot switch is connected, FtSw R controls its behavior.

Tip: To learn how to connect foot switches to the ASR-X, see Chapter 1.

FtSw L and FtSw R can be set to any of the following values:

- Unused—pressing the pedal will have no effect.
- Sustain—holding the pedal will cause notes to continue sounding after the key is released, much like the sustain pedal on a piano.
- Sostenuto—any keys that are held down when you press the pedal will sustain until you release the pedal; keys pressed down after you press the pedal will not sustain.
- SysCTRL1—pressing the pedal down will send a value of 127 to any aspect of a sound or effect that's modulated by the controller designated as CTRL1; releasing the pedal will send a value of 0 to any aspect of a sound or effect that is modulated by the controller designated as CTRL1. (For details on setting the CTRL1 parameter and descriptions of CTRL1 settings, see "CTRL1, CTRL2, CTRL3 and CTRL4" later in this chapter.)
- SysCTRL2—pressing the pedal down will send a value of 127 to any aspect of a sound or effect that's modulated by the controller designated as CTRL2. This functions in the same manner as the SysCTRL1 value described above.

- SysCTRL3—pressing the pedal down will send a value of 127 to any aspect of a sound or effect that's modulated by the controller designated as CTRL3. This functions in the same manner as the SysCTRL1 value described above.
- SysCTRL4—pressing the pedal down will send a value of 127 to any aspect of a sound or effect that's modulated by the controller designated as CTRL4. This functions in the same manner as the SysCTRL1 value described above.
- Play / Stop—pressing the pedal will have the same effect as pressing the Stop button if a sequence is playing; it will have the same effect as pressing the Play button if a sequence isn't playing.
- PlayTop / Stop—pressing the pedal once will have the same effect as double-clicking the sequencer Play button; pressing it twice will stop the sequence if it's playing.
- RecPlay / Stop—pressing the pedal will start recording on the currently selected track. If the sequencer is already recording, pressing the pedal down will stop recording. This setting can be used for punching ins on a track.
- Record—pressing the pedal will have the same effect as pressing the sequencer Record button.
- Stop—pressing the pedal will have the same effect as pressing the sequencer Stop button.
- Rewind—pressing the pedal acts like pressing the sequencer Stop and Record buttons together.
- FastForward—pressing the pedal acts like pressing the sequencer Stop and Play buttons together.
- Mute—pressing the pedal will have the same effect as pressing the track Mute button.
- Step Advance—pressing the pedal will advance a track currently being step-recorded by one step.

Warning: If you're using a single foot switch, FtSw L should always be set to "Unused."

AutoSelect FXBus

The AutoSelect FXBus parameter allows you to program the ASR-X to assign an appropriate effect to a sound when it's chosen for use by a track. Each sound in the ASR-X has a parameter called the Alt Bus that assigns it to a non-insert effect routing. If AutoSelect FXBus is set to "On":

- when you select a sound that contains an insert effect for use by a track other than the Insert Control Track, the sound is routed to the FX bus designated by its Alt Bus value.
- when you select a sound that doesn't contain an insert effect for use by any track, the sound is routed to the FX bus designated by its Alt Bus value.

When the AutoSelect FXBus parameter is set to "Off," the track's FX Bus routing is unchanged when a new sound is selected for the track.

Tip: To learn about the Insert Control Track , see Chapter 4. To learn how to program the Alt Bus for sounds you've sampled, see Chapter 3.

Track ParamReset

The Track ParamReset parameter determines whether or not certain track parameters will be reset to their default values when a new sound is selected for a track. This helps ensure that each sound will be heard when it's selected for a track as its programmers intended; on the other hand, if you've set a track's parameters just so, you may want them to remain in place when a new sound is selected. A complete list of the affected track parameters—as well as the values to which they're reset—can be found in Chapter 9. When Track ParamReset is set to "On," these parameters will be reset whenever a new sound is selected for a track; when it's set to "Off," each track's parameters will be unaffected by the selection of a new sound for the track.

Auto-Zero Cross

The Auto-Zero Cross parameter enables or disables the ASR-X's zero-crossing search feature. This feature automatically offers locations within waves that are most likely to produce trouble-free loops when the Loop Start and Loop End Pad parameters (see Chapter 3) are adjusted.

Alter system pitch?

The System Pitch Bend Setup

A Pitch Bend Wheel is a spring-loaded wheel typically located to the far left of a MIDI keyboard. It's most commonly used to bend the pitch of notes up or down by pushing the wheel forward (up) or pulling it back (down). Some manufacturers employ a left/right scheme.

ASR-X sounds are programmed to respond to MIDI Pitch Bend messages in ways appropriate to the sound. The ASR-X also offers a system pitch bend setup that can be accessed by setting any track's Pitch Bend Up and Pitch Bend Down parameters to the "Sys" setting (see Chapter 2). There are three parameters that determine the behavior of the system pitch bend setup.

The system Pitch Bend Up parameter can be set to:

- 1-12dn or 1-12up—the pitch of any sound on a track whose Pitch Bend Up parameter is set to "Sys" will be lowered or raised by the number of equal-temper semitones set here when a Pitch Bend value of 127 is received.
- Off—the pitch of any sound on a track whose Pitch Bend Up parameter is set to "Sys" will ignore MIDI messages received from a Pitch Bend Wheel pushed forward.

The system Pitch Bend Up parameter can be set to:

- 1-12dn or 1-12up—the pitch of any sound on a track whose Pitch Bend Down parameter is set to "Sys" will be lowered or raised by the number of equal-temper semitones set here when a Pitch Bend value of 0 is received.
- Off—the pitch of any sound on a track whose Pitch Bend up parameter is set to "Sys" will ignore MIDI messages received from a Pitch Bend Wheel pulled all the way back.

The PitchBendMode parameter unlocks a powerful feature that allows you to decide which notes will be affected by received Pitch Bend messages. It can be set to one of three values:

- Normal—received Pitch Bend messages will affect all notes currently sounding.
- Held—received Pitch Bend messages will affect only those notes sounding from keys which are being physically held down. Notes held with the sustain pedal or in their release stage will remain at their original pitch.
- Prog—the system Pitch Bend will respect the Normal/Held settings programmed into sounds using the system pitch bend set-up.

Tip: This PitchBendMode feature can be used to create guitar-style pitch bends or to "paint" with pitch, leaving different notes sustaining at different pitches.

Fine Tuning

The Fine Tuning parameter allows you to raise or lower the overall pitch of ASR-X sounds in cents—100ths of a semitone. This parameter can lower pitch by as much as -50 cents or raise it by up to +49 cents.

PitchTbl

The intervals (or relationships) between notes in a scale can be altered to create special pitch tables. The ASR-X pitch tables have a tuning resolution of 256 cents per semitone. The default pitch table is "EqualTemper," the western 12-tone equi-tempered pitch table. However, you can select from a large assortment of traditional, modern, ethnic, and exotic pitch tables in the ASR-X. A detailed list of these pitch tables can be found in Chapter 9.

The ASR-X also provides a RAM location for a custom pitch table, and supports the MIDI pitch table Bulk Tuning Dump and Single Note Tuning Change standards. If you've got the appropriate computer program, you can create your own pitch tables, and transmit them to the MR-61 and MR-76 via SysEx. This feature is described in detail in Chapter 9.

The ASR-X provides a system pitch table that can be accessed by setting any track's PtchTbl parameter to the "Sys" setting (see Chapter 2). The System/MIDI parameter allows you to select the tuning that will be used by the system pitch table. Any of the onboard pitch tables, or the RAM pitch table, can be selected.

Edit MIDI settings?

Local-Off Operation of the ASR-X

The Pads Play Local and Local Off Channel parameters allow you to disable the ASR-X's response to its pads, Patch Select buttons and foot switch while using them to send data to an external MIDI sequencer—the external sequencer can then send the data back to the ASR-X sounds via MIDI. Turning off the ASR-X's response to the pads when working with an external sequencer ensures that what you hear is being correctly captured and played by the external sequencer; it also prevents the accidental simultaneous playing of ASR-X sounds from two MIDI sources.

The Pads Play Local parameter enables or disables the ASR-X's response to the pads, Patch Select buttons and foot switch. It can be set to:

- On—causing the pads, Patch Select buttons and foot switch to function normally in the playing, construction and editing of ASR-X sounds. MIDI data will be transmitted on the currently selected track according to the setting of the TrackMIDIOut track parameter (see Chapter 2).
- Off—the pads, Patch Select buttons and foot switch function only as MIDI controllers transmitting data on the MIDI channel determined by the Local Off Channel parameter.

The Local Off Channel parameter determines the MIDI channel on which the ASR-X will transmit data from the pads, Patch Select buttons and foot switch when the Pads Play Local parameter is set to "Off."

ClockSource

Various activities within the ASR-X depend on a timing source, or clock. Obviously, the sequencer needs such a reference; in addition, synchronized LFOs and noise generators within sounds, and certain effects such as delays, also depend on a timing reference. The ASR-X contains its own internal clock—it can also use timing information received from an external MIDI device that transmits MIDI clocks. The ClockSource parameter determines which timing reference will be used. The parameter can be set to:

- Internal—so that ASR-X's internal clock is used. When this is the case, the sequencer tempo sets the timing of synchronized LFOs, noise generators and effects.
- MIDI—so that received MIDI clocks control the timing of the sequencer, LFOs, noise generators and effects. With this setting, the ASR-X responds to Song Position Pointer messages.

Xmit MIDI Clocks

The ASR-X can generate MIDI clocks to provide a timing reference for external MIDI devices, allowing them to be synchronized to the ASR-X sequencer. The XMit MIDI Clocks parameter enables or disables transmission of MIDI clocks when the ASR-X is running. It may be set to "Off" or "On." This parameter also enables or disables transmission of MIDI Song Position Pointer messages from the ASR-X sequencer.

Bank&ProgChgRecv

Each track has parameters that allow you to enable or disable the track's response to Bank Select and Program Change messages. The Bank&ProgChgRecv parameter provides a master switch for this feature, simultaneously enabling or disabling all 16 tracks' response to Bank Select and Program Change messages. The parameter may be set to "Off" or "On."

ResetControlRecv

The ResetControlRecv System parameter allows you to determine how the ASR-X will respond to Reset All Controllers MIDI messages. When the parameter is set to "On," and the ASR-X receives a Reset All Controllers message, it will return all of its real-time controllers and any parameters that respond to MIDI controllers to their default values, clearing up any hung values or unexpected settings. When

ResetControlRecv is set to Off, the ASR-X will not respond to Reset All Controllers messages. For more information on the ASR-X's response to Reset All Controllers messages, see "Reset All Controllers (MIDI controller 121) Reception Behavior" in Chapter 9.

AllNotesOff Recv

The ASR-X can respond to All Notes Off (controller 123) and All Sounds Off (controller 120) MIDI control messages. When the ASR-X receives either of these messages, any of its notes that are currently sounding are silenced. The AllNotesOff Recv System parameter is a combined filter for both of these messages. When it's set to "On," the ASR-X will respond to them—when it's set to "Off," it will ignore them.

SysEx Device ID

When sending System Exclusive messages to the ASR-X in a MIDI system that contains more than one ASR-X, it's vital to have a way of distinguishing one ASR-X from another. To accomplish this, each ASR-X should be set to its own SysEx ID number. The SyEx Device ID parameter may be set from 000 to 127.

CTRL1, CTRL2, CTRL3 and CTRL4

The ASR-X responds to the following real-time MIDI controllers and messages:

- Data Entry Slider
- Pitch Bend Wheel
- Mod Wheel
- Foot Pedal
- Sustain/Sostenuto pedals
- MIDI Volume messages
- MIDI Pan messages
- MIDI Expression messages

In addition, you can define four additional real-time MIDI controllers: CTRL1, CTRL2, CTRL3 and CTRL4. These can be assigned to any MIDI controller number, and can used to modulate the ASR-X sounds or effects. (see Chapters 3 and 4, respectively, to learn about modulation). Each track offers parameters for enabling or disabling the track's response to any of the four CTRLs. See Chapter 2.

When the ASR-X is shipped from the factory, the CTRLs are set to the following default values:

- CTRL1 is assigned to Breath Controller (MIDI controller #002).
- CTRL2 is assigned to FXControl1 (MIDI controller #012). This is the controller transmitted by the FX-SW modulator on ENSONIQ's TS-10 and TS-12.
- CTRL3 is assigned to PatchSelct (MIDI controller #070). The Patch Select buttons can be used for real-time modulation when you select CTRL3 as a sound or effect modulator.
- CTRL4 is assigned to Timbre (MIDI controller #071).

Tip: Some of the ENSONIQ-programmed sounds in the ASR-X use CTRL3 as the mechanism by which they respond to the front-panel Patch Select buttons. If you'd like to use an external MIDI controller—such as a continuous controller—instead of the Patch Select buttons, you can change CTRL3 to any controller number that's convenient. Remember, however, that this will have the effect of disabling the Patch Select buttons on the ASR-X for these sounds.

Access disks utils?

Note: The following options are available only when a floppy disk is in the ASR-X disk drive.

Format floppy disk?

The ASR-X can read and save files to DOS-formatted high-density (HD) and double-density (DD) floppy disks. Before a disk can be used by the ASR-X to store data, it must be in DOS format. You can use your ASR-X to format any HD floppy disk that's been properly inserted into the drive. When you press the Yes button in response to "Format floppy disk?" the ASR-X presents a second display as a safety feature to make sure you're prepared to erase the floppy in the drive.

Warning: Make sure that each disk you format doesn't contain anything you want to keep. All data on a disk will be lost when the disk is formatted.

You can format DD disks on any device capable of DOS formatting using the DOS command "format (the letter designator of your floppy drive): /F:720".

Erase disk files?

The "Erase disk files?" feature lets you permanently delete any file from a floppy inserted in the ASR-X floppy drive. When you answer the question by pressing the Yes button, the following display appears:

```
Erase this file?
SESSION    :APRIL24
```

The type of file selected — The file to be erased

Turn the Parameter knob to select the type of file you want to erase, and then the Value knob to select a specific file. When you've selected the file you want to delete from the disk, press the Yes button. A display will appear asking you if you're sure—press the Yes button to erase the file.

Rename disk files?

The ASR-X allows you to rename any files you've already saved to disk by pressing the Yes button in response to "Rename disk files?" When you've done this, the display will show:

```
Rename this file?
SESSION    :APRIL24
```

The type of file selected — The file to be renamed

Turn the Parameter knob to select the type of file you want to erase, and then the Value knob to select a specific file. When you've selected the file you want to rename, press the Yes button. The ASR-X will present the file-naming display:

The file's current name

```
Rename HITSTUFF as?
New Name=   HITSTUFF
```

Spell the file's new name here

Turn the Parameter knob to select each character location in turn—the selected character will be underlined—and then turn the Value knob to select the desired character for that location. When you've finished spelling out the new name for the file, press the Yes button to write the name to disk.

Note: When you rename a sound or sequence file, you're renaming only the disk file for the sound or sequence—its original upper-and-lower case name as shown in RAM is unaltered.

Set disk prefs/info?

Answering "Set disk/prefs info?" with a press of the Yes button results in the presentation of two sub-displays, either one of which can be selected by turning the Parameter knob.

The Directory Sorted parameter allows you to set whether or not the files on the floppy disk will be displayed in alphabetical order or not. When Directory Sorted is set to "Yes," within each file type, individual files will be presented alphabetically. when the parameter is set to "Off," they'll be displayed in the order in which they were saved to disk.

The Bytes Free display shows how much free space is currently available on the floppy in the drive. A freshly formatted high-density disk has about 1,400 bytes of space available; a just-formatted double-density disk about 720 bytes.

Enter MemoryManager?

Show free memory?

The ASR-X Memory Manager provides a handy way to keep track of how much RAM is available for sequences and waves. Pressing the Yes button in response to "Show free memory?" reveals two read-only sub-displays:

- Sound & Wave RAM—This shows the amount of free RAM currently available for sounds and waves. The amount displayed will depend on the amount of memory installed in your ASR-X. A stock ASR-X will show 37,778 bytes free when all of its memory is available.
- Sequencer RAM—This shows the amount of free memory currently available for sequencing. When no sequences have been created or loaded from floppy, this display shows 229,488 bytes free.

Erase memory banks?

The ASR-X Memory Manager allows you to easily clear the sound and wave RAM, or the sequencer RAM. Pressing the Yes button in response to "Erase memory banks?" reveals the following sub-display, from which you can turn the Value knob to select either "All Sounds&Waves" or "All Sequences":

```
Erase memory banks?
* All Sounds&Waves *
```

What will be erased is shown here

When you've selected the type of RAM you'd like to erase, press the Yes button.

Erase sound?

The ASR-X allows you to erase any sound from RAM by pressing the Yes button in response to "Erase sound?" When you do so, the display will show:

The amount of memory allocated to the selected sound

The sound's bank and program number

```
Erase  0.4k?  00:000
SOUND    :Garbage Kit
```

The sound to be erased

Turn the Value knob to select the sound you'd like to delete, and press the Yes button to erase it.

Rename sound?

To rename a sound in RAM, press the Yes button is response to "Rename sound?" The following display will appear:

The sound's bank and program number

```
Rename ?      00:000
SOUND    :Rock Kit_01
```

The sound to be renamed

Turn the Value knob to select the sound you'd like to rename, and then press the Yes button to invoke the sound-renaming display:

The sound's current name

```
Old Name:Rock Kit_01
New Name:Rock Kit_01
```

The new name will be spelled out here

Turn the Parameter knob to select each character location in turn—the selected character will be underlined—and then turn the Value knob to select the desired character for that location. When you've finished spelling out the new name for the sound, press the Yes button to finish renaming it.

Change sound type?

The MemoryManager allows you to change the SoundFinder category to which a sound is assigned. Press the Yes button is response to "Change sound type?" and the following display appears:

```
Change sound type?
SOUND   :Rock Kit_01
```

The sound you want to assign to a different SoundFinder category

Turn the value knob to select the sound whose category you'd like to change, and press the Yes button. The displays shows:

```
Change Rock Kit_01?
SoundFinder= *CUSTOM
```

The SoundFinder category to which the sound is currently assigned

Turn the Parameter knob clockwise to reveal the FinderPref parameter, which allows you to assign the selected sound to the USER-SND and/or DEMO-SND SoundFinder types. Turn the Value knob to select:

- None—to assign the selected sound to neither the USER-SND or DEMO-SND category.
- DEMO-SND—to assign the selected sound to the DEMO-SND category.
- USER-SND—to assign the selected sound to the USER-SND category.
- USER+DEMO—to assign the selected sound to both USER-SND and DEMO-SND categories.

Turn the Value knob to select the desired SoundFinder designations, and press the Yes button to re-assign the sound to the new categories.

Exp Name

The EXP Name is a read-only display that shows the name of the ENSONIQ EXP Series Wave Expansion Board you've installed in your ASR-X.

```
Memory Manager:
Exp Name:Dance:EXP-3
```

If no board is installed, the display will show "**EMPTY**."

8 Expanding the ASR-X

Overview

The ASR-X provides some exciting opportunities for expansion, described in this chapter—follow all of the instructions provided carefully, to ensure that you don't injure your ASR-X, or yourself,.

An Important Note About Electro Static Discharge

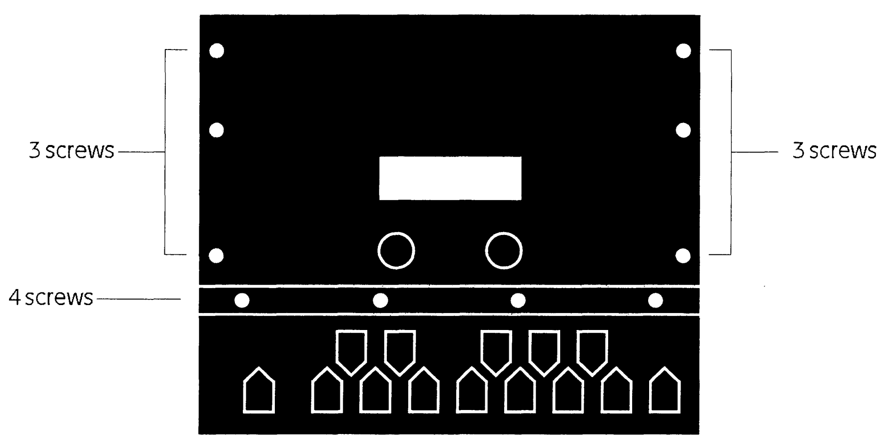
Many of the internal components in the ASR-X and areas of its expansion boards are susceptible to Electro Static Discharge (ESD), commonly known as "static." Electro static discharge can damage or destroy electronic devices. Here are some procedures you can follow when handling electronic devices in order to minimize the possibility of causing ESD damage:

- Before opening your ASR-X or handling the expansion boards you should be grounded. Use a ground strap to discharge any static electric charge built up on your body. The ground strap attaches to your wrist and any unpainted metal surface within the ASR-X.
- Avoid any unnecessary movement, such as scuffing your feet when handling electronic devices, since most movement can generate additional charges of static electricity.
- Minimize the handling of the expansion boards. Keep them in their static-free packages until needed. Transport or store the expansion boards only in their protective packages.
- When handling the expansion boards, avoid touching the connector pins. Try to handle the expansion boards by the edges only.

If you have any questions concerning the installation of ASR-X expansion options, or for additional technical support, please contact your authorized ENSONIQ dealer or ENSONIQ Customer Service at (610) 647-3930 Monday through Friday 9:30 a.m. to 12:15 p.m. and 1:15 p.m. to 6:30 p.m. Eastern Time.

Opening the ASR-X

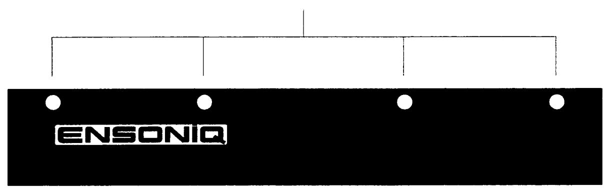
1. Turn of the ASR-X's power and unplug the AC cable from its rear-panel jack.
2. Place your ASR-X on a flat surface, normal-side up, leaving an empty space on the surface to the left of the ASR-X left equal to the width of the ASR-X.
3. Locate the ten hex screws along the left, right and lower edges of the ASR-X's upper panel.



The ASR-X when viewed from the top

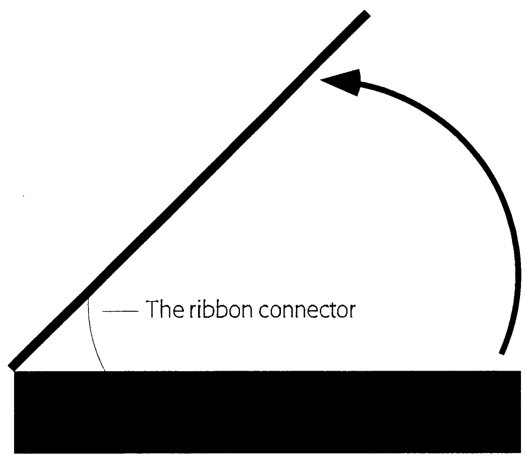
4. Using the hex wrench supplied in your ASR-X accessory kit, remove the ten screws. Put the screws in a safe location—you'll need them again when you close up the ASR-X.

5. Turn the ASR-X around so that its rear panel is facing you, and locate the four Phillips-head screws along its top edge.



The ASR-X when viewed from the back.

6. Remove the four screws, and place them with the first ten.
7. Rotate the ASR-X back to its original position.
8. Gently lift the right-hand edge of the ASR-X's lid, opening it out and towards the left as you would a book. Be careful—the lid is not hinged to the chassis of the ASR-X—take care not to break or damage the ribbon connector connecting the lid to the ASR-X's main board.



Carefully lift the lid as you would open a book.

9. Rest the lid, display-side-down, on the surface to the left of the ASR-X.

Installing Additional Sampling/Resampling Memory

Increasing Sample Memory

The ASR-X ships from ENSONIQ with 2 megabytes of memory in which to store sampled / resampled waves. You can install up to 32 megabytes of additional memory (for a total of 34 megabytes) by installing standard 4, 8, 16 or 32 megabyte SIMM chips.

With additional memory, the amount of time available for sampling grows:

Amount of memory installed	mono sampling	stereo sampling
2 megabytes (as shipped from the factory)	17 seconds	8.7 seconds
6 megabytes (with 4-meg SIMM installed)	65 seconds	32 seconds
10 megabytes (with 8-meg SIMM installed)	112 seconds	56 seconds
18 megabytes (with 16-meg SIMM installed)	207 seconds	103 seconds
34 megabytes (with 32-meg SIMM installed)	398 seconds	198 seconds

What is a SIMM Chip?

"SIMM" is an acronym for "Single In-line Memory Module." SIMM chips, or "SIMMs" are actually small circuit boards onto which have been placed smaller DRAM—for "dynamic random access memory"—chips.

Which SIMMs Can be Installed in the ASR-X?

The ASR-X accepts any standard 72-pin SIMM that meets these standards:

- The SIMM is 70ns (nanoseconds) or faster.
- The SIMM is a 5-volt chip (3-volt SIMMs will not properly fit the ASR-X SIMM socket).
- Either extended data output (EDO) or non-EDO SIMMs can be used.
- The SIMM is either a x32 or x36 chip.

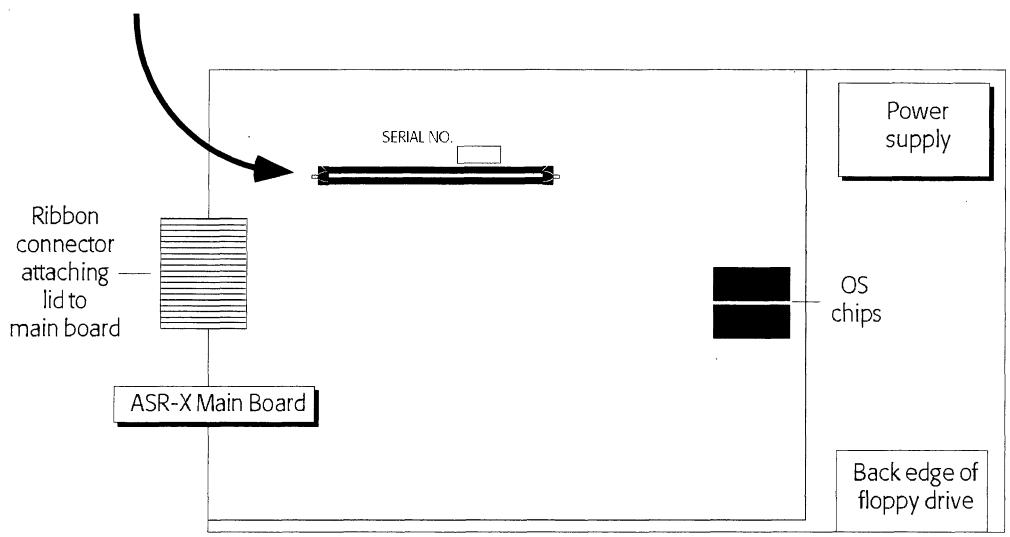
SIMM Installation Procedure

Warning: It's worth taking a moment to read through the following procedures before actually performing them, so you'll know what to expect along the way. Don't forget to follow the guidelines in "An Important Note About Electro Static Discharge" at the beginning of this chapter.

Locating the SIMM Socket on the ASR-X Main Board

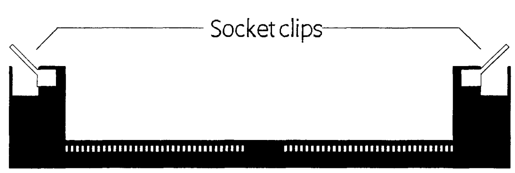
Before proceeding, you'll need to open your ASR-X.
Follow the instructions in "Opening the ASR-X," earlier in this chapter.

1. Looking down into the ASR-X with the pads towards you, the SIMM socket is located as shown by the arrow in the illustration below:



The main board contains many elements not shown.
Drawing not to scale.

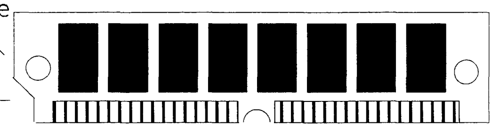
2. Viewed from the front, the SIMM socket looks something like this:



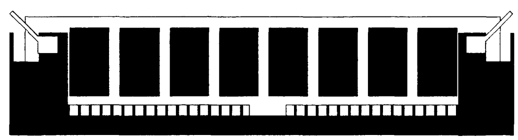
Installing a SIMM chip into the SIMM Socket

Before proceeding, you'll need to open your ASR-X. Follow the instructions in "Opening the ASR-X," earlier in this chapter. "Locating the SIMM Socket on the ASR-X Main Board," above, tells you how to find the SIMM socket.

1. Orient your SIMM chip so that its edge connector is downward, and its alignment notch is facing to the left, as shown.



2. Approaching the SIMM socket from its back side—the side closest to the ASR-X's rear-panel connectors—place your chip's edge connector in the slot in the center of the socket.
3. Using two hands, gently press the upper edge of the back of your chip so that it tilts forward between the two metal clips on the socket.
4. Continue pressing forward until both socket clips snap into place against the front edge of the chip —you'll hear a click when this occurs.



5. Replace the ASR-X's lid and eight screws removed in Steps 3-6 of "Opening the ASR-X," earlier in this section.
6. Reconnect the ASR-X's AC power.

   When you turn your ASR-X back on, your new memory will be available for use.

SIMM Removal Procedure

Before proceeding, you'll need to open your ASR-X. Follow the instructions in "Opening the ASR-X," earlier in this chapter. "Locating the SIMM Socket on the ASR-X Main Board" tells you how to find the SIMM socket.

1. Gently pull outward each of the SIMM socket's metal clips, one at a time—you'll be able to hear or feel when each clip lets go of the chip's edge.
2. When both clips have let go of the SIMM's edges, you can lift the chip out of the socket.
3. Replace the ASR-X's lid and eight screws removed in Steps 3-6 of "Opening the ASR-X."
4. Reconnect the ASR-X's AC power.

Installing an ENSONIQ EXP-Series Wave Expansion Board

The ASR-X can accommodate an ENSONIQ EXP expansion board. These boards add new wave data and sounds to the ASR-X. For more information on the EXP-Series Wave Expansion boards, call ENSONIQ at 610-647-3930 or visit the ENSONIQ World Wide Web site at *http://www.ensoniq.com*.

EXP-Series Wave Expansion Board Installation Procedure

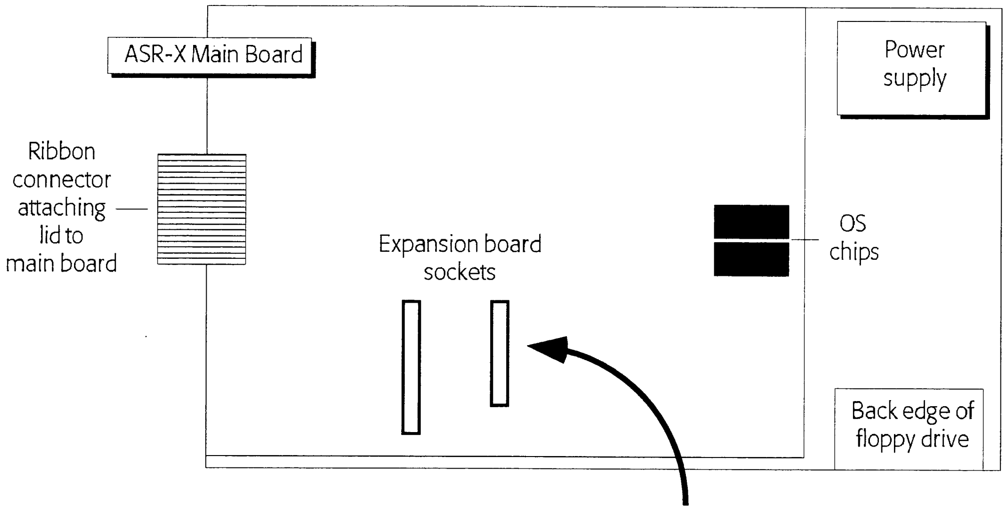
Warning: It's worth taking a moment to read through the following procedures before actually performing them, so you'll know what to expect along the way. Follow the guidelines in "An Important Note About Electro Static Discharge" at the beginning of this chapter.

Locating the Wave Expansion Board Sockets on the ASR-X Main Board

Before proceeding, you'll need to open your ASR-X.
Follow the instructions in "Opening the ASR-X," earlier in this chapter.

1. Looking down into the ASR-X with the pads towards you, the expansion board sockets are located as shown by the arrow in the illustration below:

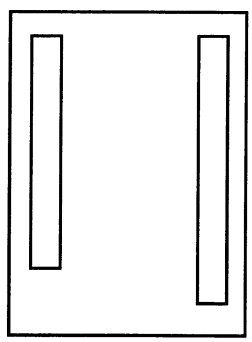
The main board contains many elements not shown.
Drawing not to scale.



Installing an EXP-Series Wave Expansion Board

Before proceeding, you'll need to open your ASR-X by following the instructions in "Opening the ASR-X," earlier in this chapter. "Locating the Wave Expansion Board Sockets on the ASR-X Main Board," above, tells you how to find the EXP board sockets.

1. Examine your expansion board. Notice that it has a 50-pin and a 40-pin connector.



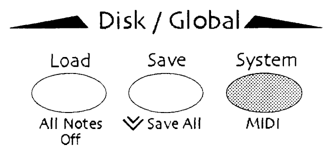
When you turn the expansion board over, connector-side-down, its connectors will line up with the sockets in the ASR-X. It's in this position that the board is installed.

2. Align your expansion board above the sockets on the ASR-X main board.
4. Press the expansion board down firmly into the main board sockets so that it makes a physical (and electrical) connection with your ASR-X. The expansion board's connectors must be inserted into both of the main board's sockets in order to work properly.
5. Replace the ASR-X's lid and screws, reconnect its power cord, power up and follow the instructions in "To Identify an Installed Expansion Board," below, to verify that the ASR-X is properly recognizing the expansion board.

Note: To remove and expansion board, lift it gently from its sockets on the ASR-X main board.

To Identify An Installed Expansion Board

1. Press the System / MIDI button.



2. Turn the Parameter knob until the display shows:

```
System/MIDI:
Enter MemoryManager?
```

3. Press the Yes button.
2. Turn the Parameter knob until the display shows:

```
MemoryManager:
Exp Name:Dance:EXP-3
```

The name of the board you've installed

This will show the name of the installed expansion board.

Note: If you've installed an expansion board and the ASR-X does not show its name, carefully repeat the instructions in "Installing an EXP-Series Wave Expansion Board." If the ASR-X still doesn't recognize the expansion board, call your authorized ENSONIQ dealer or ENSONIQ Customer Service at 610-647-3930.

Replacing the ASR-X Operating System Chips

With most electronic devices, operating system (O.S.) upgrades have become common. For ENSONIQ products, an operating system upgrade provides system enhancements, and at times offers additional features. The ASR-X O.S. is contained on EPROM chips installed in sockets on the ASR-X main board. Any O.S. changes require changing the O.S. EPROMs.

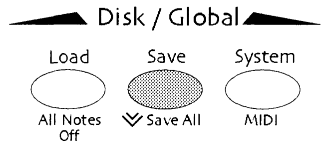
You can find out the current ASR-X operating system version by visiting ENSONIQ's World Wide Web site—you'll find a list of current operating system versions for all ENSONIQ products at *http://www.ensoniq.com/html/pi.htm*—or by calling Customer Service at 610-647-3930. An up-to-date O.S. list for all ENSONIQ products can also be found in the Transoniq Hacker, a third-party monthly publication (for more information, call 1-503-227-6848).

Operating System Updating Procedure

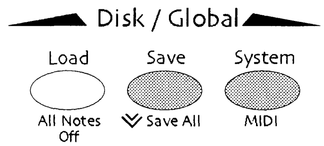
Warning: It's worth taking a moment to read through the following procedures before actually performing them, so you'll know what to expect along the way. Follow the guidelines in "An Important Note About Electro Static Discharge" at the beginning of this chapter.

Learning The Version Number Of the Currently Installed Operating System

1. Press the Disk/Global Save and hold it down.



2. While continuing to hold down the Save button, press the System/MIDI button.



   The display briefly shows you the version number of the operating system installed in your ASR-X:

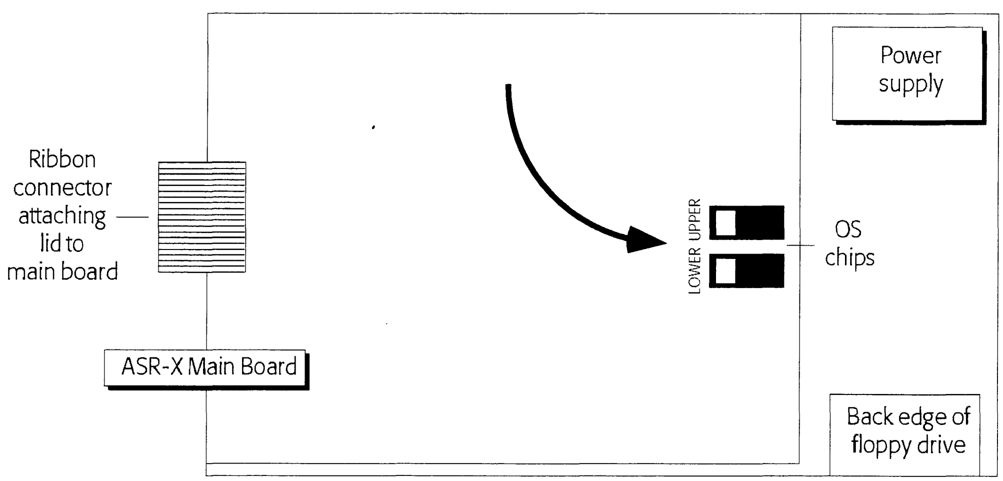
```
   ENSONIQ ASR-X
O.S. Version:   1.00
```

If you'd like to upgrade your ASR-X, contact your authorized ENSONIQ dealer or ENSONIQ's Customer Service at (610) 647-3930 to obtain the special ASR-X EPROM upgrade kit.

Locating the Currently Installed OS Chips on the ASR-X Main Board

Before proceeding, you'll need to open your ASR-X.
Follow the instructions in "Opening the ASR-X," earlier in this chapter.

1. Looking down into the ASR-X with the pads towards you, the two OS chips are located as shown by the arrow in the illustration below:



The main board contains many elements not shown.
Drawing not to scale.

Items Included in the EPROM Replacement Kit

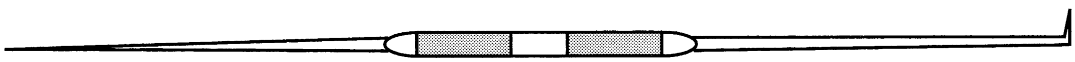
- Two software update EPROM chips
- A self-addressed stamped envelope
- An anti-static wrist strap

Do not remove the EPROM chips from the protective black foam until you are ready to install them. Be sure to use a grounding strap when handling these chips to avoid damage from static discharge.

A disposable grounding strap is included in this kit. You will not use the wrist strap until you've removed the lid of your ASR-X. A grounding wrist strap will discharge any static built up on your body to ground, and prevent you from damaging your software chips or your ASR-X.

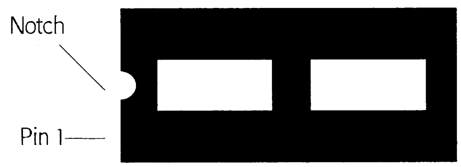
The Tools You'll Need

- #2 Phillips screwdriver
- A thin bladed, flathead screwdriver or a scribe as shown here:



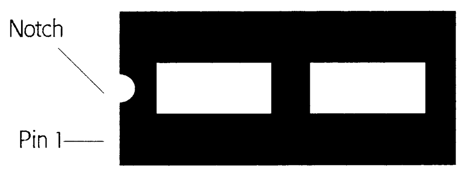
A Visual Examination of EPROMs and Sockets

The EPROMs go in sockets that look like this:

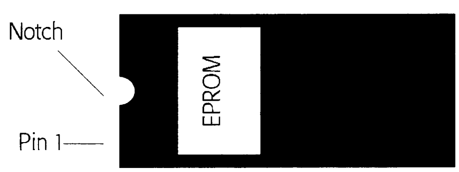


EPROM
socket

EPROM chips and their sockets have a notch on one end. Looking down on the EPROM or socket, with its notch on the left, pin 1 will always be on the bottom. Pin 1 of the EPROM will always go into pin 1 of the socket.



EPROM
socket



EPROM
chip

The ASR-X O.S. EPROM chips must always be justified to pin 1 on the chip and the socket. Look at your chips through their pink plastic bag and make sure you can locate pin 1.

Replacing the ASR-X OS Chips

Before proceeding, you'll need to open your ASR-X by following the instructions in "Opening the ASR-X," earlier in this chapter. "Locating the Currently Installed OS Chips on the ASR-X Main Board," above, tells you how to find the OS chips you'll be replacing.

1. Save all of your work to disk—see Chapter 7 for instructions.
2. Open the envelope with the disposable wrist strap. Unwrap the first two folds of the band and wrap the exposed adhesive side firmly around your wrist. Unroll the rest of the band and peel the liner from the copper foil at the opposite end. Attach the sticky side of the copper foil to any non-painted metal area of the ASR-X case.
3. Locate the LO and UP O.S. EPROM chips on the ASR-X main board.
4. Remove the O.S. EPROMs from the ASR-X main board. ENSONIQ recommends using the angled end of a scribe, or a thin bladed, flathead screwdriver to slowly lift each end of the EPROM until it is free from the socket. Gently wedge the scribe or screwdriver between the black socket and the chip (not the green board and the socket). When the scribe or screwdriver is in place, work it slowly up and down between the chip and the socket, raising the chip a little at one end and reaching underneath to lift the other, until the chip is free.
5. Lift the chips out of the ASR-X and set them aside.
6. Replace the EPROM chips with the new ones supplied, labeled LO and UP. Remember to line up the notches in the sockets with the notches in the chips. The pins of each chip should be inserted into the holes in the socket. In a new EPROM, it's not uncommon for the left and right sets of pins on a chip to be spread a bit wider than the socket. You can very carefully bend the pins inward slightly by resting the edge of the chip on a flat non-metal surface and tipping the chip while applying pressure gently.

Warning: The sockets are labeled on the main board. Make sure the LO chip is in the LOWER socket and the UP chip is in the UPPER socket.

7. Replace the ASR-X's lid and screws, reconnect its power cord and power up. The ASR-X will now utilize the new OS software.

9 Supplemental Information

List of ROM Sounds

ROM08:000	Thump Kick
ROM08:001	Muff Kick
ROM08:002	Tite Kick
ROM08:003	808 Kick
ROM08:004	AmbientKick
ROM08:005	Electro Kik
ROM08:006	Wolf Kick
ROM08:007	2001 Kick
ROM08:008	Cosmo Kick
ROM08:009	Bang Kick
ROM08:010	PZ Kick
ROM08:011	Wild Kick
ROM08:012	Snick Kick
ROM08:013	WooBox Kick
ROM08:014	RapBoomKick
ROM08:015	BBM Kick
ROM08:019	SideStick 1
ROM08:020	SideStick 2
ROM08:021	Chill Snare
ROM08:022	Big RockSnr
ROM08:023	Jamm Snare
ROM08:024	Wolf Snare
ROM08:025	Gated Snare
ROM08:026	Live Snare
ROM08:027	Spak Snare
ROM08:028	Ludwig Snr
ROM08:029	Real Snare
ROM08:030	Classic Snr
ROM08:031	909 Snare
ROM08:032	808 Snare
ROM08:033	Brush Slap
ROM08:034	Clean Snare
ROM08:035	Cosmo Snare
ROM08:036	House Snr 1
ROM08:037	House Snr 2
ROM08:038	House Snr 3
ROM08:039	Bang Snare
ROM08:040	Slang Snare
ROM08:041	Zee Snare
ROM08:042	Mutt Snare
ROM08:043	Rimshot
ROM08:047	Studio Tom
ROM08:048	Rock Tom
ROM08:049	909 Tom
ROM08:050	808 Tom
ROM08:052	Studio Hat
ROM08:053	Tight Hat
ROM08:054	Techno Hat
ROM08:055	Smack Hat

ROM08:056	Snick Hat
ROM08:057	PZ Hat
ROM08:058	Compresd Ht
ROM08:059	808 Hat Cl
ROM08:060	909 Hat Cl
ROM08:061	R&B Hat Cl
ROM08:062	Trance Hat
ROM08:063	CR78 Hat
ROM08:064	Pedal Hat
ROM08:068	Compr OpnHt
ROM08:069	StudioOpHt1
ROM08:070	StudioOpHt2
ROM08:071	808 OpenHat
ROM08:072	909 OpenHat
ROM08:073	CR78-O-Hat
ROM08:077	CrashCymbal
ROM08:078	RideCymbal
ROM08:079	Ride Bell
ROM08:080	China Crash
ROM08:084	Rap Clap
ROM08:085	808 Clap
ROM08:086	808 Rimshot
ROM08:087	808 Cowbell
ROM08:088	808 Clave
ROM08:090	Tamb. Down
ROM08:091	Tamb. Up
ROM08:092	Triangle Cl
ROM08:093	Triangle Op
ROM08:094	AfroCowbell
ROM08:095	Agogo
ROM08:096	Bongo
ROM08:097	Conga Slap
ROM08:098	Conga Mute
ROM08:099	Conga Hi
ROM08:100	Conga Lo
ROM08:101	Timbale Hi
ROM08:102	Timbale Lo
ROM08:103	Timbale Rim
ROM08:104	Cabasa
ROM08:105	Maracas
ROM08:106	Shaker
ROM08:107	Shekere Up
ROM08:108	Shekere Dn
ROM08:109	Guiro Long
ROM08:110	Guiro Short
ROM08:111	Vibraslap
ROM08:112	Clave
ROM08:113	Woodblock
ROM08:114	Stick Click

ROM08:115	Cuica
ROM08:116	Gt. SlideDn
ROM08:117	Scratch 1
ROM08:118	Scratch 2
ROM08:119	Scratch 3
ROM08:120	Scratch 4
ROM08:121	Scratch 5
ROM08:122	Scratch 6
ROM08:123	Scratch Lp
ROM08:124	Whistle 1
ROM08:125	Whistle 2
ROM08:126	Hiss
ROM09:000	Poppy Piano
ROM09:001	Digby Piano
ROM09:002	Clavinot
ROM09:003	Orgcussion
ROM09:004	NewOrgleans
ROM09:005	Snare-Imba
ROM09:006	NaturalBass
ROM09:007	Less Frets
ROM09:008	SlapYo'Self
ROM09:009	BuzzSawBass
ROM09:010	Sweep Bass
ROM09:011	Snot-T-Bass
ROM09:012	Barkin'Bass
ROM09:013	RaveTheWave
ROM09:014	Tite'T'Bass
ROM09:015	Snoot Guit
ROM09:016	Classic Syn
ROM09:017	Squared Off
ROM09:018	Cat's Meow
ROM09:019	Sin-Stringz
ROM09:020	String Hit
ROM09:021	Horn Hit
ROM09:022	Sax Hit
ROM09:023	Raunch Hit
ROM09:024	Clangerous
ROM09:025	Spackle Me
ROM09:026	The Birds !
ROM09:027	Noise Sync
ROM09:028	Sync'O'Goob
ROM10:000	Gizmo Kit
ROM10:001	Dance Kit
ROM10:002	HeavyDrmKit
ROM10:003	Big Kit
ROM10:004	Rock Kit
ROM10:005	Ol'SkoolKit
ROM10:064	GM Kit
ROM10:127	Silence

List of ROM Waves

KEYBOARD	ELEC PIANO
	PERC ORGAN
	DRAWBAR ORGAN
	PAD SYNTH
STRING-SOUND	STRING HIT
	MUTE GUITAR
	MUTE GUITARWF
	GTR-SLIDE
BRASS+HORNS	HORN HIT
WIND+REEDS	BARI SAX HIT
BASS-SOUND	UPRIGHT BASS
	BS HARMONICS
	FM BASS
	ANALOG BASS 1
	ANALOG BASS 2
	FRETLESS BASS
	MUTE BASS
	SLAP BASS
DRUM-SOUND	2001 KICK
	808 KICK
	AMBIENT KICK
	BAM KICK
	BANG KICK
	BBM KICK
	BOOM KICK
	COSMO KICK
	ELECTRO KICK
	MUFF KICK
	PZ KICK
	SNICK KICK
	THUMP KICK
	TITE KICK
	WILD KICK
	WOLF KICK
	WOO BOX KICK
	808 SNARE
	808 RIMSHOT
	909 SNARE
	BANG SNARE
	BIG ROCK SNAR
	CHILL SNARE
	CLASSIC SNARE
	CLEAN SNARE
	COSMO SNARE
	GATED SNARE
	HOUSE SNR 1
	HOUSE SNR 2
	HOUSE SNR 3
	JAMM SNARE
	LIVE SNARE
	LUDWIG SNARE
	MUTT SNARE
	REAL SNARE
	RIMSHOT
	SLANG SNARE
	SPAK SNARE
	WOLF SNARE
	ZEE SNARE
	BRUSH SLAP
	SIDE STICK 1
	SIDE STICK 2
	STICKS
	STUDIO TOM
	ROCK TOM
	909 TOM
	SYNTH DRUM
CYMBALS	808 CLOSED HT
	808 OPEN HAT
	909 CLOSED HT
	909 OPEN HAT
	HOUSE CL HAT
	PEDAL HAT
	PZ CL HAT
	R&B CL HAT
	SMACK CL HAT
	SNICK CL HAT
	STUDIO CL HAT
	STUDIO OPHAT1
	STUDIO OPHAT2
	TECHNO HAT
	TIGHT CL HAT
	TRANCE CL HAT
	CR78 OPENHAT
	COMPRESS OPHT
	CRASH CYMBAL
	CRASH LOOP
	RIDE CYMBAL
	RIDE BELL
	CHINA CRASH
PERCUSSION	808 CLAP
	808 CLAVE
	808 COWBELL
	AGOGO
	BONGO
	CABASA
	CLAVE
	CONGA HIGH
	CONGA LOW
	CONGA MUTE
	CONGA SLAP
	CUICA
	ETHNO COWBELL
	GUIRO
	MARACAS
	SHAKER
	SHEKERE DN
	SHEKERE UP
	SLAP CLAP
	TAMBOURINE DN
	TAMBOURINE UP
	TIMBALE HI
	TIMBALE LO
	TIMBALE RIM
	TRIANGLE HIT
	VIBRASLAP
	WHISTLE
	WOODBLOCK
TUNED-PERCUS	BIG BELL
	SMALL BELL
	GAMELAN BELL
	MARIMBA
	MARIMBA WF
SOUND-EFFECT	SCRATCH 1
	SCRATCH 2
	SCRATCH 3
	SCRATCH 4
	SCRATCH 5
	SCRATCH 6
	SCRATCH LOOP
WAVEFORM	SAWTOOTH
	SQUARE WAVE
	TRIANGLE WAV
	SQR+SAW WF
	SINE WAVE
	ESQ BELL WF
	BELL WF
	DIGITAL WF
	E PIANO WF
	DIG VOCAL WF
	DEEP PAD WF
INHARMONIC	HISS
	NOIZZZ
	TEXTURE
TRANSWAVE	BROKEN TWF

List of SoundFinder Categories

If there are no sounds of a particular type currently in the ASR-X memory, the type will not be displayed:

Category	Description
USER-SND	This special category is ideal for storing the sounds you create—sounds will also appear in their appropriate SoundFinder musical instrument type list. All RAM kits are designated as USER-SNDs.
DEMO-SND	Demo sounds are designed to demonstrate the scope of sounds in the ASR-X. Whenever this is selected, the first sound in the type will be selected; the ASR-X will not reselect the last sound selected in the DEMO-SND type. Demo sounds also appear in their appropriate Sound Type list.
EXP-SND	Expansion board sounds.
DRM-SND	ROM drum key sounds.
ROM-SND	All sounds in ROM.
RAM-SND	All sounds in RAM.
ALL-SND	All sounds.
BASS	Acoustic and electric basses.
BASS-SYN	Synth basses, and processed electric basses with a "synthy" quality.
BELL	Acoustic and synth bell sounds, both pitched (e.g., glockenspiel, celesta). and non-pitched (e.g., church bells).
BRASSECT	Trumpet, trombone, tuba, French horn, saxophone, and mixed brass sections (including sampled sections) and small ensembles (with more than one distinct pitch/"player" on a single key).
BRASSOLO	Solo brass (e.g., trumpet, trombone, tuba, French horns).
DRUM-KIT	Drum kits that use the ENSONIQ drum map.
DRMKITGM	Drum kits that use the General MIDI drum map.
GUITAR-A	Steel, nylon, and gut-stringed acoustic guitars.
GUITAR-E	Clean electric guitars and distortion guitars.
HITS	Hits of all kinds.
KEYS	Other stringed keyboard sounds (e.g., harpsichord and clavinet).
LAYERS	Unnatural layered combinations of acoustic elements (e.g., a bass harmonic layered with a string section), excluding pianos/electric-pianos/organs layered with other sounds in which the piano/electric-piano/organ element is dominant. Also excludes multi-instrumental orchestral layers.
LOOPGRUV	Looped, repeating musical passages and drum rhythm loops (sampled or wave-sequenced) that play on one key.
MALLET	Tuned mallet-struck percussion instruments (e.g., marimba, xylophone, timpani, steel drum, log drum).
ORCHSTRA	Multi-instrumental orchestral Sounds (e.g., mixed strings/brass/woodwinds/reeds/orchestral percussion) layered with one another.
ORGAN-A	Acoustic pipe and pump organs.
ORGAN-E	Electric and electronic organs.
ORGANLYR	Any organs layered with other sounds in which the organ element is dominant.
PERC-KIT	Percussion kits that use either the ENSONIQ or General MIDI percussion maps.
PERCSOLO	Solo untuned percussion (e.g., taiko, synth-tom) includes most drum key sounds.
PIANO-A	Acoustic pianos, honky-tonk, toy pianos, and piano forte.
PNOLYR-A	Acoustic pianos layered with other sounds in which the acoustic piano element is dominant.
PIANO-E	Electric and electronic piano sounds, and electric pianos layered with acoustic pianos.
PNOLYR-E	Electric pianos layered with other sounds in which the electric piano element is dominant.
PLUCKED	Plucked strings (e.g., harps, banjo, dulcimer, sitar), pizzicato strings, and other plucked instruments (e.g., kalimba).
SAX-SOLO	Solo saxophones.
SOUND-FX	Realistic sound effects (e.g., broken glass, animal sounds, record scratches) and entirely non-pitched fantasy and chaos sound effects.(e.g., spacecraft, environments)
SPLITS	Combination keyboard splits of two or more different types of sounds. Also includes splits of similar sounds that have discontiguous key ranges (e.g., a bassoon/oboe split that covers the natural ranges of both instruments).
STRGSECT	Bowed string sections (including sampled sections) and small string ensembles (with more than one distinct pitch/"player" on a single key).
STRGSOLO	Bowed solo strings (e.g., violin, viola, cello).
SYN-COMP	Non-vintage, sustaining and non-sustaining, polyphonic synth sounds with a pitched or non-pitched, highly obtrusive attack component that lend themselves toward comping (i.e., you can always play successive 1/8 note chords with these funky sounds).
SYN-LEAD	Monophonic lead synth sounds (excluding monophonic synth basses).
SYN-PAD	Non-vintage, sustaining, polyphonic synth sounds with a pitched, less obtrusive attack component, and an appropriate release, that lend themselves toward pad playing.
SYN-VINT	Polyphonic, signature vintage "analog" synth sounds (excluding monophonic vintage synth leads and synth basses). Normally these are named after the synth that they evoke.
SYNOTHER	Other types of pitched, polyphonic, hybrid synth sounds with sustaining, disparate components (e.g., sample & hold sync sounds).

VOCALS	Vocal sounds (e.g., choirs, synth-vox).
WINDREED	Solo woodwinds/reeds (e.g., flute, oboe, bassoon, clarinet, recorder, English horn, ocarina, bandoneon, shakuhachi, bagpipes, harmonica, accordion, melodica, didjeridoo).
*UTILITY	Utility resources (e.g., default template sounds used for programming and other special non-musical purposes).
*CUSTOM	The category in which the sounds that play waves are stored. When you send waves to pads, the pad sounds that play the waves are stored in this category.

Drum and Percussion Maps

ENSONIQ Drum Map

ZONE	KEY RANGE	DESCRIPTION
1 (6 keys)	B1 to E2 KICK	The key C#2 allows for non-finish envelope sounds.
2 (10 keys)	F2 to D3 SNARE	Includes sidestick—the keys from A2-C3 allow for non-finish envelope sounds (Snare rolls, brush swirls, etc.)
3 (10 keys)	D#3 to C4 HATS	The keys G#3 and B3 allow for non-finish envelope sounds (closed hats first, opens on A#3 and B3; foot closed on C4).
4 (9 keys)	C#4 to A4 CYMBL	The key A4 allows for non-finish envelope sounds (rides C#4 to E4; followed by crashes).
5 (9 keys)	A#4 to F#5 TOMS	All keys in finish envelope mode.
6 (7 keys)	G5 to C#6 PERC1	Shaken or small hits—tambourine (G5 to A5); shaker, cabasa, or maracas (A#5 to C6); claps (C#6); snap; woodblock
7 (6 keys)	D6 to G6 PERC2	Latin non-pitched Percussion—bongo; conga slap; low conga; high conga; timbale
8 (7 keys)	G#6 to D7 PERC3	Pitched and Bell-like Percussion—Triangle (A6 closed, A#6 long); cowbell (G#6); high agogo: low agogo; claves (B6, or at D#6 if there are no congas); vinyl surface noise (C7). The keys from B6-D7 allow for non-finish envelope sounds.

ENSONIQ Percussion Map

ZONE	KEY RANGE	DESCRIPTION
1 (6 keys)	B1 to E2	Low Drums—the key C#2 allows for non-finish envelope sounds.
2 (10 keys)	F2 to D3	Medium drums such as Conga, Tabla, Udu—the keys from A2-C3 allow for non-finish envelope sounds.
3 (10 keys)	D#3 to C4	Small things that keep time (shakers, small drums, etc) Clave (G#3); sleighbells, castanets (C4). The keys G#3 and B3 allow for non-finish envelope sounds.
4 (9 keys)	C#4 to A4	Small time-keeping instruments including ride cymbals and instruments like Guiro (C#4 to E4);. crash cymbals, or other accent instruments like windchime, vibraslap, gong (F4 to A4). The key A4 allows for non-finish envelope sounds.
5 (9 keys)	A#4 to F#5	Things struck that play fills—like timbali, woodblocks, log drums, small pitched drums.
6 (7 keys)	G5 to C#6	Tambourines or similar shaken instruments (G5-A5); small high-pitched shakers like maraccas, egg shakes (A#5 - C6); claps, clave (C#6)
7 (6 keys)	D6 to G6	Multi hits of bongos, high drums, cuica, guiro (D6-E6); multi hits of agogo, or other metallic inst. (F6-G6)
8 (7 keys)	G#6 to D7	Cowbell (G#6); Triangle (A6 closed, A#6 long); Long sounds like rainsticks (B6-D7) The keys from B6-D7 allow for non-finish envelope sounds.

GM Kit Map

MIDI Note #		GM Kit
35	B1	AcoustcKick
36	C2	Bright Kick
	C#2	SideStick 1
38	D2	Snare-GM
	D#2	HouseClap1
40	E2	Rock Snare
41	F2	Dry Tom 1
	F#2	4xCl Hat3
43	G2	Dry Tom 1
	G#2	Pedal Hat
45	A2	Dry Tom 1
	A#2	OpenHat-GM
47	B2	Dry Tom 1
48	C3	Dry Tom 1
	C#3	Crash 1-GM
50	D3	Dry Tom 1
	D#3	Ride 1-GM
52	E3	China 1-GM
53	F3	RideBell-GM
	F#3	Tambourine
55	G3	Splash1-GM
	G#3	Cowbell
57	A3	Crash 1-GM
	A#3	Vibraslap
59	B3	Ride 1-GM
60	C4	Bongo
	C#4	Bongo
62	D4	Conga Mute
	D#4	Conga High
64	E4	Conga Low
65	F4	Timbali
	F#4	Timbali
67	G4	Agogo
	G#4	Agogo
69	A4	Cabasa
	A#4	Maracas
71	B4	Whistle B
72	C5	Whistle A
	C#5	Guiro Short
74	D5	Guiro Long
	D#5	Clave
76	E5	Woodblock 1
77	F5	Woodblock 1
	F#5	Cuica 1
79	G5	Cuica 5
	G#5	Tri Mute-GM
81	A5	Tri Open-GM
	A#5	Shaker
83	B5	Sleighbell
84	C6	WindchimeGM
	C#6	Castanets 1
86	D6	Mt Surdo-GM
	D#6	Op Surdo-GM
	E6	Silence

List of Quantize Templates

The following is a list of all the quantize parameters and their settings for the available quantize templates (there is no data recorded for High Key and Low Key):

Name	Q. to:	Strength	Swing	Random	Shift	Win. Min	Win. Max.	Q Offs?	Move Offs?	Deltas
Strict 1/4	1/4	100	50	0	0	0	50	off	on	off
Strict 1/8	1/8	100	50	0	0	0	50	off	on	off
Strict 1/16	1/16	100	50	0	0	0	50	off	on	off
Strict 1/8T	1/8T	100	50	0	0	0	50	off	on	off
Tighten 1	1/8	5	50	0	0	0	50	off	on	off
Tighten 2	1/8	20	50	0	0	0	50	off	on	off
Tighten 3	1/8	50	50	0	0	0	50	off	on	off
Tighten 4	1/8	70	50	0	0	0	50	off	on	off
Tighten 5	1/16	5	50	0	0	0	50	off	on	off
Tighten 6	1/16	20	50	0	0	0	50	off	on	off
Tighten 7	1/16	50	50	0	0	0	50	off	on	off
Tighten 8	1/16	70	50	0	0	0	50	off	on	off
Randomize 1	1/8	50	50	3	0	0	50	off	on	off
Randomize 2	1/8	60	50	15	0	0	50	off	on	off
Randomize 3	1/16	50	50	3	0	0	50	off	on	off
Randomize 4	1/16	60	50	15	0	0	50	off	on	off
Note Offs 1	1/8	100	50	0	0	0	50	on	on	off
Note Offs 2	1/16	100	50	0	0	0	50	on	on	off
Swing 1	1/16	90	55	0	0	0	50	off	on	off
Swing 2	1/16	92	57	1	0	0	50	off	on	off
Swing 2	1/16	100	63	0	0	0	50	off	on	off
Humanize 1	1/16	75	51	2	0	0	50	off	on	off
Delta 1/8	1/8	100	50	0	0	0	50	off	on	on

What Is MIDI?

Musical instrument and computer manufacturers have agreed upon a set of standards that allows their products to communicate with each other. It's called "MIDI," an acronym for "Musical Instrument Digital Interface." There are two basic aspects to the MIDI standards: the kind of wiring to be used for connecting MIDI devices, and the nature of messages that will be sent through those wires.

Life In The MIDI World

MIDI has opened up incredible possibilities for musicians and music lovers alike. Here are some of the things MIDI has made possible:

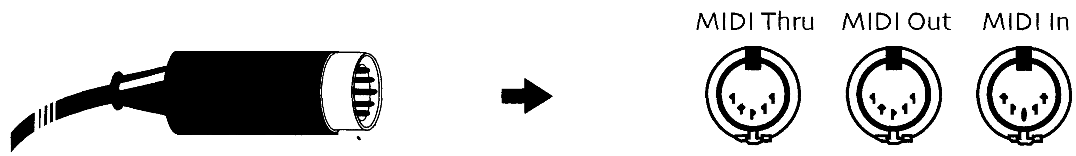
- Musicians can record their performances into MIDI recorders—called *sequencers*—which are found in keyboard workstations, groovestations such as the ASR-X, in stand-alone boxes, and in computers. Once recorded, MIDI-recorded performances can be tweaked and nudged to perfection. Musical arrangements can be re-orchestrated after they've been recorded. Full-blown multi-instrument recordings can be easily created.
- Keyboardists can connect their instruments to a myriad of sound-producing MIDI boxes. MIDI allows a conventional-looking keyboard, to control a number of such devices at the same time, providing for the creation of new, complex timbres. Keyboardists can also set up specific areas on their keyboards to control specific external MIDI devices. These same capabilities are available to computer users. Actually, pretty much any musical instrument can be outfitted to control MIDI devices.

- Musicians can benefit from the communication possible between MIDI instruments and computers to program sounds for their instruments on their computers, taking advantage of the computers' large graphic displays.
- Internal data from one MIDI device can be transmitted to another for storage.
- Recording engineers can control mixing consoles and effects devices with MIDI.

Understanding MIDI

MIDI Hardware

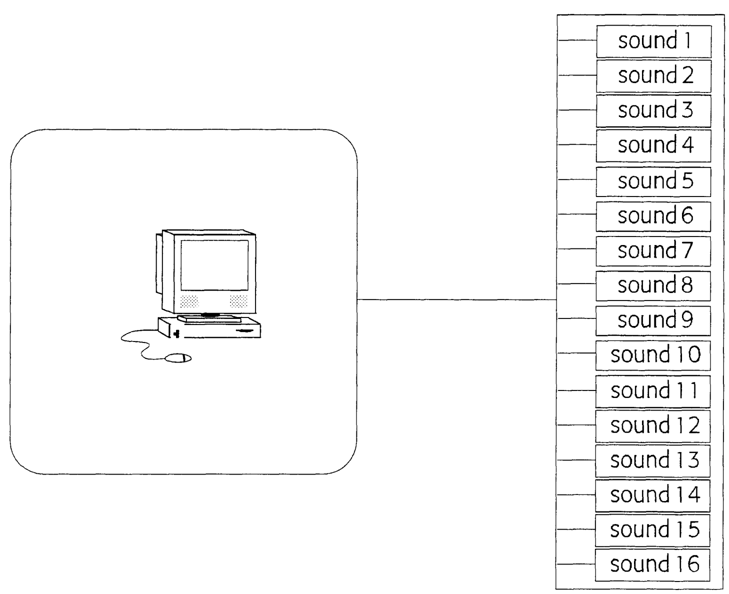
The architects of MIDI had to settle, first of all, on the MIDI hardware: the wires. All MIDI cables have the same kind of plug on either end.There are three MIDI sockets, or jacks, on the back of most MIDI instruments. The MIDI Thru jack is for MIDI data that passes through the instrument unchanged, on its way to some other MIDI device. The instrument sends out its own MIDI information through the MIDI Out jack. The MIDI In jack is for MIDI information coming into the instrument.



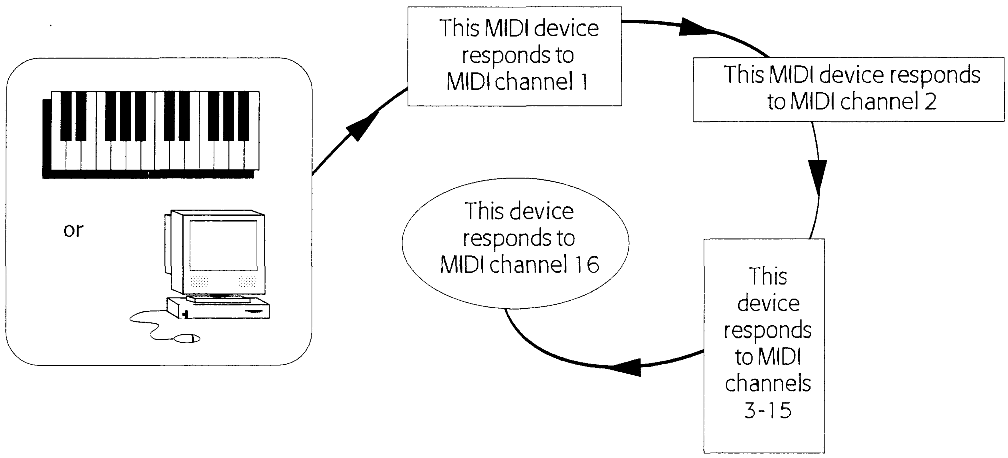
The MIDI cable itself can carry 16 independent channels of MIDI information that travel together through the wire. This means that you can have 16 separate MIDI conversations going on at once among instruments and/or computers connected together with MIDI cables.

How MIDI Channels Work

MIDI instruments can be set up to listen to specific channels and ignore everything else that's going on. This allows a central device such as a keyboard or your personal computer to control each instrument individually. Some instruments—such as the ASR-X—are capable of responding to as many as 16 channels at once. Such instruments are referred to as being *multi-timbral*—it's as if there are up to 16 musical instruments in one box, and MIDI allows you to control each sound separately.



MIDI rigs can also combine both possibilities, with some instruments programmed to respond to one MIDI channel or another, and multi-timbral devices set up to receive up to 16 channels at once.



MIDI messages travel up and down all these channels, and these constitute the second major component of the MIDI Spec.

How MIDI Messages Work

MIDI works in a manner reminiscent of the old player pianos, whose sheets of hole-punched paper told the keyboard mechanism which keys to press down and when. It's not sound that's sent through MIDI cables; it's instructions from one MIDI device—called the "controller"—to another. Of course, MIDI generally doesn't cause any keys to physically move.

Suppose a keyboardist presses a note on a keyboard which is controlling some sound-producing MIDI box. The controller would send out a *Key Down* (or "note-on") message for that note. The MIDI box receiving such a message would play the note. When the keyboardist lets go, the controller would send out a *Key Up* message, and the receiving device would stop sounding the note. It's as simple at that.

MIDI captures the expressive nuances in a performance by sending out other kinds of messages. Controllers can sense how hard a musician plays—referred to in the MIDI world as velocity—and can instruct other devices to respond accordingly. Sustain and sostenuto foot pedals also send out MIDI messages. There are many tools for expression that can be transmitted and responded to via MIDI.

To tell a MIDI instrument which sound program you want to hear, you would send a MIDI Program Change.

MIDI can also send messages that have the same effect as pushing buttons and twirling knobs on a receiving device. To make sure that only the intended instrument listens to such instructions, MIDI sends it a special greeting in a language only it can understand. Every MIDI device has such a language, and these "hey there" messages are referred to as "System Exclusive headers." System Exclusive data is often referred to as SysEx data.

In MIDI recording, all of the messages that a controller produces are sent to a sequencer. Most sequencers have Record, Stop and Play buttons, since they're usually designed to resemble tape recorders. When the Record button is pressed, the sequencer captures incoming MIDI information. Pressing Stop tells the sequencer to store that information in its memory. When Play is pressed, it sends it back out.

The Art of MIDI

The fact that MIDI is so simple to use is a testament to the cleverness of its designers. Its true magic, however, lies in MIDI's power as a tool in the creative process, and in the imaginations of those artists who wield it.

List of MIDI Controller Names

Bank Select #000 - Bank Select	Expression#043 - Expression LSB	MIDIContrl#086 - UNDEFINED
Mod Wheel #001 - Mod Wheel or Lever	FXControl1#044 - Effect Control 1 LSB	MIDIContrl#087 - UNDEFINED
Breath #002 - Breath Controller	FXControl2#045 - Effect Control 2 LSB	MIDIContrl#088 - UNDEFINED
MIDIContrl#003 - UNDEFINED	MIDIContrl#046 - UNDEFINED	MIDIContrl#089 - UNDEFINED
FootContrl#004 - Foot Controller	MIDIContrl#047 - UNDEFINED	MIDIContrl#090 - UNDEFINED
Glide Time#005 - Portamento Time	GenPurpse1#048 - UNDEFINED	FX Depth 1#091 - Effects Depth 1
Data Entry#006 - Data Entry MSB	GenPurpse2#049 - General Purpose 1 LSB	FX Depth 2#092 - Effects Depth 2
Volume #007 - Volume	GenPurpse3#050 - General Purpose 2 LSB	FX Depth 3#093 - Effects Depth 3
Balance #008 - Balance	GenPurpse4#051 - General Purpose 3 LSB	FX Depth 4#094 - Effects Depth 4
MIDIContrl#009 - UNDEFINED	MIDIContrl#052 - General Purpose 4 LSB	FX Depth 5#095 - Effects Depth 5
Pan #010 - Pan	MIDIContrl#053 - UNDEFINED	Data Inc #096 - Data Inc
Expression#011 - Expression	MIDIContrl#054 - UNDEFINED	Data Dec #097 - Data Dec
FX Control1#012 - Effect Control 1	MIDIContrl#055 - UNDEFINED	NonRgPmLSB#098 - Non-Reg param Num LSB
FX Control2#013 - Effect Control 2	MIDIContrl#056 - UNDEFINED	NonRgPmMSB#099 - Non-Reg param Num MSB
MIDIContrl#014 - UNDEFINED	MIDIContrl#057 - UNDEFINED	RgParamLSB#100 - Reg param Num LSB
MIDIContrl#015 - UNDEFINED	MIDIContrl#058 - UNDEFINED	RgParamMSB#101 - Reg param Num MSB
GenPurpse1#016 - General Purpose 1	MIDIContrl#059 - UNDEFINED	MIDIContrl#102 - UNDEFINED
GenPurpse2#017 - General Purpose 2	MIDIContrl#060 - UNDEFINED	MIDIContrl#103 - UNDEFINED
GenPurpse3#018 - General Purpose 3	MIDIContrl#061 - UNDEFINED	MIDIContrl#104 - UNDEFINED
GenPurpse4#019 - General Purpose 4	MIDIContrl#062 - UNDEFINED	MIDIContrl#105 - UNDEFINED
MIDIContrl#020 - UNDEFINED	MIDIContrl#063 - UNDEFINED	MIDIContrl#106 - UNDEFINED
MIDIContrl#021 - UNDEFINED	Sustain #064 - Sustain	MIDIContrl#107 - UNDEFINED
MIDIContrl#022 - UNDEFINED	PortOn/Off#065 - Portamento On/Off	MIDIContrl#108 - UNDEFINED
MIDIContrl#023 - UNDEFINED	Sostenuto #066 - Sostenuto	MIDIContrl#109 - UNDEFINED
MIDIContrl#024 - UNDEFINED	Soft Pedal#067 - Soft Pedal	MIDIContrl#110 - UNDEFINED
MIDIContrl#025 - UNDEFINED	LegatoFtsw#068 - Legato Ftsw	MIDIContrl#111 - UNDEFINED
MIDIContrl#026 - UNDEFINED	Hold 2 #069 - Hold 2	MIDIContrl#112 - UNDEFINED
MIDIContrl#027 - UNDEFINED	PatchSelct#070 - Snd Variation (Patch Select)	MIDIContrl#113 - UNDEFINED
MIDIContrl#028 - UNDEFINED	Timbre #071 - Harmonic Content (Timbre)	MIDIContrl#114 - UNDEFINED
MIDIContrl#029 - UNDEFINED	Release #072 - Release	MIDIContrl#115 - UNDEFINED
MIDIContrl#030 - UNDEFINED	Attack #073 - Attack	MIDIContrl#116 - UNDEFINED
MIDIContrl#031 - UNDEFINED	Brightness#074 - Brightness	MIDIContrl#117 - UNDEFINED
BankSelect#032 - Bank Select LSB	SoundCntl6#075 - Sound Controller 6	MIDIContrl#118 - UNDEFINED
Mod Wheel #033 - Mod Wheel LSB	SoundCntl7#076 - Sound Controller 7	MIDIContrl#119 - UNDEFINED
Breath #034 - Breath Controller LSB	SoundCntl8#077 - Sound Controller 8	
MIDIContrl#035 - UNDEFINED	SoundCntl9#078 - Sound Controller 9	
FootContrl#036 - Foot Controller LSB	SoundCtl10#079 - Sound Controller 10	
Glide Time#037 - Portamento Time LSB	GenPurpse5#080 - General Purpose 5	
Data Entry#038 - Data Entry LSB	GenPurpse6#081 - General Purpose 6	
Volume #039 - Volume LSB	GenPurpse7#082 - General Purpose 7	
Balance #040 - Balance LSB	GenPurpse8#083 - General Purpose 8	
MIDIContrl#041 - UNDEFINED	Portamento#084 - Portamento Control	
Pan #042 - Pan LSB	MIDIContrl#085 - UNDEFINED	

ASR-X MIDI Implementation

The ASR-X features an extensive MIDI (Musical Instrument Digital Interface) implementation. For normal applications, you will find all the information you need regarding the ASR-X's MIDI functions in this manual. You can also refer to the MIDI Implementation Chart on the following page for a summary of the ASR-X implementation.

If you'd like a free copy of the full ASR-X System Exclusive Specification, write to: "ENSONIQ Corp./MIDI Specification Desk/155 Great Valley Parkway/P.O. Box 3035/Malvern PA 19355-0735/USA

Include in your written request your name and address, and indicate that you would like a copy of the "ASR-X System Exclusive Specification." Please allow 2 to 3 weeks for delivery.

ASR-X **MIDI Implementation Chart** **Version: 1.00**

Function...		Transmitted	Recognized	Remarks
Basic Channel	Default	1	1-16	
	Changed	1-16	1-16	
Mode	Default	POLY	MULTI	
	Messages	X	X	
	Altered	X	X	
Note Number	True voice	36-96	21-108	Note reception is filtered by Key Lo and Key High track parameters
Velocity	Note On	0	0	Note On velocity reception is filtered by VelocityRange Lo and VelocityRange Hi track parameters
	Note Off	0	0	Transmitted Note Off velocity is always 64
After Touch	Key	X	0	
	Channel	X	0	
Pitch Bend		X	0	supports held mode
Control Change		0-119	0-119	see "MIDI Controllers Reception Behavior" below
Program Change		0-127	0-127	select sounds from the currently selected bank
	True#	0-127	0-127	
System Exclusive		0	0	see ASR-X SysEx Specification recognizes MIDI Tuning Dump Standard and Single-Note Tuning Change messages
System Common	Song Position	0	0	
	Song Select	X	X	
	Tune Request	X	X	
System Real Time	Clock	0	0	
	Commands	X	X	
Aux Messages	Local On/Off	X	X	
	All Notes Off	0	0	
	Active Sensing	X	X	
	System Reset	X	X	
Notes	Response to received Controllers varies depending on the nature of the ASR-X parameter affected—see parameter descriptions for details.			

Mode 1: Omni On, Poly   Mode 2: Omni On, Mono   O : Yes
Mode 3: Omni Off, Poly   Mode 4: Omni Off, Mono   X: No

MIDI Controllers Reception Behavior

Control Change	Description	Remark
0-119	SysCTRL 1-4	assignable controllers
0	Bank Select MSB	always 0
1	Mod Wheel	
4	Foot (Pedal)	
5	Portamento Time	
6	Data Entry MSB	for editing of registered and non-registered parameters only, after registered or non-registered parameter MSB and LSB are received
7	Volume	
10	Pan	
11	Expression Controller	
32	Bank Select LSB	
64	Sustain	
65	Portamento On/Off	
66	Sustenuto	
72	Release Time	Amp Env Release
73	Attack Time	Amp Env Attack
74	Brightness	Filter Cutoff
75	Sound Controller 6	Normal LFO Rate
76	Sound Controller 7	Amp Env Decay
77	Resonance	Filter Resonance
91	Effects 1 Depth	FX Bus Select, described in Chapter 2.
98	Non-Reg. Param. Select LSB	Track parameter descriptions in Chapter 2 list track parameters' Non-Registered parameter LSB values
99	Non-Reg. Param. Select MSB	always 0
100	Reg. Param. Select LSB	always 0, 1 or 2 only
101	Reg. Param. Select MSB	always 0
119	Mute	values mute or un-mute track corresponding to MIDI channel: 127=mute track; 000=un-mute track; 064=remove track from group solo

Reset All Controllers (MIDI controller #121) Reception Behavior

When the system ResetControlRecv=Off, the reset all controllers message will be ignored.

When system ResetControlRecv=On, the following MIDI messages and parameters on all tracks assigned to the MIDI channel on which the message was received will be reset to the following values:

Assignable SysCtrl1-4=000	Controller 008=064	Controller 070 to 071=000
Pitch Bend=center	Controller 009=000	Controller 072 to 079=064
Channel Pressure=000	Controller 010=064	Controllers 080 to 097=000
Polyphonic Pressure=000 for all 88 keys	Controller 011=127	Controller 098 to 101=cleared
Controllers 001 to 004=000	Controllers 012 to 031=000	Controllers 102 to 119=000
Controller 005=064	Controllers 033 to 064=000	Controllers 120 to 127=left unchanged
Controller 006=000	Controller 065=000	
Controller 007=127	Controllers 066 to 069=000	

When Track ParamReset=Off:

Controllers 005, and 070 to 079 will be left unchanged.

When Track ParamReset=On:

Controllers 005, and 070 to 079 will be reset to the values listed above.

Note: Track MIDI reception filters do not affect reception of the Reset All Controllers message.

Track ParamReset Behavior

When the System/MIDI Track ParamReset parameter is set to "On," selecting a new sound for a track causes certain parameters on the track to reset to default values. The following details the behavior of all of the track parameters in this regard.

Track parameter	Is parameter reset on sound selection?	Parameter's default value
Track Volume	no	n/a
Mix (Expression)	no	n/a
Vol/MixPolarity	no	n/a
Track Pan	no	n/a
FX Bus	see "AutoSelect FXBus "	n/a
Pitch Bend Up	yes	Prog
Pitch Bend Down	yes	Prog
Octave Shift	yes	0oct
Semitone Shift	yes	0st
Fine Tuning	yes	0cents
PitchTbl	yes	Prog
Glide Mode	yes	Prog
Glide Time	yes	Prog
Delay Offset	yes	0ms
SyncLFO&Noise	yes	Prog
Normal LFO Rates	yes	0
LFO Depth	yes	0
LFO Delay Time	yes	0
Amp Env Attack	yes	0
AmpEnv Decay	yes	0
AmpEnv Release	yes	0
Filter Cutoff	yes	0
Filter Resonance	yes	0
FiltEnv Attack	yes	0
FiltEnvDecay	yes	0
FiltEnvRelease	yes	0
Amp&Filt Env Vel	yes	0
Key Range Lo	no	n/a
Key Range Hi	no	n/a
VelocityRange Lo	no	n/a
VelocityRange Hi	no	n/a
VelocityMode	no	n/a
PressureMode	yes	Auto
ProgramChngeRecv	no	n/a
Bank Select Recv	no	n/a
Data Entry Recv	no	n/a
Pitch Bend Recv	no	n/a
Mod Wheel (1) Recv	no	n/a
FootPedal (4) Recv	no	n/a
Volume (7) Recv	no	n/a
Pan (10) Recv	no	n/a
Expressn (11) Recv	no	n/a
Sustain/SostRecv	no	n/a
SysCtrl1 Recv	no	n/a
SysCtrl2 Recv	no	n/a
SysCtrl3 Recv	no	n/a
SysCtrl4 Recv	no	n/a

Using RPNs and NRPNs to Edit Parameters

MIDI allows for a special category of controllers called RPNs (for "Registered Parameter Numbers") and NRPNs (for "Non-Registered Parameter Numbers"). Many sound parameters can be edited via RPNs and NRPNs. If this is the case, the parameter's description found in this chapter will list the appropriate RPN or NRPN. If a parameter is displayed while being edited via MIDI, the display will reflect the changes you make.

RPN MIDI messages must adhere to a specific structure in order to be properly understood by receiving devices such as the ASR-X. They must include the following components:

- A continuous controller status byte for the appropriate MIDI channel—this will be the MIDI channel of the selected track (see Chapter 2)
- MIDI controller 101—the RPN MSB—with a value of 000
- MIDI controller 100—the RPN LSB—with the RPN value listed in the description of the relevant parameter
- MIDI controller 006—Data Entry—with the value to which you'd like to set the parameter. The values displayed for each parameter correspond to one of 128 possible MIDI values (which run from 000 up to 127). You can count the parameter values displayed on the ASR-X, beginning from 000, to locate the corresponding Data Entry value you'll want to send to the ASR-X.

NRPN MIDI messages must also adhere to a specific structure in order to be properly understood by receiving devices such as the ASR-X. They must include the following components:

- A continuous controller status byte for the appropriate MIDI channel—this will be the MIDI channel of the selected track (see Chapter 2)
- MIDI controller 099—the NRPN MSB—with a value of 000
- MIDI Controller 098—the NRPN LSB—with the NRPN value listed in the description of the relevant parameter
- MIDI Controller 006—Data Entry—with the value to which you'd like to set the parameter. The values displayed for each parameter correspond to one of 128 possible MIDI values (which run from 000 up to 127). You can count the parameter values displayed on the ASR-X, beginning from 000, to locate the corresponding Data Entry value.

Registered Parameters

Registered parameters 0, 1 and 2 are received multi-timbrally by the ASR-X. When received on a track's MIDI channel, RPN 0 affects the track's pitch bend up and down simultaneously: Pitch bend up is raised and pitch bend down is lowered by the same RPN value. RPNs 1 and 2 edit Semitone Shift and Fine Tuning parameters, respectively, when received on the track's MIDI channel.

Registered parameters must be transmitted to the ASR-X as a continuous controller status byte followed by three consecutive continuous controller messages: The registered parameter MSB and LSB values select the track parameter that will be edited, and a Data Entry value invokes the parameter's setting.

Controllers

Number	Name	Value
101	Registered Parameter Select MSB (Most Significant Byte)	always 0
100	Registered Parameter Select LSB (Least Significant Byte)	00, 01 or 02 (see below)
6	Data Entry MSB	0-127, desired track parameter setting

Registered Parameters

Number	Name	ASR-X Parameter Range
00	Pitch Bend Range	0-12 (displayed as Pitch Bend Up =0-12 up; raises pitch; Pitch Bend Down=0-12 down
01	Fine Tuning	0-127 (displayed as -50 cents to +49 cents)
02	Coarse Tuning	0-127 (displayed as -64st to +63st)

Non-Registered Parameters

Non-registered parameters are received multi-timbrally by the ASR-X, affecting track parameters when received on the track's MIDI channel.

Non-registered parameters must be transmitted to the ASR-X as a continuous controller status byte followed by three consecutive continuous controller messages. The non-registered parameter MSB and LSB select the track parameter, and a data entry value invokes the track parameter's desired setting.

Controllers

Number	Name	Value
99	Non-Registered Parameter Select MSB (Most Significant Byte)	always 0
98	Non-Registered Parameter Select LSB (Least Significant Byte)	see track parameter descriptions in Chapter 2 for each parameter's Non-Registered parameter LSB value
6	Data Entry MSB	0-127, desired track parameter setting

List of RPNs and NRPNs

Track Parameter	Editing via MIDI
Expression	Responds to MIDI controller 011 and NRPN LSB 034.
FX Bus assignment (Insert, LightReverb, MediumReverb, WetReverb, Dry)	Responds to MIDI NRPN LSB 033.
Pitch Bend Up	Responds to MIDI NRPN LSB 022 (also responds to RPN LSB 000).
Pitch Bend Down	Responds to MIDI NRPN LSB 023 (also responds to RPN LSB 000).
Octave Shift (-4oct to +4oct)	Responds to MIDI NRPN LSB 011.
Semitone Shift	Responds to MIDI RPN LSB 002.
Fine Tuning	Responds to MIDI RPN LSB 001.
Pitch Table	Responds to MIDI NRPN LSB 021.
Glide Mode	Responds to MIDI controller 065 (see below) and NRPN LSB 031. When a value of 64 or greater for MIDI controller 065 is received, glide will be enabled for the part; values below 64 will not disable glide.
Glide Time	Responds to MIDI controller 005 and NRPN LSB 032.
Delay Offset (positive-only)	Responds to MIDI NRPN LSB 024.
SyncLFO&Noise (system tempo time division)	Responds to MIDI NRPN LSB 025.
Normal LFO Rates	Responds to MIDI controller 075 and NRPN LSB 008.
LFO Depth	Responds to MIDI NRPN LSB 009.
LFO Delay Time	Responds to MIDI NRPN LSB 010.
Amplitude Envelope Attack time	Responds to MIDI controller 073 and NRPN LSB 014.
Amplitude Envelope Decay time	Responds to MIDI controller 076 and NRPN LSB 015.
Amplitude Envelope Release time	Responds to MIDI controller 072 and NRPN LSB 016.
Filter Cutoff (lo-pass & hi-pass)	Responds to MIDI controller 074 and NRPN LSB 012.
Filter Resonance	Responds to MIDI controller 077 and NRPN LSB 013
Filter Envelope Attack time	Responds to MIDI NRPN LSB 017.
Filter Envelope Decay time	Responds to MIDI NRPN LSB 018.
Filter Envelope Release time	Responds to MIDI NRPN LSB 019.
Amp & Filter Envelope Velocity sensitivity	Responds to MIDI NRPN LSB 020.
Key Range Low limit	Responds to MIDI NRPN LSB 026.
Key Range High limit	Responds to MIDI NRPN LSB 027.
Velocity Range Low limit	Responds to MIDI NRPN LSB 028.
Velocity Range High limit	Responds to MIDI NRPN LSB 029.
VelocityMode	Responds to MIDI NRPN LSB 035.
Pressure Mode	Responds to MIDI NRPN LSB 030.
Mute button	Responds to MIDI NRPN LSB 036 (0=normal muted, 1=unmuted, 2=solo muted, 3=solo, 4-127=solo).

Pitch Tables and the MIDI Tuning Standard Format

Pitch tables created using an external computer can be downloaded into the ASR-X's RAM pitch table using the MIDI Tuning Standard format. The ASR-X can accommodate one user-defined RAM pitch table in addition to the many alternate pitch tables stored in ROM. The ASR-X's pitch tables can be accessed by any of its 16 tracks through the setting of the track's PitchTbl parameter, or via NRPN LSB 021 values sent on the track's MIDI channel. You can also select a system-wide special pitch table by selecting the desired table with the PitchTbl System parameter.

The MIDI Tuning Standard is comprised of two kinds of messages: the MIDI Tuning Dump, a SysEx bulk dump which transmits tunings for all keys, and a Single-Note Tuning Change, which alters the tuning of a specific note. The SysEx bulk dump format is supported by several tuning editors for the Apple Macintosh and Microsoft Windows 95. It is anticipated that the Single-Note Tuning Change message will be employed by third-party tuning controllers to achieve Middle-Eastern music scales.

The ASR-X's response to the Single-Note Tuning Change message has been extended to allow users to apply a single tuning change to the ASR-X's entire pitch range. If a Single-Note Tuning Change message is sent to user-tuning number 7F (127), and if the note is between Middle C and an octave above (note numbers 60 to 71 inclusive), the tuning change will be applied to all notes in the current RAM pitch table. In all other cases, the note-change message only changes the tuning for the note specified. If a Single-Note Tuning Change message is received during playback of a note (between the key-down and key-up messages), the tuning change takes effect on the next note.

It is suggested that third-party tuning controllers should send a zero-pitch-detune message for each of the twelve notes supported by the Single-Note Tuning Change message and also select the RAM tuning for the receiving channel. The zero-pitch messages need only be sent once before sending their note-change messages.

For more information on the MIDI Tuning Standard, contact:

MIDI Manufacturer's Association
c/o Tom White, President
P.O. Box 3173
La Habra, CA 90632-3173
Phone/FAX: (310) 947-4569
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Just Intonation Network
535 Stevenson Street
San Francisco, CA 94103
Phone: (415) 824-5325
FAX: (415) 864-8726
WWW: http://www.dnai.com/~jinetwk

List of ROM System Pitch Tables

Pitch Table	Description
EqualTemper	The Western 12-tone equal-temperament tuning is used for the default pitch table.
Pythagrn-C	Early tuning derived by calculating 12 perfect fifths and adjusting the octaves downward as necessary. Leaves all fifths except the one between G# and D# very pure. The entire mathematical anomaly encountered by tuning up 12 perfect fifths (called the Pythagorean comma) is accounted for in the interval between G# and D#.
Just Int-C	Designed so that the major intervals in any scale are very pure, especially the third and fifth.
Meantone-C	One of the earliest attempts to derive a tuning which would accommodate music played in a variety of keys. The major third interval is very pure.
Wrkmeistr-C	Derived by Andreas Werkmeister, a contemporary of Bach, this is a further attempt to create a temperament which would accommodate music played in any key.
Vallotti-C	A variation of Pythagorean tuning in which the first 6 fifths in the circle of fifths are flat by 1/6 of the Pythagorean Comma. This is probably close to the tuning used by Bach for his Well-Tempered Clavier.
Grk-Diatonc	The basic building block of ancient Greek music (in which most modern Western music has its roots) was the tetra chord - four notes and three intervals spanning a perfect fourth. The placement of the two inner notes of the tetra chord determined its genus — diatonic, chromatic or enharmonic. This pitch table is derived from two diatonic tetra chords, combined to form a seven-note scale similar to the modern diatonic scale. It is to be played only on the white keys. Tone center is E.
Grk-Chromat	This pitch table is derived from two chromatic tetra chords (the intervals are, roughly, quarter-tone, half-step, major third), combined to form a seven-note scale. It is meant to be played on the white keys. Tone center is E.
Grk-Enharm	This pitch table is derived from two enharmonic tetra chords (the intervals are, more or less, two quarter-tones followed by a major third), combined to form a seven-note scale. It is meant to be played on the white keys. Tone center is E.
Turkish-A	This is a typical Turkish octave-based scale using only one quarter tone. The second note in the scale is tuned 40 cents flat from the equal-tempered equivalent. In this tuning B is 40 cents flatter from B natural. The scale rises from A.
Arabic-1	The intervals in this table form the basis for much Middle Eastern music. Here the octave is divided into 17 intervals, corresponding to the fret intervals of some stringed instruments used in this area. The scale rises from the base pitch of C4 in a series of three repeating intervals (in cents) of 90, 90, 24 and so on. From C4 to F5 represents an octave.

Arabic-2	Similar to Arabic 1, except that here the octave is divided into 24 intervals. This makes one pitch octave cover two keyboard octaves, meaning that the fingering will be the same in any octave. This scale rises from the base pitch of C4 in a series of four repeating intervals (in cents) of 24, 66, 24, 90 and so on.
Arabic-3	This is a 12-tone scale using quarter tones (notes tuned sharp or flat by 50 cents from their equal-tempered equivalents) on the C#, E, G# and B keys.
Arabic-4	Another octave-based scale with an Arabic flavor. In this case the "quarter tones" are not perfectly equal, imparting a distinctive character to the notes.
Java-Pelog1	One of the two main scales of the gamelan orchestras of Java and Bali is the seven-tone scale called Pelog. The notes C, D, F , G, and A (which are reproduced on the black keys) are considered primary, with E and B used for grace notes. The octaves are stretched (tuned a little sharp) due to the harmonic content of the instruments in the gamelan. (There are many variations of these tunings, almost as many as there are gamelan ensembles. These tunings are to be considered typical, not definitive.)
Java-Pelog2	Another version of the seven-tone Pelog scale used in gamelan music. The notes C, D, F , G, and A (which are reproduced on the black keys) are considered primary, with E and B used for grace notes. The octaves are stretched (tuned a little sharp) due to the harmonic content of the instruments in the gamelan.
Java-Pelog3	A third version of the seven-tone Pelog scale used in gamelan music. The notes C, D, F , G, and A (which are reproduced on the black keys) are considered primary, with E and B used for grace notes.
Java-Slndro	A 15-tone equal tempered tuning from Java. Playing every third note (as in a diminished chord) yields a typical 5-tone scale of the gamelan. Other notes can be used as passing tones.
Java-Combi	This is actually two pitch tables in one. The white keys play the seven-tone Pelog scale, same as the table JAVA-PELOG1. The black keys play a five-tone scale called Slendro, which is close to a five-tone equi-tempered scale. Both tunings have their octaves stretched (tuned a little sharp) due to the harmonic content of the instruments in the gamelan.
Indian-Raga	Indian scale used to play ragas, based on 22 pure intervals called Srutis. This pitch table uses two keyboard octaves to play one octave in pitch. The 22 Srutis are mapped to keys in this two-octave range omitting the A#s, which play the same pitch as the adjacent A.
Tibetan	This tuning is based on a pentatonic scale from Tibet. Notice that playing the black keys yield a scale similar to the 5-tone Slendro tuning from Indonesia.
Chinese-1	This is a seven-tone scale used widely in China. It is meant to be played on the white keys.
Chinese-2	A seven-tone scale based on an ancient Chinese lute tuning. It is meant to be played on the white keys.
Thailand	This is a seven-tone equi-tempered scale from Thailand. It is meant to be played on the white keys.
24-Tone-Equ	Centered on C4, this scale has an even quarter tone (50 cents) between each keyboard note, and each pitch octave covers 2 keyboard octaves. This tuning has been used by many contemporary composers and can be used in some Middle Eastern music.
19-Tone-Equ	Centered on C4, this scale divides the octave into 19 equal steps. From C4 to G5 forms an octave. This scale yields very pure thirds and sixths, but not fifths. Like the 24-tone scale, this has been used by some modern composers.
31-Tone-Equ	Centered on C4, this scale divides the octave into 31 equal steps. From C4 to G6 forms an octave. Similar to 19-tone in the purity of its intervals.
53-Tone-Equ	This scale divides the octave into 53 equal steps. From C2 to F6 forms an octave. It yields very pure thirds, fourths and fifths.
Harmonic	This is a mathematically generated scale based on the relationships of the partials in the harmonics of the fifth octave of the linear harmonic spectrum. It is interesting mostly from a theoretical standpoint.
CarlosAlpha	Derived mathematically by Wendy Carlos in the search for scales with the maximum purity of primary intervals, This is based on the division of the octave into 15.385 equal steps (78 cents per key). One pitch "octave" covers 16 keys, though because the Carlos scales are asymmetric (not based on whole number divisions of the octave) they do not yield pure octaves.
Carlos-Beta	Wendy Carlos' Beta scale is based on the division of the octave into 18.809 equal steps 63.8 cents per key. One pitch "octave" covers 19 keys; though, being asymmetric, it yields no pure octaves.
CarlosGamma	Wendy Carlos' Gamma scale is based on the division of the octave into 34.188 equal steps (35.1 cents per key). This scale has essentially perfect major thirds, fourths and fifths. One pitch "octave" covers 35 keys, though, again, being asymmetric it yields no pure octaves.
Partch-43	Harry Partch was a pioneer of micro-tonality in the early 20th century. He developed this 43-tone-per-octave scale of pure intervals, and even designed an entire orchestra of instruments using this scale. The tonal center is found on key D2 (the low D on the 76-note keyboard). This pitch table has been transposed up an octave to bring the notes into a more usable range.
Reverse	This pitch table simply reverses the pitch-tracking of the keyboard, putting the highest notes at the bottom of the keyboard and the highest notes at the top. Hours of fun.
Bagpipe	This is the tuning of a traditional Scottish bagpipe.
ShonaMbira1	One tuning of the African Mbira, similar to the Kalimba or thumb-piano. Each Mbira player uses his own "tuning" which is his signature.
ShonaMbira2	Another Mbira tuning.
SuperJust	This is a Just Intonation scale created by Wendy Carlos.
88CET	88CET is a scale with a constant interval of 88 cents. It features three different thirds and close approximations to many just intervals. This keyboard mapping omits the G# / Ab key from the system.
Pierce-Bohl	An octave-repeating stretched scale invented by John Pierce which is derived from a pure twelfth divided into thirteen steps.
WS1	The WS scales are for single samples which span the entire keyboard. WS1 maintains 12 tones per octave for two octaves centered on middle C, then continues to high and low ends of the keyboard with 1/4 of a semitone or 48 tones per octave.
WS2	WS2 maintains 12 tones per octave for three octaves centered on middle C from G to G.
WS3	WS2 maintains 12 tones per octave for four octaves centered on middle C.
Stretch	A stretch tuning, in which the middle C is at unity, C1 is detuned flat 40 cents and C8 is detuned sharp 40 cents. The stretch is a linear ramp between these two offsets.
RandomDetun	Each note has been "tweaked" by + or - up to 10 cents, giving chords a chorused effect which is different for each note.
RAM	Selects pitch tables that can be downloaded via MIDI. See earlier for more information about RAM pitch tables.

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"INSTRUCTIONS PERTAINING TO A RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS"

IMPORTANT SAFETY INSTRUCTIONS

WARNING—When using electric products, basic precautions should always be followed, including the following:

1. Read all the instructions before using the product.

2. Do not use this product near water - for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.

3. This product should be used only with a cart or stand that is recommended by the manufacturer.

4. This product, either alone or in combination with an amplifier and headphones or speakers, may be capable of producing sound levels that could cause permanent hearing loss. Do not operate for a long period of time at a high volume level or at a level that is uncomfortable. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.

5. The product should be located so that its location or position does not interfere with its proper ventilation.

6. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.

7. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.

8. This product may be equipped with a polarized line plug (one blade wider than the other). This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the plug.

9. The power supply cord of the product should be unplugged from the outlet when left unused for a long period of time.

10. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

11. The product should be serviced by qualified service personnel when:

    a. The power supply cord or the plug has been damaged; or
    b. Objects have fallen, or liquid has been spilled into the product; or
    c. The product has been exposed to rain; or
    d. The product does not appear to operate normally or exhibits a marked change in performance; or
    e. The product has been dropped, or the enclosure damaged.

12. Do not attempt to service the product beyond that described in the user-maintenance instructions. All other servicing should be referred to qualified service personnel.

SAVE THESE INSTRUCTIONS

LEADING THE WORLD IN SOUND INNOVATION