User's Guide

ASRX PRO

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.

CAUTION

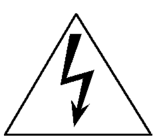
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION : TO REDUCE THE DANGER OF ELECTRIC SHOCK

DO NOT REMOVE COVER (OR BACK)

NO USER SERVICEABLE PARTS INSIDE

REFER SERVICING TO QUALIFIED SERVICE PERSONNEL



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 PRO

User's Guide

Version 3.00

ASR-X Pro User's Guide

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World Wide Web—http://www.ensoniq.com
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 Pro 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 9310 0231 01-A Model Number MM-169

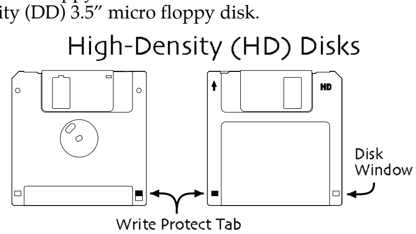
Temperature Guidelines

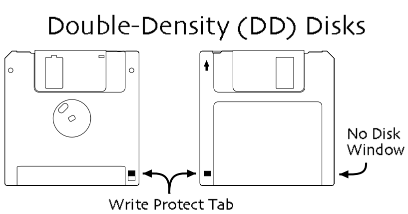
The ASR-X Pro contains a substantial amount of computerized and electronic circuitry that can be susceptible to damage when exposed to extreme temperature changes. When the ASR-X Pro is brought inside after sitting in a cold climate (i.e., the back seat of your car), condensation builds up on the internal 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 Pro:

- Avoid leaving the ASR-X Pro in temperatures of less than 50 degrees Fahrenheit or more than 100 degrees Fahrenheit.
- When bringing the ASR-X Pro 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 Pro inside a vehicle exposed to direct sunlight.

Care and Feeding of the Disk Drive

The ASR-X Pro'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.





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 Pro. 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:

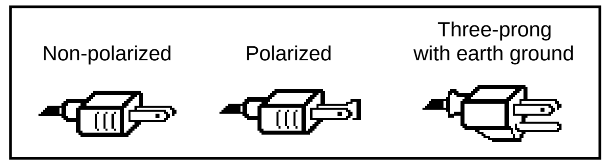
- Don't use single-sided (SD) disks. These disks have not passed testing on both sides. While a single-sided disk might work with the ASR-X Pro, it is possible that you will eventually lose important data to a disk error if you try using single-sided disks.
- Don't put anything other than a disk into 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 Pro 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 Pro 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 Pro 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 Pro in lots of different locations with varying or unknown AC line conditions, you might consider investing in a line conditioner.

Table of Contents

Tutorial

Introduction

Welcome to the ASR-X Pro User's Guide, and congratulations on your purchase of the ENSONIQ ASR-X Pro. This book contains a step-by-step tour of the major features of the ASR-X Pro. For more detailed information on the topics discussed—and more—see the ASR-X Pro Reference Manual.

The first part of the User's Guide is structured as a tutorial, meant to be read and followed in the order in which it's presented. This approach will allow you to become familiar with ASR-X Pro concepts and procedures one at a time, and will let you build up your understanding of—and comfort with—the way the ASR-X Pro works. The tutorial will only take about a half hour to complete. Along the way, you'll get a sense of how much fun it is to create music on your new ASR-X Pro. The remainder of the User's Guide contains a list of the sounds built into your ASR-X Pro, and a list of its insert effects and their parameters.

Note: At the end of various sections of the tutorial, you'll see a "Before proceeding..." section. The instructions in each these sections help set up the tutorials that follow.

What Else is in the ASR-X Pro Box?

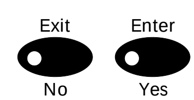
The following items are included with every ASR-X Pro shipped from the ENSONIQ factory:

- ENSONIQ X-Audio Sampling CD Volume 1—Producers' Mix
- hex wrench
- ENSONIQ ASR-X Pro User's Guide
- ENSONIQ ASR-X Pro Reference Manual
- AC power cable

Getting Around on the ASR-X Pro

The Display and Yes/No Buttons

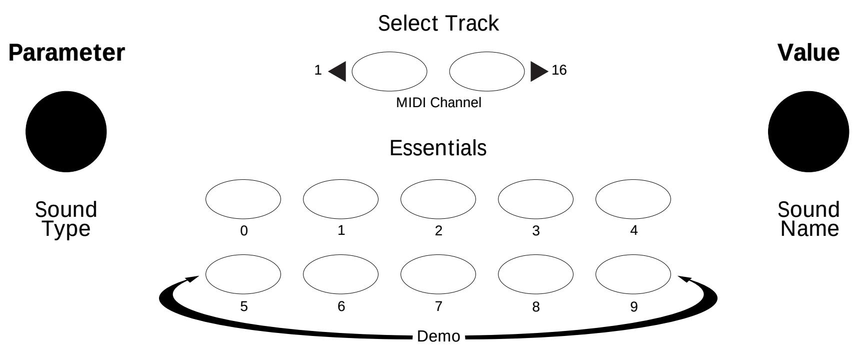
The display located in the center of the ASR-X Pro front panel will always tell you what you need to know as you do different things on the ASR-X Pro. Many ASR-X Pro procedures will be presented as questions that you can answer by pressing the No or Yes button.



Tip: When a question is being asked, the LEDs in the No and Yes buttons will flash.

Some activities involve a series of parameters and/or procedures. In such cases, you'll begin by answering "Yes" to a top-level question. From there you'll encounter parameters and/or further questions presented on sub-displays that relate to what you're doing. To exit back out to the top level of the ASR-X Pro, you can press the Exit/No button.

The Knobs



Below the display in the center of the front panel are two knobs used during most ASR-X Pro activities. These knobs are known by two different sets of names, since they serve two general purposes.

When you're selecting sounds for tracks or for pads in the ASR-X Pro, the central knobs should be thought of as the "Sound Type" and "Sound Name" knobs.

For every other activity, the knobs will be referred to as the "Parameter knob" and the "Value knob."

Both sets of names are printed on the front panel, as you can see.

Some ASR-X Pro Terms You Really Need to Know

It's important that you understand the meaning of these terms as you use the ASR-X Pro:

- parameter—This is any characteristic of the ASR-X Pro software that can be changed.
- value—This is the setting of a parameter.
- select—This is the act of choosing a sound for playing or recording, or choosing a parameter to be edited.
- standard sound—A sound is a program that plays one or more digital audio recordings arranged in layers. The recordings, or *waves*, can be data permanently stored in the ASR-X Pro's ROM (for "read-only memory"), or something you've sampled or loaded into the ASR-X Pro. Standard sounds based on ROM waves can have up to 16 layers; sounds that play sampled waves use one layer for mono waves, or two layers for stereo waves.
- kit sound—A kit is a sound in which each of its 64 notes can play a different standard sound. Since the ASR-X Pro is designed for beats and loops, kit sounds are its most potent sounds.

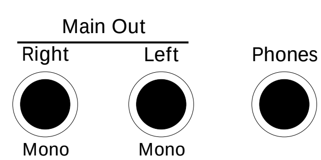
Tip: See Chapter 3 of the ASR-X Pro Reference Manual to learn more about standard and kit sounds.

Getting Ready to Go

Making Connections

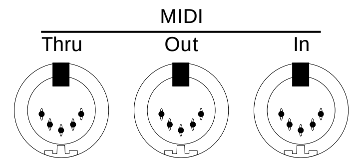
There are a few types of connections you'll want to make as you set up the ASR-X Pro —all of the jacks necessary for these connections are located on the rear panel of the ASR-X Pro. You'll want to set up:

- a way of listening to the ASR-X Pro—You can connect the ASR-X Pro's Left and Right Main Out jacks to a mixer, amplifier or stereo system. If you'd like to use the ASR-X Pro 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. If you'd rather listen to the ASR-X Pro through headphones, you can plug yours in to the ASR-X Pro's Phones jack—your headphones will need to have a 1/4" stereo plug or adapter to work correctly with the Phones jack.



Warning: If you connect the Main Outs to a stereo system, set your ASR-X Pro Volume level carefully, or you risk damaging components of your stereo. See "Setting Levels" below.

- If you plan to use any external MIDI devices with the ASR-X Pro—such as a keyboard, sequencer or computer—you'll need to connect:
  - the MIDI In of the ASR-X Pro to the MIDI Out of the external device.
  - the MIDI OUT of the ASR-X Pro to the MIDI In of the external device.



Tip: You'll be able to verify that your external MIDI device is properly connect to the ASR-X Pro by transmitting some MIDI data to the ASR-X Pro —when the ASR-X Pro receives the data, its front-panel MIDI LED will light.

- AC power for the ASR-X Pro—by connecting one end of the supplied power cord to the AC Line jack on the back of the ASR-X Pro, and the other end to a grounded AC outlet.

Powering Up

Warning: Before powering up your ASR-X Pro for the first time, turn its front-panel Volume knob all the way down (counter-clockwise) before turning it on. This will help make sure that you've set safe levels for all of your equipment before you make any sound on the ASR-X Pro.

To power up the ASR-X Pro, press in the top of its rear-panel power switch. When you do so, the ASR-X Pro will start up and display:

```
 ENSONIQ  ASR-X PRO
Resampler w/ Effects
```

Setting Levels

When Connected to a Mixer or Amplifier

The ASR-X Pro will produce its best sound when its volume knob is turned all the way up, so the best way to set up your ASR-X Pro levels is to:

1. Turn the volume of the channels to which you've connected the ASR-X Pro all the way down. If you're connected to a mixer, turn down the channel preamps as well.
2. Turn the ASR-X Pro Volume knob all the way up.
3. Play the ASR-X Pro pads with a good amount of force.
4. Set the levels on your mixer or amplifier to a workable setting.

When Connected to a Home Stereo System

Since the dynamic range produced by the ASR-X Pro is greater than that of a CD, record or cassette, set your levels carefully:

1. Set the stereo to your normal listening level.
2. Play the ASR-X Pro pads with a good amount of force.
3. While playing, slowly bring up the setting of the ASR-X Pro volume knob to an acceptable level that doesn't cause your stereo to distort.

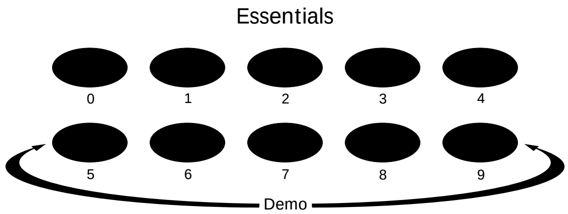
When Using Headphones

1. Play the ASR-X Pro pads with a good amount of force.
2. Slowly bring up the setting of the ASR-X Pro volume knob to a comfortable listening level.

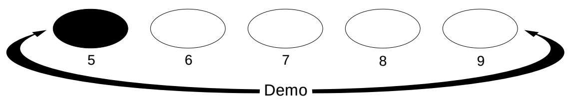
Playing the ASR-X Pro Demos

The ASR-X Pro contains demos to give you an idea of what it sounds like. To play the main demo:

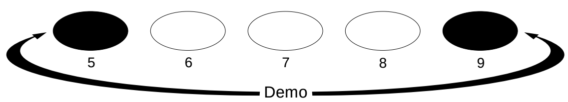
1. Locate the Essentials buttons on the ASR-X Pro front panel.



2. Hold down the Essentials button with "5" printed beneath it.



3. While still holding the button down, press the Essentials button with "9" printed beneath it.



4. Release both buttons.
   The display shows:

```
Start demo playback?
MAINDEMO:   Internal
```

5. Press the Enter/Yes button to hear the demo.
6. To stop the demo, press the Track Sound button in the center of the ASR-X Pro front panel.

Note: You can actually press any button to stop the demo—the Track Sound button was specified for the purposes of this tutorial.

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7. To listen to any of the other built-in demos, turn the Parameter knob to select a demo category, the Value knob to pick an individual demo from the selected category, and press the Yes button.

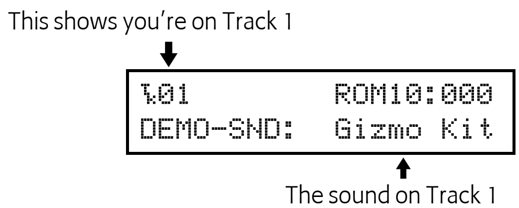
Selecting Tracks

The ASR-X Pro is organized into a framework of 16 tracks that play an important role in the ASR-X Pro:

- When you're playing sounds or sampling/resampling in the ASR-X Pro, you can think of each track as being a container for the sound with which you're working.
- When you're recording—or *sequencing*—each track contains a recorded performance, as well as the sound that plays it.
- The ASR-X Pro is a multi-timbral MIDI receiver—the 16 tracks correspond to MIDI channels 1-16. Each track receives MIDI data on its like-numbered channel. You can play a track's sound or record on a track via MIDI at any time.
- Each track transmits MIDI data on its like-numbered channel when it uses a MIDI-OUT sound.

Whenever you do anything in the ASR-X Pro, you're always on one or another of these tracks, which is referred to as "the currently selected track." The ASR-X Pro display tells you which track is currently selected—that is, which track you're on.

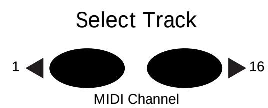
Since you've just powered up and played the demo, you're on Track 1:



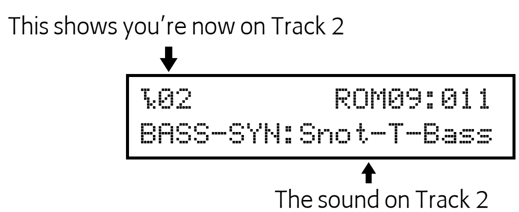
Tip: The currently selected track is shown on the display during many ASR-X Pro activities.

To Select One of the 16 Tracks

1. Locate the Select Track buttons.



2. Press the right-hand Select Track button once. The display shows that you've just selected the next track, Track 2.



   Higher-numbered tracks are selected by pressing the right-hand Select Track button, while lower numbered tracks are selected by pressing the left Select Track button.

3. Press the left Select Track button to go back to Track 1.

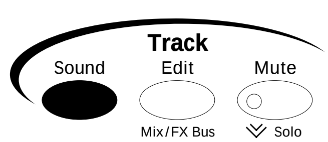
Selecting Track Sounds

Each track can play any of the sounds in the ASR-X Pro. Finding just the right sound is simple, thanks to the ASR-X Pro's SoundFinder™ feature. With SoundFinder, you select the type of sound you want, and then choose a specific sound of that type.

Tip: For a list of SoundFinder categories, see Chapter 9 of the ASR-X Pro Reference Manual. For a list of all of the ASR-X Pro's built-in sounds, see "List of ROM Sounds" later in the User's Guide.

To Select a Sound for a Track

1. Check the display's upper left-hand corner to make sure that you've got the desired track selected—for this tutorial, you'll use Track 1.
2. Press the Track Sound button.



   (Later in this section you'll learn why "DEMO-SND" just changed to "DRUM-KIT" on the display.)

3. Turn the Sound Type knob to view the SoundFinder categories that are currently available. SoundFinder categories are shown in the lower-left corner of the display. (When there are no sounds in memory of a particular type, its SoundFinder category isn't shown.)

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

The currently selected SoundFinder type

4. Select the DRUM-KIT SoundFinder type.
5. Turn the Sound Name knob to view the various drum kits in the DRUM-KIT SoundFinder category.

```
%01        ROM10:005
DRUM-KIT:Ol'SkoolKit
```

As you turn the Sound Name knob, the sounds in the selected SoundFinder category are displayed

You can also select sounds using the Essentials buttons. Each button can be pressed to instantly place a favorite sound onto the current track. To learn more about the Essentials buttons, see Chapter 2 of the ASR-X Pro Reference Manual

6. Press some of the Essentials buttons to see how they work, and to check out the sounds that ENSONIQ assigned to the buttons at the factory. You can also press any upper Essentials button and the button just beneath it—an additional five sounds can be accessed in this manner.
7. Press the Essentials button with the "2" beneath it to select the sound HeavyDrmKit. The display provides information about the sound you've selected:

The track you're on — The sound's bank number — The sound's program number

```
%01        ROM10:002
DRUM-KIT:HeavyDrmKit
```

The selected SoundFinder type — The selected sound

Sounds are stored in groups called *banks*. Each bank can contain up to 127 sounds, each of which is numbered. This system allows you to select sounds via MIDI, since each bank's number is also its MIDI Bank Select number, and each sound's program number is also its MIDI Program Change number. By sending the correct MIDI Bank select and Program Change values on a track's MIDI channel, you can select its sound from an external MIDI device.

Tip: Each sound can be found in the bank in which it's stored and in an instrument category. You can press the Track Sound button to toggle between these. In Step 1, "DEMO-SND" changed to "DRUM-KIT" on the display when we pressed Track Sound for the second time in this tutorial.

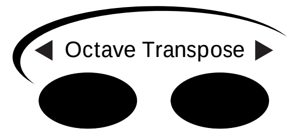
Playing with the Pads

Now that you've selected HeavyDrmKit for Track 1, why not play it? The pads on the ASR-X Pro are a keyboard of sorts that plays the sound you've chosen—it's a percussion-oriented keyboard perfect for a groovestation like the ASR-X Pro. Bang out a few notes to hear some of the bass drum and snare sounds in HeavyDrmKit. Since it's a kit sound, each pad can play a different standard sound.

ASR-X Pro sounds cover the entire pitch range typical of any modern MIDI instrument. The 13 ASR-X Pro pads play a section of a sound at a time—an octave-plus-one-note's worth—in this case, the bass drum and snare range of HeavyDrmKit. To the left of the pads are the Octave Buttons. These are used for re-aiming the pads at different octaves within the currently selected sound.

To Change the Octave Played by the Pads

1. Locate the Octave Transpose buttons.



2. Press the right-hand Octave Transpose button once. The display briefly shows:

```
%01         ROM10:002
Pad Xpose=0oct C2-C3
```

The number of the octave being played by the pads   The note range being played

The Octave Transpose display shows the octave the pads are currently playing. The pads can play five different octaves, each of which has been assigned a number as shown below (Middle C is C4):

- Octave 0 plays C2-C3
- Octave 1 plays C3-C4
- Octave 2 plays C4-C5
- Octave 3 plays C5-C6
- Octave 4 plays C6-C7

3. Tap the right-hand Octave Transpose button twice to set the pads to play Octave 1. The first tap takes you to the Octave Transpose display, and the second tap raises the octave setting by one.

Tip: You can also select an octave by pressing either of the Octave Transpose buttons and turning the Value knob.

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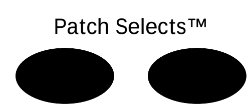
4. Play the pads—now you're playing the sounds contained in the second octave of HeavyDrmKit.
5. Use the Octave buttons to find the other sounds in HeavyDrmKit.

Before Proceeding...

1. Press the left Octave Transpose button to bring the pads down to Octave 0.

Note: If you see "Kit Mapper" on the display, press the right-hand Octave Transpose button once. The Kit Mapper is described in Chapter 3 of the ASR-X Pro Reference Manual.

Using the Patch Select Buttons



The Patch Select buttons call up variations of the currently selected sound. Each sound is made up of layers of waves, and the Patch Select buttons are programmed to turn layers or groups of layers on and off (see Chapter 3 in the ASR-X Pro Reference Manual for more on layers). Depending on the nature of the sound, this can lead to subtle or radical changes in the sound.

Patch Select buttons are able to operate in a couple of different ways. The default setting is for them to act as switches that only work as long as they're physically being held down. There are four possible positions for the Patch Select buttons:

- left Patch Select button pressed
- right Patch Select button pressed
- both Patch Select buttons pressed
- neither Patch select button pressed

The Patch Select buttons can also be set to "stick," as described in Chapter 7 of the ASR-X Pro Reference Manual.

As part of the demonstration of the Patch Select buttons, let's select a synthesizer bass sound for Track 2—we'll need it later on anyway.

1. Press the right-hand Select Track button once to select Track 2.
2. Using the Sound Name knobs, select the sound BuzzSawBass—since the BASS-SYN category was already selected, you didn't have to turn the Sound Type knob to select a SoundFinder category.
3. Play a few notes of BuzzSawBass to get an idea of its sound with no Patch Select buttons pressed.
4. Play the pads—with a standard sound such as BuzzSawBass, all the pads play the same basic sound at different pitches.

   BuzzSawBass is one of the sounds built into the ASR-X Pro's permanent ROM. Since ROM sounds can contain up to 16 layers of waves, the Patch Select buttons are most useful when used with these sounds.
5. Hold down either of the Patch Select buttons and play some notes on the pads—hear how that sound has changed? You can release the button to return BuzzSawBass to its original sound.
6. Try the other Patch Select positions with BuzzSawBass.

Tip: Patch Select button-presses will be recorded if they occur during the recording of a sequence track. They're also transmitted via MIDI so that they can be used for controlling other ENSONIQ products that use Patch Select buttons, and so that they can be recorded in an external sequencer.

Before Proceeding...

1. Press the left Select Track button to return to Track 1.

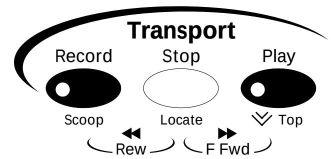
Sequencing in the ASR-X Pro

The ASR-X Pro contains a recording device called a *sequencer* that is both powerful and easy to use. A sequencer records the music you play on each of its 16 tracks. The music is recorded on each track as MIDI data, and is therefore highly editable. Each segment of music you record is called a *sequence*. There is always a sequence selected in the ASR-X Pro—even when you haven't yet recorded anything—and the track that you're on is in that sequence.

To Record a Sequence Track

Tip: Read through Steps 1 through 4 once before actually doing them so that recording doesn't sneak up on you before you're ready.

1. Press and hold down the Transport Record button.
2. While still holding the Record button, press the Play button, and then release both buttons. The Record and Play button LEDs light to show you're recording.



   The sequencer begins counting off four beats before recording begins.

   The display shows the countoff in negative numbers in the upper-right corner of the display—when the numbers get to 0, recording actually starts.

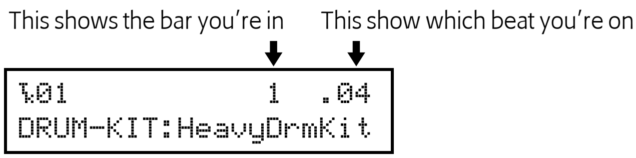
The numbers count up to the first beat of actual recording

```
%01            -1   .01
DRUM-KIT:HeavyDrmKit
```

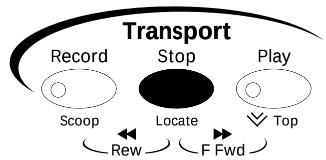
Tip: This countoff can be customized or turned off. See Chapter 6 in the ASR-X Pro Reference Manual.

You'll be recording two measures. On the first and third beats, play the lowest pad, a bass drum sound. On the second and fourth beats, play the highest pad, a snare drum sound.

3. At the first beat after the countoff, play the pads as described above. The display will show you where you are in each measure, in bars and beats.



4. At the end of the second measure, press the Stop button.



You've just recorded your first track in your first sequence.

To Play Back a Sequence Track

1. Hold down the Record button, press the Stop button, and then release both buttons to rewind to the top of the sequence.

Tip: The Transport buttons perform many tasks—see Chapter 6 in the ASR-X Pro Reference Manual to learn more.

---

2. Press the Play button.

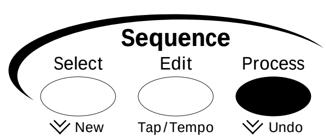
   The track you've just recorded plays back—the sequence loops, playing your recording over and over (this default sequence setting can be changed; see Chapter 6 in the ASR-X Pro Reference Manual).
3. Press the Stop button to end playback.

Undo

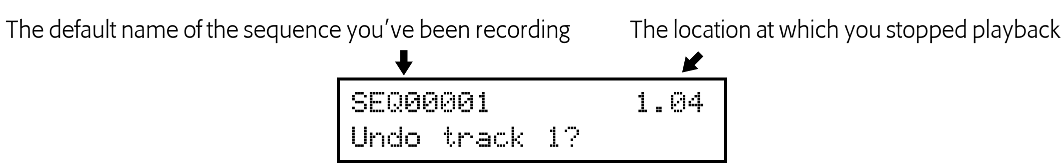
For situations in which you've recorded something you'd like to get rid of, or when you've done something to a track that you wish you hadn't done, the ASR-X Pro provides a handy Undo feature. As long as you haven't yet recorded other tracks or performed other procedures, you can undo whatever you've just done to a track.

To Undo a Track

1. Double-click the Sequence Process button.



   The display shows:



   The No/Yes LEDs flash to indicate that you're being asked a question.
2. If you'd like to undo your recording and try again, press the Yes button and repeat the steps in "To Record a Sequence Track" above. Press the No button if you're satisfied with your performance.

Correcting the Timing of a Performance

A sequencer such as the one in the ASR-X Pro can correct timing inconsistencies in a recorded performance—this process is called *quantizing*. The ASR-X Pro offers a sophisticated set of quantizing tools, as well as quantizing *templates*, in which common types of quantization are already pre-programmed, and that can hold your own favorite quantization settings.

To Quantize a Track

1. We'll be quantizing the just-recorded Track 1, which is therefore already selected. Normally, however, you'll want to make sure that the track you want to quantize is selected before proceeding.
2. Press the Sequence Process button. The display shows:

```
SEQ00001        1.01
Undo track 1?
```

   In this case, you don't want to undo the track, so don't press the Yes button. The Undo function is only one of the track and sequence operations accessed by pressing the Sequence Process button.
3. Turn the Parameter knob clockwise until the display shows:

```
SEQ00001        1.01
Quantize track 1?
```

4. Press the Yes button. The display now shows:

```
Quantize track 1?
Template=Strict 1/16
```

   This display is presenting you one of the pre-programmed quantizing templates. However, the track you've recorded contains 1/4 notes, not 1/16th notes.
5. Turn the Value knob counter-clockwise to select the Strict 1/4 template.
6. Press the Yes button.
7. Play back your track—the timing of the quarter notes in your track is now perfect.
8. Press the Stop button to end playback.

Tip: To learn more about quantizing, see Chapter 6 of the ASR-X Pro Reference Manual.

Adding to an Already-Recorded Track

The ASR-X Pro sequencer provides a variety of recording modes, described fully in Chapter 6 of the ASR-X Pro Reference Manual. These modes change the way in which the sequencer records what you play. In:

- Replace mode, what you record replaces anything already on the track.
- Add mode, what you record is added to what's already on the track. In this way, you can build up complex recordings, an element at a time.
- Step mode, you can enter notes or chords one at a time with the sequencer at rest.
- Track Mix allows you to record volume, panning and other track setting changes on a track.
- Final Mix lets you to record whole-sequence volume changes—such as fade-outs— and tempo changes.

Since you're in Add mode (it's the default setting), you can now add hihat 1/8th notes to your track.

1. Pressing the right-hand Octave Transpose button twice to aim the pads at HeavyDrmKit's second octave. Play the pads to locate a hihat sound you like.
2. Press the Record and Stop buttons to return to the beginning of the sequence.
3. Hold down the Record button, press Play and release both buttons when the countoff begins.

4. Record 1/8th notes on the hihat to accompany the two measures of bass drum and snare.
5. Press the Stop button at the end of the second bar—in Add mode, recording continues until you press the Stop button.
6. Play back your track to hear the bass drum, snare and hihat playing together.
7. If you'd like to redo the hihat, double-click the Sequence Process button, undo the track, and then record it again.
8. Correct the hihat's timing by repeating the steps in "Correcting the Timing of a Performance." You'll want to select the Strict 1/8th template this time, since you don't want your hihat 1/8th notes turned into 1/4 notes.

Before Proceeding...

1. Press the left Octave Transpose button twice to re-direct the pads back to the lowest octave.

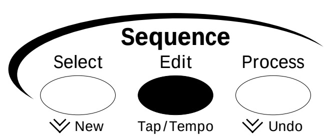
Recording Another Track

Let's record synth bass quarter notes on Track 2 to accompany the drums on Track 1. This time, however, don't record any notes during the first measure—wait until Bar 2 to begin playing. You can use any of the pads to play your quarter notes.

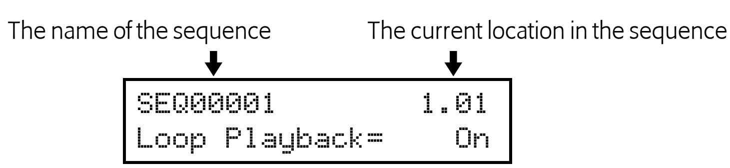
1. Press the right-hand Select Track button to select Track 2.
2. Hold down Record, press Stop and let go of both buttons to make sure you're at the beginning of the sequence.
3. Record your bass part using the same technique you used on Track 1.
4. Once you're satisfied with your performance, quantize the track using the Strict 1/4 quantization template.
5. Play back your bass-and-drums sequence, and press the Stop button when you're done listening.

Before Proceeding...

1. Locate and press the Sequence Edit button.



   This button accesses various settings relating to the behavior of the sequence.
2. Turn the Parameter knob until the display shows:



3. Turn the Value knob counter-clockwise to turn the playback looping off.

A Few Final Sequencing Thoughts

The Sequence Select, Edit and Process buttons provide access to a wealth of tools used in sequencing, from the creation and selection of sequences, to all of the settings that establish your recording environment, to the various ways that recorded tracks and sequences can be manipulated and processed after recording. You can also chain sequences together to try out arrangements, and create songs as well. Sequencing is described in detail in Chapter 6 of the ASR-X Pro Reference Manual.

The Track buttons—Sound, Edit and Mute—provide the means of setting up tracks and changing the way they play their sounds, changes you can make even as sequences play back. You can also mute and solo tracks using a highly musical Mute/Solo system. Chapter 2 in the ASR-X Pro Reference Manual is the place to go for information relating to tracks.

Effects

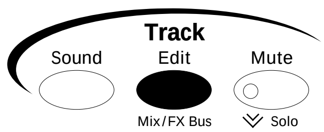
The ASR-X Pro contains a powerful ENSONIQ ESP2 effect chip, capable of providing a variety of exceptional effects. Each sequence contains its own:

- insert effect—An insert effect is one of a collection of 40 high-quality effects. Insert effects handle a wide range of processing jobs.
- global reverb—The global reverb in each sequence is intended as the all-purpose reverb you'll use for the sequence's tracks. Each sequence can use any one of the eight global reverbs available.

Sending a Track to an Effect

Tracks and their sounds are heard through one of their sequence's effects when they've been assigned to an *effect bus*, which routes the track to the effect you want. Each track has a parameter for this purpose. Let's change the effect through which we're hearing our bass.

1. Locate the Track Edit button.



2. Press the Track Edit button until the display shows:

The currently selected effect bus

The FX Bus parameter assigns the selected track to an effect bus, and therefore, an effect. The values available for this parameter vary slightly, depending on what kind of sound is on the selected track, and whether or not you've added an X-8 output expander to your ASR-X Pro. You can select:

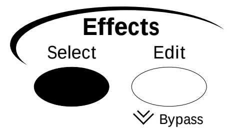
- Prog—to retain each note's individual effect bus in a kit sound. When a standard sound is on the selected track, this setting causes the sound to use its Alt Bus setting, described in Chapter 4 of the ASR-X Pro Reference Manual.
- Insert—to route the selected track and its sound to the sequence's insert effect.
- LightReverb—to apply a small amount of reverb to the track's sound.
- MediumReverb—to apply an average amount of reverb to the track's sound.
- WetReverb—to apply a large amount of reverb to the track's sound.
- Dry—to leave the track's sound un-effected.
- AuxOut1-4—to send the track's sound out of one of the auxiliary output jacks provided by an X-8 output expander. These values are only available when an output expander is installed.

3. Turn the Value knob to set the FX Bus parameter to each of the available settings, playing a few notes on the pads with each setting to hear how the bass sound changes.
4. Set the FX Bus parameter to "Insert."

Changing a Sequence's Insert Effect

Now that the bass is being sent to the sequence's insert effect, let's find out what that effect is, and then change it.

1. Locate the Effects Select button.



2. Press the Effects Select button once. The display shows the sequence's insert effect:

```
Insert Effect=
    20 DDL→Chorus
```

3. Turn the Value knob to select different insert effects and play some notes on the pads to hear what they sound like.

Note: Insert effects sometimes take a few moment to be installed, or "downloaded."

---

4. When you're done experimenting, select insert effect 25 Dist→Chorus.
   Play a few notes—the bass has now become something rough and wide.

A Few Final Effect Thoughts

A sequence's global reverb can be selected in the same way as its insert effect (press the Effects Select button a second time to view the global reverb selection page). The ASR-X Pro effects provide a wide selection of parameters that let you get each effect sounding just the way you want it to—these parameters can be accessed by pressing the Effects Edit button, selecting the appropriate question and pressing the Yes button.

Effect selection and editing is described in Chapter 4 of the ASR-X Pro Reference Manual. For a listing of the insert effects and their parameters, see "Insert Effect Parameters" later in the User's Guide.

Sampling/Resampling

The ASR-X Pro captures audio using a process called *sampling*, which is another term for digital recording. When you sample in the ASR-X Pro, you create a wave, or if you've sampled in stereo, a pair of waves. The ASR-X Pro creates a sound based on each wave (or pair of waves) you create, and when you play that sound, you play the wave.

The ASR-X Pro can sample audio that you feed it through its rear-panel Audio Input jacks, or it can *resample* sounds and sequences from the ASR-X Pro itself—or both sources at once. In addition, you can set up the ASR-X Pro to automatically begin sampling when it "hears" audio—a feature we'll try out in the following tutorial.

Note: The techniques for sampling and resampling are the same—the only difference is the source of the audio being sampled.

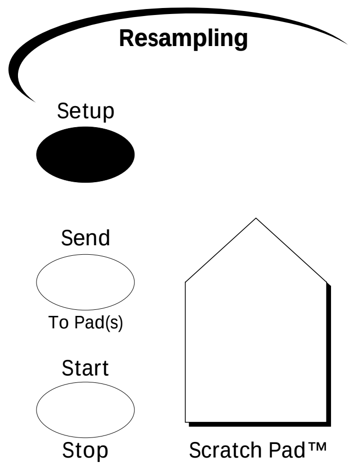
Resampling a Beat

Remember that you're always on a track in the ASR-X Pro. This is even true when you sample or resample. So that we don't get in the way of the tracks we've recorded, let's select an unused track to begin.

1. Press the right-hand Select Track button to select Track 3.

You're going to be putting the samples you create into one of the kit sounds, so let's select one that we're not already using in our sequence.

2. Turn the Sound Type knob to select the DRUM-KIT SoundFinder category, and turn the Sound Name knob to select Dance Kit.
3. Hold down the Record button, press the Stop button, and then release both buttons to make sure that we're at the beginning of our sequence.
4. Locate and press the Resampling Setup button.

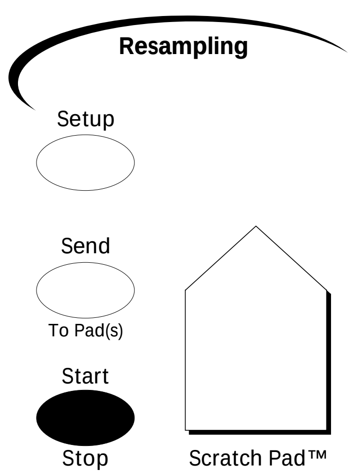


5. Turn the Parameter knob until the display shows:

```
Meter:
Trig=
```

This is where you can set a level that determines how loud audio has to be to cause the ASR-X Pro to begin sampling. This is referred to as the *trigger threshol*d.

6. Turn the Value knob counter-clockwise to move the level indicator as far leftward as it will go.
   The farther to the left the indicator is set, the quieter the audio that will cause sampling to begin.
7. Turn the knob the other way to move the indicator to the right by one position.
8. Locate the Resampling Start/Stop button.



9. Press the Resampling Start/Stop button once.
   The display shows "Waiting for trigger" on its top line, since the ASR-X Pro is waiting to hear some audio before it starts sampling.
10. Press the Transport Play button to play the sequence.
    On the display, you'll see "Sampling In Progress" to show that you're now sampling.

11. When the sequence ends, press the Sample Start/Stop button again to stop sampling.

    You've just created a stereo sample that's captured—as a pair of waves—the sequence you recorded earlier. Whereas a sequence is just MIDI data, sampling records the sound that MIDI data produces.

    The waves are in the Scratch Pad, which you can now play to hear what you've resampled.

12. The display is showing:

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

    At this point, you must send the waves to one or more pads in a drum kit in order to fully use and edit what you've sampled.

13. Strike the lowest pad. The ASR-X Pro takes this to mean that you want to play your waves from that pad. The display shows the pad you've hit.

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

This pad symbol represents the pad you hit

14. Strike the middle pad and the highest pad—the waves will be assigned to three pads, as shown on the display.

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

All three pads you hit are now shown on the display

Tip: You can un-select a pad by hitting it a second time, but for the purposes of the tutorial, you do want the waves sent to all three of the pads you've hit.

Tip: Though you shouldn't do this for the tutorial, you can use the Octave Select buttons to send your waves to a pad in any octave.

15. Press the Yes button to send the waves to the three pads.
16. Play the pads, you'll hear what you've sampled on the three pads you selected, and other elements of Dance Kit on the other pads.

What Happened When We Sent the Waves to the Pads?

A RAM Kit was Created

When you send waves to pads, those pads have to be in a sound that can accept them. Since all of the sounds built into your ASR-X Pro are permanently stored in ROM, none of them qualify—they're unchangeable. As a result, when you try to send waves into a ROM sound—if that's what's on the currently selected track—the ASR-X Pro makes an editable copy of the sound, into which you can send your waves. Such a sound is called a *RAM* (for "random access memory") *sound*, since that's where all editable items are stored in the ASR-X Pro's memory.

Kit sounds are ideal for sending waves into, since in kit sounds, each pad can have its own sound. If you try to send waves into a sound that's not a kit, the ASR-X Pro turns it into one—in such a kit, each pad will play a different pitch of the original sound.

Therefore, when you send waves into a sound, it has to be a *RAM kit* sound—if it's not, the ASR-X Pro automatically makes a RAM kit copy of the sound on the currently selected track. In our tutorial, that's just what happened. Dance Kit was a ROM sound, so the ASR-X Pro created a RAM kit copy of it that could accept your waves.

Press the Track Sound button to see what's on Track 3 now—it's a sound called DanceKit_01. When the ASR-X Pro makes a RAM kit copy of a sound, it names the copy after the original, adding a number onto it (if necessary, it automatically abbreviates the name of the original ROM sound).

Sounds Were Created that Play the Waves

When you sent your waves to pads, the ASR-X Pro created a standard sound in RAM that plays the waves. For each pad to which you sent the waves, it also created a copy of that sound. This allows each pad to play the waves in its own individual way, as you'll see in a few moments. Let's take a look at all of these new standard sounds that have been created.

Turn the Sound Type knob to select the *CUSTOM SoundFinder category. If necessary, turn the Sound Name knob all the way counter-clockwise so that the sound SMPL1 is displayed. This is the standard sound created when you sent your wave to pads. Each such sound is assigned a number—if you were to sample something else and send it to pads, its standard sound would be called "SMPL2."

Turn the Sound Name knob clockwise—now you can see SMPL1_01, SMPL1_02 and SMPL1_03, the three copies of SMPL1 that are played by the three pads in DanceKit_01. You can see that the copies of SMPL1 retain its name and are numbered.

Tip: You can use the Memory Manager to rename sounds in RAM. See Chapter 7 of the ASR-X Pro Reference Manual.

Before Proceeding

1. Go back to the DRUM-KIT SoundFinder category and select DanceKit_01.

A Few Final Sampling/Resampling Thoughts

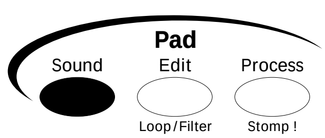
The ASR-X Pro's sampling/resampling features are designed to be easy and fun to use. To learn about sampling/resampling other kinds of audio, auto-normalization, the different trigger modes, using the rear-panel Audio Inputs and more, see Chapter 5 of the ASR-X Pro Reference Manual.

Editing Sampled Waves

All wave editing in the ASR-X Pro occurs in the Pads section of the front panel. Let's tie together what we've just discussed by pressing the Pad Sound button to see what sound each pad in DanceKit_01 is playing.

Viewing Pad Sounds

1. Locate and press the Pad Sound button.



When you've pressed the Pad sound button, you can hit the pads to see the sounds they're playing.

Tip: If the pads you're viewing are part of a RAM kit, you can—but don't, for the tutorial—select a new sound for each pad when this display is visible.

2. Play the lowest pad. The display shows:

The pad you've selected — The currently selected track

```
padC2  %03 RAM00:002
*CUSTOM :   SMPL1_01
```

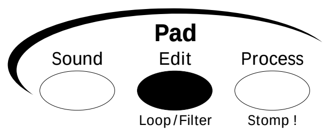
The sound being played by the pad

As you might expect, this pad—the first pad to which you sent your waves—is playing the first copy of SMPL1: SMPL1_01.

3. Hit the other two pads to which you sent waves—they play SMPL1_02 and SMPL1_03.
4. You can check out the sounds being played by the other pads in DanceKit_01, if you like.

Editing Pad Sounds

1. Hit the lowest pad—we're going to edit SMPL1_01.
2. Locate and press the Pad Edit button.



3. Turn the Parameter knob until the display shows:

The pad whose sound you're editing — The sound being edited

```
C2  WAVE    SMPL1_01
PlayMode=OnceForward
```

The PlayMode parameter determines how the selected sound—in this case, SMPL1_01—will play the waves on which it's based. It can be set to:

- OnceForward—so that the wave will play through once and stop.
- OnceBackwrd—so that the wave will play backwards once and stop.
- LoopForward—so that the wave will play forwards over and over again.
- LoopFwd&Bwd—so that the wave will play once forward, then play backwards, then forward again, over and over.

4. Turn the Value knob clockwise to set PlayMode to LoopForward.
5. Turn the Parameter knob clockwise one tick so that the display shows the Start/Loop parameter. On this single page, you can dial in a coarse setting for the:
   - sample start—the location from which the wave will play when you strike a pad.
   - loop start—the location from which the wave will start playing again when it loops around if PlayMode is set to LoopForward or LoopFwd&Bwd.
   - loop end—the point at which the wave will return to loop start when PlayMode is set to LoopForward or LoopFwd&Bwd. When the wave is not set to loop, loop end functions as sample end, determining where the wave will stop playing.

   All of the above are expressed on this display as percentages of the entire wave.

Tip: Though you don't need them for this tutorial, sample start, loop start and loop end all have fine adjustments that you can use to find perfect loop settings when working on your music.

```
C2  WAVE    SMPL1_01
Start/Loop=00 00 99%
```

Sample Start   Loop start   Loop end

Since you sampled using the threshold trigger, the sample and loop start settings should already be set properly for what we're about to do.

6. Turn the Parameter knob so that the loop end field is underlined, which means it's selected for editing.
7. Press and hold down the lowest pad, and turn the Value knob to find a setting for loop end that causes the wave to return smoothly to its beginning after the four bass notes. The auto-zero cross feature offers you locations within the wave that are likely to result in satisfactory loops.

   You've just made your first ASR-X Pro loop.
8. Make note of your loop end setting—you'll need it again in a moment (write it down, if you've got paper handy).
9. Press the middle pad, and if necessary, turn the Parameter knob until the PlayMode parameter is displayed:

The pad whose sound you're editing   The sound being edited

```
F#2 WAVE    SMPL1_02
PlayMode=OnceForward
```

Notice how the display is now showing that we're working with a new pad, and a new sound: SMPL1_02.

10. Turn the Value knob to set PlayMode for this pad's sound to LoopForward as well.
11. Turn the Parameter knob so that the Start/Loop settings are displayed, and the loop end value is underlined.
12. Set loop end to the same setting you used with SMPL1_01.
13. Turn the Parameter knob counter-clockwise to underline the sample start field.
14. Set sample start to half the setting of loop end.
15. Hit and hold down the middle pad—the wave now starts playing halfway through, when the bass comes in, and then starts over again from the beginning. You've turned the beat around by setting sample start later than the loop start.
16. Hit and hold the lowest pad—it's still playing the whole wave.

    You can do this because each pad is playing its own sound, and we've been editing those sounds, not the wave itself. Let's do one more.
17. Press the highest pad and set its PlayMode parameter to OnceForward.
18. Turn the Parameter knob to display the Start/Loop parameter.
19. By adjusting the sample start and loop end settings, grab a single bass/bass drum hit.
20. When you're done, you can play all three pads to hear three different edits of your sampled waves.

    Don't forget: you've also got all of the other pads in DanceKit_01 to jam with. If you want, you can record your jam into the sequencer and resample the whole thing all over again. And on and on....

A Few Final Wave Editing Thoughts

The Pad buttons offer a great selection of tools for processing your waves, including bit reduction for that ratty sound, and a special warp volume scaling feature. Under the Pad Edit button, you'll find an amazing collection of parameters—including resonant filters—that will let you shape your waves into most anything you can imagine. The pads and all of the things you can do to the sounds they play are described in detail in Chapter 3 of the ASR-X Pro Reference Manual.

Saving Your Music to a Floppy Disk

You can save your work to any writable SCSI device or to the ASR-X Pro's built-in floppy drive, which can write and read files to and from standard 3.5" high-density or 3.5" double-density floppy disks. Since the ASR-X Pro uses a DOS-based format, its floppy disks can be read by any Macintosh or PC-compatible computer. ASR-X Pro sequences are saved to floppy as Standard MIDI Files that can be played by any sequencer or sequencing program that can accommodate SMFs. ASR-X Pro waves are saved to disk as AIF files so that they can be edited by most major sound-editing programs.

The easiest way to save your ASR-X Pro music is to save an ALL-SESSION file. ALL-SESSION files save all of the sounds and waves in RAM and all of your sequences—they even save your System/MIDI, Resampling Setup and sequencer Click settings, and be automatically loaded when you power up.

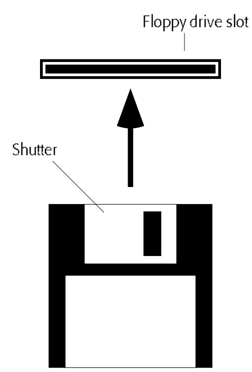
ASR-X Pro floppy and SCSI disk operations are described in detail in Chapter 7 of the ASR-X Pro Reference Manual.

Before you can save anything to floppy, the disk must be formatted. You can format the floppy on a computer or, if you're using a high-density disk, on the ASR-X Pro.

Note: In order to be formatted or written to, a floppy must not be write-protected. The sliding window on its lower edge should be in the closed position.

Formatting a Floppy

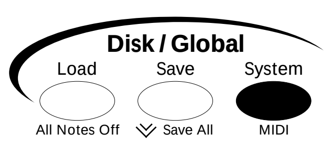
1. Insert a fresh 3/5" high-density floppy disk into the ASR-X Pro disk drive label-side up, as shown:



Position the floppy so that its shutter looks like this.

As you insert the disk into the drive's slot, stop pushing it in when it's seated in the drive mechanism—you can usually feel when this occurs.

2. Locate and press the System/MIDI button.



3. Turn the Parameter knob until the display shows:

```
System/MIDI
Access disk utils?
```

4. Press the Yes button.

The display shows:

```
Select device?
Floppy Disk
```

This display allows you to choose the storage device—your floppy drive or any attached SCSI device—with which you want to work.

5. Since we want to use the floppy drive in the tutorial, press the Yes button.

   The ASR-X Pro takes a few moments to load the floppy's directory.
6. If necessary, turn the Parameter knob until the display shows:

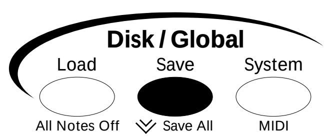
```
Disk Utilities:
Format floppy disk?
```

7. Press the Yes button.
8. The ASR-X Pro asks you to again if you want to format the disk, since doing so will permanently erase any files currently on the disk.
9. Press the Yes button if you're prepared to format the floppy.

   The formatting process takes a minute or so. When the ASR-X Pro is finished formatting the disk, the ASR-X Pro will briefly display "Format successful!" If there's a problem with the floppy, the ASR-X Pro will display a message telling you so.

Saving an ALL-SESSION File

1. Locate and press the Disk/Global Save button.



2. If necessary, turn the Parameter knob until the display shows:

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

The type of file to be saved

The name of the file to be saved

By turning the Parameter knob you can select the different types of file that can be saved to disk. For this tutorial, you want to save an ALL-SESSION file, as shown.

Tip: There's a faster way to jump to this Save ALL-SESSION display: simply double-click the Disk/Global Save button. We've used the longer method in Steps 1 and 2 to help you become familiar with the procedure for saving files of all types.

You can assign an eight-character name for your ALL-SESSION file. Each of the eight character positions can be selected—it will be underlined—and the desired character dialed in. Let's name this file "MYFIRST" to get a sense of how files are named.

Since the first "S" in "SESSION" is already underlined, it's selected for editing.

3. Turn the Value knob to dial in an "M" as the ALL-SESSION file name's first character.
4. Press the right-hand Select Track button to underline—and select—the second character position for editing.
5. Turn the Value knob to dial in a "Y."
6. Repeat Steps 4 and 5 until you've named the file.
7. When you're done, press the Yes button.

Saving an ALL-SESSION file can take a few minutes, since it's saving so many things. In our tutorial, it's saving the following separate files. It's saving:

- all of the sounds you created in RAM—DanceKit_01, SMPL1, SMPL1_01, SMPL1_02 and SMPL1_03.
- the waves you sampled.
- the current System/MIDI setup.
- a sequence bank containing your sequence.
- an ALL-SESSION file to keep track of all of the above files.

When the ALL-SESSION file has been saved, the ASR-X Pro returns to the display shown in Step 2, with "MYFIRST" now shown as the file name.

Before Proceeding...

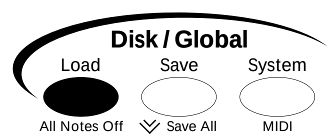
Since everything you've done has been saved to floppy, you can safely turn off your ASR-X Pro and restore your music whenever you power it back up. In fact:

1. Turn off your ASR-X Pro.
2. Turn it back on.

Loading an ALL-SESSION File

Since everything you do in the ASR-X Pro is stored in its RAM—and RAM is cleared when you power down—your ASR-X Pro no longer holds anything you've done in the tutorial. You can press the Track Sound button, select the *CUSTOM SoundFinder category and look for your sounds, or press the Transport Play button to play the sequence to verify this—there's nothing there. Let's load it all back in.

1. Locate and press the Disk/Global Load button.



After the ASR-X Pro loads the floppy's directory, the display shows:

There's our ALL-SESSION file.

2. Turn the parameter knob to see all of the different types of files that were saved when you saved the ALL-SESSION file.

Note: You won't see the sequence bank file, since it's a type of file that can only be loaded as part of an ALL-SESSION load.

---

3. Select the ALL-SESSION file.
4. Press the Yes button.

The ASR-X Pro loads everything you've done back into your ASR-X Pro. Start by playing the currently selected sequence, and then press the Track Sound button and check out the sounds in the *Custom SoundFinder category, as well as DanceKit_01—from the DRUM-KIT category.

The End

Congratulations! You completed the ASR-X Pro User's Guide tutorial. You're well on your way to many enjoyable sessions with this portable groovestation. Don't forget to check out the ASR-X Pro Reference Manual, where all the details live—the ASR-X Pro is a powerful little machine and the Reference Manual is the key to harnessing all the musical muscle now at your fingertips.

List of ROM Sounds

The following is a list of all of the sounds built into the ASR-X Pro.

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
ROM11:000	Vintg Bs 1
ROM11:001	Vintg Bs 2
ROM11:002	Vintg Bs 3
ROM11:003	Vintg Bs 4
ROM11:004	Vintg Bs 5
ROM11:005	Snappy Bass
ROM11:006	Bubbly Bass
ROM11:007	Vel Wow Bs
ROM11:008	BowWow Bass
ROM11:009	TB Tech Bs
ROM11:010	Sqr Whl Bs

ROM11:011	Bass Line
ROM11:012	Mono Boy
ROM11:013	PaddedEPno
ROM11:014	CheapoOrgan
ROM11:015	Slo Pad
ROM11:016	BandPassPad
ROM11:017	Snappy Pad
ROM11:018	Square Pair
ROM11:019	ThruThePass
ROM11:020	Slow HiPass
ROM11:021	Xtra Xtra
ROM11:022	Drippy
ROM11:023	Wheel Works
ROM11:024	Slumber
ROM11:025	Cyclon
ROM11:026	Wheel It In
ROM11:027	Planet X
ROM11:028	Odd-A-Sea
ROM11:029	Xpose'
ROM11:030	Top Secret
ROM11:031	Square Deal
ROM11:032	PS HarmLead
ROM11:033	Dual Pad
ROM11:034	Padded Cell
ROM11:035	Mellow Pad
ROM11:036	Brassy Saw
ROM11:037	Poly Pad
ROM11:038	Hol O'Lead
ROM11:039	PS Harm Sqr
ROM11:040	PS HarmMono
ROM11:041	Sigh-o-Nara
ROM11:042	WHL Swept
ROM11:043	Padly
ROM11:044	Sync2Wheel
ROM11:045	Chord Pulse
ROM11:046	Techno Sync
ROM11:047	Sweep Sync
ROM11:048	In Time...
ROM11:049	Auto Jam
ROM11:050	R. Peggio
ROM11:051	C Quencer
ROM11:052	Thing Pad
ROM11:053	Step Pad
ROM11:054	Tightwad
ROM11:055	Quasar
ROM11:056	Synced Fltr
ROM11:057	Trill It
ROM11:058	Pulsar
ROM11:059	1 FingerSeq
ROM11:060	Water Drops
ROM11:061	Pan Bells
ROM11:062	'D'-X-Flies
ROM11:063	Spacey S/H
ROM11:064	S/H Dance
ROM11:065	S/H Fltr 1
ROM11:066	NoiseGroove
ROM11:067	SickPercEFX
ROM11:068	Sci-Fi
ROM11:069	Hissy Fit
ROM11:070	Junk Box
ROM11:071	MonsterSnak
ROM11:072	Arcade

Insert Effect Parameters

The following pages list all of the ASR-X Pro's Insert effects and their parameters. To learn how to select and edit Insert effects, see Chapter 4 of the ASR-X Pro Reference Manual.

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 Pro 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	-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.
ER 1 Level	-99% to +99%	
ER 1 Level	-99% to +99%	
ER 1 Level	-99% to +99%	
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.

"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

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