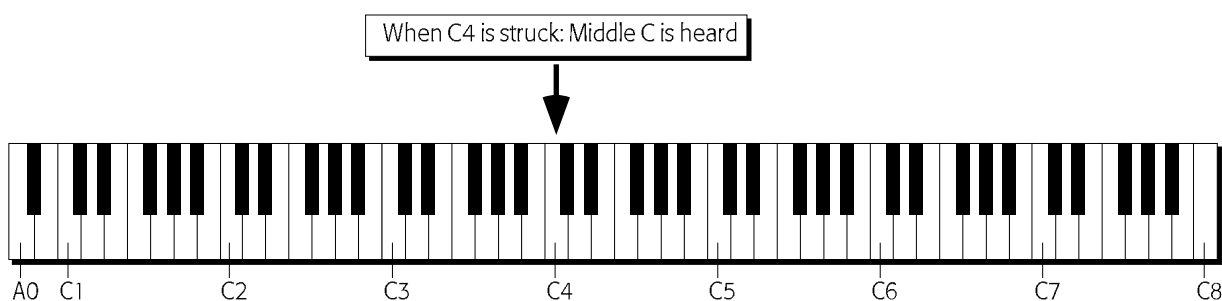


3 Pads

The Pads: Overview

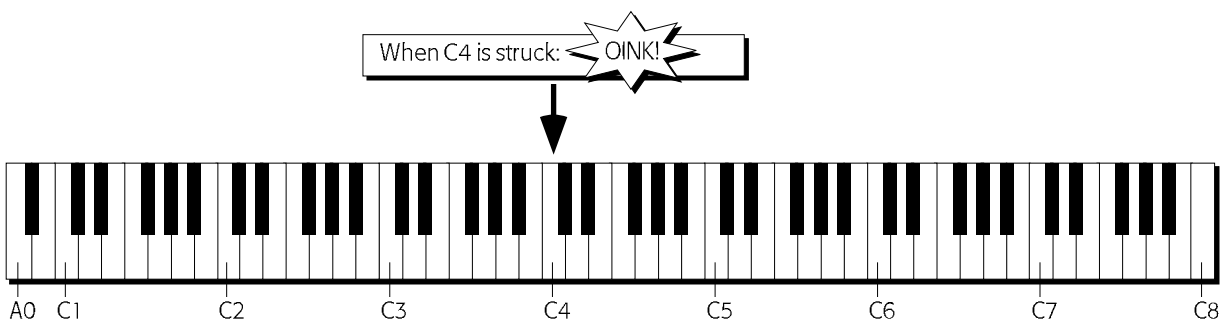
What are the Pads?

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



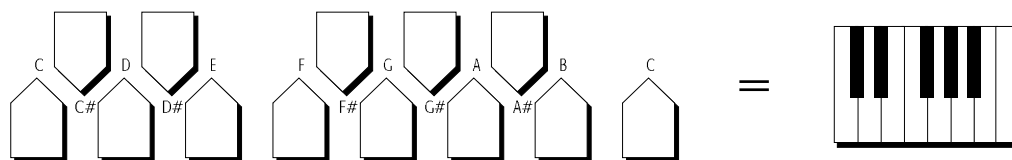
Each semitone is represented by a MIDI note name. The octaves—which begin at each C natural—are numbered, as shown above. The ASR-X Pro can address MIDI notes from A0 to C8.

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



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

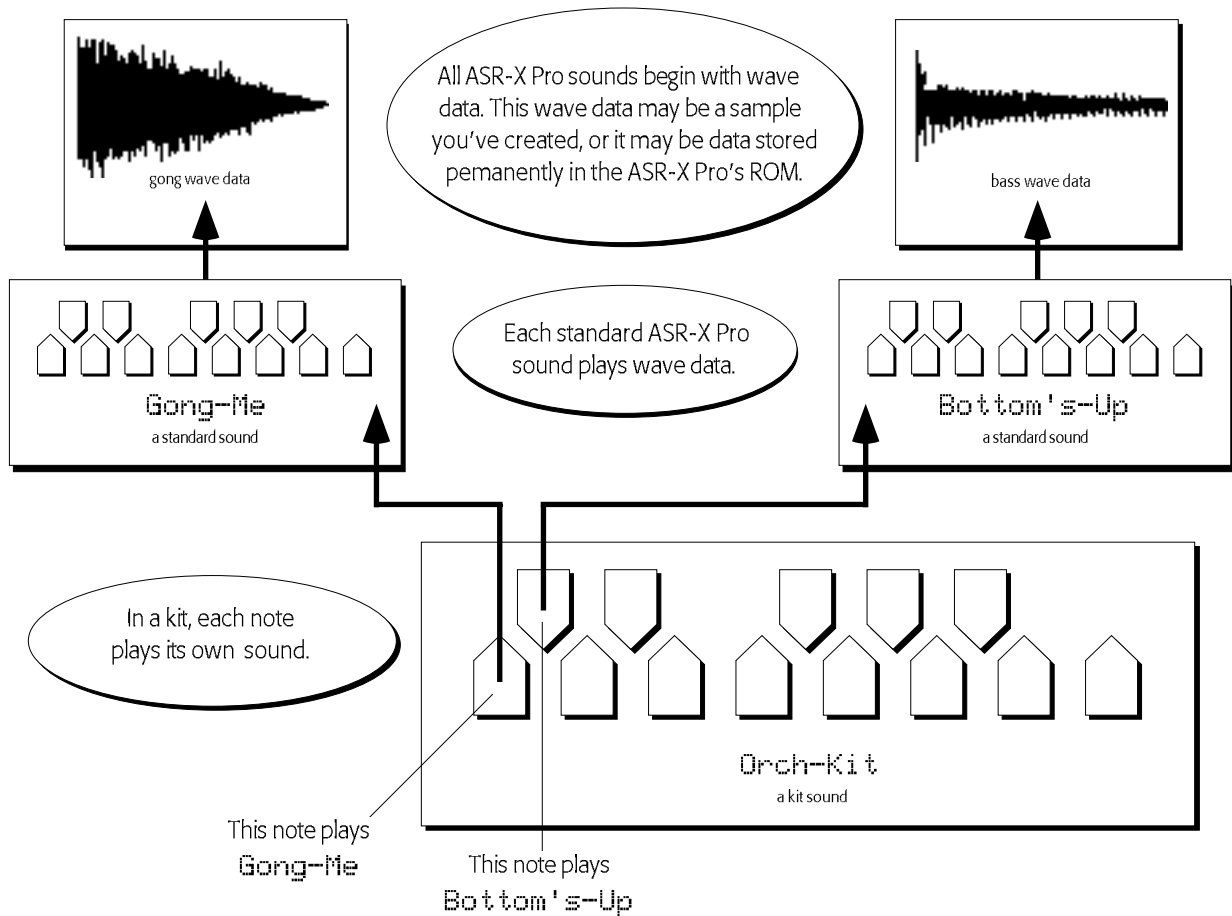
The 13 ASR-X Pro pads trigger 13 adjacent MIDI note numbers, the equivalent of 13 adjacent keys on a piano-style keyboard (unless the Kit Mapper, described later in this chapter, is on). You can use the pads to play single notes or chords.



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

What the Pads Play

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



Standard Sounds

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

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

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

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

Kit Sounds

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

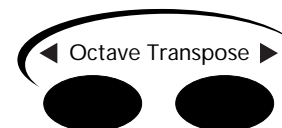
If you've assigned the same standard sound to more than one pad, they play the same sound. Since each pad has its own set of PAD parameters (described later in this chapter), you can program the pads to play

different variations of the same sound, perhaps setting them to play at different pitches. You can also program each pad in a kit to play a sound that's unrelated to what the other pads are playing—in this case each pad triggers something completely unique.

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

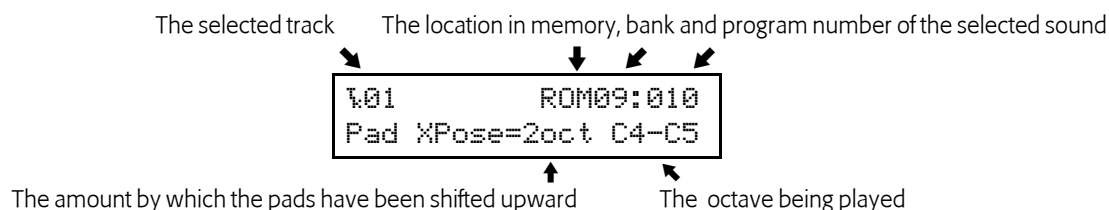
Octave Transpose Buttons

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



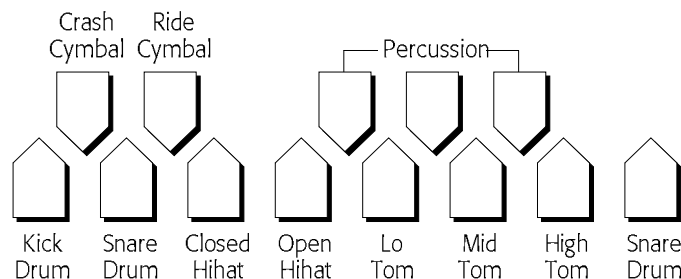
- repeatedly press the either Octave Transpose button to redirect the pads upward or downward.
- press either Octave Transpose button once, and turn the Value knob to select the desired octave.

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



The Kit Mapper

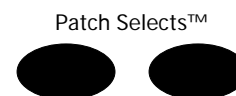
Typically, the pads in the ASR-X Pro play 13 adjacent notes. When you're using a kit sound that conforms to the ENSONIQ drum or percussion maps (described in Chapter 9), these 13 notes may be variations of the same kit component. The Kit Mapper re-assigns the pitches played by the pads so that the important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



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

Patch Select Buttons

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



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

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

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

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

Patch Selects and MIDI

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

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

Programming the Pads

Overview

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

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

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

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

Note: If you attempt to perform wave operations by pressing the Pad Process button when the selected sound is not playing an ASR-X Pro-created wave the display will show the “Synthesize Stomper sound?” prompt described later in this chapter.

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

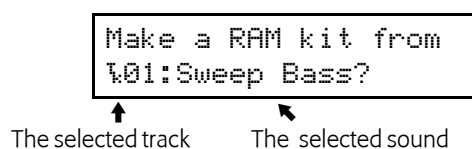
In order to program a sound's pads, two conditions must be met:

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

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

To Prepare the Selected Track's Sound for Pad Editing

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



When you press the “Yes” button in response to this question, the ASR-X Pro creates a copy of the selected sound as a kit in RAM, and assigns it to the selected track. The newly created kit will add an

underscore and a two-digit number to the end of the sound's original name—abbreviating the original name if necessary— to show that it's based on the original sound. The new kit can be found in the USER-SND and DRUM-KIT SoundFinder categories.

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

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

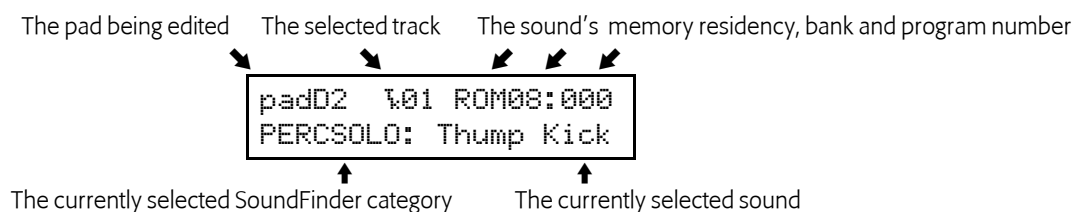
Selecting a Pad for Editing

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

Choosing a Pad's Sound

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

The pad sound-selection resembles the track sound-selection display:



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

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

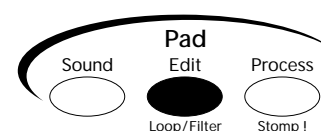
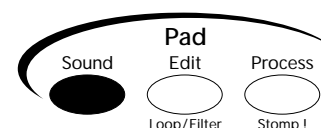
Tip: You can quickly erase a RAM sound from a pad by holding down the Pad Sound button and, while still holding it, pressing the No button. The sound Silence will be assigned to the pad.

Overview of the Pad Edit Parameters

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

To simplify navigation, the Pad Edit parameters are divided into 12 sub-groups.

The PAD parameters are always available, regardless of the type of sound being played by the selected pad. They allow you to determine the manner in which the pad will play its sound, and are described



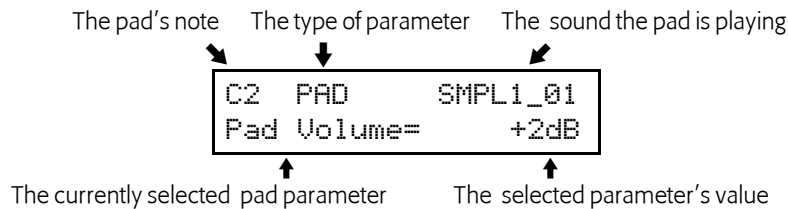
below in “Determining a Pad’s Behavior.” The PAD parameter settings become part of the selected track’s RAM kit.

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

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

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

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



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

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

Determining a Pad’s Behavior

PAD Parameters

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

Pad Volume

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

Pad Pan

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

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

FX Bus

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

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

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

Tuning Shift

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

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

Editing a Pad's Sound

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

The ASR-X Pro Modulators

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

Off	for no modulation
Full Amt	The maximum amount of modulation is applied to the modulation destination
LFO	the selected wave's LFO
Stepped	a significant amount of random noise modulation at a rate determined by the NoiseSource Rate parameter (see later in this section)
Smoothed	a subtle amount of random noise modulation at a rate determined by the NoiseSource Rate parameter (see later in this section)

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

WAVE Parameters

The Playback of Waves

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

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

A Couple of WAVE Ideas

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

PTCH Parameters

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

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

Env1PitchModAmt	-127 to +127	Env1PitchModAmt provides a special routing that endows Envelope 1 with unique capabilities in the modulation of the sound's pitch. When applied to the sound's pitch via the Env1PitchModAmt parameter, Envelope 1 automatically sustains at the pre-enveloping pitch, regardless of its Sustain Level (4) setting. Instead, its Sustain Level (4) setting serves to determine which Envelope 1 level values will cause the pitch to rise above the un-enveloped pitch and which level values will drive it below. Envelope 1 level values equal to the Sustain Level (4) value will cause the sound to play at the un-enveloped pitch. Higher level values will shift the pitch upward, and lower values will shift the pitch downward. This feature allows for the creation of bi-directional pitch envelope shapes, while conveniently ensuring that the pad's sound will always sustain at the un-enveloped pitch.
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ENV1 Parameters

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

Parameter	Range	Description
Envelope Mode	Normal, Finish, Repeat	Envelope 1 may function in one of three ways: <ul style="list-style-type: none"> • Normal—Envelope 1 plays through normally. When the key is released, the envelope takes the Release Time (5) to go from the current level down to zero. • Finish—Envelope 1 finishes playing through all its stages, ignoring the key-up event. The envelope spends no time at the Sustain Level (4) stage. When the Decay Time (4) interval is finished, instead of stopping at the Sustain Level (4) stage, the envelope immediately goes into the Release Time (5) stage. This is good for percussive-type sounds where you want the envelope to be the same for every note, no matter how long the key is held down. • Repeat—At the end of the Ramp Time (3) stage, instead of sustaining, Envelope 1 goes immediately back to the beginning and repeats, starting with the Attack Time (1) stage. When the key is released, the envelope stops repeating and moves into the release stage, taking the Release Time (5) interval to go from the current level down to zero. This type of envelope can be used to create complex LFO-type effects.
Attack Time (1)	0 to 99	Determines the time it takes for the envelope's level to travel from zero (when a note-on is received) to Attack Level (1). The higher the value, the longer the time.
Attack Level (1)	0 to 127	Determines the level the envelope will reach at the end of the time defined by Attack Time (1).
Ramp Time (2)	0 to 99	Determines the time it takes the envelope to go from Attack Level (1) to Ramp Level (2).
Ramp Level (2)	0 to 127	Determines the level the envelope will reach at the end of Ramp Time (2).
Ramp Time (3)	0 to 99	Determines the time it takes the envelope to go from Ramp Level (2) to Ramp Level (3).
Ramp Level (3)	0 to 127	Determines the level the envelope will reach at the end of Ramp Time (3).
Decay Time (4)	0 to 99	Determines the time it takes the envelope to go from Ramp Level (3) to the Sustain Level (4) stage. At the end of Decay Time (4,) the envelope will remain at Sustain Level (4) until the key is released.

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

FILT Parameters

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

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

FLT1 and FILT2 Parameters

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

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

ENV2 Parameters

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

AMP Parameters

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

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

Alt Bus	Default, LightReverb, MediumReverb, WetReverb, Dry	Determines the effect bus to which the sound will be routed when it's selected for a track if the System/MIDI AutoSelect FXBus parameter is set to "On."
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ENV3 Parameters

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

MOD Parameters

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

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

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

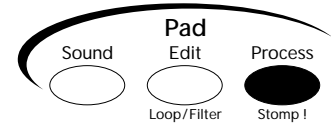
MISC Parameters

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

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

Processing a Sound's Wave

When a pad contains a sound based on an ASR-X Pro-created wavesample, or “wave,” the Pad Process button provides access to a number of tools for processing the sound’s wave. In addition, whether or not the pad contains a sound based on a wave, the Pad Process button provides access to Stomper, a unique synthesis algorithm that lets you to create your own new sounds.



Since these tools modify and create wave data, when you perform one of the pad processes:

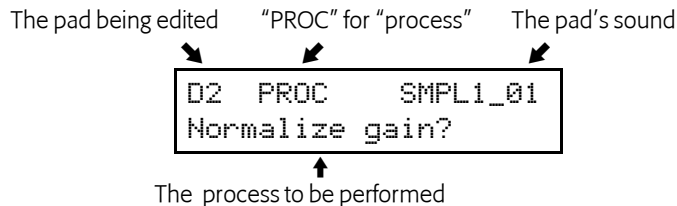
- the ASR-X Pro makes a copy of the wave
- it performs the selected operation
- it places the processed copy on the Scratch Pad. You can then play the scratch pad to audition the results of the process you’ve performed.

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

Note: The Stomper algorithm—unlike other pad processes—is available at all times, regardless of the nature of the currently selected pad sound. It’s described at the end of this chapter.

The Pad Process Display

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



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

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

The Pad Processes

Normalize gain?

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

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

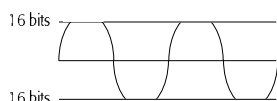
Scale loudness?

The ASR-X Pro lets you lower or raise the overall volume of a wave by percentage you set through the use of its scaling facility. When you press the Yes button in response to “Scale loudness?” two settings

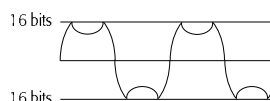
become available that allow you to set the manner in which the wave will be scaled. To view these two settings, turn the Parameter knob; to adjust them, turn the Value knob.

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

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



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



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

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

Reduce sample bits?

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

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



Turn the Value knob to change this value

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

Invert sample data?

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

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



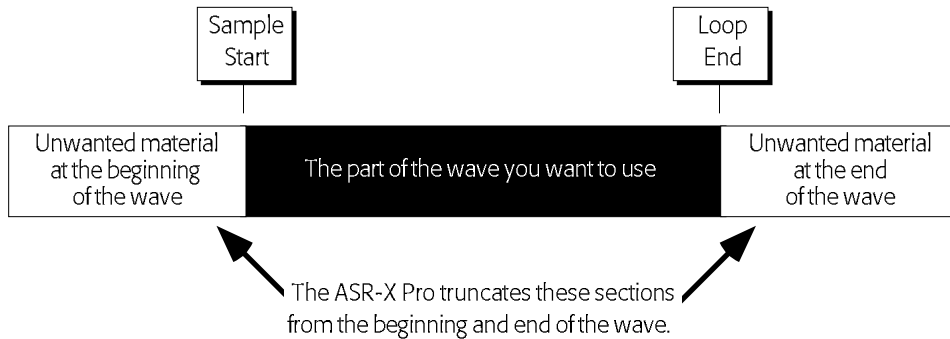
...and ends up looking like this:



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

Truncate length?

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



Copy sound?

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

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

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

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

The octave that the pads are currently playing

```

C2...0oct..C3 CopyTo
X          Pads?
  
```

The pad from which you're copying

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

Scale time?

The "Scale time?" command alters the duration of a wave without altering its pitch, allowing you to stretch or shrink a wave to fit a particular tempo—such as when you want to re-size rhythms for use as loops. When you press the "Yes" button in response to "Scale time?" the following parameters can be accessed by turning the Parameter knob:

- Amount—This parameter sets the percentage by which the wave's duration will be made longer or shorter. A value of 100% will leave the wave at its current length; values lower than 100% will

shrink the duration of the wave, while values higher than 100% will increase it. Try different values for this parameter to establish the percentage of time scaling required for your situation.

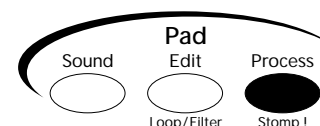
- **Quality**—sets the fidelity of the time-scaled wave. The High(slow) value produces cleaner-sounding waves, but will take a greater amount of time to process. When experimenting with the Amount parameter to determine its correct value for your timing needs, set the Quality parameter to Low(fast) to save time. Once you've settled on an Amount value, set Quality to the desired setting and re-scale the wave.

Stomp!

Stomper is a non-real-time algorithm created by Håkan “Zap” Andersson that allows you to construct your own vintage-synth-style sounds using the ASR-X Pro's processor. You create a Stomper sound by setting parameters that describe the sound's characteristics, and then hit the Yes button to instruct Stomper to build the sound and place it in the Scratch Pad—from there it can be assigned to pads a RAM kit sound in the same manner as any other wave. Stomper allows you to select the sound's waveform content, its filtering—including resonant filtering—and volume, or amplitude, characteristics. Since Stomper creates your sound right in the ASR-X Pro, the resulting 16-bit sound is terrific.

To learn more about Stomper, visit its Web site at <http://www.Master-Zap.com.stomper>

Stomper is accessed by pressing the Pad Process button. When the currently selected pad uses a sound that's not based on a loaded wave, “Synthesize Stomper sound?” is displayed. When the sound on the pad is based on a wave, scroll all the way clockwise after pressing the Pad Process button to reveal “Synthesize sound?” In either case, the displayed question provides access to Stomper's parameters. As you move through the Stomper parameters, the Yes and No LEDs will flash to indicate that you can build your sound at any time by pressing the Yes button, or leave Stomper by pressing the No/Exit button. As the sound is being created, a progress indicator will be displayed.



Sound Type

Stomper provides a set of presets that can be used as is, or as a starting point for your own sounds. Turn the Value knob to select any of the following presets:

KICK1 KICK2 SNARE TOM CRASH HAT

When you change the value of any parameters, an additional USER preset is created.

Oscillator

Each Stomper sound can contain up to four active oscillators, each of which has its own set of parameters and can be configured to function as a waveform oscillator or as a low-pass resonant filter. Before setting up an oscillator, you must first select it by turning the Value knob when the Oscillator # display is visible.

Mode

The Mode display allows you to set the currently selected oscillator as you wish. An oscillator can be set to Off, Oscillator or Filter. Turn the Value knob to the desired setting.

Tip: When an oscillator is set to Filter mode, it filters all lower-numbered oscillators.

Note: The ASR-X Pro presents only those parameters relevant to the selected oscillator's mode. As a result, the remaining Stomper displays you'll see depends on the selected oscillator's mode.

Oscillator Mode Stomper Parameters

The following Stomper parameters are available when the selected oscillator is set to oscillator mode.

<i>Parameter:</i>	<i>What it does:</i>
Waveform	Selects the waveform to be used by the oscillator. Choices are: Sine, Saw, Square, Triangle
Noise Factor	Controls the amount of random frequency deviation applied to the oscillator from 0.00 to 1.00 in 1/100ths steps.
Noise Rate	Sets the rate of the random frequency deviation. A value of 0 is off; a value of 1 means that noise will be applied every sample; 2 would be every second sample, and so on.
Start Time	Sets the oscillator's start time in the final sound in milliseconds. A typical setting would be 0; increasing the value delays the sounding of the oscillator.
End Time	Determines the duration of the oscillator by setting its end point, in milliseconds.
Start Freq	Sets the starting frequency, or pitch, of the oscillator, from 0 to 20,000Hz in steps of 10.
End Freq	Sets the final frequency, or pitch, of the oscillator, from 0 to 20,000Hz in steps of 10.
FreqCurveShape	Sets the shape of the curve as the oscillator travels from its start frequency to its end frequency—this can be set from 0.01 to 10.00 in .01 steps and from 10 to 100 in steps of 1.
Start Amp Scale	Sets the starting amplitude, or volume, of the oscillator, from 0% to 100%.
End Amp Scale	Sets the ending amplitude, or volume, of the oscillator, from 0% to 100%.
AmpCurveShape	Sets the shape of the curve as the oscillator travels from its start amplitude to its end amplitude—this can be set from 0.01 to 10.00 in .01 steps and from 10 to 100 in steps of 1.
Tone CurveShape	Sets the amount of distortion added to the shape of the oscillator's waveform, from 0.01 to 10.00 in .01 steps and from 10 to 100 in steps of 1.

Filter Mode Stomper Parameters

These Stomper parameters are available when the selected oscillator is set to low-pass filter mode.

<i>Parameter:</i>	<i>What it does:</i>
Start Cutoff	Sets the starting cutoff frequency, determining the point above which frequencies will be attenuated at the beginning of the sound. Parameter can be set from 0Hz to 20,000Hz.
End Cutoff	Sets the ending cutoff frequency, determining the point above which frequencies will be attenuated at the end of the sound. Parameter can be set from 0Hz to 20,000Hz.
Start Resonance	Sets the amount of resonance (Q) at the start of the sound, from 0.00 to 0.99 in steps of .1.
End Resonance	Sets the amount of resonance (Q) at the end of the sound, from 0.00 to 0.99 in steps of .1.