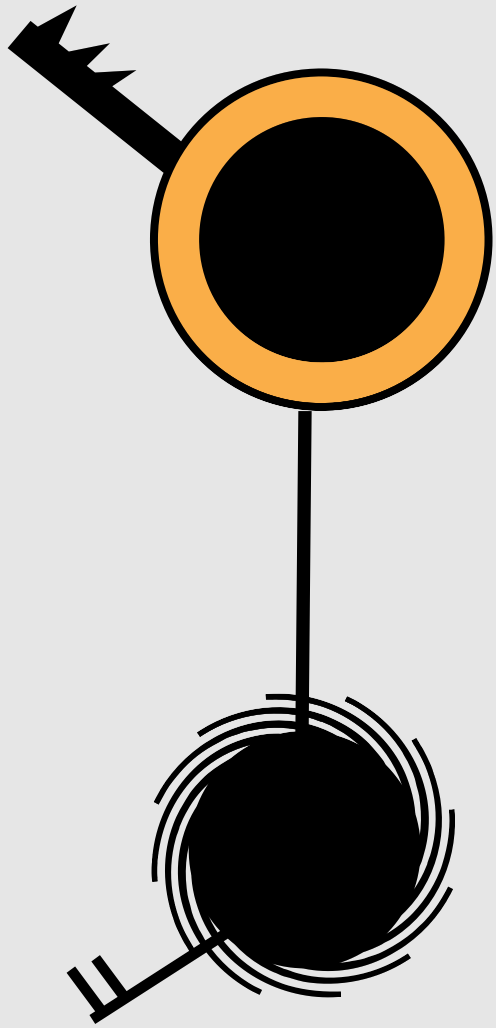
7 Digital Processing

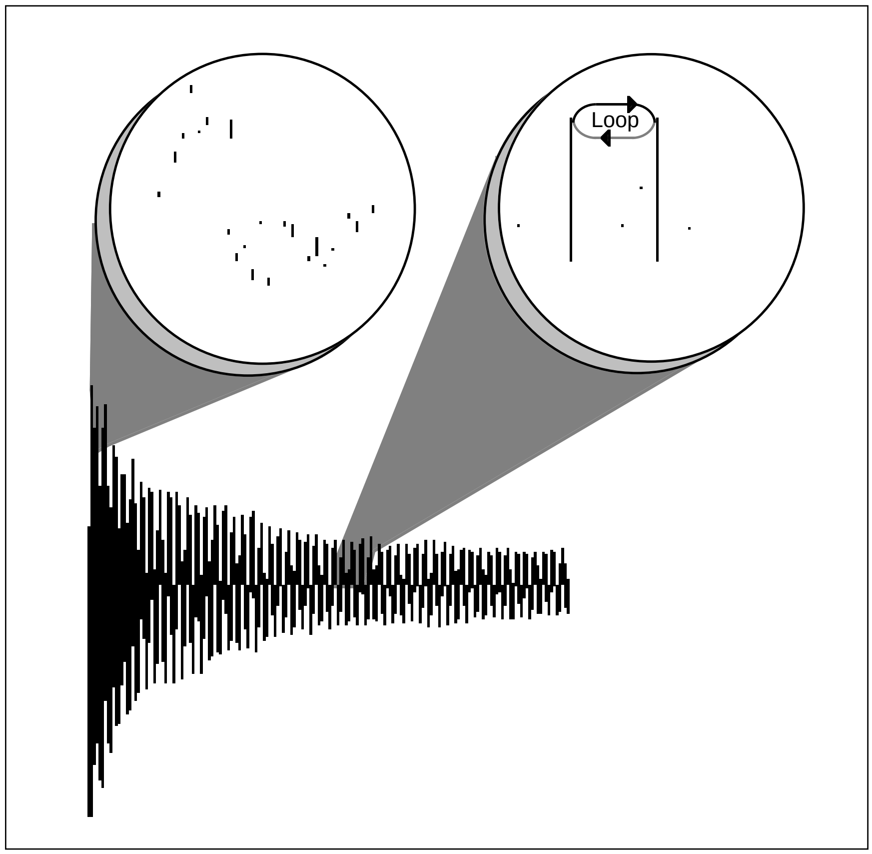


Background: About Looping

Sampling is the process of storing sounds in digital memory. Since each individual sample (not the complete sample, but each element of the sample) requires one memory slot, memory requirements increase if you sample long sounds or use high sampling rates.

How Looping Works

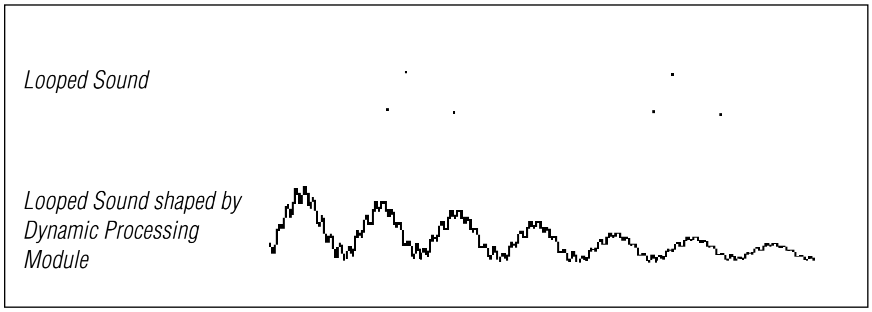
Looping is a technique that can extend a sound's length without using lots of memory. It is based on the fact that many sounds start off with a complex attack transient, then settle down to a comparatively steady sound. Listen carefully to a plucked guitar string. The first part of the sound consists of a complex mixture of pick noise and several harmonics; after a while, the string decays down to a pretty steady repetitive waveform.



Since the latter part of the waveform is repetitive, there is no need to waste memory sampling several seconds of it. Instead, you can mark off a loop of the repetitive section, and instruct the ESI to play that looped section for as long as the key is held down. After playing to the end of the loop, ESI jumps back to the beginning of the loop and plays through the loop again. This process repeats until you release the key playing back the sample.

Creating Attack & Decay Characteristics for the Looped Portion

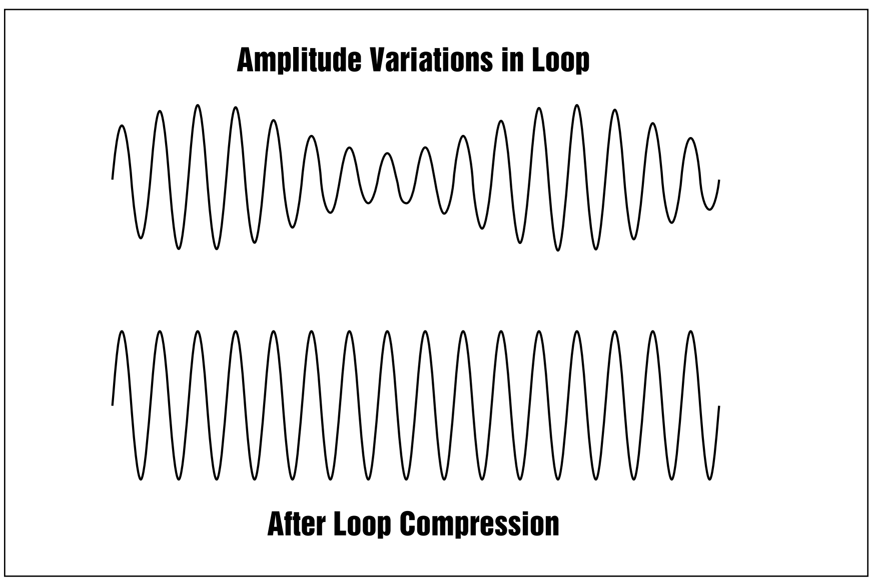
One potential problem is that the loop repeats at the same level. This is usually acceptable for sustaining instruments (flute, organ, brass, etc.), but is unacceptable for plucked or struck sounds, which decay over time. Fortunately, the Dynamic Processing module provides a means to shape the attack, sustain, and decay characteristics of a sample. You can create a decay during the looped portion, and simulate pluck or struck sounds.



Artificial Decay can be applied to a looped sound.

Loop Compression

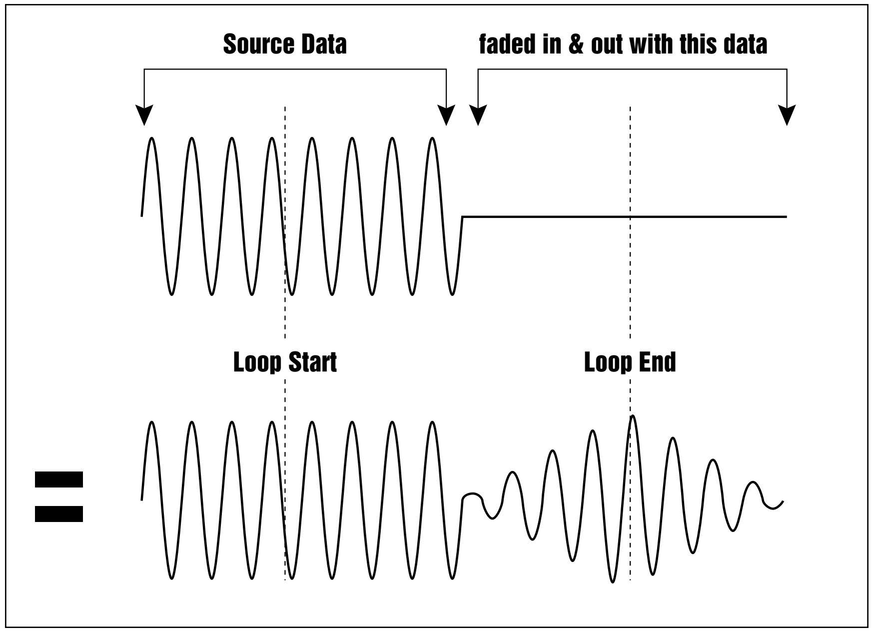
The looped section of the sample can be compressed, which "evens out" any changes in amplitude. Changes in amplitude can cause "breathing" effects, thumping or clicks.



Loop Compression smooths out amplitude variations during the loop period.

Crossfade Looping

The ESI fades between the beginning and end of the loop so that as the end fades out, the beginning fades in. This virtually eliminates the clicks and pops that can occur with other types of looping.



Crossfade Looping takes sound data from around the loop start point and fades it into the sound data around the loop end point so that the data at those loop points is identical. In the example above, a sine wave is crossfaded with silence to illustrate the process.

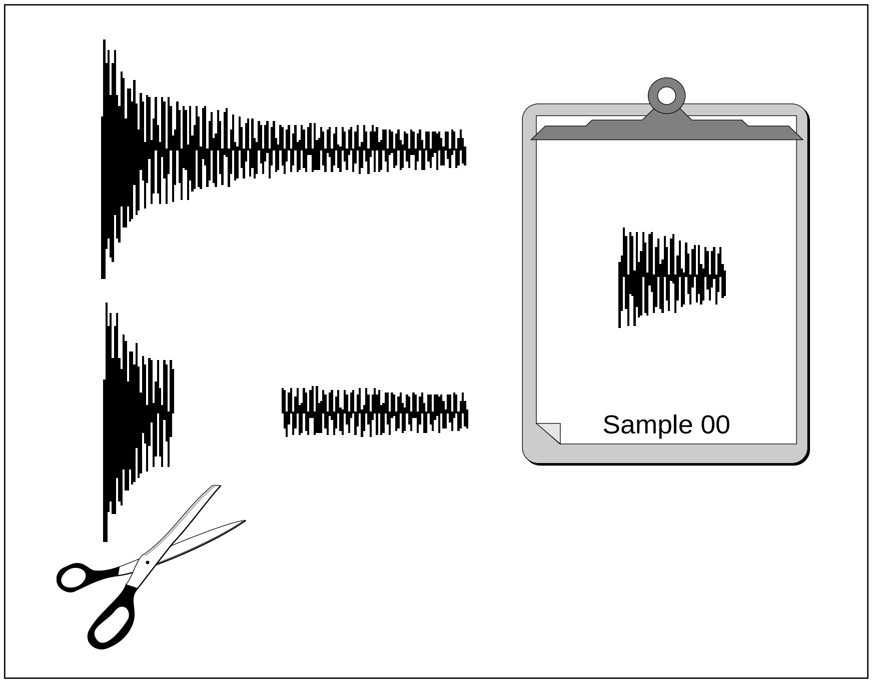
✶ **Tip:** *IIf you notice a timbral shift during the crossfade, try increasing the loop length so that the timbral shift is spread out over more time.*

Crossfade looping is very effective with complex ensemble sounds which can be virtually impossible to loop otherwise. In most cases, even the most complex sounds can be looped without any clicks or pops. Bear in mind, however that crossfade looping actually changes the sample data. You may want to keep a backup of your raw sample data in case the loop is not satisfactory.

Background: Cut, Copy, Paste & Undo

Cut, Copy & Paste let you cut out, or make a copy of a segment of a sample and then paste it back in the same sample or in another sample. Imagine a printout of a sample. Now imagine that you have a pair of scissors capable of cutting a piece out of that sample (or a copy of that piece if you don't want to affect the original), as well as a clipboard to neatly hold the cut or copied piece. Now imagine that you can either perfectly insert the clipboard contents at any given point in any sample, or mix the clipboard contents in with any sample, starting at any designated point. The ESI can do all this, and all electronically. The ESI also provides Auto Correlation and optional Crossfade functions that result in seamless transitions between the cut, copied, and pasted parts.

✶ **Tip:** *The clipboard contents are designated as Sample 00.*



Clipboard Data: *The clipboard will retain data until replaced by other data to be copied, cut, or backed up. This occurs with several ESI operations where you want to be able to undo an action that doesn't work out as anticipated. Since clipboard data stays intact when you call up another sample, data can be cut or copied from one sample and pasted to another.*

Undo and Redo

! Note *The Undo function only works when there is a hard disk connected to the system.*

During several ESI digital processing operations, such as cut, paste, and sample rate conversion, the ESI automatically backs up the sample being processed and stores this backup in the digital clipboard on the system drive. If you do not like the results of the processing, you can call up the Undo function, and restore the original sample. (Ah, if only life itself were so simple.) Best of all, the processed sample now moves onto the system drive. If you decide you like the processed version better after all, you can actually undo the Undo (Redo).

Insufficient System Drive Memory

If the system runs out of hard disk storage space the display informs you that there is not enough memory to back up a sample. You then have two choices: either free up some additional memory by erasing banks on the system drive, or call up the Undo function (Digital Processing, 9. Undo) and disable the backup process. Of course, if backup is disabled, you cannot undo an operation.

Typical Applications

Typical cut/copy/paste applications would be to splice the beginning of one sample to the end of another, or to mix two samples together to conserve memory. (Do this by copying an entire sample, then pasting it at the beginning of the second sample using the mix option.) You can splice an attack transient on to a synthesizer waveform loop to produce realistic sounds that take up virtually no memory, or take a pop or click out of a sample. Another application is flanging and chorusing. Paste (mix) a sample to itself, offset from the beginning by a few hundred samples or so, to thicken up the sound.

We suggest you practice cut, copy, and paste techniques on a spoken phrase. Samples of political speeches can be particularly amusing when subjected to cut and paste operations.

Background: Auto Correlation

Auto Correlation

First of all, just what does auto correlation mean? Auto correlation simply means to automatically compare and tweak into a complementary or parallel relationship. The computer analyzes the signal around the loop points you have specified and then moves the end point of the loop until it finds a section of the wave that closely matches the section around the start point. Auto correlation may be used again and again with the computer moving the analysis window slightly each time to try to zero in on the optimum loop.

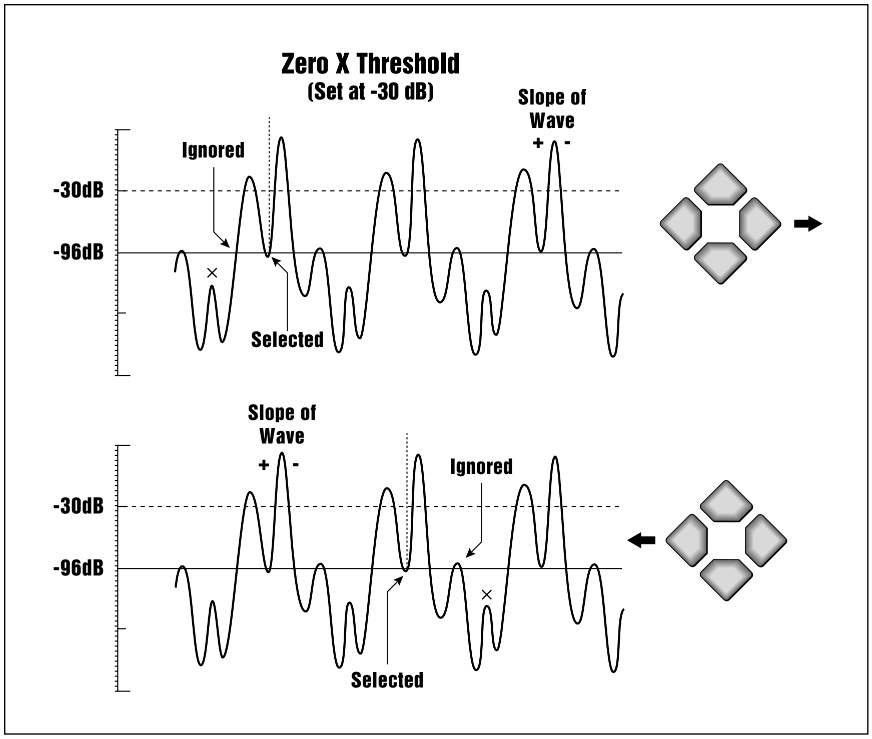
Background: Zero Crossing

Zero Crossing

The term zero crossing refers to the point at which the positive slope of a waveform passes through zero. In many digital processing applications, such as splicing and looping, it is useful to locate zero crossings in order to make glitch-free joins and loops. On some signals, however, a simple zero crossing may not be effective because the signal contains excessive noise or low-level, high-frequency harmonics. In these cases, every few samples may cross through zero. By setting a zero crossing threshold, we can ignore low-level zero crossings and wait for the signal to reach a certain level before choosing the next zero-crossing. Zero crossing threshold sets a level that a signal must exceed before the next zero crossing with a positive slope is selected. The selected zero crossing threshold is used in the Auto-Truncate function in the Sample Management module, Sample Setup (5), or whenever you manually select a zero crossing using the cursor keys.

The diagram on the following page illustrates zero crossing in action. The small x marks the initial position. To move forward through the sound to the next zero crossing, press the right cursor button, as shown in the upper diagram. The ESI will find the first zero crossing on the positive slope after the signal has crossed the designated threshold.

To move backward through the sound to the next zero crossing, as shown in the lower diagram, press the left cursor button. Again, the ESI finds the first zero crossing on the positive slope after the signal has crossed the designated threshold.

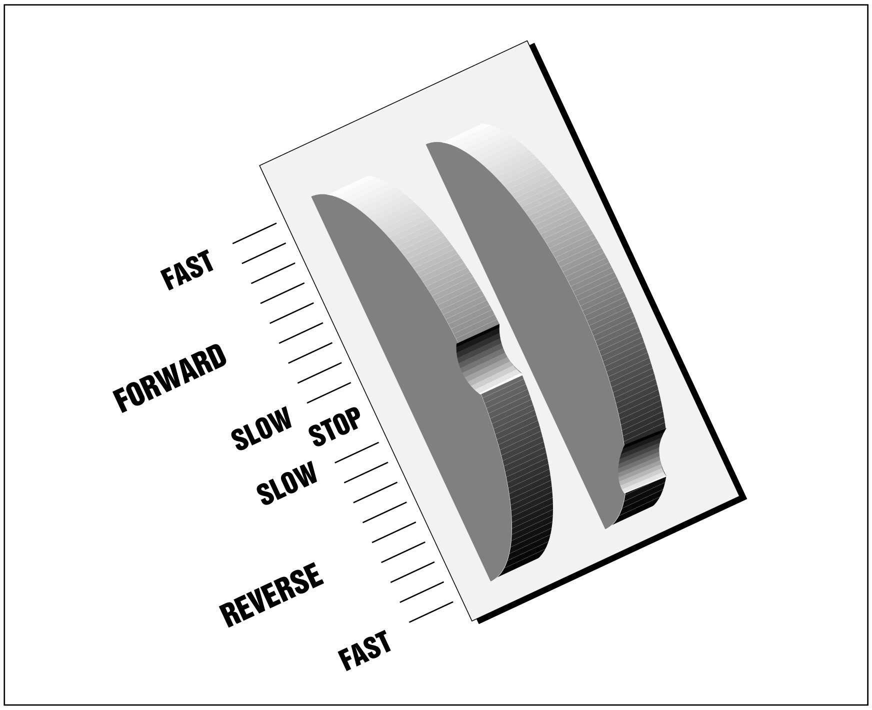


Adjust the zero crossing threshold according to the type of wave you are processing. A setting of -96 dB is most sensitive and can be used for finding the start point of a sound. Settings closer to -30 dB are less sensitive and suited for finding zero-crossings in complex waves.

Background:
Scrub Wheel

The Scrub Wheel is an ESI function that allows you to use the pitch wheel in many Digital Processing functions to quickly move through a sound, similar in concept to rocking the reels of a reel-to-reel tape recorder. The scrub wheel makes it easy to locate a particular section of a sound because you are able to hear the sound as you move through it.

To use the scrub wheel, move the pitch wheel of your MIDI keyboard while in a Digital Processing function such as Truncation or Looping. If the pitch wheel is moved slightly forward, the sound will play slowly through its length. If the wheel is pushed forward all the way, the sound will play through faster. If the wheel is moved backwards (towards you) the sound will play backwards through the sound. The scrub wheel operates like the accelerator on your car: the harder you press it, the faster it goes.



ESI SCRUB WHEEL. Move the wheel slightly to advance slowly through the sound. Move the wheel more to advance quickly through the sound.

0. Select Sample

This function lets you choose the sample you want to process. The resulting current sample remains as designated until you select a different sample, change presets, or load another bank. While in the Digital Processing module, the current sample is placed over the full keyboard range, and all other samples are muted.

1. **Activate Digital Processing module.**
2. **Select Submodule Select Sample (0).**
3. **Select a sample to be processed.** As you scroll through non-empty samples, the display shows the sample number, name, sampling rate, sample length, how many presets use the sample and whether or not the sample is stereo, left or right. As you scroll, each sample is playable on the keyboard over its entire available range.

```
   SELECT SAMPLE
001 Piano C6
Stereo      44100 Hz
5 Presets   1.6 secs
```

4. **Press ENTER to exit the submodule.** The ESI will return to the Module Identifier.

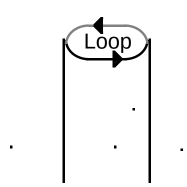
1. Setup

This submodule lets you turn looping on and off, loop in release on and off and sets various parameters concerning other Digital Processing functions. For more information, see Background: About Looping at the beginning of this section.

! Caution: *Samples imported from the EIII having forwards/backwards loops will play back on the ESI, however the loop data will be permanently modified to contain the forwards/backwards sound data. Changing the loop points on imported forwards/backwards loops could have unpredictable results and is not recommended.*

1. **Activate Digital Processing module.**
2. **Select Setup (1).**
3. **Select Loop Enable On/Off on line two.**

```
       SETUP       ⇢
Loop enable:      on
Loop in Release:  on
```



Your choices are:

- **Off:** The loop is disabled and the sample plays as recorded.
- **On:** The sample plays normally until it reaches the end of the loop. It then jumps back to the start point and replays through the loop. Upon reaching the loop end, it again re-plays the loop. This looping process continues for as long as the key is pressed.

4. **Select whether Loop in Release on line three is on or off.** Here are your choices:

- **Off:** Lifting your finger off a key initiates the release phase of the VCA envelope. If Loop in Release is off, the loop will not continue during the release phase. The portion of the sample after the loop will be played during release. The signal may cut off abruptly after you remove your fingers from the keys if the sample has been truncated after the loop.
- **On:** With Loop in Release on, the loop will keep playing—even after you lift your finger off the key—for the duration of the VCA release.

5. **Select page two by pressing the right cursor button.** The second page displays the following parameters:

```
←      SETUP
Beep:            off
Zero X Thresh: -54dB
```

- **Beep:** when on, the ESI gives an audible indication when time intensive digital processing operations are finished.
- **Zero X Threshold:** Use the Data Entry Control or INC/DEC buttons to select the zero crossing threshold. A setting of -96 dB will be the most sensitive; a setting of -30 dB will be the least sensitive. The selected zero crossing threshold is used in the auto-truncate function or whenever you manually select a zero crossing using the left/right cursor keys. If auto truncate does not seem to be working well, try adjusting the zero crossing threshold.

6. **Press ENTER to exit the submodule.** The ESI will return to the Module Identifier.

2. Loop

If you have a difficult time finding good loop points, the ESI can assist you using the Auto Correlation function. If Auto Correlation doesn't produce acceptable results, the beginning and end of a loop can be crossfaded to help mask loop discontinuities. For more information, see Background: About Looping at the beginning of this section.

1. **Activate Digital Processing module.**
2. **Select the Loop submodule (2).**

```
LOOP   secs  samples
Start: 1.96    43280
End:   3.54    78162
Size:  1.58    34882
```

✶ **Tip:** *The smallest possible loop start point is four samples into the sound.*

3. **Select the values for the start point on line two and the loop size on line four that give the best looping effect, then press ENTER.** The left and right cursor will change the start point and/or size so that the loop automatically falls on positive zero crossing points in the waveform.

The sample size equals the difference in samples between the start and end points. These settings interact in the following ways:

! **Caution:** *You cannot Undo the loop settings.*

- Changing the size automatically changes the end point so that the difference in samples between the start and end points remains equal to the size.
- Changing the start point changes the end point to maintain a constant size.
- Changing the end point changes the size, and the start point remains constant.

! **Caution:** *If a sample's loop disappears as soon as you have left the Digital Processing module, check to see if the loop function has been disabled in Dynamic Processing-1, Setup.*

4. **The display asks if you want to Auto Correlate.** Press YES to Auto Correlate, or NO to proceed directly to step six, Compress Loop. Auto Correlation uses artificial intelligence techniques to choose optimum loop points. Without Auto Correlation, unless you are proficient with sampling techniques, loops will usually have discontinuities between the splice points that can produce annoying ticks, pops and other glitches.

```
LOOP   secs  samples
Start: 1.96    43280
End:   3.54    78162
Auto Correlate? Y/N
```

✶ **Tip:** *If a sample ends in a loop, playing it backwards will repeat the loop. The sample will not play back prior to the loop start.*

With Auto Correlation, the ESI looks for loop points, near the ones you chose, that can be spliced together with minimum discontinuity. Occasionally it will be impossible for either you or the ESI to locate a perfect splice point which means the sound cannot be looped, but in most cases you'll find that Auto Correlation, combined with practice and experimentation, can produce very smooth loops.

5. **If desired, repeat steps three and four, until the best possible loop results.** When you're finished looping, press NO in step four and carry on.

```
LOOP   secs  samples
Start: 1.96    43280
End:   3.54    78162
 Compress Loop? Y/N
```

6. **The display asks if you want to Compress the Loop.** Press Yes to Compress, or No to proceed directly to step seven, Crossfade Loop. Compressing just the loop portion of the sound is yet another way to achieve a smooth sounding loop. Compression "evens out" the changes in level during the loop which are perceived by the listener as amplitude modulation.

7. **Select the Crossfade Loop size and type, then press ENTER.** Crossfading means that as one part of the loop fades out, the other fades in. Instead of butt-splicing the end of the loop back to the beginning when forward looping, (or butt-splicing the loop end and start points with forward/backward looping), Crossfade looping smoothly blends the two sounds on either side of the splice. Crossfading virtually eliminates any loop glitches, although there may be level variations instead. These variations are not as noticeable as loop glitches. For more information, see Background: About Looping at the beginning of this section.

! Caution: *If there is not enough disk memory to back up a sample, the ESI will not let you Crossfade unless you disable the backup process in Digital Processing-9, Undo.*

```
LOOP   secs  samples
Start: 1.96    43280
End:   3.54    78162
 Compress Loop? Y/N
```

The two types of crossfade looping are:

- **EqPwr: (Equal Power)** This is a weighted crossfade that produces no apparent level shift. It is the most commonly used mode.
- **Linear:** This provides a straight mathematical crossfade. Use Linear mode with samples whose splice points are already close to optimum, such as a sample that has already been Auto Correlated.

When you press ENTER, the sample will be backed up so that crossfade looping can be undone (in Digital Processing, 9. Undo) if you are not happy with the results.

8. **Press YES to truncate (discard) all samples after the loop end, or NO to retain the samples after the loop end.** If you might want to re-loop the sample later, or try different loop points, press NO. If you are satisfied with the loop, press YES. This will save memory and the ESI will return to the Module Identifier.

9. **If you are not satisfied with the resulting crossfade or compression, or didn't really want to truncate those samples past the end point, proceed to Digital Processing, Undo (9).**

3. Truncate

The Truncate operation shortens a sample's length by trimming off individual samples from the beginning and/or end. Truncation is most often used to remove unneeded portions of a sample to conserve memory, but it can also be used to change instrument characteristics such as removing the attack from a plucked string note or isolating a particular section of a sample.

1. **Activate Digital Processing module.**
2. **Select Truncate (3).**
3. **Truncate the desired amount of samples from the start on line two and/or the end on line three, then press ENTER.**

✶ **Tip:** *The left and right cursor keys will change the start and end points so that they fall on positive zero-crossing points in the waveform.*

```
TRUNC  secs  samples
Start: 0.00    00001
End:   2.43    53610
Size:  2.43    53610
```

The sample size equals the difference in samples between the start and end points. Because of this, these settings interact with the size value on line four in these ways:

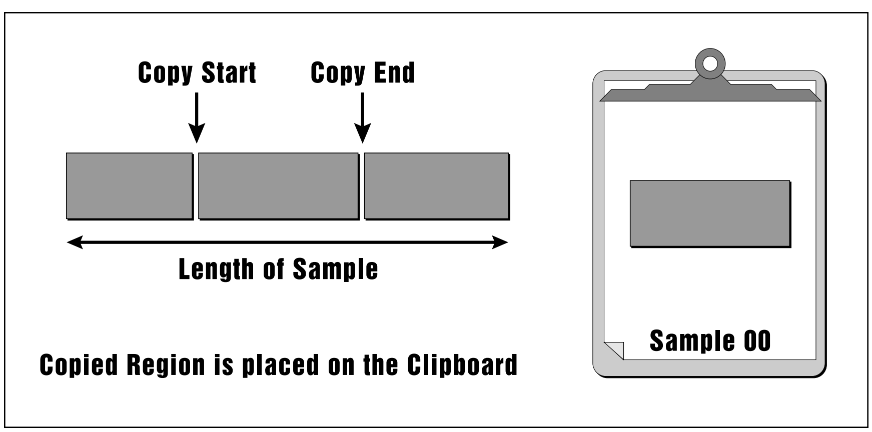
! **Caution:** *You cannot Undo the truncation settings.*

- Truncating the **Start** causes the size to change to maintain a constant **End Point**.
- Truncating the **End** causes the size to change to maintain a constant **Start Point**.

4. **If you are not satisfied with the results, or want to compare before and after, proceed to Digital Processing, Undo (9).**

4. Copy Region

Portions of a sample can be cut, copied, and pasted to other samples, or into the samples from which they came. The Copy function allows you to duplicate a section of a sample, and store that duplicate in a special part of memory called the clipboard. For more information see the section Background: Cut, Copy, Paste, and Undo.



1. **Activate Digital Processing module.**
2. **Select Copy Region** (4).
3. **Select the sample that you want to copy a section from, then press ENTER.**

```
  COPY REGION from
001 Selected Sample

  Select a Sample
```

4. **If the sample is stereo, the following screen will appear. Select the right side, left side, or both sides (stereo), then press ENTER.** Otherwise, proceed to the next step.

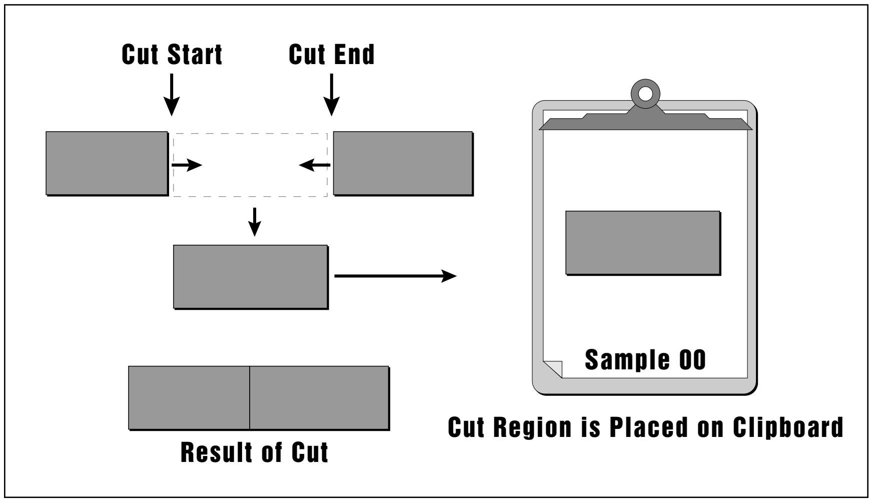
```
  COPY REGION from
001 Selected Sample
Side: Stereo
Select L/R/Stereo
```

5. **Specify the start point on line two and end point on line three for the portion of the sample to be copied, then press ENTER.** Use the up and down cursor to choose the appropriate line. The size of the portion to be copied on line four will change to reflect changes in the start and end points. Pressing ENTER saves the selected portion in the clipboard. Copying does not affect the original sample. The clipboard will retain this data until replaced by something else to be copied, cut, or backed up.

```
COPY   secs  samples
Start: 0.00    00001
End:   1.61    57881
Size:  1.61    57881
```

5. Cut Region

Portions of a sample can be cut, copied, and pasted to other samples, or the samples from which they came. The Cut function removes a section of a sample, and stores the cut portion in a special part of memory called the clipboard. For more information see the section Background: Cut, Copy, Paste, and Undo.



1. **Activate Digital Processing module.**
2. **Select Cut Region (5).**
3. **Select the sample that you want to cut a section from, then press ENTER.**

✶ **Tip:** *The left and right cursor keys will change the start and end points so that they fall on positive zero-crossing points in the waveform.*

```
 CUT REGION from
001 Selected Sample

 Select a Sample
```

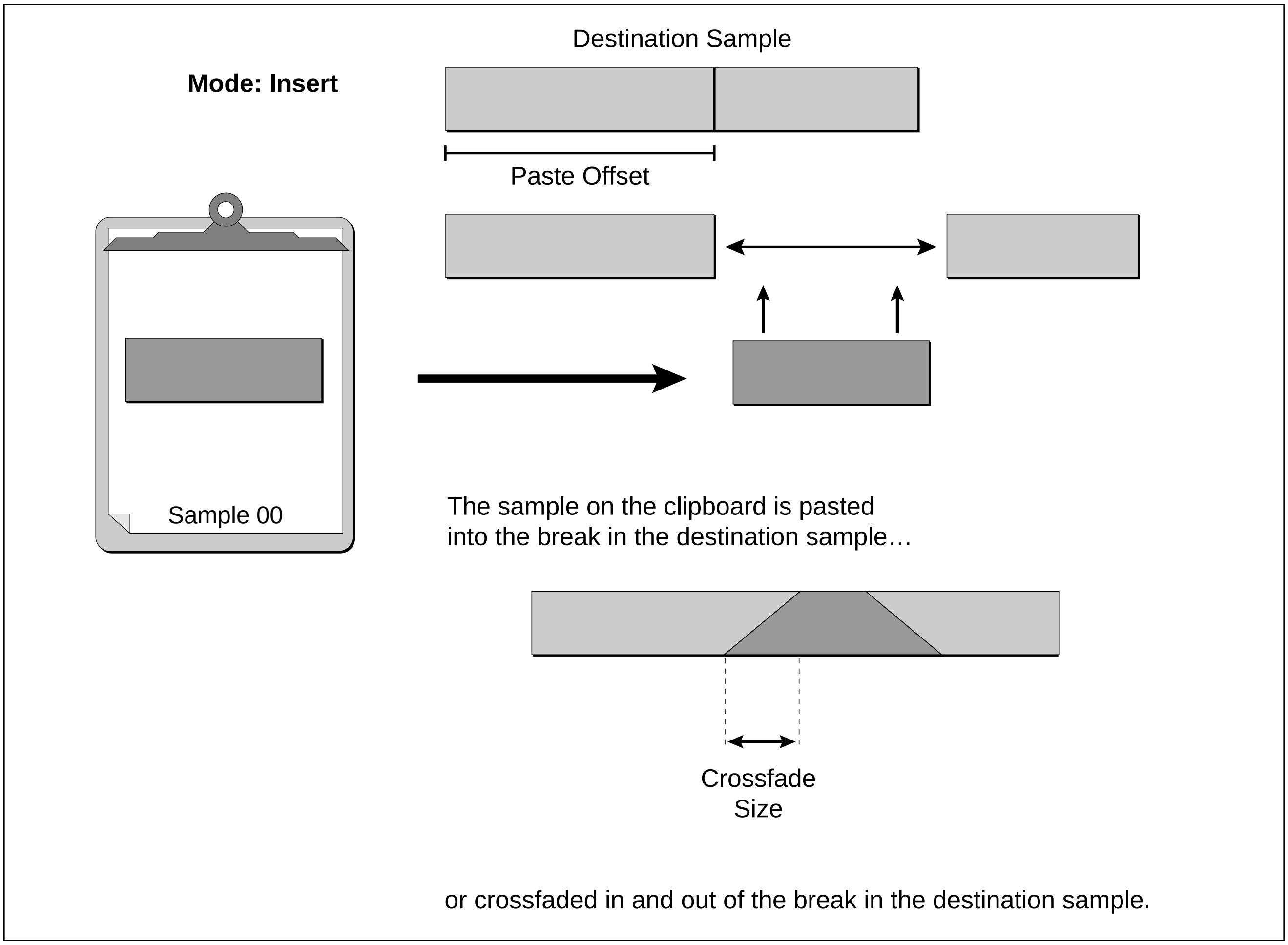
4. **Specify the start point on line two, and the end point on line three for the portion of the sample to be cut, then press ENTER.** Use the up and down cursor to choose the appropriate line. The size of the portion to be cut on line four will change to reflect changes in the start and end points. Cutting affects the original sample by deleting the part being cut. The clipboard will retain this data until replaced by something else to be copied or cut.

```
CUT    secs  samples
Start: 0.00   00001
End:   1.61   57881
Size:  1.61   57881
```

5. **If you are not satisfied with the resulting cut, proceed to Digital Processing, Undo (9).**

6. Paste Region

Portions of a sample can be cut, copied, and pasted to other samples, or the samples from which they came. Paste takes the clipboard contents (which holds the last cut or copied sample segment) and either inserts it in a sample at a specified point, or mixes it with a sample starting at a specified point. For more information see the section Background: Cut, Copy, Paste, and Undo.



1. **Activate Digital Processing module.**
2. **Select Paste Region (6).**
3. **Select the sample that you want to paste the clipboard contents into, then press ENTER.**

```
PASTE REGION into
S00 Selected Sample

 Select Dest Sample
```

✶ **Tip:** *The left and right cursor keys will change the Paste Offset point so that it falls on positive zero-crossing point in the waveform.*

4. **Specify the paste point on line two as an offset (in samples) relative to the beginning of the sample, then press ENTER.**

```
PASTE  secs  samples
Offset 0.00  000001

  Select Location
```

5. **Select the Paste Mode.** Choose whether to insert or mix the clipboard contents at the point selected in step three, then press ENTER.

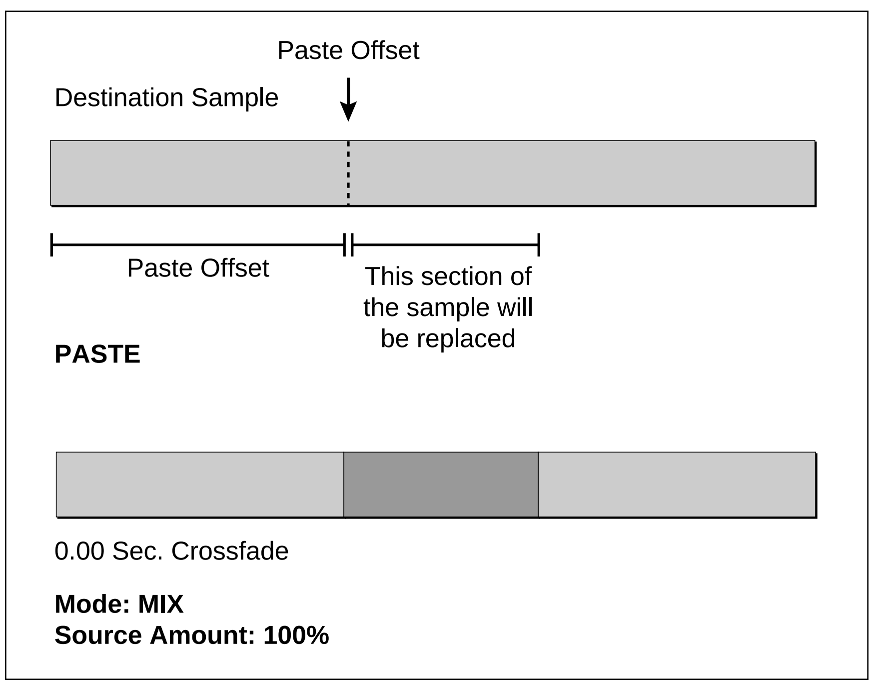
✶ **Tip:** *Use the Copy Sample function (Sample Management, 5) to paste the clipboard contents to an empty sample location.*

```
PASTE  secs  samples
Offset 1.06    33943
Mode:  Insert
Select Insert/Mix
```

Insert opens up a space in the sample into which the clipboard contents fit. Equal Power Mix and Linear Mix combine the clipboard contents with existing samples, starting at the point selected in step three. Normally, you will always use Equal Power. Use Linear Mix when the signals are very similar or when Equal Power causes a gain in amplitude.

6. **If you chose insert in step five, proceed to the next step. If you chose mix in step five, select the level of the contents to be mixed, then press ENTER.** 100% indicates a replacement.

```
PASTE  secs  samples
Offset 1.06    33943
Mode:Equal Power Mix
Source Amount:  100%
```

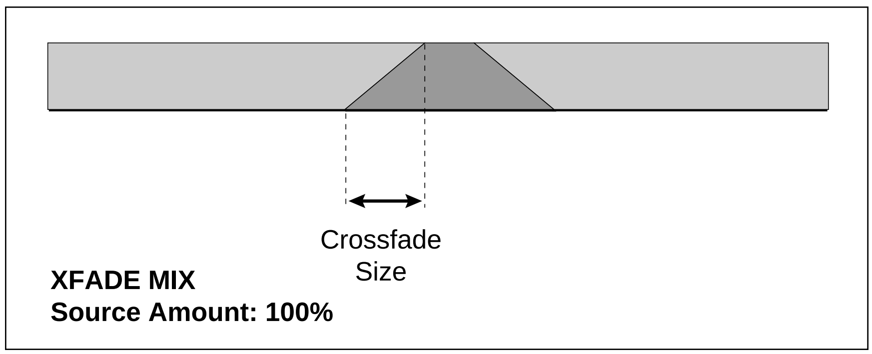


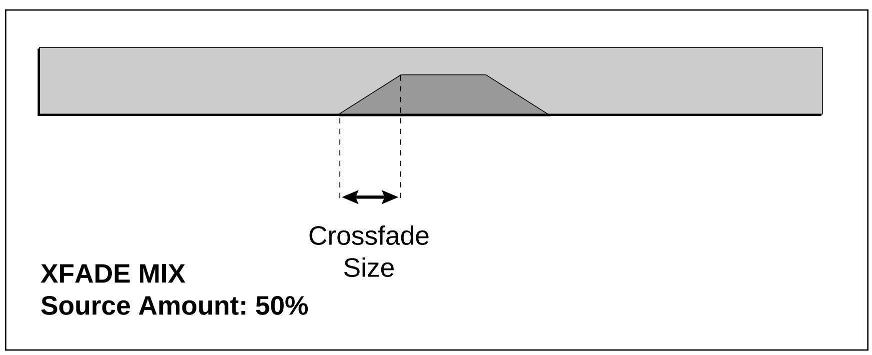
7. Select the Crossfade size and type, then press ENTER.

```
XFADE  secs  samples
Size:  0.00    00000
Type:         Linear
```

Crossfading minimizes glitches from pasting dissimilar sections of samples. When creating time delay and flanging/chorusing effects, do not select any Xfade time for best results. Your choices are:

- **Equal Power:** This is a weighted crossfade that produces no apparent level shift. It is the most commonly used mode.
- **Linear:** This provides a straight mathematical crossfade.





8. Press ENTER to perform the paste.

9. If you are not satisfied with the paste, or want to compare before and after, proceed to Digital Processing, Undo (9).

7. Digital Tools I

This submodule contains several additional numbered functions. These are extremely useful utilities for manipulating samples. The following is a short description of each function.

0. **Sample Calculator:** Calculates and displays the optimum pitch to sample rate ratios for single cycle loops at the desired pitch.
1. **Taper:** Adds a fade-in and or fade-out to the sample, thus smoothing out samples with abrupt beginnings or endings.
2. **Gain Change:** Alters the level of all or a part of a sample.
3. **Reverse:** Reverses all or part of a sample.
4. **Stereo <-> Mono:** Converts a stereo sample to mono or a mono sample to stereo.
5. **Left <-> Right:** Swaps sides if the current sample is stereo or moves a mono sample to the other side.
6. **DC Filter:** Removes the DC component from a sample, centering the waveform around the zero axis.
7. **Sample Integrity:** Samples may occasionally have slight looping problems due to data corruption of the sample header file. This function remedies the problem by reconstructing the sample header.

0. Sample Calculator

Use the Sample Calculator to compute the sample rate for a perfect single cycle loop. You provide the pitch and sample rate, the Single Cycle field displays the single cycle value based on the information you provide. Use the Single Cycle value to adjust the loop length. This provides an “in tune” single cycle loop.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Sample Calculator (0).** The display will reveal the Sample Calculator.

! Caution: *If the Single Cycle value shown is not an integer number, adjust the Sample Rate until it is an integer. Then use the Sample Rate Convert function (Digital Tools II, submodule SampleRate Convert (0) to convert the sample to the new rate.*

```
  SAMPLE CALCULATOR
Pitch:   440Hz   A3
Sample Rate: 44001Hz
Single Cycle: 100.00
```

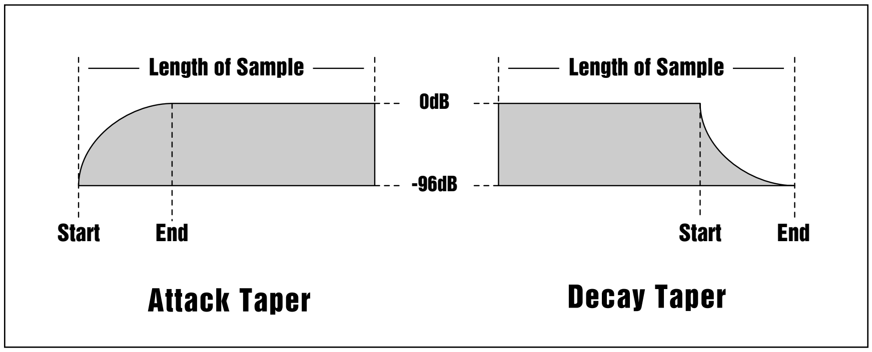
3. **Enter the pitch of the sample using either the keyboard or Data Entry Control.** The number of samples in a single cycle changes for each note.
4. **Select sample rate on line three. Use the INC/DEC buttons to change the value.** The number of samples in a single cycle changes for each sample rate.

To adjust the loop refer to the Digital Processing, submodule Loop (2).

1. Taper

Taper allows you to create an artificial decay on percussion samples in which the original decay is absent, create an artificial fast attack on a sound (such as a bowed violin with a slow attack), or clean up background noise when editing dialog. The diagram above shows the effect of Taper gain and attenuation on a sample using the Linear and Exp 3 curves.

✶ **Tip:** *Taper the ends of sounds before splicing to avoid clicks or pops at the splice point.*



1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Taper (2).**
3. **If the current sample is stereo, the following screen appears. Select the left side, right side, or both sides (stereo), then press ENTER.** Otherwise, proceed to the next step.

```
       TAPER

Side: Stereo
Select L/R/Stereo
```

4. **Select the desired taper points and press ENTER.** The display will show the current taper points, which will be the endpoints of the current sample.

✶ **Tip:** *The left and right cursor buttons will change the start and end points so that they fall on positive zero-crossing points in the waveform.*

```
TAPER  secs  samples
Start: 0.00   000000
End:   3.13   137873
Size:  3.13   137873
```

```
       TAPER
Start Amount: 0.00dB
End Amount:    -96dB
Type:         Linear
```

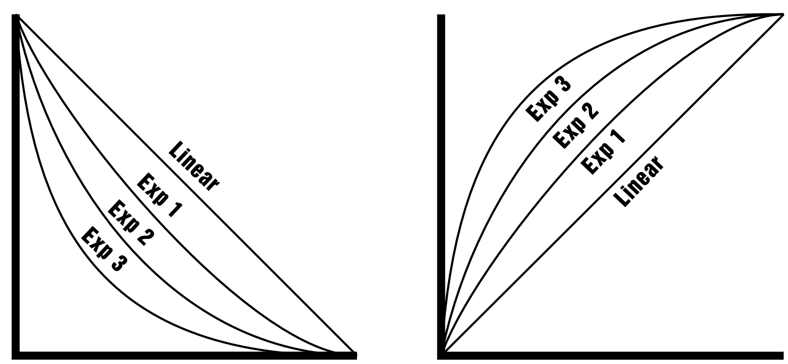
! **Caution:** *When using Taper Gain, avoid clipping the signal with too much gain. The Normalize display in the Gain Change submodule can be used as a sample headroom indicator.*

5. **Move the cursor to the parameter to be selected, select the desired value(s) with the Data Entry Control and press ENTER.** The sample will be tapered between the selected start and end points with the type of curve selected.

- **Start Amount:** is the amount of gain or attenuation applied at the start of the taper and is variable from -96 to +96 dB.
- **End Amount**: is the amount of gain or attenuation applied at the end of the taper and is also variable from -96 to +96 dB.
- **Type:** selects the type of taper curve: Linear, Exp 1, Exp 2, Exp 3. Graphs of these curves are shown below.

✶ **Tip:** *Percussive and Plucked sounds are notoriously hard to loop. By Taper boosting the entire plucked sound +3 dB to +5 dB), using the default Start and End settings, a very natural kind of compression is applied which makes looping these sounds much easier.*

Start with 0 dB and Taper up to +3 dB. As the sound decays, the gain will be boosted, leaving the attack of the sound unaffected.



Taper Attenuation Curves **Taper Gain Curves**

6. **If you are not satisfied with the resulting taper, or want to compare before and after, proceed to Digital Processing, Undo (9).**

2. Gain Change

Gain Change alters the level of all or part of a sample.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Change Gain (2).**
3. **If the current sample is stereo, the following screen will appear. Select the left side, right side, or both sides (stereo), then press ENTER.** Otherwise, proceed to the next step.

```
      CHANGE GAIN

Side:  Stereo
Select L/R/Stereo
```

4. **Select the desired gain change points and press ENTER.**

```
GAIN   secs  samples
Start: 0.00   000000
End:   3.13   137873
Size:  3.13   137873
```

The display will show the current gain change points, which are the end points of the current sample. After pressing ENTER, the ESI computes the amount of gain change necessary to achieve normalization or 0 dB headroom.

5. **Select the desired boost or cut with the Data Entry Control and press ENTER.** The amount of boost or cut is variable from -96 dB to +96 dB in 1 dB steps.

✶ **Tip:** *Boosting the gain more than the amount specified in Normalize will result in clipping which may or may not be audible. Use the Normalize display as a headroom indicator to limit the maximum amount of gain.*

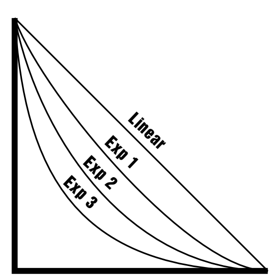
```
    GAIN CHANGE

Amount:        +00dB
 +04dB = Normalize
```

6. **Move the cursor to the parameter(s) to be adjusted, select the desired value(s) with the Data Entry Control and press ENTER.** The sample will be tapered between the selected start and end points with the type of curve selected.

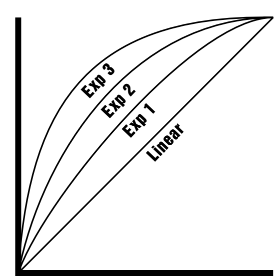
```
FADE   secs  samples
Size:  0.00   000000
Type:         Linear
```

- **Size:** sets the size of the crossfade between no gain change and gain change. This is variable from 0 to 1/2 the sample size.
- **Type:** selects the type of gain curve: Linear, Exp 1, Exp 2, Exp 3.

7. **If you are not satisfied with the resulting gain change, or want to compare before and after, proceed to Digital Processing, Undo (9).**



Gain Change Attenuation Curves



Gain Change Gain Curves

3. Reverse Section

Reverses all or part of a sample.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Reverse Section (3).**
3. **If the current sample is stereo, the following screen appears. Select the left side, right side, or both sides (stereo), then press ENTER.** Otherwise, proceed to the next step.

```
   REVERSE SECTION

Side: Stereo
Select L/R/Stereo
```

4. **Select the desired section of the sample to be reversed and press ENTER.**

✶ **Tip:** *The left and right cursor keys will change the start and end points so that they fall on positive zero-crossing points in the waveform.*

```
REVERSE secs  samples
Start:  0.00  000000
End:    3.13  137873
Size:   3.13  137873
```

The display shows the current points at which the sound will be reversed, which will be the endpoints of the current sample. The sample will be reversed between the selected start and end points.

5. **If you are not satisfied with the reversal, or want to compare before and after, proceed to Digital Processing, Undo (9).**

4. Stereo <-> Mono

Converts a stereo sample to mono or a mono sample to stereo. Mono samples are created using the left side of a stereo sample. Stereo samples are created by simply duplicating the mono sample on the other side.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Stereo <-> Mono (4).** The display asks if you want to convert the current sample to stereo (if it is mono) or to mono (if it is stereo).

```
  STEREO <-> MONO


  Convert to Mono?
```

3. **Press YES to convert the sample to stereo or mono or NO to exit the submodule.** The ESI will return to the Module Identifier.

5. Left <-> Right

Swaps sides if the current sample is stereo or moves a mono sample to the other side.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Swap Sides (5).** The display asks if you want to swap sides if the current sample is stereo, or move the sample to the other side if it is in mono.

```
     SWAP SIDES


  Swap Sides? Y/N
```

3. **Press YES to swap or change sides or NO to exit the submodule.** The ESI returns to the Module Identifier.

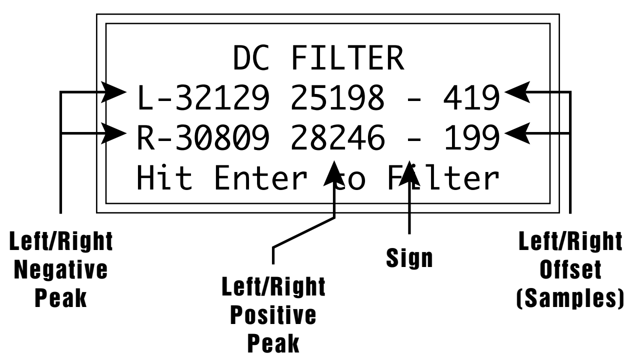
6. DC Filter

Removes the DC component from a sample, centering the waveform around the zero axis.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), DC Filter (6).** The ESI immediately begins scanning the current sample for DC offset.

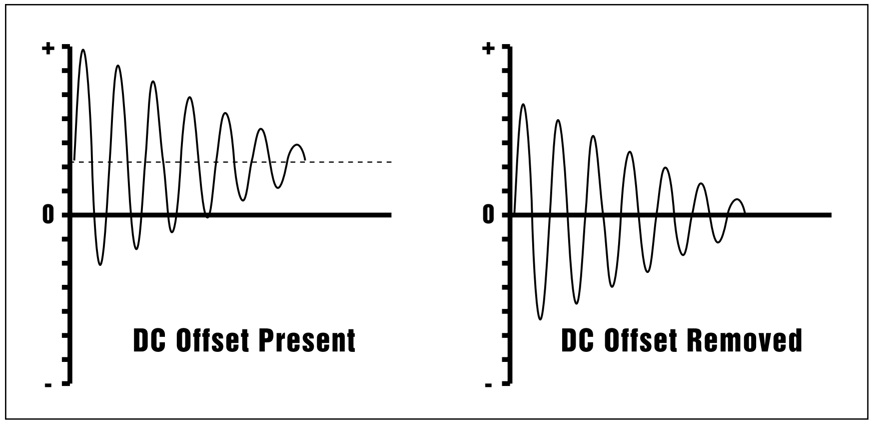
```
     DC FILTER


     Scanning...
```

When scanning is complete, the display will show:



The positive and negative peaks are expressed in 16-bit samples. The offset is the DC bias present in the sample. Full scale is the maximum level of a 16-bit number.

+32767	=	Full Scale (positive)
-32768	=	Full Scale (negative)



3. **Press ENTER to filter DC offset or Escape to exit the submodule.** The ESI will return to the Module Identifier.

7. Sample Integrity

Samples may occasionally have slight looping problems due to data corruption of the sample header file. This function remedies the problem by reconstructing the sample header. If a sample has been corrupted, the display may show an error message such as “Mono Start Zero!”.

1. **Activate Digital Processing module.**
2. **Select Digital Tools I (7), Sample Integrity (7).** The display shows:

```
FIX SAMPLES

Fix all samples? Y/N
```

3. **Press ENTER to fix samples in the bank.** The ESI will only modify loops which are in need of repair.

8. Digital Tools II

This submodule contains eight more digital functions. These are extremely useful utilities for manipulating samples. Following is a short description of each function.

0. **Sample Rate Convert:** Converts the sample to any sample rate between 7000 Hz and 50000 Hz.
1. **Digital Tuning:** Digitally re-tunes a sample within a range of ±1 octave.
2. **Compressor:** The compressor dynamically changes the gain of the sample based on the amplitude envelope of the sample. This is a non-realtime implementation of a full featured dynamic range compressor.
3. **Parametric EQ** (Equalizer)**:** This is a digital (non real-time) implementation of a one band parametric EQ, with +12 dB boost or -48 dB of cut and completely variable frequency and bandwidth controls.
4. **Time Compression:** This function changes the length of a sample without changing the pitch. This can be very useful for fitting samples to the beat of a song or fitting dialog into a spot. Samples can be compressed or expanded in length from 50% to 200%.
5. **Pitch Change:** This function changes the pitch of a sample without changing the time relationships between events. The maximum amount of pitch change is ±1200 cents (± one octave).
6. **Transform Multiply:** This function merges two sounds together in a unique way which can create many strange and beautiful sonic textures. Frequencies common to the original sounds are accentuated while uncommon frequencies are discarded.
7. **Doppler/Pan:** This unique function allows you to dramatically move a sound from front to back and side to side in a 2-D space. Several pre-computed paths as well as 10 user-definable paths are available.
8. **Sonic Enhancer:** The Sonic Enhancer adds brilliance and "cut" to a sample making it stand out in a mix.

0. Sample Rate Convert

Sample Rate Conversion can be used for saving memory, increasing the upward transposition range of a sound, or exactly matching the sample rate to a multiple of the sounds frequency for perfect single cycle loop (use the sample calculator).

Sampling at a high sample rate provides better frequency response than sampling at a slower rate, but uses up more memory. If you need to reclaim some of that memory, and are willing to trade off sample frequency response, samples can be converted from a higher rate to a lower one. Use your ear to compare sounds at the two sample rates. If you can't tell the difference, use the lower rate.

1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Sample Rate Convert (0).** The display will show the current sample rate and sample size.

```
SAMPLE RATE CVT  f:0
Rate:        22050Hz
Size:         107220
Loop Size:   5035.00
```

3. **Select the new sample rate and press ENTER.** The display updates the sample size to reflect the new sample rate. The Loop Size display shows the size of the resulting loop. If a fractional number is shown, you may want to adjust the sample rate until the loop size is an integer, otherwise the loop size will be slightly altered by the conversion process.

The "**f**" in the upper right corner indicates the reconstruction filter that is used in the resampling process. Higher numbers indicate more interpolation filtering (cutting more high frequencies). There are six interpolation filter frequencies (0-5).

Tip: By performing successive sample rate conversions with lower reconstruction filter numbers, the automatic filtering system can be subverted and the sample can be kept brighter.

4. **If you are not satisfied with the resulting sample rate conversion, or want to compare before and after, proceed to Digital Processing, Undo (9).**

1. Digital Tuning

Digital Tuning allows you to change the pitch of a sound in order to splice or combine it with another sound of a different pitch. Try offsetting the pitch of a copied sample by a few cents, then combine it with the original for flange and chorus effects. If a sample is slightly out of tune, it is usually better to fix the sample once and for all, rather than re-tune the sample when it is placed on the keyboard.

✶ **Tip:** *Re-tuning pitch downward increases the sample size. Re-tuning pitch upwards decreases the sample size.*

1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Digital Tuning (1).** The display will show the current tuning offset, sample size and loop size.
3. **Select the desired amount of re-tuning, then press ENTER.** The display will update the sample size to reflect the new tuning. The loop size display shows the size of the resulting loop. If a fractional number is shown, listen to the resulting loop carefully. The loop size will be slightly altered by the conversion process.

The "**f**" in the upper right corner indicates the reconstruction filter that will be used in the resampling process. Higher numbers indicate more interpolation filtering (cutting more high frequencies). There are six interpolation filter frequencies (0-5).

Tip: By performing successive digital tunings with lower reconstruction filter numbers, the automatic filtering system can be subverted and the sample can be kept brighter.

```
DIGITAL TUNING    f:0
Tuning:   +1200 cents
Size:          209882
Loop Size:   69930.00
```

4. **If you are not satisfied with the resulting re-tuning, or want to compare before and after, proceed to Digital Processing, Undo (9).**

2. Compressor

The Digital Compressor is a digital (non-realtime) equivalent of an analog dynamic range compressor with attack and release times, adjustable threshold, adjustable ratio, and three modes of operation.

1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Digital Compressor (2).**
3. **If the current sample is stereo, the following screen will appear. Select the left side, right side, or both sides (stereo), then press ENTER.** Otherwise proceed to the next step.

```
  DIGITAL COMPRESSOR

Side: Stereo
 Select L/R/Stereo
```

4. **Select the desired area to be compressed and press ENTER.**

! Caution: *The gain setting which is in effect at the end point of the selected section will remain in effect for the rest of the sample. This is done to prevent nasty pops and phase problems caused by an abrupt transition back to the normal gain.*

```
COMP.  secs  samples
Start: 0.00   000000
End:   3.13   137873
Size:  3.13   137873
```

The display will show the current points of the sample to be compressed, which will be the start and end points of the current sample. Select the section of the sample to be compressed, then press ENTER.

```
COMPRESSOR Mode: rms
Thres:   center 100%
Ratio:        4.00:1
Atk:999ms  Rel:999ms
```

5. **Position the cursor under the desired parameter and adjust the value using the Data Entry Control.**

Mode: RMS or Peak.

> **RMS -** Root-mean-square or an “average of the magnitude of the signal”. RMS represents the “true” energy content of a signal.
>
> **Peak** - Uses the peak amplitude of a signal to determine the amplitude. The peak amplitude is a meaningful measurement in a digital system because of the 96 db (16-bit) headroom limit.

Threshold: Above, Center, Below, %

> **Above -** Only signal levels above the threshold % will be affected by the compressor.
>
> **Center -** Signal levels above as well as below the threshold % will be affected by the compressor.
>
> **Below -** Only signal levels below the threshold % will be affected by the compressor.
>
> **% -** Determines the threshold level as a percentage of 100% of 16 bits.

Compression Ratio: Variable from 0.01 : 1 to 99.8 : 1.

Determines the amount of compression or expansion. Ratios of greater than 1:1 compress dynamic range. Ratios of less than 1:1 expand dynamic range.

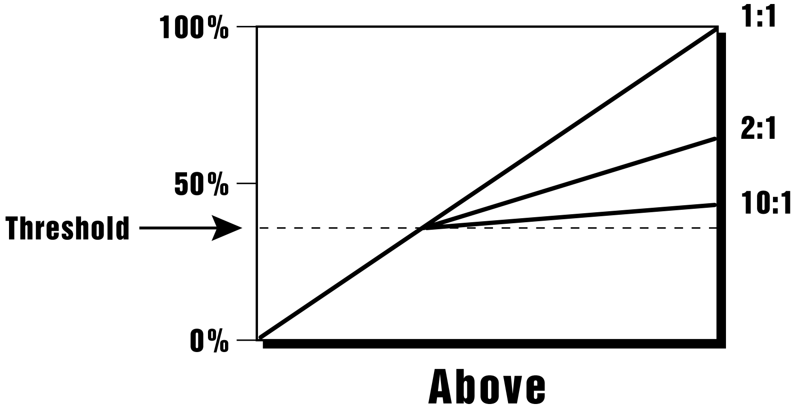
> **Compression -** Reduces the amount of dynamic range. Levels remain more constant.
>
> **Expansion** - Expands the amount of dynamic range. Changes in level are exaggerated.

Atk (Attack Time)

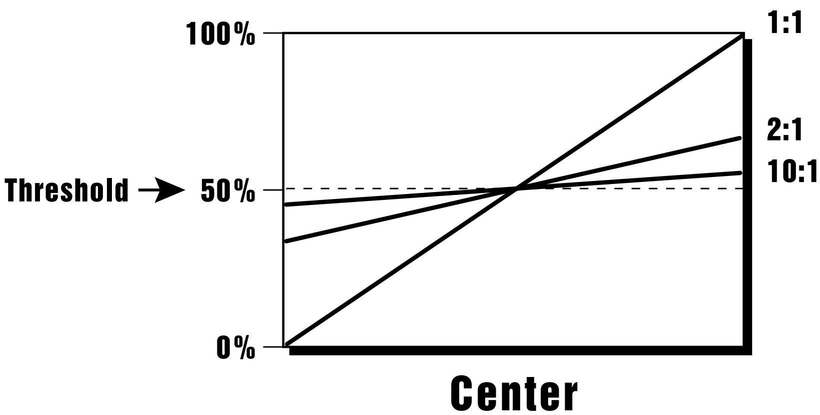
Determines how quickly the gain will be turned down. The attack time is variable from 0 to 999 milliseconds.

Rel (Release Time)

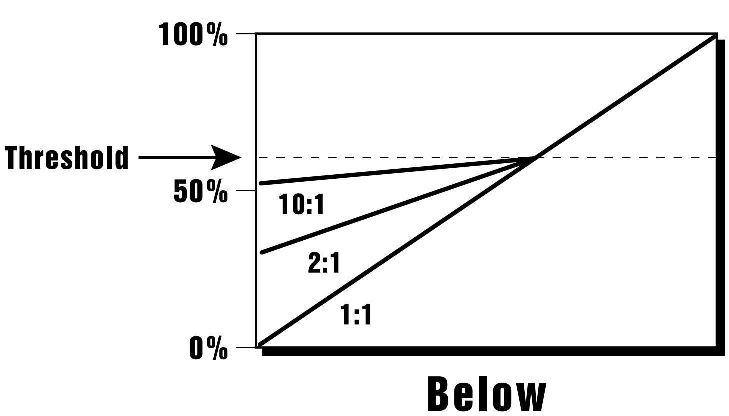
Determines how quickly the gain will be turned up. The release time is variable from 0 to 999 milliseconds.



Only signal levels ABOVE the Threshold % will be affected by the compressor.



Signal levels ABOVE as well as BELOW the Threshold % will be affected by the compressor.



Only signal levels BELOW the Threshold % will be affected by the compressor.

Using the Digital Compressor

Limiter

A limiter prevents the signal from exceeding a preset level (threshold). Signal levels below the threshold will be unaffected.

Set the controls as follows:

Threshold: Above, XX% (where XX is the limit point).

Ratio: >10:1

Attack Time: 1 mS

Release Time: approximately 100 mS

Musical Compression (e.g. Guitar)

This type of compression tries to keep the volume constant, generally to increase the sustain of the instrument. As the note dies away, the compressor will boost the level in an effort to keep the level constant.

Set the controls as follows:

Threshold: Center, XX% (where XX is the compression point).

Ratio: approximately 4:1

Attack Time: 1 mS to 100 mS

Release Time: > 100 mS

Noise Reduction

Noise reduction will reduce low levels even further in the assumption that low levels are noise.

Set the controls as follows:

Threshold: Below, approx. 30%

Ratio: approximately 0.7:1

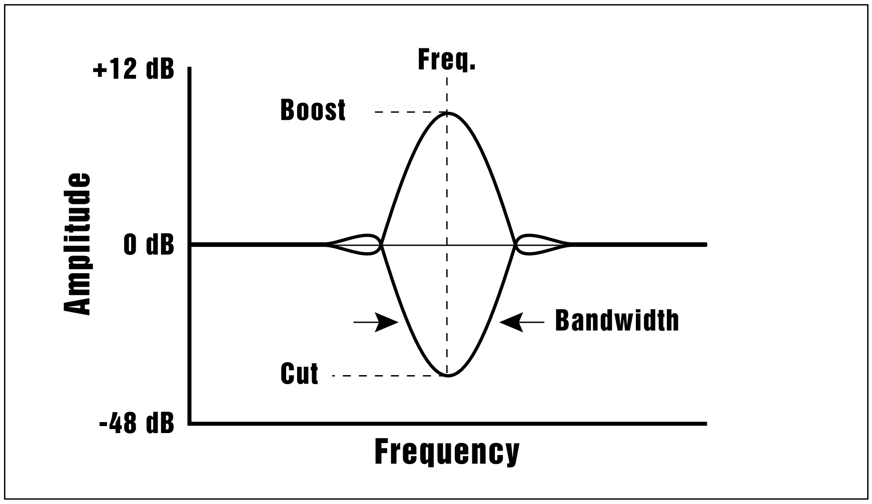
Attack Time: approximately 100 mS

Release Time: approximately 100 mS

3. Parametric Equalizer

Parametric EQ allows you adjust the individual parameters of the filter. Boost and Cut controls "how much" of the signal will be boosted or cut. Center Frequency sets the center frequency to be boosted or cut, and the Bandwidth control sets the width of the band to be boosted or cut. The three parameters are diagrammed below.

The parametric equalizer is a digital (non real-time) equivalent of an analog equalizer with +12 db of boost and a whopping -48 dB of cut. Accurate center frequency and bandwidth controls are precise and exceptionally wide range.



1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Parametric EQ (3).**
3. **If the current sample is stereo, the following screen will appear. Select the left side, right side, or both sides (stereo), then press ENTER.** Otherwise proceed to the next step.

```
   PARAMETRIC EQ

Side: Stereo
 Select L/R/Stereo
```

4. **Select the desired points to be EQ'ed and press ENTER.**

✶ **Tip:** *The left and right cursor buttons will change the start and end points so that they fall on positive zero-crossing points in the waveform. This will minimize clicks in the sound when processing only part of a sample.*

```
EQ      secs  samples
Start:  0.00  000000
End:    3.13  137873
Size:   3.13  137873
```

The display shows the current points of the sample to be equalized, which will be the start and end points of the current sample. Select the section of sample to be EQ'ed, then press ENTER.

5. **Move the cursor to the parameter(s) to be adjusted, select the desired values with the Data Entry Control, and press ENTER.**

✶ **Tip:** *A standard analog parametric equalizer may be useful in order to locate the frequencies which need EQ. The digital EQ can then be used to actually filter the sample with ultra-low noise and phase linear response.*

```
   PARAMETRIC EQ
Gain:           +12dB
Center Freq:  1000Hz
Bandwidth:      50Hz
```

6. **If you are not satisfied with the resulting EQ or want to compare before and after, proceed to Digital Processing, 9. Undo.**

4. Time Compression

This function allows you to change the length of a sample without changing the pitch. This is a useful function for fitting samples to the beat of a song or fitting dialog into a spot. Samples can be compressed or expanded in length from 50% to 200%.

1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Time Compression (4).** The display will show the current time compression ratio and signal type.

Time Compression Ratio:
200% = Double Length
50% = Half Length

```
  TIME COMPRESSION
Ratio:           110%
Type:           broad
Length:  3.1=>  3.4s
```

3. **Select the desired time compression (or expansion) ratio.** The lower line of the display changes to show the resulting length of the compressed or expanded sample.
4. **Select the general type of sample to be processed.** Refer to the chart on the following page for a list and brief description of the sample types.
5. **If you are not satisfied with the resulting Time Compression or want to compare before and after, proceed to Digital Processing, Undo (9).**

✶ **Tip:** *If you are not happy with the results of Time Compression, simply Undo, then choose another algorithm.*

5. Pitch Change

This function is the exact opposite of Time Compression in that it changes the pitch of a sample without changing the time. The maximum amount of pitch change is ±1200 cents (± one octave).

1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Pitch Change (5). The** display will show the current amount of pitch change in cents and signal type. (1 cent = 1/100 of a semitone.)

```
    PITCH CHANGE
Tune:        +  52cts
Type:           mid-2
```

3. **Select the desired pitch change amount.** The lower line of the display changes to show the resulting length of the compressed or expanded sample.
4. **Select the general type of sample to be processed.** The choices are:

deep	Predominant deep bass (to 14 Hz)
bass	Predominant bass (to 20 Hz)
mid-1	Average source material
mid-2	Average source material (high-mids)
high	Source material with high frequencies
tight	Maintains time accuracy - Drum loops
broad	Low bass energy but critical highs
broad-smooth	Both high & low frequencies - Smooth output
difficult	Inharmonic or broadband material
noisy	Non-pitched - Sound Effects, etc.
tight-smooth	Preserves rhythmic accuracy. Use small ratios.
x-smooth	Preserves rhythmic accuracy. Use small ratios.

5. **If you are not satisfied with the resulting Pitch Change or want to compare before and after, proceed to Digital Processing, Undo (9).**

6. Transform Multiply

✭ **Tip:** *Transform Multiplication can take quite a bit of time with longer samples. Begin your experiments with short samples or even sample attacks as these will give good initial results.*

This unique function merges two sounds together in a way that accentuates frequencies common to both sounds while discarding uncommon frequencies. Because of this characteristic, Transform Multiplication tends to work best with sounds that are harmonically rich. Using this function is easy. Just pick two sounds and multiply! Although you have to wait to hear the results, Transform Multiplication can produce many strange and beautiful textures unattainable by any other means. The length of the resulting sample will be equal to that of the current sample.

1. **Activate Digital Processing module.**
2. **Select submodule Select Sample (0).**
3. **Select the first source sample to be multiplied, then press Enter.**
4. **Select Digital Tools II (8), Transform Multiplication (6).** The display shows:

```
 TRANSFORM MULTIPLY
072 Flute C4
L    0.1secs 29312Hz
Select Second Sample
```

Transform Multiplication Ideas:

● *Try using the same sound for both samples.*

● *Splicing silence to the beginnings or ends of short samples can change the spectral characteristics of the result.*

● *Using speech as one of the sources, it is possible to "speak from within" violins, bassoons, cymbals, etc.*

5. **Select the second sample to be multiplied using the INC/DEC buttons, the Data Entry Control or the keypad.** The third line of the display shows the characteristics of the second sample (L, R, or L/R), the length and the sample rate.
6. **Press ENTER to select the second sample.** The display shows the time required to process the samples.

```
 TRANSFORM MULTIPLY

Will take 26 mins..
   Continue? Y/N
```

7. **Press Yes to start the computation or No to return to the Module Identifier.**
8. **If you are not satisfied with the resulting Transform Multiplication or want to compare before and after, proceed to Digital Processing, Undo (9).**

! Note *Transform Multiplication needs extra memory to perform its thousands of calculations. If you get a memory error, load just the two samples into the ESI and try again.*

7. Doppler/Pan

This function allows you to apply either a pre-programmed or user-definable sound path to a mono sample or the left side of a stereo sample. The result is a stereo sample with pitch shifted and left-right gains adjusted according to the path. The sound can move dramatically forward-back and left-right in a 2-D space in front of the listener. Several pre-programmed paths and up to 10 user-definable paths are available with up to 26 points per path.

1. **Activate Digital Processing module.**
2. **Select Digital Tools II (8), Doppler/Pan (7).** The display will show the current path, level threshold and Doppler status.

```
    DOPPLER/PAN    ⇢
Path:   small circle
Doppler:          on
Threshold:     -44dB
```

3. **Select the desired Path for the Doppler/Pan.** The lower line of the display changes to show the resulting length of the compressed or expanded sample. The choices are:

Fast-by (L->R)	Sound moves left to right quickly
Slow-by (L->R)	Sound moves left to right slowly
Far->Near (fast)	Sound moves far away to close quickly
Far->Near (slow)	Sound moves far away to close slowly
Near->Far (fast)	Sound moves close to far away quickly
Near->Far (slow)	Sound moves close to far away slowly
Small Circle	Sound moves in an 8 ft. circle in front
Medium Circle	Sound moves in a 50 ft. circle in front of you
Large Circle	Sound moves in a 120 ft. circle in front of you
Huge Circle	Sound moves in a 250 ft. circle in front of you
Random 1	Sound moves in a random path in front of you
Random 2	Sound moves in a different random path
User Path 1-10	Sound moves in a user-programmed path

The pre-computed paths are always initially scaled to the length of the sample, although this time may be changed using the Duration parameter (read on). The 10 user-defined paths are saved along with the bank so that they may be applied to other samples over multiple-user sessions.

4. **Turn Doppler Pitch Shift On or Off.** With Doppler turned Off, only the panning effect will be enabled.

★ **Tip:** *Pressing the right cursor button from the initial screen selects the Path Management screen described on the following page.*

5. **Set the amplitude Threshold**. Certain sound paths may take the sound source very far away from the listener, resulting in little or no resulting amplitude. The "Threshold" parameter specifies the maximum amplitude attenuation of the original sample that will ever occur. By setting this parameter appropriately, you can ensure that the resulting sample will produce at least a minimum amplitude output at every point along the path. A setting of 0 dB would prevent any amplitude changes whatsoever. A setting of -96dB would not prevent the sound from completely fading away at some points.
6. **Press ENTER to continue.** The Path Parameters screen (shown below) will appear.

```
   PATH PARAMETERS

Duration:      3.06s
Auto-repeat:     off
```

7. **Select a shorter Path Duration if desired.** The Path Duration allows you to scale the path time so that it runs to completion in a time shorter than the length of the sample. The default value is always the length of the sample. If auto-repeat is off (see below), the sound will remain at the end of the path until the sample is complete.
8. **Turn Auto-repeat On or Off as desired.** Auto-repeat causes the path to repeat if the end is reached before the sample has completely played. Auto-repeat will only occur if path duration is set to a value less than the sample length. The default setting for Auto-repeat is Off.
9. **Press ENTER to begin processing the sample.** The display shows:

```
     DOPPLER/PAN

Processing path...
```

10. **If you are not satisfied with the resulting Doppler/Pan or want to compare before and after, use Digital Processing, Undo (9).**

Path Management Screen

Sound paths may optionally be edited and processed in a number of ways. **If the right cursor is pressed in the initial Doppler/Pan screen, the Path Manager screen appears:**

```
←   PATH MANAGER

Path:    user path 1
Function:    reverse
```

! Note *The pre-programmed paths must first be copied to a user location before they can be edited.*

The path may be selected in this screen as well. “Function” specifies the path processing function. The available functions are:

- **Edit** -------- edit a user path
- **Copy** ------ copy a path into one of the user paths
- **Reverse** -- cause a user path to be played in reverse
- **Flip** -------- flip a user path around the X-axis
- **Offset** ----- apply an x and/or y offset to a user path
- **Clear** ------ initialize a user path

Copy Path

This function copies a pre-programmed or user path to a user path. When Copy Path is selected, the following screen will appear:

```
     COPY PATH
From:  medium circle
To  :    user path 1
 Select Source Path
```

Select the source and destination paths, then press ENTER to copy the path.

Reverse Path

This function reverses the direction of a user path. When Reverse Path is selected, the following screen will appear:

```
    REVERSE PATH
Reversing path:
    user path 2
 Are You Sure? Y/N
```

Flip Path

This function causes a user path to be mirrored around the "X" axis so that points that were to the left side of the listener are now on the right and vice-versa. Selecting Flip Path causes the following screen to appear:

```
     FLIP PATH
Flipping path:
    user path 2
 Are You Sure? Y/N
```

Offset Path

This function allows you to add an offset to the left and right path points, moving the entire path closer or farther, or more to the left or right. The X offset adds to all X values; the Y offset adds to all Y values in the path. The resulting sums are clipped to the coordinate system (shown under Path Edit). When Offset Path is selected, the following screen will appear:

✶ **Tip:** *The "X" offset varies the L-R position. The "Y" offset varies the front-back position.*

```
    OFFSET PATH
X offset:    +265
Y offset:    +  0
  Enter X offset
```

Press ENTER to add the offsets to the path.

Clear Path

This function clears all x,y and time values to zero. If this "zero" path were used to process a sample, the sound would not move. When Clear Path is selected, the following screen will appear:

```
    CLEAR PATH
Clearing path:
   user path 2
 Are You Sure? Y/N
```

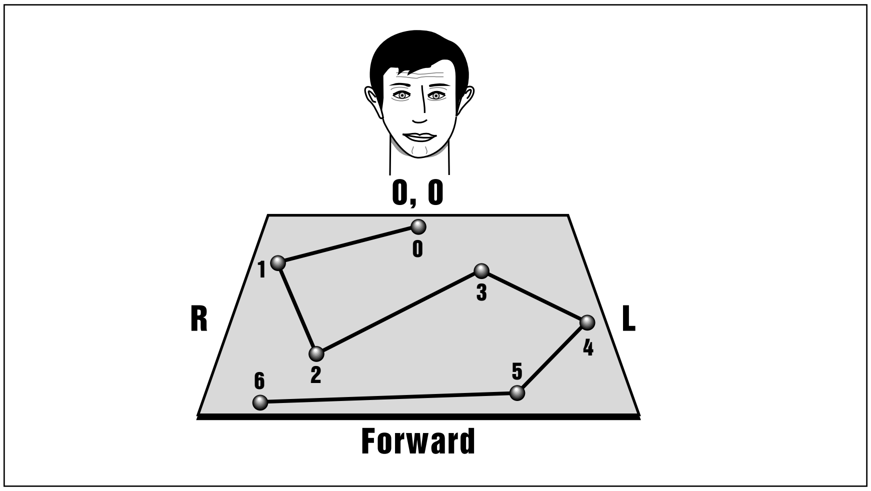
Press YES to clear the path.

Path Edit

The Edit Path function allows you to specify up to 26 points and times for each of the ten available paths. When Path Edit is selected, the following screen will appear:

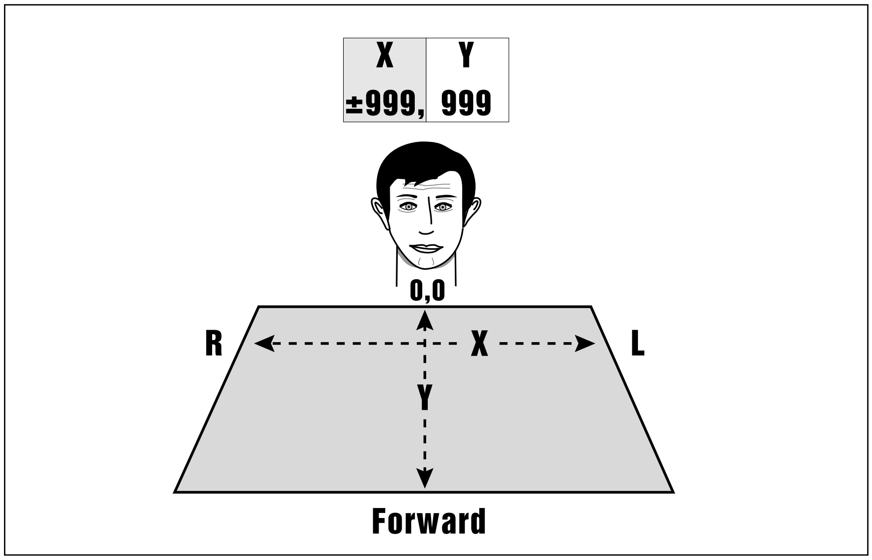
```
     PATH EDIT
0  0.00s [-999,500]
1  0.52s [+  0,  0]
2  0.86s [+224,999]
```

Each line of this screen is a point in the path, specified by [X,Y] position (numbers inside the [brackets]) and a time, in seconds. The point number is the left-most number of each line and may not be edited. By using the INC/DEC buttons or the Data Entry Control while the cursor is under a point number, the path edit screen can be scrolled to display and edit all 26 points.



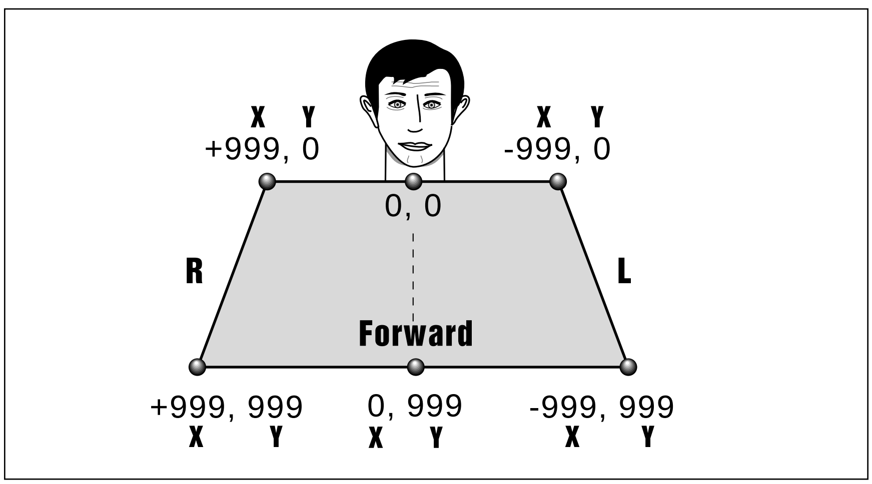
Up to 26 points defining the sound location can be specified. Each point has an associated time parameter to move the sound from the previous point.

X-Y coordinates are specified in 10ths of feet. X values range from -999 (-99.9 feet) to +999 (99.9 feet), Y values from 0 to 999 (99.9 feet). The listener is considered to be at [0,0] as shown below:



The first part of the coordinate designates the left/right position. The second part of the coordinate designates the forward positioning.

Time values range from 0 to the length of the current sample. Note that if this path is applied to a different sample, all time values will be scaled to the length of the different sample.



The diagram above shows the extreme positions of the path coordinates.

Once path editing is complete, press ENTER to return to the Path Management screen. When the path is ready to apply to the sample, press the left cursor button while in the Path Management screen to return to the main Doppler/Pan screen. Press ENTER again to move ahead to the Path Parameters screen and once more to process the sample.

8. Sonic Enhancer

The Sonic Enhancer adds brilliance and “cut” to a sample, making it stand out in a mix. This effect is especially effective on vocal samples to which it adds sheen, clarity and presence. The Sonic Enhancer works by generating new harmonics related to the source material.

The parameters are as follows:

- **Amount:** Sets the intensity of the effect. Lower values (<40%) tend to work best as they keep the effect subtle.
- **Tune:** Sets the frequency range of the effect, variable from 0-9. Higher numbers emphasize higher frequencies.
- **Fade In:** Sets the amount of time from the start point for the effect to fade in to the programmed amount.
- **Fade Out:** Sets the amount of time before the end point that it will take for the effect amount to fade out to zero.

▶ To Enhance a Sample:

1. **Press the Digital Processing key**.
2. **Select Digital Tools II (8), Sonic Enhancer (8).**
3. If the current sample is stereo, the following screen appears. **Select the left side, right side, or both sides (stereo), then press Enter**. Otherwise proceed to the next step.

```
   SONIC ENHANCER

Side:  Stereo
 Select L/R/Stereo
```

4. **Select the section of the sample to be processed, then press Enter.** The display shows the current endpoints, which are the endpoints of the sample.

```
ENHANCE secs  samples
Start:  0.00   00000
End:    0.58   25745
Size:   0.58   25745
```

```
   SONIC ENHANCER
Amount:           40%
Tune:               9
FadeIn:000 Out:000ms
```

5. **Move the Cursor to the desired parameter, and adjust the value** using the Data Entry Control. The Sonic Enhancement process will be applied to the sample between the selected start and end points.

6. **If you are not satisfied with the resulting process, or want to compare before and after, proceed to Digital Processing, Undo (9).**

9. Undo

Have you ever wanted a time machine so that you could go back and undo a mistake? This function may be the next best thing.

To Restore A Sample To Its Original State:

1. **Activate Digital Processing module.**
2. **Select Submodule Undo (9).**

```
UNDO  TRUNCATION
001  Selected Sample

Backup:      enabled
```

! Caution: *The Undo function will not work unless you have a hard disk drive connected.*

The display will show whether backup is currently enabled or not. If backup is not enabled, you will not be able to restore the sample. Otherwise, press ENTER. The original sample will be restored, the processed sample will be stored in the clipboard, and the ESI will return to the Module Identifier.

To Compare a Processed Sample with the Original Sample:

1. **Activate Digital Processing module.**
2. **Select Undo (9).**

```
REDO  TRUNCATION
001  Selected Sample

Backup:      enabled
```

The display will show whether backup is currently enabled or not. If backup is not enabled, you will not be able to compare samples. Otherwise, press ENTER. The original sample will be restored, the processed sample will be stored in the clipboard, and the ESI will return to the Module Identifier.

3. **Re-select Undo (9).** The processed sample will be restored, and the original sample will be stored in the clipboard. You can continue switching back and forth between the processed and original samples by repeating this step until you decide which sample you want to keep.

It is best to use Undo ONLY when you have a fixed hard disk drive installed on the SCSI bus. If you have only removable hard disks installed and remove the one containing the backup data, the ESI may become confused. The ESI will select a fixed hard disk as the backup drive, if available. If you have only removable disks and still want to use Undo, insert one removable disk in the drive. **Mount Drives in the Master menu sets the Undo device.** Leave the removable disk in the drive until you have finished all DSP operations.

To Disable or Enable the Backup Process:

1. **Activate Digital Processing module.**
2. **Select Undo (9).**

```
REDO  TRUNCATION
S00  Selected Sample

Backup:     disabled
```

The display will show whether backup is currently enabled or not.

3. **Use the INC/DEC buttons to choose whether the backup function is enabled or disabled, then press ENTER.**