9 Dynamic Processing

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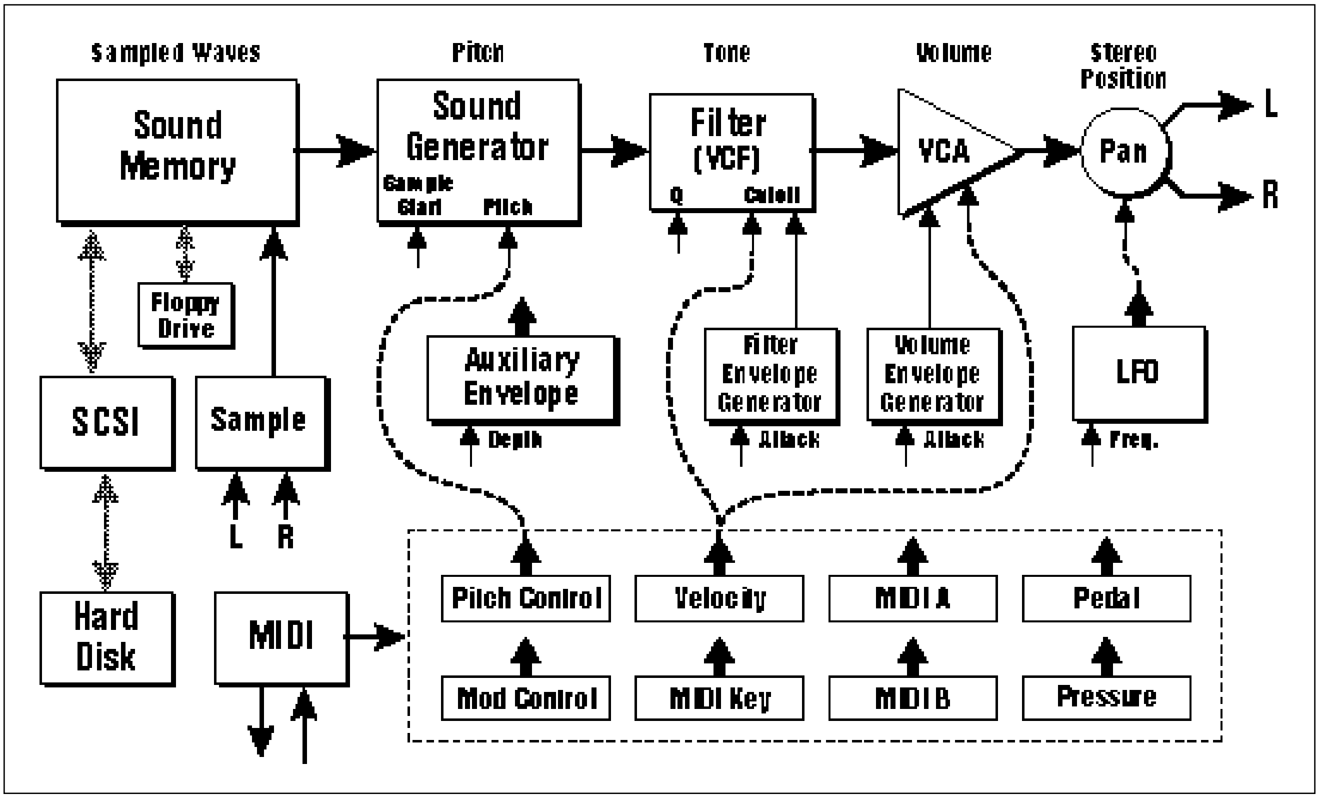
Background

The Dynamic Processing module of the ESI provides you with all the intuitive control options of an analog synthesizer. But, since you are starting with complex digitally sampled sounds instead of simple square or sawtooth waveforms, the possibilities are multiplied dramatically. Each of the 64 channels contains three AHDSR type envelope generators, one multi-waveform LFO with delay and variation, 19 ultra powerful filters (VCF) with resonance, one level VCA, one stereo panning network, and an extremely flexible routing scheme which ties everything together. The diagram below illustrates the layout of an ESI channel.



This block diagram illustrates the general architecture of the ESI. The dotted lines show how realtime controllers, envelopes and LFOs can be routed to modulate the sound.

Each key on the keyboard can contain two zones (primary and secondary, just like samples), and each of these can have completely different sets of analog parameters applied to it. The zone concept makes the programming and modifying of parameters quite straightforward.

◩ **Note:** Zone selection works slightly differently depending on the module you are using. In the Dynamic Processing module, selecting a Zone only affects the Dynamic Processing parameters. In the Preset Definition module however, Dynamic Processing parameters and Samples are selected when you select a zone.

The Zone Concept

A zone is simply a selected range of the keyboard. That's it! Nothing mysterious about it. Dynamic processing parameters can be programmed for any range of the keyboard (zone) regardless of where the samples lie. For instance, a completely different pan, LFO rate, and VCA envelope can be set for each individual key, or range of keys.

Zones can be set differently for each parameter you are adjusting. For example, the VCF cutoff can be set to one value for the entire keyboard, the pan position can be set for each individual key, and then the VCA envelope can be set for a portion, or all of the keyboard. Simply define the zone for the desired keyboard range and change the parameter. That's all there is to it.

Quick Zone

! Caution: Be careful! Quick Zone can be a bit confusing at times. Quick Zone was designed as a convenience feature for advanced programmers. It's probably best to leave quick zone turned Off until you have a good feel for programming presets.

Quick Zone offers an especially fast and efficient means of creating and accessing zones in the Dynamic Processing module. When Quick Zone is “On”, all you need to do in order to create or access a zone is press down the keys on the keyboard defining the range and access the Dynamic Processing parameters. For example, if two keys such as C2 and A#3 are held down and the filter cutoff is changed, then all samples in the range C2-A#3 will be assigned the new filter cutoff value.

To access Quick Zone, enter the Dynamic Processing module. The display will show:

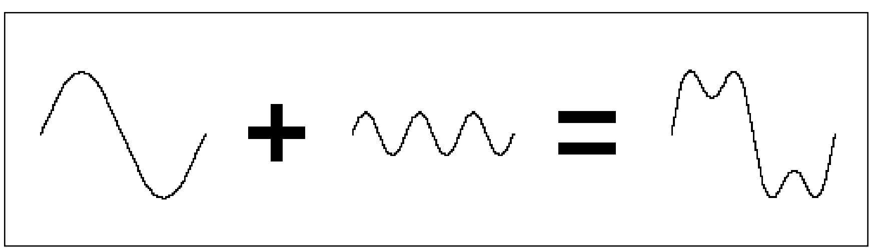
```
DYNAMIC PROCESSING
P00 both C#1 to C#5
Quick Zone:     off
 Select a Submodule
```

Use the left and right cursor buttons to select: Off, Pri, Sec or Both. Quick Zone remains on until you turn it off or until the ESI is rebooted.

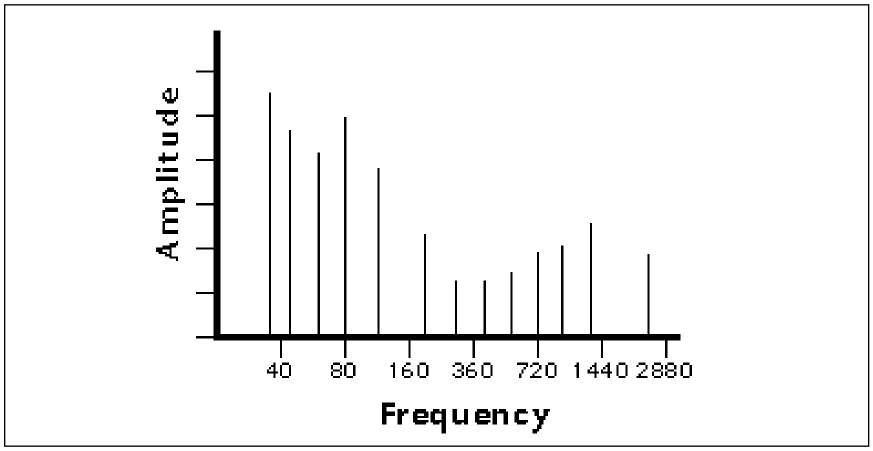
Filter Background

The ESI has 19 different filter types. These ultra powerful filters were originally developed for the Morpheus Z-plane synthesizer. In addition to Lowpass, Highpass and Bandpass filters, we have included Swept EQ filters, Phasers, Flangers and Vocal filters as well as a special bass distortion filter we call the “Bottom Feeder.” Before the actual filter descriptions, we have included a short section explaining how the different types of filters work.

To understand how a filter works we need to understand what makes up a sound wave. A sine wave is the simplest form of sound wave. Any waveform except a sine wave can be analyzed as a mix of sine waves at specific frequencies and amplitudes.

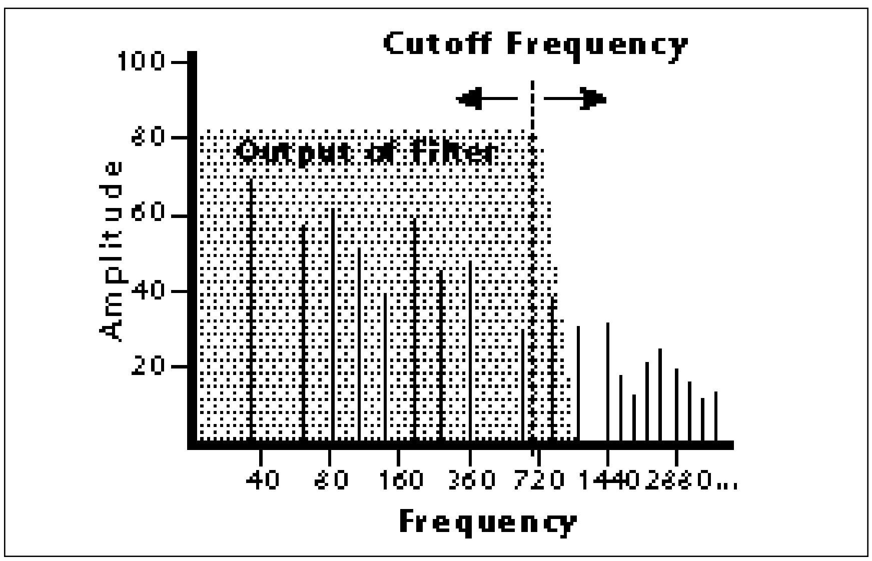


One way to represent complex waveforms is to use a chart with frequency on one axis and amplitude on the other. Each vertical line of the chart represents one sine wave at a specific amplitude.



Each vertical line of the chart represents one sine wave at a specific amplitude.

Most sounds are complex waves containing many sine waves of various amplitudes and frequencies. A filter is a device which allows us to remove certain components of a sound depending on its frequency. For example, a Low Pass Filter, lets only the low frequencies pass and removes only the high frequencies.

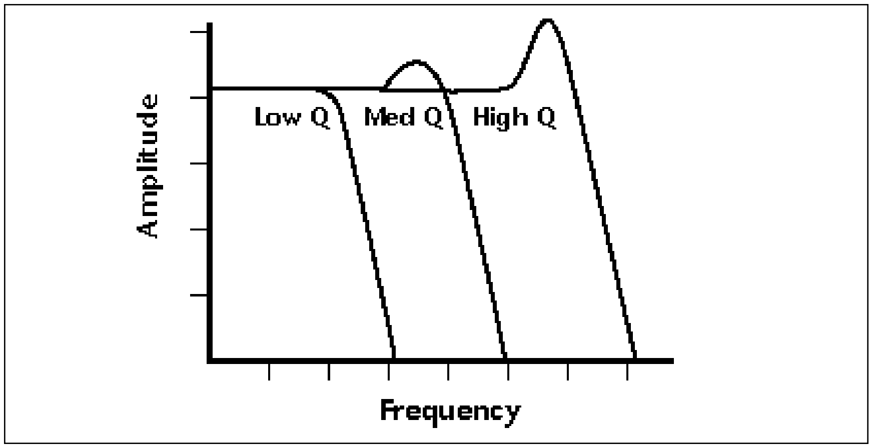


The point at which the frequencies are rejected is called the Cutoff Frequency (or Fc for short). A filter that lets the frequencies above the Fc pass is called a High Pass filter. Using a filter, we now have a way to control the harmonic content of a sampled sound. As it turns out, a low pass filter can simulate the response of many natural sounds.

For example, when a piano string is struck by its hammer, there are initially a lot of high frequencies present. If the same note is played softer, there will be fewer of the high frequencies generated by the string. We can simulate this effect by routing the velocity of the keyboard to control the amount of high frequencies that the low pass filter lets through. The result is expressive, natural control over the sound.

The auxiliary envelope generator is commonly used to control the cutoff frequency of the Z-plane filter. This allows the frequency content to be varied dynamically over the course of the note. Dynamic filtering, coupled with all the different instruments available, makes for endless possibilities in the final sound.

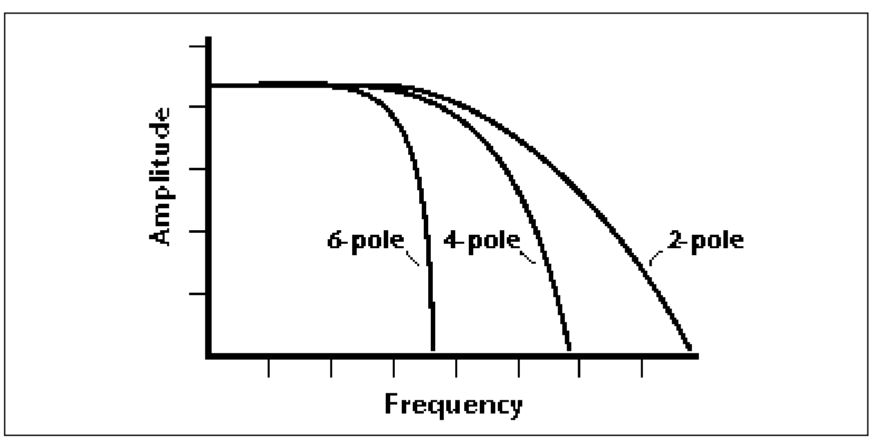
Another control on the filter is called Q, or resonance. On a lowpass or highpass filter, turning up the Q of the filter emphasizes the frequencies around the cutoff frequency. The chart below shows how different levels of Q affect the lowpass filter response.



Turning up the “Q” emphasizes the frequencies around the cutoff point.

In terms of sound, frequencies around the cutoff tend to “ring” with high Q settings. If the filter is swept back and forth slowly with a high Q, various overtones will be “picked out” of the sound and amplified as the resonant peak sweeps over them. Bells and gongs are real world examples of sounds which have a high Q.

Another important feature of a filter is the number of poles it contains. The lowpass filters can be either 2-pole, 4-pole or 6-pole filters. The highpass and bandpass filters can be either 2nd or 4th order filters, which is another way to describe the number of filter sections they contain. The number of poles in a filter describes the steepness of its slope. The more poles the steeper the slope, which in turn affects the sound. In general, the 2-pole filter will have a buzzier sound and the 4-pole filter has the classic low pass resonant filter sound. ESI's 6-pole low pass filters create a tight, modern sound.



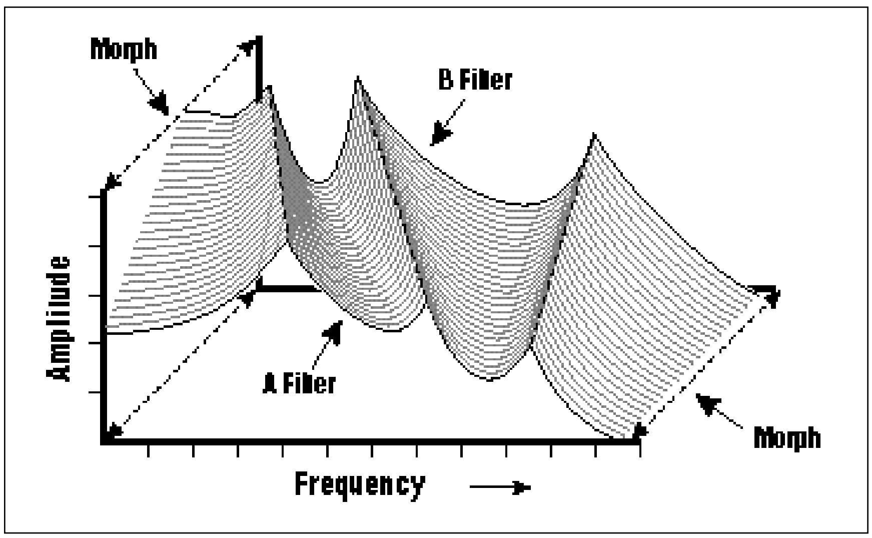
Parametric Filters

A more complex type of filter is called a parametric filter or Swept EQ. A parametric filter allows control over three basic parameters of the filter. The three parameters are: Frequency, Bandwidth, and Boost/Cut. The Frequency parameter (Labelled cutoff on the ESI filters) allows you to select a range of frequencies to be boosted or cut. The Bandwidth parameter allows you to select the width of the range. (Bandwidth and Frequency are combined into the Cutoff parameter in ESI's EQ filters.) The Boost/Cut parameter (Q on the ESI filters) either boosts or cuts the frequencies within the selected band by a specified amount. Frequencies not included in the selected band are left unaltered. This is different from a band pass filter which attenuates (reduces) frequencies outside the selected band.

The parametric filter is quite flexible. Any range of frequencies can be either amplified or attenuated. Often times, several parametric sections are cascaded (placed one after another) in order to create complex filter response curves.

The Z-Plane Filter

The Z-plane filter can change its function over time. In a simple Z-plane filter, we start with two complex filter types and interpolate between them using a single parameter. Refer to the diagram below.



The Z-plane filter has the unique ability to change its function over time.

Filters A and B represent two different complex filters. By changing a single parameter, the Morph, many complex filter parameters can now be changed simultaneously. Following along the Morph axis you can see that the filter response smoothly interpolates between the two filters.

This is the essence of the Z-plane filter. Using interpolation, many complex parameters are condensed down into one manageable entity.

Consider, as an example, the human vocal tract, which is a type of complex filter or resonator. There are dozens of different muscles controlling the shape of the vocal tract. When speaking, however, we don't think of the muscles, we just remember how it feels to form the vowels. A vowel is really a configuration of many muscles, but we consider it a single object. In changing from one vowel to another, you don't need to consider the frequencies of the resonant peaks! You remember the shape of your mouth for each sound and interpolate between them.

Filter morphing can be controlled by an envelope generator, an LFO, modulation wheels or pedals, keyboard velocity, key pressure, etc. The filter Cutoff parameter controls morphing on certain filters. The Q parameter can control various parameters such as boost/cut and mouth cavity size and of course, resonance or Q.

ESI Filter Types

2-pole Lowpass

Lowpass filter with 12dB/octave rolloff and Q control.

4-pole Lowpass

Lowpass filter with 24dB/octave rolloff and Q control.

6-pole Lowpass

Lowpass filter with 36dB/octave rolloff and Q control.

2nd Order Highpass

Highpass filter with 12dB/octave rolloff and Q control.

4th Order Highpass

Highpass filter with 24dB/octave rolloff and Q control.

2nd Order Bandpass

Bandpass filter with 6dB/octave rolloff on either side of the passband and Q control.

4th Order Bandpass

Bandpass filter with 12dB/octave rolloff on either side of the passband and Q control.

Contrary Bandpass

A novel bandpass filter where the frequency peaks and dips cross midway in the frequency range.

Swept EQ, 1-octave

Parametric filter with 24 dB of boost or cut and a one octave bandwidth. Cutoff controls center frequency and Q controls boost or cut.

Swept EQ, 2->1-octave

Parametric filter with 24 dB of boost or cut. The bandwidth of the filter is two octaves wide at the low end of the audio spectrum, gradually changing to one octave wide at the upper end of the spectrum. The filter Cutoff controls the center frequency and the Q parameter controls the boost or cut.

Swept EQ, 3->1-octave

Parametric filter with 24 dB of boost or cut. The bandwidth of the filter is three octaves wide at the low end of the audio spectrum, gradually changing to one octave wide at the upper end of the spectrum. The filter Cutoff controls the center frequency and the Q parameter controls the boost or cut.

Phaser 1

Creates a comb filter effect typical of phase shifters. Filter Cutoff moves the position of the notches. Q varies the depth of the notches.

Phaser 2

Comb filter with slightly different notch spacing than Phaser 1. Filter Cutoff moves the position of the notches. Q varies the depth of the notches.

Bat-Phaser

Phase shifter with peaks as well as notches. Filter Cutoff moves the position of the peaks and notches. Q varies the depth of the peaks and notches.

Flanger Lite

Contains three notches. Filter Cutoff moves frequency and spacing of the notches. Q increases flanging depth.

Vocal Ah-Ay-Ee

Vowel formant filter which sweeps from the “Ah” sound, through “Ay” sound to “Ee” sound at maximum Cutoff. Q varies the apparent size of the mouth cavity.

Vocal Oo-Ah

Vowel formant filter which sweeps from the “Oo” sound, through “Oh” sound to “Ah” sound at maximum cutoff. Q varies the apparent size of the mouth cavity.

Bottom Feeder

This is a specialized distortion filter, useful for adding punch and drive to low frequency sounds such as bass and drums. Set the Cutoff frequency low (less than 100) for best effect. Q has no effect on this filter.

ESI/E3X Lowpass

This is the original ESI-32 lowpass filter. It has been included to maintain backward compatibility. Banks created on the ESI-32 will use this filter by default.

0. Select Zone

This submodule lets you specify the keyboard zone to be processed.

1. Activate Dynamic Processing module.
2. Select Submodule Select Zone (0).
3. If there are both primary and secondary samples assigned to the preset, select whether the zone will contain both samples, just the primary, or just the secondary. Press ENTER. If the preset contains only primary or secondary samples, the ESI will bypass this screen and proceed immediately to step 4.

```
    SELECT ZONE
P00 both

 Select pri/sec/both
```

! Caution: If a zone contains more than one sample, the new original key will be assigned to all samples in the selected zone.

4. Select the lowest key of the zone and press ENTER. The default is the lowest note of the lowest sample. You can select a different low key in two ways. The Data Entry Control scrolls through the lowest key of each sample on the keyboard. (The Data Entry Control is the fastest selection method if you want the lowest key of the zone to coincide with the lowest note of a sample.) Or, you can use the keyboard to specify any note as the lowest note of the zone.

```
     SELECT ZONE
P00 both C#1

   Select Low Key
```

The second line shows the note being played on the keyboard (or scrolled with the Data Entry Control). After selecting a note, the third line will display the primary sample number, and the fourth line will display the secondary sample number associated with the note on line two.

5. Select the highest key of the zone and press ENTER. The default is the highest note of the sample that contains the previously specified low note. You can select a different high key in two ways. The Data Entry Control scrolls through the highest key of each sample on the keyboard. (The Data Entry Control is the fastest selection method if you want the highest key of the zone to coincide with the highest note of a sample.) Or, you can use the keyboard to specify any note as the highest note of the zone.

```
    SELECT ZONE
P00 both C#1 to C#5

  Select High Key
```

The second line shows the note being played on the keyboard (or scrolled with the Data Entry Control). After selecting a note, the third line displays the primary sample number, and the fourth line displays the secondary sample number associated with the note on line two. After pressing ENTER, the zone selection process is complete and the ESI returns to the Module Identifier.

1. Setup

This module establishes the controls for several functions, including tuning, delay before onset of a note, and chorus on/off.

1. Activate Dynamic Processing module.
2. Select Setup Submodule (1).
3. Move the cursor to the parameter(s) to be adjusted, and select the desired value(s) with the Data Entry Control. The display shows:

```
   DYNAMIC SETUP   →
Tuning:   +0.0 cents
Delay:        0.000s
Chorus:          off
```

- Tuning: varies the zone pitch over a range of -100 to +100 cents. (One hundred cents equals 1 semitone.)
- Delay: varies the time between when a MIDI Note On message is received and the onset of the note up to 1.53 seconds.
- Chorus: “thickens” the sound by doubling the primary sample and detuning the doubled sample somewhat. Chorus uses two channels. When chorus is on, the number of available channels in the zone will be cut in half. Chorus cannot be used with stereo samples.

APPLICATION: Creating an Alternate Tuning “Template.”

1. Tune each key by selecting it as a zone and then adjust the tuning.
2. To use the “template” with other samples, load in new samples to the existing sample numbers.
3. Return the samples to their true pitches using the Original Key function.

4. Select page two by pressing the right cursor button. The second page displays the following parameters for the selected zone.

! Caution: If there are no samples in the selected zone, the ESI will return to the beginning of the zone selection process.

```
←   DYNAMIC SETUP   →

Original Key:     E0
S01 Piano E1
```

- Original Key: allows you to change the original key of any samples in the selected zone.
- Change Sample Number: allows you to reassign any sample in the bank to the selected zone.

5. Select page three by pressing the right cursor button. The third page displays the following parameters for the selected zone.

```
←  DYNAMIC SETUP

Disable Loop:    off
Disable Side:    off
```

- Disable Loop: turns off the loop for any samples within the selected zone for the current preset and overrides any loop settings made in the Digital Processing module for the current preset.
- Disable Side: turns off playback of either the left or right side of a stereo sample within the selected zone.

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

2. VCA

The Voltage Controlled Amplifier (VCA) submodule contains two pages. Page one sets the overall zone level and stereo placement (pan). Page two sets the AHDSR envelope characteristics that alter VCA dynamics with respect to time.

1. Activate Dynamic Processing module.
2. Select submodule VCA (2).
3. Move the cursor to the parameter(s) to be adjusted, and select the desired value(s) with the Data Entry Control. The first page reads:

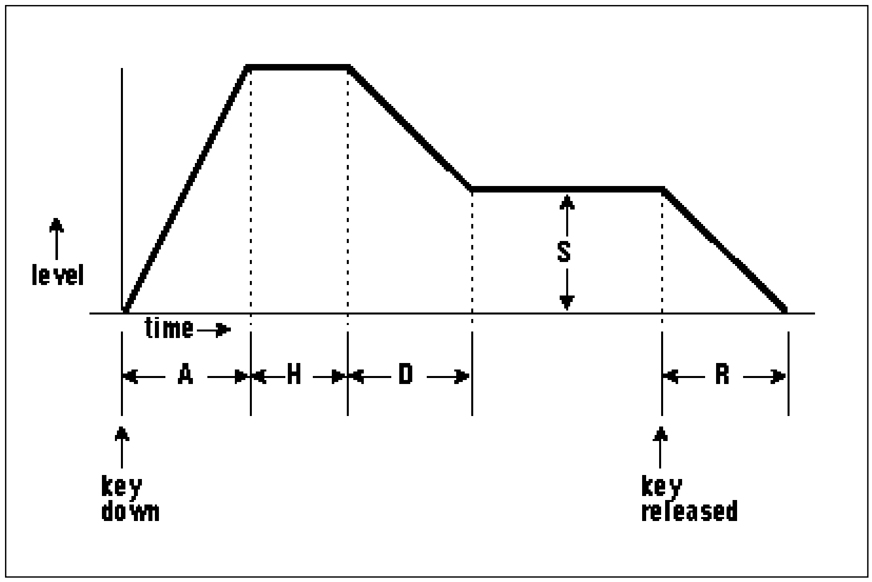
! Caution: When VCA Level is set to 100%, additional modulation from pedals, LFOs, etc. will not increase the level. You must reduce the VCA level from the 100% mark for additional modulation to have any effect.

```
        VCA        →
Level:          100%
Pan:          +   0%
L        |         R
```

- Level: is variable from -100% to 100%.
- Pan: is continuously variable from +100, where the zone appears at the right extreme of the stereo image; through 0, where the zone appears at the center of the stereo image; to -100, where the zone appears at the left extreme of the stereo image. The display graphically indicates the stereo position on the bottom line. The initial placement may have to be adjusted as modulation is applied to pan.

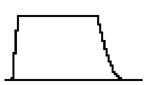
4. Select page two by pressing the right cursor button. The second page displays the following parameters for the VCA AHDSR envelope generator:

```
←VCA Attack:   0.00s
 Hold:          0.00s
 Decay:         0.00s
 Sus:100%  Rel: 1.04s
```

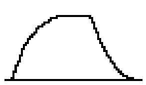


AHDSR ENVELOPE STAGES. If the key is released during the Hold or Decay stages, the Release stage begins.

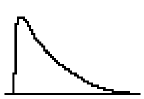
Piano



Organ



Strings



Percussion

The amplitude envelope shape gives important clues to the ear about what type of sound is being produced.

Refer to the following diagrams and definitions to understand how each parameter affects dynamics.

- Attack: varies the VCA envelope attack time from 0 to 163.69 seconds.
- Hold: sets the duration of the peak, from 0 to 21.69 seconds. If a key is held down longer than the hold duration, the decay phase will begin. If a key is released during the hold duration, the release phase will begin.
- Decay: varies the initial decay time from 0 to 163.69 seconds.
- Sus: (Sustain) varies the envelope sustain level from 0 to 100% of the peak level.
- Rel: (Release) varies the release time from 0 to 163.69 seconds.

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

3. VCF

The Voltage Controlled Filter (VCF) submodule contains three pages. Page one determines the filter, the filter's initial cutoff frequency and Q (resonance). Page two determines the effect of keyboard position on cutoff frequency, and the extent to which the associated AHDSR envelope affects the filter cutoff frequency (envelope amount). Page three sets the VCF AHDSR envelope characteristics. These alter timbre with respect to time.

1. Activate Dynamic Processing module.
2. Select VCF (3).
3. Move the cursor to the parameter(s) to be adjusted, and select the desired value(s) with the Data Entry Control. The first page reveals:

```
        VCF        →
Type: 4 Pole Lowpass
Cutoff:      22049Hz
Q:                0%
```

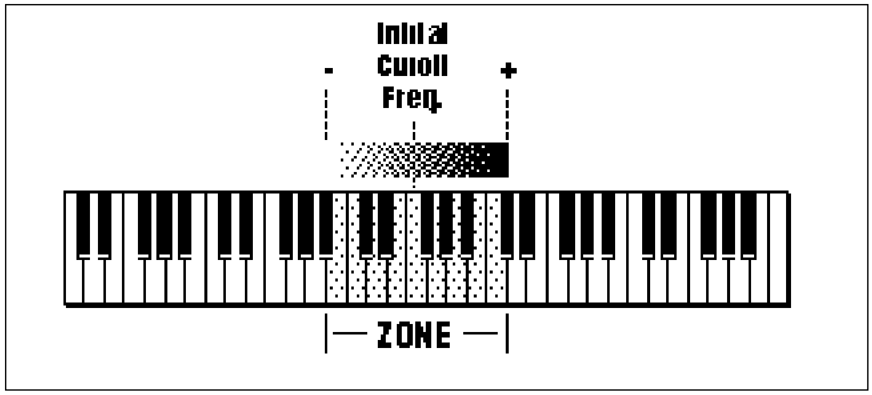
- Type: selects the type of filter used. Refer to the Filter Background section earlier in this chapter.
- VCF Cutoff: varies the filter cutoff frequency from 0 Hz to 22049Hz. Higher values correspond to higher filter cutoff frequencies, hence a sound with more treble. The maximum filter sweep range is most obvious with envelope amount set to +0.
- Q: varies the resonance from 0 to 100%. Higher values correspond to increased resonance, which accentuates the frequency response at the filter cutoff frequency. This produces a sharper, more whistling sound.

4. Select page two by pressing the right cursor button. The second page shows:

```
←       VCF        →

Tracking:      +1.00
Envelope Amt:  +  0%
```

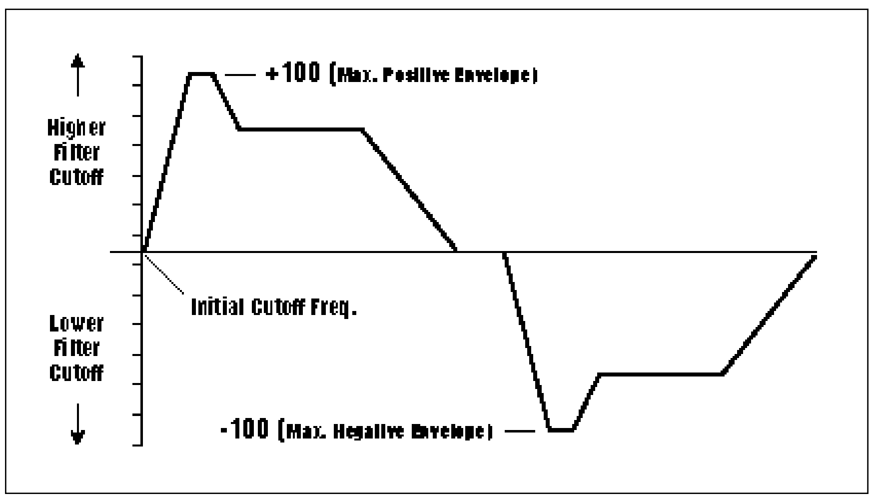
- Tracking: varies the filter cutoff with respect to the note(s) being played on the keyboard from -2.00 to +2.00. With 0.00 tracking, the filter cutoff will not be affected by the keyboard pitch. With tracking set to -2.00, the filter cutoff increases at one-quarter the rate at which pitch increases. With tracking set to +2.00, the filter cutoff increases at twice the rate at which pitch increases. Intermediate values give intermediate degrees of tracking. With negative tracking, a sample becomes progressively less bright as you play higher up in the zone. This is useful with some bass sounds, and to even out frequency response variations between samples when multi-sampling.



Keyboard Tracking varies the filter cutoff frequency as you play up and down the keyboard.

With 0.00 tracking, a sample's timbre remains constant over the selected zone. This generally gives the most realistic synthesizer and instrument sounds. With positive tracking, the sample's timbre becomes brighter as you play higher up in the zone.

- Envelope Amt: varies the effect of the filter envelope on the filter cutoff frequency from -100, which is the maximum inverted envelope, through +00, which has no effect from the envelope, to +100, which is the maximum positive envelope. It is usually necessary to raise the filter cutoff value when using inverted envelopes.



The filter cutoff frequency can be modulated with a positive or negative envelope amount.

5. Select page three by pressing the right cursor button. The third page displays the following parameters for the VCF AHDSR envelope generator:

```
← VCF Attack:   0.00s
 Hold:          0.00s
 Decay:         0.00s
 Sus:100%  Rel: 1.04s
```

- Attack: varies the VCA envelope attack time from 0 to 163.69 seconds.
- Hold: sets the duration of the peak, from 0 to 21.69 seconds. If a key is held down longer than the hold duration, the decay phase will begin. If a key is released during the hold duration, the release phase will begin.
- Decay: varies the initial decay time from 0 to 163.69 seconds.
- Sus: (Sustain) varies the envelope sustain level from 0 to 100% of the peak level.
- Rel: (Release) varies the release time from 0 to 163.69 seconds.

6. Press ENTER to exit the submodule. The ESI returns to the Module Identifier.

4. LFO

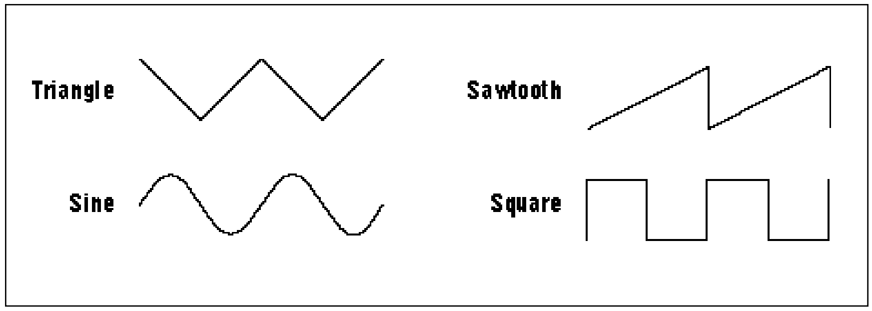
A Low Frequency Oscillator (LFO) is a wave which repeats at a slow rate. The LFO waveforms in ESI are Triangle, Sine, Square and Sawtooth. If using LFO to modulate the pitch of an instrument, the waveform determines the course of the pitch. For example, a Sine wave is smooth and therefore smoothly changes pitch. The Square waveform changes abruptly causing the pitch to change abruptly.

The LFO submodule contains three pages. Page one sets the LFO rate, shape of the LFO waveform, and the delay before onset of modulation. Pages two and three determine the degree to which the rate is varied as you play different notes (LFO variation), the LFO's destination(s): pitch (which produces vibrato), the VCF cutoff, the VCA amplitude (produces tremolo—attention surf music and Bo Diddley fans), and panning (spatial modulation). The amount of modulation sent to each destination is variable from 0 to 100%.

1. Activate Dynamic Processing module.
2. Select the LFO submodule (4).
3. Move the cursor to the parameter(s) to be adjusted, and select the desired value(s) with the Data Entry Control.

```
          LFO          →
Rate:            4.25Hz
Shape:         triangle
Delay:            0.00s
```

- LFO Rate: varies the LFO speed from 0.08 Hz to 18.14 Hz.
- LFO Shape: selects the waveshape of the LFO. The available LFO waveforms are: triangle, sine, sawtooth and square.
- LFO Delay: sets the amount of time between hitting a key and the onset of modulation. This simulates an effect often used by string players, where the vibrato is brought in only after the initial note pitch has been clearly established. The delay range is from 0 to 21.69 seconds.



LFO Waveforms.

4. Select page two by pressing the right cursor button. The second page shows:

! Caution: The LFO routing will be ignored if a realtime controller is assigned to the same destination. For example, if you have routed LFO->Pitch in this screen, and then route the mod. control to LFO->Pitch, the setting made in this screen will be ignored.

```
←        LFO           →
Variation:            0%
LFO->Pitch:           0%
LFO->Cutoff:          0%
```

- LFO Variation: provides a way to create the illusion of multiple players, each having an individual modulation rate. The depth of LFO modulation applied to each parameter is variable from 0 to 100%. With 0% variation, each key will have the same LFO rate. Increasing variation (to a maximum of 100%) alters the LFO rate for each key you play. The higher the number, the greater the variation in LFO rate. For effects such as ensemble playing, the variation feature is invaluable.
- LFO ->Pitch: produces vibrato effects.
- LFO ->Cutoff: varies the VCF's cutoff frequency in a cyclic fashion. This is useful for adding shimmering effects to a sound.

5. Select page three by pressing the right cursor button. The third page displays the following LFO routing options:

! Caution: When VCA Level is set to 100%, additional modulation from pedals, LFOs, etc. will not increase the level. You must reduce the VCA level (Dynamic Processing, 2) from the 100% mark for additional modulation to have any effect.

```
←        LFO

LFO->VCA:             0%
LFO->Pan:             0%
```

- LFO -> VCA: produces tremolo effects by altering the overall level of a zone in a cyclic fashion.
- LFO -> Pan: cyclically varies the placement of the audio output of a zone within the stereo field.

6. Press ENTER to exit the submodule. The ESI returns to the Module Identifier.

5. Auxiliary Envelope

An envelope is descried as a contour used to shape a sound over time. The Auxiliary envelope is a general purpose envelope that can be placed at any of several points in the Dynamic Processing signal path. This submodule contains two pages. Page one establishes the auxiliary envelope destination and amount, and page two determines the AHDSR envelope characteristics.

1. Activate Dynamic Processing module.
2. Select the submodule Auxiliary Envelope (5).
3. Move the cursor to the parameter(s) to be adjusted, and select the desired value(s) with the Data Entry Control. The first page shows:

```
  AUXILIARY ENVELOPE→

 Dest:              off
 Envelope Amt:       0%
```

- Destination: determines which parameter will be controlled by the envelope. These are: Pitch, Pan, LFO rate, or one of the four LFO destinations (LFO->Pitch, LFO->VCA, LFO->VCF, LFO->Pan). Using the auxiliary envelope to control pitch gives automatic pitch-bending. Controlling the LFO rate can increase (or decrease) the LFO speed over the duration of a note. Adding envelope control to an LFO destination allows modulation to fade in or out over the duration of a note.
- Envelope Amt: varies the depth of the envelope's effect on the chosen destination. It is variable from -100, the maximum inverted envelope, through +00 (no effect from the envelope), to +100, the maximum positive envelope. When using inverted envelopes, it is usually necessary to raise the LFO depth for the destination selected in step 5 of Dynamic Processing, 4. LFO.

4. Select page two by pressing the right cursor button. The second page displays the following parameters for the auxiliary AHDSR envelope generator:

```
←Aux Attack:  0.00s
 Hold:        0.00s
 Decay:       0.00s
 Sus:100% Rel: 1.04s
```

The envelope parameters are identical to the VCF and VCA envelopes.

5. Press ENTER to exit the submodule. The ESI returns to the Module Identifier.

6. Velocity To

Velocity data from your MIDI keyboard, sequencer or other controller can control any or all of nine different parameters, as selected on three pages in this submodule.

1. Activate Dynamic Processing module.
2. Select Velocity To (6).
3. Move the cursor to the parameter(s) to be adjusted, and select the desired value(s) with the Data Entry Control. The first page shows:

```
    VELOCITY TO    →
Pitch:          +  0%
VCA Level:      +  0%
VCA Attack:     +  0%
```

✶ Tip: Since each zone can have its own keyboard velocity settings, lower register bass sounds can have minimum dynamics to provide a constant bottom, while upper register lead sounds can be played more dynamically.

Values are adjustable from -100% to +100%.

- Pitch: ties velocity to pitch. With negative values, playing more forcefully lowers pitch. With positive values, playing more forcefully raises pitch.
- VCA Level: ties velocity to overall amplitude. At 0%, the overall level remains at the maximum level set in Dynamic Processing, VCA (2), and produces the loudest possible dynamics, no matter how forcefully or softly you play the keyboard. Progressively higher positive values give a progressively wider dynamic range by lowering the level as you play more softly (standard dynamics). Progressively higher negative values give a progressively wider dynamic range by lowering the level as you play more forcefully (reverse dynamics). Use velocity to add dynamics to sampled sounds that don't contain dynamics, such as samples from older synthesizers.

! Caution: When VCA Level is set to 100%, additional modulation from pedals, LFOs, etc. will not increase the level. You must reduce the VCA level (Digital Processing, 2) from the 100% mark for additional modulation to have any effect.

- VCA Attack: ties velocity to the VCA envelope attack time. With negative values, playing softly gives shorter attack times and playing more forcefully lengthens the attack. With positive values, playing softly gives longer attack times, while playing more forcefully shortens the attack. VCA attack is useful for string and horn sounds, where bowing or blowing softly produces a slower attack than rapid bowing or blowing, which produces a much faster attack.

✶ Tip: The VCA Attack settings interact with the VCA AHDSR attack time setting (Dynamic Processing, 2). You will probably need to fine tune the AHDSR attack time setting for optimum results.

4. Select page two by pressing the right cursor button. The second page shows:

```
←   VELOCITY TO    →
VCF Cutoff:       +0%
VCF Q:            +0%
VCF Attack:       +0%
```

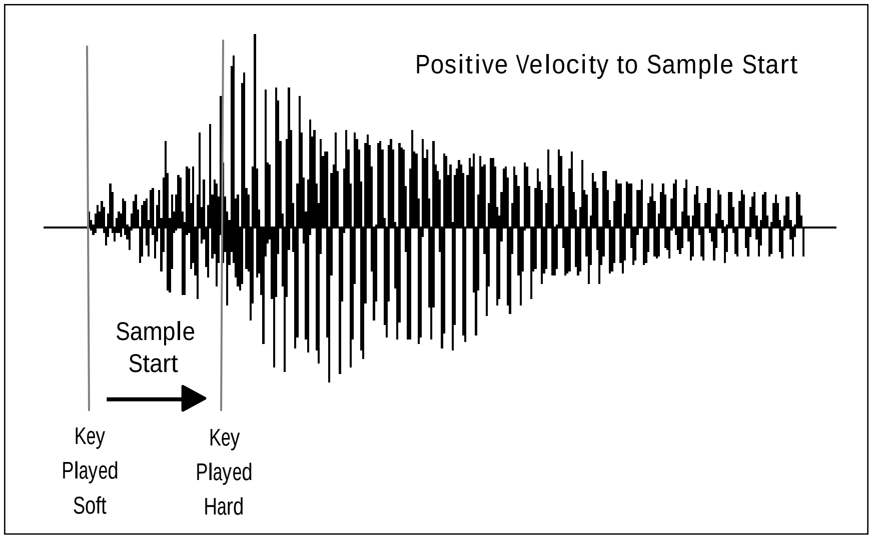
✶ Tip: The Cutoff and Q settings interact with the VCF controls (Dynamic Processing, 3). You will probably need to fine tune the VCF settings for optimum results.

- VCF Cutoff: ties velocity to the filter cutoff frequency. At 0%, the cutoff remains as set in Dynamic Processing, VCF (3), no matter how forcefully or softly you play the keyboard. Progressively higher positive values give a progressively wider cutoff frequency range by lowering the cutoff as you play more softly. Progressively higher negative values give a progressively wider cutoff frequency range by lowering the cutoff as you play more forcefully. Acoustic instruments often sound brighter when played forcefully. You can simulate this effect by tying VCF Cutoff to keyboard dynamics.
- VCF Q: ties velocity to filter Q. At 0%, the Q remains as set in Dynamic Processing, VCF (3), no matter how forcefully or softly you play the keyboard. Progressively higher positive values give a progressively wider Q range by lowering the Q as you play more softly. Progressively higher negative values give a progressively wider Q range by lowering the Q as you play more forcefully. Increasing Q thins out a sound, yet also increases its sharpness. Tying this to velocity can work well with percussive samples when you want a more intense, but not necessarily louder, sound. Overall volume levels can change with changes in Q; this is normal.
- VCF Attack: ties velocity to VCF envelope attack time. With negative values, playing softly gives shorter attack times, while playing more forcefully lengthens the attack. With positive values, playing softly gives longer attack times. And, as you might expect by now, playing more forcefully shortens the attack.

5. Select page three by pressing the right cursor button. The third page displays:

```
←   VELOCITY TO
Pan:             +   0%
Sample Start:    +   0%
Auxiliary Env:   +   0%
```

- Pan: ties velocity to stereo placement. At 0%, the Pan position remains as set in Dynamic Processing, VCA (2) no matter how forcefully or softly you play the keyboard. Progressively higher positive values shift the stereo image further to the right as you play more forcefully. Progressively higher negative values shift the stereo image further to the left as you play more forcefully. The higher the value, the greater the difference in stereo spread between soft and forceful keyboard playing.
- Sample Start: ties velocity to where the sample begins playing when you hit a key. At 0%, the sample plays normally no matter how forceful or soft you play the keyboard. Progressively higher positive values move the sample start point further towards the sample's end, thus cutting off the attack portion of the sample. Progressively higher negative values move the start point backward, beginning at the end of the sample.



The Velocity to Sample Start parameter allows you to change the attack characteristics of the sample with velocity. This technique is especially effective with percussion samples.

- Auxiliary Env: ties velocity to the Auxiliary Envelope's depth. At 0%, the effect of the Auxiliary Envelope remains as set in Dynamic Processing, Auxiliary Envelope (5) no matter how forceful or soft you play the keyboard. Progressively higher positive values increase the envelope depth as you play more forcefully. Progressively higher negative values decrease the envelope depth as you play more forcefully.

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

7. Keyboard Mode

This submodule alters the way the keyboard processes the notes (not the sounds) that you play. Options include a gate or trigger Envelope Mode, a monophonic Solo Mode, and a Nontranspose function for maintaining a constant pitch throughout a zone.

1. Activate Dynamic Processing module.
2. Select Keyboard Mode (7).
3. Move the cursor to the parameter(s) to be adjusted, and select on or off with the Data Entry Control. The first page reads:

```
    KEYBOARD MODE
Mode:         trigger
Solo:             off
Nontranspose:     off
```

✶ Tip: If you have a sound effect layered behind a melodic line, but don't want the effect to transpose as you play the melody, simply Nontranspose the sound effect.

- Env Mode: chooses between gate mode or trigger mode. In gate mode, the AHDSR envelopes react as described earlier. Holding down a key cycles through the AHDS stages, and releasing initiates the Release phase. In trigger mode, pressing a key, however briefly, cycles through the AHR stages and ignores the decay and sustain phases. Trigger mode is usually the best mode to use when triggering the ESI from external drum pads.

✶ Tip: Although the ESI version 3.00 is different from version 2.10 in that it is more "legato" in its envelope following, you can get a totally "un-legato" solo mode using one of the Mono Channel assignments. Mono Channel assignments always preserve the attack, even with the unlooped samples .

- Solo: provides the playing action of a monophonic synthesizer with single triggering and last-note priority. Solo mode produces more realistic effects when working with monophonic instrument sounds such as solo trumpet, flute, or sax, since this mode does not allow you to play a chord.
- Nontranspose: lets a sample play throughout its assigned zone at its original pitch only—there will be no transposition. This is useful when determining the original note at which a sample was recorded. Nontranspose is also useful if you're playing drum parts from the keyboard. Assigning a drum sound to a range consisting of several keys provides an easier target than being forced to hit a single key.

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

8. Realtime Control Enable

This function lets you to exempt specific realtime control destinations within a zone from being affected by the realtime controls. Off prevents the realtime control destination from being affected by the realtime controls. On (the default) allows the destinations to be affected by the realtime controls.

Example: Suppose you wanted the pitch wheel to bend only the upper half of the keyboard and not the lower half. Simply select the lower half of the keyboard as the current zone and turn Pitch Off in this screen. That's it!

1. Activate Dynamic Processing module.
2. Select Realtime Enable (8).
3. Move the cursor to the parameter(s) to be adjusted, and select On or Off with the Data Entry Control. The first page reads:

```
CONTROLLERS ENABLE →
Pitch:             on
VCF Cutoff:        on
VCF NoteOn Q:      on
```

4. Select page two by pressing the right cursor button. The second page controls the following realtime control destinations:

```
← CONTROLLERS ENABLE →
 LFO->Pitch:        on
 LFO->VCF Cutoff:   on
 LFO->VCA:          on
```

5. Select page three by pressing the right cursor button. The third page controls the following realtime control destinations:

```
← CONTROLLERS ENABLE
 VCA Level:         on
 Attack:            on
 Pan:               on
```

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

9. Channel Assignment

This function lets you to restrict the number of output channels assigned to a zone. The “Assign Group” system allows you to assign a certain number of output channels and designate a submix to each zone.

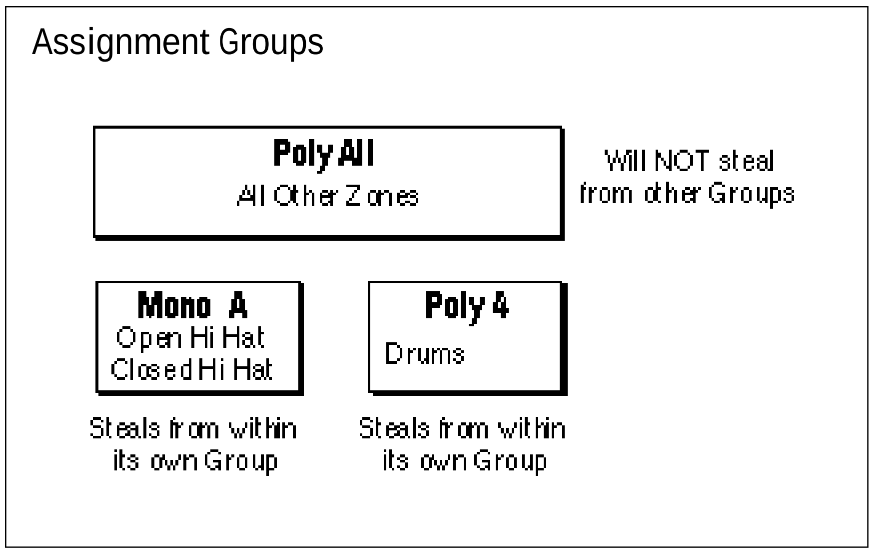
Zones are allowed to use output channels within their assigned group of channels only and do not interfere with other groups. Poly All is the default mode which allows free use of all output channels, but Poly All will not interfere with Zones assigned to other groups. By assigning zones in the preset to groups, important parts can be protected from being “stolen” if ESI's polyphony is exceeded. Alternatively, a zone, such as an open high hat, can be assigned to a mono channel bin so it will be cancelled by a closed high hat assigned to the same group.

The modes are as follows.

- Poly All: Poly All is the default group and is used when specific assignments are not needed. Notes are played polyphonically with dynamic channel assignment, using all channels.
- Poly 16 A-B: Two groups of 16 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 16 channels.
- Poly 8 A-D: Four groups of 8 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 8 channels.
- Poly 4 A-D: Four groups of 4 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 4 channels.

- Poly 2 A-D: Four groups of 2 channels each. Notes are played polyphonically with dynamic channel assignment, but using no more than 2 channels.
- Mono A-I: Nine monophonic channels. Any voices assigned to the same letter interrupt each other, but does not affect other voices.

As an example of how this is used, suppose you have a sequence containing a drum pattern. You want the drum pattern to remain solid and free from stolen channels no matter how complex the sequence. Furthermore, you want the closed high hat to cut off the open high hat to simulate a real drum kit.



Zones assigned to groups will only steal channels from within that same group. Alternatively, multiple zones can be assigned to the Mono Groups so that selective channel stealing will occur.

A. Select the zone for the open high hat and assign it to one of the Mono groups (Mono A). Assign the closed hi hat to the same group (Mono A).

B. Select the zone for the rest of the drums and assign it to the Poly 4 Assignment.

! Caution: If a plug is not inserted into a submix jack, any zone assigned to that jack is automatically re-routed to the main outputs.

▶ To Assign Zones to Assignment Groups:

1. Press the Dynamic Processing key.
2. Select Channel Assignment (9), The following screen appears.

```
CHANNEL ASSIGNMENT

AssignGroup:Poly All
Submix:         main
```

3. Move the cursor down to the third line and select an Assignment Group.
4. Move the cursor down to the last line and select a submix.
5. Press ENTER to exit the submodule. The ESI returns to the Module Identifier.