14 WaveLab Real-Time Plug-Ins

Delay

Delay effects usually simulate an acoustic environment. A grand example of a natural delay would be the Grand Canyon (or any canyon, for that matter). When you stand inside a canyon and clap your hands, it will take a moment for that sound to travel at 1,126 feet per second (343 meters per second) from your hands to the canyon wall and back to your ears. In nature, this is usually referred to an *echo*. In the electronics world, the terms *echo* and *delay* are synonymous.

Some natural environments (such as canyons) create very long delays, while others produce shorter delays, sometimes only a few milliseconds apart. If you sing in the shower, you'll probably enjoy a little delay during your aria.

Common Delay Parameters

All three of the WaveLab Delay plug-ins have the following settings.

Delay

This controls the time interval of the echoes. Large settings sound like canyons, while small settings sound more like your shower. The values range from 0.1 to 5,000 milliseconds (5 seconds).

Feedback

This sets how many subsequent echoes you will hear following the first echo. It simulates how many canyon walls there are or how many reflective surfaces there are in your shower. A large value would increase the subsequent echoes, or a value of 0 would allow only the first echo to be heard. Be careful when setting the Feedback higher than 50%, because the number of repeats can start to feed back on themselves, eventually becoming endless.

Lo Filter

This controls how many low frequencies will be contained in the echoes. The values range from 10 to 800 Hz. The lowest setting will ensure that the echoes will have the same low-frequency content as the dry signal. Higher settings will make the echoes sound thinner. Below the knob is a Lo Filter on/off button.

High Filter

This functions the same as the Lo Filter but will attenuate the high-frequency content of the echoes. The values range from 1.2 to 20 kHz. The highest setting will ensure that the echoes will have the same high-frequency content as the dry signal. Lower settings will make the echoes sound darker. Below the knob is a Hi Filter on/off button.

MonoDelay

All of the parameters of the MonoDelay (see Figure 14.1) are defined above.

Figure 14.1 The MonoDelay control panel.

StereoDelay

What could possibly make a Delay effect more fun? When there are two of them. The StereoDelay (see Figure 14.2) is two independent delay effects in one plug-in. Each delay has the same parameters as the MonoDelay and is individually programmable.

Figure 14.2 The StereoDelay control panel.

Pan

The left and right control panels don't necessarily correspond to the left and right audio channels. Rather, each delay can be distributed to both channels equally by setting the Pan control to the center position. Fully counterclockwise will make the echoes from that delay appear only in the left channel, while fully clockwise will send the echoes to the right channel.

PingPongDelay

This is a delay that will produce echoes that will alternate between the left and right audio channels. For example, if the Delay is set to 300 milliseconds, the first echo will appear 300 milliseconds later in the left channel and 600 milliseconds later in the right channel. It gets its name from watching table tennis if you were seated parallel to the net. See Figure 14.3.

Figure 14.3 The PingPongDelay control panel.

Spatial

This will control the width of the echoes between the left and right channels. Turning the Spatial control fully counterclockwise will make the PingPongDelay function identically to the MonoDelay. Fully clockwise will send the echoes hard left and then hard right, again and again, depending on the Feedback setting.

Dithering

Dithering is a very easy, yet very specialized process. Please refer to the "Intern" and "The Apogee UV22HR" sections at the end of this chapter for detailed descriptions.

Dynamics

Dynamics processors, like compressors and limiters, are designed to narrow the dynamic range of the audio material. Basically, they make the louds quieter while making the quiets louder. Dynamics processors usually have controls such as Threshold, Ratio, and Make-Up Gain. The easiest way for me to describe how a compressor works is by asking you to go get a rubber band. It doesn't have to be anything special, and you can use your imagination if you don't have a rubber band nearby.

Understanding Threshold

Begin by touching the sides of your extended index fingers together and place the rubber band around both fingers. The band should now be loose around your fingers. Now pull your fingers apart, but not far enough to make the rubber band taut. It should still be loose around your fingers.

Now pull your fingers apart so that the band becomes taut but offers no resistance. If you try to increase the distance between your fingers now, the band will start to offer an increasing amount of resistance.

How does this relate to audio? Let's complete the metaphor. The distance between your fingers is the audio volume. The taut rubber band offering no resistance is the compressor threshold. The resistance offered by the rubber band when you pull your fingers farther apart is the compressor in action.

In this example, as long as the volume didn't become too loud, the compressor didn't do anything. In other words, it offered no resistance below a certain threshold. Once the volume increased, the compressor offered an increasing amount of resistance.

The Threshold value is expressed in dB from −60 to 0 dB.

Understanding Ratio

The other parameter you'll find on a dynamic processor is Ratio. Let's use the rubber band metaphor to understand how it works. Pretend the rubber band isn't rubber, but made of air. (Yes, I know you can't make a rubber band out of air, but indulge me for a moment.) Because it's made of air, it can offer no resistance even when it becomes taut.

Now, magically return the band to its original rubber material. (You didn't know you were magic—or maybe you did!) Now the band offers resistance past the point where it became taut. But now double the rubber band around itself, and you'll have to use more force to move your fingers as far apart as you could before. The more times you double up the rubber band, the more force will be required to separate your fingers.

Using your magic powers again, make the rubber band out of titanium. Now when you pull the band taut (unless you have a big red S on your blue leotard), you cannot move your fingers beyond that point.

The degree of resistance that is offered by the band is the *ratio*. When it's made of air, it's a ratio of 1:1 (one-to-one). When it's rubber, the two stretching sides of the band make it 2:1. When the band is doubled over, it has four sides, therefore 4:1. When it's made of titanium, the ratio is set so high that it becomes a limiter.

To put this in audio terms, the first number of the ratio is the amount of volume in dB the volume will have to exceed the Threshold to add 1 dB of volume, which is the second number in the ratio. So, let's say your Threshold is set to −20 dB, and your Ratio is set to 2:1. If you exceed the Threshold by 2 dB, the compressor will restrain the level so that only 1 dB of that volume is allowed through. If you increase the volume level 4 dB past the Threshold, the compressor will allow 2 dB of volume through. Increase the volume 16 dB past the Threshold, and your compressor will allow only 8 dB through. That's a 2:1 ratio.

Let's increase the ratio 4:1. Now if you exceed the Threshold by 16 dB, the compressor will allow only 4 dB through. If you set it to 8:1, the compressor gives you only 2 dB.

The Ratio value is expressed in dB from 1:1 to 8:1 dB.

Understanding Make-Up Gain

This is not to be confused with the positive advantage of reconciling an argument with your significant other. Rather, because a compressor is restraining the output of the audio material, there needs to be a way to restore the level to an appropriate listening level. After you've added a compressor to the signal flow, you cannot use the (channel) volume fader to increase the level. To do so would be sending more signal into the compressor, where it would get restrained even more. So the volume adjustment must be made on the output of the compressor. In other words, you're making up the overall volume (gain) you lost by using the compressor. So while the Threshold and Ratio make the louds quieter, it's the Make-Up Gain that makes the quiets louder, thereby narrowing the dynamic range of the audio material.

Make-Up Gain is expressed in dB and has a range of 0 to +24 dB. Most of the Make-Up Gain controls on the WaveLab plug-ins will have an Auto button that will have the plug-in automatically adjust the amount of Make-Up Gain based on how much volume is being restrained. If the Auto control is enabled, the knob will be grayed out and non-adjustable.

Common Dynamics Parameters

We've already talked about Threshold, Ratio, and Make-Up Gain. These are controls found mainly on compressors and limiters. But there are some other parameters that are common to most Dynamics processors.

Knee

Some Dynamics processors have a Soft Knee or Hard Knee setting. This determines the shape of the dynamic curve. That curve is known as a *knee* because it looks like a human knee. (This will be more apparent when looking at a compressor control panel.) A Hard Knee setting makes the Dynamic control abrupt. Most limiters use a Hard Knee setting by default. A Soft Knee makes the Dynamic control smoother and more natural-sounding.

Input VU Meter

The Input VU meter will allow you to monitor the volume of the audio material as it enters the plug-in during playback. Low levels might indicate quiet passages but could also inform you that the audio material might be too quiet for a Dynamics processor to function properly. The meter is represented in dB, with −60 dB at the bottom and 0 dB at the top.

At the bottom of the meter is a numeric indicator of the loudest volume received at the plug-in's input. Normally, the value will be a negative number in light blue. If you notice that the numeric value becomes red (thereby becoming a positive number), that indicates that the maximum input level of the plug-in has been exceeded and will be adding unwanted distortion to your audio material. If this happens, you should adjust the audio material volume level. The values will hold their values even after you stop playback, but they will be reset the instant you restart the playback. You can also reset the indicator at any time by clicking on it.

Output VU Meter

After the Dynamics processor has adjusted the volume level, the Output VU meter will show you how much signal is being delivered at the plug-in output. Low Output readings might indicate that your Dynamics processor is reducing the volume by a large degree, and therefore a larger amount of Make-Up Gain might be required to bring the volume back to a nominal level. The meter is represented in dB with −60 dB at the bottom and 0 dB at the top.

Just like the Input VU meter, you will find the numeric indicator at the bottom of the Output VU meter. It functions the same as the Input VU meter numeric indicator except that it monitors the Output meter. If you notice that the numeric value becomes red, thereby clipping, you will need to lower your Make-Up Gain to ensure that the Dynamics processor isn't adding unwanted distortion.

GR (Gain Reduction) Meter

This is a somewhat confusing meter because it works backwards from the other meters. That's because the GR meter displays how much the Gain is being reduced. Small amounts of reduction indicate that the Threshold is not being exceeded very much or that the ratio is set to a low value, such as 1.5:1 or 2:1. Therefore, very little Dynamics processing is taking place. Conversely, large amounts indicate that the Dynamics processor is reducing the volume aggressively, either by a low Threshold setting or a high Ratio setting. The meter is represented in dB, with 0 dB at the top and −60 dB at the bottom.

Just like the Input and Output meters, you will find the numeric indicator at the bottom of the GR meter. This will always be represented by a light-blue negative integer. Although you can't clip the GR meter, large values might indicate overly compressed audio material. The rule of thumb here is that unless you're going for a pronounced compressor effect on purpose, if the sound of the compressor is obvious, you may need to relax the Threshold and/or Ratio settings.

The Graph

The Input, Output, and GR VU meters surround the Graph. The best way to think of it is as an Input versus Output Graph. The vertical axis is the input of the compressor, and the horizontal axis is the compressor output. The blue diagonal wedge represents the character of compression as well as the Threshold and Ratio settings. The light-blue Graph lines represent 10s of dB from −60 to 0 dB. We'll discuss how to read the Graph when we talk about the Compressor plug-in in a moment.

Attack

This is the amount of time that will pass before the Dynamics processor engages the audio material. Smaller values will engage the Dynamics processor earlier, and longer values will allow more of the unaffected signal to pass through the plug-in before processing. In the case of a compressor, larger values make the audio material sound punchier, while larger values will make a Gate allow signal to pass more gradually. The value is measured in milliseconds and usually ranges from 0.1 to 100 ms. The Gate plug-in has an extended Attack range of up to 1,000 ms (1 second).

Hold

This is how long after the Threshold has been exceeded that the Dynamics processor will continue to effect the audio material. Short values can make the Dynamics processor sound bumpy, because the Hold will be processing the signal above the Threshold more often. Longer values will smooth the sound by processing the signal less frequently. The value is measured in milliseconds and usually ranges from 0.1 to 2,000 ms (2 seconds); however, the Compressor plug-in has an extended range of 5,000 ms (5 seconds).

Release

This is how long the Dynamics processor continues to affect the audio material once the volume drops below the Threshold. Short values will return the volume to its pre-processed level more quickly but can make the audio material sound as if it's being turned up quickly during quiet passages. This is an effect referred to as *breathing*. Longer values will leave the Dynamics processor engaged for a longer time when the volume drops below the Threshold. That will make the introduction of any abrupt audio material pre-processed, because the Dynamics processor will still be engaged. In other words, the rubber band will still be stretched.

Most of the Release controls will have an Auto button that will adjust the Release constantly by analyzing the downward taper of the audio material as it drops past the Threshold. This is a set-it-and-forget-it control and usually provides very musical and natural-sounding results.

Live (Rhymes with Five) Button

Some Dynamics processors have this button. One of the biggest advantages to plug-in Dynamics processors is that they can look ahead to the oncoming audio material and be prepared for it when it finally arrives. Hardware processors cannot make a look-ahead calculation because they cannot read the future, as plug-ins can. The Live button is disabled on the plug-in by default to exploit this advantage. The disadvantage is that the processor will introduce a larger degree of latency into the signal path. This usually isn't a problem unless you're recording into WaveLab and through a Dynamics processor. (See Chapter 15.) In that case, you'd want to disable the look-ahead capability of a Dynamics processor by turning on the Live button.

Compressor

The Compressor plug-in (see Figure 14.4) has all of the parameters we've already discussed. However, you will need to learn about a few other features of this and other Dynamic processors—specifically, the Graph.

Figure 14.4 The Compressor control panel.

Reading the Graph

As I mentioned earlier in this section, this is the Input versus Output Graph, with Input on the vertical axis and Output on the horizontal axis. Figure 14.4 shows a Threshold setting of −20 dB. That is represented on the Graph by the lower white dot on the blue wedge. You'll notice that white dot is positioned on the second blue Graph line, also indicating a Threshold of −20 dB. Adjusting the Threshold will move that dot on the Graph. (You can also click and drag the dot itself to adjust the Threshold.)

At the top of the blue wedge is another dot that represents the Ratio setting. You can alter the Ratio by turning the knob or clicking and dragging the Ratio dot. In Figure 14.4, the Ratio is set to 2:1, so with a Threshold of −20 dB, the Ratio dot is located at the −10 dB Graph line. Adjusting the Threshold will also adjust the position of the Ratio dot but not the ratio itself.

Here's how to use the Graph most effectively. The Input VU meter relates to the Graph lines. If the Compressor is configured as it is in Figure 14.4, watching the Input meter will help you determine where the Threshold should be set. If the Input meter is bouncing past the −20 dB Graph line frequently, then −15 or −20 dB would be appropriate places to set the Threshold. Lower input levels would require a lower Threshold for the compressor to be effective.

The Output meter also relates to the Graph lines. Using Figure 14.4 as an example, if the Output meter is getting clipped, you might need to increase the ratio so that more volume above the Threshold is being reduced. It might also mean you need to adjust your Make-Up Gain.

Finally, GR meter also relates to the Graph lines. If you see a large amount of Gain Reduction (from the top of the Graph down) in the −15 to −30 dB range, it means your compressor is working really hard and reducing a lot of volume. You'll also probably be able to hear the over-processing being created by the compressor. In that case, you should refine your compressor Threshold and Ratio settings.

Analysis

Adjusting the Analysis will determine whether the Compressor will process the audio material based on its Peak volume or Average (RMS) volume, or anywhere in between. Underneath the knob you can see Peak – RMS listed on the control panel. Turning the knob fully counterclockwise will set the Compressor to process peak volume, while fully clockwise will process only RMS (Average) volume. But you can set the control anywhere in between for a less stringent analysis. Basically, a peak-leaning analysis works better for audio material with strong attack transients, such as drums or piano, while an RMS-leaning analysis will work better for audio material with more gentle attacks, such as vocals, horns, or bowed string instruments.

VintageCompressor

Now that we've discussed Threshold, Ratio, and Make-Up Gain at great length, here's a Compressor that has none of those controls. That's because the VintageCompressor (see Figure 14.5) has its controls preset to mimic the response of a vintage tube or transistor compressor. Because the Threshold and Ratio are preset, it also doesn't have a Graph.

Figure 14.5 The VintageCompressor control panel.

Input

Because the Threshold and Ratio of the VintageCompressor are preset, increasing the Input control will push the volume of the audio material past the preset Threshold. Doing so probably will instantly push the output of the plug-in into clipping. That's why I've always adjusted the Output control down about −6 dB before adjusting the Input control. Don't worry about clipping the input of the Vintage Compressor. The preset nature of the plug-in will reduce this possibility.

Output

This is the Output of the VintageCompressor. Because the Make-Up Gain is preset, this control simply sets the plug-in output. Lowering the output will allow you to set the Input control higher to introduce more compression. You should take care to set it so that the Output of the plug-in is never clipped (the red numeric indicator at the bottom of the Output VU meter).

Punch Button

Tube and transistor compressors are not known for having lightning-fast responses to audio material with fast attack transients. The old vintage models like this plug-in is modeled after were known for having very musical qualities when processing fast, loud transients. So even though the VintageCompressor has an adjustable attack, turning on the Punch will mimic the musical quality of a vintage hardware compressor.

MultibandCompressor

One of the limitations of a standard or single-band compressor is that full-range audio material can confuse the Threshold to be triggered at an inappropriate time. For instance, if there was a sudden increase in low frequencies, the Threshold would be triggered, and the compressor would start reducing *all* of the volume, regardless of the frequency range. That's why a MultibandCompressor (see Figure 14.6) becomes the most powerful processor in your mastering toolbox.

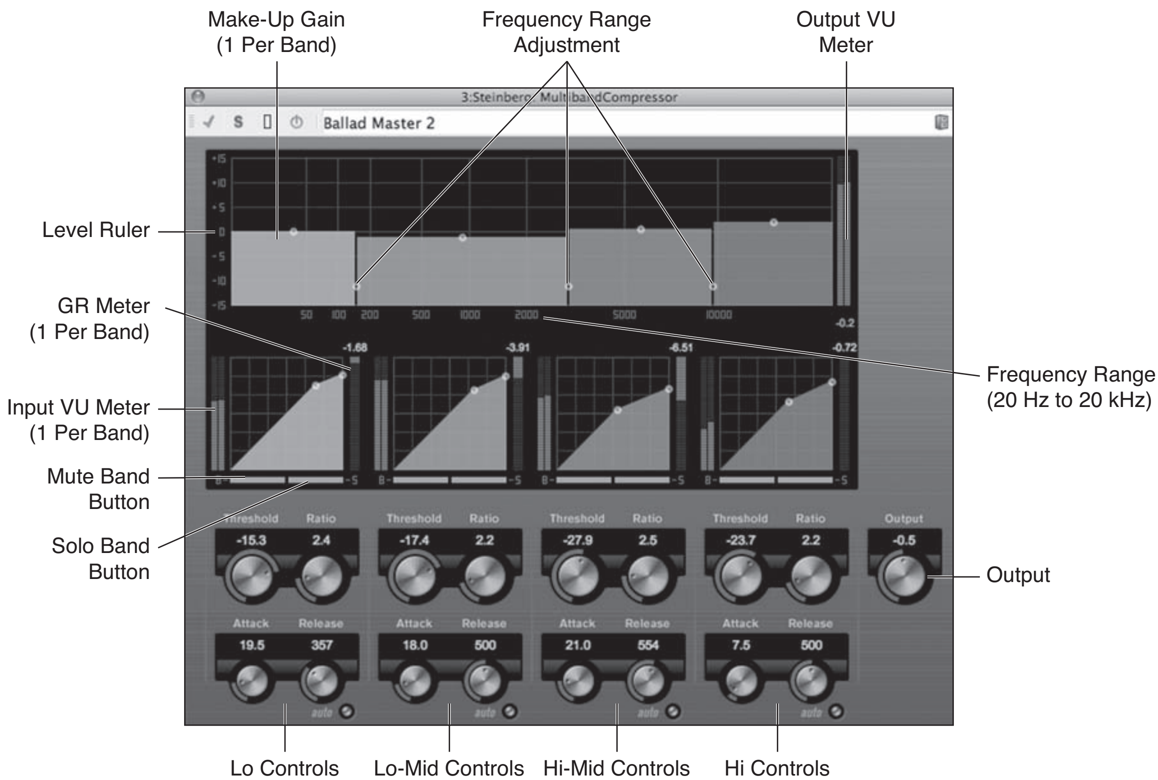

Figure 14.6 The MultibandCompressor control panel.

The MultibandCompressor is made up of four independent compressors with definable frequency bands (ranges). Each compressor has its own Threshold, Ratio, Attack, and Release settings. This will allow you to process each frequency band with its own customized compression settings. That way, the lo compressor won't affect any of the other compressors. The bands are divided into lo, lo-mid, hi-mid, and hi frequency ranges.

The MultibandCompressor was also discussed in Chapter 6. So for now, I'm going to describe only the controls we haven't learned about to this point.

Mute Band Button

Each of the bands has a Mute Band button. Clicking this button will disable the band compressor. When muted, the band controls will be grayed out, and the button will be yellow. Signal will still pass through the rest of the band compressors normally. You can have multiple bands muted simultaneously.

Solo Band Button

Each band has its own Solo Band button. Clicking this button will prevent signal from passing through the other band compressors, allowing you to listen to the processing of an individual band compressor. The Solo Band button will be red when activated, and only one Solo button can be engaged at a time. (The appearance of the other bands will not be altered in any way when a Solo Band button is activated.)

Frequency Range Adjustment

Between the frequency bands is a small white dot. Clicking and dragging this dot to the left or right will alter the bandwidth between compressors. This will allow you to adjust each compressor to work most appropriately with the ranges of audio material.

VSTDynamics

The VSTDynamics plug-in (see Figure 14.7) is actually three Dynamics processors wired in series. It is composed of Gate, Compressor, and Limiter modules. The

Figure 14.7 The VSTDynamics control panel.

advantage to this plug-in is that the modules can interact with one another. Another advantage is that all three modules can exist inside of one Master Section effects slot.

Because most of the controls on the VSTDynamics plug-in are identical to the individual components from which this plug-in is built, please refer to the Gate, Compressor, and Limiter plug-ins in the chapter for more detail. However, this plug-in does not include Analysis and Live buttons for the Gate and Compressor. Also, because the Limiter Input is controlled from the Output level of the Compressor, there is no Limiter Input knob.

Module Configuration Button

In the lower-right corner of the plug-in is the Module Configuration button. Repeatedly clicking this button will rearrange the order of the modules. Because they're wired in series (one into the next, into the next), different orders can produce a variety of results. The possible orders are the default of Gate > Compressor > Limiter, Compressor > Limiter > Gate, and Compressor > Gate > Limiter.

DeEsser

The DeEsser (see Figure 14.8) is a specialized type of compressor designed to remove vocal sibilance. That's the S sound in human speech. For example, the word Mississippi has three S sounds. Sometimes the S sound can be overbearing. You could use an EQ to notch out those frequencies, but the constant muting of those frequencies would compromise the sound of the recording when no S sounds exist. Instead, the DeEsser monitors the frequencies where excess sibilance usually occurs and compresses those narrow frequency bands.

Figure 14.8 The DeEsser control panel.

Reduction

This is the amount of S attenuation offered by the DeEsser. The values are depicted in dB. Higher values will reduce the S sound more aggressively.

Auto Threshold

This is the only WaveLab Dynamics processor that has an Auto button for Threshold. When activated, the processor will constantly monitor the audio material for the S sound and set the Threshold automatically.

Other Uses for the DeEsser

Every so often, a client will bring me a mastering project in which the hi-hat cymbals have been mixed inappropriately loudly. Because the S sound and hi-hats share a similar frequency band, I've actually had really good luck using the DeEsser to reduce hi-hat and other cymbal volumes.

Limiter

A limiter (see Figure 14.9) is similar to a compressor except that the Ratio is usually set extremely high. This is known as a *brick wall*. In other words, a limiter will radically clamp the volume of the audio material as if it ran into a brick wall and never let the volume exceed the Limiter Output setting. Or, to use my rubber band metaphor, this is when you turn the band into titanium. (Or remember the old guy guarding the bridge in *Monty Python and the Holy Grail*? His famous line was, "None shall pass." Well, if King Arthur is the audio material, the old guy is the limiter.)

Note: If you want a true brick-wall limiter, use the Maximizer or Peak Master instead.

Figure 14.9 The Limiter control panel.

The Input and Output controls work identically to those on the VintageCompressor plug-in. But in the Limiter, the preset Threshold and Ratio are much more extreme.

Overcooking a Limiter

If you use a limiter to raise the average volume level beyond a certain level, you'll really start to hear the limiter. That's not a good thing. It's commonly known as *overcooking*. There's only so loud you can raise the average volume before it starts to fall apart. Keep that in mind when a client continually pesters you for "More volume! MORE volume! MORE, MORE VOLUME!"

Maximizer

The Maximizer (see Figure 14.10) is another type of limiter, but it is more adept at raising the average volume levels of audio material, thereby increasing the perceived volume. Another term for the Maximizer is a *mastering limiter*. Like a limiter, it will also prevent volumes from exceeding the output setting.

Figure 14.10 The Maximizer control panel.

The Maximizer was also discussed in Chapter 6. So for now, I'm going to be describing only the controls we haven't learned about to this point.

Optimize

This is an arbitrary value, but raising this value will increase the average volume level of the audio material. Just like any limiter, setting the volume too loud will overcook the sound, even when the level is guaranteed not to exceed 0 dB.

Soft Clip Button

When enabled, the Soft Clip will relax the limiting of the peaks. It results in a more vintage or tube-like response from the Maximizer. I usually have Soft Clip enabled. However, if my client is asking for more volume, I'll turn this off so that the Maximizer offers a more mathematically strict increase in perceived volume.

Gate

A Gate (see Figure 14.11) is similar to a compressor, except that instead of reducing the volume that crosses the Threshold, a Gate will not allow sound to pass through until it crosses the Threshold. This is a great tool for automatically reducing background noise.

Figure 14.11 The Gate control panel.

The Threshold control is at the upper-left corner of the control panel. You would set the Threshold right above the level of the background noise so that only signal louder than the background noise will be allowed through.

State Light

The State light has three colors, similar to a traffic-control light. Red indicates that the Gate is closed and not letting any audio through. Green indicates that the Gate is wide open and not impeding the flow of audio material. Yellow indicates that the Gate is partially open. This occurs when the audio level is hovering around the Threshold level.

Side Chain Controls

Enabling the Side Chain will install a filter into Gate. That filter will prevent the Threshold from being triggered by non-essential frequencies. You'd start by choosing a filter type: LP (Low Pass), BP (Band Pass), or HP (High Pass). The Center control sets the frequency of the filter and is variable between 50 Hz and 20 kHz. The Q-Factor sets the bandwidth of the Center frequency. The Monitor button allows you to hear the audio material as the filter is affecting it.

A practical example of using the Side Chain controls would be recording a keynote speaker in a hotel that happens to be under renovation. The large trucks outside might be making loud rumbling noises that falsely trigger the Gate Threshold. I would select an HP filter and set the Center to 250 Hz and the bandwidth to 1.0. That way, only frequencies above 250 Hz (such as human speech) would trigger the Threshold.

Expander

Take a compressor and flip it upside down. What you'll end up with is an Expander (see Figure 14.12). Instead of reducing volumes above the Threshold, an Expander reduces volumes below the Threshold. In other words, it makes quiet volumes even quieter.

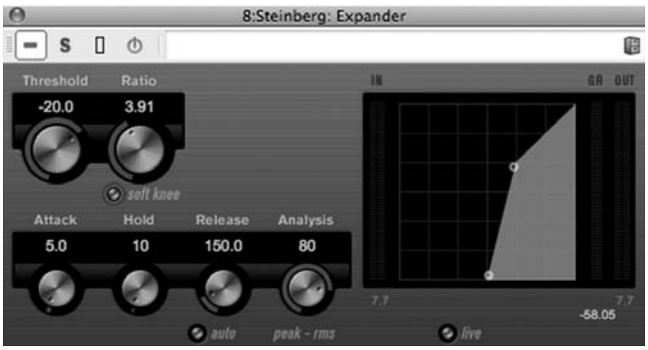

Figure 14.12 The Expander control panel.

Reading the Expander Graph

Compared to the Compressor Graph, the Expander Graph literally has been turned on its head. The Ratio is now the lower dot on the graph, and the Threshold is the upper dot. In an Expander, the Ratio controls how much quieter the audio material will be below the Threshold. The Threshold itself will determine the point where the audio material will be unprocessed by the Expander.

For example, the settings in Figure 14.12 show the Expander with a Threshold of −20 dB and a Ratio of almost 4:1. That means that an input of −25 dB (−5 dB below the Threshold) would be attenuated to a level of −45 dB. That's the level below the Threshold times the Ratio and subtracted from the amount of input: $(-5\times4)-25$.

EnvelopeShaper

The EnvelopeShaper (see Figure 14.13) is a really interesting and useful plug-in. It's very similar to an ADSR on a synthesizer amplifier section. But instead of controlling the Attack, Decay, Sustain, and Release of a synthesizer sound, the EnvelopeShaper crafts the Attack, Length, and Release of audio material.

Figure 14.13 The EnvelopeShaper control panel.

The EnvelopeShaper can make percussive audio material (such as drums and piano) sound even more percussive. But you can also craft the Attack setting to let sound rush in more slowly. On a piano, for example, this would create bowed-piano effect (as if the piano was being played with a violin bow).

Even though some of the controls are named identically to other Dynamics processors, the unique nature of the EnvelopeShaper requires that we discuss each one.

Attack

This will control the slope of the audio material volume as it comes into the plug-in. It is measured in dB from –20 to 20 dB. A setting of 0 dB would leave the initial attack of the audio material unaffected. Negative settings will let the sound rush in slowly, while positive values will increase the level of the initial attack.

Length

This is how long, in milliseconds, the Attack will continue to alter the volume. Its range is 5 to 200 milliseconds. Lower values will make the Attack more pronounced, while higher values will relax the Attack.

Release

When the volume drops below –20 dB, the Release controls how quickly the sound will fade away. It is measured in dB from –20 to 20 dB. A setting of 0 would leave the fade-out unaffected, leaving the original audio material unaffected. Negative values lower the fade-out volume, making the sound fade out more quickly. Positive values raise the fade-out volume, making the sound fade out more slowly.

Output

This is the Output level of the plug-in. Because the EnvelopeShaper is capable of adding as much as 20 dB to the Attack of the audio material, it's easy to clip the Output. Keep your eye on the numeric indicator at the bottom of the Output VU meter to make sure it never turns red. If the does, the Output of the plug-in has been clipped, and you should lower the Output so that it does not introduce unwanted distortion.

Reading the EnvelopeShaper Graph

The Attack, Length, and Release controls are also represented by the small white dots on the Graph. You can adjust their values by clicking and dragging the dots.

The graph itself is a representation of the shape of the EnvelopeShaper. But underneath the bright-blue Envelope Graph is a darker-blue Graph that indicates the natural envelope of the audio material. This will help you to visualize the effect of the Envelope-Shaper versus the original envelope.

Distortion

Now that we have inexpensive digital audio editors that can provide us with pristine sonic quality and character, here's a plug-in that can take all that away (see Figure 14.14). I always thought it was odd that grunge music came into vogue during the birth of the digital recording revolution. All of a sudden, every grunge band wanted distortion effects on their vocals. After spending a lifetime dreaming of having a digital recording studio, void of unwanted distortion, I was getting paid to add distortion to vocal tracks.

Figure 14.14 The Distortion control panel.

Boost

This is the amount of distortion that will be added to the audio material. It's an arbitrary value from 0 to 10. (What, no 11?)

Feedback

This will route some processed signal back into the Boost control, creating more distortion. It's another arbitrary value from 1 to 10.

Tone

This will craft the tonal characteristic of the distortion. Lower values will darken the distortion, giving it a warmer tube-like quality. Higher values will boost the high frequencies, making the distortion fuzzier.

Spatial

If you're working with stereo audio material, the Spatial control will add separation between the left and right audio channels, creating a wider stereo effect. When you're working with mono material, the control will not add any Spatial effect. If you want to add the effect to a mono file, copy the mono channel into both channels of a new stereo file.

EQ (Equalization or Tone Control)

EQ is a tool for crafting the tonal characteristics of audio material. Most of the adjectives used to describe a sound relate to its tone. For example, if someone describes a sound as *boomy*, that means it has a lot of low-frequency energy. Rich means an abundance of lo-mid frequencies. Bright means a lot of hi-mids. Airy means a lot of highs. By crafting the tone during the mastering of audio material, you can make it have a smoother and more even tone.

Understanding EQ will involve a little discussion of emphasis/de-emphasis and psychoacoustics. When increasing the volume of a frequency range, you are increasing its emphasis. Conversely, when you decrease the volume of a frequency range, you are decreasing the emphasis. This is known as *de-emphasis*.

The way the human ear perceives emphasis/de-emphasis is related to psychoacoustics. For example, if you emphasize the high frequencies, the listener might perceive the change as a de-emphasis of low frequencies. When a client asks me to make the mastering sound warmer, I don't necessarily emphasize the lows and lo-mids. Rather, I'll de-emphasize the hi-mids and highs. For that reason, EQ should always be considered an emphasis/de-emphasis tool. In the other words, the controls can be turned up *and* down. Because we've grown up with the concept of more is better, new users tend to only turn things up, up, and up.

Visualizing EQ Emphasis/De-Emphasis

One of the best ways to see what any of the WaveLab EQs are doing to the sound is to use the Signal Generator (refer to Chapter 12) to create a white noise file. I recommend setting the Single Generator Global Gain control to −12 dB before creating the file. Then put that file in loop playback by pressing / (forward slash). Next, load the GEQ-10 into Master Section Effects Slot 1, the GEQ-30 into Slot 2, and the StudioEQ into Slot 3. With the Spectroscope Meter visible (refer to Chapter 9), start playback and turn down a slider on the GEQ-10. Watch what happens on the Spectroscope, and you'll get an idea of how an equalizer works. You can do the same thing with the GEQ-30, StudioEQ, PostFilter, and DualFilter.

GEQ-10

GEQ refers to Graphic Equalizer, while 10 refers to the number of adjustable frequency bands. Using a Graphic EQ is a little like having volume controls for different frequencies.

The GEQ-10, shown in Figure 14.15, has 10 frequency bands, starting at 31.5 Hz and ending at 16 kHz. Even though the frequency range isn't listed as 20 Hz to 20 kHz, the GEQ-10 is full-range.

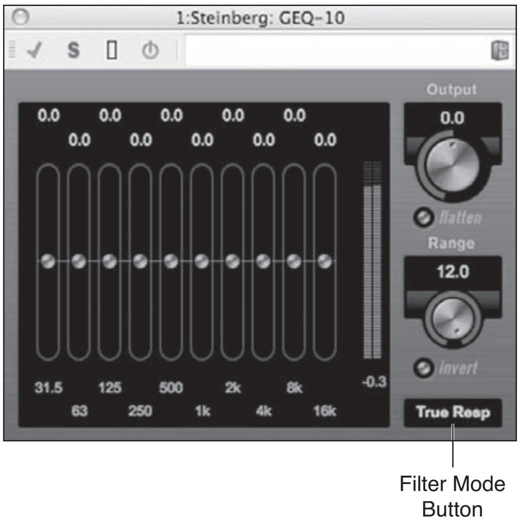

Figure 14.15 The GEQ-10 control panel.

The GEQ-10 has an Output VU meter and an Output control. Because an EQ will be adding and subtracting from the overall volume, it is important to monitor the Output VU meter to make sure you haven't clipped the Output of the plug-in. Conversely, if you're de-emphasizing several frequencies, you might have to raise the Output control to make up the gain.

Frequency Sliders

These are volume controls for each frequency range. The center frequency of each is located directly below the slider, while the slider position is displayed directly above the fader. The values are represented in positive and negative dB. Positive values will indicate the slider is emphasized, while negative values will indicate de-emphasis.

Clicking and dragging the slider up and down will adjust the amount of emphasis/de-emphasis. You can also draw across multiple sliders by clicking and dragging one slider and then moving your pointer up and/or down while dragging left and/or right. If you know the value you want to input, double-click on the slider's Value indicator at the top of the control panel and type in the desired value. The maximum value will be constrained by the Range control. (See the upcoming "Range" section.)

Ctrl/Command-click on a slider, and it will return to the 0-dB position. You can use the same method to draw multiple sliders back to 0 dB.

The bandwidth (Q-Factor) of each slider is preset but can be determined by looking at the frequencies of the surrounding sliders. In other words, if you're adjusting the 1-kHz slider, the slider to the left is 500 Hz, and the slider to the right is 2 kHz. So the bandwidth of the 1-kHz slider is 750 Hz to 1.5 kHz. Determining bandwidth in this manner is accurate only when using the default Filter mode of True Resp (True Response; see the "Filter Mode Button" section).

Flatten Button

Clicking the Flatten button will reset all the sliders to their 0-dB position.

Range

This will control the GEQ-10 dB Range. The value is represented in dB, and the default setting of 12 will offer +/– 12 dB of slider movement. For finer control, you can lower the Range as low as 0 dB, but that Range would render the EQ disabled. A more sensible setting of 6 would allow each slider to be moved +/– 6 dB.

Invert Button

Activating the Invert button will flip the EQ on its head. Any emphasized frequencies would be de-emphasized and vice versa. When the EQ is inverted, you can monitor the antithesis of your EQ curve. Bear in mind that the visual positions of the frequency faders themselves will not invert.

Filter Mode Button

Clicking this button will allow you to select from one of six filter modes. Some modes are serial, where the signal flows from one slider into the next, whereas other modes are parallel, where the signal flows through each slider simultaneously. The rule of thumb is that using parallel filters will work better with complex audio material. That's why parallel filter modes usually offer a more musical response when mastering. I'll do my best to accurately describe how the Filter modes work, but I recommend doing some critical listening with your audio material to really hear the sonic personalities of each mode.

True Resp (True Response) is a serial mode and will keep the slider bandwidths at mathematically accurate distances. (See the "Frequency Sliders" section.)

Digi Stand (Digital Standard) will allow the highest frequency slider (the slider farthest to the right) to extend its high-frequency curve to the limit of the audio material sampling frequency. In other words, in the case of the GEQ-10, if the audio material sample frequency is 96 kHz, the 16-kHz slider will extend its high curve to 48 kHz. (Refer to Chapter 1 for an explanation of sampling frequency and the Nyquist theorem.)

Classic puts all the sliders into parallel. In this mode, the filters work more efficiently because they all work on the same input. That's how Classic mode works so well on complex audio material. Interestingly, the tone will be altered even when the sliders are all set to 0 dB. Although I can't explain why, I can tell you that Classic mode works really well for mastering. But it's a good idea to audition the EQ using the Bypass button so that you can hear how the Classic mode is crafting the tone even when the sliders are flat.

ConstQ U (Constant Q-Factor, Unsymmetrical) is a parallel mode where the first and last sliders will extend their frequency ranges in relation to the sampling frequency of the audio material. In other words, the bandwidth of the first and last slider will be much wider and therefore will affect more frequencies.

ConstQ S (Constant Q-Factor, Symmetrical) is a parallel mode where the bandwidth increases with the gain of the slider. The more you move the slider, the wider the bandwidth becomes.

Resonant is a serial mode where moving one slider will result in a simultaneous yet smaller opposite movement of the adjacent frequency bands. The sliders won't actually move, but if you emphasize 1 kHz, then the 500-Hz and 2-kHz frequency ranges will be de-emphasized.

GEQ-30

The GEQ-30 (see Figure 14.16) is a graphic EQ with 30 frequency bands.

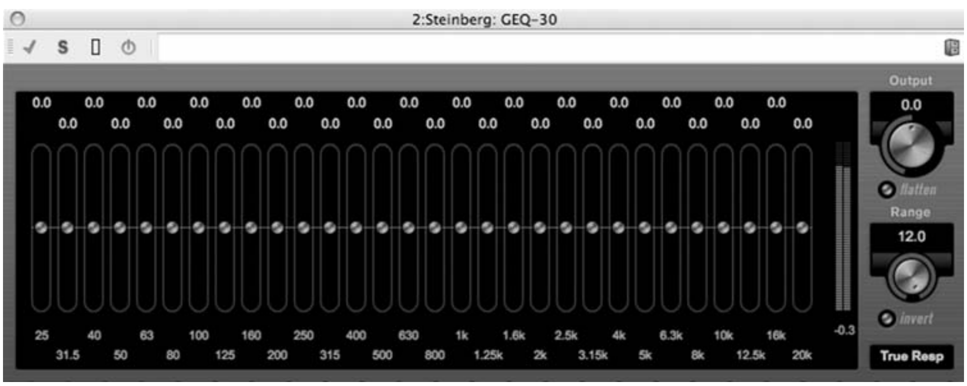

Figure 14.16 The GEQ-30 control panel.

All of the operations and controls are identical to the GEQ-10. The GEQ-30 simply has more faders and therefore narrower frequency ranges.

StudioEQ

The StudioEQ (see Figure 14.17) is a parametric EQ. It has four bands with selectable emphasis/de-emphasis, center frequency, and Q-Factor. While a graphic equalizer has more bands than the StudioEQ, the frequency is adjustable within 1 Hz unit. That makes a parametric EQ much pickier than a graphic EQ.

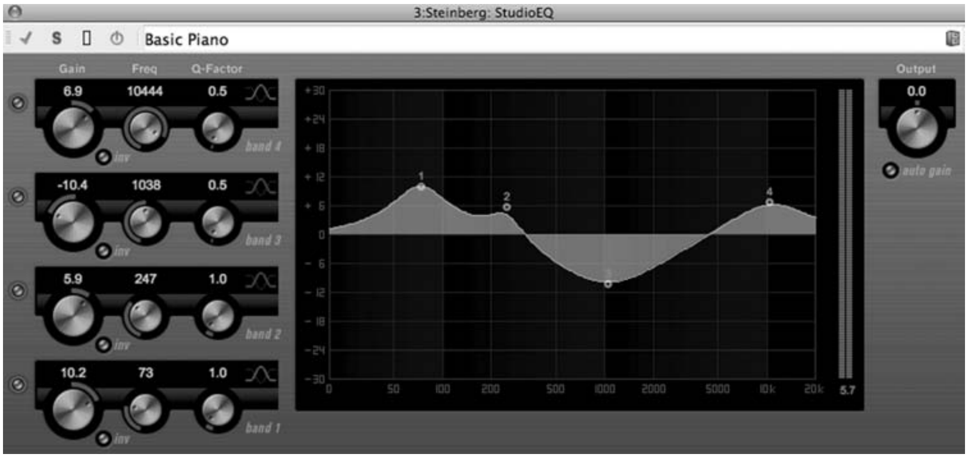

Figure 14.17 The StudioEQ control panel.

The four bands have their individual controls on the left-hand side of the control panel. On the right-hand side, there is a graph. You can program the EQ from either the knobs or the Graph. The bottom-to-top orientation of the bands might lead you to believe they are designed to be lo, lo-mid, hi-mid, and hi EQ filters, but that's not the case. The four bands are arbitrarily named: Band 1, Band 2, and so on. That's because the center position of each band is selectable between 20 Hz and 20 kHz. So it's possible that, for example, the center frequency of Band 4 could be lower than that of Band 3. However, Bands 1 and 4 have features that make them better for lo and hi EQ.

Band On/Off Button

To the left of control panel are four Band On/Off buttons. These will allow you to audition each filter separately. When you click one of the filters on, its corresponding dot will appear on the Graph.

Gain

This is the amount of emphasis/de-emphasis for each band. Gain is represented in dB and is variable +/– 24 dB.

Inv (Invert) Button

Each band can be flipped on its head to audition the antithesis of the band. This is useful when you've emphasized a frequency and swept across the frequency range to find a specific frequency you'd like to de-emphasize. When you've found it, clicking the Inv button will turn the emphasis into a de-emphasis.

Freq (Frequency)

This will define the center frequency of the band. Each frequency has a range of 20 Hz to 20 kHz.

Q-Factor

This will control the bandwidth of each frequency band. A setting of 0.5 will set the bandwidth as wide as possible, while a setting of 10 will make the filter into a very narrow notch.

Filter Mode

Bands 2 and 3 are preset to be Peak filters. However, Bands 1 and 4 have selectable modes. The effect of the different filter modes, especially the Shelf and Cut modes, will depend largely on the Q-Factor setting.

One of the terms I'll be using is *resonance*. Resonance is a narrow band of frequencies located near the band frequency. When you add or subtract resonance, you're working with the frequencies in that narrow band, which will add either a whistling or a growling, depending on whether it's a hi or lo filter. The easiest way to see what resonance is adding to your EQ setting is by looking at the Graph on the right-hand side of the control panel.

- **Shelf I.** Adds filter resonance in a direction opposite the gain. The resonance on Band 1 will appear above the center frequency, while Band 4 resonance will appear below the center frequency.
- **Shelf II.** Adds resonance in the same direction as the gain and at the center frequency.
- **Shelf III.** Combines the qualities of Shelf I and Shelf II.
- **Cut.** This will turn the filter into a low-cut or high-cut filter. Basically, it will cut out all the frequencies below (Band 1) or above (Band 4) the center frequency. In this mode, although you can still adjust the Gain setting, doing so will have no effect because the Cut mode has a preset Gain.
- **Peak.** This is the same filter mode used by Bands 2 and 3. It's known as a *band-pass filter* because it has an equally spaced, albeit adjustable, bandwidth. Peak mode is similar to a graphic EQ slider in True Response mode, except that you can adjust the Frequency and Q-Factor.

Making EQ Adjustments in the Graph Window

Whenever you have a band enabled, it's correspondingly numbered EQ point will appear on the Graph. Clicking and dragging the EQ point up and down will adjust the Gain, while left and right will adjust Frequency. There are some modifier key/mouse dragging options that make adjustments easier and more precise.

Holding Ctrl/Command while dragging up and down will leave the Frequency setting at its current position and allow you to adjust the Gain setting.

Holding Alt/Option while dragging left and right will leave the Gain setting at its current position and allow you to adjust the Frequency setting.

Shift-dragging up/down or left/right will adjust the Q-Factor setting while leaving the Frequency and Gain settings unchanged.

Being able to drag the EQ points inside of the Graph will make it much easier to zero in on certain frequencies. You'll need to be listening to the audio while you do this, but bear in mind that some audio editors (even to this day) don't allow you to monitor the audio while making adjustments. (How silly is that?)

Note: Even though the Graph Level ruler ranges from +/– 30 dB, the Gain controls have a maximum range of +/– 24 dB. However, when you're using Shelf filter modes on Bands 1 and/or 4, it is possible to push the EQ curve beyond the visual limit of the Graph. Don't worry, because the audio is still there. In other words, even though you can't see it, you can still hear it.

Auto Gain Button

Because the StudioEQ (like any EQ) will be adding volume to or subtracting volume from the audio material, you will need to monitor the Output VU meter and the Output knob to make sure you don't clip the StudioEQ. However, the Auto Gain button will allow the plug-in to set its optimum output automatically.

Filter

WaveLab comes with two specialized filters. These filters are similar to EQ, but they offer some specialized capabilities (specifically the PostFilter) that a normal EQ cannot offer.

DualFilter

This is essentially a low-pass or high-pass filter with optional resonance. See Figure 14.18.

Figure 14.18 The DualFilter control panel.

In its default (off) state, the DualFilter has a green Graph that indicates only the low- and high-pass filter positions. Because there are no points that appear on the Graph, the adjustments must be made with the two knobs.

Position

At the default position of 0, the DualFilter will leave the signal unprocessed. Increasing the Position value will slide the high-pass filter to the right and filter out low frequencies. The higher the Position value, the wider the range of low frequencies that will be filtered. Conversely, decreasing the Position value will slide the low-pass filter to the left and filter out high frequencies. The lower the Position value, the wider the range of high frequencies that will be filtered.

Resonance

This will increase the Resonance curve of the high or low filter. Resonance is a narrow band of frequencies located near the band frequency. When you add or subtract resonance, you're working with the frequencies in that narrow band, which will add either a whistling or a growling, depending on whether it's a high- or low-pass filter.

PostFilter

The PostFilter (see Figure 14.19) is a combination of a low-cut filter, a notch filter, and a high-cut filter. This is a very powerful tool when you need to filter out very discrete or narrow frequency bands.

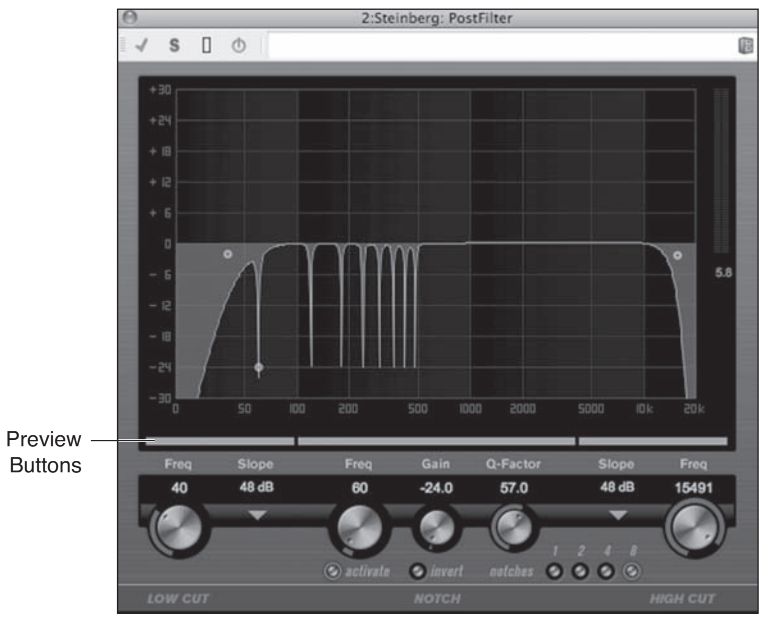

Figure 14.19 The PostFilter control panel.

The PostFilter has both knobs and buttons on the bottom, as well as a Graph on the top. The most unique feature of the PostFilter is the Notch filter. A Notch filter is normally used to correct AC noise, buzz, and hum that has made its way into audio recordings. That noise is usually caused by ground loops, bad cables, or overhead fluorescent-light tubes. You could try to remove the noise with EQ, but the frequency bandwidth of even a parametric EQ is usually too wide to be effective.

Low-Cut Frequency

This control is found in the lower-left corner of the control panel. This is the frequency of the Low Cut filter. Any low frequencies below this setting will be reduced dramatically.

Low Cut Slope

This defines the slope of the Low Cut filter. A setting of 12 dB (per octave) will make the slope very gentle, while a higher setting of 24, 36, or 48 dB (per octave) will set a much steeper slope.

The settings of the Low Cut filter in Figure 14.19 would be effective for reducing rumble below 40 Hz.

Notch Frequency

This will set the fundamental (lowest) frequency of the Notch filter. If you're experiencing an electrical hum, the frequency is usually 60 Hz (50 Hz in the UK). Or, if you don't know the fundamental frequency, you can turn the Gain control (see below) to a high setting (above 10 dB) and then sweep the Frequency knob across the frequency spectrum until you pinpoint the offending hum. Then, pressing the Invert button (see the upcoming "Invert Button" section) will reverse the direction of the notch from positive to negative, thereby reducing the hum.

Gain

This will be the amount of volume that the Notch filter reduces (or adds if the Invert button is off). Louder hum and buzz will require higher Gain settings.

Q-Factor

This is the bandwidth of the Notch filter. Higher values will make the filter more narrow and selective. You will want to use as high a setting as possible so that adjacent frequencies are not affect by the Notch filter. However, setting the level too high might not cover enough of a frequency range to be effective.

Activate Button

This turns the Notch filter on and off. The Low Cut and High Cut filters will remain active even if the Notch filter is deactivated.

Invert Button

This will flip the Notch filter upside down. If the Notch has positive Gain, pressing the Invert button will make it negative, and vice versa. The Invert button is typically used after the frequency has been determined, to quickly reverse the direction of the notch Gain.

Notches Buttons

These buttons let you enable more notches at harmonic locations based on the frequency. Hums are often accompanied by buzzing above the fundamental frequency. Activating more notches (up to eight) can reduce the volumes of the buzzing. Try not to use more notches than you need to attenuate the buzzing, because extra notches might remove frequencies that aren't associated with the buzz.

High Cut Slope

This defines the slope of the High Cut filter. A setting of 12 dB (per octave) will make the slope very gentle, while a higher setting of 24, 36, or 48 dB (per octave) will set a much steeper slope.

High Cut Frequency

This control is found in the lower-right corner of the control panel. This is the frequency of the High Cut filter. Any high frequencies above this setting will be reduced dramatically.

The settings of the High Cut filter in Figure 14.19 would be effective for reducing hiss above 15,491 Hz.

Preview Buttons

The Low Cut, Notch, and High Cut sections all have a Preview button. This will allow each section to be auditioned for the frequencies that are being filtered. Only one Preview button can be active at one time. When a Preview button is activated, the Graph will turn red and black. The red area is a representation of the frequency spectrum unaltered by the filter, while the black area represents the frequencies that are being filtered.

Reading the PostFilter Graph

At the upper part of the control panel is the PostFilter graph. Each filter you have activated will appear on the Graph with a frequency adjustment point. You can move the Low Cut and High Cut points to the left and right to adjust the corresponding filter frequency. You can manipulate the Notch frequency point by clicking and dragging in the same way as you did for the StudioEQ plug-in. (See earlier.) If you have more than one notch activated, they will appear on the Graph as well.

Modulation

Modulation effects get their name by taking the audio material and splitting it into two separate signal paths. One path remains unaltered, but the other has an LFO (*low-frequency oscillator*) and a short delay applied to it. The frequency of the LFO is usually well below the limit of human perception, so the effect doesn't add any sound. Rather, the pitch (or volume) is altered based on the frequency and amplitude of the LFO.

Common Modulation Parameters

All of the Modulation effects that come with WaveLab will have some of the following parameters.

Rate

This is the frequency of the LFO and is measured in Hz units. The setting ranges from 0.01 Hz to 5 Hz (or 10 Hz for the AutoPan). Lower settings will provide a gentle modulation, while higher values will make the modulation faster.

Width

This is the amplitude of the LFO and is measured in percent from 0 to 100%. Lower values will make the modulation less audible, while higher values will increase the modulation effect.

Spatial

This will adjust the stereo image of the modulation effect. At 0, the modulation effect will still be audible, but present equally between the left and right audio channels. Increasing the Spatial amount will modulate the effect between the left and right audio channels and provide a more pronounced stereo image. If you're using mono audio material, the Spatial control will have no effect.

Mix

This will control how much dry (unaffected) signal will be mixed with the wet (effect) signal. A setting of 0 will allow only dry signal through, whereas a setting of 100 will allow only wet signal through. Usually, a middle setting (between 30 and 60) will produce a good balance.

Delay

This controls how long the second (effect) signal is delayed. It is measured in milliseconds and ranges from 10 to 30 milliseconds. Increasing the delay will make for a richer-sounding modulation but might also make the delay overly audible.

Wave Shape

This is the LFO waveform type. You have the choice of sine or triangle waveform shapes. A sine waveform has a smoother up and down motion, similar to the serpentine movements of a snake. The triangle waveform shape will bounce back and forth more sharply, like a billiard ball bouncing off a pool-table cushion.

Lo and Hi Filter

These are low- and hi-cut filters on the wet (effect) signal only. The values are represented in Hz from 10 to 1,000 Hz and 1,200 to 20 kHz, respectively. These are useful when a full-range setting might reduce the clarity of the audio material. For example, an electric bass guitar would benefit from setting the lo filter to a value of 150 to 250, thereby keeping the low frequencies out of the effect signal.

Chorus

The Chorus plug-in (see Figure 14.20) is pitch-based and has all of the parameters we've discussed.

Figure 14.20 The Chorus control panel.

StudioChorus

The StudioChorus plug-in (see Figure 14.21) is pitch-based and has all the parameters we've discussed, but it also has two identical Chorus effects. The audio material will

Figure 14.21 The StudioChorus control panel.

enter the left Chorus first and then feed directly into the right Chorus. This will allow you to make some very rich and deep-sounding modulation effects.

AutoPan

The AutoPan (see Figure 14.22) is volume-based and will modulate the amplitude of the audio material.

Figure 14.22 The AutoPan control panel.

There are fewer controls on the AutoPan, but they function in the same way as the other modulation effects. The only difference is that instead of modulating the pitch like a chorus effect, the AutoPan will modulate the volume.

Unlike the Spatial control of the Chorus and the StudioChorus, the AutoPan will create a stereo effect even with mono audio material.

Pitch

Pitch effects alter the pitch of the audio material but usually don't offer modulation capabilities. Rather, they alter the pitch by a much wider margin.

Octaver

The Octaver (see Figure 14.23) creates octaves of the audio material from one octave or two octaves lower than the audio material. The effect will work properly only on

Figure 14.23 The Octaver control panel.

monophonic audio material, such as a solo vocal, woodwind, or brass instrument. Because the processed signals are lower than the source material, the effect works better when the fundamental frequencies of the audio material are higher than 250 Hz (middle C).

Direct

This controls how much of the original audio material will be present in the processed signal.

Octave 1 and Octave 2

These controls allow you to change the level of the first octave and the second octave below the fundamental frequency of the direct signal. By combining the Direct signal with the Octave signals, you can deepen the frequency content of audio material.

Reverb

Reverb is another type of spatial effect used to simulate an acoustic environment. The original uses for early reverbs were to mix the direct studio-recorded sound of music with the environmental ambiance of a performance venue. Whether it was a concert hall, a church, or a recital room, a reverb made it possible to record an ensemble in a studio and add the ambience in post-production. These early reverbs were based on springs or metal plates. In fact, a well-told story of one early reverb relates back to my hometown.

I grew up in Salt Lake City, Utah. One of the most famous downtown buildings is known as the Tabernacle. It was built in the 1860s by the Church of Jesus Christ of Latter-Day Saints and was used for many purposes, including as a performance venue for the Mormon Tabernacle Choir and an early venue for the Utah Symphony. The building offers some truly outstanding acoustical properties even to this day.

Back in the mid-1960s, the LDS Church owned a recording studio across the street from the Tabernacle. One ingenious engineer realized that by running cables through the underground tunnel from the studio to the Tabernacle, they could pipe recordings into the venue through speakers and then record the natural reverb of the room with a microphone. That signal could then be returned to the recording studio, and that fantastic sound could be mixed into studio recordings.

Spring tanks and plate reverbs were later invented and offered a more feasible way of providing reverb effects. But it wasn't until the digital reverbs hit the market that truly worldclass acoustic simulation became available to everyone.

Common Reverb Parameters

WaveLab comes with two reverb processors, one of which is a smaller, less CPU-intensive version of the other. Therefore, the basics of the Reverb effects are identical.

Pre-Delay

This basically controls how far away the audio material is from the venue's opposite wall. In other words, if you're standing on the stage of a symphony hall and you clap your hands, the sound travels at 1,100 feet per second (343 meters per second) from your hands to the back wall (the cheap seats) and then bounces off of that wall and back into your ears. The bigger the room, the more delay there will be. So, setting a larger Pre-Delay will simulate a larger environment more accurately.

The Pre-Delay amount relates to time and is adjustable from 0 to 500 milliseconds (1/2 of a second).

Reverb Time

This is the duration of the reverb from the beginning to the end of its audibility. It is also a measure of time and is variable from 0.10 to 20 seconds. Larger values will be used to simulate concert halls and traffic tunnels, while smaller values will simulate the garage we've all rehearsed in or even the shower we sing in.

This is another setting that you should adjust carefully. Setting the Reverb Time too high will make for a very cloudy and confusing effect. Typically, a pop vocal reverb will have no more than one or two seconds of Reverb Time. But ensemble recordings of symphony orchestras or choirs might have as many as four to six seconds. Anything higher than that can quickly create a cacophony of sound that will become difficult to control.

Diffusion

This will control how many sonically reflective surfaces you have in your simulated environment. More surfaces will create more reflections of the reverb. For example, a school gymnasium has only six reflective surfaces, made up of four walls, one ceiling, and one floor. However, a typical Christian church has columns, pews, tables, statues, and taller multi-angled ceilings. Therefore, there are more surfaces that create more diffusion. So, larger diffusion settings make for more surfaces for the sound to bounce off of.

Lo Level

This controls the decay time of the low frequencies in the reverb effect. It's measured in a percentage from 10 to 400 percent, with 100% being the baseline. Lower settings will reduce the amount of low frequencies in the reverb and will sound as if you've put people in the church pews. The clothing the people are wearing (depending on the kind of church you attend) will absorb more bass frequencies. But if you want to simulate an empty church, you may need to raise the Lo Level above 100% so that those bass frequencies will be unencumbered by absorbent surfaces.

Hi Level

This works identically to the Lo Level but affects the high-frequency decay of the reverb. If the room you're simulating has more glass, mirrors, or metal, the high frequencies in the reverb will get absorbed less. High percentages will make the reverb sound glassier, while lower settings will make the room sound as if it has been acoustically treated to dampen the acoustic liveliness.

Mix

Once again, the Mix control manages the wet to dry level. (See earlier in this chapter.)

RoomWorks

RoomWorks (see Figure 14.24) is the highest quality reverb that comes with WaveLab. It's also the more programmable and better-sounding reverb of the two. However, it does take more processing power to run RoomWorks than it does the SE version.

Figure 14.24 The RoomWorks control panel.

RoomWorks is surround-sound compatible, and there are several presets that capitalize on this functionality. So, if you're working in a mono or stereo environment, try to avoid the presets that have a name starting with Surr., or else you may end up with some strange-sounding results.

Lo and Hi Frequency and Gain

These are the four controls on the far-left side of the control panel. These allow you to craft the audio-material frequency content as it enters the plug-in. It's similar to the Lo and Hi levels except that instead of altering the frequency content of the reverb, the Lo and Hi Frequency and Gain can limit the frequencies that get into RoomWorks in the first place.

The Lo Frequencies range from 25 to 6 kHz, while the Hi Frequencies range from 250 to 22 kHz. The Gains of each frequency are variable from −18 to 6 dB.

Variation Button

This is the roulette wheel of RoomWorks. Clicking on this button will alter the reverb characteristics ever so slightly and offer gentle variations to the same reverb settings. There are 1,000 variations.

Hold Button

Pressing this button will make the reverb signal repeat in an infinite loop. The loop is smooth and without pops or clicks. Although not a very musically useful option, the Hold button does offer some very interesting sound-effect possibilities.

Size

The sound in an acoustic environment can bounce off of sonically reflective surfaces that are located closer to the listener than the back wall. Therefore, that reverb signal will be heard before the big back-wall reverb. These are known as *early reflections*, and the Size control allows you to adjust how many early reflections you'll hear. From what I can tell, it's an arbitrary number from 20 to 250. Larger values will simulate more reflective surfaces located closer to the listener. Usually, larger rooms have more early reflections.

Damping Lo and Hi Frequency

While the Lo and Hi Level work as described earlier in this section, the Hi and Lo Frequencies are also adjustable. The Lo is adjustable from 25 Hz to 1 kHz, and the Hi is adjustable from 500 Hz to 22 kHz.

Envelope Amount, Attack, and Release

The envelope allows you to customize the Attack time and Release time of the reverb effect. The Amount control adjusts the strength of the envelope, or, in other words, how much the envelope will affect the reverb signal. You'll notice that the 50% mark of the Amount is at almost the 3 o'clock position. Settings above 50% can make the envelope stronger than required to be audible. Therefore, settings below 50% will get you better results.

The Attack and Release are both measured in time and adjustable from 10 to 1,500 milliseconds. The Attack controls how long it will take for the envelope to engage, while the Release controls how long the reverb will last. Shorter release times will create what is known as a *Gated Reverb*. If you listen to any Phil Collins drum tracks from the '80s or '90s, you'll get a good idea of what this effect sounds like.

Wet Only Button

Chances are that you won't be using this button very often, at least not in WaveLab. This button allows only the wet (reverb) signal to pass through to the RoomWorks output. In a DAW program like Cubase, a reverb effect is usually set up to be a send effect where the dry signal comes through the track, but only the reverb (or any other time-based effect) should be allowed to come back into the signal path. But because WaveLab runs all of its plug-ins in series (out of one into the next, and so on), clicking this button will remove all of the original audio material from the signal path. Therefore, unless you're making your audio sound as if it's coming from a faraway distance (as if you're standing backstage), you'll never turn on the Wet Only button.

Note: Turning the Mix control fully clockwise is the same as turning on the Wet Only button.

Efficiency

This control determines how much computer processing power will be used by the plug-in. Reverbs use a larger volume of computer processing than other plug-ins, so it's nice to be able to set the Efficiency to work properly with your computer. Lower values will be less efficient and will take more processing power. If you start to hear pops and clicks during playback, you may need to raise the Efficiency.

Export Button

Chances are, you'll always want this button on. This will set the Efficiency control to 0 during rendering. Because you'll probably want the highest-quality reverb signal in your rendered files, just leave this button on.

RoomWorks SE

This is the Special Edition version of the larger, more powerful RoomWorks plug-in (see Figure 14.25). Although it's a good-sounding reverb, it cannot be customized as extensively as its larger counterpart.

Figure 14.25 The RoomWorks SE control panel.

Because the SE version won't draw as much computer power as RoomWorks, it's a good choice to use on older, less powerful computers. Personally, every computer I use WaveLab with is less than two years old. If you have a fairly modern computer, chances are that you won't need to use the RoomWorks SE just to save on processing power. Not that there aren't some good-sounding presets in RoomWorks SE, but I usually go for the best-sounding, most programmable reverb possible.

Spatial

The two Spatial plug-ins that come with WaveLab are a little tricky to set up. That's because the behavior of each processor is completely dependant on whether you're working with mono or stereo files. I'll go over those operational idiosyncrasies when discussing each plug-in, but suffice it to say that both processors are designed to create or enhance a stereo image to your audio material.

Common Spatial Parameters

The MonoToStereo and StereoEnhancer plug-ins are virtually identical to each other, so they both have the exact same parameters.

Width

This controls how wide the stereo effect will be. It's represented in an arbitrary value from 0 to 200. Increasing the value will increase the left-to-right stereo image of the audio material. A horizontal bar at the bottom of the plug-in control panel will also represent the Width. (See Figures 14.26 and 14.27.)

Delay

Adding Delay will further augment the stereo image. Delay is a time value that ranges from 1 to 30 milliseconds. Even though the Delay times are very short, it is possible to induce a sort of stuttering effect when the values are near the maximum position. Use lower values to avoid stuttering.

Color

Adjusting the Color control will offer a wide range of stereo imaging. From what I can tell, the Color control uses phase shifts and EQ filters to further separate the stereo image. It's an arbitrary value that ranges from 0 to 100. Adjustment of the Color control requires a delicate touch, because there are subtle differences from one integer to the next.

Mono Button

Because a Spatial plug-in is altering the stereo image, it's possible for phase problems to creep in. Pressing the Mono button will virtually sum the left and right audio channels of the Master Section to mono by placing the effect equally in each channel. That way, you can listen for phase problems, such as cancellation.

MonoToStereo

The MonoToStereo plug-in (see Figure 14.26) is designed to create a stereo image in mono audio material.

Figure 14.26 The MonoToStereo control panel.

All of the parameters of the MonoToStereo plug-in were discussed already. However, getting the plug-in to work properly can be a little baffling.

Getting MonoToStereo to Work Properly

This plug-in has one input because it's designed to work with mono audio material. The Master Section, on the other hand, is stereo (two Master Level faders) by default. So if you try to add the MonoToStereo plug-in while the Master Section is in stereo, you will get an error warning:

> Error reported by plug-in Steinberg: MonoToStereo:
>
> Cannot handle the required number of input channels. Plug-in is switched off.

Plus, the MonoToStereo plug-in works only with mono audio material. But it's possible to get the error message even when a mono audio file is all you have loaded into WaveLab. (This situation made me swear a few times while researching the Spatial plug-ins.)

Here are the steps to make sure the MonoToStereo plug-in works properly for you:

1. Load only mono audio material into WaveLab.
2. Right after you load the mono file, start playback so that the Master Section gets configured into a mono channel (single Master Level fader) configuration.
3. Add the MonoToStereo plug-in to the first Master Section effects slot. Doing so will keep the Master Section input as mono but will reconfigure the output to stereo.
4. If you need to add additional stereo plug-ins, they must be placed in an effects slot *after* the MonoToStereo.
5. Avoid loading and/or playing back stereo audio material after you've loaded the MonoToStereo plug-in. Otherwise, you may get the error warning when you start playback on a mono file. The only way to get the plug-in to work properly without the error message is to reset the Master Section (refer to Chapter 5) and repeat Steps 1 through 4.

StereoEnhancer

The StereoEnhancer (see Figure 14.27) is virtually identical to the MonoToStereo plug-in. The only difference is that the StereoEnhancer is designed to work with stereo audio material.

All of the controls of the StereoEnhancer are identical to those on the MonoToStereo plug-in except for two buttons. The Delay and Color controls each have their own On/Off button.

Figure 14.27 The StereoEnhancer control panel.

Because the StereoEnhancer has two inputs, you probably won't get the channel warning message when using it. If you do, make sure you've loaded only stereo audio files into WaveLab and started playback on a stereo file to configure it for stereo operation (two Master Level faders).

Tools

The Tools plug-ins are highly specialized processors. Although they are highly capable, chances are you'll never use them unless you're recording through a plug-in (later in this chapter) or working in a surround-sound environment. All the Tools plug-ins should be placed in the last effects slot.

TestGenerator

This is a simpler plug-in version of the Signal Generator (refer to Chapter 12). It is designed to create test tones that can be used to test the performance or configuration of audio equipment. See Figure 14.28.

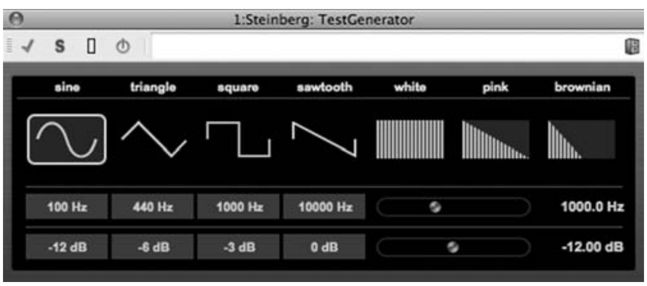

Figure 14.28 The TestGenerator control panel.

There are a few situations where I've used the TestGenerator plug-in rather than the Signal Generator. Because the TestGenerator works in real time, it has the advantage of working in live recording situations.

I've used the TestGenerator when recording through a plug-in and needing a slate tone. A slate tone is used to identify locations in an audio recording either audibly or due to their unique appearance in audio editing software, visually. For example, if I'm recording a long dialog session, it's much easier to record one long audio file instead of recording new files for every new take. With the TestGenerator loaded into an effects slot, I can leave it bypassed until I need to mark a new location. Then when I need the tone, I can click the Bypass button momentarily to record some tone onto the track. That tone will identify to any other audio editor who might be working on this material that some sort of edit is required at that location. (If I were the editor, I'd be dropping markers onto the track while I'm recording. But not all programs read markers in the same way. For that reason, sometimes it's better to use markers and a slate tone so that you become the audio editor's new best friend.)

I've also used the TestGenerator to test other audio equipment that might be connected to my audio interface. For example, I've recorded audio for film into WaveLab but sent the audio to the cameras for review and sync purposes. I'll use the TestGenerator to send a −12-dB tone to the cameras so the camera operators can set a matching input level of −12 dB.

Note: Even −12 will create very loud test tones. If you want to win the friendship of the camera operators, producers, and directors alike, make sure none of them is wearing headphones when you match up the audio levels of the equipment.

Waveform Buttons

The Waveform buttons are located across the top of the control panel. By clicking on the Sine, Triangle, Square, Sawtooth, White, Pink, or Brownian buttons, you can define what type of waveform you'd like to use.

Frequency Settings

There are four common waveform frequencies you can select by pressing the 100, 440, 1,000, or 10,000 Hz button. For different frequencies or fine-tuning, you can use the slider to the right of the frequency buttons, or you can double-click the value at the far right for manual entry. When selecting custom frequencies, the values range from 1 to 20 kHz. Because noise waveforms contain broad ranges of many frequencies, the frequency controls are dimmed out and not necessary.

Output Volume Settings

There are four common output level buttons of −12, −6, −3, and 0 dB. For different levels or fine-tuning, you can use the slider to the right of the level buttons, or you can double-click the value at the far right for manual entry.

Mix6To2

This plug-in (see Figure 14.29) is designed to work in a DVD-audio-compatible surround-sound Audio Montage. If you're using WaveLab in a surround-sound

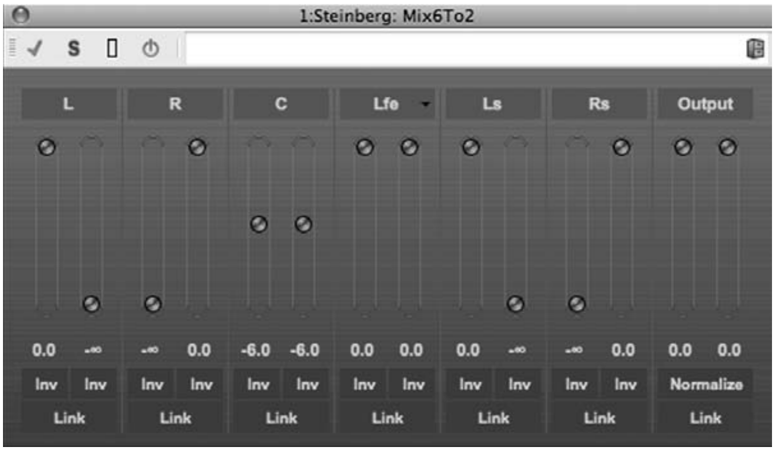

Figure 14.29 The Mix6To2 control panel.

environment, the Mix6To2 plug-in can take six audio channels and create a stereo version of the surround mix. A 5.1 surround-sound configuration consists of six discrete audio channels: Left Front, Center, Right Front, Left Surround, Right Surround, and the .1 is a LFE (*low-frequency effects*) or subwoofer channel.

Each of the audio channels has a fader to send the signal to the left output, the right output, or a combination of the two. There are also Inv (Invert Phase) buttons on each channel and a LINK button that connects both channel faders together. The LFE channel also has a drop-down box (the black triangle in the channel name) that allows you to set the gain of the channel to 0, 6, 10, or 20 dB. The Master Output fader includes a Normalize button that will adjust the output automatically to 0 dB without clipping.

Mix8To2

This plug-in (see Figure 14.30) is identical to the Mix6To2, but as its name implies, it can mix eight audio channels into a two-channel stereo mix. It is designed for use in a multichannel Audio Montage.

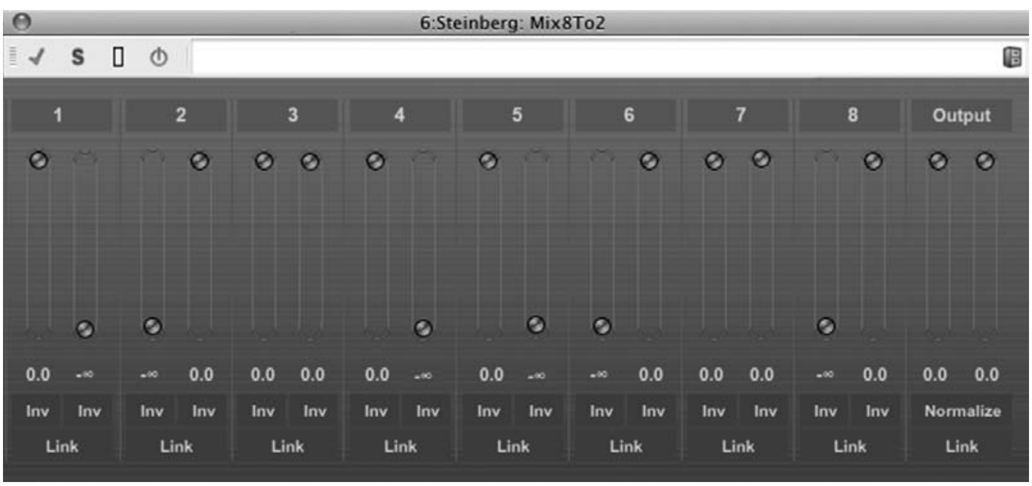

Figure 14.30 The Mix8To2 control panel.

The controls of the Mix8To2 plug-in are the same as those of the Mix6To2 plug-in. (See the preceding section.)

Sonnox Restoration Tools

WaveLab comes with the three world-class restoration plug-ins from Sonnox Ltd. These plug-ins can clean up audible anomalies, such as pops, clicks, crackles, hum, buzz, and hiss. Although most of those anomalies are associated with analog recordings, such as tape and vinyl records, there are sonically similar artifacts that can creep into the most sophisticated digital recording studios. Using these plug-ins can remove those artifacts with a minimum of fuss.

Online Quick Start Guides

Each Sonnox tool has a quick-start guide you can find in the online WaveLab manual. While you're in WaveLab, click on the Help About Active Window button on the Command Bar or press F1 (Mac users press Shift+Command+/). Then search for Sonnox, select the desired plug-in, and then scroll to the bottom of the page for the quick-start guide.

Common Sonnox Parameters

The Sonnox tools share some common parameters, and they're all located in the same spots on each control panel.

Sonnox Box

In the upper-left corner of each control panel is the Sonnox logo. By clicking on that logo, you can customize the response of the Input and Output meter clip lights. The Clip Lights Hold Indefinitely setting will leave the clip lights on forever or until you click on the light to turn it off. You can also choose a two-second or five-second hold.

On the DeBuzzer, there's also a setting for Default to 50 Hz. Because Sonnox Ltd. is in the UK, and electricity there operates at 50 Hz, the plug-in loads with that frequency as the default. If you're in a country where the electrical frequency is 60 Hz (such as the USA), uncheck the setting so that the DeBuzzer loads with 60 Hz as the default.

Input/Output Meters

On the left and right sides of each control panel are vertical meters that show you the Input and Output levels of the plug-in. At the top of each meter is a red Clip indicator that will warn you when the input or output level of the plug-in has been exceeded and has induced clipping. To reset the Clip indicator, click on the Clip light.

Trim Input/Output Level

Below each meter is a value button that shows the Input and Output level. Clicking and dragging up and down on the corresponding button will raise or lower the input, while the output can only be lowered from 0 dB.

Knob, Slider, and Value Box Adjustments

All of the variable controls on the Sonnox plug-ins consist of a knob, a slider, and a value box. Operating the knobs is a little different from with the other WaveLab plug-ins. The Sonnox knobs all work in circular mode, where you click and drag your mouse on the knob in a circular motion. While you're holding down the mouse button, you can increase the pointer radius away from the knob for a finer level of adjustment. (The other plug-ins we've talked about all operate in linear mode, where you can make adjustments by clicking and dragging up and down or left and right, and holding Shift will make finer adjustments.) But if you'd rather the knobs work in linear mode, hold down Alt/Option while dragging.

Although you can make adjustments by clicking and dragging on the value boxes, you can also right-click and drag on a value box for a finer degree of adjustment. This right-click/left-click method works on the sliders, too.

You can also double-click the value boxes to make manual entries, and you can Ctrl/Command-click any control to set it to the default value.

Enable Button

While the DeClicker has In buttons to enable or disable each of its three processes, the DeBuzzer and DeNoiser have an Enable button for turning the entire plug-in on and off. However, the standard Bypass and effect On/Off buttons across the top of the control panel will perform the same function.

Be Patient and Make Small Adjustments

Because the Sonnox tools all track the audio material dynamically, the adjustments you make on the control panel may not be apparent immediately. It's better to make the adjustment and then let the plug-in reanalyze the audio material with the new settings.

Also, I've found that very slight adjustments can make big differences in the quality of each plug-in. For that reason, making gross control panel adjustments may not allow you to find the best possible settings. Making smaller adjustments usually garners better results.

Sonnox DeBuzzer

The DeBuzzer (see Figure 14.31) is used to remove hum and buzz from your audio material. These types of anomalies are usually caused by noisy electricity, bad cables, or ground loops (multiple sources of electrical ground or earth). Things such as fluorescent and neon light tubes or any device that uses an AC adapter or external transformer can also cause hum and buzz.

Sometimes the default settings will do a fantastic job of reducing hum, but making some gentle adjustments, especially when it comes to removing buzz, can make a big difference.

Figure 14.31 The Sonnox DeBuzzer control panel.

Frequency Controls

The largest knob on the control panel manually adjusts the fundamental frequency of the hum. It's used when the DeBuzzer is set to Freeze tracking. (See the upcoming "Tracking Mode" section.) If it's an electrical buzz, this is usually set to 50 or 60 Hz, but you can adjust it to a range from 20 to 440 Hz. However, if you click on the Fine Adjust button, the range is narrowed from 58.8 to 61.2 Hz. Plus, if you adjust past either extreme, the range will be reconfigured automatically. It also puts the DeBuzzer into Freeze tracking mode.

The Tone On button will produce an oscillator at the current Frequency setting. This will allow you to listen to and match the frequency of hum and make frequency adjustments accordingly. With the Tone On active, a value box will appear above the button, allowing you to change the tone volume. The default is −18 dB.

Detect Meter

This will visualize the amount of hum and buzz that the DeBuzzer is sensing at its input.

Tracking Mode

You can set the DeBuzzer to operate in either Auto or Freeze tracking mode. Auto will have the plug-in search for the fundamental (lowest pitch) frequency of the hum and set the Frequency control dynamically. During playback, the precise frequency of the Auto detection is represented by a small red line appearing on the Frequency knob, along with a numeric readout in Hz.

Although you can still manually adjust the Frequency knob in Auto mode, it is better to let the plug-in determine the frequency. But I have run into situations where I've transferred a vinyl recording into the computer and found hum at a higher or lower frequency than 60 Hz. That's because the hum may have been 60 Hz when the record was cut, but when played back on my phonograph, even through great care was taken to set a precise 33 or 45 RPM speed, the frequency of the hum was altered by the speed of the phonograph.

In Freeze mode, the DeBuzzer is in full manual mode. The frequency is set by you and is not automatically detected. The Fine Adjust and Tone On controls will become very useful in Freeze mode.

A good practice for proper hum removal is to loop a silent section of audio material, such as the space in between each song. That silent section will contain an easily detectable hum frequency. I'll usually set WaveLab to play that silent section in a loop, and when the DeBuzzer detects the Frequency in Auto mode, I'll flip it to Freeze mode so that the Frequency doesn't shift. However, if your turntable doesn't maintain a constant speed very well, leave the DeBuzzer in Auto mode.

Sensitivity

This will control how sensitive the DeBuzzer is to hum and buzz. If the hum is loud, you can set the Sensitivity lower. If the hum is apparent but not very loud, you may need to set the Sensitivity higher. However, high Sensitivity settings might remove too much of the hum frequencies and take a larger amount of the audio material with them.

Mode Button

You can set the DeBuzzer to Hum mode or Buzz mode; however, Buzz mode still has the hum-removal component. In Hum mode, the fundamental and harmonic frequency range is 0 to 800 Hz. In Buzz mode, the range increases to 4 kHz, making it more effective at removing high-frequency buzzing. However, the widened frequency response will remove more high frequencies and therefore might remove too large a portion of audio material. The rule of thumb is to leave it in Hum mode unless there's an accompanying high-frequency buzz.

Reduction Meter

This will visualize the amount of hum and buzz that is being removed from the audio material. If the hum is loud, it is not unusual to see a large amount of Reduction. However, if the hum is somewhat quiet, and the meter is still showing a large amount of Reduction, you may need to adjust the Attenuation (see the next section) or Sensitivity.

Attenuation

This is how much detected hum will be removed. It has a very wide range from 0 to −96 dB, so if it's set too high, it can remove not only hum, but also other non-hum frequencies from the audio material. For that reason, it's best to listen to a loop of a silent section (see the earlier "Tracking Mode" section) and adjust the Attenuation downward until the hum is reduced just below the point of audibility.

Sonnox DeClicker

The DeClicker (see Figure 14.32) has three filters—one for each type of artifact. Each filter is preset to identify the somewhat unique sonic fingerprint left by the artifact:

- A pop will be a loud and fast transient with a wide frequency range, like a deep scratch on a vinyl record.
- A click will be an even faster loud transient with a higher-frequency content, like the discharge of static electricity.
- Crackle is a consistent anomaly that sounds like vinyl surface noise or radio static.

You can adjust or activate each filter separately, depending upon what artifacts are present in the audio material.

Figure 14.32 The Sonnox DeClicker control panel.

Each filter has its own Sensitivity slider with value box, a Detect meter, and In button. It's important that you turn on only the filter you need. If you were to leave the DePOP filter on when working with audio material that has no pop artifacts, the filter can falsely detect other transients (specifically drums) as popping and reduce them unnecessarily.

Sensitivity Sliders

Each filter has its own Sensitivity slider. Raising the Sensitivity will increase the strength of the filter. However, setting the Sensitivity too high can cause the filter to falsely identify desirable audio material as an artifact, thereby removing too much of the original audio material. It's best to listen to a section of audio material with a popping, clicking, or crackling in it and adjust the corresponding slider until the artifact becomes inaudible.

Detect Meters

Each filter has its own Detect meter. At the bottom of every meter is a red light that will indicate that the filter has detected a sonic artifact. The strength of the meter above the red light will indicate the volume of the artifact. The Detect meters always represent the input of the plug-in and won't be affected by higher sensitivity settings.

Sonnox DeNoiser

The DeNoiser (see Figure 14.33) looks for the kinds of hiss inherent in analog tape and vinyl recordings. However, other electronic devices can add noise. For example, the Trim (Gain) control set too high on a mic preamp can induce noise very similar to tape hiss. Therefore, the DeNoiser would be effective at removing that type of noise, too. The noise is usually in the higher frequency band, from 1 to 20 kHz.

Figure 14.33 The Sonnox DeNoiser control panel.

Reading the Graph

Unlike the other Sonnox tools, the DeNoiser has a graph at the top of the control panel. The graph visually represents the audio material, noise, Sensitivity control, and HF Limit control all at the same time. The 20 Hz to 20 kHz frequency range is depicted from left to right, but because noise is usually found above 1 kHz, the frequency range has been compressed to a smaller portion of the graph. The volume scale of –144 to 0 dB is depicted from top to bottom.

During playback, the audio material will appear as a blue waveform. The Sensitivity control will be represented by a yellow vertical line extending from left to right. The HF (High Frequency) Limit will appear as a magenta area extending from right to left.

Sensitivity

The Sensitivity slider is adjustable from +/−18 dB with 0 dB as the default. At 0 dB, the filter will set the Sensitivity at about the same volume as the audio material. This setting is useful for removing a very small amount of noise. Adjusting the Sensitivity higher will increase the noise detection, while lower settings will move the Sensitivity further into the audio material. Basically, if the noise is loud, you should set the Sensitivity above 0 dB, but if the noise is quiet, then you should set the Sensitivity below 0 dB.

Tracking Mode

The DeNoiser has two tracking modes. Adapt (Adaptive) will dynamically analyze the audio material for noise. This mode works best when the noise level is inconsistent. When using Adapt mode, the settings of the Sensitivity and Attenuation will require great care. Large settings can create metallic, swirly artifacts that are usually more distracting than the noise would be.

The Freeze mode will make an analysis of the noise and use that as the noise fingerprint. Freeze mode works best when you have a portion of audio material that is void of all other sound except the noise (such as the quiet area in between song tracks on a vinyl record). During playback of that quiet section, switch from Adapt to Freeze mode. The DeNoiser will do an analysis of the noise and create a noise fingerprint that will appear in the graph as a red horizontal line. With the Sensitivity set to 0 dB, adjusting the Attenuation slider (see the upcoming "Attenuation" section) downwards will remove frequencies that resemble the noise fingerprint. In Freeze mode, you can increase the Attenuation more dramatically than in Adapt mode.

HF Limit

The HF (High Frequency) Limit is basically a high-cut filter that will remove all frequencies above a certain setting. It's measured in Hz units and is adjustable from 1 to 22 kHz. The default setting of 22 kHz will work best for most full-range audio. However, some audio is not full-range, and any noise might be outside the frequency range of the audio material.

Let's use a tape-based Dictaphone as an example. Dictaphones are the pocket-sized tape recorders used to dictate speech for later transcription into text. Compared to digital recorders, these small, slow-moving tapes are designed to capture only the frequency range of human speech. However, the tape hiss they create can extend far beyond the upper frequency range of speech. You'll never really notice the hiss when you're playing the tape back through the Dictaphone's built-in speaker. But that hiss becomes *very* noticeable and distracting when played back on a full-range audio system, such as studio monitors or high-quality headphones.

Because the sibilance of human speech usually tops out around 8 to 12 kHz, adjusting the HF Limit to an equal amount will remove any high frequencies above that setting.

And because most Dictaphone tapes get re-recorded over and over again, the high-frequency capabilities of the tape get worse each time and might require an even lower HF Limit setting.

Attenuation

This slider will set, in dB, how much of the noise fingerprint will be removed. The louder the noise, the more Attenuation will be needed to sufficiently reduce it. However, setting the level too low (especially when in Adapt mode) can create some digital artifacts that are more distracting than the noise would be. The basic rule of thumb is to lower the slider only as far as required to remove an appropriate amount of noise.

Recording through a Plug-In

WaveLab comes with two specialized plug-ins that are designed to utilize the enhanced capabilities of ASIO audio interfaces. ASIO is an acronym for *Audio Streaming Input/Output* and is a hardware-driver standard developed by Steinberg. If you're using an ASIO-compatible audio interface, you can use the Audio Input plug-in to record (via the Render process) real-time audio through the Master Section. That allows you to record through any of the plug-ins. This is especially useful when you have to deliver audio material immediately after recording it.

Using the ASIO Audio Input Plug-In

Click on the first Master Section effects slot and then click on the ASIO heading and select the Audio Input plug-in.

Programming the Audio Input control panel, shown in Figure 14.34, is very simple. You can record from as many as eight inputs. Moving the Input slider to the right will allow you to select more than default of two inputs. You can also click on the numeric value of inputs to open a dialog box for the manual entry of up to eight inputs. If you need to record more than two inputs, your audio interface will need to have more than two input connectors.

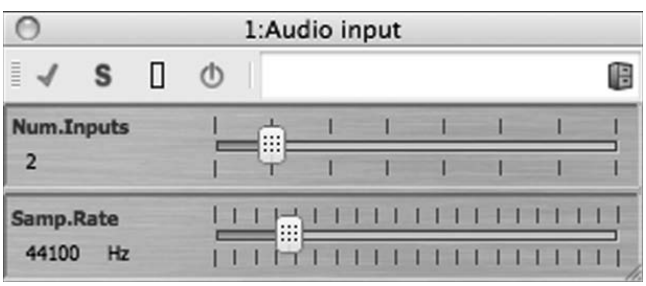

Figure 14.34 The Audio Input control panel.

The Sample Rate slider will control the frequency of the recording. It is constantly adjustable from 11.025 to 192 kHz. Because it's possible to set sample rates down to the last integer, it's usually better to click on the numeric value below Samp.Rate and type in the exact value. For example, to set the sample rate to 48 kHz, click on the value, then type 48000 into the dialog box that will appear on the left-hand side of the control panel, and then press Enter. If you're using settings below 44.1 kHz or above 48 kHz, make sure your audio interface is capable of lower or higher sample rates.

The Audio Input plug-in works in conjunction with the settings on the ASIO Plug-Ins tab of the Audio Streaming Settings dialog box. Go to the Options menu and select Audio Streaming Settings or press Shift+Ctrl/Command+P. Then click on the ASIO Plug-Ins tab. See Figure 14.35.

The ASIO Plug-Ins tab is divided into two columns. The left column is the Device Output (to Gear) column, and the right is the Device Input (from Gear) column. We're only going to use the Device Input column for the Audio Input plug-in.

Properly programming this dialog box has everything to do with using the Audio Input plug-in. Input 1 of the Audio Input plug-in corresponds to the first Device Input setting. Input 2 corresponds to the second Device Input setting, and so on. You'll notice that in Figure 14.35, I have the first and second devices set to MR816CSX(2) Analog 5 and MR816CSX(2) Analog 6. That routes the fifth and sixth inputs of my audio interface to the Audio Input plug-in's first and second input. This allows you to freely configure which hardware device goes into which input of the Audio Input plug-in without having to physically plug and unplug the cables from your audio interface.

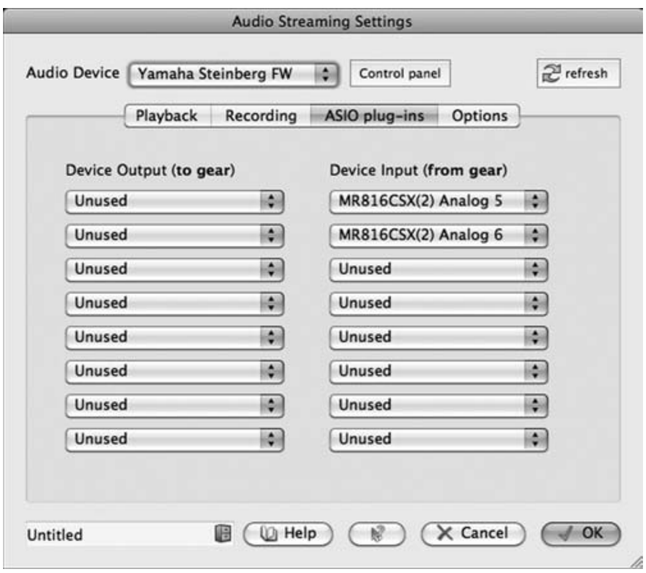

Figure 14.35 The Audio Streaming Settings ASIO Plug-Ins tab for the Audio Input plug-in.

Adding Plug-Ins to Record Through

With the Audio Input plug-in installed in the first Master Section effects slot, any plug-ins you add will become part of the recording signal path. For example, if I added the Limiter plug-in to Effects Slot 2, then the audio would flow from the Audio Input plug-in to the Limiter. Any effect you add will become part of the recorded file.

Monitoring the Input

When you're using the Audio Input plug-in, clicking the WaveLab Play button or pressing your spacebar will turn on the recording monitor. That way, you can make adjustments to the plug-ins and hear what the audio will sound like before it gets recorded. That will allow you to make any customizations to the plug-ins before committing them to the recording.

Note: Normally, pressing your spacebar would start playback of the currently loaded audio file. But when the Audio Input plug-in is being used, you cannot play back an audio file, because the playback operation is now the monitor-enable operation.

Recording the File

Click the Render button at the bottom of the Master Section or press A. Because you're using the Audio Input plug-in, the Render dialog box will look different (see Figure 14.36).

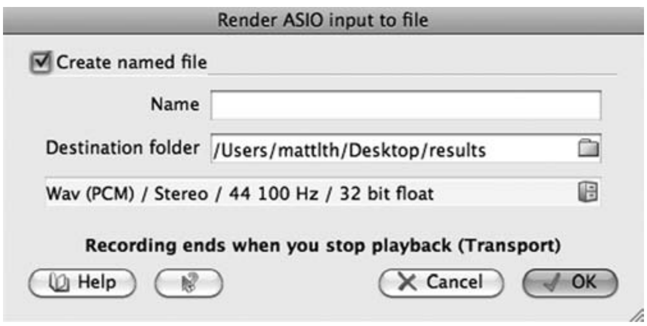

Figure 14.36 The Render ASIO Input to File dialog box.

At the top of the dialog box is a check box for Create Named File. If you don't check this box, WaveLab will create a temporary file named Untitled X. (X is the number of the currently loaded Untitled file.) I recommend that you check this box so that you can save a permanent file as you're recording. You'll also be able to set the destination folder and audio file properties in the bottom of the dialog box.

When you click OK, the recording will begin. You won't actually see a waveform being drawn, but you will notice that all of your meters will be operating. When you're ready to stop the recording, click the Stop button on the Command Bar or press the spacebar. The recording will then appear in the Document window, complete with a waveform.

When you're finished using the Audio Input plug-in, make sure to remove it from the Master Section so that WaveLab will operate normally.

Note: Because the computer takes time to process the incoming audio signal through the Master Section, there will be some latency. The more plug-ins you load, the more latency there will be. This usually won't be a problem, because WaveLab will be the final destination for the audio signal.

Using External Hardware Processors

If you have a hardware processor that you'd like to include in WaveLab, you can do so simply by connecting the device to your audio interface and using the External Gear ASIO plug-in. Your audio interface will need to have more than two inputs and outputs to make using this plug-in possible. You can use only one External Gear plug-in in the Master Section, but by daisy-chaining the ins and outs of your external hardware devices, you can run as many devices together as you'd like. Click on the Master Section effects slot where you'd like to install the plug-in, click on ASIO plug-ins, and select External Gear. See Figure 14.37.

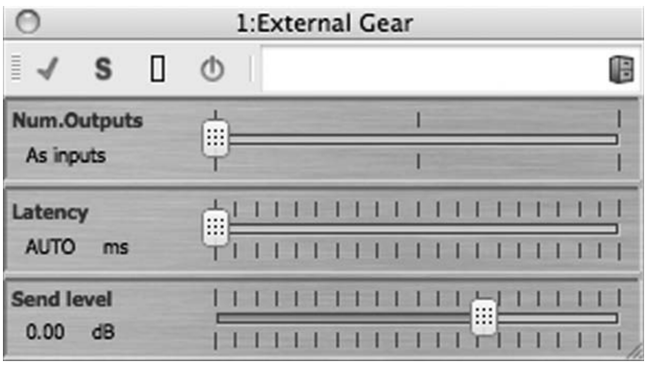

Figure 14.37 The External Gear control panel.

Normally, the Num.Outputs slider is set to As Inputs. This setting will work in most situations. However, it is possible to set it to 1 for mono or 2 for stereo.

Latency Slider

The Latency slider is set to Auto by default. Because the External Gear plug-in will probably introduce some latency into the audio path, the Auto setting will compensate automatically for the latency. (The latency is corrected only during the rendering process, not during monitoring.) If you need to, you can manually adjust for up to 2,000 milliseconds (2 seconds) of Latency.

Send Level Slider

The default setting of 0 dB should work well most of the time. However, because most hardware devices have their own input and output volume controls, you might need to make sure to balance them first and then adjust the Send Level slider to make any final corrections.

Configuring the ASIO Plug-Ins Tab

As with the Audio Input plug-in, the Input and Output settings for the External Gear plug-in are programmed in the ASIO Plug-Ins tab. Click on the Options menu, click on Audio Streaming Settings, and click the ASIO Plug-Ins tab. See Figure 14.38.

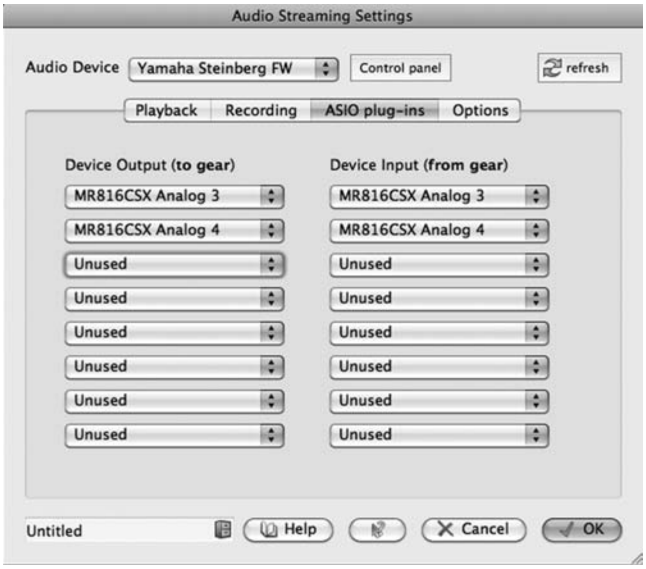

Figure 14.38 The Audio Streaming Settings ASIO Plug-Ins tab for the External Gear plug-in.

The ASIO Plug-Ins tab has two columns, with Device Output (to Gear) on the left and Device Input (from Gear) on the right. The left-to-right orientation of the ins and outs is critical. The topmost output flows to the topmost input, and so on. For example, let's say I had a Neve Portico 5042 tape saturation emulator I wanted to use in my Master Section. I would connect a pair of outputs from my audio interface to the inputs of the 5042 and then the outputs of the 5042 back into a pair of my audio interface inputs. In Figure 14.38, I've used the third and fourth analog ins and outs of my Steinberg MR816CSX to connect the 5042 ins and outs.

Now let's take a moment to consider the signal flow. The audio file you have loaded will play back into the Master Section. The signal will flow through the effects slots from top to bottom. When the signal hits the External Gear plug-in, it will get sent out of the Device Outputs (to Gear) ports defined in the ASIO Plug-Ins tab (refer to Figure 14.38). That puts the signal flow into your hardware processor. Then, when the signal flows from your processor's audio output, it will arrive at the Device Input (from Gear) ports defined in the ASIO Plug-Ins tab. If there are additional Master Section plug-ins below the External Gear plug-in, the signal will flow through those as well. Finally, the entire signal flow will arrive at the Master Level faders and the Dithering Slot, waiting to be rendered.

Monitoring and Rendering with the External Gear Plug-In

When you're playing back your audio material through the External Gear plug-in, you'll be able to monitor the sound of all your plug-ins, including the External Gear plug-in. However, during rendering, the process will change from the procedure described in Chapter 6. When you're using software plug-ins, the rendering is done at much faster than real time. Because of that, you won't be able to monitor the playback of the rendering. But when you're using the External Gear plug-in, the rendering process must be done in real time. In other words, if your audio material is five minutes long, then the render will take five minutes. That's because your hardware processors operate in real time, not in the accelerated world of software plug-ins. You also won't be able to monitor the playback during rendering.

Tips on Using Hardware Processors

If your hardware processor is volume-based (such as a compressor or an EQ), then you will have to worry only about the input and output levels. Make sure you never clip the signal flow of any hardware processor, or else you will be adding unwanted distortion to the rendered file.

If you're using a hardware processor that is time-based (such as a delay or reverb), you'll need to watch the input/output levels *and* the wet-to-dry mix control of the processor. Most time-based hardware devices have a knob to control wet-to-dry levels, but some might have that control in software (accessible from a menu) or might also be set to pass only wet signal. Because passing some of the dry signal will be paramount when using time-based devices with the External Gear plug-in, make sure you know how to set the hardware processor wet/dry balance.

Using Legacy, VST, and Third-Party Plug-Ins

The layout of your plug-in folders might look very different from mine. Because I'm using the Mac version while writing this book, the three main folders I see when I click on an effects slot are ASIO Plug-Ins, VST, and VST-3. However if you're working on a PC and/or you have third-party plug-ins installed, there will be more plug-ins from which to choose.

Figure 14.39 shows one of my Windows-based mastering computers. You can see that there's a Legacy folder and a UAD folder. (There's also a DirectX folder, but because this plug-in standard is all but dead, I won't go over it.)

Legacy Plug-Ins

Because WaveLab has been available on the Windows platform since the mid-1990s, and some of you have upgraded from a previous version of WaveLab, you can access all of the older plug-ins from the Legacy folder. This will allow you to access the older versions of the more modern VST3 plug-ins. (There are no Legacy plug-ins for the Mac, because WaveLab 7 is the first version that runs on the Mac platform.)

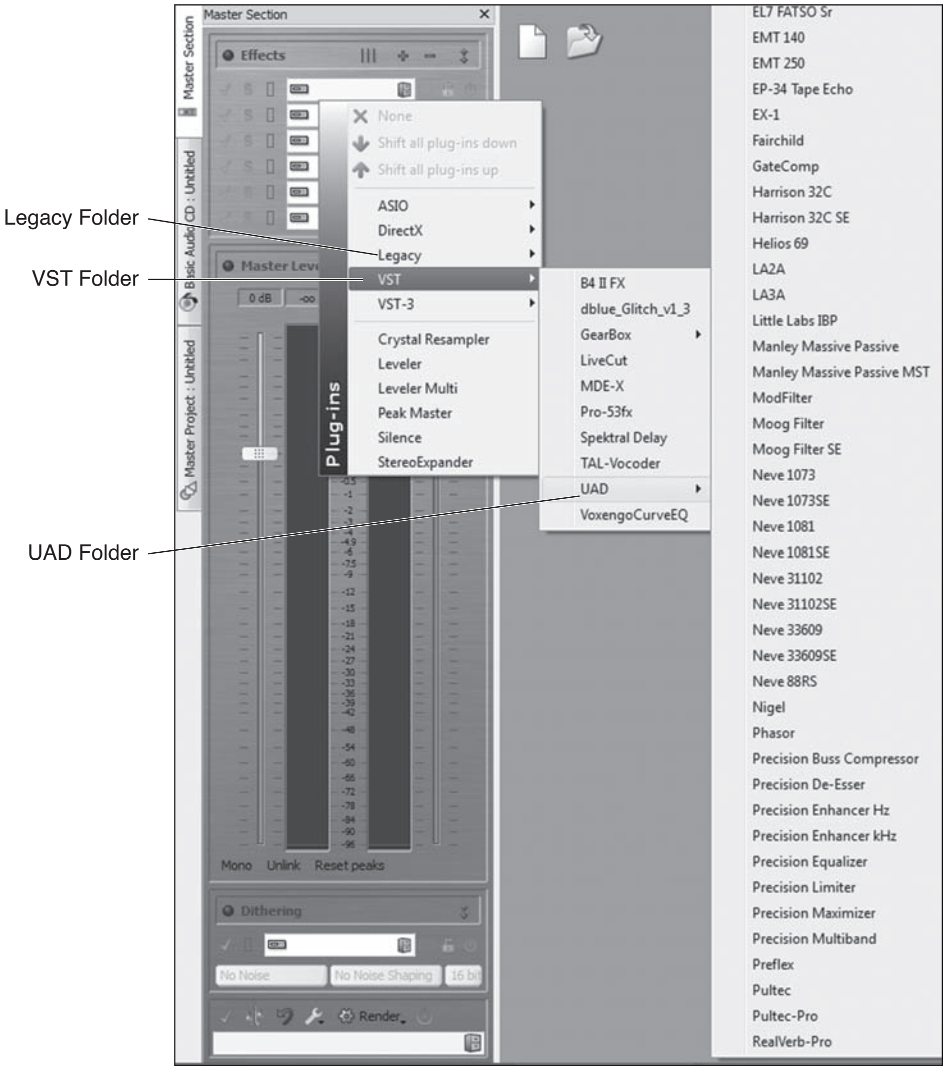

Figure 14.39 Additional folders on a PC with third-party plug-ins.

VST and Third-Party Plug-Ins

The Steinberg VST plug-in standard stimulated a lot of third-party plug-in development. Because of that, there are some fantastic plug-ins you can add to WaveLab. When you run the installer for the third-party plug-ins, they'll usually be placed into a folder that WaveLab can see. So the next time you launch WaveLab, it will detect those plug-ins so that you can use them in the Master Section.

I'm a big fan of the Universal Audio UAD series of plug-ins. Currently, they are all VST2 compatible, not VST3. For that reason, I need to find them in the VST folder (refer to Figure 14.39), not the VST-3 folder. Finding out what VST standard your plug-ins are based on will help you find them in WaveLab after you've installed them.

VST Plug-In Settings

WaveLab allows you to customize where your plug-ins are located, which ones are loaded and ignored, and how the knobs on the plug-in control panels will react to mouse editing. Go to the Options pull-down menu and select VST Plug-In Settings.

At the top of the Plug-In Settings dialog box, shown in Figure 14.40, is a check box for Search Standard VST Plug-In Shared Folder. On a PC, that's usually C:\Program Files\Steinberg\VSTPlugins. On a Mac, it's usually /Library/Audio/Plug-Ins/, and then there are folders for both VST and VST3. If you don't want WaveLab to look in those folders, uncheck this box.

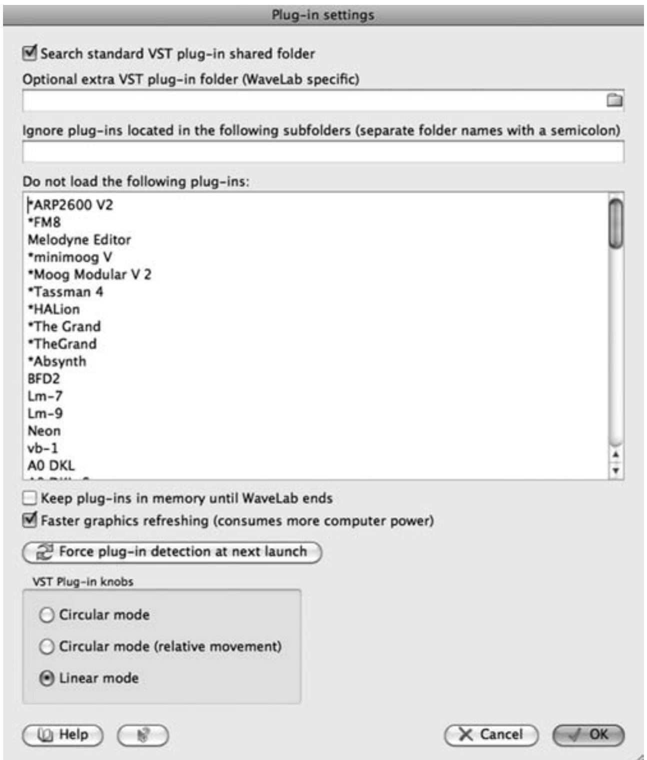

Figure 14.40 Plug-In Settings dialog box.

The next setting allows you to define the location of your Optional Extra VST Plug-In Folder (Wavelab Specific). Sometimes, a third-party developer will have their installer put the plug-ins into a different VSTPlugins (Windows) folder. If you'd like to use those plug-ins in WaveLab, you might need to define the folder in this field. Or you can copy and paste them into the standard VST plug-ins folder by using Windows Explorer or the Mac Finder.

If you're doing your own customized plug-in folder management, and you've defined that location as your Optional Extra VST Plug-In Folder (WaveLab Specific), you can also tell WaveLab to ignore the subfolders that you list in the Ignore Plug-Ins Located in the Following Subfolders (Separate Folder Names with a Semicolon) field. As the field name implies, the names of the folders will need to be separated with a semicolon (Folder 1; Folder 2; Folder 3; and so on).

If you know that there are some plug-ins in the standard VST plug-ins folder that you don't want to load, you can type the plug-in name(s) into the Do Not Load the Following Plug-Ins list. Each plug-in name will need to be typed verbatim and separated by a carriage return (Enter key). Because WaveLab detects and analyzes all the plug-ins for compatibility, your list should resemble Figure 14.40. You can see that, because VST instrument and synthesizer plug-ins are not compatible with an audio editing program like WaveLab, all of my synthesizer plug-ins are already added to that list and will therefore be ignored. However, if there's a problematic plug-in that is crashing or causing other problems but is supposed to be WaveLab-compatible, you can enter its name here so that WaveLab will ignore it until the developer can update the plug-in.

The Keep Plug-Ins in Memory until WaveLab Ends option will keep plug-ins loaded in memory even when they have been removed from the Master Section effects slots. If you're loading and unloading plug-ins a lot, this might be a good check box to enable. However, because the memory required to keep unused plug-ins can cause performance issues, especially on RAM-starved computers (2 GB or less), you should probably leave this at the default setting of unchecked.

The Faster Graphics Refreshing (Consumes More Computer Power) option is checked by default. This allows the visual meters and graphs on the plug-in control panels to respond more quickly. Usually, you'd want more accurate metering. But if you're relying on batteries for power or the increased fan noise of your computer is distracting, you might need to disable this setting.

Force Plug-In Detection at Next Launch Button

Clicking this button will require WaveLab to re-detect and reanalyze the plug-in folder(s) the next time you launch WaveLab. If you've installed new plug-ins or updated plug-ins that were deemed incompatible (therefore, not loaded), you can click this button and then quit and re-launch WaveLab. Your plug-ins should appear the next time you look for them in the Master Section.

VST Plug-In Knobs Settings

At the bottom of the dialog box are three different VST knob behaviors. Circular mode will require you to make circular mouse movements when turning a knob clockwise or counterclockwise. Circular mode will snap the position of the knob to where the pointer is clicked. But in Circular Mode (Relative Movement), you can make adjustments to the knob from anywhere on it, and it won't snap to the pointer position. (It's similar to turning a knob with your fingers and then moving your fingers and moving the knob again.) Linear mode is the default knob behavior and was described earlier in this chapter.

Note: When you change the VST knob settings, the new behavior won't work on plug-ins that were already loaded into an effects slot. You can engage the new knob setting by unloading and reloading the plug-in(s) or by closing and re-launching WaveLab.

Plug-In Organization

You can customize how and where different plug-ins will appear in the Master Section effects slots. Go to the Options menu and select Plug-In Organization.

The Plug-In Organization dialog box, shown in Figure 14.41, is divided into six columns with some command buttons on the right-hand side. The plug-in names will appear in the first column and inside their default group folder.

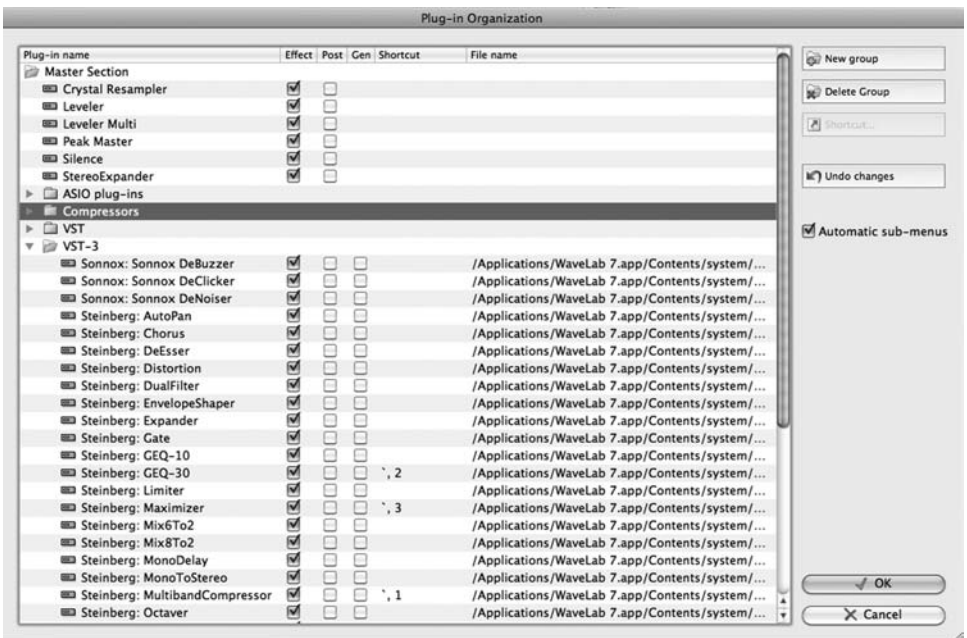

Figure 14.41 Plug-In Organization dialog box.

The Effect check box will allow a plug-in to appear either in Master Section or Audio Montage effects slots. The Post check box will allow a plug-in to appear in the Master Section Dithering slot (post-fader).

Then Gen check box will enable the plug-in to be displayed in its Generic Editor style control panel. This will reduce the plug-in's fancy control panel to a series of horizontal sliders.

The Shortcut column allows you to define keyboard shortcuts or MIDI commands from a MIDI controller to load plug-ins into a highlighted effects slot. In

Figure 14.41, you'll notice that the MultibandCompressor, the GEQ-30, and the Maximizer have `, 1; `, 2; and `, 3 as their respective shortcuts. ( ` is the tilde key located to the left of the number 1 key on the top row of numeric keys.) With those shortcuts, I can click on a Master Section effects slot, type the shortcut, and load the plug-in. Double-clicking in the plug-in Shortcut column will open a dialog box for programming the keys and MIDI commands.

Note: The shortcuts work only with the Master Section and not the Audio Montage effects slots.

The File Name column shows you where the plug-in is located on your computer.

Adding a New Group

Sometimes it's nice to have your plug-ins show up by their nature, not in alphanumeric order. For example, in Figure 14.41, you'll notice a group folder named Compressors. I added that group folder by clicking the New Group button on the right-hand side of the dialog box. Then I named the group Compressors. If I wanted to arrange all of my compressor plug-ins into that folder, I would click and drag each plug-in from the other groups into the Compressors group. The next time I clicked on an effects slot, a Compressors folder would be visible along with the ASIO Plug-Ins, VST, and VST-3 folders. You could do the same thing with limiters, chorus, EQs, and so on.

WaveLab-Specific Plug-Ins

The following six WaveLab-specific plug-ins will not appear in any other VST3-compatible program. Although most of them fall into an aforementioned category, I will describe them here individually.

Crystal Resampler

The Crystal Resampler is a high-quality plug-in that will convert the sample frequency of an audio file. One common situation would be to take files that were originally recorded at 44.1 kHz and convert them to the video-compatible frequency of 48 kHz, or vice versa. Another example is to take audio files recorded at ultra-high sample frequencies (such as 96, 192, and 384 kHz) and resample them to an audio CD–compatible frequency of 44.1 kHz.

Because you'll be using the Crystal Resampler in the Master Section, the processing happens in real time.

The appearance of the control panel (see Figure 14.42) varies depending on whether you're using WaveLab 7.0 (left) or 7.1 (right). But the functionality and processing are identical. The difference is merely cosmetic.

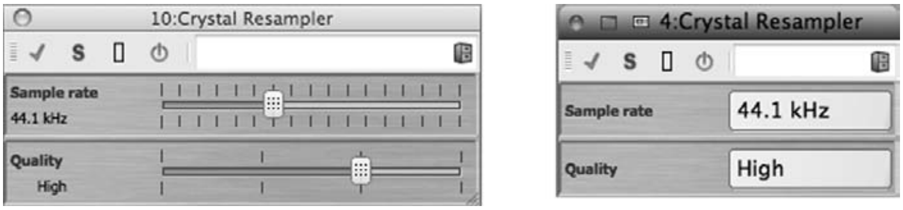

Figure 14.42 The Crystal Resampler control panels.

Sample Rate

This sets the desired sample frequency to which you'd like the file to be resampled. The choices range from 6 kHz to 384 kHz.

Quality

This will allow you to customize the quality of the resampling. The settings include Preview (fast), Standard, High, and Ultra (slow). A modern computer usually can process at the Ultra setting, and that is the setting I recommend. However, if you're using a lot of other plug-ins that are drawing a lot of power, and the Ultra setting is causing dropouts, pops, or clicks, you can use a lower setting. But when it comes time to render the results, make sure you move the setting back to Ultra.

Leveler

The Leveler (see Figure 14.43) is a dynamic processor in that it affects the audio volume. However, it does not provide any compression or limiting, as other dynamic processors do. Instead, it allows you to adjust the volume on the left and right channels separately. That's very useful if your audio file is louder or quieter on one channel than on the other.

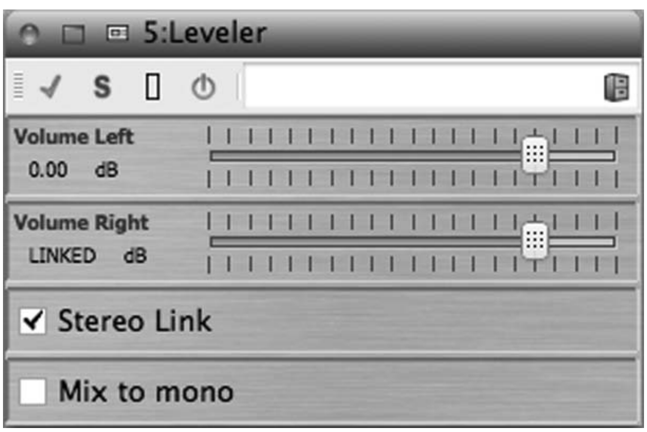

Figure 14.43 The Leveler control panel.

Volume Left

This is the output level fader for the left channel. It is adjustable from –48 dB to 12 dB.

Volume Right

This is the output level fader for the right channel and is also adjustable from –48 dB to 12 dB.

Stereo Link

With this setting enabled, the Volume Right fader is linked to the Volume Left fader. However, the visible position of the Volume Right fader will not reflect the Volume Left fader position.

Mix to Mono

When enabled, both the left and right audio channels are mixed together to create a monophonic output.

Leveler Multi

The Leveler Multi (see Figure 14.44) basically is a simpler version of the Leveler; however, it has more than two inputs. Therefore, it can be used on a multichannel Audio Montage like a 5.1 surround-sound configuration. It also can be used on mono and stereo audio files or Montages.

Figure 14.44 The Leveler Multi control panel.

Volume

This is the output level fader for the entire plug-in. It is adjustable from –48 dB to 12 dB.

Peak Master

The Peak Master (see Figure 14.45) is basically a limiter. As its name implies, it is designed to limit the peak volume, thereby allowing more overall gain to be applied to the audio without clipping. Because it is a limiter, it's best to place the Peak Master in the last effects slot.

Input Gain

This controls how much signal comes into the plug-in. The setting is variable from –12 dB to 24 dB.

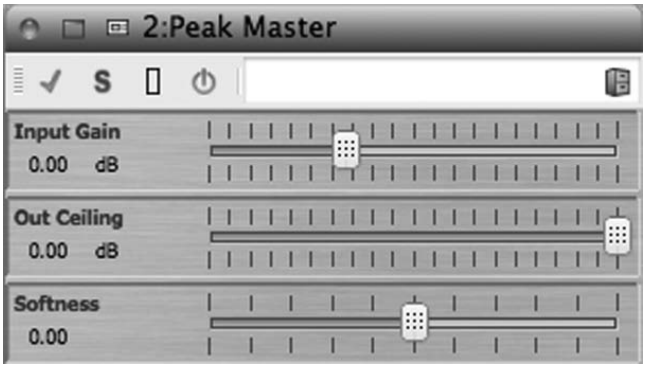

Figure 14.45 The Peak Master control panel.

Out Ceiling

This sets the maximum level allowed at the output of the plug-in. The setting is variable from −18 dB to 0 dB.

Softness

This controls how softly the peaks are rounded. It has arbitrary values from +/−5. Increasing the Softness will make the peaks more tube- or tape-like, while lowering the Softness will create fewer audible limiting artifacts.

Silence

The Silence plug-in (see Figure 14.46) is a dynamic processor, but its job is to add a very precise amount of absolute silence (−∞ dB) at the beginning and/or end of an audio file. Although you can use it as a real-time plug-in, the silence will not be apparent during playback; the silence will be added only during the rendering process.

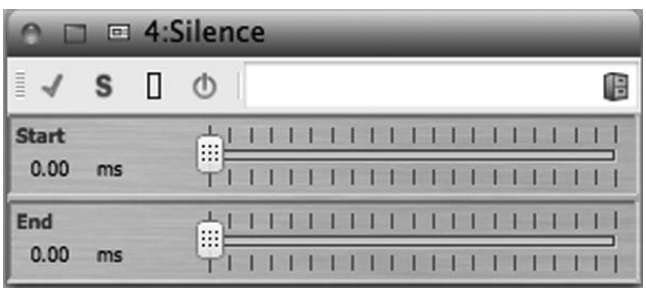

Figure 14.46 The Silence control panel.

The Silence plug-in is commonly used in the Batch Processor Workspace (see Chapter 19, "The Batch Processor Workspace") for adding silence to a series of audio files.

Start

This controls the amount of silence that will be added to the start or beginning of the audio file. It is adjustable from 0.00 milliseconds to 60,000.00 milliseconds (60 seconds).

End

This control is identical to the Start control, but the silence will be added to the end of the audio file.

StereoExpander

StereoExpander (see Figure 14.47) is basically a modulation effect. It works best on audio files that have very similar left- and right-channel data (which would make it sound monophonic) and adds more stereo separation.

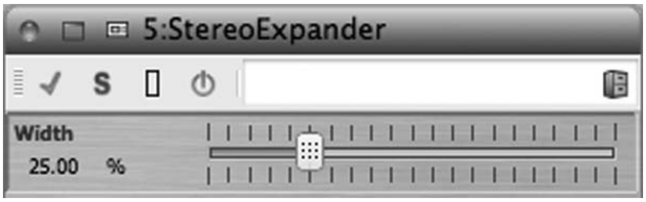

Figure 14.47 The StereoExpander control panel.

Width

This control defines how much artificial stereo separation is added to the sound from 0% to 100%. Values below 25% will usually create a pleasant stereo effect. Values above 25% will increase the stereo separation but can introduce more audible processing artifacts.

Dithering

Dithering is a process by which bits are removed from an audio file, thereby making it compatible with audio devices that require a lower bit depth. For example, an audio CD requires a 16-bit audio file format. Therefore, you cannot burn 20-, 24-, or 32-bit (or higher) audio files onto an audio CD unless you dither them down to 16-bit.

Intern

The name of this plug-in is an abbreviation of Internal (see Figure 14.48). Basically, it's WaveLab's built-in dithering plug-in. Unlike other plug-ins, Intern does not have a separate control panel. Rather, it simply enables the Noise Type, Noise Shaping, and Output Bit Resolution settings in the Dithering slot (located in the Master Section).

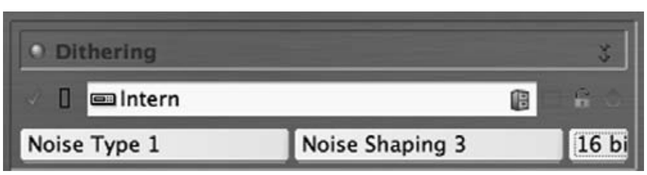

Figure 14.48 The Intern settings.

Noise Type

The term "noise" is a little misleading. It's not the kind of noise you would associate with hiss or buzz. Dithering noise is the process by which bits are removed from the quiet sections of the original audio file. So when you are selecting different noise types, the audible difference is hardly discernable unless you're auditioning the quiet (levels below −18 dB) parts of the audio file.

There are three different noise types: No Noise, Noise Type 1, and Noise Type 2.

Noise Shaping

There are four different noise shapes: No Noise Shaping and Noise Shaping 1 through 3. Different settings will have different results depending on the frequency content of the quiet part of the audio file.

Output Bit Resolution

This setting allows you to define the bit depth of the rendered audio file. It is variable from 8-, 16-, 20-, and 24-bit depths.

The Apogee UV22HR

The UV22HR (see Figure 14.49) was developed by Apogee and is one of the most widely used and well-respected dithering processes in the industry. Steinberg has licensed the Apogee technology and includes the UV22HR in WaveLab.

Figure 14.49 The Apogee UV22 High-Resolution dithering plug-in.

The UV22HR is not only a state-of-the-art dithering plug-in, but it's also one of the easiest plug-ins to use.

Output Bits

This is where you click on the desired bit depth of the rendered file. For example, if you're converting 32-bit audio files to 24-bit, click on the 24 button. If you're converting 24-bit audio files to 16-bit, click on the 16 button.

Dither Level

This setting is a little trickier. But frankly, I've never used any setting other than Hi, which is the default setting. The Lo setting applies the lowest level of dithering noise, while the Auto Black setting basically disables the dithering process whenever the playback volume reaches $-\infty$.

New WaveLab 7.1 Plug-Ins

One of the most wonderful things about software is that when a manufacturer updates the software, they usually include new functionality. WaveLab 7.1 is no exception. There is a new Stereo Tools plug-in, along with two simplified versions of the Stereo Tools functionality.

Stereo Tools

The Stereo Tools plug-in (see Figure 14.50) provides real-time phase inversion, channel swapping, and M/S (mid/side) decoding. (For more information about inverting phase, refer to Chapter 7.) M/S audio files are created by recording with two microphones that have specific characteristics. The mid microphone (usually a cardioid, directional mic) is pointed directly at the sound source, while a microphone with a figure-eight pickup pattern is turned 90 degrees and placed so that the diaphragms are oriented at the same distance from the sound source. Because the equal distance from the source is critical, the mics usually are positioned with the mid mic directly over the side mic.

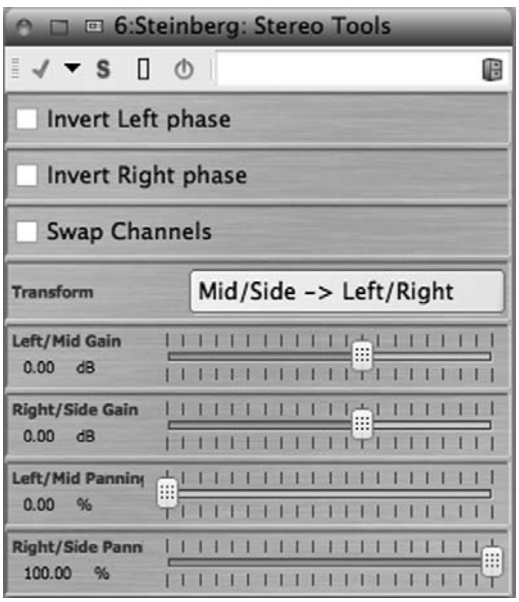

Figure 14.50 The Stereo Tools control panel.

For a more detailed description of M/S and other microphone techniques, check out *Big Studio Secrets for Home Recording and Production* (Course Technology PTR, 2010).

As its name implies, the Stereo Tools plug-in is functional only when used on stereo audio files.

Invert Left Phase and Invert Right Phase

These settings allow you to invert the phase of the left, right, or both audio channels.

Swap Channels

This setting will reroute the audio file's left channel to the right output, while simultaneously rerouting the right channel to the left output.

Transform

The Transform setting should be changed from the default setting of Nothing only when using the Stereo Tools plug-in to decode M/S stereo files. Left/Right -> Mid/Side will decode a stereo file to M/S, while Mid/Side -> Left/Right will decode an M/S file to stereo.

Left/Mid Gain

This controls the volume of the left or mid channel volume, depending on the Transform setting (see above).

Right/Side Gain

This controls the volume of the right or side channel volume, depending on the Transform setting (see above).

Left/Mid Panning

One of the advantages of M/S recording is being able to virtually reposition the microphone after the recording. Therefore, this controls the placement of the left or mid channel in the stereo field.

Right/Side Panning

This controls the placement of the right or side channel in the stereo field.

The Two Additional M/S Plug-Ins

WaveLab 7.1 also includes two M/S plug-ins that are basically simpler versions of the M/S functions found in the Stereo Tools plug-in. These are the LR -> M/S and M/S -> LR plug-ins. They offer the M/S and left/right decoding of the Stereo Tools plug-ins, but that's all they do. They're so basic, in fact, that they don't have a control panel of their own. Therefore, the Control Panel button on the Master Section will be grayed out.