

36B

Audio Effects

After Effects comes with several audio processing effects. They are all located in the menu item Effect > Audio. We discussed the basics of audio plus the Stereo Mixer effect in the book in Chapter 36; here we'll go over some of the other effects, including how they work and some suggested applications for their use. Of course, experimentation is the best way to master them.

Our personal bias is more toward audio “sweetening” than special effects creation, and that will be our focus in the effects we discuss. We've grouped together effects with similar purposes (such as those that work with echoes, or that alter the tonal balance of a sound). Of course, almost any effect can be driven to extremes to create weird sounds, and again, that's where experimentation comes in.

Delay

This plug-in is used to create echo effects. When it's rendering, it creates a copy of the original sound and plays it back delayed in time compared with the original. Delay Amount determines the relative volume of this echo. Feedback decides how much of this delayed sound to feed back through the chain, creating subsequent echoes. The higher the Feedback amount, the louder the subsequent echoes, and the longer it will take the final effect to die away. This is one of the reasons you might need to artificially lengthen sounds (see Chapter 36), to prevent the echoes from getting cut off prematurely.

You can time the echoes produced by Delay to specific numbers of frames for rhythmic animation. All it takes is a little math (really):

$$[\text{desired delay (in frames)} \div \text{frame rate}] \times 1000 = \text{delay time parameter (msec)}$$

The maximum delay time is 5 seconds (5000 milliseconds), which works out to just under 150 frames at 29.97 frames per second (fps).

[Ex.01] demonstrates an echo trick using Delay. It also is an example of using Time Remapping not just to extend the length of a sound file, but to “freeze” the source audio at a specific point, letting the layer continue in time so the echoes generated can continue past our freeze point.

An overview of some of the more useful audio effect plug-ins.

TIP

Audio Preview

Press . (decimal point key on keypad) to preview the audio from the current frame forward; as of CS5, the optional Mac OS X shortcut is Control + period. The duration of this preview is set in Preferences > Previews. Set it to at least 30:00 for the examples in this chapter.

Example Project

Explore the 36B-Example Project.aep file as you read this bonus chapter; references to [Ex.##] refer to specific comps within the project file.

Reverb

When we hear reverberation, we're actually hearing thousands of individual echoes blending together. Instead of the archetypal echo case in which your voice bounces off a canyon wall and bounces back to you, inside more normal rooms your voice (or any sound) is scattering out in all directions, bouncing off any surface it meets, and occasionally zooming past your ears on its various ways to other surfaces to bounce off. Get enough of these echoes and reflections together, and they start to smear into an overall ambiance rather than distinct sounds. The sound loses some energy with each bounce, eventually decaying away into silence. How much energy is lost depends on the surfaces in the room – for example, reverberation takes longer to die away in a marble bathroom than in a carpeted living room with heavy drapes.

Reverb is simulated in software by setting up a number of individual echoes of varying lengths, then feeding them back on themselves to get the additional reflections. After Effects' Reverb effect provides the following parameters to control the reverb simulation:

Reverb Time adjusts the spacing between the original reflections; longer times simulate larger rooms (where there are longer distances for the sound to travel between walls).

Diffusion adjusts how random these individual reflections are – less diffuse means a more orderly room with patterns to the reflections; more diffuse means the reflections are more ragged in timing.

Decay is how strongly, and therefore often, the reflections bounce. Use smaller decay values to simulate carpeted and draped rooms where the reverb dies away quickly, and larger decay values to simulate glass and stone where the reverb would linger much longer.

Brightness controls a very important but often overlooked parameter of reverb simulation: that imperfect real-world reflective surfaces tend to attenuate (reduce in level) higher frequencies more than lower frequencies. Crank this value up for more metallic sounds; keep it low for more realistic, organic sounds.

The default settings for Reverb are pretty good. It does not have quite the quality of a more expensive dedicated hardware or software reverb, but it can help add a sense of room (or distance) when it's used in small Wet amounts. Tweak the Reverb Time first to change the virtual room size, followed by the Decay to control how echoey it is. Varying Diffusion too much from its default quickly makes the simulation break down. Also, note that Reverb takes longer to render than most other audio effects.

fx Reverb	Reset
▶ Reverb Time (ms)	100.00
▶ Diffusion	75.00%
▶ Decay	25.00%
▶ Brightness	10.00%
▶ Dry Out	0.00%
▶ Wet Out	30.00%

TIP

Sound Check

If you want to hear just what the reverb is doing, set the Dry Out to 0% and the Wet Out to around 30% to 70%, depending on how loud the original audio is. This is shown in [Ex.02].

Dry Out and Wet Out

Many of the audio effects feature a pair of parameters called Dry Out and Wet Out. This is because many of the processings sound best if you mix together the unprocessed sound with the processed version. The Dry Out is the original version; the Wet is the processed one. Think of it as Blend With Original, but with more control.

Most of the effects sound more realistic with relatively high Dry and low Wet amounts; some, like Flange/Chorus, work better set closer to a 50/50 mix. A nice trick is to fade up the Wet Out from 0 to some nominal setting over time, causing the effect to fade in without reducing the volume of the original sound.

Flange & Chorus

A funny thing happens when an audio file is delayed only a very short amount and mixed back in against itself. Rather than hearing two distinct sounds, they start to cancel or reinforce just certain frequency components of each other. This is what the Flange & Chorus effect produces.

When the value for Voice Separation Time is set to roughly 10 milliseconds or shorter, as in [Ex.03a], the phenomenon is known as *flanging* – so called because it was originally created by playing two identical audio tapes in synchronization, then dragging your hand on the tape reel flange of one of the copies, causing this slight delay. Some describe it as a “jet taking off” effect.

Voice Separation Time values of roughly 10 to 50 milliseconds, as in [Ex.03b], produce a phenomenon known as *chorusing* because it sounds like multiple singers or identical instruments trying to sound exactly the same note slightly out of time with each other. The result is a more watery sound. Larger values get into very loose chorusing, to the point of being called a “slap back echo” – you can hear this in [Ex.03c] – because it sounds like an echo bouncing off a very hard object nearby.

Typically, only one Voice would be used to create this effect, but this parameter can be increased to make it deeper and more dramatic. Note that each Voice is delayed by the Voice Separation Time, meaning the total delay is set by the number of Voices multiplied by the Voice Separation Time. To keep a flange or chorus sound, you may need to reduce the Time as you increase the number of Voices to avoid hearing a slapback echo effect. More Voices will also require you to reduce the Wet Out amount lest you start overloading. Enabling the Voice Phase Change parameters emphasizes some of the frequency cancellation effects at lower Voice Separation Time settings.

The real secret to this class of effects is a touch of Modulation Depth and Rate. This causes the Voice Separation Time to wander (modulate), resulting in a slightly watery, unstable sound. Modulation Rate values of under 1.0 are most common; higher values produce more watery or even drunken effects. A touch of this effect will help add depth and “magic” to some sound effects such as cymbal crashes and jet engine takeoffs; some also like to use it on vocals.

Bass & Treble

These are basic tone controls. They affect a broad range of frequencies: Bass affects those below 500 Hz; Treble affects those above 1000 Hz. When they’re set to boost low (Bass) and high (Treble) frequencies, they have relatively subtle effects; they have a much more drastic effect when they are set to cut (negative values).

fx Flange & Chorus		Reset
▶	Voice Separation Time (ms)	2.00
▶	Voices	1.00
▶	Modulation Rate	0.30
▶	Modulation Depth	100.00%
▶	Voice Phase Change	0.00
		<input type="checkbox"/> Invert Phase
		<input type="checkbox"/> Stereo Voices
▶	Dry Out	50.00%
▶	Wet Out	50.00%

TIP

Intoxication Modulation

If you want to make your audio really sound drunken, try the Modulator effect. This effect is aimed more at special effects creation than normal audio sweetening. Set its Amplitude Modulation to 0 and its Modulation Depth to higher values, such as 20% on up.

fx Bass & Treble		Reset
▶	Bass	100.00
▶	Treble	0.00

Bass and treble boost effects are demonstrated in [Ex.04a] and [Ex.04b]. Boosting the highs can increase intelligibility but can also increase any hiss present; listen carefully. Decreasing highs can make sounds appear more distant, or less intelligible – which can be helpful if you have a music or sound effect track that is distracting too much from the voiceover.

Preview the Bass & Treble parameters with the Effect switch in the Timeline panel toggled off and on to hear their effect. Boosting Bass gives more low-end emphasis but can also quickly result in clipping distortion; again, preview and be ready to pre-lower the volume with the Stereo Mixer effect.

The High-Low Pass and Parametric EQ effects have more precision and more predictable results. However, Bass & Treble is still very useful for a quick, general, subtle enhancement.

High-Low Pass

The definition of a High Pass filter is one that passes frequencies higher than a set cutoff and tries to remove any frequencies present below that cutoff. A Low Pass filter, as you would expect, does the opposite: passes all frequencies below a set cutoff.

Why is this useful? From a purely corrective point of view, unwanted sounds often exist at the extremes of the audio spectrum. For example, traffic noise, wind across a microphone, and even some ventilation system noise often exists below 100–200 Hz in frequency; a High Pass filter with a Cutoff of 100 or so can help remove this while leaving most other sounds untouched. Likewise, hiss often exists at the highest end, above 10 kHz or so; a Low Pass filter with a Cutoff of 10,000 or higher can remove some of it with minimal damage on voice.

More creatively, reducing the portions of the audio spectrum we hear can be used to mimic lower fidelity devices, such as AM radios or the telephone. In these cases, you will probably need to apply this effect twice: one copy set to Low Pass, and one to High Pass. Setting the Cutoff of the High Pass to around 800–1500 Hz and the Cutoff of the Low Pass to around 2000 is a good start at a cheap television or radio effect, as heard in [Ex.05a].

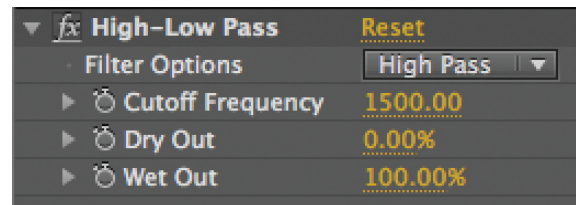
Most of the time, you will want to set the Wet Out to 100% and the Dry Out to 0%. If you are cutting out the frequencies you want, but otherwise find the effect too drastic, start adding in a little Dry Out until you get the desired mix.

This effect tends to reduce the volume of a sound; you may need to boost it back up again using Audio Levels or the Stereo Mixer.

TIP

Controlled Response

If you want effects that generate graphics from audio to react only to certain portions of the audio spectrum, precompose the audio layer and apply audio effects in this precomp to focus on the desired frequencies.



FACTOID

1 is for Gun

By the way, the audio for [Ex.05] comes from an award-winning short movie by Keith Snyder and Blake Arnold called *1 is for Gun*; that's Blake's voice and Keith's music. Information on the making of it can be found at www.woollymammoth.com/gun/.

Parametric EQ

Parametric equalizers allow you to focus on specific bands of frequencies and either boost or remove them. This is the finest knife you have for altering the sound character of audio files. Examples of uses include making a voice more “present” by enhancing just parts of its range, reducing annoying harmonic peaks or *resonances* in a sound file, and cutting out specific narrow spectrum sounds such as fluorescent light buzz or other annoying sounds that are constant in pitch.

After Effects’ Parametric EQ effect gives you three of these filters to work with; you can enable and disable each individually, and each is drawn on the frequency graph in its own color: red for 1, green for 2, and blue for 3. Inside the graph, pitch or frequency goes from low to high as you go from left to right; a curve that goes above the centerline is boosting a range of frequencies, while a curve that goes below that line is cutting them. Here are what the various parameters mean:

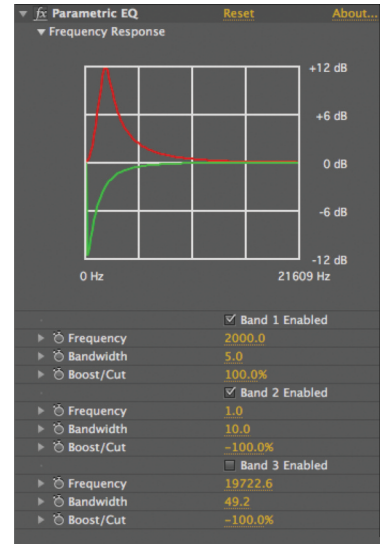
The **Frequency** parameter sets the center of the frequency spectrum range you are going to be targeting. Unfortunately, this slider (and the graph above it) has a linear range; our perception of pitch is actually exponential, meaning this effect is a bit tricky to dial in. You’ll find that the lower quarter of its range has the most audible effect, getting touchier as you get farther to the left.

Bandwidth sets how narrow or wide a range of frequencies you are going to be affecting. It is defined as a percentage of the total bandwidth of the original audio. Since common audio sample rates (such as 44.1 or 48 kHz) have a bandwidth of just over 20 kHz, this means that setting the bandwidth to 1% affects roughly a 200 Hz-wide swath of frequencies. (Note to those with an audio background: This is not the same as “Q” or width on typical audio equipment.)

Again, the problem with this parameter is that the way we perceive pitch is not linear: A 200 Hz range is perceived to be a very wide range at lower frequencies and a very narrow range at higher frequencies. As you set the Frequency parameter higher, you may find you also need to set the Bandwidth parameter higher to get the same perceptual effect.

Boost/Cut sets how much you are increasing or decreasing the prominence of the selected frequency range.

A common working practice with parametric EQs is to set the Bandwidth fairly small (such as 2) and the Boost/Cut to a moderate Boost (such as +6 or higher), then to go back and forth adjusting the Frequency and previewing the results until you have isolated the portion of the sound spectrum you want to work on. You can then adjust the Bandwidth to set how broad



EQ Pointers

What frequencies should you use as starting points for equalization? Voice enhancement/adjustment usually takes place in the 100 Hz to 5 kHz range. Buzzes from bad dimmers and so on tend to happen around 8 kHz and up. You can also search and boost or reduce general musical instrument ranges by adjusting the range of frequencies you treat.

Ground circuit hum in the United States starts at exactly 60 Hz, and in nasty cases, also appears at integer multiples of that frequency. In Europe, the magic number is 50 Hz. Because the narrowest Bandwidth setting (0.1) corresponds to roughly 200Hz, unfortunately, you’ll have trouble making surgical adjustments down that low; you will need a dedicated audio program to go after these problems.

of an effect you want, and set Boost or Cut depending on whether you want to enhance it or remove it.

[Ex.05b] is another pass at our cheap speaker effect, this time using Parametric EQ to make it sound more akin to a telephone.

Tone

Tone can be used to create sounds from scratch. Think of it as the aural equivalent of a solid. It replaces any audio that came with the layer it was applied to, including any previous Tone effects – so you need separate layers if you want to mix the results of multiple Tone effects together. A basic test tone generator is set up in [Ex.06_starter]; open it and experiment with the settings.

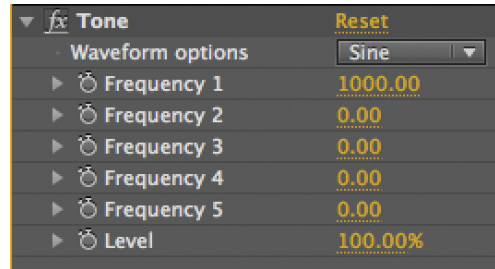
Tone can create up to five pure pitched sounds with a single master volume for all five. The layer's Levels parameter can also adjust its volume. You can disable any of the five component sounds by tuning their Frequency parameters down to 0 Hz. You can set only one Waveform, which is applied to all five component sounds. Their names describe the shape of their waves, and their general sound characters are as follows:

- **Sine** is the purest tone, containing no harmonics above the frequency it is set at.
- **Triangle** sounds just a touch brighter than a sine wave.
- **Saw** is short for Sawtooth and is the most raucous-sounding of the waveforms.
- **Square** is strong too, but has a more “hollow” sound than Saw.

You can also set the fundamental frequency of each component sound. Waveforms other than Sine have additional higher-pitched components called *harmonics* that exist at integer multiples of the sound's fundamental frequency – 1×, 2×, 3×, 4×, 5×, and so forth – so you will often not need to set this frequency as high as you might expect to get a bright or high-pitched sound. Remember that you can hold the Shift key while scrubbing a value to make it jump by larger increments.

At its simplest, you can use Tone to create the audio portion of the bars and tone reference signals that often get laid down at the head of videotapes. Apply Tone to a layer, leave the Waveform at Sine, set Frequency 1 to 1000 and all the other frequencies to 0, and Level to 100%. Now adjust the volume of your reference tone with the layer's Levels. A common –18 dB reference tone is set up in [Ex.06-reference –18dB].

You can also create all sorts of sci-fi radio and electronic sounds by animating the Frequency of Tone. Again, we tend to hear pitches in a logarithmic fashion, rather than linear; that means as you keyframe Frequency to increase, it will seem to rise quickly and then slow down, even if you are using linear keyframes. When you're animating Frequency, try using Easy Ease In or Out for the keyframe with the lower value to get a more natural glide in pitch.



TIP

Sound Keys

One of our favorite plug-ins is Trapcode Sound Keys (distributed by Red Giant Software), which derives keyframe data from user-defined frequency ranges in audio.

CONNECT

RAM Previewing: Chapter 2.

Keyframe interpolation (including Easy Ease keyframe assistants): Chapter 4.

Effects that generate graphics from audio: Chapter 23B.

Time remapping: Chapter 28.

Expressions: Chapter 37 and 37B.