With effects, your mix sounds more like a real “production” and less like a bland home recording. You might simulate a concert hall with reverb. Put a guitar in space with stereo chorus. Make a kick drum punchy by adding compression. Used on all pop-music records, effects can enhance plain tracks by adding spaciousness and excitement. They are essential if you want to produce a commercial sound. But many jazz, folk, and classical groups sound fine without any effects.

This chapter describes the most popular signal processors and effects, and suggests how to use them.

Effects are available both as hardware and software (called plug-ins). To add a hardware effect to a track, you feed the track’s signal from your mixer’s aux send to an effects device, or signal processor (Figure 10.1). It modifies the signal in a controlled way. Then the modified signal returns to your mixer, where it blends with the dry, unprocessed signal.

SOFTWARE EFFECTS (PLUG-INS)

Most recording programs include plug-ins: software effects that you control on your computer screen. Each effect is an algorithm (small program) that runs either in your computer’s CPU or in a DSP card. Some plug-ins come already installed with the recording software; others you can download or purchase on CD, then install them on your hard drive. Each plug-in becomes part of your recording program (called the host), and can be called up from within the host.
You can use plug-ins made by your recording software company or by others. Some manufacturers make plug-in bundles, which are a variety of effects in a single package.

Plug-ins are the usual way to create effects in a DAW. Some DAWs let you configure your audio interface to produce an aux-send signal, which you feed to an external hardware processor. The processed signal returns to the interface and blends with the dry signal.

All the effects described below are available as plug-ins as well as hardware.

**EQUALIZER**

Recall from Chapter 2 that an equalizer (usually in the mixer) is a sophisticated tone control, something like the bass and treble controls in a stereo system. Equalization (EQ) lets you improve on reality: add crispness to dull cymbals or add bite to a wimpy electric guitar. EQ also can make a track sound more natural; for instance, remove tubbiness from a close-miked vocal.

To understand how EQ works, we need to know the meaning of a spectrum. Each instrument or voice produces a wide range of frequencies called its spectrum—the fundamentals and harmonics. The spectrum gives each instrument its distinctive tone quality or timbre.

If you boost or cut certain frequencies in the spectrum, you change the tone quality of the recorded instrument. EQ adjusts the bass, treble, and midrange of a sound by turning up or down certain frequency ranges; that is, it alters the frequency response. For example, a boost (a level increase) in the range centered at 10 kHz makes percussion sound bright and crisp. A cut at the same frequency dulls the sound.
Types of EQ

Equalizers range from simple to complex. The most basic type is a **bass and treble control** (labeled LF EQ and HF EQ). Figure 10.2 shows its effect on frequency response. Typically, this type has up to 15 dB of boost or cut at 100 Hz (for the low-frequency EQ knob) and at 10 kHz (for the high-frequency EQ knob).

With **3-band EQ** you can boost or cut the lows, mids, and highs at fixed frequencies (Figure 10.3). Sweepable EQ is more flexible because you can “tune in” the exact frequency range needing adjustment (Figure 10.4). If your mixer has **sweepable EQ**, one knob sets the center frequency while another sets the amount of boost or cut.

**Parametric EQ** lets you set the frequency, amount of boost/cut, and bandwidth—the range of frequencies affected. Figure 10.5 shows how a parametric equalizer varies the bandwidth of the boosted part of the spectrum. The “Q” or quality factor of an equalizer is the center frequency divided by the bandwidth. A boost or cut with a low-Q setting (like 1.5) affects a wide range of frequencies; a high-Q setting (like 10) makes a narrow peak or dip.

A **graphic equalizer** (Figure 10.6) is usually outside the mixing console. This type has a row of slide pots that work on 5 to 31 frequency bands. When the controls are adjusted, their positions graphically show the resulting frequency response. Usually, a graphic equalizer is used for monitor-speaker EQ, or is patched into a channel for sophisticated tonal tweaking.
Equalizers can also be classified by the shape of their frequency response. Peaking EQ shapes the response like a hill or peak when set for a boost (Figure 10.7). With shelving EQ, the shape of the frequency response resembles a shelf (Figure 10.8). Audio clip 27 at
A filter causes a rolloff at the frequency extremes. It sharply rejects (attenuates) frequencies above or below a certain frequency. Figure 10.9 shows three types of filters: lowpass, highpass, and bandpass. For example, a 10-kHz lowpass filter (high-cut filter) removes frequencies above 10kHz. Its response is down 3dB at 10kHz and more above
that. This reduces hiss-type noise without affecting tone quality as much as a gradual treble rolloff would. A 100-Hz highpass filter (low-cut filter) attenuates frequencies below 100 Hz. Its response is down 3 dB at 100 Hz and more below that. This removes low-pitched noises such as air handler rumble or breath pops. A 1-kHz bandpass filter cuts frequencies above and below a frequency band centered at 1 kHz.

The crossover filter in some monitor speakers consists of lowpass, highpass, and bandpass filters. They send the lows to the woofer, mids to the midrange, and highs to the tweeter.

A filter is named for the steepness of its rolloff: 6 dB per octave (first order), 12 dB per octave (second order), 18 dB per octave (third order), and so on.

**How to Use EQ**

If your mixer has bass and treble controls, their frequencies are preset (usually at 100 Hz and 10 kHz). Set the EQ knob at 0 to have
no effect (flat setting). Turn it clockwise for a boost; turn it counterclockwise for a cut. If your mixer has multiple-frequency EQ or sweepable EQ, one knob sets the frequency range and another sets the amount of boost or cut.

Table 10.1 shows the fundamentals and harmonics of musical instruments and voices. The harmonics given represent an approximate range. Percussion, cymbals, and muted trumpet actually have some
energy up to 80 to 100 kHz. For each instrument, turn up the lower end of the fundamentals to get warmth and fullness. Turn down the fundamentals if the tone is too bassy or tubby. Turn up the harmonics for presence and definition; turn down the harmonics if the tone is too harsh or sizzly.

Here are some suggested frequencies to adjust for specific instruments. If you want the effects described below, apply boost. If you don’t, apply cut. Try these suggestions and accept only the sounds you like:

- **Bass:** Full and deep at 60 to 100 Hz, growl at 600 Hz, presence at 2.5 kHz, string noise at 3 kHz and up. Cut around 200 to 500 Hz for clarity.
- **Electric guitar:** Thumpy at 60 Hz, full at 100 Hz, puffy at 500 Hz, presence or bite at 2 to 3 kHz, sizzly and raspy above 6 kHz.
- **Drums:** Full at 100 to 200 Hz, wooly at 250 to 800 Hz (try cutting in that range), trashy snare at 1 to 3 kHz, attack at 5 kHz. Try cutting toms around 600 Hz to reduce boxiness. Dull cymbals sound more sizzly and crisp with a boost at 10 to 12 kHz.
- **Kick drum:** Full and powerful below 60 Hz, papery at 300 to 800 Hz (cut at 400 to 600 Hz for better tone), click or attack at 2 to 6 kHz.
- **Sax:** Warm at 500 Hz, harsh at 3 kHz, key noise above 10 kHz.
- **Acoustic guitar:** Full or thumpy at 80 Hz, presence at 5 kHz, pick noise above 10 kHz.
- **Acoustic guitar pickup:** To make the guitar sound more “acoustic,” try a narrow cut at 1.2 to 1.5 kHz and maybe some high-frequency cut.
- **Voice:** Full at 100 to 200 Hz (males), full at 200 to 400 Hz (females), honky or nasal at 500 Hz to 1 kHz, presence at 5 kHz, sibilance (“s” and “sh” sounds) around 3 to 10 kHz.

**Example:** Suppose a vocal track sounds too full or bassy. Reach for the LF EQ knob (say, 100 Hz) and turn it down until the voice sounds natural. Or suppose a snare drum sounds dull or muffled. Grab the mid-frequency EQ knob, set it to 5 to 10 kHz, and turn it up until the snare sounds clear and crisp.

Set EQ to the approximate frequency range you need to work on. Then apply full boost or cut so the effect is easily audible. Finally, fine-tune the frequency and amount of boost or cut until the tonal balance is the way you like it.
What if an instrument sounds honky, tubby, or harsh, and you don’t know what frequency to tweak? Set a sweepable EQ for extreme boost. Then sweep the frequencies until you find the frequency range matching the coloration. Cut that range by the amount that sounds right. For example, a piano miked with the lid closed might have a tubby coloration—maybe too much output around 300 Hz. Set your low-frequency EQ for boost, and vary the center frequency until the tubbiness is exaggerated. Then cut at that frequency until the piano sounds natural.

In general, avoid excessive boost because it can distort the signal. Try cutting the lows instead of boosting the highs. To reduce muddiness or enhance clarity, cut 1 to 2 dB around 300 Hz—either on individual instruments or on the entire mix. Don’t boost everything at the same frequency.

**When to Use EQ**

Before using EQ, try to get the desired tone quality by changing the mic or its placement. This gives a more natural effect than EQ. Many purists shun the use of EQ, complaining of excessive phase shift or ringing caused by the equalizer—a “strained” sound. Instead, they use carefully placed, high-quality microphones to get a natural tonal balance without EQ.

The usual practice is to record flat (without EQ) and then equalize the track during mixdown.

Sometimes the instruments need a lot of EQ to sound good. If so, you might want to record with EQ so that the playback for the musicians will sound good. When you play the multitrack recording through your monitor mixer, the recording may not sound right unless the tracks are already equalized. (That’s assuming the monitor mixer in your board has no EQ.)

**Uses of EQ**

Here are some applications for EQ:

- **Improve tone quality.** The main use for EQ is to make an instrument sound better tonally. For example, you might use a high-frequency rolloff on a singer to reduce sibilance, or on a direct-recorded electric guitar to take the “edge” off the sound.
You could boost 100 Hz on a floor tom to get a fuller sound, or cut around 250 Hz on a bass guitar for clarity. Cut around 100 Hz to reduce bass buildup on massed harmony vocals. The frequency response and placement of each mic affect tone quality as well.

Although you can set the EQ for each track when it is soloed, a better way is to set the equalizers when the entire mix is playing. That’s because one instrument can mask or hide certain frequencies in another instrument. For example, the cymbals might mask the “s” sounds in the vocal, making the vocal sound dull—even though it might sound fine when soloed.

Create an effect. Extreme EQ reduces fidelity, but it also can make interesting sound effects. Sharply rolling off the lows and highs on a voice, for instance, gives it a “telephone” sound. A 1-kHz bandpass filter does the same thing. To make a mono keyboard track sound stereo, send it to two mixer channels. Boost lows and cut highs in one channel panned left; cut lows and boost highs in the other channel panned right.

Reduce noise and leakage. You can reduce low-frequency noises—bass leakage, air-conditioner rumble, mic-stand thumps—by turning down the lows below the fundamental-frequency range of the instrument you’re recording. This information is shown in Table 10.1.

For example, a fiddle’s lowest frequency is about 200 Hz, so you’d use a low-cut filter (highpass filter) set to 200 Hz (if possible). This low-cut filter won’t change the fiddle’s tone quality because the filtered-out frequencies are below the fiddle’s lowest frequency. Similarly, a kick drum has little or no output above 9 kHz, so you can filter out highs above 9 kHz on the kick drum to reduce cymbal leakage. Filtering out frequencies below 100 Hz on most instruments reduces air-conditioning rumble and breath pops. Try rolling off the lows on audience mics to prevent muddy bass. To reduce hum, set a parametric EQ for a 24-dB cut, Q of 30, at these frequencies: 60, 120, and 180 Hz (in the United States) or 50, 100, and 150 Hz (in Europe).

Compensate for the Fletcher-Munson effect. As discovered by Fletcher and Munson, the ear is less sensitive to bass and treble at low volumes than at high volumes. So, when you record a very loud instrument and play it back at a lower level, it might
lack bass and treble. To restore these, you may need to boost the lows (around 100 Hz) and the highs (around 4 kHz) when recording loud rock groups. The louder the group, the more boost you need. It also helps to use cardioid mics with proximity effect (for bass boost) and a presence peak (for treble boost).

**Make a pleasing blend.** If you mix two instruments that sound alike, such as lead guitar and rhythm guitar, they tend to mush together—it’s hard to tell what each is playing. You can make them more distinct by equalizing them differently. For example, make the lead guitar edgy by boosting 3 kHz, and make the rhythm guitar mellow by cutting 3 kHz. Then you’ll hear a more pleasing blend and a clearer mix. The same philosophy applies to bass guitar and kick drum. Because they occupy about the same low-frequency range, they tend to mask or cover each other. To make them distinct, either fatten the bass and thin out the kick a little, or vice versa. The idea is to give each instrument its own space in the frequency spectrum; for example, the bass fills in the lows, synth chords emphasize mid-bass, lead guitar adds edge in the upper mids, and cymbals add sparkle in the highs.

**Compensate for mic placement.** Sometimes you’re forced to mike very close to reject background sounds and leakage. But a close mic emphasizes the part of the instrument that the mic is near. This gives a colored tone quality, but EQ can partly compensate for it. Suppose you had to record an acoustic guitar with a mic near the sound hole. The guitar track will sound bassy because the sound hole radiates strong low frequencies. But you can turn down the lows on your mixer to restore a natural tonal balance.

This use of EQ can save the day by fixing poorly recorded tracks in live concert recordings. During a concert, the stage monitors might be blaring into your recording/PA microphones, so you’re forced to mike close to reject monitor leakage and feedback. This close placement, or the monitor leakage itself, can give the recording an unnatural tone quality. In this case, EQ is the only way to get usable tracks.

**“Re-mix” a single track.** If a track contains two different instruments, sometimes you can change the mix within that track by using EQ. Imagine a track that has both bass and synth. By using LF EQ, you can bring the bass up or down without
affecting the synth very much. Mixing with EQ is more effective when the two instruments are far apart in their frequency ranges.

**Improve the tonal balance of an entire mix.** During mastering, you can EQ the stereo mix of each song to make it better, to make the songs on an album sound more similar, or to make the album sound more like commercial albums. An effective tool for this purpose is Harmonic Balancer (www.har-bal.com). Whenever you record, the ideal situation is to use the right mic in the right position, and in a good-sounding room. Then you don’t need or want EQ. Otherwise, though, your recordings will sound better with EQ than without it.

**COMPRESSOR**

A compressor acts like an automatic volume control, turning down the volume when the signal gets too loud. Here’s why it’s necessary.

Suppose you’re recording a female vocalist. Sometimes she sings too softly and gets buried in the mix; other times she hits loud notes and blasts the listener. Or she may move toward and away from the mic while singing, so that her average recording level changes.

To control this problem, you can ride gain—turn her down when she gets too loud; turn her up when she gets too quiet. But it’s hard to anticipate these changes. You might prefer to use a compressor, which does the same thing automatically. It reduces the gain (amplification) when the input signal exceeds a preset level (called the threshold). The greater the input level, the less the gain. As a result, loud notes are made softer, so the dynamic range is reduced (Figure 10.10). Play audio clip 28 at www.elsevierdirect.com/companions/9780240811444.

Compression keeps the level of vocals or instruments more constant, so they are easier to hear throughout the mix, and it prevents loud notes that might clip. Also, it can be used for special effects—to make drums sound fatter, or to increase the sustain on a bass guitar. In pro studios, compression is used almost always on vocals; often on bass guitar, kick drum, and acoustic guitar; and sometimes on other instruments.
Doesn’t compression rob the music of its expressive dynamics? Yes, if overdone. But a vocal that gets too loud and soft is annoying. You need to tame it with a compressor. Even then, you can tell when the vocalist is singing loudly by the tone of the voice. It also helps to compress the bass and kick drum to ensure a uniform, driving beat.

You can avoid vocal compression if the singer uses proper mic technique. He or she should back away from the mic on loud notes, and come in close on soft notes. To tell whether you need a compressor, listen to your finished mix. If you can understand all the words, and no notes are too loud, omit the compressor.

**Using a Compressor**

Normally, you compress individual tracks or instruments, not the entire mix. You want to compress only the stuff that needs it. To compress a stereo mix, you need a 2-channel compressor with a stereo link, which keeps the left–right balance from changing during compression.

Multiband compression (covered later) is usually a better choice for compressing the stereo mix.

Let’s describe the controls on the compressor. Some compressors have few controls; most of their settings are preset at the factory. Figure 10.11 shows a compressor plug-in.

**COMPRESSION RATIO OR SLOPE**

This is the ratio of the change in input level to the change in output level. For example, a 2:1 ratio means that for every 2 dB change in input level, the output of the compressor changes 1 dB. A 20-dB change in input level results in a 10-dB change in the output, and so on.
Typical ratio settings are 2:1 to 4:1. A “soft knee” or “over easy” characteristic is a low compression ratio for low-level signals and a high ratio for high-level signals. Some manufacturers say that this characteristic sounds more natural than a fixed compression ratio.

**THRESHOLD**

This is the input level above which compression occurs. Set the threshold high (about −5 dB) to compress only the loudest notes; set it low (−10 or −20 dB) to compress a broader range of notes. A setting of −10 is typical. If the compressor has a fixed threshold, adjust the amount of compression with the input level control.

**GAIN REDUCTION**

This is the number of dB that the gain is reduced by the compressor. It varies with the input level. You set the ratio and threshold controls
so that the gain is reduced on loud notes by an amount that sounds right. The amount of gain reduction shows up on a meter—3 to 10 dB is typical.

COMPRESSOR GRAPH

Most compressor plug-ins display a compression graph of output level versus input level in dB (Figure 10.11). As the input level increases from zero up to the threshold level, the graph is an upward-sloping straight line in which the change in output level matches the change in input level. Where the input level is above the threshold, the graph transitions to a flatter slope where compression occurs: the change in output level is less than the change in input level. The transition point is called the knee of the compression curve. The transition from no compression to compression can be abrupt (hard knee) or gradual (soft knee).

The slope of the compressed part of the curve (the part on the right) indicates the compression ratio. The more horizontal this part is, the greater the ratio. A horizontal line indicates limiting (explained later in this chapter under the heading Limiter). As you vary the compression ratio and threshold, the compression curve changes accordingly.

ATTACK TIME

This is how fast the compressor reduces the gain when it’s hit by a musical attack. Typical attack times range from 0.25 to 10 msec. Some compressors adjust the attack time automatically to suit the music; others have a factory-set attack time. The longer the attack time, the larger the peaks that are passed before gain reduction occurs. So, a long attack time sounds punchy; a short attack time reduces punch by softening the attack.

RELEASE TIME

This is how fast the gain returns to normal after a loud passage ends. It’s the time the compressor takes to reach 63% of its normal gain. You can set the release time from about 50 msec to several seconds. One-half second to 0.2 second is typical. For bass instruments, the release time must be longer than about 0.4 second to prevent harmonic distortion.
Short release times make the compressor follow rapid volume changes in the music, and keep the average level higher. But because the noise rises along with the gain, short release times can give a pumping or breathing sound. Long release times sound more natural. If the release time is too long, though, a loud passage will reduce the gain during a subsequent quiet passage. In some units, the release time varies automatically, or is factory-set to a useful value.

Some compressors disable the attack and release settings when the compressor is set to RMS or average mode. Those settings are adjusted automatically.

Figure 10.12 shows the effects of compressor attack time and release time on the envelope of a waveform. The gray block represents a musical note fed into the compressor. As Figure 10.12 shows, a long
attack time tends to increase the attack portion of the envelope, giving a sound with a sharper attack or “edge.” A short release time tends to bring up the level in between notes, giving a louder but more fatiguing sound. A release time similar to the note’s decay time increases sustain.

OUTPUT-LEVEL CONTROL

Also called make-up gain, this control is used to increase the output level of the compressor by the amount of gain reduction. For example, if a compressor is causing 6 dB of level reduction, set the make-up gain to 6 dB to achieve unity gain. This also brings up the quiet parts of the track by 6 dB. Some compressors keep the output level constant when other controls are varied.

Spend some free time playing with all the settings so you learn how they affect the sound. Play various instruments and vocals through a compressor, vary the settings, and take notes on what you hear.

Some compressors have a side chain. This is a pair of in/out jacks for connecting an equalizer. To compress only the sibilant sounds on a vocal track, boost the side-chain EQ around 3 to 10 kHz. To compress only the breath pops on a vocal track, boost the side-chain EQ around 20 Hz.

A multiband or split-frequency compressor divides the audio band into three to five bands (bass, mids, treble) and compresses each band separately. That way, the compressor can squash a loud bass note, or soften “s” sounds, without bringing down the overall level. Multiband compression is sometimes applied to the final mix of each song during mixdown or mastering.

Connecting a Compressor

To compress one track in DAW, select a compressor plug-in for that track. Don’t use an aux send for compression. Connect a hardware compressor in line with the signal you want to compress in one of the following ways:

- To compress one instrument or voice while recording: Locate the input module of the instrument you want to compress. On the back of that module, connect the insert send jack to compressor in; connect compressor out to the insert return jack.
(Chapter 11 explains these terms.) Or, take a signal from the input module’s direct out. Feed that into the compressor, and feed the compressor output to the recorder track input.

- To compress a group of instruments while recording: Locate the bus output of the instruments you want to compress. Go from bus out to compressor in, and go from compressor out to recorder-track in. If the bus has insert jacks, you could connect to them instead.

- To compress one track during mixdown: Go from track out to compressor in, and go from compressor out to mixer channel in. Or locate the mixer input module for that track, then find the insert send and return jacks in that module. (There might be a single insert jack with send and return terminals.) Connect the insert send to compressor in; connect compressor out to insert return.

Suggested “Ballpark” Compressor Settings

- Vocals: Ratio 2:1 to 3:1, fast attack, 1/2 second release, set threshold for 3 to 6 dB of gain reduction. Singers with extreme dynamic range might need 12 dB of gain reduction and a ratio of 4:1.

- Bass and drums: Ratio 4:1, slow release, set threshold for 3 to 6 dB of gain reduction on loud bass “pops.” Adjust attack time depending on how much you want to soften the attack. Short attack time = soft attack, long attack time = loud attack.

- Electric guitar: 4:1 to 8:1 ratio, 10-dB gain reduction, 400 msec release.

- To reduce breath pops: Use a multiband compressor. Enable only the lowest frequency band. Try these settings: ratio 30:1, upper frequency 100 Hz, make-up gain 0 dB, attack 1 msec, release 100 msec, threshold –18 dB. Experiment with the threshold setting.

- To reduce sibilance (de-ess mode): First, set an equalizer for a narrow boost around 6 kHz. While playing the vocal track, sweep the frequency up and down until you find the frequency area where sibilance (“s” and “sh” sounds) is exaggerated the most. Note that frequency. Remove the equalizer and insert a multiband compressor. Disable all bands except a band from about
3 to 10kHz. Try to set up a narrower band around the frequency you noted earlier. Suggested settings: ratio 10:1, attack 1 msec, release 100 msec, threshold −32 dB. Experiment with the threshold setting to get the desired amount of de-essing.

- To compress the stereo mix: Try these settings: 2:1, soft knee, attack 20 msec, release 200 msec. Set the threshold to get 5 to 10 dB of gain reduction. If you can hear the compressor squashing the sound, back off the gain reduction unless that’s the sound you want.

Should you compress while tracking or mixing? If you compress while tracking, it will be difficult or impossible to change the amount of compression during mixdown. If you compress tracks during mixdown, you can change the settings at will.

When applying low-cut EQ to a track, insert the EQ before the compressor. Often there is too much bass on a track, and that extra bass will trigger the compressor unless you EQ it out first. When you apply a boost, put the EQ after the compressor so that the boost isn’t compressed.

**LIMITER**

A limiter keeps signal peaks from exceeding a preset level. While a compressor reduces the overall dynamic range of the music, a limiter affects only the highest peaks (Figure 10.13). To act on these rapid peaks, limiters have a very fast attack time—1 microsecond to 1 millisecond. The compression ratio in a limiter is very high—10:1 or greater—and the threshold is set high, say at 0 dB. For input levels up to 0 dB, the output level matches the input. For input levels above 0 dB, the output level stays at 0 dB. This prevents overload in the device following the limiter.

![Figure 10.13](image-url)
A compressor/limiter carries out both of the functions in its name. It compresses the average signal levels over a wide range, and limits peaks to prevent overload. It has two thresholds: one low for the compressor and one high for the limiter.

Limiters can be used to prevent recorder overload during field recording, or to prevent PA power amps from clipping. When you master a program of several mixed songs in a DAW, you might use limiting to reduce the level of signal peaks in the program. Set the threshold about 6 dB below the highest peak level. Then apply normalization, which raises the level of the entire program until the highest peak in the program reaches maximum level. Limiting and normalization create a louder program on your finished CD without compressing the music’s dynamics.

A limiter plug-in with the “look-ahead” feature checks the upcoming audio for peaks. Audio enters a look-ahead buffer, and the limiter measures that audio signal and reacts quickly to reduce the incoming peaks.

NOISE GATE

A noise gate (expander) acts like an on-off switch that removes noises during pauses in an audio signal. It reduces the gain when the input level falls below a preset threshold. That is, when an instrument stops playing for a moment, the noise gate drops the volume, which removes any noise and leakage during the pause (Figure 10.14).

Note: The gate doesn’t remove noise while the instrument is playing.

Where is it used? The noise gate helps to clean up drum tracks by removing leakage between beats. It can shorten the decay time of the
drums, giving a very tight sound. If you’re recording a noisy guitar amp, try a gate to cut out the buzz and hiss between phrases.

How do you use a noise gate? Patch it between a recorder-track output and a mixer line input, or use a gate plug-in in a DAW. Solo the track that you want to gate. Set the gate’s threshold so that noise and leakage go away during pauses. If the gate chops off each note, the threshold is set too high—turn it down. Set the release time as short as possible, but long enough so that you hear the entire note (or tom hit) before the gate cuts off the sound. To fix a boomy kick drum, adjust the threshold until the kick sounds as “tight” as you want. That is, use the gate to shorten the decay portion of the kick-drum’s envelope.

Excellent recordings can be made without gating. But if you want a tighter sound, gates come in handy. Some signal processors have compression, limiting, and noise gating in a single package.

Some gates have a side-chain input or key input. It’s an input for an external signal that controls the gating action. The control signal triggers the output of the gate’s main audio path. For example, you could feed a bass guitar through the noise gate, and gate the bass with a kick-drum signal fed into the side chain. Then the bass will follow the kick drum’s envelope.

**DELAY: ECHO, DOUBLING, CHORUS, AND FLANGING**

A digital delay (or a delay plug-in) takes an input signal, holds it in memory, then plays it back after a short delay—about 1 msec to 1 second (Figure 10.15). Delay is the time interval between the input signal and its repetition at the output of the delay device.

If you listen to the delayed signal by itself, it sounds the same as the undelayed (dry) signal. But if you combine the delayed and dry signals, you may hear two distinct sounds: the signal and its repetition.
By delaying a signal, a processor can create several effects such as echo, repeating echo, doubling, chorus, and flanging.

**Echo**

If the delay is about 50 msec to 1 second, the delayed repetition of a sound is called an echo. This is shown in Figure 10.16 by the two pulses. Echoes occur naturally when sound waves travel to a distant room surface, bounce off, and return later to the listener—repeating the original sound. A delay unit can mimic this effect. Many people use the term delay to mean echo.

In setting up a mix with echo, you want to hear both the dry sound and its echo. You do this by creating an effects loop: from the mixer, to the effects box, back to the mixer. Here's how to set up echo with a hardware mixer and effects unit:

1. On the delay unit, set the dry/wet mix control all the way to wet or 100% mix. Then the output of the delay unit will be only the delayed signal.
2. Suppose you want to use aux1 as the echo control. Connect aux1 send to delay unit IN. Connect delay unit OUT to Bus 1 and 2 IN (or to the effects-return jacks).
3. Find the mixer module for the instrument you want to add echo to.
4. Assign the instrument to busses 1 and 2. Monitor busses 1 and 2.
5. Find the knobs labeled Bus 1 IN and Bus 2 IN. They might be called Aux Return or Effects Return. Turn them up to 0, about three-fourths of the way up.
6. Turn up the aux1 send knob, and there's your echo.

![Diagram of echo setup](image)
The delayed sound mixes with the dry sound in busses 1 and 2. You hear both sounds, which together make an echo. Each aux knob controls the amount of echo on each track, while the effect-return knobs control the overall amount of echo on all tracks that are feeding the echo unit.

To set up echo in a DAW, follow this procedure:

1. Create or use a stereo aux bus that has an Echo or Delay plug-in enabled.
2. In a track that you want to have echo, enable an aux send to that bus.
3. Open the Echo plug-in. Set its dry/wet mix control all the way to wet or 100% mix.
4. In the track that you want to have echo, gradually turn up the aux-send until you hear the desired amount of echo. In the Echo plug-in, adjust the delay parameters for the desired effect.

SLAP ECHO

A delay from about 100 to 130 msec is called a slap echo or slapback echo. It was often used in 1950s rock 'n' roll and rockabilly tunes, and still is used today.

REPEATING ECHO

Most delay units can be made to feed the output signal back into the input, internally. Then the signal is re-delayed many times. This creates a repeating echo—several echoes that are evenly spaced in time (Figure 10.17, and audio clips 29 and 30 at www.elsevierdirect.com/companions/9780240811444). The regeneration (feedback) control sets the number of repeats.
Repeating echo is most musical if you set the delay time to create an echo rhythm that fits the tempo of the song. The formula is

\[
\text{Delay in seconds} = \frac{60}{\text{tempo}}
\]

So if the tempo is 120 bpm, the delay is 0.5 sec (500 msec). That’s one echo per quarter note. Use half that delay to get one echo per eighth note. Use one-third that delay for triplets. A slow repeating echo—0.5 second between repeats; for example, gives an outer-space or haunted-house effect. It often sounds good on lead vocal in a ballad.

**Doubling**

If you set the delay around 23 to 30 msec, the effect is called doubling or automatic double tracking (ADT). It gives an instrument or voice a fuller sound, especially if the dry and delayed signals are panned to opposite sides. The short delays used in doubling sound like early sound reflections in a studio, so they add some “air” or ambience.

Doubling a vocal can be done without a delay unit. Record a vocal part, then overdub another performance of the same vocal part. Mix the parts, pan them both to center, or pan them left and right.

**Chorus**

This is a wavy or shimmering effect. The delay is 15 to 35 msec, and the delay varies at a slow rate. Sweeping the delay time causes the delayed signal to bend up and down in pitch, or to detune. When you combine the detuned signal with the original signal, you get chorusing.

**STEREO CHORUS**

This is a beautiful effect. In one channel, the delayed signal is combined with the dry signal in the same polarity. In the other channel, the delayed signal is inverted in polarity, then combined with the dry signal. Thus, the right channel has a series of peaks in the frequency response where the left channel has dips, and vice versa. The delay is slowly varied or modulated. *Hear a demonstration with audio clip 33 at www.elsevierdirect.com/companions/9780240811444.*
BASS CHORUS
This is chorus with a highpass filter so that low frequencies aren’t chorused, but higher harmonics are. It gives an ethereal quality to the bass guitar.

Flanging
If you set the delay around 0 to 20 msec, you usually can’t resolve the direct and delayed signals into two separate sounds. Instead, you hear a single sound with a strange frequency response. The direct and delayed signals combine and have phase interference, which puts a series of peaks and dips in the frequency response. This is called a comb-filter effect (Figure 10.18). It gives a very colored, filtered tone quality. The shorter the delay, the farther apart the peaks and dips are spaced in frequency.

The flanging effect varies or sweeps the delay between about 0 and 20 msec. This makes the comb-filter nulls sweep up and down the spectrum. As a result, the sound is hollow, swishing, and ethereal, as if the music were playing through a pipe. Flanging is easiest to hear with broadband signals such as cymbals but can be used on any instrument, even voices. Hear a demonstration with audio clip 34 at www.elsevierdirect.com/companions/9780240811444.

Some examples of flanging are on many Jimi Hendrix records, and on the oldies “Itchycoo Park” by the Small Faces and “Listen to the Music” by the Doobie Brothers. The first use of flanging was on “The Big Hurt” sung by Toni Fisher.

Positive flanging refers to flanging in which the delayed signal is the same polarity as the direct signal (Figure 10.18). With negative flanging, the delayed signal is opposite in polarity to the direct

FIGURE 10.18
Flanging (or positive flanging).
signal, which makes a stronger effect. The low frequencies are canceled (the bass rolls off), and the “knee” of the bass rolloff moves up and down the spectrum as the delay is varied. The high frequencies are still comb-filtered (Figure 10.19). Negative flanging makes the music sound like it's turning inside out.

When the flanger feeds some of the output signal back into the input, the peaks and dips get bigger. It's a powerful “science fiction” effect called resonant flanging.

REVERBERATION

This effect adds a sense of room acoustics, ambience, or space to instruments and voices. To know how it works, we need to understand how reverb happens in a real room. Natural reverberation in a room is a series of multiple sound reflections that make the original sound persist and gradually die away or decay. These reflections tell the ear that you’re listening in a large or hard-surfaced room. For example, reverberation is the sound you hear just after you shout in an empty gymnasium.

A reverb effect simulates the sound of a room—a club, auditorium, or concert hall—by generating random multiple echoes that are too numerous and rapid for the ear to resolve (Figure 10.20). Digital reverb is available either in a dedicated reverb unit, as part of a multi-effects processor, or as a plug-in.

The most natural sounding digital reverb is a sampling reverb or convolution reverb, which creates the reverb from impulse-response samples (WAVE files) of real acoustic spaces, rather than from algorithms. One convolution reverb plug-in is SIR at www.knufinke.de/sir/. Free impulse-response samples are at www.noisevault.com and www.cksde.com/p_6_250.htm.
Reverb and echo aren’t the same thing. Echo is a repetition of a sound (HELLO hello hello); reverb is a smooth decay of sound (HELLO-OO-oo-oo).

Multichannel digital reverbs are available for surround sound, both as hardware and software. Examples include: Eventide’s Orville, Sony’s DRE-S777, TC Electronics’ System 6000, Lexicon’s 960L, and Kind of Loud’s RealVerb 5.1 Pro Tools plug-in. Surround reverb plug-ins for Steinberg’s Nuendo platform include Steinberg’s Surround Edition plug-in bundle and TC Works SurroundVerb plug-in.

**Reverb Parameters**

Here are some controls in a reverb unit or plug-in (Figure 10.21):

- **Reverb Time (RT60):** The time it takes for reverberation to decay 60 dB below its original level. Set it long (1–1/2 to 2 seconds) to simulate a large room; set it short (under 1 second) to simulate a small room. Generally you use short reverbs (or no reverb) for fast songs, and long reverbs for slow songs.

- **Pre-delay (pre-reverb delay):** A short delay (10 to 100 msec) before the onset of reverb to simulate the delay that happens in real rooms before reverb starts. The longer the pre-delay, the bigger the room sounds. Using pre-delay on an instrument’s reverb often helps to clarify the sound by removing the onset of reverb from the direct sound of the instrument. If your reverb unit doesn’t have pre-delay, you can create it by connecting a delay unit between your mixer’s aux-send and the reverb input. If your reverb plug-in lacks pre-delay, insert a delay plug-in before the reverb plug-in. *Audio clip 31 at www.elsevierdirect.com/companions/9780240811444 demonstrates reverb: short reverb time, long reverb time, and pre-delay.*

- **Density or diffusion:** A high density setting produces many echoes spaced close together. It gives a smooth decay but...
increases the load on the CPU. A low-density setting produces fewer echoes spaced farther apart, and may be adequate for vocals, synth pads, and organ. Use high density for percussive sounds to prevent grainy-sounding reverb.

- **Damping:** Adjusts the reverb time or decay at high frequencies. Set the damping frequency high (say, 7 kHz) to simulate a hard-surfaced room; set it low (say, 2 kHz) to simulate a soft-surfaced room. The latter is also called a “warm room” reverb.

- **Presets:** Factory-supplied reverb settings of small rooms, auditoriums, halls, and so on. A plate reverb setting duplicates the bright sound of a metal-foil plate, which used to be the most popular type of reverb in pro studios. Unnatural effects are available, such as nonlinear decay, reverse reverb that builds up before decaying, or gated reverb. With gated reverb, the reverb cuts off suddenly shortly after a note is hit. In the 1980s it was often used on a snare drum. A good example is the oldie “You Can Call Me Al” on Paul Simon’s album Graceland.
Reverb Connections

To connect a reverb unit to a hardware mixer, connect a cable from the mixer aux-send to the reverb input. Connect a cable (two for stereo) from the reverb outputs to the mixer aux returns (effects returns or bus inputs). Set the mix control on the reverb unit all the way to wet or reverb. Turn the mixer’s aux-return or bus-in knobs (if any) about two-thirds of the way up, and adjust the amount of reverb on each track with the aux-send knobs. Try to get an overall reverb-send level near 0 on the meter; then fine-tune the aux return level for the desired amount of reverb.

To enable reverb in a DAW, use the same procedure as for setting up echo, but choose a reverb plug-in.

PREVERB

Preverb is reverb that precedes a note rather than follows it. The reverb starts from silence and builds up to a note’s attack (audio clip 32 at www.elsevierdirect.com/companions/9780240811444). When used on a snare drum, preverb gives a whip-cracking kind of sound, like “shSHK!”

Here’s how to add preverb to a snare drum track in a DAW:

1. Set up an aux1 bus with reverb. Set the reverb all the way to wet or 100% mix.
2. Solo the drum track.
3. On the drum track, turn up the aux1 send and set it to pre-fader. Find a good snare hit and play it.
4. Turn down the drum-track fader and play the snare hit. You should hear only the reverb from the aux1 bus.
5. Highlight the snare hit up until the next hit. Export or save the mix (the snare hit) as Drum reverb.wav.
6. Select a blank track and call it Drum Reverb. Import Drum reverb.wav into that track.
7. Select the drum-reverb clip, then select the Reverse processing in your DAW. This reverses the drum-reverb clip.
8. On the original snare-drum track, reset the aux1 send to post-fader and turn it back down. Turn up the track fader where it used to be.
9. Slide in time the reversed drum-reverb clip so that it ends just as a snare-drum hit starts (check the waveforms).

10. Play the reversed drum-reverb track along with the snare-drum track. You should hear preverb.

Some signal processors have a reverse reverb effect in which the reverb comes after the note that produced it, but builds up before it fades out. This isn’t quite the same as preverb. Reverse reverb can upset the musical timing; preverb doesn’t.

**ENHANCER**

If a track or a mix sounds dull and muffled, you can run it through an enhancer to add brilliance and clarity. An enhancer works by adding slight distortion (as in the Aphex Aural Exciter).

**OCTAVE DIVIDER**

This unit takes a signal from a bass guitar and provides deep, growling bass notes one or two octaves below the pitch of the bass guitar. It does this by dividing the incoming frequency by 2 or 4: If you put 82 Hz in, you get 41 Hz out. Some MIDI sound modules have bass patches with extra-deep sound, and 5-string bass guitars have an extra string tuned especially low.

**HARMONIZER**

Basically a delay unit with delay modulation, a harmonizer makes a variety of pitch-shifting effects. It can create harmonies, change pitch without changing the duration of the program, change duration without changing pitch, and many other oddities. You’ve heard harmonizers on radio-station spots when the announcer’s voice sounds like a Munchkin or Darth Vader. Play audio clip 35 at www.elsevierdirect.com/companions/9780240811444.

**VOCAL PROCESSOR**

This device or plug-in can affect the vocal’s inflection, add growls or whispers, correct the pitch, add vibrato, make the voice more-or-less nasal or chesty, and so on. The latest vocal formant-corrected pitch-shifters maintain the voice formant structure when they shift pitch;
this prevents the “chipmunk” effect. An example includes TC-Helicon VoiceModeler plug-in.

Another type of vocal processor is called a channel strip. It includes a high-quality mic preamp or two, plus EQ, compression, gating, de-esser, and perhaps some tube saturation distortion. An example is Focusrite VoiceMaster Pro.

**PITCH CORRECTION**

This plug-in provides automatic or manual pitch correction of a single track (but not of chords). In automatic mode, it corrects flat or sharp notes by changing their pitch to match a musical scale of your choice. In manual mode, you see a graph of the notes’ pitches on your monitor screen, and slide certain notes up or down to correct their pitch. Manual mode is less obvious than automatic. You also can use this effect as a “robotic” effect where the sung notes change pitch in a step-wise, jerky way rather than smoothly. Some pitch-correction plug-ins are Antares Auto-Tune, Celemony Melodyne, TC Helicon Intonator, and Roland’s V-Vocal plug-in for Cakewalk Sonar Producer.

Melodyne Direct Note Access lets you edit individual notes within chords. See www.celemony.com.

**TUBE PROCESSOR**

This device or plug-in uses a vacuum tube or a simulation of one. Tubes have euphonic even-order harmonic distortion, which is claimed to add “richness” or “warmth” when the tube distorts (audio clip 36 at www.elsevierdirect.com/companions/9780240811444). There are tube mics, tube mic preamps, tube compressors, and stand-alone tube processors.

**ROTARY SPEAKER SIMULATOR**

This effect simulates the sound of a Leslie organ speaker, which plays music through rotating horns. It’s a complex sound effect of pitch shifting, tremolo, and phase shifting. The speed and depth of the effect are adjustable.
ANALOG TAPE SIMULATOR

Analog tape saturation is mainly third-harmonic distortion and compression. An analog tape simulator adds this distortion to digital recordings in an attempt to smear or warm up the sound in a pleasant way (Audio clip 37 at www.elsevierdirect.com/companions/9780240811444).

SPATIAL PROCESSOR

Spatial processors enhance the stereo imaging or spatial aspects of a mix heard over two speakers. Some units have joystick-type pan pots, which move the image of each track anywhere around the listener. Other units make the stereo stage wider, so that images can be placed to the left of the left speaker, and to the right of the right speaker. The listener might hear images toward the sides of the listening room. In 5.1 surround systems, this spatial processing is done by surround panning and surround reverb.

MICROPHONE MODELER

Antares and Roland offer a microphone modeler or simulator. You tell it which mic you’re using and which mic you want it to sound like. A wide variety of vintage and current mic simulations are available. Mic modeling comes in three forms: hardware device, plug-in, and firmware (programmed into a chip) in a recorder–mixer.

GUITAR AMPLIFIER MODELER

Another simulator takes the sound of a direct-recorded guitar, and makes the guitar sound like it is played through a guitar amp (Audio clip 38 at www.elsevierdirect.com/companions/9780240811444). Several amp models can be simulated, as well as effects, tone, drive, the mic used to pick up the amp, and the mic’s position.

Two hardware examples of amp modelers are the Line 6 Pod and the Johnson J Station. Amp Farm is a guitar modeling plug-in for Pro Tools. Roland’s DAWs offer COSM mic modeling and guitar-amp modeling.

Guitar processors or guitar stomp boxes can be used on any instrument or vocal to add distortion.
DISTORTION

Some plug-ins add intentional distortion of various types to generally “shred” the sound for a low-fi effect. Two examples are Izotope Trash and Camel Audio CamelCrusher (Figure 10.22).

DE-CLICK AND DE-NOISE

Also called “Audio restoration programs,” these are plug-ins—or stand-alone programs—that can remove the clicks and pops from LP records, or remove hiss and hum from noisy recordings. Examples include: Dart Pro 24, iZotope RX, Bias SoundSoap, and Waves Native Restoration Bundle.

SURROUND SOUND

Recent plug-ins for surround sound are surround panning, surround reverb, and surround encoding/decoding.
MULTI-EFFECTS PROCESSOR

This provides several effects in a single device or plug-in. Some units let you combine up to four effects in any order. Others have several channels, so you can put a different effect on different instruments. With most processors, you can edit the sounds and save them in memory as new programs. At www.elsevierdirect.com/companions/9780240811444 audio clips 29–38 demonstrate various effects on voice.

Some processors offer 100 or more programmable presets with MIDI control over any parameter. For example, with some units you can place an instrument in a simulated room, and use a MIDI controller to continuously change the size of that room. Many signal processors can be controlled by MIDI program-change commands. You can quickly change the type of effect, or effect parameters, by entering certain program changes into a sequencer.

Suppose you want each tom-tom hit in a drum fill to have a different size room added to it. For example, put the high-rack tom in a small room; put the low-rack tom in a concert hall; and put the floor tom in a cave. To do this, first assign a different program number (patch or preset number) to each effects parameter. You do this with the effects device. Then, using the sequencer, punch in the appropriate program number for each note.

A MIDI program-change footswitch lets guitarists call up different effects on MIDI signal processors. By tapping a footswitch, they can get fuzz, flanging, wah-wah, spring reverb, and so on.

A MIDI mapper lets you control some effects parameters with any controller. For example, vary reverb decay time with a pitch wheel, or vary a filter with key velocity.

LOOKING BACK

We’ve come a long way with effects. Looking back over the past few decades, each era had its own “sound” related to the effects used at the time. The 1950s had tube distortion and slap echo; the 1960s used fuzz, wah-wah, and flanging. Much of the early 1970s sounded dry, and the early 1990s emphasized synthesizers, drum machines, and gated reverb. Now vacuum tubes and acoustic instruments are back, along with occasional low-fi (tinny, distorted, or noisy) sounds.
and dry vocals. Whatever effects you choose, they can enhance your music if used with taste.

**SOUND-QUALITY GLOSSARY**

The sound of mic techniques, effects, and EQ in a recording can be hard to translate into engineering terms. For example, what EQ should you use to get a “fat” sound or a “thin” sound? The glossary below may help. It’s based on conversations with producers, musicians, and reviewers over 30 years. Not everyone agrees on these definitions, but they are common. This glossary doesn’t suggest the cause of the sound quality or how to change it; that’s up to you to determine.

**AIRY**—Spacious. The instruments sound like they are surrounded by a large reflective space full of air. A pleasant amount of reverb or early reflections. High-frequency response that extends to 15 or 20 kHz.

**BALLSY** or **BASSY**—Emphasized low frequencies below about 200 Hz.

**BLOATED**—Excessive mid-bass around 250 Hz. Poorly damped low frequencies, low-frequency resonances.

**BLOOM**—Adequate low frequencies. Spacious. Good reproduction of dynamics and reverberation. Early reflections or a sense of “air” around each instrument in an orchestra.

**BOOMY**—Excessive bass around 125 Hz. Poorly damped low frequencies or low-frequency resonances.

**BOXY**—Having resonances as if the music were enclosed in a box. Speaker cabinet diffraction or vibration. An emphasis around 250 to 600 Hz.

**BREATHY**—Audible breath sounds in vocals, flute, or sax. Good high-frequency response.

**BRIGHT**—High-frequency emphasis. Harmonics are strong relative to fundamentals.

**BRITTLE**—High-frequency peaks, or weak fundamentals. Slightly distorted or harsh highs. Opposite of round or mellow. See **Thin**.
Objects that are physically thin and brittle emphasize highs over lows when you crack them.

CHESTY—A vocal signal with a bump in the low-frequency response around 125 to 250 Hz.

CLEAN—Free of noise, distortion, and leakage.

CLEAR—See Transparent.

CLINICAL—Too clean or analytical. Emphasized high-frequency response, sharp transient response. Not warm.

COLORED—Having timbres that aren’t true to life. Non-flat response, peaks, or dips.

CLOUDY—See Wooly.

CONSTRUCTED—Poor reproduction of dynamics. Dynamic compression. Distortion at high levels. Also see Pinched.

CRISP—Extended high-frequency response. Like a crispy potato chip or crisp bacon frying. Often referring to cymbals.

CRUNCH—Pleasant guitar-amp distortion.

DARK—Opposite of bright. Weak high frequencies.

DELICATE—High frequencies extending to 15 or 20 kHz without peaks. A sweet, airy, open sound with strings or acoustic guitar.

DEPTH—A sense of closeness and farness of instruments, caused by miking them at different distances. Good transient response that reveals the direct/reflect sound ratio in the recording.

DETAILED—Easy to hear tiny details in the music; articulate. Adequate high-frequency response, sharp transient response.

DISTANT—Too much leakage. Low direct-to-reverb ratio.


DULL—See Dark.

ECHOEY—Having audible echoes or reverberation.
EDGY—Too much high frequencies. Trebley. Harmonics are too strong relative to the fundamentals. When you view the waveform on an oscilloscope, it even looks edgy or jagged, because of excessive high frequencies. Distorted, having unwanted harmonics that add an edge or raspiness to the sound.

EFFORTLESS—Low distortion, usually coupled with flat response.

ETCHED—Clear but verging on edgy. Emphasis around 10 kHz or higher.

FAT—See Full and Warm. Also, a diffuse spatial effect. Also, smeared out in time, with some reverberant decay. Also, the sound of a snare drum tuned low.

FOCUSED—Referring to the image of a musical instrument which is easy to localize, pinpointed, having a small spatial spread.

FORWARD—Sounding close to the listener. Emphasis around 2 to 5 kHz.

FULL—Opposite of thin. Strong fundamentals relative to harmonics. Good low-frequency response, not necessarily extended, but with adequate level around 100 to 300 Hz.

GENTLE—Opposite of edgy. The harmonics—highs and upper mids—aren’t exaggerated, or may even be weak.

GLARE, GLASSY—A little less extreme than edgy. A little too bright or trebley.

GRAINY—The music sounds like it’s segmented into little grains, rather than flowing in one continuous piece. Not liquid or fluid. Suffering from distortion. Some early A/D converters sounded grainy, as do current ones of inferior design. “Powdery” is finer than “grainy”!

GRUNGY—Lots of distortion.

HARD—Too much upper midrange, usually around 3 kHz. Or, good transient response, as if the sound is hitting you hard.

HARSH—Too much upper midrange. Peaks in the frequency response from 2 to 6 kHz. Or, excessive phase shift.
HEAVY—Good low-frequency response below about 50 Hz. Suggesting great weight or power, like a diesel locomotive or thunder.

HOLLOW—See Honky. Or, too much reverberation, a mid-frequency dip, or comb filtering.

HONKY—The music sounds the way your voice sounds when you cup your hands around your mouth. A bump in the response around 500 to 700 Hz.

IN YOUR FACE—Dry (without effects, without reverb), possibly with compression.

LIQUID—Opposite of grainy. A sense of seamless flowing of the music. Flat response and low distortion. High frequencies are flat or reduced relative to mids and lows.

LOW-FI (low fidelity)—“Trashy” sounding. Tinny, distorted, noisy, or muddy.

MELLOW—Reduced high frequencies, not edgy.

MUDDY—Not clear. Weak harmonics, smeared time response, distortion. Too much reverb at low frequencies. Too much emphasis around 200 to 350 Hz. Too much leakage.

MUFFLED—The music sounds covered up. Weak highs or weak upper mids.

MUSICAL—Conveying emotion. Flat response, low distortion, no edginess.

NASAL—The vocalist sounds as if she is singing with the nose closed. Also applies to strings. Bump in the response around 500 to 1000 Hz. See Honky.

NEUTRAL—Accurate tonal reproduction. No obvious colorations. No serious peaks or dips in the frequency response.

PAPERY—Referring to a kick drum that has too much output around 400 to 600 Hz.

PHASEY—Having phase interference (comb filtering). The sound of a direct signal and its delayed repetition mixed to the same channel (delay usually under 20 msec). Might be due to multiple mics picking up the same source, or one mic picking up direct sound and
delayed reflected sound. Or a delayed signal mixed with itself unde-
layed. Or, some opposite-polarity crosstalk between stereo channels.
Or one monitor speaker is reversed in polarity.

PIERCING—Strident, hard on the ears, screechy. Having sharp, nar-
row peaks in the response around 3 to 10 kHz.

PINCHED—Narrowband. Midrange or upper midrange peak in the
frequency response. Pinched dynamics are overly compressed.

PRESENT, PRESENCE—Adequate or emphasized response around
5 kHz for most instruments, or around 2 to 5 kHz for kick drum and
bass. Having some edge, punch, detail, closeness, and clarity.

PUFFY—Bump in the response around 400 to 700 Hz.

PUNCHY—Good reproduction of dynamics. Good transient
response. Or conversely, referring to highly compressed transients
(especially snare drum and kick drum) that sound like hitting a
punching bag. Sometimes a bump around 5 kHz or 200 Hz.

RASPY—Harsh, like a rasp. Peaks in the response around 6 kHz
which make vocals sound too sibilant or piercing.

RICH—See Full. Also, having euphonic distortion made of even-
order harmonics.

ROUND—High-frequency rolloff or dip. Not edgy.

SHARP—See Crisp, Strident, and Tight.

SIBILANT, ESSY—Exaggerated “s” and “sh” sounds in vocals, too
much output around 5 to 10 kHz.

SIZZLY—See Sibilant. Also, too much highs on cymbals.

SMEARED—Lacking detail. Poor transient response. This may be a
desirable effect in large-diameter mics. Also, poorly focused images.

SMOOTH—Easy on the ears, not harsh. Flat frequency response,
especially in the midrange. Lack of peaks and dips in the response.
Low distortion.

SPACIOUS—Conveying a sense of space, ambience, or room around
the instruments. To get this effect, mike farther back, mix in an ambi-
ence mic, add reverb, or record in stereo. Components that have
opposite-polarity or out-of-phase crosstalk between channels may add false spaciousness.

SQUASHED—Overly compressed.

STEELY—Emphasized upper mids around 3 to 6 kHz. Peaky, non-flat high-frequency response. See Glassy, Harsh, Edgy.

STRAINED—The component sounds like it’s working too hard. Distorted. Inadequate headroom or insufficient power. Opposite of effortless.

STRIDENT—See Harsh and Edgy.

SWEET—Not strident or piercing. Flat high-frequency response, low distortion. Lack of peaks in the response. Highs are extended to 15 or 20 kHz, but they aren’t bumped up. Often used when referring to cymbals, percussion, strings, and sibilant sounds.

THIN—Fundamentals are weak relative to harmonics. Note that the fundamental frequencies of many instruments aren’t very low. For example, violin fundamentals are around 200 to 1000 Hz. So if the 300 Hz area is weak, the violin may sound thin—even if the mic’s response goes down to 40 Hz.

TIGHT—Good low-frequency transient response. Absence of ringing or resonance when reproducing the kick drum or bass. Good low-frequency detail. Absence of leakage.

TINNY, TELEPHONE-LIKE—Narrowband, weak lows, peaky mids. The music sounds like it’s coming through a telephone or tin can.

TRANSPARENT—Easy to hear into the music, detailed, clear, not muddy. Wide, flat frequency response, sharp time response, very low distortion and noise.

TUBBY—See Bloated. Having low-frequency resonances as if you’re singing in a bathtub.

VEILED—The music sounds like you put a silk veil over the speakers. Slight noise or distortion, or slightly weak high frequencies.

WARM—Good bass, adequate low frequencies, adequate fundamentals relative to harmonics. Not thin. Or, excessive bass or midbass. Or, pleasantly spacious, with adequate reverberation at low frequencies. Or, gentle highs, like from a tube amplifier. See Rich.
WASHED OUT—Phase interference from multiple mics picking up the same source. Too much leakage or reverberation.

WOOLY, BLANKETED—The music sounds like there’s a wool blanket over the speakers. Weak high frequencies or boomy low frequencies. Sometimes, an emphasis around 250 to 600 Hz.