

CHAPTER 3

Equalization

The most intuitive effect —

OK EQ is EZ

U need a Hi EQ IQ

MIX SMART QUICK START: Equalization

GOALS

- Fix spectral problems such as rumble, hum and buzz, pops and wind, proximity, and hiss.
- Fit things into the mix by leveraging of any spectral openings available and through complementary cuts and boosts on spectrally competitive tracks.
- Feature those aspects of each instrument that players and music fans like most.

GEAR

- Master the user controls for parametric, semiparametric, program, graphic, shelving, high-pass and low-pass filters.
- Choose the equalizer with the capabilities you need, focusing particularly on the parameters, slopes, and number of bands available.

Home stereos have tone controls. We in the studio get equalizers. Perhaps better described as a “spectral modifier” or “frequency-specific amplitude adjuster,” the equalizer allows the mix engineer to increase or decrease the level of specific frequency ranges within a signal. Having an equalizer is like having multiple volume knobs for a single track. Unlike the volume knob that attenuates or boosts an entire signal, *the equalizer is the tool used to turn down or turn up specific frequency portions of any audio track.*

Out of all signal-processing tools that we use while mixing and tracking, EQ is probably the easiest, most intuitive to use—at first. Advanced applications of equalization, however, demand a deep understanding of the effect and all its

possibilities. Don't underestimate the intellectual challenge and creative potential of this essential mix processor.


When you want more of something spectrally, EQ offers the solution. The successful engineer must also be able to make a related, more difficult judgment: how to identify when a mix element needs *less* of something. So don't be fooled. EQ is straightforward at first, but with its ability to both boost and cut, it requires careful thought, practice, and excellent critical listening skills while also demanding an accurate and informative studio-monitoring environment.

This chapter defines the functions of the user-adjustable controls found on most equalizers in the recording studio and then focuses on how the mix engineer uses EQ to support the art by various means of fixing, fitting, and featuring elements of the mix as desired.

3.1 TOOLS FOR SPECTRAL MODIFICATION

The equalizer is easy and intuitive to use, and when pronounced settings are applied it is easy to hear. A collection of equalizers—whether plug-ins or outboard gear—is likely to include a few variations on the user interface: parametric, semiparametric, program, shelf, graphic, as well as simple—but no less important—high-pass and low-pass filters.

3.1.1 Parametric EQ

 The most flexible type of equalizer is the parametric equalizer, so called because it provides three adjustable parameters for altering the spectral shape of any audio signal. The other types of equalizers have just one or two of these three parameters, so mastering these three parameters makes understanding all other types of equalizers much easier. When you learn how to use a parametric equalizer, you are pretty much learning how to use all types of equalizers.

FREQUENCY SELECT

Perhaps the most obvious parameter needed is the one that determines the frequency range to be altered. The center frequency of the spectral region being affected is dialed up on a knob, switch, or slider labeled *frequency*. In search of bass, you might decide that the signal needs additional low-frequency content in the area around 80 Hz. Or is it closer to 40 Hz? These decisions are made at the frequency select control.

CUT/BOOST

▶ Having aimed for the desired frequency range, you determine how much to alter its amplitude by adjusting a second parameter called *cut/boost* or *gain*. This control determines the amount of decrease or increase in amplitude at the center frequency chosen via the frequency selection parameter mentioned previously. To reduce the shrillness of a brassy horn track, select a high frequency (perhaps around 8 kHz) and *cut* it by a small amount (maybe about 3 dB). To add a lot of

bass, *boost* 9 to 12 decibels at the low frequency that sounds best, somewhere between 40 and 120 Hz perhaps. These two parameters alone—frequency select and cut/boost—give the engineer a terrific amount of spectral flexibility.

Q

! The parametric equalizer's third and final parameter is just a little trickier to understand. Consider a boost of 6 dB at 3,000 Hz. This step could be the EQ move needed to add presence to a ukulele track so that it becomes more distinct from the similar, competitive performance on the guitar. Choosing the amount of gain and the center frequency isn't enough to completely define this modification to the ukulele track's frequency content. Figure 3.1 demonstrates two possible results for the same frequency select and cut/boost settings. When a center frequency for this increase in level is selected, it affects not just that single frequency but also the neighboring frequencies as well. The degree to which it also boosts other frequencies nearby is defined by the third parameter, *Q*. The *Q* describes the frequency width of the cut or boost region. It is perhaps counterintuitive, but the industry standard is that a small value for *Q* indicates a wide spectral area being boosted. A large value for *Q* specifies a narrower, more specific area of influence. Engineers frequently say "low *Q*" and "high *Q*" to describe wide and narrow equalization settings, respectively.

3.1.2 Multiband EQ

Frequency select, cut/boost, and *Q* are the three basic, adjustable parameters needed to achieve almost any kind of alteration to the spectral content of a signal, from broad and subtle enhancements to pronounced and aggressive

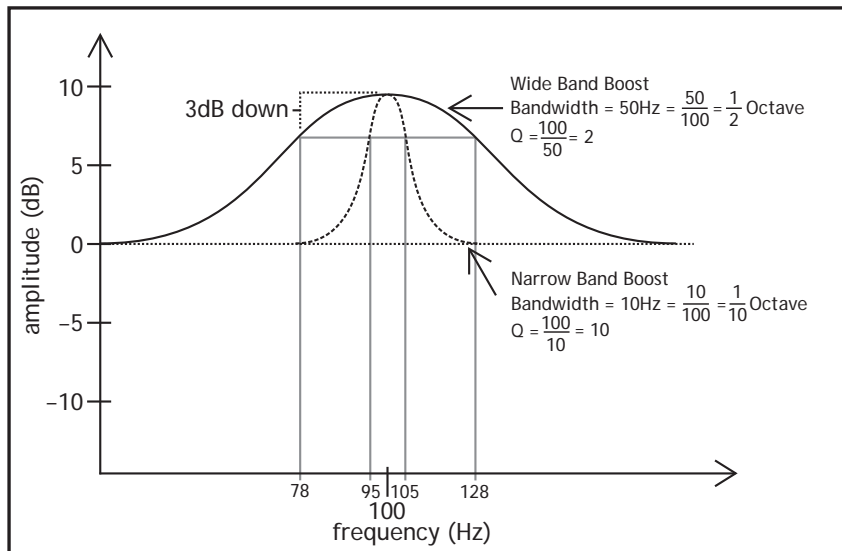


FIGURE 3.1

Any cut or boost at any frequency may be wide or narrow.

notches. This trio of controls makes up a single band of parametric equalization. A parametric equalizer may then offer several (usually three or four) of these bands, with overlapping frequency ranges. You coordinate the action of these multiple bands of EQ to achieve a single, beneficial result for the signal.

! A four-band parametric EQ has 12 controls on it: the 3 parameters \times 4 bands = 12 controls in all! It offers the three parameters four different times so that you can select four different spectral targets and shape each of them with their own amount of cut or boost, and each with whatever unique Q is needed. In the frequency ranges where any two adjacent bands overlap, the equalization changes accumulate. Boosts overlapping with boosts lead to still more of a boost at those frequencies. Cuts overlapping with cuts behave similarly. When the boost of one parametric band overlaps with the cut of an adjacent band, they work against each other, and the net effect is simply the algebraic sum of the boost minus the cut. The larger magnitude wins.

The result, if our ears can follow it all, is the ability to drive a tremendous amount of change in the spectral content of a signal. Figure 3.2 shows one possible result of four-band parametric equalization. Clearly, there is no limit to the spectral reshaping an engineer can explore. The terrific amount of sonic shaping power that four bands of a parametric equalizer offer makes it a popular piece of gear in any studio. However, other useful EQ control sets exist as well.

3.1.3 Semiparametric EQ

Some equalizers set the Q internally, providing the mix engineer access to only the frequency select and cut/boost parameters. Because of this reduction from three user-adjustable parameters to two, this type of EQ is sometimes called a

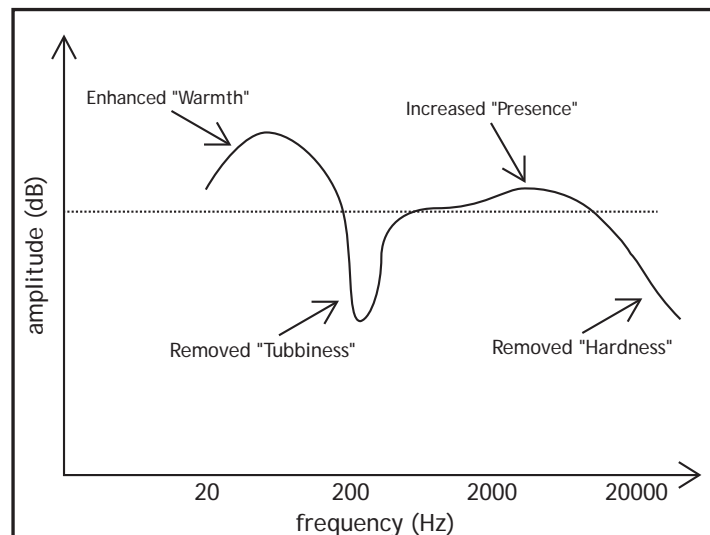


FIGURE 3.2

A four-band parametric EQ has great production potential.

semiparametric equalizer. You can't directly adjust the bandwidth—just the center frequency and the amount of cut or boost. Emphasizing the adjustability in the frequency domain, this type of equalizer is also sometimes called a “sweepable” EQ.

💡 This configuration in which each band offers only two adjustable parameters is still quite desirable. With the removal of controls that might be distracting, the engineer can work faster and thereby stay creative without as much searching and fumbling for the right parameter. The simpler semiparametric design also represents an opportunity for the manufacturer to offer an equalizer at a somewhat lower price or to sell a higher-quality design at the same price. Make no mistake: the semiparametric is not simply a compromise, rather it is still a very useful tool in the sound recording.

🔊 A semiparametric equalizer's lack of a Q control doesn't mean the concept of a broad or narrow cut or boost is gone. There is still a region of neighboring frequencies affected by the selection of frequency and gain on a semiparametric equalizer. However, the spectral width of any EQ adjustment is set by the design of the equalizer, not by an adjustable control available to us as we mix. Through careful listening, we must develop a sense of the width of each make and model of semiparametric EQ we use. This knowledge influences our thinking when we reach for EQ in a future recording session. Over time, you must collect knowledge of the approximate bandwidth of all semiparametrics available to you. You'll choose Brand X, Model A over Brand Y, Model B because you know it to have the Q best-suited to your production needs at that instant—and these various production needs are discussed in detail later in this chapter.

🔊 Much fuss is made about the sound quality of certain equalizers. Some sound better than others, it is said. To be sure, the quality of the design, components, and manufacturing processes can have a noticeable effect on the sound quality of the device. But in the case of semiparametric equalizers, a critical distinguishing characteristic is the design-determined Q. Some music production situations leave us in need of a wide cut or boost—time to reach for a low-Q equalizer. Other production challenges are solved with a sharp notch to more surgically remove a troublesome resonant frequency. In this case, a high-Q processor is needed. The better-sounding EQ is the one that is right for the task at hand. Some semiparametric EQs have low Q; others have higher Q. Know your gear, and listen carefully. An equalizer ill-suited to one task might sound great on another. Accidentally using too narrow a bandwidth when a wide one was needed will sound disappointing. But it isn't fair to blame the EQ, categorically condemning it as a bad-sounding equalizer. When a production situation requiring a narrow Q arrives, this “bad-sounding” equalizer may sound perfect. We must defend ourselves from rumors and misinformation and must not be too quick to judge the quality of audio devices until we have had the chance to study, use, and hear them across a range of actual mix sessions.

! A semiparametric equalizer's lack of a Q control doesn't mean that this parameter is fixed. Some equalizer designs deliberately allow the Q to increase (narrower bandwidth) as the cut or boost becomes more extreme. The idea—and it's a good one—is that a small gain setting, say ± 3 dB or less, reflects an

engineer's need for subtle reshaping of spectrum. A low Q accompanies such a gain setting. A larger gain setting, perhaps ± 12 dB, implies that more specific frequency ranges must be highlighted or removed. A high Q is offered. This useful design cleverly allows Q to increase as the absolute value of gain increases to extreme cut/boost settings. This feature—limited to only a few semiparametric designs—can come in handy.

In the hands of a thoughtful engineer, semiparametric equalizers offer great spectral shaping flexibility.


3.1.4 Program EQ

💡 There is room for further simplification of the user interface on an equalizer. Sometimes we are offered control only over the amount of cut/boost and can adjust neither the frequency nor the Q of the equalization shape. Generally called *program EQ*, this is the sort of equalizer found on, for instance, your parents' stereo (labeled “treble” and “bass”). This type of EQ is also found on many consoles, vintage and new. It appears most often in a two- or three-band form: three knobs labeled high, mid, and low that are fixed in frequency and Q and offer the recording engineer only the choice of how much cutting or boosting they are going to apply. In the case of mixing consoles, remember that there may be the same equalizer repeated over and over on every channel of the console. If it costs an extra 20 dollars to make the equalizer sweepable, that translates into a bump in price of more than \$600 on a 32-channel mixer. If it costs \$100 to make them fully parametric, and it's a 64-channel console... now we are talking about a price increase equivalent to that of a really good used car. The good news is that well-designed program equalization can sound absolutely gorgeous. It often offers frequencies that are close enough to the ideal spectral location to get the job done on many tracks. Sometimes the engineer doesn't even miss the frequency select parameter; program EQ does the job just fine.

3.1.5 Graphic EQ

▶ A variation on the program EQ concept is the *graphic equalizer*. Like program EQ, this device offers the engineer only the cut/boost decision, with no user-adjustable controls for Q or frequency. On a graphic EQ, the several frequency bands are presented not as knobs, but as sliders, like faders on a console. The graphic EQ presents sliders from low to high frequency, left to right. The visual result of such a design is that the fader positions provide a good visual description of the frequency response modification that is being applied—hence the name “graphic.” Handy also is the fact that the faders can be made quite compact. It is not unusual to have from 10- to 30-band graphic equalizers that fit into one or two rack spaces: compact, visually informative, and easy to use.

When the graphic equalizer lives in software, adjustments can be as simple as clicking on a frequency response plot and dragging the line into the shape that is desired.


 Graphic EQ is an extremely intuitive and comfortable way to work. Being able to see an outline of the current frequency response modification makes it easier and quicker to achieve the spectral goal at hand. Turning knobs on a four-band parametric equalizer is more of an acquired skill.

There are moments in the course of a project when one must reshape the harmonic content with great care, using a parametric EQ. In other instances, there is no time for such careful tweaking, and a graphic EQ is the perfect, efficient solution. All engineers should plan to master both.

3.1.6 Which Is Best?

When building a studio and investing over time in the hardware and software needed to create ever better recordings, one naturally wonders: what type of EQ is best? Are parametric EQs better than semiparametric? Are graphic EQs better than program EQs? Life isn't that simple. All design types have the potential to be very-high-quality equalizers—or not.

It's a bit complicated. In all types of hardware EQ, we find models on which the knobs move smoothly and continuously across their available range of settings and other models, which have knobs that “click” to discrete values. Frequency select may be continuously sweepable from, say, 125 Hz to 250 Hz, or the knob may snap directly from 125 Hz all the way over to 250 Hz, offering nothing in between. If, in the latter case, you wanted the equalization contour to be centered on exactly 180 Hz, you are out of luck.

 What seems to be a reduction in engineering control and flexibility, however, may offer an improvement in equalizer sound quality. Clicking a knob on the faceplate may physically select different electronic components inside the device. The equalizer is literally swapping components in the circuit path for different frequency selections. It isn't just adjusting some variable piece of the circuit; it is physically changing the circuit. In choosing which type of equalizer to acquire and use, engineers have to trade off sound quality versus price and processing flexibility versus ease of use. Some companies have such high standards for sound quality that they take away a little bit of user flexibility (continuous controls) to get a better sound. Conversely, if one finds an equalizer that is fully parametric and sweepable across four bands yet costs less than a large pizza, it would be wise to investigate how the manufacturer made the EQ so infinitely adjustable and how much sound quality was sacrificed in the name of this flexibility. Don't value an equalizer based on the number of controls it has. A simple program EQ with cut and boost controls might contain only extremely high-quality algorithms or components inside. The best studios acquire the gear that sounds best and keeps the engineer most comfortable.

Don't fall for the sexiest graphics or coolest user interface; what you see isn't necessarily what you hear. Don't be seduced by the power a multiband, fully parametric equalizer offers when it might only lead to confusion in the heat of a

session and might represent some sacrifice in sound quality necessary to offer all of those knobs and switches. Listen carefully and consult experienced experts.

3.1.7 Shelving EQ

The most elaborate, feature-rich equalizer envisioned so far—the one with the most knobs on the faceplate—is the multiband parametric equalizer. With four bands of parametric EQ, the device provides us with 12 controls. There is room for still more processing flexibility. Here's how.

Each parametric band offers a region for spectral emphasis when boosting or deemphasis when attenuating. This shape is called a *peak/dip contour* because of the visual change it makes in the frequency response. Roughly shaped like a bell curve, it offers a bump up or down in the frequency response. Alternative EQ contours exist.

► The *shelving equalizer* offers half of a peak/dip response on one side of the selected center frequency, and a flat cut or boost region on the other (Figure 3.3). You choose a frequency to boost or cut, and the shelving equalizer achieves that

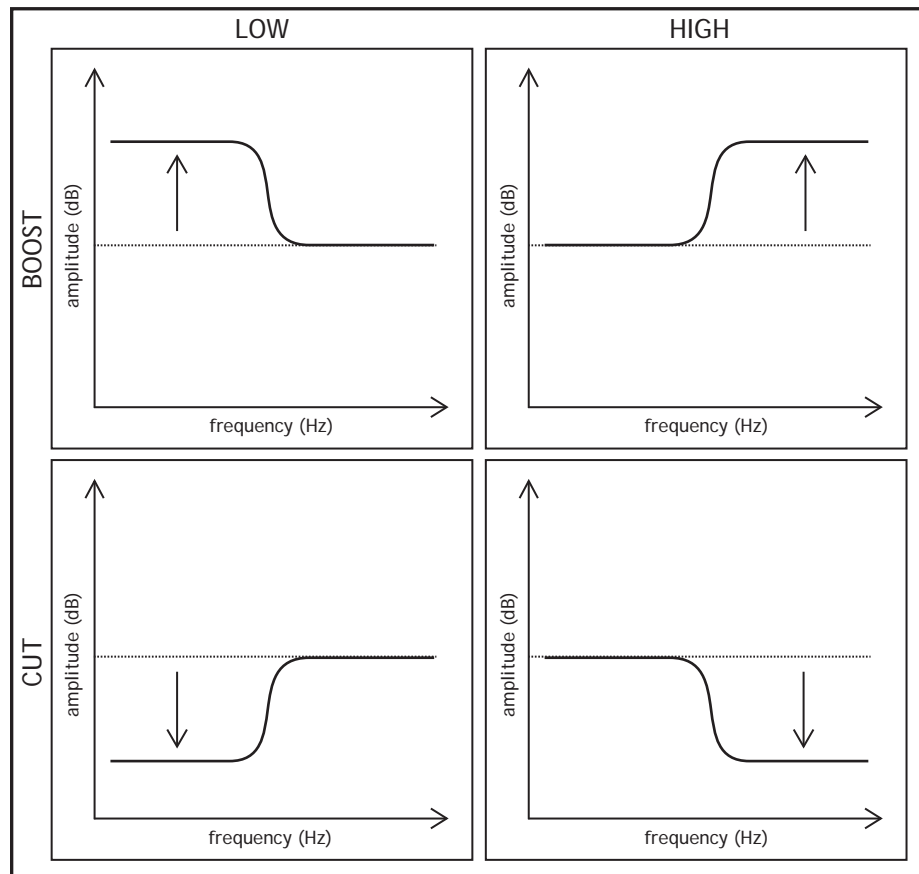


FIGURE 3.3
Shelving equalization.

and then holds the boost or cut beyond that frequency. A broad equalization desire might be to add brightness to the sound in general. A high-frequency shelving EQ bumped up 4 to 6 dB at 8 kHz will raise the output from about 8 kHz and above. It isn't limited to a center frequency and its associated bandwidth; rather, it raises 8 kHz and the frequencies above. The resulting alteration in the frequency response is flat (like a bookshelf) beyond the selected frequency.

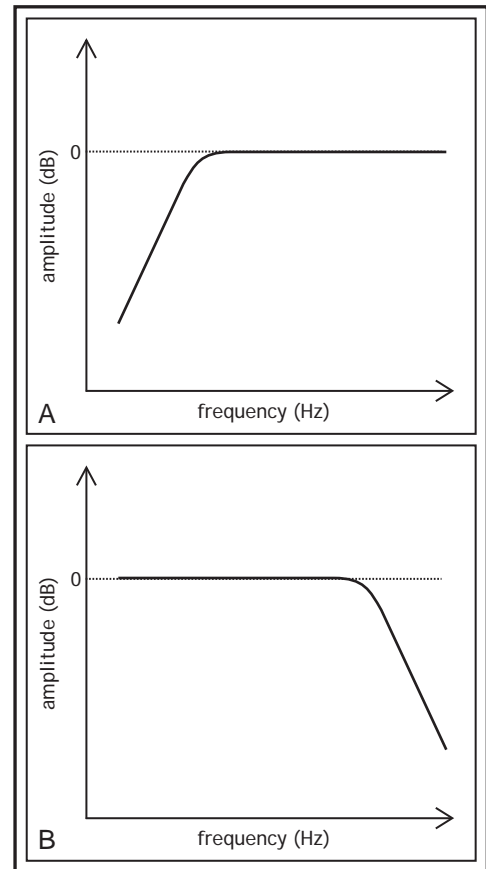
As Figure 3.3 shows, the concept of shelving EQ applies to low frequencies as well as high, and to cuts as well as boosts. In all cases, there is a flat region beyond the selected center frequency—above the selected high frequency or below the selected low frequency—that may be boosted or attenuated.

3.1.8 Filters

Beyond the peak/dip and shelf, an important third option exists for reshaping the frequency content of a signal: the *filter*. Engineers speak generally about filtering a signal whenever they change its frequency response in any way. Under this loose definition, all of the equalizers discussed so far are audio filters. But to be more precise, a true filter must have one of the two shapes shown in Figure 3.4. A high-pass filter (Figure 3.4a) allows high frequencies through with no change in amplitude but attenuates lows. A low-pass filter does the opposite. A low-pass filter (Figure 3.4b) allows low frequencies to pass through the device without a change in amplitude, but high frequencies are attenuated.

► Because the sonic result can be rather similar to a shelf EQ cutting out extreme high or low frequencies, there is the potential for some confusion between them. Filters distinguish themselves from shelf EQ in two key ways. First, filters are attenuation-only devices. They do not deliberately raise the amplitude at any frequency. In contrast, shelf EQ can cut or boost. Second—and this is more important—filters offer a never-ending amount of attenuation beyond the selected frequency. They do not flatten out like the shelf. They just keep cutting and cutting. If there is some unwanted low-frequency air conditioner rumble on a track that you never, ever want to hear, a filter can offer significant attenuation. A shelf equalizer has a finite limit to the amount of attenuation it can achieve—perhaps only 12 or 16 dB down. The weakness of using a shelving equalizer in this case is easily revealed on every quiet passage whenever that track is being played, as it might still be possible to hear the air conditioner rumbling merrily along faintly in the background.

FIGURE 3.4
(a) High-pass filter;
(b) low-pass filter.



! Filters and shelving equalizers introduce some complexity to the faceplate of an equalizer. The four-band parametric (12 controls) gets upgraded with a low-pass and a high-pass filter at each end of the spectrum, as well as a switch that toggles the lowest parametric band from a peak/dip bell curve to a low-frequency shelf and the top parametric band from a peak/dip bell curve to a high-frequency shelf. Such an equalizer contains a rich amount of capability with which the engineer can freely alter the spectral content of any signal in the studio. These knobs and switches enable you to bend and shape the frequency response of the equalizer into almost any contour imaginable.

3.2 TECHNIQUE

With knowledge of what EQ can do for us, what are we to do with EQ?

3.2.1 Non-EQ Equalization

💡 Before the details of applying EQ are discussed, it should be noted that many of the goals associated with equalization are realized through other means. If we are mixing somebody else's tracks, the spectral content of the audio provided to us is very much influenced by a range of factors beyond our control. The frequency makeup of each track was largely determined by the performer who played it and the engineer who recorded it.


When you are the tracking engineer *and* the mix engineer, things go more smoothly. You have a vision for the final mix and can track accordingly so that every incremental decision at the tracking session coaxes the tone towards that vision. Ideally, when you are at last ready to mix the project, no EQ will be needed. Realistically, we've done our job well if we can at least track things so that *less* EQ will be needed.

🔊 Less is better because the electrical circuit or signal-processing code associated with the equalizer isn't without sonic drawbacks. Phase distortion, noise, harmonic distortion, and worse sneak into the signal when we insert an equalizer. As our listening skills develop, our understanding of EQ matures, and our experience with recording and mixing deepens, we can loop back around through these decisions so that—from the beginning of the project—all relevant actions support the desired spectral outcome:

Equalizing without EQ

- Instrument selection
 - Instrument
 - Make
 - Model
 - Vintage
 - Size
 - Maintenance
 - Tone settings
 - Effects

- Picks
- Bows
- Strings
- Pickups
 - Type
 - Selection
- Sticks
- Mallets
- Heads
- Mouthpiece
- Mute
- Tuning/alternative tuning
- Composition
- Arrangement
- Musicianship
 - Chops
 - Technique
 - Intonation
 - Embouchure
 - Confidence
 - Influences
 - Strength
 - Control
 - Maturity
- Performance decisions
 - Left-hand and right-hand techniques
 - Articulation
 - Voicings
 - Extended techniques
- Room acoustics
- Microphone selection/placement
- Microphone preamplifier
- Other effects, especially
 - Distortion (see Chapter 4)
 - Compression (see Chapter 5)
 - Delay (see Chapter 7)
 - Reverb (see Chapter 9)

 Navigating through all these spectrum-driving decisions toward the goal of getting the right tone for every track in the mix—perhaps even obviating any need for EQ—seems to require time travel. How can we make the right decisions during a guitar overdub regarding its ultimate place in the mix when we have yet to record the piano and the accordion and the mix session might be months away? Simple: we guess. We make an informed guess based on our experience in past projects. We try very hard to have a strong vision for the final mix from the very first recording session. And we chase that vision diligently, refining it as necessary as we learn


more about the piece and as we welcome the surprise moments of inspiration and exploration that come from creative recording sessions along the way. To the best of our ability, we make all the decisions that influence tone in ways that lead us toward an appropriate goal, and we rely on things other than EQ to achieve it. You know you are getting good at EQ when you hardly use it. Leverage every line in the list above from the beginning of the project to make the final tone you desire at mixdown an inevitability from the first tracking session.

3.2.2 Patching and Plugging In

Equalizers are often found built into the console or DAW, available on each and every strip. Additional outboard and plug-in equalizers are typically incorporated into the audio signal flow using the *insert* of the mixer or digital audio workstation. In most applications, we use EQ to alter the signal from its current form into a better-sounding tone. We no longer want to hear the old sound. We don't want to hear both the improved spectral content and the old, non-EQed signal—just the equalized track please.


The insert is the patch that effectively forces the track through the EQ, altering its tone and giving us no chance to accidentally mix together both the equalized and the nonequalized versions of the sound. Alternatively, EQ can be patched between the multitrack return and the mixer line input, which effectively inserts the EQ into the signal flow. The use of auxiliary sends or buses is unusual for this type of effect, as we rarely need to send multiple different tracks to the same equalizer, and we don't care to return the equalized signal to the mix in parallel with the nonequalized signal (see Appendix A).

3.2.3 Boost, Search, and Set

 The most common approach for finding a useful EQ setting is to *boost*, *search*, and then *set* the equalizer. *Boost* by a clearly audible amount, maybe 9 dB to 12 dB or more. *Search* by sweeping the frequency select control until the desired spectral region is audibly highlighted. Finally, *set* the EQ to the level and bandwidth needed. Either attenuate the gain and cut the frequency if it is unwanted or back off to a less radical amount of boost and fine-tune the bandwidth if it is a positive contributor to the mix. It's that simple.

Over time, through experience and ear training, you'll find that you can still be productive in this process while boosting by less than 12 dB, eventually skipping the boost and search steps entirely and instead reaching directly for the frequency range you wish to manipulate.

3.2.4 Anticipate

 Early in your career, you should occasionally challenge yourself to first listen to the problematic tracks. Then, before touching any equipment, imagine—in your mind's ear—the frequency ranges in need of alteration. You reach for the frequency select first, not the cut/boost control, and aim immediately for


those frequencies. With the frequency set, you raise/lower the gain and tailor the bandwidth and create—on your first try—the right alteration to tone. The goal is to skip the boost and search parts.

This premeditated approach to equalization is the method of the experienced engineer. The less-experienced engineer should try this from time to time during low-pressure recording sessions. In fact, audio listening skills develop more quickly when we discipline ourselves to make an informed guess of the target frequency without boosting and searching first. Learning from mistakes made this way will help you develop a deep feeling for the qualities of different spectral regions.

However, until you've had several years in the mix position, know that there is no harm in taking advantage of those first two steps. Doing so is fast, relatively easy, and has the extra benefit that the producer and any band members who might also be in the control room will be able to follow along sonically and be supportive of your EQ decisions. Even the famous, expensive engineers resort to the boost, search, and set approach on occasion. Remember, the music-buying public doesn't care how it is done—just that it sounds great.

3.2.5 Improve

! Three things slow our progress as we become proficient with equalization. The first challenge is the sheer range of possibilities that EQ offers. Using boosts and cuts, high Q to low Q, peak/dip, shelf, or filter, the humble effect known as EQ represents a nearly limitless range of options from 20 Hz to 20,000 Hz. Such wide-open possibilities can paralyze. Don't get lost wandering among the spectral possibilities with EQ, waiting to see/hear if anything interesting happens; seek out specific motivations to equalize a signal, as we discuss in the following section (“Mix Strategies—EQ”).

 Second, learning how to hear with high resolution isn't easy. Critical listening skills are developed over a lifetime and require careful concentration, good equipment, and a good monitoring environment. It is straightforward to distinguish the difference between 1 kHz and 10 kHz and even between 1 kHz and 4 kHz. But can you easily hear the difference between a 6 dB boost at 1 kHz versus a 6 dB boost at 2 kHz? What if it's just a 2dB boost? How 'bout 1 kHz versus 1.2 kHz? No one learned the difference between 1 kHz and 1.2 kHz overnight. It's a journey of practice and increasing resolution. Take ear training seriously.

Third, interfering with this already challenging learning process is the temptation to imitate others or repeat equalization moves that worked on the last song. “Magic” settings that make every mix sound great simply don't exist. If a rookie got the chance to write down the equalizer settings used on, say, Stevie Ray Vaughan's guitar track on *Pride and Joy*, it might be tempting to apply it to some other guitar track, thinking that this unique equalizer setting goes a long way toward improving the tone. But the fact is, the tone of Stevie's guitar is a result of countless factors: the playing, the tuning, the type of strings, the

kind of guitar, the amp, the amp settings, the stomp boxes, the placement of the amp within the room, the room, the microphones used, the microphone placement, the tape machine, the tape formulation, the tape calibration, and so on. The equalizer alone doesn't create the tone. In fact, in the scheme of things, EQ played a relatively minor role in the development of Stevie Ray Vaughan's tone. Those many other factors have a much bigger influence. So repeating EQ moves that worked in the past simply will not ensure a good sound today—they are pretty much irrelevant.

💡 The way to take out the guesswork and gain fluency with this infinitely variable, difficult-to-hear thing called EQ is to develop a *process* that provides a crisp strategy for when and how to equalize a sound. We do that next.

3.3 MIX STRATEGIES: EQ

💡 When does a recording engineer boost, search, and set? What are they listening for? Why and when do they equalize? EQ is simple in concept but difficult in application. Some important words of comfort: all engineers have a lot to learn about EQ. All engineers—apprentices, hobbyists, veterans, and award winners—are still exploring the sonic variety and musical capability of equalization.

As discussed in Chapter 1, distilling the infinite possibilities available to an engineer into specific actions to take while mixing can be made much simpler by focusing efforts into the three categories of the smart mix mindset: the fix, the fit, and the feature.

💡 It is a mistake to insert an equalizer on the vocal just because the vocal “is supposed to be EQed.” The use of EQ on a vocal depends on the qualities of that vocal track that might need spectral adjustment. And there is certainly a chance that no EQ is needed at all. When the vocal could benefit from equalization, what specifically is to be done when the EQ is inserted? A deliberate strategy should motivate the use of EQ on vocals or any other track. Mix engineers don't EQ just for the sake of EQing. They EQ to solve problems and support the music. The challenging task of how best to solve any problems and somehow support the music is made easier when we cycle through three questions:

- What, if anything, needs to be *fixed* using EQ?
- Can EQ be used to help the various elements of the multitrack mix *fit* together better?
- Do any tracks need some aspect or sound quality *featured* through EQ?

3.3.1 Fix

A major motivation for engaging an equalizer is to correct things spectrally, getting rid of problems that lie within specific frequency ranges. For example, out-board equalizers, console channel inputs, microphone preamplifiers, and even microphones themselves often have low-frequency roll-off filters. Their prevalence suggests a common need for a high-pass filter. These devices remove

low-frequency energy less for creative “this’ll sound awesome” reasons and more to fix the common problems of rumble, hum, buzz, pops, wind, and proximity effect. We’ll take these in order.

RUMBLE

In many recording situations, the microphone picks up a very-low-frequency (40 Hz and *below*) rumble. This low-end energy likely comes from the building’s heating, ventilation, and air conditioning system, or the vibration of the very structure of the building due to traffic on nearby highways and train tracks. (Note to self: Don’t build a studio next door to major freeways, train stations, or helipads.)


! Rumble can be really-low-frequency stuff, so low that singers and most musical instruments are incapable of creating it. Rumble is often so deep that many loudspeakers—even expensive professional-tier models—can barely reproduce it, if at all. Because the loudspeakers won’t reproduce it, rumble can be difficult to notice from the comfort of the control room. To keep it out of the audio tracks, rumble must be carefully diagnosed in the recording room by the tracking engineer.

Because it is so difficult to hear and nearly impossible to reproduce, it is appropriate to wonder whether removing rumble is worth the trouble. If the rumble is easily overlooked, can’t the problem be ignored? There are two motivations for keeping deep rumble out of our recordings: dynamic range and production standards.

💡 The dynamic range issue is straightforward. Just because music listeners can’t easily hear rumble doesn’t mean that the audio system doesn’t use up precious dynamic range in its effort to process the signal. In fact, an inaudible low-frequency signal might drive an audio device into distortion. The listeners at home may not hear the cause, but they will hear the effect. The distortion creates upper harmonics that are easily within the more audible mid and high frequencies (see Chapter 4). The audio engineer’s careful use of the dynamic range of all of their audio devices throughout the whole recording process is corrupted by the presence of unwanted, low-frequency rumble. When we record a vocal, snare drum, guitar, or whatever signal a session presents, we want to raise the level of the signal well above the noise floor of the system but not so much that the device runs out of headroom and begins to clip. Rumble will use up some of that precious headroom, causing us to run into overload at lower signal levels. To prevent rumble from causing overload at any stage—within the microphone, the preamp, the console, any signal processors, the storage device, the storage medium, the power amplifiers, or the monitors—it should be prevented, attenuated, or removed if possible.

Regarding production standards and rumble, is it really acceptable to allow rumble into your recorded production knowing that maybe 80 or 90 percent of listeners won’t notice it because their systems and listening environments won’t reveal it? Most audio engineers have a natural desire to look out for that 10 or 20

percent of other listeners who will hear it. If you set your production standards high enough to satisfy the pickiest listeners and to far exceed the expectations of the rest, then you share the audio standards of most successful engineers. Rumble is unwelcome because it undermines those high standards.


 To prevent problematic rumble, recording engineers listen carefully to the recording space even as the band is being situated and the microphones are being selected and placed. During the sometimes-chaotic time of getting ready to record, we must assess the noise floor of the recording space. If a deep sound is heard and/or possibly felt (in the chest cavity or through the feet), try to identify the source and, if possible, to stop it. It is not uncommon in some facilities and especially in some on-location recording sessions (e.g., in symphony halls or houses of worship) to turn off the heating or air conditioning and/or plan to use one of the bass roll-off features of your recording chain (on the microphone, the microphone preamp, the console, or an equalizer). Because very little music happens at such low frequencies, it is often appropriate to insert this high-pass filter to remove most of the problematic, very-low-frequency energy. The intent is for this filter to address *mechanical* problems in the room and have little to no effect on the *musical* signal sitting higher up the frequency axis.

! Listen carefully, however. Filters can have a noticeable effect on the signal far above the cutoff frequency. Compare the signal with and without the high-pass filter engaged. Try not to be seduced by the obvious benefit of rumble removal and pay careful attention to the audio signal, even though it may be well above the filter frequency. If the signal isn't diminished in quality, the rumble removal is a success. Unwanted attenuation of nonrumble portions of the signal and phase distortions that blur the transients of the signal, removing realism and detail, are a potential side effect of these filters. Try a small set of different filters that you may have available and choose the one that attenuates the rumble while doing the least damage to the audio that remains in the frequencies above.

That's the rumble fix. A slightly different problem is hum.

HUM AND BUZZ

Hum is typically the interference from our power lines and power supplies that is born from 60-Hz AC power (or 50-Hz power, depending on your locale). The alternating current in the power provided by the utility company often leaks into our audio through damaged, poorly designed or failing power supplies and audio interconnects. It can also be induced into our audio through proximity to electromagnetic radiation of other power lines, transformers, electric motors, light dimmers, frozen margarita machines, and such. As more harmonics above 60 Hz appear—120 Hz, 180 Hz, and 240 Hz—the hum blossoms into a full-grown *buzz*. Buzz finds its way into almost every old guitar amp, helped out a fair amount by fluorescent lighting and single-coil guitar pickups.

 The best solution is to stop the hum or buzz by identifying and removing the cause. This removal can be done through steps as simple as turning off all

lights on dimmers, turning off all fluorescent lights, or asking the guitarist to move or turn slightly. Sometimes plugging the equipment into a different outlet helps. On the other hand, chasing hum and buzz problems can point to more significant, difficult solutions that can't be implemented during a recording session: rewire the patch bay, redesign the power distribution and grounding for the entire recording studio, modify the equipment, and so forth. Such solutions require deep knowledge of audio circuitry and power and grounding. These solutions can be dangerous and expensive if done incorrectly. Clearly, a specialist is needed. There is no single cure.

► When the source of the hum can't be found, and therefore the cause of the hum can't be directly stopped, a high-pass filter helps. To remove hum, engage a roll-off filter—a high-pass filter—starting at a frequency just above 60 Hz (50 Hz) or perhaps an octave above, at 120 Hz (100 Hz). Although rumble is so low that this is not a big problem, filtering out hum requires a filter that is high enough in frequency that it can audibly affect the musical quality of many types of sound. Exercise care and listen carefully when filtering out hum. Many instruments (some vocals, most saxophones, and a lot of hand percussion, to name a few examples) may not be changed much sonically by a high-quality filter. But low-frequency-based instruments (e.g., kick drum, bass guitar) aren't going to tolerate this kind of spectral alteration. The lowest note on a four-string bass has a fundamental frequency of 40 Hz, clearly below the hum frequency. To filter out the hum is to filter out some spectral portion of the bass guitar tone. The lowest note on a six-string guitar has a fundamental of about 80 Hz, frustratingly close to the fundamental hum frequencies. Fortunately, the hum might be less noticeable on a guitar, as the performance can mask a low-level hum.

When hum and buzz can't be prevented and must be removed by EQ, it is worth noting that the spectral components are rather discrete. They are born of pure-tone sine waves. A 60-Hz hum is not a complex sound. It has energy at 60 Hz (or very close to it), and that's it. Buzz built on 60-Hz hum will retain its spectrally discrete character. It will have energy at 60 Hz and multiples thereof. It contributes no noise in between these frequencies, which suggests that very-narrow-band notch filters placed at these frequencies might remove the audible artifacts of hum and buzz yet allow the music through in and around these notched frequencies.

It's never comforting to introduce deep spectral cuts in your favorite tracks. Notching out the problem is our last resort. Clearly, it would be far better to fix the hum and buzz at the source.

POPS AND WIND

🔊 Other low-frequency problems fixed by a high-pass filter are the woofer-straining pops of a breath of air hitting the microphone whenever the singer or voiceover artist articulates a plosive sound. One must pay attention to words containing a "P" or a "B." A similar problem can occur whenever your recording work takes you outdoors (gathering news, doing live sound in outdoor venues,

or collecting natural sounds in the field). Any breeze across the microphone isn't just bad for the microphone. It creates unwanted, distracting, dynamic-range-consuming, low-frequency noise. The typical defense is a pop filter and/or windscreen. These devices attempt to keep breath and breeze off of the capsule allowing the desired audio signal through. In fact, pop filters and windscreens are acoustic high-pass filters. As with the insertion of filter circuits, listen carefully to the pass band—the portion of the audio signal that is supposed to be above the filter cutoff frequency. Windscreens and pop filters may have side effects in the mid and high frequencies as they take care of the low-frequency problems caused by pops and wind. Try several different ones and listen for the best low-frequency control with the least mid- and high-frequency compromise. If these acoustic devices can't keep the wind off the microphone sufficiently, then filter the low frequencies out with a low-frequency roll-off switch, preferably at the microphone itself.

PROXIMITY

When the instrument being recorded is positioned very near a directional microphone, an increase in low-frequency content caused by proximity effect appears. Sometimes this is sonically beneficial. Radio DJs love it—it makes them sound larger than life. Sometimes the low-frequency boost is just too much.

► Place a directional microphone about 6 inches or less from an acoustic guitar, and the resulting sound will have a distracting, low-frequency thud associated with each full strum of the guitar. This thud generally sounds unpleasant and can mask or interfere with the otherwise great sound of the rest of the captured tone of the instrument. If possible, reduce or remove the unwanted proximity effect by backing the microphone away from the instrument or switching to an omnidirectional microphone (omnidirectional microphones based on pure pressure transduction do not suffer from proximity effect). Alternatively, reach for a filter. Roll off some low end with a gentle high-pass filter and tame the unwanted proximity effect.

HISS

Not all spectral headaches live in the low frequencies. The noise floor of many devices in the studio—most especially from microphones, microphone preamplifiers, and analog magnetic tape—is often random energy, or a signal containing a broad range of frequencies. Such white noise is hopefully so low in level that it is rarely noticed. If a track has a range of dynamic levels from very loud to very soft, with any periods of silence in between phrases, the low-level noise may become audible and distracting.

The spectral character of white noise points to a partial fix. Though it may contain energy across the entire audible spectrum, white noise (the noise of many simple electronic components) is perceived as having a bias toward the higher frequencies. It sounds bright, possessing more energy in the high octaves than the low octaves (versus pink noise, which contains equal energy per octave).

► A low-pass filter, or high-frequency shelf set up for gentle attenuation of the highs, might be coaxed into settings (high-frequency, gentle filter slope or low Q) that attenuate enough highs to make the noise less distracting without rendering the audio signal itself too dull. The hiss isn't removed, but it is substantially reduced.

GENERAL FIXES

The list of “fixes” motivating the use of filters and equalizers goes on. Any undesirable quality of any sound that we can attribute to a specific frequency range may be addressed through equalization. The most common problems were discussed earlier, but other challenges will appear. To fix these spectral problems with EQ, you need to learn the spectral qualities of all the signals that you encounter. It's daunting at first, but with focus and practice, you can learn the major spectral features of the instruments you record and mix most.

Contemporary pop production is strongly oriented toward some common instruments: vocals, drums, bass, guitar, and piano, and so on. We discuss the major instruments in detail later in this chapter, but it should be noted that with each new track you encounter in your multitrack mix, you will scan it (through critical listening) for isolatable frequency problems in need of correction. Each problem found may motivate a specific use of EQ to fix it. As you develop experience with these popular instruments, you will encounter some common problems that repeat themselves.

💡 Ever faced a snare drum with an annoying ring? In some tunes, the ring sounds great. In others, such a snare sound is decidedly unpleasant. If the ringing tone is unwanted, it is of course best to fix it at the drum. Before recording the music, dampen or retune the drum. All too often, however, the mixing engineer is not the same person as the tracking engineer. We can inherit problems not fixed at the earlier recording sessions. In this situation, we simply find the frequency range most responsible for the ring (boost, search, etc.) and try attenuating it (as much as 12 dB or more) at a narrow bandwidth.


What qualifies as a narrow Q? Greater than 4. Many equalizers don't offer a Q any sharper. Rare EQs will offer a Q of 10 or greater. These are powerful tools for this kind of EQ approach. A high-Q notch at the troublesome frequency that is wide enough to remove the unwanted ring but narrow enough not to diminish the rest of the instrument's tone is ideal. Often, turning down that ring reveals an exciting snare sound across the rest of the frequency range.

Ever track a guitar with old strings? Dull and lifeless, it is unlikely that this is fixable, but don't rule it out until you've tried a bit of a boost somewhere up between 6 kHz and 12 kHz.

► Sometimes a gorgeous spectral element of a sound is hidden by some other, much-less-appealing frequency component. A good example of this can be found in drums. Big-budget commercial releases often have wonderfully powerful, punchy drum sounds. Yet when the less-experienced engineer goes searching for the correct low frequency to boost on their lower-budget projects, it never

quite sounds right. The low-frequency stuff that makes a drum sound punchy often lives just a few hertz lower than some rather muddy energy. Boosting the lows invariably boosts some of the mud.

The solution is to keep the low-end power but remove the mud. Search at narrow bandwidth for the ugliest, muddiest component of the drum sound (likely somewhere between about 200 and maybe as high as 400 Hz for kick and toms) and cut it. As this problematic frequency is attenuated, listen to the low end below the mud-range. Often this approach reveals plenty of low-end power and punchiness that just wasn't audible before the well-placed cut was applied. This application highlights low-frequency elements of the sound by applying a strategic EQ cut at a frequency range just above it.

 Removal of a narrow band somewhere within this often undesirable 200- to 400-Hz “mud-range” must be done carefully. Cutting too wide an area or attenuating too deeply can rob the instrument of fullness.

This effect must be used sparingly as well. Applying it to all tracks leaves the entire production thin and powerless. This spectral region must have some energy somewhere in the mix. Save this frequency range for the vocals, guitars, strings, and/or keys but consider pulling it out of some of the drums (e.g., kick drum and tom-toms).

In the instrument-specific discussions that follow, you'll see that we identify some parts of the sound that might—but won't always—be considered a negative attribute. Squeaks, thumps, clunks, clicks, clatter, and other mechanical noises, excessive breathiness, and so on need your thoughtful attention. If they aren't adding to the sound in some way, identify their dominant general spectral locations and try to attenuate them with an ear for the potential impact that this noise has on other desirable sounds in the same spectral area.

3.3.2 Fit

A key reason to equalize tracks in multitrack production is to help fit all these different tracks together. One of the simplest ways to bring clarity to a component of a crowded mix is to get everything else out of the way spectrally. The presence of one signal in some specific range of frequencies can make it difficult to hear any other signals at that same range of frequencies. EQ is one of the key tools that a mix engineer uses to minimize these spectral conflicts.

COMPLEMENTARY CUTS AND BOOSTS

Wanting listeners to be able to hear the acoustic guitar while the synthesized string pad is sustaining, you might find a satisfyingly present midrange boost for the guitar and perform a complementary cut in the mid frequencies of the string pad. This EQ cut on the string pad keeps the sound from competing with or drowning out the acoustic guitar.

In this way, we piece together a spectral jigsaw puzzle. The trick is to find a spectral range that highlights the good qualities of the guitar without doing significant

damage to the tone of the synth patch when subjected to the necessary cut. It'll take some trial and error to get it just right, but this approach allows clever recording engineers to layer several details into a mix.


► This EQ strategy often motivates us to simply narrow the frequency range of some tracks. Consider a crowded mix, with 48 or more tracks of audio. With so much going on, it is difficult to create a mix in which the various tracks aren't competing with each other and undermining each other, spectrally. There is room for some aggressive equalization. Maybe a piano track is one contributor to a particularly full part of the musical arrangement, when guitars, strings, synths, and background vocals have all joined in. When this sort of everybody-play-at-once crescendo happens in a pop production, the piano might be treated to a rather severe EQ. A low-frequency shelf might be engaged to attenuate everything from about 200 Hz and below by a solid 12 dB or more. So radical an EQ move may strike you as surprising at first. Solo the equalized piano track and it likely has a low-quality, cheap, thin sound. Listen to the piano in the context of the mix, however, when the drums, bass, guitars, strings, synths, vocals, and so on are all playing along, and the piano sounds, well, like a piano. With so much else happening, the listener's experience seems to fill in or assume the low-frequency energy is there.

So fitting tracks together with complementary boosts and cuts can happen at even a crude level. It's not always a complicated jigsaw puzzle of spectral shapes. Sometimes you simply lop off a broad portion of the lows or the highs and let the remaining parts of the production fill those spectral spaces. Shelf EQ and filters are great for getting this job done.

Expect to apply this thinking in a few critical areas of the mix. Around the bass guitar, engineers almost always encounter low-frequency competition that needs addressing. Anyone who plays guitar or piano and does solo gigs as well as band sessions has likely discovered this already. Performing alone, the musician has low-frequency responsibilities as he or she covers the bass line and pins down the harmony. In the band setting, on the other hand, musicians are free to pursue other chord voicings, such as leaving the root of the chord out entirely, or inverting the chord so that the root is an octave or two higher—well out of the way of what the bass player is doing. The guitarist and the pianist don't want to compete with the bass player musically, and the same is true spectrally, which means that the engineer might be able to pull out a fair amount of low end from an acoustic guitar sound to make room for the bass track. Alone, the acoustic guitar might sound too thin, but with the bass guitar playing, all is well. There is spectral room for the low frequencies of the bass because the acoustic guitar no longer competes there. But the acoustic guitar still has the illusion of being a full and rich sound because the bass guitar is playing along, providing uncluttered, full bass for the song and for the mix.

🔊 In the highs, competition appears among the obvious high-frequency culprits like the cymbals, hand percussion, and steel-string acoustic guitar, as well as the not-so-obvious: distorted electric guitars. It is always tempting in rock music to

add distortion to guitars, vocals, and anything that moves. Spectrally speaking, this kind of distortion (see Chapter 4) leads to the addition of some upper harmonic energy. This harmonic distortion will overlap with the cymbals and any other distorted tracks. You get a spectral pileup. Make them fit by using the same complementary EQ moves. Maybe the cymbals are informally allocated the highs above 10 kHz, the lead guitar has emphasized distortion around 8 kHz, and the rhythm guitar hangs out at 6 kHz. Mirror-image cuts on the other tracks will help ensure that all these high-frequency instruments are clearly audible in the mix.

 Fitting together the spectral pieces this way requires us to focus on what we *can't* hear. When we delete—or at least diminish—the parts that are perceptually less important, our mix gets a lot easier. With clutter and conflict removed, the other tracks become easier to hear. We better reveal additional sophistication and detail in the mix.

We don't always seek out spectrally distinct tracks. We might allow spectral overlap in an effort to create a *metainstrument*. These are built from the deliberately vague blending of complementary, similar-sounding tracks. The most common example is the way we often connect the kick drum attack to the electric bass sustain. Some drummers and bass players are so tight, musically, that we mix them into a single instrument. The performers know each other so well, and they listen to each other so intently, that almost every kick played by the drummer's foot is accompanied by a perfectly timed note from the bass player's fingers. We mix them as one, for a brief full sustain in the low end punctuated by punchiness in articulation—a concoction possible only through the coordinated combination of bass guitar plus bass drum. The casual listener may not notice—and arguably doesn't care—that it comes from two different instruments. They just like the sound, and feel the beat.

Use a similar philosophy for merging a gated room sound with a close microphone snare hit (see Chapter 6 on expansion and gating). As mix engineers, we know that it takes many tracks and a lot of finesse to create the overall metasnare sound, but the end user just hears it—and enjoys it—as a thrilling, moving source of tempo and energy.


Take it even further and combine any instruments into a logical new sound, perhaps an electric bass plus a very low, left-hand piano part plus a bit of lower brass. If the performances are consistent enough to act as one, you can deliberately unite them into a new texture—a single new sound fusion. It might sound like electric bass, but with a twist, an added inexplicable. Or we might take it to an extreme, and the bass sound might now have a tone like no other. It's clearly not an electric bass guitar, nor is it a piano, and it's certainly not a tuba. The bass/piano/tuba metacombination has the aural presence of some sort of synthesized bass sound that is perfect for the tune, even though the casual listener may have no idea what instrument was played.

As mix engineer, we must consciously control this. Some tracks are sculpted, panned, and processed so that they can make their uniquely identifiable contribution


to the overall mix. Other tracks are creatively blurred and obscured, tucked into and around other tracks for a more subliminal role.

We seek to please the end user through any and all means appropriate. Delete what they can't hear, convert obscurities into opportunities, and recognize that although we can audition the individual multitrack pieces, the end user hears only the final, artistic whole.

SEIZING THE SPECTRAL OPPORTUNITY


 Boosting one track while applying complementary cuts in a spectrally competing track is a proven EQ strategy, and it reveals the logic of another, preferred strategy. The better approach is to listen for and identify the existence of half of this process. That is, once your critical listening skills have gained sufficient confidence, you listen for a spectral opening (e.g., a preexisting lack of energy in the string pad at 4,500 Hz) and seize the opportunity with the mix-enhancing boost at that frequency on the acoustic guitar.


Not every boost on one track requires a symmetric cut on another track. Good arrangers and orchestrators know how to piece all the instruments together for a natural fit. The producer and recording engineer must learn this too, using equalization only as needed.

 The mid frequencies are definitely the most difficult spectral region to sort out. It is very competitive space spectrally as almost all instruments have something to say in the mids. When listening to full-range music, it can be the most difficult place to hear in detail. Less-experienced engineers tend to gravitate toward the more obvious low- and high-frequency areas when they reach for the equalizer. On the road to earning golden ears, plan to focus on the middle frequencies as a key challenge and learn to hear the subtle differences that live between 500 Hz and 6,000 Hz. It's much easier to fit all the pieces of the spectral puzzle together when you can exploit the full potential of the lows, the mids, and highs.

3.3.3 Feature

A natural application of equalization is to enhance a particular part of a sound in order to bring out the positive spectral components of the sound. What are the positive spectral components of a sound? That is a terrifying question—at first.


 To be a good mix engineer and use EQ to highlight the necessary features of each and every track so that the resulting mix serves the music best, you must have a point of view regarding what sounds good and what sounds bad. It is subjective, to be sure, but it is an opinion informed by deep experience with music, frequent encounters with moments of exquisite sonic beauty, and necessary, accidental exposure to mistakes of gloriously extreme magnitude.

 Mix engineers take the time to study their art so that they can navigate the decisions of what “sounds good” and what “sounds bad,” and be so good at it that artists and producers hire them to express their opinion and mix

their work. A mix engineer needs to remain an avid music fan, no matter how many times the clinical, technical side of engineering rears its head. A mix engineer must seek out and listen to a vast range of recordings, making time for critical analysis of why—to them—one piece of art is better than another. The successful mix engineer grows to rely on a community of equally passionate colleagues with whom they share music, ideas, and opinions. All great engineers must have periods of eating, drinking, sleeping, and living all things audio. It mustn't be allowed to take over your whole life—in fact, family and friends will remind you to turn off your intense audio side every now and then. But if you are ever going to get good at this—really good at this—some controlled obsession is going to help. And it is in those moments of controlled obsession that you will take on the challenge of adding to your ever-expanding, ever-deepening understanding of the spectral components of the most important tracks in the mix.

The detailed instrument-by-instrument discussion and data that follows should be digested slowly. If necessary, come back to this section later, when a session challenge requires you to tackle some specific detail of a particular instrument. Know that this information is typically absorbed over years of practice, not in a single sitting.

No two guitars, no two drums, and no two singers are ever exactly alike. Issues of design, manufacture, tuning, performance, and so on all conspire to make precise spectral knowledge of all types of instruments in all styles of music impossible. Yet some spectral trends and traits are fairly universal, plus or minus a few hundred Hz and plus or minus a few dB.

 More important than the instrument-by-instrument discussion and data is the *process*: the idea of building a mental template of key spectral features of the instruments. They don't have to be exact. You always listen carefully to each track as its own unique entity. But the mental template remains a good starting point.

The following information is introduced as a way of thinking about the tracks you wish to EQ, so that you can get your head around the challenge of deciding what attributes need to be featured in your mix.

Start with the instruments you play or the instruments played by the clients you work with most. Work your way out from those instruments to similar ones—from guitar to bass, from piano to other keyboards, from female vocals to male vocals, and so on. Feel no obligation to commit every detail to memory. The goal is to build an easy-to-remember oversimplification of the major spectral features of every instrument you encounter:

- The fundamental frequencies of the playable range of the instrument
- A general sense of the harmonic content that accompanies these fundamentals
- Any unique spectral landmarks associated with the instrument
- The spectral qualities of any mechanical noises or artifacts


Enhance a bit of the consonant range with a modest, typically less than 6-dB boost, with low Q, in the frequency range that sounds best for that singer, likely somewhere in the 2-kHz to 6-kHz range.

! For added intimacy and human fragility, perhaps bring out some air and breathiness through a lift at the very high frequencies, a 3- to 6-dB boost using a very high-quality shelving EQ at 10 kHz, maybe 12 kHz if the EQ sounds good that high. A side effect of this emphasis of air is the creation of overly sizzling sibilant “S” sounds. In between the “Ss,” the voice sounds brilliant with this EQ move. But the high-frequency boost makes each “S” into an aural assault of too much high-frequency energy. The solution is to momentarily attenuate these overly loud “S” sounds by using a De-Esser, a compressor with a bit of side-chain processing (see Chapter 5), leaving us free to emphasize some of the human expressiveness of the singer taking a big breath right before a screaming chorus.

SNARE DRUM

One mental template for the snare drum is to think of it as a sudden, short burst of broadband noise. Snare can be difficult to EQ in isolation, as it reacts to almost any spectral change. The context of the other tracks will be essential when choosing spectral regions for emphasis/deemphasis.

A starting point might be to divide the snare sound into two parts (Figure 3.6). One is the lower-frequency energy coming from the drum itself—the sound the snare would have if it were just a tom, without the snares stretched across the bottom, resonant head. Second is the mid- to high-frequency energy up to 10 kHz and beyond due to those rattling snares. Narrow the spectral possibilities by looking for power in the drum-based lows and exciting, raucous emotion in the upper mid frequencies of the noisy snares. Vocals, guitars, pianos, strings, and reeds are going to want their own various upper-midrange areas to themselves. The snare, with much to offer in the mids, welcomes a complementary set of gentle cuts to make room for these instruments, balanced by a gentle boost in any midrange area that remains so that it can express itself.

 A common area of focus for more aggressive cutting is in the mid frequencies, usually narrow and in the 1-kHz to 2-kHz range. This narrow midrange area often has a sharp sort of ringing “ping” sound which, when attenuated (high Q, greater than 4, with a 6- to 12-dB cut), reveals a more open, easier to enjoy mid-range character above. Yes, it is counterintuitive, but a cut here highlights desirable spectral features in the frequency range just above.

KICK/BASS DRUM

Consider reducing this instrument to two components (Figure 3.7). There is the upper-mid-frequency click of the beater hitting the drum followed by the low-frequency pulse of the large, ringing drum. The attack lives broadly up in the 3 kHz range. The tone is down around 80 Hz and below. These are two good targets for tailoring a kick sound.

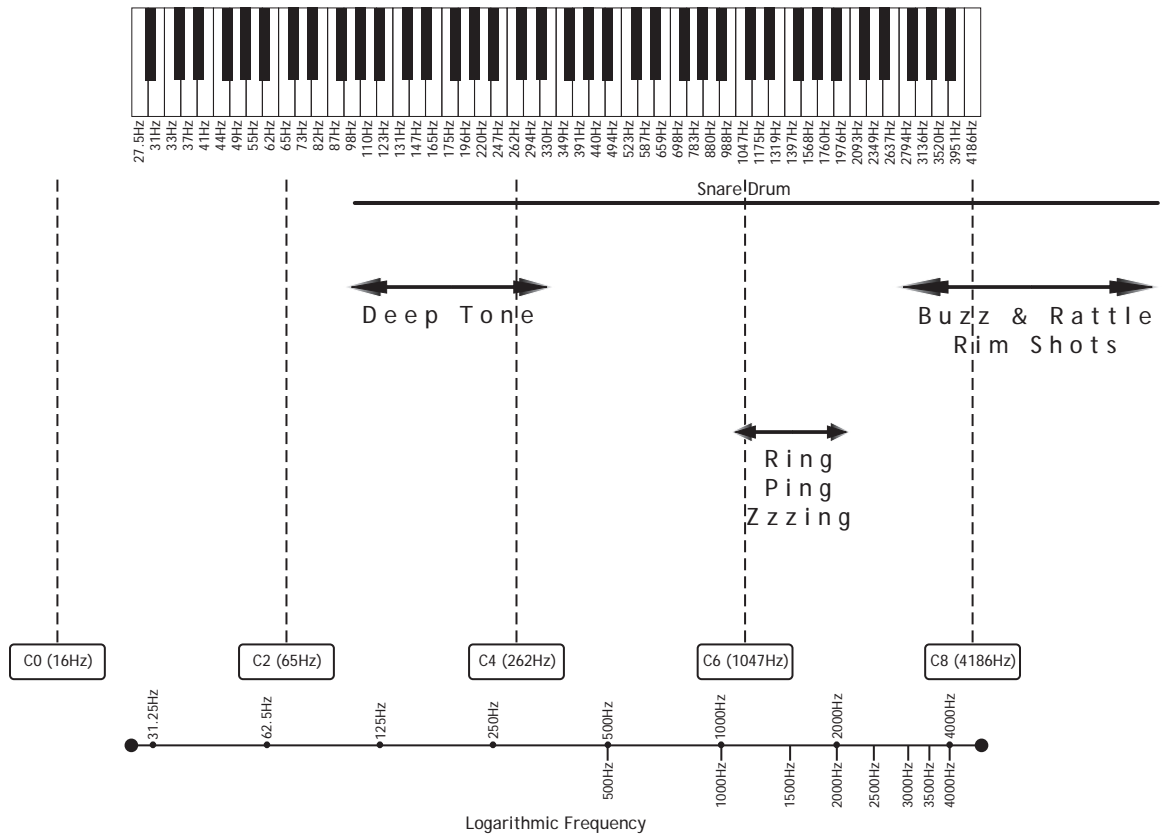
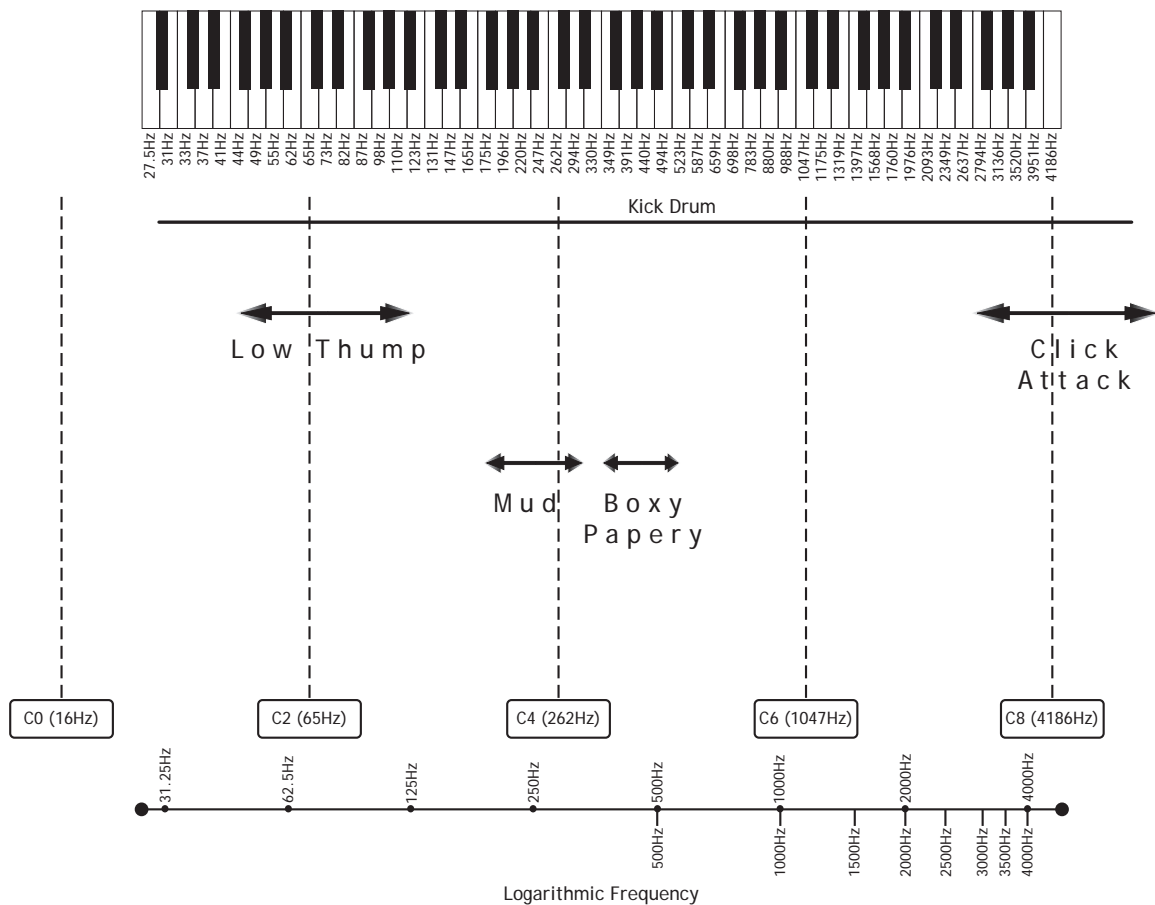


FIGURE 3.6
Frequency range of spectral landmarks (snare drum).

► If the kick lacks low-frequency excitement, remember this before you dial in a simple low-frequency boost (discussed earlier in this chapter, see “Fix” above.): kick is a common candidate for a narrow cut in that 200- to 400-Hz mud-range (high Q, greater than 4, with a cut of 6 to 12 dB, sometimes more!). Find it by boosting 6 to 9 dB with a narrow Q, and sweep around slowly from just below 200 Hz to just above 400 Hz. Find the ugliest, muddiest frequency range and then suck it out with an aggressive sweep from boost to cut. Listen carefully when you remove the muddiness. This EQ approach is productive if lower-frequency (100 Hz and lower) energy seems to rise into audibility, leading to a punchier and more powerful kick.

Depending on the tuning of the instrument, there is often a signature papery/boxy sort of sound associated with the front resonant head. Often somewhere in the 300- to 600-Hz range, some kick tracks will benefit from a gentle, surgical cut here (2 to 6 dB of attenuation with a medium to narrow Q at the most problematic frequency).

**FIGURE 3.7**

Frequency range of spectral landmarks (kick drum).

ELECTRIC GUITAR

The fundamental frequencies in the playable range of the typical, in-tune electric guitar covers the range from about 80 Hz to about 1200 Hz (Figure 3.8). Of course, the harmonics of every note played represent energy at multiples of these frequencies. Any amp- or stomp-box-induced harmonic distortion adds yet more energy to the sound, from 80 Hz well up to 6 kHz and beyond.

! The equalizer doesn't empower the engineer to separate fundamentals from harmonics, only to access a specific frequency range. So the engineer should note that an EQ setting above 1200 Hz affects only the harmonics of the instrument. EQ processes below 1200 Hz are a bit messier to keep up with, adjusting the harmonic balance for some notes and changing the level of the fundamental for others. Shaping timbre above 1200 Hz is more straightforward than dipping into that 80-Hz to 1200-Hz range, where the engineer's cuts and/or boosts will affect the very phrasing and note-to-note level of the guitarist's performance. EQ changes in the 80–1200-Hz range aren't forbidden. In fact, such settings are

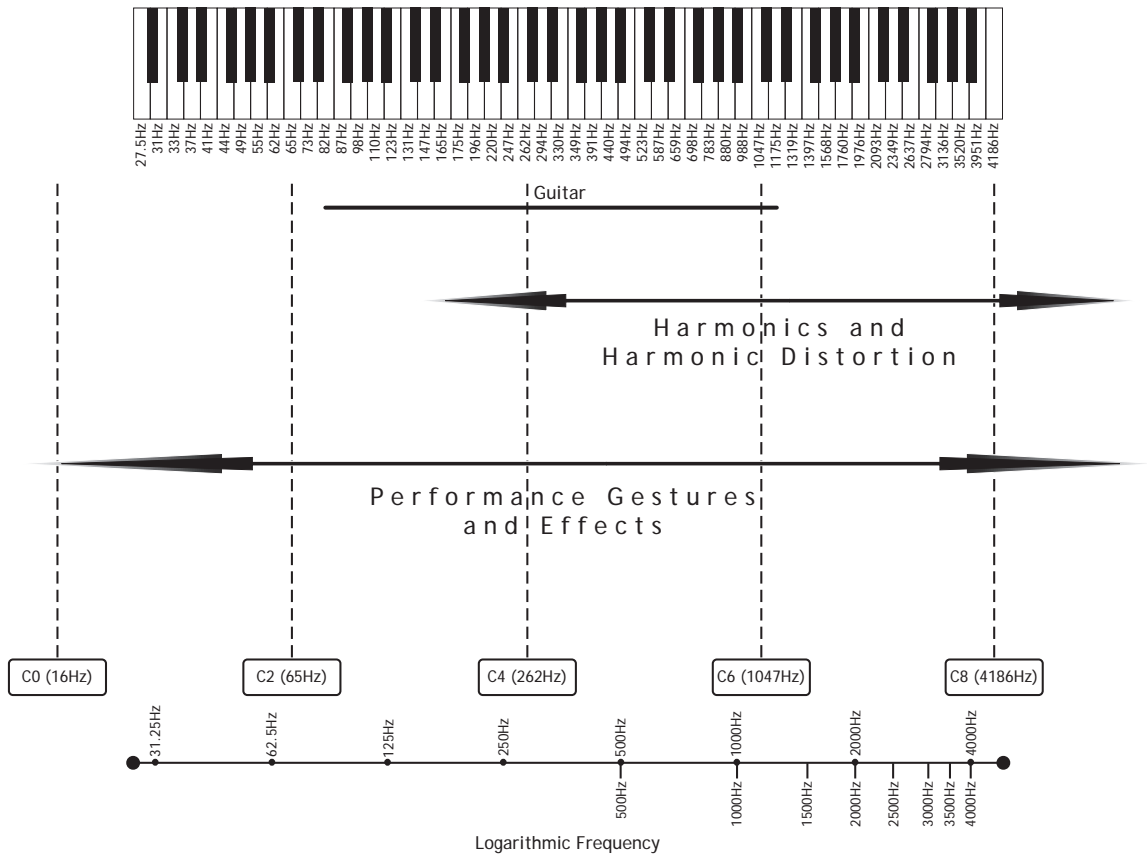



FIGURE 3.8
Frequency range of fundamentals and spectral landmarks (guitars).

quite common. But expect to use more restrained gain settings (less than +/-6 dB) and lower Q settings (less than 2), and pay particular attention to single-note lines in the performance, if there are any.

 Although the lowest sustained pitch for the instrument is that 80 Hz fundamental, this doesn't mean that the instrument has no output below 80 Hz. As with all instruments, there can be audible broad band bursts of energy associated with the onset of every note or chord, creating instantaneous bursts of energy that include energy below the lowest fundamental frequency. Similarly, the ends of notes and chords can have broadband energy associated with them, very much influenced by the performance gestures invoked. This spectral interest at the beginning and end of notes is very important to the electric guitar sound, so pay particular attention below 80 Hz. Playing with a pick, playing with fingertips, playing with fingernails, hammering on and pulling off, palm mutes, and so on. The spectral subtleties of these techniques are most revealed in the upper midrange (2 to 6 kHz), and in the subharmonic range (below 80 Hz).


ACOUSTIC GUITAR

Try separating the acoustic guitar into its musical tone and its mechanical sounds. Listen carefully to the tone as you seek frequencies to highlight. Frustratingly, this covers quite a range from lows (fundamental frequencies start at about 80 Hz) to highs (harmonics for a steel-string acoustic played with a pick reach 10 kHz and above).

In parallel, consider its more peculiar noises that may need emphasis or suppression, depending on your production wishes: finger squeaks, fret buzz, pick noise, and the resonant sound of the wooden box of the instrument itself that thumps along with every aggressive strum. Though their exact frequency location varies from instrument to instrument and performer to performer, look for these frequency landmarks in every acoustic guitar you record and mix. EQ is a powerful way to gain control of the various elements of this challenging instrument. Manipulate the relative loudness of these elements as part of shaping the tone and character of the acoustic guitar.

ELECTRIC BASS GUITAR

The four-string bass guitar reaches an octave lower than a regular guitar, leading to a fundamental near 40 Hz (see Figure 3.8). The highest fundamental is close to 350 Hz. As with regular guitar, mentally distinguish between and among the fundamentals, the harmonics of the instrument, and the harmonics of any distortion device.

 As the bass rarely plays chords and usually plays single-note lines, particular attention must be paid to any EQ moves in that range of overlapping fundamentals and harmonics (40 to 350 Hz). The bass player has likely worked hard to make the level of each note suit the performance intent of the overall tune. The mix engineer's EQ settings shouldn't undermine the music by crudely altering the relative level of the notes.

Landmarks to be on the lookout for include fret buzz (when the string is allowed, usually mistakenly, to vibrate partially against a fret), pick noise, finger squeaks, and the sound of fingers and/or a pick striking the pickups.

At the risk of oversimplifying: the primary role of the bass in most pop music arrangements is harmonic—it lays out the bottom of the arrangement, orienting listeners to the very chord being played and giving essential meaning to the rest of the notes being played and sung by others. The bass plays a significant second role, often underestimated by engineers new to their craft: time. Not only is the pitch of the note played important to the tune, but the instant at which the note is played is critical to the feel and the groove of the tune. Is it just a little late and lazy, lending a laid-back and relaxed quality to the tune? Is it a fraction of an instant early, adding urgency? In either case, the bass player is in league with the drummer to make these things happen. Listen carefully to the interaction of the kick drum and the bass guitar. They usually act as a single instrument, each kick accompanied simultaneously by a bass note. With bass, it's about pitch and time.

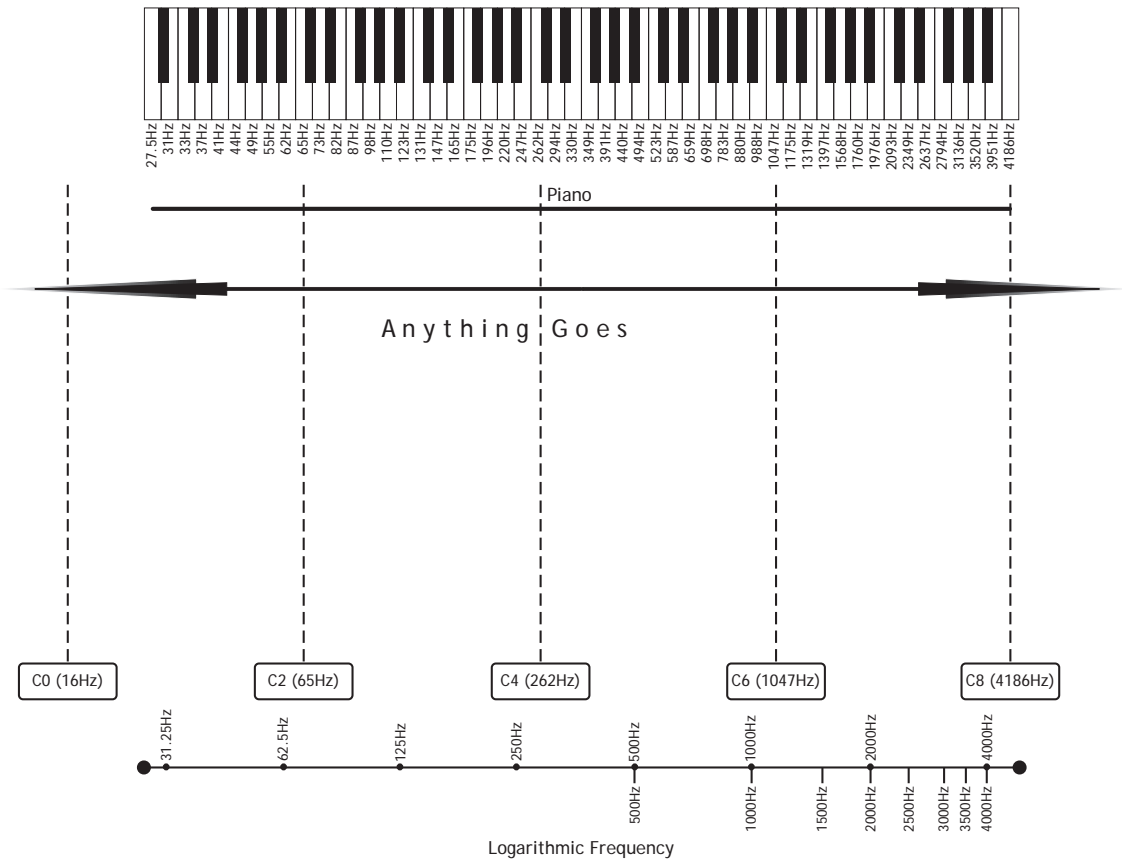


FIGURE 3.9
Frequency range of
fundamentals (piano).


PIANO

Ah, piano. The mother instrument. Its 88 keys span fundamental frequencies from just under 30 Hz to just over 4 kHz (see Figure 3.9). It reaches about as low and as high as any instrument we are likely to encounter (the pipe organ is a rare exception).

As if that vastness weren't enough, the piano is also a percussion instrument. That is, the strings are excited into action by the pounding of hammers (the frequent use of tuxedos and fancy gowns seems to make it look less crude in concert). So, like a drum, the notes begin with a broadband, pitchless burst of energy that spans the audible range. The notes sustain—well, the notes sustain with that characteristic rise and fall of any and all harmonics in a complex, ever-changing, unique piano way.

So the sound begins with a bang, and sustains with an ever-evolving harmonic recipe. For many of us, we half-panic when we think of applying equalization to a piano track. EQing a piano is like hugging Jell-O. The harmonic content of the sound keeps moving. Just when we think we've found a spectral region

worthy of highlighting or in need of a bit of attenuation, the piano itself allows that harmonic region to swell or recede or both. Prepare to be challenged by this instrument—hopefully, pleasantly so.

 For those instruments that a mix engineer plays and records often, it is wise to spend some time examining their sounds with an equalizer. Insert an equalizer, dial in a narrow ($Q = 3$ or 4) 9-dB boost, and sweep the frequency parameter slowly throughout the audible range. Look for and make note of any defining characteristics of instruments and their frequency ranges. Also look for the less-desirable noises that some instruments make and file those away on a “fix” list. These mental summaries of the spectral qualities of some key instruments will save you time in the heat of a session when the client wants more punch in the snare (aim low) and more breathiness in the vocal (aim high). The ability to successfully feature any aspect of any instrument depends directly on both your general knowledge of the overarching attributes of the entire audible spectrum from 20 Hz to 20,000 Hz as well as an intimate knowledge of all instruments in the multitrack arrangement.

3.4 SUMMARY

Equalizers enable the mix engineer to alter the relative levels of the spectral content within any signal. The parameters of frequency select, cut/boost, and Q are variously available on parametric, semiparametric, program, and graphic EQs. Peak/dip, shelving, high-pass, and low-pass filters provide the necessary alteration to frequency content so that the engineer can fix, fit, or feature audio signals as needed to help the mix best realize the artistic intent of the musicians who created it.