Craig Anderton

Sonar 3—Mixing & Mastering

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Welcome ...

Cakewalk, the company formerly known as Twelve Tone Systems that started business in 1987, launched their first MIDI sequencer back in the days of MS-DOS. At that time, the Macintosh ruled for music and was joined to a lesser degree by the Atari ST series. PCs, particularly after IBM's disastrous launch of the PC jr, were considered machines suitable only for spreadsheet jockeys, corporate bean counters, and the terminally uncreative.

Eventually Cakewalk added hard disk recording to their sequencer, and thus the Cakewalk Pro Audio line was born. While it acquired tens, and eventually hundreds, of thousands of devotees, it still was looked upon by many as being "not quite as professional" as what was on the Mac—never mind that it was highly cost-effective, had no onerous copy protection, enjoyed fine tech support, and even had niceties like a reliable video window to expedite audio-for-video projects.

Then in 2001 Cakewalk ended the Pro Audio line after version 9 and introduced Sonar, a quantum leap for the company. It featured an efficient, "flat" interface that smoothed workflow; it also took advantage of Windows' low-latency WDM sound drivers—a vast improvement over previous drivers—and had the ability to run DXi (DirectX instruments) software synths. But it was a gamble: Few sound cards supported WDM and few manufacturers made DXi software. Still, the legions of Cakewalk fans were all too happy to move up to the next step, and pro users, who had started to get impatient with Apple's vacillations regarding music on the Macintosh, saw something not only viable, but in some ways ahead of the pack for the Windows platform.

But two other developments played into Sonar's hands: The performance-to-price ratio for Windows machines was skyrocketing, and Microsoft finally delivered Windows XP, a rock-solid consumer-oriented operating system based on NT code. Even though Apple eventually triumphed with OS X and a recommitment to audio, Windows had become firmly estab-

lished as a pro audio platform not just because of Sonar, but due to other innovative products like Sonic Foundry's Vegas and Acid, Steinberg's Wavelab, and Sekd's Samplitude.

Cakewalk, capitalizing on their momentum, continues to develop and refine Sonar. An overhaul of the graphic look, a redesigned Console view, extremely flexibly bus structure, and compatibility with Steinberg's low-latency ASIO protocol as well as WDM have made Sonar a more mainstream application. It's also broken away from the hold of Microsoft's DirectX protocol for plug-ins and soft synths by including an adapter that allows running VST-spec devices (another cross-platform protocol developed by Steinberg).

All these factors have combined to make Sonar one of the most popular, if not the most popular, pro audio application for Windows. Yet it retains the sleek workflow that endeared it to fans in the first place, as well as the foundation that users have come to love. As a result, this book will remain current for some time, even as newer versions appear.

And speaking of the book, please note this is not a Sonar "power user" book, although certainly, there's a lot of power user-type material in here. Part 1 covers the essentials of Sonar's operation, but it is most definitely not a rewrite of the manual; if the manual doesn't cut it for you, then seek out a "getting started with Sonar" type of book. Part 2 moves more heavily into the art of mixing and mastering music, and concentrates on the artistic implications of the technology available in Sonar. Think of Sonar as the "platform" for the book; many of these concepts translate just as easily to other programs.

In any event, the whole point of software is not to provide features, but to let you make music—and that's what we're going to cover here. And I hope you indeed make some great music, if for no other reason than so I can listen to it!

Craig Anderton

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Introduction to Mixing

You've written your song, recorded your parts, and did a few fixes with overdubs. You like what you're hearing from the monitors, and now you're ready for the next step: mixing and ultimately, mastering the fruits of your creativity.

For those who are impatient and don't want to take the time to read the rest of this book, here's the complete story on mixing:

"Adjust the levels, tonal balance, stereo or surround placement, and add effects as needed so that the music sounds really, really great."

Okay, I guess we can all go home now ... but wait! It's not quite that simple. That description is like saying "To play the piano, hit a combination of white and black keys with your fingers until you come up with a combination of notes that sounds wonderful." The hard part, of course, is knowing which notes to hit, when to hit them, and to have the physical and mental ability to do both without errors.

Mixing is similar: you have to make a huge number of value judgements. Which instrument should be most prominent at any given time? Do you want to mute some sections that seem redundant? Do you want a raw, in-your-face sound, or a smooth, well-produced sound? Do you want a massive guitar sound, or something that shares its space with other instruments? Who is your target audience?

How successfully you answer these sorts of questions determines the success of your mix. Mixing is a combination of art—you have to be able to judge what sounds good—and science, where you need to know what technologies and processes will produce the sounds you want. As a result, this portion of the book will include aspects of both art and science.

8.1 The Mindset: Producer, Engineer, Musician

In professional situations, the musician is part of a team of (hopefully) experienced and musically intelligent people. Two of the people who play an important role on this team are the producer and the engineer. In a home or project studio environment, the musician doesn't necessarily have access to these high-powered talents and has to perform those roles from within. Although this may seem difficult at first, this experience is probably one of the greatest teachers you can have in learning how to be objective about your playing, your style, and your sounds.

During the mixing process, it helps to be aware of the ideal role of each of the three participants (musician, producer, engineer) so that you can assume those roles at will.

- The *producer* oversees the process, rides herd on the arrangement, gauges the overall emotional impact, and makes artistic judgements about what does and does not work. To fulfill the function of a producer, you need to see each piece as part of a whole, and each track as part of a final composition. If you know where you are going, it's a lot easier to get there; the job of the producer is to figure out where you are going.
- The *musician* participates in the mix on any one of several levels, from simply observing the producer to making sure the production remains true to the original intent of the music.
- The *engineer* is the one at the session who doesn't drink, smoke, talk much, or complain, and is responsible for translating the producer's needs into a technological solution. If the producer says the vocals need more "presence," it's up to the engineer to decide which tweaks

will result in that particular effect. Of course, this is a stereotype and no stereotype is accurate, but every engineer I ever worked with respected the job and took it seriously. It can be helpful to adopt an engineer's attitude when mixing; forget about whether you could have done a better solo, and simply work with what you have.

By becoming familiar with these roles, you can apply their differing outlooks to your music and obtain a more balanced perspective. Above all, don't just mix the music; *produce* it. Turn the collection of tracks into a cohesive statement.

A common mistake among beginning producers is to overproduce. Sometimes tracks are best left unprocessed, and sometimes parts should be removed to create space for other parts. Don't fall in love with the elements that make up a particular piece of music; keep your focus on the final result. Sure, that may have been a great guitar lick—but does it support the song, or just shows that the guitarist can play lots of notes in a very short period of time?

8.2 Left-brain vs. Right-brain Activity

The human brain is a dual processing system. The left hemisphere is involved in more analytical tasks, such as math, decoding directions, reading, and so on. The right hemisphere is more involved with creative tasks and emotional responses; it's the part that "feels" rather than "thinks." This is not some weird new age philosophy; it's possible to hook up electrodes to people's heads and see which hemisphere of the brain is working during a particular task.

So what does this have to do with mixing? *Everything*—here's why.

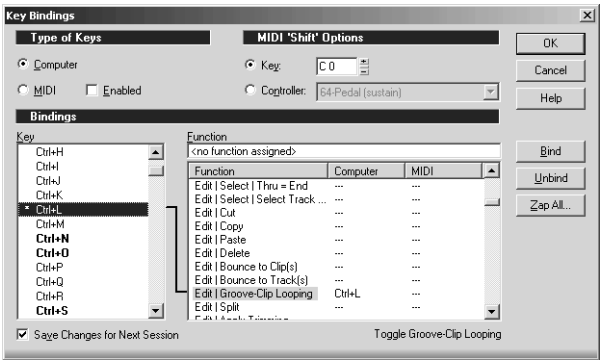
In general, it is difficult for people to switch back and forth between the two hemispheres. Every musician knows what I'm talking about: suppose you're in a right-brain groove, generating an idea a minute, when all of a sudden there's a technical glitch. Now you have to switch over to left-brain

mode and begin the troubleshooting process. When you start playing music again, the groove is gone, because your brain became stuck in left-brain mode.

In a conventional recording studio situation, the engineer lives in left-brain mode, the artist stays in the right brain (e. g. doesn't have to worry about level-setting and such because the engineer takes care of that), while the producer has the difficult job of trying to integrate the two. If you're trying to perform all these functions at once by yourself, you'll find it's not all that easy. This is why it's always great if you can have associates to help during the mixing process.

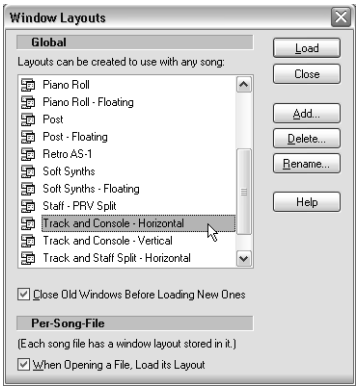
However, if you're flying solo, there are still ways to reconcile the right brain/left brain dichotomy. The most important goal is to make sure you don't have to think about left-brain activities, so you can stay in right-brain mode. If working with Sonar becomes second nature, it will be that much easier to stay in right-brain mode. Here are some tips on how to do this:

- Learn the keyboard equivalents for various operations. Once memorized, it takes less effort to just hit a couple of keys than to locate a specific area on the screen, move your mouse to it, go down a menu, select an item, etc. Sonar makes it easy to create your own set of *key bindings*, which tie particular functions to particular keystrokes on your computer keyboard. It's also possible to trigger functions from a keyboard, MIDI controller, or control surface.



Go *Options > Key Bindings*, and you can tie functions to particular keystrokes, as well as see a list of existing commands. In this example, ctrl L calls up the *Edit > Groove-Clip Looping* function.

- Use Layouts to organize specific combinations of windows for certain tasks, like mixing, overdubbing, different types of editing, etc. This requires less effort than opening windows and dragging them around.



Any time you want to save a particular layout, go *View > Layouts > Add.* This brings up a window where you can name the layout. After naming it, click on OK, which adds the new layout to the list.

- The use of color and graphics goes well with right-brain thinking, as your brain can decode colors more easily than words. This is why it's important to customize Sonar for the way that works best for you. For example, in Sonar you can change colors (go *Options* > *Colors*) for the background, tracks, faders (e. g. volume and pan), and many other graphic elements; you can even choose a picture or pattern behind the tracks, like your studio's logo.

8.2.1 Feel Versus Perfection

Some older recordings, created under technically primitive conditions, still conveyed a joyousness and enthusiasm—a "feel"—that made for great music. And some newer albums are so perfect, so automated and equalized, that the sound is sterile and somehow mechanical. There are some producers who believe that feel is all important; if a musician does a great part but blows a couple of phrases, that's all right if the feel was good. Other producers insist on doing a part over and over and over until it's technically perfect. Both approaches have their advantages and pitfalls, so try to strike a balance. Don't fall into the trap of being so self-critical that you never complete anything, but also don't get so loose that everything sounds "great" and you lose the ability to evaluate.

One of the wonderful aspects of computer-based recording is you can save your mixes as you go along. You may find out that it was the first or second mixes, not the last ones, that had a certain quality. You might also find that combining different parts of different mixes can do wonders for a tune—maybe you were really inspired when doing the intro to one mix, but nailed the middle of a different mix. Although most people do this kind of editing in a two-track digital audio editor, you can import your final mixes into Sonar, get out the scissors tool, cut and move, then export the edited version.

8.3 Challenges Facing the Solo Artist

There's more to being a solo artist than just dealing with the right/left brain dichotomy. The fact that one person can write, play, produce, record, master, and even duplicate music is unique to modern times. But just because we *can*, does that mean we *should?* There's much to recommend human interaction, and the reality check that comes from a trusted associate who can give honest, objective feedback (and in the case of a producer or engineer, offload some of the left-brain activities).

Many readers are fortunate enough to work with friends and associates, while others, for any number of reasons, tend to work solo. Is this an inherently flawed concept? Not necessarily, because doing all the tasks yourself is highly educational. Programming drum parts made me a better bass player. Producing myself forced me to be more objective, and engineering—well, I learned that mostly from working with some really fine engineers and have tried to continue on from there. Over the years, I've gotten reasonably skilled at these arts. Sure, I'd rather have Sheila E. do my drum parts, but you can't have everything.

The key to pulling off the difficult task of being a solo musician is *not to fall in love with your music*. Distance yourself from what you do, so you can make the kind of objective decisions normally reserved for the producer. Following are some tips on how to create "better music through detachment."

8.3.1 The Radio Factor

A song's intro is crucial. If a radio station or A&R person doesn't like the first 10 seconds, you're through. They *might* listen to 10 seconds of your next cut, but don't count on it.

Here's a test for intros. Picture an office party filled with a variety of people, from the new mailroom guy to upper management. They're all a bit tipsy and chatting away, while the radio (whose quality won't be as good as the monitors in your studio) provides background music. A commercial

comes on, followed by an announcer saying the station logo—"K-TONE, where the music still matters"—then they lead into your song. Picture this scene as vividly as you can in your imagination.

Try to put yourself in the position of one of the partygoers, then "look" around you. How do the people react? Do they stop talking and listen? Do they listen for the first few bars, then go back to conversing? Do they ignore it entirely? Is there something in the first few seconds to grab their attention and keep it? For your tune to be played on the radio, think of it in the context of radio play. It has to be able to segue from anything to anything, appeal to short attention spans, and be different. Doing this exercise can help clarify what needs to be done to make the song stand out more.

The bottom line: every mix should, if at all possible, do something to grab people's attention in the first 10 seconds or they may tune out.

8.3.2 Got Live If You Want It

My preferred way to test a song is to play it in front of an audience (preferably non-musicians). It's the quickest way to find out what connects and what doesn't. Then you can apply that feedback to improving the song. But if you can't do that, to simulate the effect of playing a piece one-on-one to an audience, I go back to square one, pick up a single guitar or keyboard, and re-arrange the song for playing as a solo performer in real time.

Something good happens every time I try this. For example, in one song I had what I thought was a nifty little instrumental figure between the verse and chorus. It was not possible to duplicate with solo guitar, so I substituted an alternate chord pattern—which ended up being more compelling than the original, and as a side benefit, could be played live.

Remember, songs were once honed on the road, then captured in the studio. Now songs are often created in the studio, and re-created on the road. As you mix a tune, always imagine an audience is listening. It will make a difference in how the song develops.

8.3.3 The Cleverness Factor

For me, the paramount lesson from doing years of studio work behind songs was that *everything* supports the lead singer. Your licks are there *only* to make the lead vocal more effective.

Many years ago, I came up with a lyrical, melodic bass part for a verse. It was composed in isolation, while waiting for the engineer to get a good snare drum sound, and I fell in love with the part. But played behind the vocal, it was too distracting. The producer told me to simplify the part, and I ended up playing something that any moron who had just picked up a bass could play. It was hard to let my clever bass part go, but the simpler version made a far greater contribution to the tune.

When mixing, many times it's what you *mute* that makes the song work, not what you leave in. If you recorded 30 tracks, don't feel you have to use them all. The less there is going on, the more important the remaining parts become. The mixing process is your last chance to be brutally honest: If something doesn't work quite right, get rid of it, regardless of how clever it is or how good it sounds on its own.

8.3.4 What Are Your Real Goals?

Of course, all this advice assumes that you *want* to connect with an audience. But I don't necessarily advocate that. Creating music is, in the larger sense, about self-discovery, and that's the magical part. Even if told that I'd never sell another CD in my life, I'd still make music.

I feel there are only two ways to be successful. One is to be totally true to yourself, and hope that the music you create strikes a chord in others as well. This usually creates the brightest stars with the longest careers, because there is no artifice. And if it doesn't "fit" with a mass audience, at least what you have is honest, and your friends will probably love it.

The other option is to carefully study past hits, cool chord progressions, pick lyrical subjects with wide appeal, etc., and do mixes that are designed to appeal to specific audiences. I've known songwriters who take this

approach, and while there is always a kernel of soul in what they do, they approach writing more as a business than as art. That's fine too and can lead to a comfortable, well-paying, career without the drawbacks of fame. In that case, you really need to study mixes so your tunes can fit in with what's "commercially acceptable." And you may need to add a lot more compression when mastering because "everyone else does it," not because you necessarily think it's appropriate.

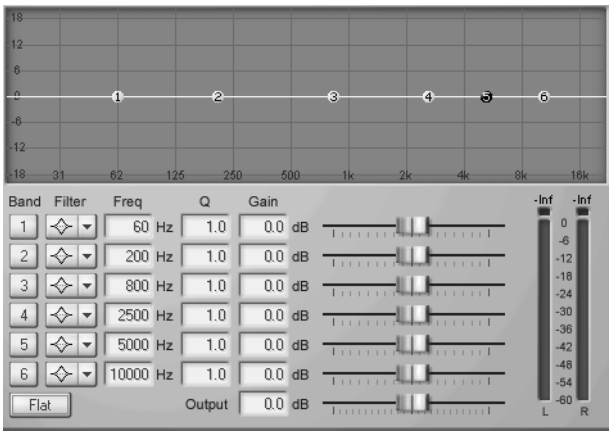
I think combining the two approaches yields the best results. Let the artist in you create, then let the hard-headed, objective part of you produce, mix, and master. While this section has concentrated on what it takes to become more objective, I don't mean to trivialize the creative factor. As in so many aspects of life, it's the synthesis of opposites that creates the best results. Go ahead, love your music—but don't be in love with it if you want to remain objective.

Okay, enough opinions ... let's get technical.

8.4 About Frequency Response and Hearing

One goal of mixing and mastering is to produce a balanced, even sound. It should have a full, satisfying bass without "muddiness," a well-defined midrange, and sparkly (not screechy) highs. To achieve this, as well as use equalization properly, we need to understand frequency response.

Frequency response defines how a system records or reproduces the spectrum of audible frequencies, which stretches from 20Hz to 20,000Hz. (Hz, short for Hertz, measures the number of cycles per second in a wave; 1kHz or kilohertz equals 1,000Hz.) This is usually shown on a graph. The Y-axis (vertical) shows level, and the X-axis (horizontal) indicates frequency.



Here, the graph shows a straight line from 0 to 20kHz. This is called a *flat response*, which means that no range of frequencies is accented or diminished. This is confirmed by the fact that the gain controls are set to 0.0dB

The audible range is further divided into bands. These are not precisely defined, but here's a rough guide.

- Bass: Lowest frequencies, typically below 200Hz.
- Lower midrange: 200 to 800Hz.
- Midrange: 800Hz to 2.5kHz.
- Upper midrange: 2.5kHz to 8kHz.
- Treble: 8kHz and higher.

While these guidelines are approximate, they are still useful as references. For example, bass guitar and kick drum occupy the bass range. Vocals are in the midrange and lower midrange. Percussion instruments like tambourine have lots of energy in the treble region.

Although electronic devices like hi-fi amplifiers often have a flat frequency response, no mechanical device does. A speaker's response falls off at high and low frequencies. Guitar pickup response falls off at high frequencies, which is why guitar amps often boost the upper midrange.

Loud, extended mixing sessions are very tough on the ears. Mixing at low levels keeps your ears "fresher" and minimizes ear fatigue; you'll also be able to discriminate better between subtle level variations. Loud mixes may get you hyped up, but they'll also trip your ear's built-in "limiting" (ears don't hear in a linear fashion).

However, because the ear's frequency response changes depending on level, if you mix or master at *too* low a level, you might boost the bass and treble too much. Mix at a comfortable listening level—neither too loud nor too soft. Then check at both high and low levels to find a good average setting.

8.5 Monitoring and Acoustics

All the effort you put into recording, overdubbing, and mixing means nothing if your monitoring system isn't honest about the sounds you hear. The issue isn't simply the speakers; the process of monitoring is deceptively complex, as it involves your ears, the acoustics of the room in which you monitor, the amp and cables that drive your monitors, and the speakers themselves. *All of these elements work together to determine the accuracy of what you hear, and therefore, how you mix and master.* If you've ever done a mix that sounded great on your system but fell apart when played elsewhere, you've experienced firsthand what can go wrong with the monitoring process.

8.5.1 The Problem with Ears

For starters, your ears—the most crucial and important components of your monitoring system—aren't perfect. Even healthy, young ears aren't perfect, thanks to a phenomenon called the Fletcher-Munson curve. Simply stated, the ear has a midrange peak and does not respond as well to low and high frequencies, particularly at lower volumes. The response comes closest to flat response at relatively high levels. The "loudness"

control on hi-fi amps attempts to compensate for this by boosting the highs and lows at lower levels, then flattening out the response as you turn up the volume.

Another limitation is that a variety of factors can damage your ears—not just loud music, but excessive alcohol intake, deep sea diving, and just plain aging. I've noticed that flying temporarily affects my high frequency response, so I wait at least 24 hours after getting off a plane before doing anything like mixing that involves critical listening. The few times I've had to break that rule, mixes that seemed perfectly fine at the time played back too bright the next day. Also note that professional audio engineers often exhibit a dip in the all-important midrange frequencies from too much day-in, day-out exposure to louder-than-average sounds.

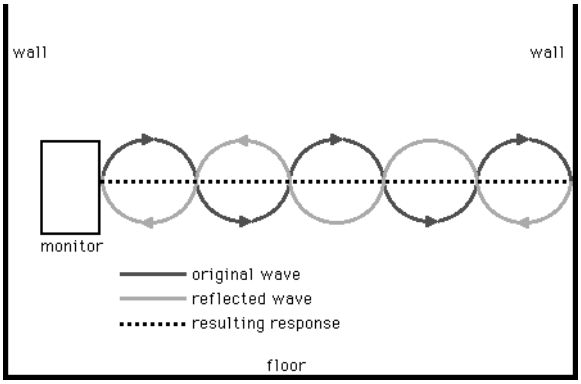
You've heard it before, but believe me: *Take care of your hearing so at least your ears aren't the biggest detriment to monitoring accuracy.* Back in my touring days when I'd often play 200 days out of the year, I wore cotton in my ears. While not as effective as present-day, high-tech earplugs, I feel it really saved my hearing. These days, I often carry the cylindrical foam ear plugs you can buy at sporting good stores. I wear them while walking city streets, at clubs, when hammering or using power tools (the impulse noise of a hammer hitting a nail is major!), or anywhere my ears are going to get more abuse than someone talking at a conversational level. I make my living with my ears, and taking care of them is a priority. Good hearing should be your priority too.

8.5.2 Other Variables

The room in which you monitor will also influence how you mix. For a real ear-opener, set up an audio level meter (e. g. the kind made by Radio Shack for monitoring workplace noise levels), sit with it in the middle of your room, run a sine wave test tone oscillator through the speakers, and watch the meter. Unless you have great monitors and an acoustically tuned room, that meter will fluctuate like a leaf in a tornado. Speakers by themselves do not have perfectly flat responses, but they look like a ruler compared to the average untreated room.

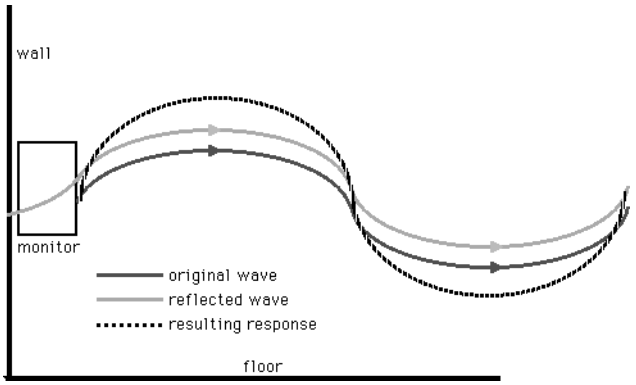
You don't even need a level meter to conduct this test: Play a steady tone around 5kHz or so, then move your head around. You'll hear obvious volume fluctuations. (If you can't hear the 5kHz tone, then perhaps it's time to look for a different line of work.)

These variations occur because as sound bounces around off walls, the reflections become part of the overall sound, creating cancellations and additions.



This illustration shows a standing-wave condition, where a wave reflects back from a wall out of phase, thus canceling the original waveform. At other frequencies, the reflection can just as easily reinforce the original waveform. These frequency response anomalies affect how you hear the music as you mix.

Another example of how acoustics affects sound is when you place a speaker against a wall, which seems to increase bass. Here's why: Any sounds emanating from the rear of the speaker, or leaking from the front (bass frequencies are very non-directional), bounce off the wall. Because a bass note's wavelength is so long, the reflection will tend to reinforce the main wave. This is a greatly simplified explanation, but it gets the principle across.



Placing a speaker with its back against the wall often gives an apparent increase in bass; placing it in a corner accentuates the bass even more.

As the walls, floors, and ceilings all interact with speakers, it's important that any speakers be placed symmetrically within a room. Otherwise, if (for example) one speaker is 3 feet from a wall and another 10 feet from a wall, any reflections will be wildly different and affect the response.

The subject of acoustically treating a room deserves a book in itself. We are just touching on the basics here to provide background on an important element of the mixing process. If you have the money, hiring a professional consultant to "tune" your room with bass traps and similar mechanical devices (this is different from room-tuning with graphic EQ) could be the best investment you ever make in your music. I can't really give specific advice here for your situation, because every room is different.

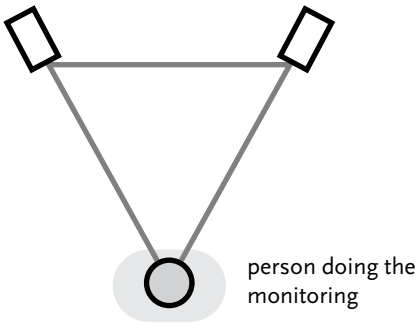
> Some people try to compensate for room anomalies by inserting a graphic equalizer just before their power amp and "tune" the equalization to adjust for room anomalies. While this sounds good in theory, if you deviate at all from the "sweet spot" where the microphone was, the frequency response will be off. Also, heavily equalizing a poor acoustical space simply gives you a heavily equalized poor acoustical space. Like noise reduction, which works best on signals that don't

have a lot of noise, room tuning works best on rooms that don't have serious response problems.

8.5.3 Near-field Monitors

Traditional studios have large monitors mounted at a considerable distance (6 to 10 ft. or so) from the mixer, with the front flush to the wall, and an acoustically-treated control room to minimize response variations. The "sweet spot"—the place where room acoustics are most favorable—is designed to be where the mixing engineer sits at the console.

In smaller, project studios, near-field monitors have become the standard way to monitor. With this technique, small speakers sit around 3 to 6 feet from the mixer's ears, with the head and speakers forming a triangle.



When using near-field monitors, the speakers should point toward the ears and be at ear level. If slightly above ear level, they should point downward toward the ears.

Near-field monitors reduce (but do not at all eliminate) the impact of room acoustics on the overall sound, as the speakers' direct sound is far greater than the reflections coming off the room surfaces. As a side bene-

fit, because of their proximity to your ears, near-field monitors do not have to produce a lot of power. This also relaxes the requirements for the amps feeding them.

However, placement in the room is still an issue. If placed too close to the walls, there will be a bass build-up. Although you can compensate for this with EQ (or possibly controls on the speakers themselves), the build-up will be different at different frequencies. High frequencies are not as affected because they are more directional. If the speakers are free-standing and placed away from the wall, back reflections from the speakers bouncing off the wall could cause cancellations and additions for the reasons mentioned earlier.

You're pretty safe if the speakers are more than 6 ft. away from the wall in a fairly large listening space (this places the first frequency null point below the normally audible range), but not everyone has that much room. My solution, crude as it is, has been to mount the speakers a bit away from the wall on the same table holding the mixer, and pad the walls behind the speakers with as much sound-deadening material as possible.

Nor are room reflections the only problem; if placed on top of a console, reflections from the console itself can cause inaccuracies. To get around this, in my studio the near-fields fit to the side of the mixer, and are slightly elevated. This makes as direct a path as possible from speaker to eardrum.

8.5.4 Anatomy of a Near-field Monitor

There are lots of near-field monitors available, in a variety of sizes and at numerous price points. Most are two-way designs, with (typically) a 6" or 8" woofer and smaller tweeter. While a 3-way design that adds a separate midrange driver might seem like a good idea, adding another crossover and speaker can complicate matters. A well-designed two-way system will beat a so-so 3-way system.

There are two main monitor types, *active* and *passive*. Passive monitors consist of only the speakers and crossovers, and require outboard amplifiers. Active monitors incorporate any power amplification needed to drive the speakers from a line level signal. I generally prefer powered monitors because the engineers have (hopefully!) tweaked the power amp and speaker into a smooth, efficient team. Issues such as speaker cable resistance become moot, and protection can be built into the amp to prevent blowouts. Powered monitors are often *bi-amped* (e. g., a separate amp for the woofer and tweeter), which minimizes intermodulation distortion and allows for tailoring the crossover points and frequency response for the speakers being used.

However, there's of course nothing wrong with hooking up passive monitors (which are less expensive than active equivalents) to your own amps. Just make sure your amp has adequate headroom. Any clipping that occurs in the amp generates lots of high-frequency harmonics (ask any guitarist who uses distortion), and sustained clipping can burn out tweeters.

8.5.5 Is There a "Best" Monitor?

On net bulletin boards, you'll see endless discussions on which near-fields are best. In truth, the answer may rest more on which near-field works best with your listening space and imperfect hearing response. How many times have you seen a review of a speaker where the person notes with amazement that some new speaker "revealed sounds not heard before with other speakers?" This is to be expected. The frequency response of even the best speakers is sufficiently uneven that some speakers will indeed emphasize different frequencies compared to other speakers, essentially creating a different mix.

Although it's a cliché that you should audition several speakers and choose the model you like best, I believe you can't choose the perfect speaker, because such a thing doesn't exist. Instead, you choose the one that colors the sound the way you prefer.

Choosing a speaker is an art. I've been fortunate enough to hear my music over some hugely expensive, very-close-to-perfect systems in mastering labs and high-end studios, so I know exactly what it should sound like. My criterion for choosing a speaker is simple: Whatever makes my "test" CD sound the most like it did over the high-end speakers wins.

If you haven't had the same kind of listening experiences, book 30 minutes or so at some really good studio and bring along one of your favorite CDs (you can probably get a price break because you're not asking to use a lot of the facilities). Listen to the CD and get to know what it should sound like, then compare any speakers you audition to that standard. For example, if the piano on your mix sounds a little understated on the expensive speakers, choose speakers where the piano is equally understated.

One caution: if you're A-B comparing a set of speakers and one set is slightly louder than the other (even 1/10th of a dB can make a difference), you'll likely choose the louder one as sounding better. Make sure the speaker levels are matched as closely as possible in order to make a valid comparison.

A final point worth mentioning is that speakers have magnets which, if placed close to monitors, can distort the monitor's display. If you plan to place a monitor near the speakers (e. g., audio for video work), go for magnetically-shielded speakers as they do not exhibit this problem.

8.5.6 Learning Your Speaker and Room

Ultimately, because your own listening situation is imperfect, you need to "learn" your system's response. For example, suppose you mix something in your studio that sounds fine, but in a high-end studio with accurate monitoring, the sound is bass-heavy. That means your monitoring environment is shy on the bass, so you boosted the bass to compensate (this is a common problem in project studios with small rooms). In future mixes, you'll know to mix the bass lighter than normal in order to have it come out okay.

Compare midrange and treble as well. If vocals jump out of your system but lay back in others, then your speakers might be "midrangey." Again, compensate by mixing midrange-heavy parts back a little bit.

Also, decide on a standardized listening level. I believe in monitoring at low levels when mixing, not just to save my ears, but also because if something sounds good at low volume, it will sound great when really cranked up. However, this also means that the bass and treble might be mixed up a bit more than they should be to compensate for the Fletcher-Munson curve. So, before signing off on a mix, I check the sound at a variety of levels. If at loud levels it sounds just a hair too bright and boomy, and if at low levels it sounds just a bit bass- and treble-light, that's about right.

8.5.7 Headphones, Hi-fi Speakers, and Satellite Systems

Musicians on a budget often wonder about mixing over headphones, as $100 will buy you a great set of headphones, but not much in the way of speakers. Although mixing exclusively on headphones is not a good idea, I highly recommend keeping a good set of headphones around as a reality check (not the open-air type that sits on your ear, but the kind that totally surrounds your ear). Sometimes you can get a more accurate bass reading using headphones than you can with near-fields. Careful, though: It's easy to blast your ears with headphones and not know it. Watch those volume levels (and be real careful about accidentally setting up a feedback loop—a loud enough squeal could cause permanent hearing damage).

As to hi-fi speakers, here's a brief story. For almost 15 years, I mixed over a set of trusted bookshelf speakers in my home studio. These were some of the least sexy-sounding and most boring speakers in the world. But they were neutral and flat, and more importantly, I had "learned" them during the process of taking my mixes to many pro studios for tweaking or mastering. In fact, when listening over expensive speakers, the sound was almost always exactly what I expected, with one exception: Signals below about 50Hz simply vanished on my speakers. Therefore, with instruments like orchestral kick drums, I had to mix visually by checking

the meters, then verifying the mix at another facility. Thankfully, I've since upgraded to "real" near field monitors that can hear signals down to about 30Hz.

So yes, you can use hi-fi speakers if you absolutely must, assuming they're relatively flat and unbiased (watch out; some consumer-oriented speakers "hype" the high and low ends). However, they often aren't meant to take a lot of power, so be careful not to blow them out. One other tip: Unless the manufacturer states otherwise, I recommend placing bookshelf speakers horizontally, with the tweeters on the outside. This gives better stereo separation than mounting the speakers vertically.

Lately, "satellite" systems have appeared where the near-fields are physically very small—in fact, too small to produce adequate bass (some would argue that no 6" or 8" speaker can really produce adequate bass, but sometimes we need to reconcile finances and space with the laws of physics). To compensate, a third element, the "subwoofer," adds a fairly large speaker and is crossed over at a very low frequency so that it reproduces only bass notes. This speaker usually mounts on the floor, against a wall; in some respects placement isn't too critical because bass frequencies are relatively non-directional.

Can you use satellite-based systems to make your computer audio sound great? Yes. If you're living space is tight, is this a good way to make your hi-fi setup less intrusive? Yes. Would you mix your major label project over them? Well, I wouldn't. Perhaps you could learn these systems over time as well, but I personally have difficulty with the disembodied bass when it comes to critical mixes.

8.5.8 Testing on Multiple Delivery Systems

I'm distrustful enough of speakers that before signing off on a mix, I'll run off a CD or two and listen through anything I can—car stereo speakers, hi-fi bookshelf speakers, big-bucks studio speakers, boom boxes, headphones, etc. This gives me an idea of how well the mix will translate over a variety of systems. If the mix works, great—mission accomplished. But if it sounds overly bright on, say, five out of eight systems, I'll pull

back the brightness just a bit. Of course, some of this can be compensated for during the mastering process, but ideally, you want any project to require the least amount of mastering possible.

Many "pro" studios will have their big speakers mounted in the wall, a pair of near-fields for reality testing, and some "junk" speakers sitting around to check what something will sound like over a lo-fi consumer system (such as the average TV). Switching back and forth among the various systems can help "zero in" on the ultimate mix that translates well over any system.

8.5.9 The Learning Curve

If all of the above sounds like there's a learning curve ahead of you, that's true. Pay attention to your hearing first, then the room acoustics, then the monitor. Once you've found a good speaker location that is unlikely to change, get to know the sound so you can mentally compensate for any response anomalies.

The more you monitor, the more educated your ears will become. Also, the more dependent they will become on the speakers you use (some producers carry their favorite monitor speakers to sessions so they can compare the studio's speakers to speakers they already know well). But even if you can't afford the ultimate monitoring setup, with a bit of practice you can learn your system well enough to produce a good-sounding mix that translates well over a variety of systems—and that's much of what mixing is all about.

8.6 Plug-ins: Tools for Mixing and Mastering

Plug-ins are accessory programs that "plug in" to the host. For example, with Sonar, a signal processing plug-in can insert in a track (via the FX bin, also called Insert Slot), just like how you'd patch a hardware signal processor into the patch points in a mixing console.

There are two main types of plug-in technology: *host-based* (also called *native)* and *hardware-based*. Hardware-based plug-ins run only with certain specialized hardware computer cards designed for digital signal processing, such as the UAD-1 series from Universal Audio/Mackie, the PowerCore plug-ins from TC Works, and CreamWare's Pulsar-based effects and soft synths. Host-based plug-ins use the computer's microprocessor to do any needed digital signal processing, and therefore require no specialized hardware.

Native plug-ins require a certain amount of CPU power, so the more plug-ins you run (especially software synthesizers), the harder the CPU has to work. As a result, there are limits as to how many plug-ins you can use with a software program. If you want to run more plug-ins, the two main solutions are to use a faster CPU (e. g., 2GHz instead of 500MHz) or increase the system latency (the time required for the system to process signals). Increasing latency means the CPU doesn't have to work as hard, but it increases the response time when moving faders, playing soft synths, etc.

Sonar has a "CPU meter" in the lower right corner (as well as a disk activity meter) that shows how much power is being used. Compare its readings before and after loading a plug-in to see how much power a plug-in requires. The advantage of hardware-based plug-ins is that they place very little load on the CPU; the hardware board does the hard work.

The CPU meter shows how much Sonar is exercising the CPU at any given moment.

As to plug-in formats, Sonar accepts plug-ins that meet Microsoft's DirectX specification. Sonar 3 also accepts VST-compatible plug-ins through the use of a "wrapper" accessory program that makes VST devices "look" like DirectX ones.

> "VST" stands for Virtual Studio Technology, a term coined by Steinberg when they first "virtualized" signal processors, mixers, and other elements into their sequencing software. In other words, these became "native" parts of the computer environment rather than outboard hardware devices.

Older versions of Sonar can use Cakewalk's VST-DX Adapter, formerly made by Fxpansion, to run VST plug-ins.

Note that if Cakewalk's VST-DX Adapter program is installed, other programs that accept only DirectX devices (e. g., Sonic Foundry Vegas or Sound Forge) can also access VST plug-ins.