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Cubase SX/SL—Mixing & Mastering

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Welcome ...

Steinberg unveiled the first version of Cubase VST in early 1996. The idea was simple yet brilliant: With Cubase VST, Steinberg put a tool into the hands of recordists that enables us to produce professional-sounding tracks using just a computer and audio card— no costly recording hardware required.

Cubase VST was endowed with virtual instruments; the range of its functions grew steadily until its user interface was bursting at the virtual seams with features. Time for a new Cubase.

The torch was handed over at the 2002 Frankfurt Musikmesse. Cubase VST was mothballed; Cubase SX/SL was born. If you're an old hand at VST looking to ease on over to SX/SL, think again. Cubase SX/SL is a completely new program rather than a major update. As such, it calls for some woodshedding. If you're a newbie, you don't need recalibrate your sequencing sensibilities; all you have to do is get hip to what Cubase SX/SL is here for—to make music.

That's exactly what this book will do for you. It gets to the point without a lot of blather, and that point is to create great sounding songs in Cubase SX/SL. This requires more than merely program handling skills. You need to know how to make the most of effects, mixer automation options, and VST Instruments. And Cubase SX/SL is more than a sequencer. It puts a virtual audio studio at your fingertips, so it's important to learn the techniques of music production. This book will help you to become proficient at recording, arranging, mixing, mastering, and burning tracks on CD.

Here's wishing you lots of fun and success with your Cubase SX/SL-powered music productions. We sincerely hope that alongside providing valuable insight, this book inspires you to use that knowledge creatively.

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Christian Deinhardt

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Chapter 7

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.

7.1 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 and rides herd on the arrangement, 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 show that the guitarist can play lots of notes in a very short period of time?

7.2 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.

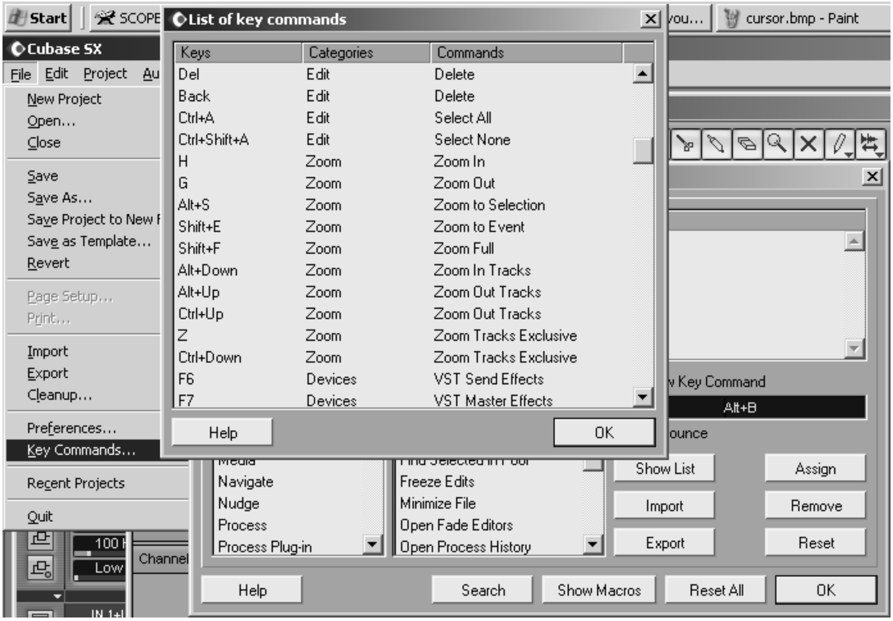
So what does this have to do with mixing? *Everything*—here's why.

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 gets to stay in left-brain mode, the artist gets to stay in 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.

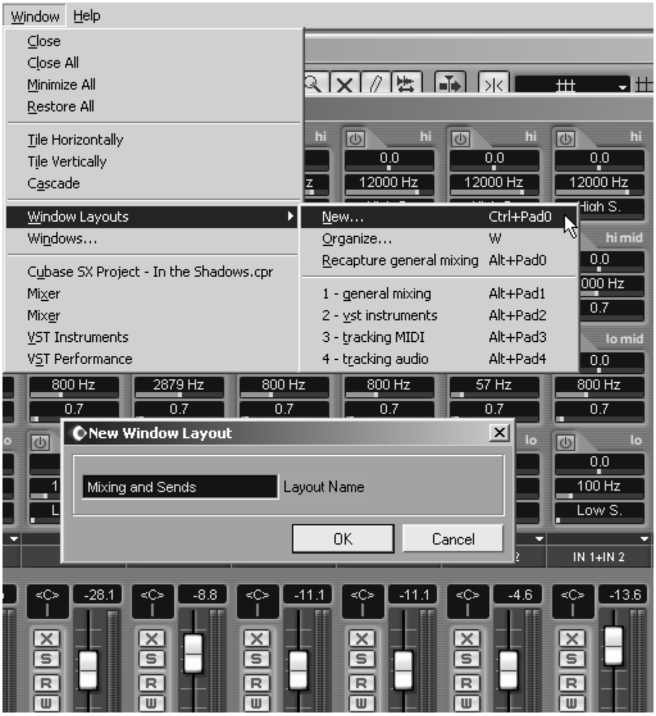
However, if you're flying solo, there are still ways to reconcile the right brain/left brain dichotomy. The most important thing is to make sure you don't have to think about left-brain activities, so you can stay in right-brain mode. If working with Cubase becomes second-nature, it will be that much 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 part on the screen, move your mouse to it, go down a menu, select an item, etc. Cubase makes it easy to create your own set of Key Commands, as well as display a list of commands. It's also possible to create Macros, which allow individual keystrokes to trigger a chain of commands (e.g., use a keystroke to remove DC offset from an audio file, then normalize it).



Go FILE > KEY COMMANDS, and you can create new key commands, see the list of existing commands, create Macros, and even export your key commands for use in other Cubase studios.

- Use Window layouts to organize specific combinations of windows for certain tasks, like mixing, overdubbing, working with VST instruments, setting up send effects, etc. This requires much less thought than opening windows and dragging them around.



You can save a particular layout of windows at any time by going WINDOW > WINDOW LAYOUTS > NEW, then typing in a name for the layout and clicking OK. For more information on Window Layouts, see Chapter 9.

- 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 Cubase for the way that works best for you. For example, in Cubase SX, you have the option of seeing EQ settings as knob positions or lines. I find it easier to scan the lines to see what's happening compared to taking a look at the knobs (and I would find it even easier if the lines changed color depending on their setting, like the way Cubase handles velocity indications in the MIDI piano roll editor). You may prefer the knobs, though—use what works best for you.

7.2.1 Feel vs. Perfection

Some older albums, recorded 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 the 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 Cubase, get out the scissors tool, cut and move, then export the edited version.

7.3 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."

7.3.1 The Radio Factor

A song's intro is crucial. If a radio station or A&R person doesn't like the first ten seconds, you're through. They *might* listen to ten 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 is nowhere near 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 ten seconds.

7.3.2 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 used to be honed on the road and 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.

7.3.3 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.

7.3.4 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 I was 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.

7.4 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 1000 Hz.) 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.

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.

7.5 Monitoring and Acoustics

All the effort you put into recording, overdubbing, and mixing is for 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.

7.5.1 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 broken 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.

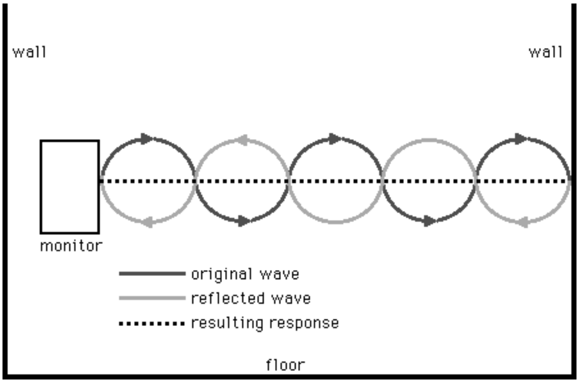
You've heard it before, but believe me: Take care of your hearing so at least your ears aren't the biggest detriment to monitoring accuracy! Back in my touring days when I'd often play 200 days out of the year, I wore cotton in my ears. While not as effective as present-day, high-tech earplugs, I feel 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.

7.5.2 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.

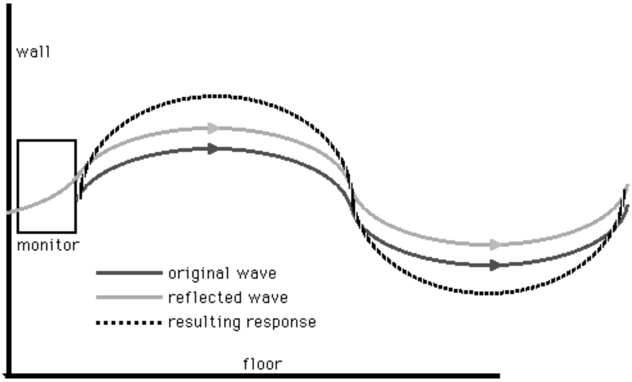
You don't even need a level meter to conduct this test: play a steady tone around 5 kHz or so, then move your head around. You'll hear obvious volume fluctuations. (If you can't hear the 5kHz tone, then perhaps it's time to look for a different line of work.)

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 three feet from a wall and another ten feet from a wall, any reflections will be wildly different and affect the response.

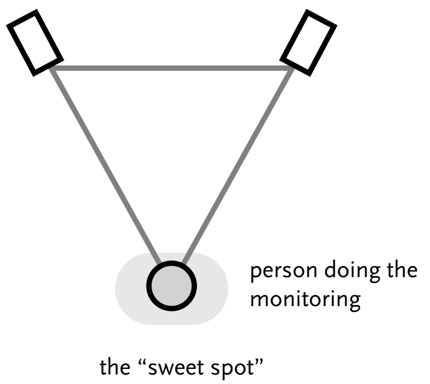
The subject of acoustically treating a room deserves a book in itself; we are just touching on the basics here to provide background on an important element of the mixing process. If you have the money, hiring a professional consultant to "tune" your room with bass traps and similar mechanical devices (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.

7.5.3 Near-field Monitors

Traditional studios have large monitors mounted at a considerable distance (six to ten feet 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 three to six 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 benefit, 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 six feet 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.

7.5.4 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.

7.5.5 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 1⁄10 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.

7.5.6 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.

You also have to 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.

7.5.7 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 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.

7.5.8 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 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.

7.5.9 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 location for the speakers 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 the good side of all this is that 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.

7.6 Plug-ins: Tools for Mixing and Mastering

Steinberg invented the VST (Virtual Studio Technology) concept where signal processors, mixers, and other elements of the studio became "native" parts of the computer environment. The VST plug-in format is cross-platform (i.e., compatible with both Mac and Windows), however on the PC, Cubase also accepts the Microsoft DirectX plug-in format.

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 XTC 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.

7.7 The Mixing Process

Mixing is not only an art, it's the ultimate arbiter of how your music sounds. A good mix can bring out the best in your music, while a bad mix can obscure it.

An effective mix spotlights a composition's most important elements, adds a few surprises to excite the listener, and sounds good on any system—from a transistor radio to an audiophile's dream setup. Translating a collection of tracks into a cohesive song isn't easy; mixing requires the same level of creativity and experience as any part of the musical process.

To understand why mixing is so tricky, we need to examine some of the problems involved in trying to make a good mix. But first, let's decide what's going to hold our final mix.

7.8 Mixdown Options

There are several ways to create a mixed file in Cubase SX/SL. Following are the most common options.

7.8.1 Exporting a Mixed File from within Cubase ("Rendering to Disk")

The easiest way to create a final stereo mix in Cubase is to adjust all the settings exactly as you want (EQ, levels, etc.), including any automation you want to use. When the mix is perfect, you can then go FILE > EXPORT > AUDIO MIXDOWN and "render" the file in any one of several formats, with sample rates up to 96 kHz, and bit resolution up to 32-bit floating point. There are two main "families" of file types:

- WAV, AIFF, and Broadcast WAV format files can be imported by CD-burning and digital audio editing programs.
- MP3, Ogg Vorbis, RealAudio, and Windows Media Audio format files use *data compression* (which can degrade the sound) so they don't need as much data, allowing them to stream more easily over the web or be used in multimedia productions. For example, a tune that requires 40MB as a WAV file, when compressed by 10:1, ends up as a svelte 4MB file.

Do not confuse data compression, which reduces the amount of data a song requires, and audio compression, which restricts a signal's dynamic range. Also, note that the data-compressed file formats available to Cubase use "lossy" compression. In other words, you cannot "uncompress" the data and end up with the original; data that is discarded to create a smaller file size cannot be retrieved.

7.8.2 Mixing to Analog Tape

Analog tape is a signal processor that some engineers feel "warms up" the sound, due to inherent distortions within the tape medium itself. These people sometimes prefer mixing to analog tape as an alternative to staying in the digital domain.

Of course, this is easy to do with Cubase: assign its master output to two channels of your audio interface's analog outputs (or digital outs, followed by a digital-to-analog converter), and patch these into the recorder's analog inputs. Pros generally prefer mixing to the widest tape possible at the fastest possible speed (e. g., ½-inch tape at 30 inches per second). Although there's always controversy about analog vs. digital, the bottom line is that 30 ips tapes can sound great. Add Dolby SR noise reduction, and you have sound quality that need make no excuses to digital systems.

7.8.3 Mixing Direct to Digital

In this case, you connect your audio interface's digital output (usually either SPDIF or AES/EBU) to the digital input of a stand-alone CD recorder, DAT machine, Minidisc, or another computer.

> SPDIF is considered a "consumer" standard, but is used a lot in pro circles as well. It has two variations: a wire-based version that terminates in RCA phono jacks, and an optical version that uses TOSLINK optical connectors. AES/EBU interfaces are balanced digital lines rather than unbalanced, allowing them to cover a greater distance without signal degradation. Although AES/EBU can be either optical or wired, it is usually carried over balanced lines terminating in XLR connectors.

Be careful to observe two important points:

- The sample rate of the Cubase project, audio interface, and mixdown device must all be the same. For CDs, the standard sample rate is 44.1kHz; broadcast generally uses 32kHz, and video runs at 48kHz. Surround sound and other "high-resolution" audio may use sample rates up to 96 or even 192kHz. However, as the most common delivery medium at present is the CD, 44.1kHz is currently the most common sample rate.

- The mixdown device should be set to slave to the clock signal coming out of the audio interface. This insures that the mixdown device will synchronize properly to a digital control signal (called *word clock)* that is carried by the digital out. Lack of sync, which can be caused by setting both devices as clock sync masters, usually leads to (at the very least) pops and clicks in the audio, and in extreme cases can cause "tearing" sounds and drop outs. Your mixdown device's manual should include information on how to set it as a slave to a master clock source.

7.8.4 Mixing to a Digital Recorder via Analog Connections

Digital offers perfect transfers, but some prefer patching in analog mastering tools between the computer's output and the mixdown device's input. This includes units like Dolby's famous Model 740 Spectral Processor, "vintage" gear like the Pultec equalizer, and so on; you'll need an audio interface with analog outputs and a mixdown device with analog inputs. Otherwise, you may need to add a stage of conversion (e.g., an analog-to-digital converter if your mixdown device has only digital inputs).

7.8.5 Mixing Back into Cubase via a Digital or Analog Mixer

An often-overlooked option is sending multiple busses from Cubase into the inputs of a digital or analog mixer, and simultaneously recording the mixer's output in two Cubase tracks. There are some good reasons for doing this:

- You can patch analog effects into the mixer (like a really high quality hardware reverb) or patch the mixer output through analog mastering tools.
- You can use the mixer's onboard EQs instead of Cubase's to lighten the host processor's load.

- You can move faders and twist knobs on the mixer in real time when mixing, thus having some of the advantages of a "control surface."
- When you save the Cubase project, you save not only the source tracks, but the final mix as well.

For more information on sending Cubase outputs to a physical mixer, see Chapter 26.

7.9 Before You Mix

Preparation for the mix begins the moment you start recording, and part of that involves recording the cleanest possible signal. Eliminate as many active stages as possible between source and recorder; many times, devices set to "bypass" may not be adding any effect but are still in the signal path, which can add some slight noise or signal degradation.

How many times do line level signals go through preamps due to lazy engineering? If possible, send sounds directly into your audio interface—bypass any mixer or preamp altogether. For mic signals, use an ultra-high quality outboard preamp and patch that directly into the audio interface rather than use a mixer with onboard preamps.

Always record with the highest possible fidelity. Although you may not hear much of a difference when monitoring a single instrument, with multiple tracks the cumulative effect of stripping the signal path to its essentials can make a significant difference in the sound's clarity.

7.9.1 The Arrangement

Before you even think about turning any knobs, scrutinize the arrangement. Solo projects are particulary prone to "clutter" because as you lay down the early tracks, there's a tendency to overplay to fill up all that empty space. As the arrangement progresses, there's not a lot of space for overdubs.

Here are a couple of suggestions when tracking that will make it much easier to create a good mix:

- Once the arrangement is fleshed out, go back and recut any overly-busy tracks that you cut earlier on. Try to play these tracks as sparsely as possible to leave room for the overdubs you've added. Sometimes I've found it very helpful to recut a song from scratch as soon as I've finished mixing it, or have played it live numerous times. Like many others, I write in the studio, and often the song will have a slightly tentative feel because of that. Recutting always seem to both simplify and improve the song.
- With vocal-based songs, try building a song around the vocalist instead of completing the rhythm section and then laying down the vocals. I often find it better to record simple "placemarkers" for the rhythm section, then immediately get to work cutting the best possible vocal. Then I go back and re-record the rhythm section. When you recut the rhythm section for real, you'll be a lot more sensitive to the vocal nuances.

7.10 Mixing: The 12-step Program

You "build" a mix over time by making a variety of adjustments. There are (at least!) twelve major steps involved in creating a mix, but what makes mixing so difficult is that these steps interact. Change a track's equalization (tone quality), and you also change the level because you're boosting or cutting some element of the sound. Alter a sound's stereo location, and you may need to shift the ambience or equalization. In fact, you can think of a mix as an "audio combination lock" since when all the elements hit the right combination, you end up with a good mix.

Let's look at these twelve steps, but remember, this is just one person's way of mixing—you might discover a totally different approach that works better for you.

7.10.1 Step 1: Mental Preparation and Organization

Mixing requires a tremendous amount of concentration and can be extremely tedious, so set up your workspace as efficiently as possible. Cubase's Window Layouts options are particularly helpful in this regard, as you can switch with a couple mouse clicks among different mixer views, sets of channels, and so on.

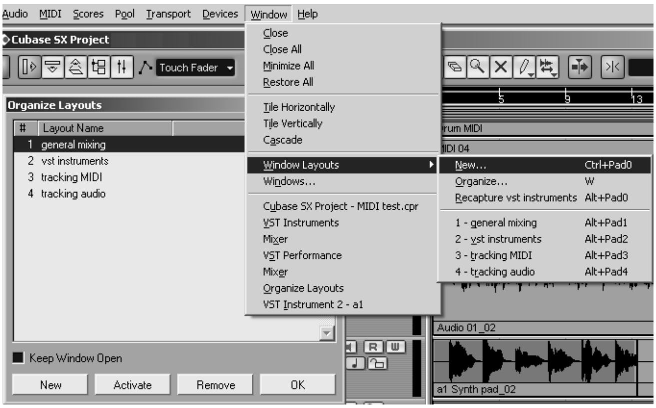
To save a particular window layout:

1 Set up your windows exactly as desired, for example, as a mixing environment.

2 Go WINDOW > WINDOW LAYOUTS > NEW.

3 Name the layout, such as "Mixer with VST Performance."

4 Click on OK.



You can create a new window layout by going to the Window menu, or by keeping the Organize Layouts window open (check the "Keep Window Open" box) and clicking on the New button. The Organize Layouts window is also where you can activate or remove window layouts.

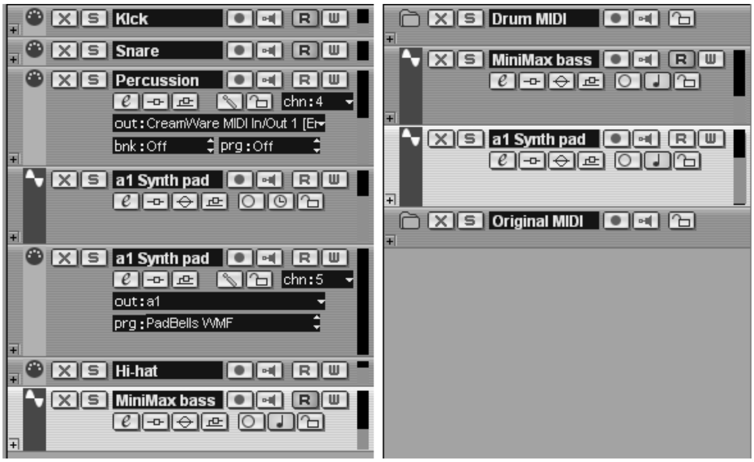
To recall a particular window layout, go WINDOW > WINDOW LAYOUTS and select the desired layout. You can also select "Organize," where you can activate particular layouts, as well as remove or create new ones.

Layouts remember only which windows are open and their spatial relationship. You cannot, for example, save one mixer view that shows EQs and another mixer view that shows Inserts.

In any event, no matter how efficiently you work, for best results take a break periodically (every hour or so is a good interval) to "rest" your ears and gain a fresher outlook on your return. Even a couple minutes of off time can restore your objectivity and, paradoxically, complete a mix much faster.

7.10.2 Step 2: Review the Tracks

Listen at low volume and familiarize yourself with the tracks. Make sure all tracks are named, note which tracks have active plug-ins that may need to be adjusted, and the like. Group sounds logically, such as having all the drum sounds on consecutive channels.



The messy view on the left shows four MIDI drum tracks for kick, snare, percussion, and hi-hat. The hi-hat was recorded after the bass and pad parts, so it shows up below them rather than with the other drum parts. Also, the a1 Synth pad part has been converted into audio, but the original MIDI track driving it has been retained in case it needs to be edited later on.

The view on the right shows what happens after cleaning this up: all the MIDI drum tracks have been placed in a folder track; a folder track has also been created for the original MIDI parts that drove synths, but placed below everything else so it's out of the way. This layout also creates a more logical mixer channel arrangement as well.

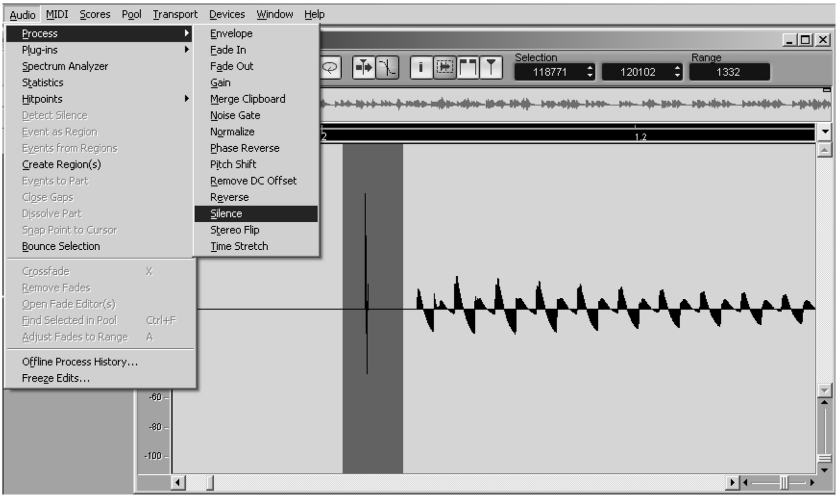
7.10.3 Step 3: Put On Headphones and Listen For Glitches

Fixing glitches is a "left brain" activity, as opposed to the "right brain" creativity involved in doing a mix. Switching back and forth between these two modes can hamper creativity, so do as much cleaning up as possible—erase glitches, bad notes, scratch tracks, and the like—before you get involved in the mix.

I highly recommended using the Solo button to solo each track and listen to it from beginning to end. With MIDI tracks, check for duplicate notes that "flam" or create chorusing-type effects and avoid overlapping notes on single-note lines (such as bass and horn parts).

With audio tracks, listen for any spurious noises just before or after audio appears (mic handling sounds if the vocalist likes using a hand-held mic, vibrating string on a guitar, hum from a bass amp, etc.). It's amazing how many little noises you'll hear on vocal tracks, like clicks from someone moving their tongue prior to singing. These low-level glitches may not seem that audible, but multiply them by a couple dozen tracks and they can definitely muddy things up.

It's easy to edit audio with Cubase, which now provides detailed audio editing. Double-click on a piece of audio, and a sample editing window appears with the audio. You can even zoom in far enough to draw out clicks with a pencil tool, as well asselect a region and apply an audio process (silence, fade in, fade out, normalize, etc.) as accessed via AUDIO > PROCESS. Fades are particularly useful if the audio has been cut and has a click at the beginning or end. This is something you'll hear when soloed, but if the click occurs on the beat, you might not notice it with other tracks playing.



It's easy to get rid of stray clicks: open your audio in the sample editor, select the region you want to remove, and click on AUDIO > PROCESS > SILENCE.

7.10.4 Step 4: Render Soft Synths as Audio Tracks

If you're sequencing VSTi devices via MIDI, consider converting them to hard disk tracks. This will free up DSP processing power for any effects you want to use during mixdown, or perhaps allow you to replace a not-so-great-sounding reverb with one that requires more CPU power. Also, these audio tracks will be saved in a Project folder, making the sound more “transportable” because the person receiving the project files doesn’t need to have the soft synth itself resident within Cubase. For more information, see Chapter 23 on Mixing and MIDI.

7.10.5 Step 5: Set Up a Relative Level Balance between the Tracks

Now that our preparations are out of the way, it’s time to get into mixing itself.

Do not add any processing for now. Concentrate on the overall effect of hearing the tracks by themselves and work on the overall sound; don’t get distracted by detail work. With a good mix, the tracks sound good by themselves—but sound even better when interacting with the other tracks.

I suggest settings levels in mono at first, because if the instruments sound distinct and separate in mono, they’ll only open up more in stereo. Also, you may not notice parts “fighting” with other if you start off in stereo.

7.10.6 Step 6: Adjust Equalization (EQ)

This can help dramatize differences between instruments and create a more balanced overall sound. Work on the most important song elements first (vocals, drums, and bass), and once these all “lock” together, deal with the more supportive parts. We’ll talk about EQ in far more detail in the chapter “About Equalization” on page 193.

The audio spectrum has only so much space, and you need to make sure that each sound occupies its own turf without fighting with other parts. Processing added to one track may affect other tracks; for example, if you boost a guitar part’s midrange, it may interfere with vocals, piano, or other midrange instruments. If you add more treble to a bass part so that it cuts better on little speakers, make sure it doesn’t start fighting with the low end of a rhythm guitar part. Sometimes boosting a frequency for one instrument implies cutting the same region in another instrument to make room.

One common mistake I hear with tapes done by singer/songwriters it that they (naturally) feature themselves in the mix and worry about "details" like the drums later. However, as drums cover so much of the audio spectrum (from the low frequency thud of the kick to the high frequency sheen of the cymbals) and as drums tend to be so upfront in today's mixes, it's sometimes best to mix the drums first, then find "holes" in which you can place the other instruments. For example, if the kick drum is very prominent, it may not leave enough room for the bass. So, boost the bass around 800 to 1,000Hz to bring up some of the pick noise and brightness. This is mostly out of the range of the kick drum, so the two don't interfere as much.

Try to think of the song as a spectrum and decide where you want the various parts to sit and their prominence.

7.10.7 Step 7: Add Any Essential Signal Processing

By essential, I don't mean "sweetening," but processing that is an integral part of the sound (such an echo that falls on the beat and therefore changes the rhythmic characteristics of a part, distortion that alters the timbre in a radical way, vocoding, etc.). As this sound will presumably be a part of the mix unless you change your mind later, you want to take it into account when mixing the other instruments.

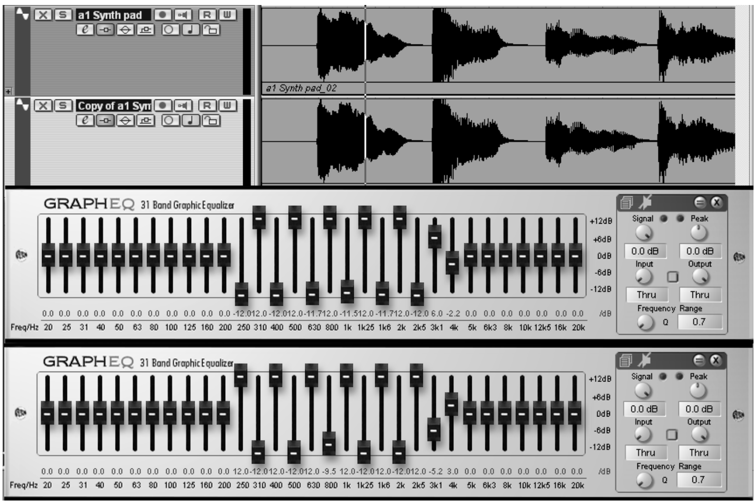
7.10.8 Step 8: Create a Stereo Soundstage

Now use the pan controls to place your instruments within the stereo field. Your approach might be traditional (i. e., the goal is to re-create the feel of a live performance) or imaginary. Pan mono instruments to a particular location, but avoid panning signals to the *extreme* left or right; they just don't sound quite as substantial as signals that are a little bit in from the extremes.

> Note that you can access the pan control for wide mixer channels, but not narrow ones. You can also find the pan control in the Inspector for audio channels, under the Channel tab.

Because bass frequencies are less directional than highs, most engineers place the kick drum and bass toward the center, unless the bass is a synth type and is in stereo. Also consider timbral balance; for example, if you've panned the hi-hat (which has a lot of high frequencies) to the right, pan a tambourine, shaker, or other high-frequency sound somewhat to the left. The same technique applies to midrange instruments as well.

Another spreading technique involves EQ. Copy a signal so it's in two channels, but equalize them differently (for example, if you have a stereo graphic equalizer plug-in, use it to cut the even-numbered bands with one channel, and the odd-numbered bands with the other channel).



Here a track has been copied. Each has a graphic EQ plug-in from CreamWare's XTC card of VST and VSTi devices. The EQs are set so the midrange boosts and cuts are equal and opposite, which adds a bit of stereo spread to mono signals. Don't forget to use the pan controls to spread these two channels.

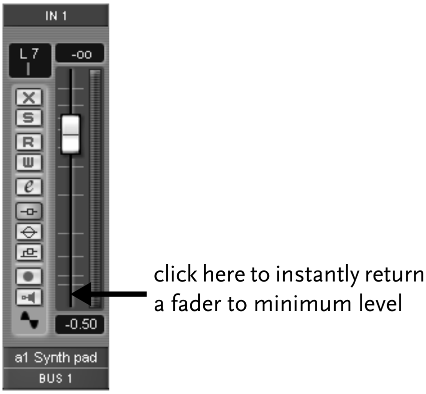
This won't work successfully on instruments with limited ranges, like voice or a lead synth part. But if you're exercising all the keys on your 88-note controller, or using a drum machine, this technique can give a very unusual type of stereo imaging.

Stereo placement can significantly alter how we perceive a sound. Consider a doubled vocal line, where a singer sings a part and then doubles it as closely as possible. Try putting both voices in opposite channels; then put both voices together in the center. The center position gives a somewhat smoother sound, which is good for weaker vocalists. The opposite-channel vocals give a more defined, sharp sound, that can really help accent a good singer.

7.10.9 Step 9: Make Any Final Changes in the Arrangement

Remember that, as with so many things in life, less is more—minimize the number of competing parts to keep the listener focused on the tune and avoid "clutter." Get rid of anything that doesn't serve the song. Conversely, if you find that a song needs some extra element, this is your final opportunity to add an overdub or two.

You can also use mixing creatively by selectively dropping out and adding specific tracks. This type of mixing is the foundation for a lot of dance music, where you have looped tracks that play continuously, and the mixer sculpts the arrangement by muting parts and doing radical level changes.



Although Cubase can automate the Mute function, you can also mute rapidly with fader automation by simply clicking at the bottom of the fader travel (just above the level indicator).

7.10.10 Step 10: The Audio Architect

Start building your space by adding reverberation and delay to give the normally flat soundstage some acoustic depth. This is also the time for more signal processing—sort of the equivalent of adding spices during the cooking process.

Generally, you'll want an overall reverb to create a particular type of space (club, concert hall, auditorium, etc.) but you may also want to use a second reverb to add effects, such as a particular "splash" on a snare drum hit or gated reverb on toms.

In the early days of recording, the general procedure was to add just enough reverb to be noticeable and simulate the effect of playing in an acoustical environment. Nowadays, reverb devices have become so sophisticated they can create effects in their own right that become as much a part of a tune as any instrumental line. However, don't drown a part in reverb. If a part is of questionable enough quality that it needs a lot of reverb, redo the part. A bad part is a bad part, no matter how much reverb you put on it.

7.10.11 Step 11: Tweak, Tweak, and Retweak

Now that the mix is on its way, it's time for fine tuning. If you're into automation, start programming your mixing moves. Remember that all of the above steps interact, so go back and forth between EQ, levels, stereo placement, and effects until you get the sound you want. Listen as critically as possible; if you don't fix something that bothers you, it will forever bother you every time you hear the mix.

7.10.12 Step 12: Check Your Mix over Different Systems

Before you sign off on a mix, check it over a variety of systems. If the mix sounds good under all these situations, your mission is accomplished.

With a home studio, you have the luxury of leaving a mix and coming back to it the next day when you're fresh, and after you've had a chance to listen over several different systems and decide what tweaks you want to make. This is one reason why automation is so wonderful—if everything was perfect about a mix except one little thing that bothers you, you can edit the automation to fix the one problem.

I can't emphasize enough that you should mix until you're satisfied. There's nothing worse than hearing one of your tunes six months later and kicking yourself because of some flaw you didn't take the time to correct, or didn't notice because you were in too much of a hurry to complete the mix.

However, you must be equally careful not to beat a mix to death. Once I interviewed Quincy Jones and he offered the opinion that recording with synthesizers and sequencing was like "painting a 747 with Q-Tips." A mix is a performance, and if you overdo it, you'll lose the spontaneity that can add excitement. A mix that isn't perfect but conveys passion will always be more fun to listen to than one that's perfect to the point of sterility. As insurance, don't always re-record over your mixes—when you listen back to them the next day, you might find that an earlier mix was the "keeper."

In fact, you may not even be able to tell too much difference among all the mixes. A record producer once told me about mixing literally dozens of takes of the same song, because he keep hearing small changes which seemed really important at the time. A couple of weeks later he went over the mixes, and couldn't tell any difference between most of the versions. Be careful not to waste time making changes that no one, even you, will care about a couple days later.

One important tip is that once you've captured your ultimate mix, you should also run a couple extra mixes, such as an instrumental-only mix or a mix without the solo instrument. These additional mixes can really come in handy at a later time, if you have a chance to re-use your music for a film or video score, or need to create extended dance mixes. Be prepared!