CHAPTER 4

Functions of Studio Equipment and Visual Representations of All Parameters

There are three components to sound: volume (or amplitude), frequency, and time. That's it. To simplify the operations of a huge variety of studio equipment, I have broken down the equipment into categories based on the function of each piece in the recording studio:

1. Sound Creators: all instruments, acoustic to electric, voice to synths
2. Sound Routers: mixing boards, patchbays, splitters
3. Sound Storers: recorders, tape players, sequencers, samplers
4. Sound Transducers: mics, pickups, headphones, speakers
5. Sound Manipulators: processing, effects

As every sound manipulator used in the studio controls either volume, frequency, or time, each can be categorized based on the main component(s) that they control:

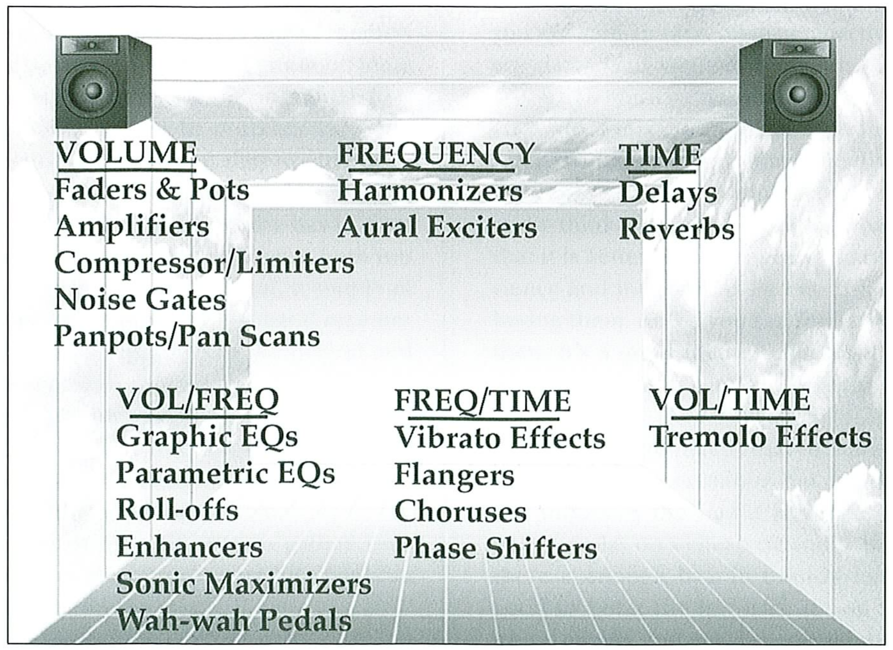
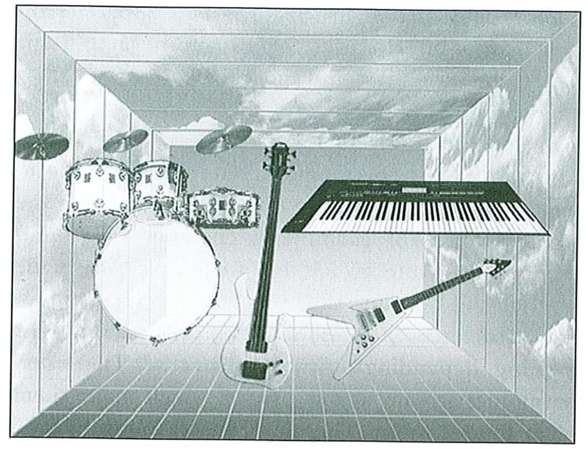


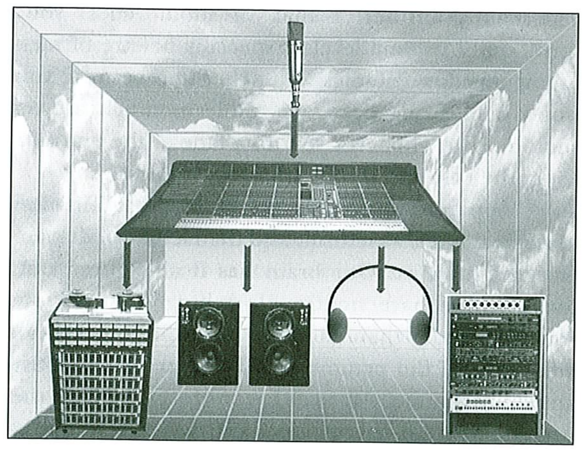
Chart 1. All Sound Manipulators

Sound creators range from acoustic to electric instruments, from voice to synthesizers.



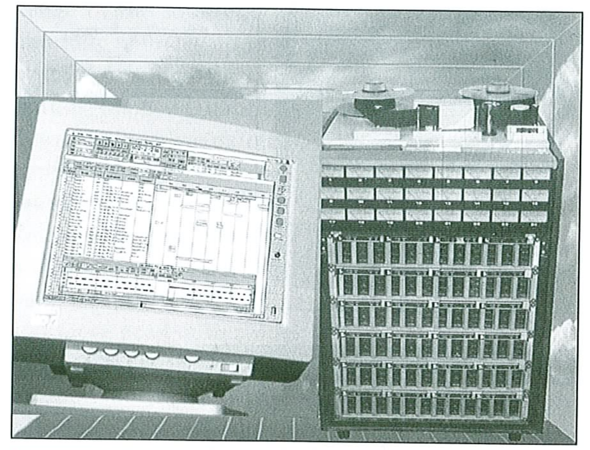
Visual 51. Sound Creators

Sound routers route sound from one place to another. Mixing boards route the signal to four places: the multitrack, the monitor speakers, cue headphones (for the band out in the studio), and the effects (so we can have a good time). Patchbays are just the back of everything in the studio—the back of the mic panels, the back of the multitrack (inputs/outputs), the back of the console (ins/outs), and the back of the effects (ins/outs)—located next to each other so we can use short cable to connect them.



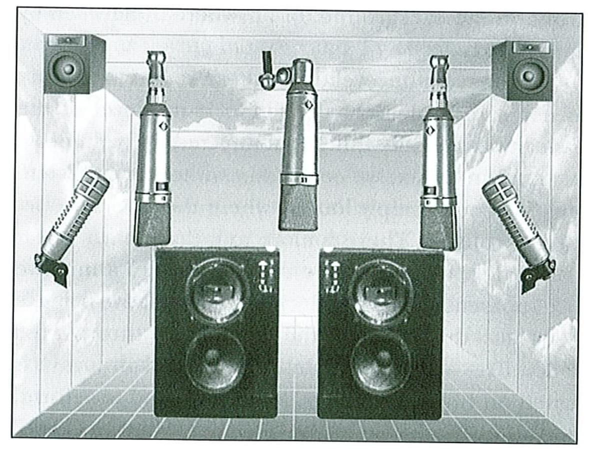
Visual 52. Sound Routers

Sound storers store sound and play it back. Tape players store digital or analog sound; sequencers store MIDI information. Some sound storers can be used to edit the sound while it is stored.



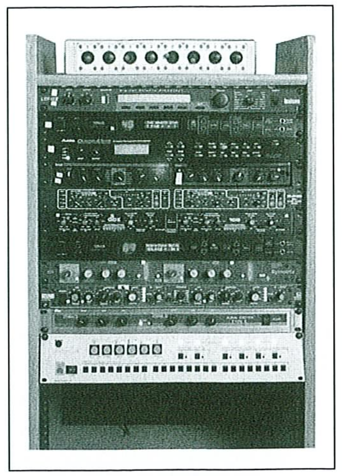
Visual 53. Sound Storers

Sound transducers take one form of energy and change it into another. Microphones take mechanical energy, or sound waves, and change it into electrical energy. Speakers take electrical energy and change it into mechanical energy, or sound waves.



Visual 54. Sound Transducers

Most of this chapter will be spent on sound manipulators. This includes processing that is used to change a sound (or effect) by adding an additional sound (or effect) to an existing sound.



Visual 55. Effects Rack

SECTION A

Volume Controls

FADERS
Volume faders control the volume of each sound in the mix, including effects. The set level of each sound is based on its relationship to the rest of the tracks in the mix. When volume is mapped out as a function of front to back, we can place any sound or effect up front, in the background, or anywhere in between by using the faders.

However, the level that we set a sound in the mix is not based solely on the fader. If the level of the faders was the only thing that affected the volume of a sound in a mix, we could mix without even listening. We could simply look at where the faders are set on the console. There is more to it than that.

When we set volume relationships in a mix, we use apparent volumes to decide on the relative balance —not just the voltage of the signal going through the fader. The apparent volume of a sound in a mix is based on two main things, fader level and waveform, and another minor one, the "Fletcher/Munson Curve" (see description next column). First, the level of the fader does affect the volume of the sound. Change the level of the fader and the sound gets louder or softer.

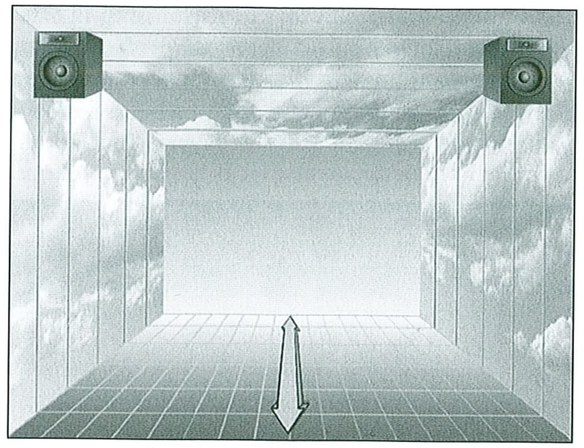
Fader Level
When you raise a fader on a mixing board, you are raising the voltage of the signal being sent to the amp, which sends more power to the speakers, which increases the "sound pressure level" (SPL) in the air that your ears hear. Therefore, when you turn up a fader, the sound does get louder. So, obviously, if you want something louder in a mix, turn it up.

We use the decibel (dB) to measure the amplitude of the signal at each stage of this circuit. In fact, there are very specific relationships between voltage, wattage, and sound pressure level. Decibels are the main variable that we use to control the apparent volume of a sound. However, there is another important factor: the waveform of the sound.

Waveform (or Harmonic Structure)
The waveform, or harmonic structure, of a sound can make a big difference as to how loud we perceive the sound to be. For example, a chainsaw will sound louder than a flute, even if they are at exactly the same level on the VU meters. This is because the chainsaw has harmonics in the sound that are irritating—or exciting, depending on your perspective. These odd harmonics scream at our psyche, which make them seem louder to us. Therefore, a screaming electric guitar will sound louder than a clean guitar sound, even if they are at the exact same volume in the mix. A minor factor contributing to the apparent volume of a sound is the Fletcher/Munson Curve.

The Fletcher/Munson Curve
The biggest problem with the human hearing process is that we don't hear all frequencies at the same volume—especially those at low volumes. (Fletcher and Munson did a study that shows just how screwed up our ears are.) This is why there are loudness buttons on stereos. However, most people like extra lows and highs, so they leave the switch on all the time. The main point here is that you should check your mixes at all volume levels. Especially beware of mixing at very low volumes all of the time because you won't be hearing bass and treble as much as you should. Also, whenever you do a fade at the end of a song, the bass and treble will drop out first.

"Apparent volume" is, therefore, a combination of decibel level, waveform, and the Fletcher/Munson Curve. But relax. Our brain has it all figured out. Most people have no trouble telling whether one sound is louder than another. Our brain quickly calculates all of the parameters and comes up with the apparent volume. All we have to do is listen to the overall apparent energy coming from each sound in the mix. You use apparent volume to set volume relationships in the mix. You don't look at the faders; you listen for the relative volumes. As previously discussed, apparent volume is most naturally mapped as a function of front to back.



Visual 56. Volume as Front to Back

COMPRESSOR/LIMITERS
Compressor/limiters were originally introduced into the studio to stop the loud peaks from distorting or saturating. Compression and limiting are volume functions; their main purpose is to turn the volume down. They turn down the volume when it gets too loud—that is, when it goes above a certain volume threshold. When the volume is below the threshold, the compressor/limiter does nothing (unless broken or cheap). The difference between compressors and limiters is explained later.

Compressor/Limiter Functions
Compressor/limiters have two main functions (and three other minor ones). The first function is to get a better signal-to-noise ratio, which means less tape hiss. The second function is to stabilize the image of the sound between the speakers, which means more presence.

Better Signal-to-Noise Ratio: Less Hiss
Recording extremely dynamic sounds, with a wide variation from soft to loud, requires turning the volume down so that the loud sounds don't overload and cause distortion. Distortion is against the law. Get distortion, go to jail. But when you turn the volume down, the soft portions of the sound barely move the needles on the tape player. And if the needles are hardly moving on the multitrack, you hear as much tape hiss as you do signal. This condition is known as a bad signal-to-noise ratio and sounds very similar to an ocean: "shhhhhhhhhhhh."

By using a compressor to turn down the volume when the signal gets too loud, you can then raise the overall volume above the tape noise. By turning down the peaks, you can record the signal hotter on tape. Then, the softer portions are loud enough so that you don't hear the tape noise.

Stabilizing the Image of Sounds: More Presence
After years of using compression to get rid of hiss, people realized that sounds often appeared more present when compressed. By evening out the volume peaks on a sound, a compressor/limiter stabilizes the image of the sound between the speakers. A sound naturally bounces up and down in volume, as shown by the bouncing pointer on a VU meter. When several sounds fluctuate naturally, their bouncing up and down can become extremely chaotic. A compressor/limiter stabilizes, or smoothes out, the movements of sounds that result from these moment-to-moment fluctuations in volume. Once compressed, the sound no longer bounces around much, so the mind can focus on it better. Therefore, the sound seems clearer and more present in a mix.

The busier the mix (the more instruments and the more notes per instrument), the more the sounds in the mix are normally compressed. This is because the more sounds and notes, the more chaos. It is difficult to keep track of a large number of instruments in a busy mix in the first place. By stabilizing the sounds, the entire mix becomes clearer.

Once a sound has been stabilized, you can then turn up the overall volume and put the whole sound right in your face. This is commonly done in radio and TV commercials to make them sound louder, so that they jump out and grab your attention. This might be annoying in radio and TV commercials, but it's great for a lead guitar or any other instrument you want extremely present in the mix.

This also works when putting sounds in the background. The problem with low volume sounds is that they can easily be lost (masked by the other sounds) in the mix, especially if the volume of the sound fluctuates much. Therefore, it is common to seriously stabilize sounds that are going to be placed low in the mix with compression. They can then be placed extremely low in a mix without fear of losing them.

NOTE: A better signal-to-noise ratio is obtained by compressing the signal on its way to the multitrack. However, many engineers will also compress the signal on its way back from the multitrack during mixdown to stabilize the sound even more.

Sharper or Slower Attack
Besides less hiss and more presence, a compressor/limiter also makes the attack of a sound sharper. Once you turn down the louder part of a signal, a sound reaches its maximum volume much quicker.

With a shorter and sharper attack, sounds are much tighter, punchier, more distinct, and more precise, which makes them easier to dance to. On the other hand, a higher quality, fast compressor will actually help to remove sharp "spikes" on the attack of a sound—softening the sound. A good compressor can mellow out the sound of a sharp guitar.

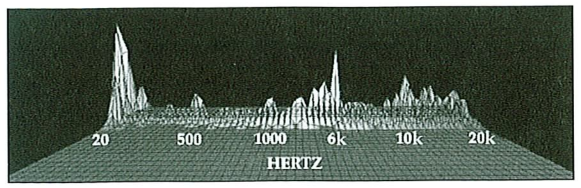
More Sustain

A compressor/limiter is also used to create more "sustain." This is commonly used on a guitar sound. Just as a compressor is used to turn down the volume peaks to raise a sound above the tape noise, it can also be used to turn down the louder parts of a guitar sound, so the guitar can be raised above the rest of the mix. Sustain is also especially helpful for obtaining feedback (when the guitar is held directly in front of a guitar amp).

Compressors are sometimes used in the same way to create more sustain on tom and cymbal sounds. The sounds seem to last longer before they fade out or are absorbed into the mix. However, the trade-off is that compressing toms and cymbals will bring their level down, so that you actually hear the bleed more. However, depending on your musical values and the project you're working on, you may want to give this a try.

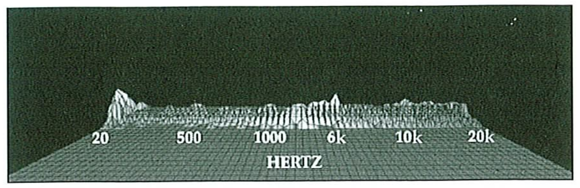
Less Resonance

A final function of a compressor/limiter is that it evens out resonances in a sound. Resonances occur in two places in instruments: hollow spaces and materials. When a hollow space (like the body of an acoustic guitar) has two parallel walls, it will boost the volume of particular resonant frequencies. Materials (like the neck of a bass guitar) will also resonate at certain frequencies, boosting the volume of those frequencies.



Visual 57. Actual EQ Curve of Resonance (Silicon Graphics "AMESH" Spectrum Analysis)

Therefore, certain notes on the instrument will actually sound louder than others. A compressor/limiter evens out the volume of these resonances by turning down the loudest part of a sound, which just happens to be the resonances.



Visual 58. Resonance Flattened Out

This is why compressor/limiters are so commonly used on resonant instruments like bass guitar, acoustic guitar, and voice.

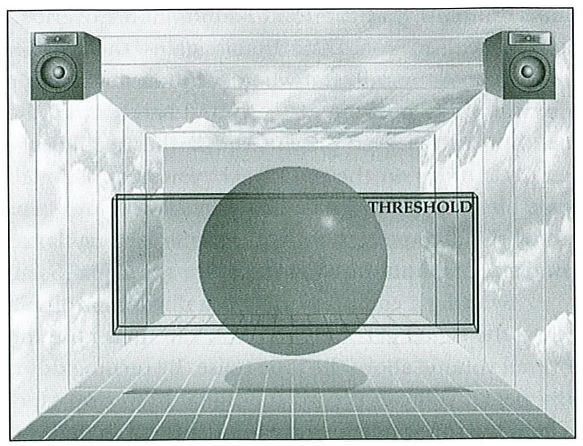
Compressor/Limiters: How to Set Them

Most compressor/limiters have two main controls, commonly known as the threshold knob and the ratio knob. On some units the threshold is called "trigger gain," "input," or "compression."

Ratio Settings

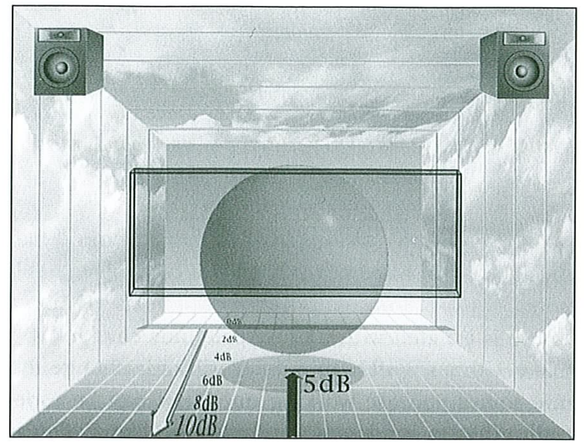
The ratio settings control how much (by percentage) the sound volume will be turned down when it goes above the threshold. For example, if a sound is 10dB above the threshold and the ratio is set to 2:1, it will be turned down 5dB. If a sound is 30dB above the threshold, it will be turned down 15dB. Ratio settings normally range from 2:1 to ∞:1 (infinity to one).

Visuals are especially effective in explaining the functions of the threshold and ratio knobs on compressor/limiters. If volume is shown as a function of front to back, the sphere will bounce back and forth based on the VU meter. It will then come out front and slam into the threshold.



Visual 59. Sound Smashing Into Threshold on Compressor/Limiter (see color Visual 59C)

The difference between a limiter and a compressor is that a limiter stops the volume from getting any louder than the threshold. The problem is that when a sound is steadily rising in volume then suddenly stops cold at the threshold, it doesn't sound natural to our ear. It sounds squashed. A compressor on the other hand, allows the volume to get a bit louder than the threshold based on a ratio, or percentage. If we set the ratio to 2:1, it will go this far:



Visual 60. 2:1 Ratio on Compressor/Limiter

A good starting point is the ratio of 4:1; this will still turn the volume down, but won't squash it. You can set the ratio wherever you like, but most people just starting out can't hear the difference between ratio settings very well. Until you can, 4:1 is a good place to start.

Threshold Settings

As the threshold is lowered on a compressor/limiter, the volume, or gain, of the sound is reduced. The compressor/limiter meters or LEDs labeled "gain reduction" will then bounce backward, showing the exact amount of volume reduction at each moment.

When adjusting the threshold, don't look at the threshold knob; rather, watch the gain reduction meters, because the threshold directly affects the amount of gain reduction. Turn the threshold knob until you get a maximum of 6dB gain reduction. If you set the threshold lower so you get more gain reduction, it will sound like it is squashed.

However, for some instruments, like lead guitar, percussion, or extremely dynamic screamer type vocals, the threshold is commonly set to provide a maximum of 10dB of gain reduction. Background vocals are also commonly compressed at 10dB max.

Again, once you can hear the nuances of various compression settings, you can set ratio and threshold the way you want for the style of music, the song, and the sound itself. Until then, try setting the ratio at 4:1 and the threshold for 6dB of gain reduction.

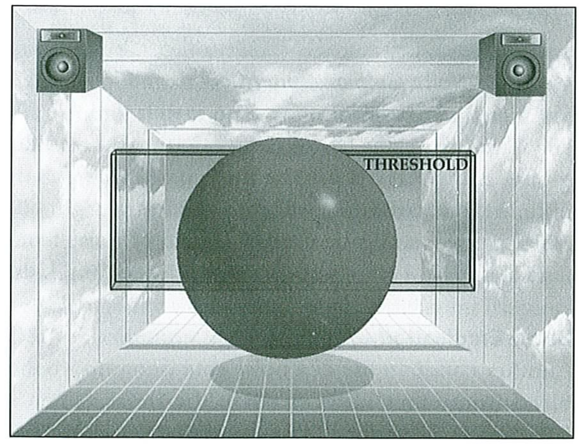
How Much Compression?

There are two main things (and several other minor ones) that determine how much you compress. The first thing is that the more instruments and the more notes you have in a mix, the more you generally compress because otherwise the mix gets too chaotic and busy. The second determining factor is the style of music; certain types of music, such as pop, are commonly compressed more.

You can also use a compressor/limiter on some sounds as a special effect. Heavy compression or limiting tends to make a sound seem unusually up front—almost as if it is inside your ear.

NOISE GATES

Operating similarly to a compressor/limiter, a noise gate turns the volume down (therefore, compressor/limiters and noise gates are often packaged together in one box). The difference is that a compressor/limiter turns the volume down above the threshold, while a noise gate drops the volume when the volume falls below the threshold.



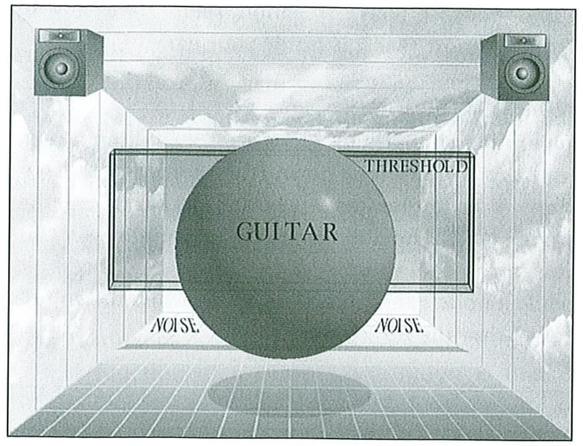
Visual 61. Sound Fading Out Past Threshold on Noise Gate (see color Visual 61C)

Noise gates have three main functions: to get rid of noise, to get rid of bleed, and to shorten the duration of a sound.

Noise Eradication

The first function of a noise gate is to get rid of noise, hiss, or anything annoying that is low in volume. However, noise gates only get rid of background noises when a sound is not playing. Noise gates don't get rid of noises while the main signal is present; however, you normally can't hear the noise when the sound is playing.

For example, one function of a noise gate is to get rid of amp noise when a guitar is not playing. Say you have a guitar amp set on "11" with lots of distortion. When not playing, the amp makes this huge "cushhhhhh" sound (when the guitar is playing, you don't hear the amp noise because the guitar is so incredibly loud). You set the noise gate by having the guitar player hit a note and sustain that note until it fades. Then the noise of the amp takes over. The threshold of the noise gate is set so as soon as the volume fades enough to hear the amp noise, it's cut off. This way, the amp noise is cut off whenever the guitar player is not playing.

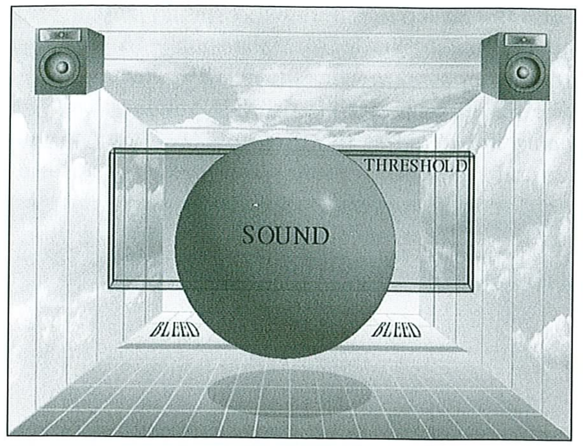


Visual 62. Noise Gate on Guitar Sound

It is important not to chop off any of the guitar sound. All it takes is for the musician to play a soft note, and the noise gate will chop the sound right off. Noise gates can also be used to get rid of noise from tape hiss, cheap effects units, dogs, crickets, and kids.

Bleed Eradication

Another common use of a noise gate is to remove the bleed from other instruments in the room. When a mic is on an instrument, the sound of that instrument will be loudest in the microphone. Therefore, it is easy to set the threshold of a noise gate between the sound and the bleed, so that the bleed gets turned off.



Visual 63. Noise Gate: Threshold Set Between Sound and Bleed

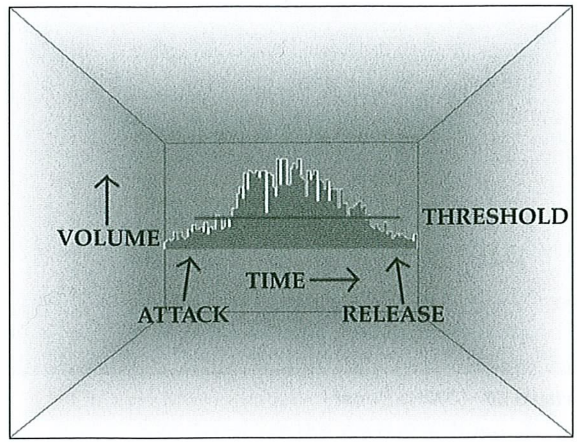
The obvious advantage of isolating a sound like this is that you have more individual control over volume, equalization, panning, and effects. Once a sound is isolated with a noise gate, any changes you make with a sound manipulator will only change the one sound you are working on. Gates can be especially effective on drums to isolate each drum. This is especially important on a snare when you have a lot of reverb. Without the gate, you end up with reverb on the hi-hat as well. Another advantage of isolation is that it helps to eliminate phase cancelation (we'll discuss this more later).

But most importantly, by removing the bleed, you will then hear the sound in only one microphone. This has the effect of putting the instrument in one precise spot between the speakers, instead of being spread in stereo. For example, consider the miking of a hi-hat cymbal. Besides being picked up by the hi-hat mic, the hi-hat is also being picked up by the snare drum mic. If the hi-hat mic is panned to one side and the snare mic (with the hi-hat bleed) is panned to the center, the hi-hat then appears to be spread in stereo between the speakers. It is no longer clear and distinct at a single spot in the mix. A noise gate can be used on the snare mic to get rid of the hi-hat bleed. The isolated image of the hi-hat will now appear to be crystal clear and precisely defined wherever the hi-hat mic is placed in the mix.

It is true that sometimes a stereo effect is desirable on a sound. However, normally you would not use the bleed from a second microphone on another instrument; instead you would put two mics on the same sound. Even so, it is rare that individual drum sounds are spread in stereo. Staccato sounds are just too bulky when recorded or mixed in stereo.

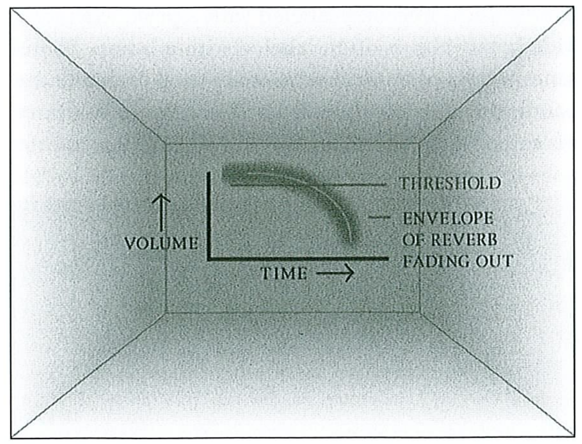
Shortening the Duration

You can also use a noise gate to shorten the duration of a sound. The noise gate will cut off both the attack and release of a sound because these are commonly the softest parts of the sound. This can be quite an unusual effect.



Visual 64. Noise Gate Cutting Off Attack and Release of Sound

A noise gate can also be put on reverb to chop off the release, resulting in the well-known effect referred to as "gated reverb."



Visual 65. Envelope (Change in Volume Over Time) of Gated Reverb

Visually, when volume is shown as front to back and the volume is less than the threshold setting, the sound will disappear. If the low volume sound is noise, bleed, or the attack and release of a sound, it gets cut off.

SECTION B

Equalizers

EQ is a change in the volume of a particular frequency of a sound, similar to the bass and treble tone controls on a stereo. It is one of the least understood aspects of recording and mixing probably because there is such a large number of frequencies—from 20 to 20,000Hz. The real difficulty comes from the fact that boosting or cutting the volume of any one of these frequencies depends on the structure of the sound itself: Each one is different. But even more complex is the fact that different sounds are equalized differently depending on the type of music, the song, and even the people you are working with.

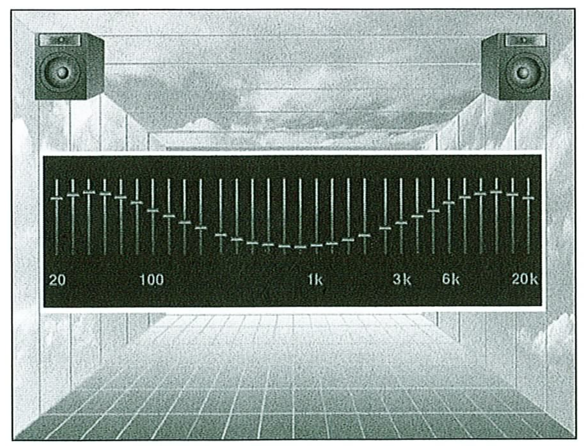
First you must learn all the frequencies or pitches by name. Then, you will see how boosting or cutting a certain frequency affects different instruments in different ways.

Types of Equalizers

There are three main types of equalizers found in the recording studio: graphics, parametrics, and roll-offs (highpass and lowpass filters).

Graphics

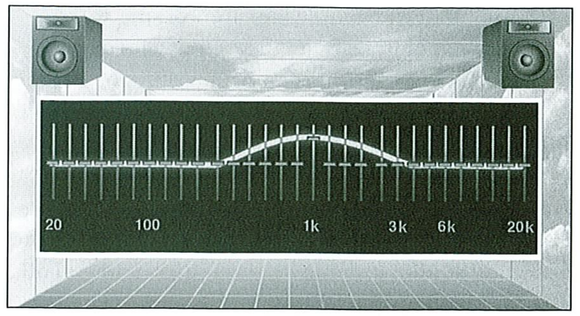
Each frequency can be turned up or down by using the volume sliders on a graphic equalizer. There are different kinds of graphic equalizers that can divide frequencies from five bands up to thirty-one bands. Five-band graphic equalizers are commonly found in car stereos (I have a 7-band in my car—at least, the last time I checked). Thirty-one band graphics (which will change the volume at thirty-one different frequencies) are common in recording studios and live sound reinforcement.



Visual 66. 31-Band Graphic EQ

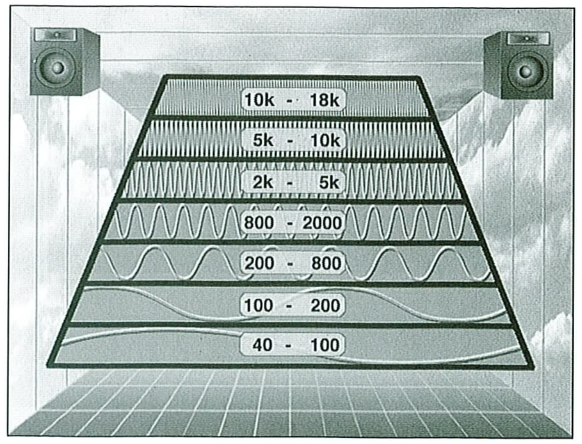
The primary advantage of a graphic equalizer is that you can make changes in volume at a number of different frequencies. Another advantage is the visual display that's easy to read for reference. (In fact, you can instantly tell what type of music someone is into by the curve of their graphic EQ.) Also, since the frequencies are mapped out visually from left to right, it is easy to find and manipulate the volume of any particular frequency.

Many people don't realize that when you turn up a particular frequency on a graphic, you are actually turning up a range of frequencies preset by the manufacturer. For example, if you turn up 1000Hz, you are actually turning up a frequency range from around 300 to 5000Hz.



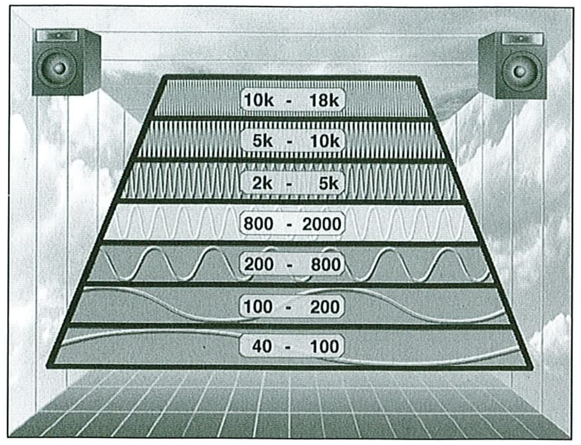
Visual 67. Wide Bandwidth on Graphic EQ

This range of frequencies is called the bandwidth and is preset by the manufacturer. You have no control over the bandwidth on a graphic. Generally, the more bands (or volume controls) there are, the thinner the bandwidth. Therefore, a 31-band graphic EQ will have a more precise frequency range for each slider than a 5-band graphic. If you turn up the volume of 1000Hz on a 5-band graphic, you could be turning up from 100 to 10,000Hz. Visually, frequency is shown as a function of up and down, so highs to lows are shown in a graphic representation.



Visual 68. Virtual Mixer Graphic EQ

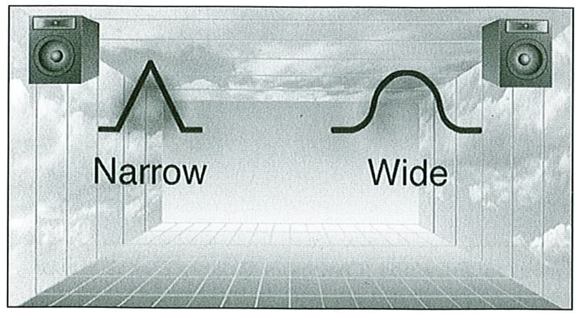
The volume of a particular frequency is shown as the brightness in that band. For example, if you turned up the highs around 1000Hz, you would see it get brighter in that frequency range, like this:



Visual 69. 1000Hz Boost

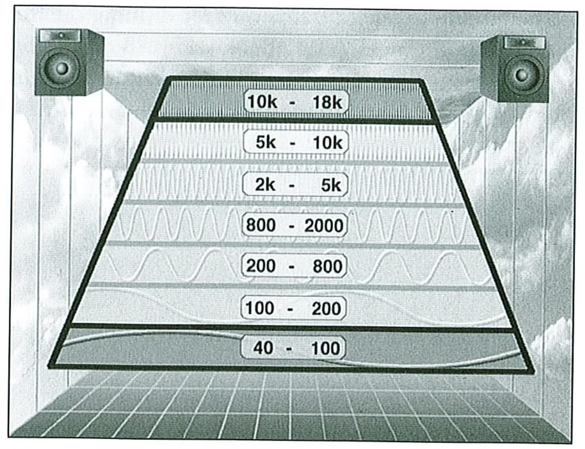
Parametrics

Engineers want to be able to control the range of frequencies, or bandwidth, they are turning up or down. With a parametric, the bandwidth knob gives you control over the width of the frequency range being manipulated. The knob is usually called "Q" because the word "bandwidth" won't fit on the knob. A thin bandwidth is normally labeled with a peak, whereas a wide bandwidth is often labeled with a hump. Sometimes ranges of musical octaves are used to define the bandwidth; for example, from .3-octaves to 3-octaves wide.



Visual 70. Wide and Narrow Bandwidths on Parametric EQ

Using visuals, the bandwidth can be shown with narrower or wider bands of color.



Visual 71. Wide Bandwidth of Frequencies Boosted

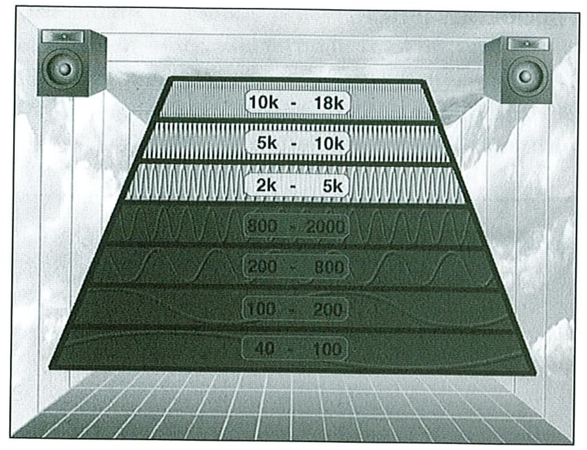
On a graphic equalizer, you select the frequency by placing your hand on the correct volume slider. On a parametric EQ, you select the frequency by turning the "frequency sweep" knob with two fingers. A separate volume knob is then used to turn the chosen frequency up or down.

Paragraphics

Many consoles have equalizers with frequency sweep knobs but do not have bandwidth knobs. This type of equalizer is commonly referred to as sweepable, semi-parametric, quasi-parametric, or paragraphic. Be careful, though, these days some manufacturers and certain salespeople are now using the term "parametric" to refer to a paragraphic or semi-parametric even though it has no bandwidth control.

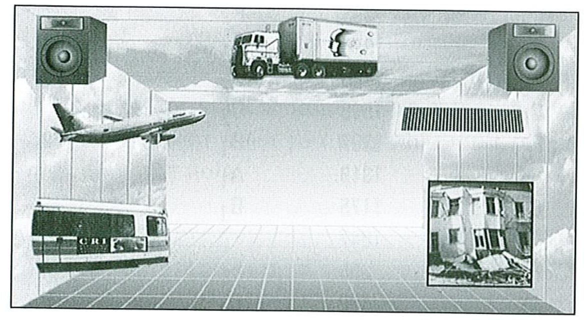
Roll-offs

A roll-off EQ rolls off low or high frequencies. They are commonly found on consoles as highpass and lowpass filters. Larger consoles often have sweepable or variable roll-off knobs, so that more of the lows or highs are rolled off. Smaller consoles often have only a button that rolls off a preset amount of lows or highs. A highpass filter rolls off the low frequencies but does nothing to the highs; it passes them.



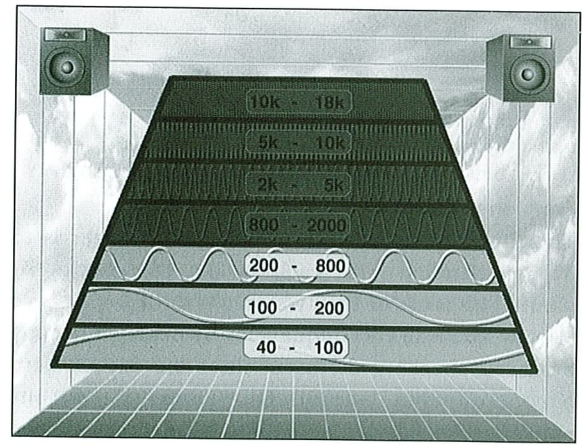
Visual 72. Highpass (Low-Cut) Filter

Highpass filters are especially helpful in getting rid of low-frequency sounds, such as trains, planes, trucks, air conditioners, earthquakes, bleed from bass guitars or kick drums, and serious foot stomping.



Visual 73. Things That Rumble

Highpass filters can be found on microphones and smaller mixing consoles as switches that simply roll off the lows when the switch is engaged.



Visual 74. Lowpass (High-Cut) Filter

A lowpass filter rolls off the high frequencies and is especially helpful in getting rid of hiss, as on a bass guitar.

FREQUENCY (PITCH)

NOTE: The difference between frequency and pitch is that frequencies are labeled with numbers and pitches are labeled with letters.

PITCH	FREQUENCY	PITCH	FREQUENCY
B_8	7902	E_4	330
A_8	7040	D_4	294
G_8	6272	C_4	262
F_8	5588	B_3	247
E_8	5274	A_3	220
D_8	4699	G_3	196
C_8	4186	F_3	175
B_7	3951	E_3	165
A_7	3520	D_3	147
G_7	3136	C_3	131
F_7	2794	B_2	123
E_7	2637	A_2	110
D_7	2349	G_2	98
C_7	2093	F_2	87
B_6	1976	E_2	82
A_6	1760	D_2	73
G_6	1568	C_2	65
F_6	1397	B_1	62
E_6	1319	A_1	55
D_6	1175	G_1	49
C_6	1047	F_1	44
B_5	988	E_1	41
A_5	880	D_1	37
G_5	784	C_1	33
F_5	698	B_0	31
E_5	659	A_0	28
D_5	587	G_0	25
C_5	523	F_0	22
B_4	494	E_0	21
A_4	440	D_0	18
G_4	392	C_0	16
F_4	349		

Chart 2. Frequencies Corresponding to Pitches

Frequency Ranges

The first step in learning to use an equalizer is to become familiar with each of the frequencies by name. This is easier than you might think because we already know all the frequencies by heart. Our body has been checking them out since the day we were born (and even before). Our entire system, our entire psyche, was designed to perceive sound. We are all professional listeners with years of experience at differentiating between frequencies.

When you learn the names of the frequencies, you can begin to remember what boosting or cutting each frequency does to each instrument. In order to make all the frequencies in the spectrum easier to remember, they can be divided into six ranges. There is no commonly accepted framework for dividing the frequencies into ranges (every book seems to divide the frequencies a little differently), so I've divided them here in the way found in most books on the subject.

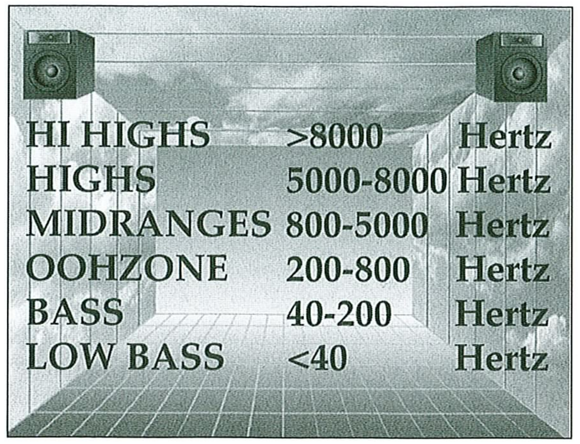


Chart 3. 6 Frequency Ranges

Low Bass: Less Than 40Hz

This range, sometimes called the "sub-bass," is commonly found in rap booms and the low bass of kick drums and bass guitars. Although it is difficult for many people to discern pitch easily at this range, it is often used in movies for earthquakes, rumbles, and explosions.

A normal vinyl LP record has about twenty-three minutes per side, usually the length of five songs. Because the grooves on a record must be wider for bass frequencies than high frequencies, you couldn't get twenty-three minutes on a record with a lot of low bass information without rolling off everything below 40Hz. This is also why you can put more low bass on a 12-inch single record. Of course, this is no longer an issue with CDs.

Bass: 40 to 200Hz

This is the approximate range boosted when you turn up the bass tone control on a stereo.

Oohzone: 200 to 800Hz

When frequencies are boosted too much in this range, they sound extremely muddy and unclear and can even cause extreme fatigue when not evened out. You'll also find that everybody in the room starts getting a bit irritated, and you'll hear people say things like, "Just mix the damn thing; I'm sick of this song anyway."

Midranges: 800 to 5000Hz

We are extremely hypersensitive to this frequency. Boosting a frequency 1dB in this range is like boosting 3dB in any other frequency range. You see, this is where we live most of the time. This is where most of our language is centered. In fact, the telephone is centered around 3000Hz because we can still understand someone when only this range is present. It is critical to be careful when boosting or cutting any frequencies here. This is doubly true on vocals because we are also hypersensitive to what vocals are supposed to sound like.

Other notable frequencies in this range include 1000Hz, which is the frequency of TV stations' test tones when they go off the air. The chainsaw frequency is 4000Hz, and it is the most irritating frequency there is, by far. It is also the frequency of fingernails on a chalkboard—eeeek!

Highs: 5000 to 8000Hz

This range, the one boosted when you turn up the treble tone control on a stereo, is commonly boosted in mastering to make things sound brighter and more present.

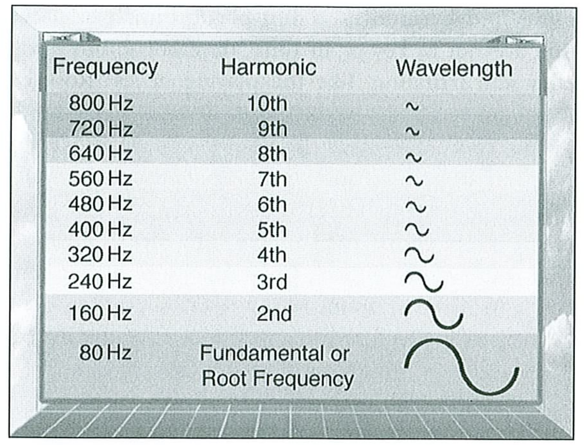
Hi-Highs: More Than 8000Hz

This is where you find cymbals and higher harmonics of sounds. Boosting this range a little on certain instruments can make the recording sound like a higher quality recording, but too much can make it irritating. By the way, that extremely high frequency that old televisions emit is 15,700Hz.

The Complexities of Frequencies: The Harmonic Structure of Sound

Specialists don't agree on how different frequencies affect our psyche, which is quite understandable because of the subjective nature of frequency perception. Psychologists and philosophers have written books on how sounds affect our mind and body and how people organize the ways they perceive frequency differences. Different frequencies affect us differently: physiologically, psychologically, and spiritually. *And even more powerful than the way specific frequencies make us feel is the way the combinations of frequencies make us feel.*

Just about every sound is made up of a combination of tones, or notes. When you hear an instrument play a particular pitch, you are hearing many other notes hidden in that sound. These other notes are called harmonics or overtones. Sounds are combinations of different harmonics.



Visual 75. Pitches in Harmonic Structure for Note "A"

For example, here is the harmonic structure of the note "A" as on an acoustic guitar. Just look at all the notes present when you play what most people think of as one note:

Frequency	Harmonic	Pitch
4400	10th	C#
3960	9th	B
3520	8th	A (3rd Octave)
3080	7th	G 1/4 flat
2640	6th	E
2200	5th	C#
1760	4th	A (2nd Octave)
1230	3rd	E
880	2nd	A
440	Fundamental or Root Frequency	A

Visual 76. Harmonic Structure of Note "A" As on Guitar

It is the particular harmonics present in a certain sound that account for the differences in sound qualities, or timbres. The term timbre refers to different sounds, such as guitar vs. piano or vocal vs. accordion, as well as the differences in the sound quality of particular instruments. For example: the difference between a Martin and a Gibson guitar.

There are two interesting things about harmonics. First, each harmonic found in a sound's timbre is a pure tone. A pure tone is the sound of a tuning fork or tone generator. It has no harmonics at all. The most amazing thing is that just about all sounds are made up of a combination of these pure tones. This means that even a screaming electric guitar sound is made of many pure tones.

So, how do you get an edgy sound from a bunch of pure tones? Well, certain combinations of harmonics will create a dissonant chord. These are the odd numbered harmonics. If you play a bunch of notes that are not in key or in tune, they will sound quite edgy and irritating, like the playing of Axl Rose or Tiny Tim (may he rest in peace). On the other hand, certain combinations of harmonics will create a chord that sounds good. These are the even-numbered harmonics. If the pitches of the harmonics combine to create a nice chord, the sound will be nice and round (like the performing of Chris Isaak or most opera singers). Whether an instrument puts out odd or even harmonics is based on the construction of the instrument and how the sound is produced.

The second interesting thing about harmonics is that they're all mathematical multiples of the root, or fundamental, frequency. The root frequency is the basic pitch we perceive when we hear a sound. For example, when we play an "A" on the guitar, even though there are numerous pitches or harmonics present in the sound, we still hear it as one pitch: "A," which is the root frequency.

Therefore, when we raise or lower the volume of a certain frequency with equalization, we are actually raising or lowering the volume of a particular harmonic in the sound. Because every sound has its own harmonic structure, every instrument sound responds to equalization differently.

USING EQUALIZERS

When to Equalize

There are five times when you might equalize a sound in a recording session. First, a sound is equalized individually while in solo when recording onto the multitrack. Second, while the entire band is rehearsing or running through the song, you doublecheck the EQ of each sound relative to all the other sounds. Then during mixdown, each sound can be equalized individually before building the mix. Most importantly, finishing touches are done on the sound's EQ (relative to all the other sounds) when listening to the whole mix at once. Finally, a bit of EQ is occasionally done during the mastering process. This is an overall EQ for the entire mix and is not necessary if a good job was done in the first place.

Equalizing in Solo When Recording Onto the Multitrack

The first step in the recording process is to equalize each sound individually. Most engineers start with the drums.

There used to be a school of thought that said you should not EQ a sound going to the multitrack. This idea was the result of inexperienced engineers screwing up the EQ on the way to the multitrack. At this point, it is very difficult to get a sound back to normal and still be able to make it sound great during mixdown. Therefore, it is important that you EQ the sound correctly onto the multitrack in the first place. The school of thought these days, though, is to definitely EQ on the way to the multitrack. In fact, professional engineers will usually try to get everything to sound like a CD on the way to the multitrack. There are some very important advantages for doing this.

First, it is much better to boost the highs on the way to the multitrack because if you boost them during mixdown you are boosting the hiss from the tape.

Secondly, the sooner you get the EQ in the ballpark, the sooner you can play ball. It is much better when you get to the final mix and the EQ already sounds close to perfect. Instead of spending all your time trying to get the mix to sound normal, you can spend your time refining extremely subtle aspects of the EQ that bring out the finer, magical aspects of the sound.

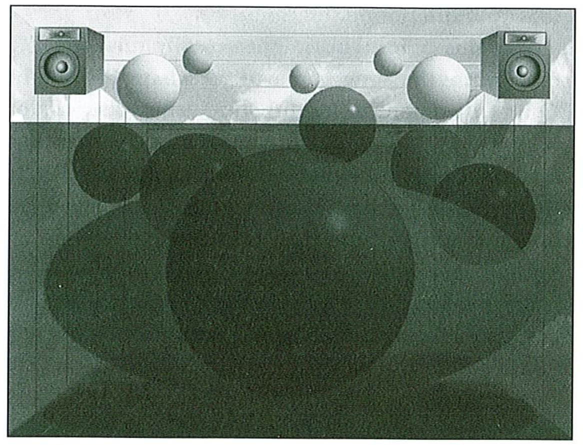
These days, most bands, especially those who have worked in major studios, expect you to get it sounding as close to a CD as possible on the multitrack. So, if you get the project sounding like a CD on the way to the multitrack, then during overdubs, everybody is thrilled with how good it sounds. And the neurons of creativity are firing everywhere because it sounds so incredible. A great mix turns people on creatively. If you don't make it sound good, you'll hear the engineer say things like, "Don't worry guys, I'll fix it in the mix." It is especially important to get things sounding great on the multitrack because overdubs can take months to complete.

The professional engineer gets to the point where he or she can guess what a sound should be alone in order for it to sound right in the mix. To do this, you must visualize what the final mix will sound like and then extrapolate how the sound should sound in solo. However, unless you have heard the band previously, you don't know what the final mix might be like. Most engineers will EQ sounds so that they sound "good" (natural or interesting) individually. The problem is that good is different for different types of music, songs, and people. However, commonly accepted values are to make sure the sound is not too muddy, too irritating, or too dull.

Equalizing in the Mix When Recording Onto the Multitrack

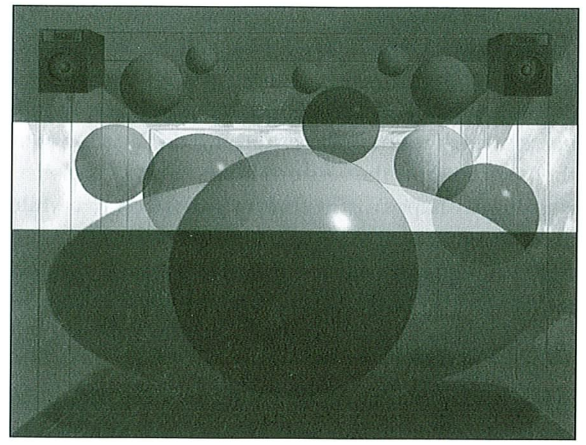
The next time to check the EQ of each sound is when the entire band starts rehearsing the song. At this point, you want to check the EQ of each instrument relative to every other instrument in the mix. You can make sounds more similar to each other or more dissimilar. You can make a lead instrument more cutting and abrasive, to really grab attention. You can give extra bass to a particular instrument to make the song more danceable or to excite the listener.

To make the process a bit easier, follow this procedure: First, scan the high frequencies and check the relative brightness of all the sounds in the song. Make sure all of them are as bright as you want them. They should have a similar amount of brightness, but sometimes you might want some sounds to be brighter or duller than others.



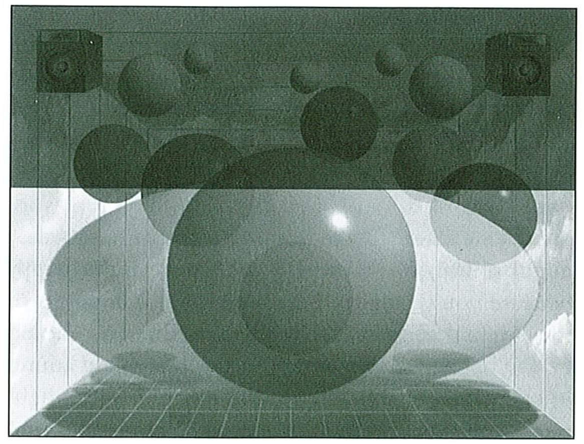
Visual 77. Song With High Frequencies Highlighted

Second, scan the midrange frequencies, checking for the relative volume of these frequencies across all the instruments. Midrange frequencies seem to stick out when boosted too much. Make sure that all instruments have the exact amount of midrange frequencies that you want. The sounds might have a similar amount of midrange frequencies, but sometimes you may want some sounds to stick out more so they grab your attention.



Visual 78. Song With Midrange Frequencies Highlighted

Third, scan the bass frequencies, checking for the relative volume of bass in each sound in the bass range. For example, check the relative amount of bass frequencies present in the kick drum compared to the bass guitar. Listen and make sure that it is the way you want it to be. This frequency range is the one most commonly missed when mixing an album or project.



Visual 79. Song With Low Frequencies Highlighted

It is critical that you check the relative EQ of each instrument in the mix at each frequency range. The amount of time you spend doing this often depends on the band. Some bands expect to be recording within a few hours of the time they arrive and have very little patience (or money). Other bands spend weeks getting the right sound and EQ before recording to the multitrack. It is a good idea to set up the band the day before the session, get all the sounds EQ'd, get a headphone mix, then go home. The next day everything is set to go and everyone is fresh.

It is also a good idea to talk with the band beforehand and let them know that you will be spending a good amount of time working on the sounds at the beginning of the session. If the band knows what is going to happen, they won't get frustrated while waiting to actually begin recording. They should appreciate the fact that you want it to sound as good as possible.

Equalizing in Solo During Mixdown

When you go to mixdown a song, the first step is to EQ each of the sounds individually. If you did your job well during the recording session (that is, if you had time), you might have little or no EQ'ing to do. "If it ain't broke, don't fix it." However, often you will need to EQ the sound again because you know what the band is going for and you have a new perspective —a fresh ear. You also have a major advantage that you didn't have when you began the recording session: You know what the whole song sounds like with all of the instruments playing together. Now you can set the EQ of each sound with the final mix in mind.

People often wonder why things don't seem to sound the same when you come back to do the mixdown. First, when using analog tape (as opposed to digital), you lose highs every time the tape is played back on the multitrack. After a couple of weeks of overdubs, the highs are dampened drastically.

Second, it is easy to think that you have something sounding right simply because you have made it sound so much better. When a band first comes in, often you listen to the sounds (and how bad they might sound), then you EQ them and you're happy because you've made the sound great compared to the original sound. The problem is that you should be EQ'ing the sound based on the real world sound of current CDs. You might have made it sound light years better, but it needed more EQ'ing . . . to sound like a CD. When you come back in to mix the song down, you have been listening to the radio or CDs in the real world. When you put on the multitrack, you automatically compare your recording to the real world and realize it didn't sound as good as you thought when you did the recording session.

Also, especially with less expensive mixers, you don't have enough bands of EQ, so you can't do all of the equalization required while recording the band. In this case, the EQ must be completed during mixdown. This is often the case when you don't have a full parametric EQ.

Equalizing in the Mix During Mixdown

The way the EQ sounds in the mix during the mixdown is the true test. Again, you should check all the sounds relative to each other at each frequency range: highs, midranges, and lows. If it is already pretty good, you can now work on fine-tuning the relative EQs. This is where you do the magical stuff. For example, you might add a tiny bit of 12,000Hz on some of the high-frequency sounds to make the overall mix sparkle. Or you might consider making a guitar solo track a little brighter and edgier, so that it grabs your attention.

At this point, you might actually turn the EQ knobs while the song is playing. Perhaps you might EQ an instrument differently for various sections of the song. Or, to really flip people out, you could change the EQ in the middle of a section of a song. It is also interesting to EQ a sound so that it seems to be coming out of a telephone.

Equalizing the Entire Mix During Mastering

There are two main types of EQ done during mastering. First, minor repairs can be done if the overall EQ doesn't sound quite right, but if the problem is very bad the entire song must be remixed. It is quite common to adjust the amount of overall bass or treble slightly. Second, the overall EQ can be adjusted to make the overall bass, midrange, and treble more similar from song to song. Again, if the difference is too great, it might require remixing the songs. There is only so much that can be done in mastering with EQ since all the sounds are no longer separate on their own tracks.

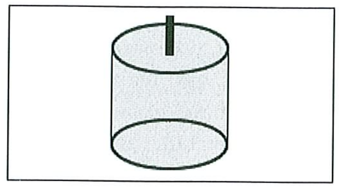
USING AN EQUALIZER: STEP-BY-STEP INSTRUCTIONS

When you approach a mixing board equalizer for the first time, it is a good idea to play with it to get to know how it works and what it does. However, when working on a project with other people, you must be quick at getting things to sound great. The following procedure will help you become more efficient whenever you use an equalizer.

A COMMON, STEP-BY-STEP PROCESS FOR EQ'ING A SOUND

1. Reset to "0"

Reset the volume controls on the equalizer to "0." This usually means setting the volume knob straight up (not all the way counterclockwise). At this position you are neither boosting nor cutting the volume of any frequency.



Visual 80. EQ Knob With Volume Set to "0"

Even if the EQ has an on/off switch, the volume knob should still be set to "0," so when the EQ is turned on, it makes no changes and you are starting from "0," not some unknown preset. In many professional studios, if you don't reset, or normal, the EQ at the end of your session, you will hear from the management later.

2. Listen

The most common mistake made by an inexperienced engineer is to begin turning the EQ knobs before listening. Don't touch the knobs until you know what you want to do. Listen to see if anything is wrong with the sound first, and if it ain't broke, don't fix it.

There are many fine details you should listen for when EQ'ing. The three main things to check are if the sound is muddy, has irritating or "honky" frequencies in the midrange, or is bright enough. If you do nothing else while recording to the multitrack, you should at least take care of each of these aspects, which make up over 75% of all EQ'ing to be done.

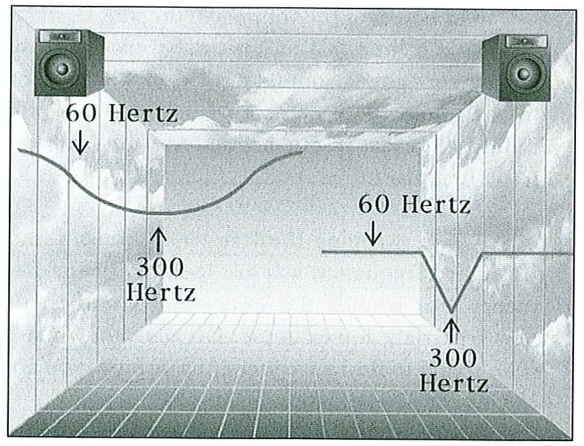
a) Cut Muddiness (100-800Hz): Check each instrument for muddiness. Kick drums almost always need to have the muddiness cut (unless it is a rap or hip hop kick). Other potentially muddy instruments include toms, bass guitar, piano, acoustic guitar, and harp. Muddiness is normally around 300Hz. If you cut the muddiness too much, the instrument will sound thin because this mud also contributes to the body of most sounds. When cutting muddy frequencies, always make sure that you haven't lost your bottom: the low lows. You might compensate by boosting the lows around 40-60Hz. Using a parametric to zone in on the specific muddy frequency will help to preserve the bottom of the overall sound.

b) Cut Irritation (1000-5000Hz): Cut any excessively irritating or honky frequencies occurring in the midrange from 1000-5000Hz. Vocals, electric guitars, and cymbals (including hi-hat) often need frequencies cut in the midrange. Depending on the type of music (and the particular snare drum used), snares sometimes need this edge cut also. The best way to detect an irritating frequency is to turn the entire sound up very loud. If you and the people in the room are cringing on the floor, then it's irritating. Never boost or cut the midrange too much and make sure you haven't made the sound too dull. At that point, you might compensate by boosting the highs around 5000-8000Hz. Using a parametric to zone in on the specific irritating frequency will help to preserve the brightness of the overall sound.

c) Boost Highs (5000-8000Hz): Boosting highs on instruments that sound dull, like the snare, is largely dependent on the style of music. R&B, dance, and certain types of rock 'n' roll require more crispness than other styles. Country, middle-of-the-road, and folk music do not need as much boost in this range, so they sound more natural.

3. Set the Bandwidth

a) When getting rid of muddy frequencies, set the bandwidth as thin as possible, because if you use a wide bandwidth, you might also drop the nice "bottom."



Visual 81. Wide and Thin Bandwidths on 300Hz Cut

b) When getting rid of irritating frequencies, set the bandwidth as thin as possible for very much the same reason as above. If you use a wide bandwidth on a vocal, guitar, or cymbal, you might lose the entire body of the sound in the midrange. Then the sound would be dull and not present.

c) When boosting highs, set the bandwidth to medium wide. This sounds more natural. If ever in doubt as to how to set the bandwidth, start with the thinnest one possible. Then you can try widening it out a bit to see if it does what you want and sounds better. By doing this, you end up with the center frequency where it should be.

4. Find the Frequency to be Boosted or Cut
Now that you have decided that a frequency needs to be boosted or cut, you must first find the frequency.
a) Boost the volume on the band of EQ where you think the problem is. When first starting out, it is a good idea to boost the volume all the way. Be careful, though. Boosting the volume all the way in the bass area can blow up your speakers. And boosting the volume all the way in the midrange can make you deaf. It's a good idea to keep your other hand on the channel fader or master volume control so as not to hurt anybody.
Boosting the volume all the way will help you locate the frequency you want to turn up or down. A good analogy is when you cook with new spices. Though tasting red hot chili peppers by themselves can be a bit extreme, you need to taste them before cooking with them to get an idea of what they taste like. Similarly, when you boost the EQ volume all the way, though it won't be that strong in the mix, it gives you an idea of what it might be like when added in moderation.

NOTE: You can also cut the volume all the way instead of boosting the volume all the way. Doing it this way is a little less annoying, because you are looking for "good" sounds instead of irritating or muddy sounds. However, you run the risk of not finding the exact frequency that was the problem in the first place.

b) When sweeping the frequency knob to find the culprit frequency, you are looking for the frequency that sounds the worst—the muddiest or most irritating, for example. On the other hand, when trying to find a frequency to turn up, you are looking for where it sounds the best.
If you are trying to get rid of a frequency and you have cut the volume knob all the way (instead of having boosted the volume knob all the way), sweep the frequency knob to find the spot where the sound seems to be the best.

5. Return the Volume Knob to "0"
With the volume boosted all the way, you are now in outer space. You have lost all touch with the reality of what the sound was like in the first place. Regain your perspective on the tone of the sound before it was EQ'd by returning the volume knob to "0" (on the EQ band you are working on).

6. Boost or Cut the Volume to Taste
If you are getting rid of a frequency, slowly turn the volume down as much as you think it needs. If you are boosting a frequency, same thing. Play with the volume knob until you figure out how much it needs to be boosted.

7. Check to See If You Like What You Did
Turn the EQ switch on and off, compare the EQ'd sound with the original sound, and make sure you like what you did. If you don't have an on/off switch on your EQ, quickly return the volume knob back to "0," then zip it back to the exact amount of boost or cut. This is also helpful when doing more than one EQ change on a sound. For example, say you have cut the muddiness on one band and you have brightened the highs on another band. If you turn off the EQ, you turn off both bands. Instead, simply return the volume control to "0" on the one EQ band you are working on, so you can see what that one change is doing and whether you like it or not.

So far I have provided you with an extensive overview of how to use EQ. However, it requires practical experience to get know its intricacies. For those of you who are just beginning, here is a listing of common EQ techniques for well-known instruments—although, in reality, every sound is different.

Frequency	40-100	100-200	200-800	800-1000	1000-5000	5000-8000	8000-12,000
SOUNDS							
BASS	Bottom	Roundness	Muddiness	Body on Small Speakers	Presence	High End	Hiss
KICK	Bottom	Roundness	Muddiness			High End	Hiss
SNARE	X	Fullness	Muddiness			Presence	X
TOMS		Fullness	Muddiness		Presence Irritation	High End	X
FLOOR TOMS	Bottom	Fullness	Muddiness		Presence		X
HI-HAT CYMBALS	X		Muddiness Bleed		Irritation	Clarity/ Crispness	Shimmer/ Sizzle
VOICE	Rumble	Fullness	Muddiness		Presence Irritation Telephone	Clarity/ Crispness Sibilance-6K	Sparkle/ Hiss
PIANO	Bottom	Fullness	Muddiness	Muddiness	Presence	Clarity/	Harmonics
HARP		Pedal Noise			Twanginess	Crispness	
ELECTRIC GUITAR	X	Fullness Crunch	Muddiness Roundness		Cut/Shred Irritation	Crispness Thinness	Hiss
ACOUSTIC GUITAR	X	Fullness	Muddiness			Clarity/ Crispness	Sparkle
ORGAN	Bottom	Fullness	Muddiness			Clarity/ Crispness	
STRINGS	Bottom	Fullness	Muddiness		Irritation Digital Sound	Clarity/ Crispness	Sparkle
HORNS	X	Fullness	Muddiness	Roundness		Clarity/ Crispness	
CONGA	Boominess	Fullness				Clarity/ Crispness	
HARMONICA	X	Fullness			Irritation	Clarity/ Crispness	X

Chart 4. Equalization Chart

Freq.	HH	Kick	Snare	Overheads	Toms
High Highs (10-12k)	+3			+3	
Highs (5-8k)		+5	+7		+6
Low Mids (200-400)	-9	-10		-6	-6
Lows (40-60)		+2			

Freq.	Bass	Dist Guitar	Clean Guitar	Acoustic Guitar	Piano	Vocals
High Highs (10-12k)				+4		
Highs (5-8k)			+3	+3	+3	+2
Mids (1-3k)	+5	+3				
Lo Mids (200-400)	-3			-5	-5	
Lows (40-60)	+2					

Chart 5. Common Quick General EQ

Instrument	Lows	Mids	Highs
Hi-Hat	Roll-off muddiness around 300Hz.	If irritating, find & roll-off irritating frequency	Around 12k, boost 3-6dB for sizzle.
Kick Drum	Roll-off muddiness around 300Hz.		Boost highs around 5-6k. Boosting up around 10-12k will only bring out hiss and cymbals.
Snare Drum	Add a little bit around 60-100Hz if snare sounds thin and wimpy.	Take out irritating frequency if apparent or going for sweet, smooth sounding mix	Add 3-10dB around 6k.
Toms	Cut boominess around 300Hz.		3-8dB boost around 5k. Less boost on floor tom.
Overheads	Cut any muddiness around 300Hz.	Be especially aware of any irritating freq's in midrange. Cut them if apparent.	Possibly boost a little around 6k and/or 12k but be wary of making them too edgy.
Bass Guitar	Possibly boost 40-60Hz if song calls for it. Possibly cut 300Hz if bass is too muddy for song.	Boost around 1-2k if more presence is needed and if string noise is not too much.	Boost around 5k for presence if mix is sparse enough to even hear it.
Electric Guitar	Boost or cut around 300Hz depending on need.	Boost around 3k for edge. Cut 3k for transparency.	Boost around 6k for presence and clarity. Boost 10k for sparkle.
Acoustic Guitar	Cut 100-300Hz where muddiness and boominess is.	Cut 1-3k to make image higher and more transparent.	Hi's: Boost around 6k for presence and clarity. Boost 10k for sparkle.
Piano	Cut muddiness around 300Hz.	Cut any honkiness around 1k.	Boost around 6k for presence and clarity.
Vocals	Cut or boost 300Hz depending on mic, voice, and use in mix.	Listen closely for any irritating or midrange honk (telephone-like sound). Cut either.	Boost around 6k for presence and clarity.
Horns		Beware of irritating or honky midrange. Cut if necessary.	

Chart 6. Typical EQ for Typical Instruments

Common Terminology for EQ Frequencies

Even if you learn all of the frequencies, understand how boosting or cutting each frequency affects various instruments differently, and master how to EQ an instrument for different types of music and songs, the people you are working with might still be using street terminology to describe what they want. Therefore, Chart 7 is a list of slang and what it means.

SECTION C

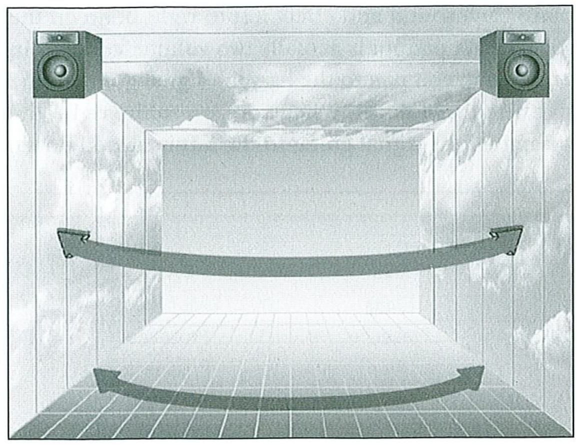
Panpots and Stereo Placement

When mixing, you use panpots (balance knobs) to place each sound and effect left to right between the speakers. A panpot is actually two volume controls in one. When you pan to the left, the signal going to the right is turned down. When you pan to the right, the volume of the signal going to the left is turned down.

40-200 Bass	200-800 Low Mids	800-5K Mids	5-8K Highs	8-12K Super Highs
Fullness Boomin' Ballsy Punchy Powerful Thumpin' Solid Thick Round Beefy Bad!!	Body Robustness Warmth Crunchy Fat	Presence Projected Forward Intelligible Articulate	Presence Airy Bright Brilliant Live Clear Smooth Crisp	Presence Crispness Sparkling Screamin' Sharp
<u>Too Much</u> Heavy Boomy Rumbly	<u>Too Much</u> Muddy Tubby Barrelly	<u>Too Much</u> Hornlike Phonelike Honky Bathroomy Boxy Woody Nasal Chunky Woofy Edgy	<u>Too Much</u> Tinny Steely Metallic Strident Cutting Piercing Shrill Screamin'	<u>Too Much</u> Crisp Sizzly Searing Glare Glassy
<u>Not Enough</u> Thin Anemic Wimpy	<u>Not Enough</u> Distant Hollow Disembodied	<u>Not Enough</u> Veiled Covered Muffled	<u>Not Enough</u> Dull Dead Dark	<u>Not Enough</u> Flat Cheap

Chart 7. Common Terminology and Slang

As previously discussed, panning in a mix is mapped out visually as a function of left to right. Panning a sound to one side or the other also seems to make the instrument a bit more distant in the mix. If the sound is panned to the center, it will seem to be a bit closer, a little more out front.



Visual 82. Left and Right Placement

If we think of the space between the speakers as a pallet on which to place instruments left to right, we are free to pan as we please. However, certain styles of music have developed their own traditions for the particular placement of each instrument left to right in the stereo field. Normally, the placement of a sound is static; it stays in the same place throughout the mix. However, the movement of a panpot during a mix creates an especially effective dynamic. We will discuss the common ways panning is used to create musical dynamics in the next chapter.

SECTION D

Time-Based Effects

DELAYS

After many failed attempts to use outdoor racquetball courts to create delays, engineers realized they could get a delay from a tape player. You could hear a delay by recording a signal on the record head, then listening to the playback head two inches later. The delay time could be set by changing the tape speed. Engineers used this technique for years with the Echoplex, which fed a piece of tape through a maze of tape heads at different distances, each giving different delay times. Not bad, but the problem with tape is that every time you record over it, you get more tape hiss.

Then came analog delays, which would put a signal through a piece of electronics to delay the signal a bit. The more you put the signal through the electronics, the longer the delay. It was a bucket brigade type of system. The only problem was that when you put a signal through a piece of electronics over and over, it gets extremely noisy after a while.

Then came digital delays, which record the signal digitally onto a chip, then use a clock to tell the unit when to play the sound back. The delayed signal can also be fed back into the input to get the well-known sound of feedback or regeneration when the signal continues to repeat.

You must learn the details of the frequency spectrum, as well as how each delay time feels and what feelings or emotions each delay time evokes. Then, when you hear a song that has a similar feeling or emotion, you will know what delay time might work. There are, of course, a wide number of other reasons for using different delay times that I will cover later.

Delay Times vs. Distance

In order to assist you in remembering what different delay times sound and feel like, it is helpful to understand the relationship between delay time and distance. Sound travels at approximately 1130 feet per second. That's around 770 miles per hour, which is extremely slow compared to the speed of sound in wires—186,000 miles per second, the speed of light. Therefore, it is easy to hear a delay between the time a sound occurs and the time it takes for a sound to travel even a few feet to a wall and back. We can also easily hear a delay when we put two microphones at two different distances from one sound. In fact, changing the distance between two microphones is exactly like changing the delay time on a digital delay.

The following chart illustrates how different distances relate to delay time. Of course, if you are calculating a delay time based on the distance between a sound source and a wall, the distance must be doubled (to and from the wall).

Feet	=	Delay (ms)
1130		1000
560		500
280		250
140		125
70		62.5
35		32.25
17.5		16.13
8.75		8. 01
4.28		4
2.14		2
1.07		1

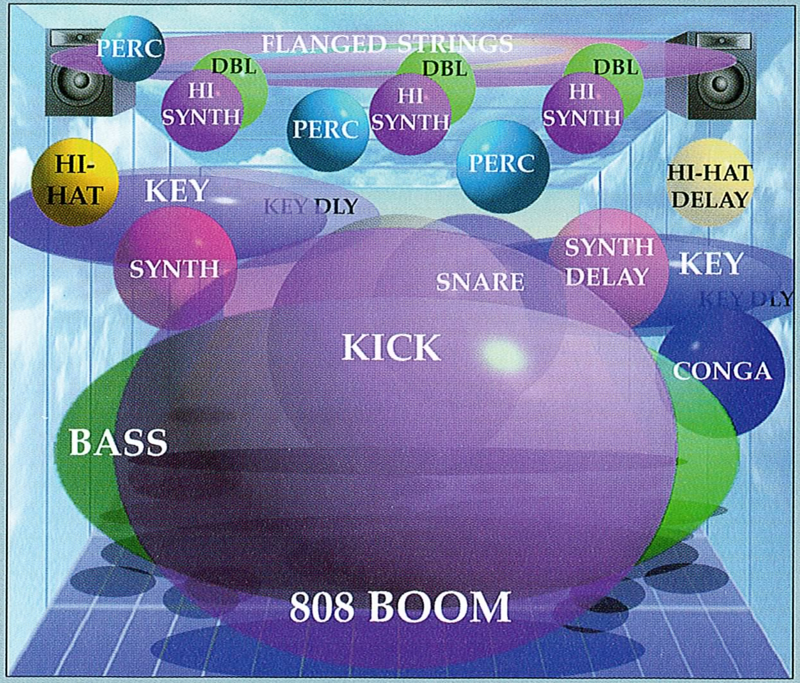
Chart 8. Distance vs. Delay Time

COLOR VISUAL SECTION

These color visuals are representations of particular moments in the mix. In order to represent a true mixing process, they would be flashing on and off to the music. Therefore, some of the visuals may look busier than the mix really is.

Of course, every song has its own personality and is mixed based on that. Therefore, don't assume that there is only one way to mix any style of music. These visuals are only a reference point from which you can begin to study what is done in mixes for various types of music.

With all this in mind . . . enjoy.

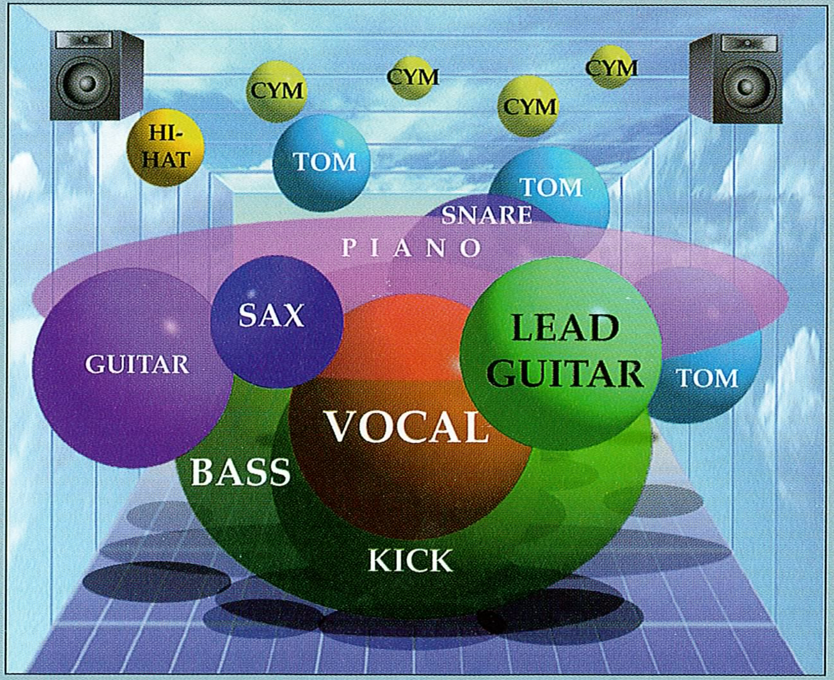


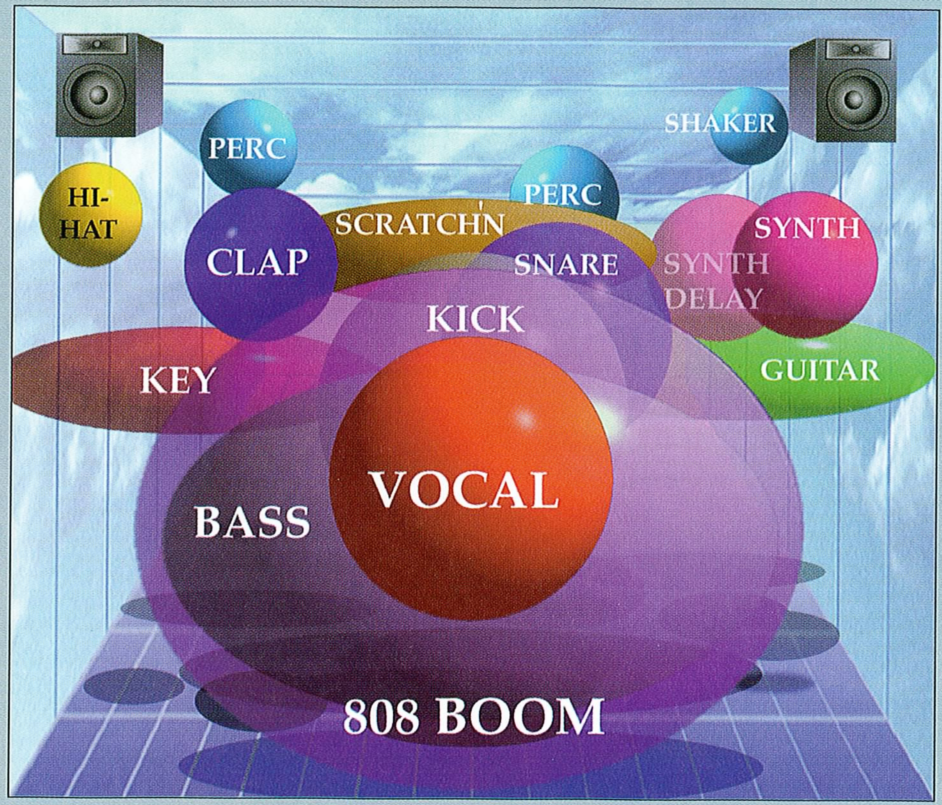
Visual A. Hip Hop Mix

Generally a pretty busy mix with an 808 boom loud and out front. Note the fattening on the bass and the doubling on each of the keys. Note the delay on the synth and hi-hat. Especially unique is the doubling of the hi-synth with another instrument. The super high strings are flanged for a subtle, spacey effect. The snare is not very loud in this particular mix.

Visual B. Blues Mix

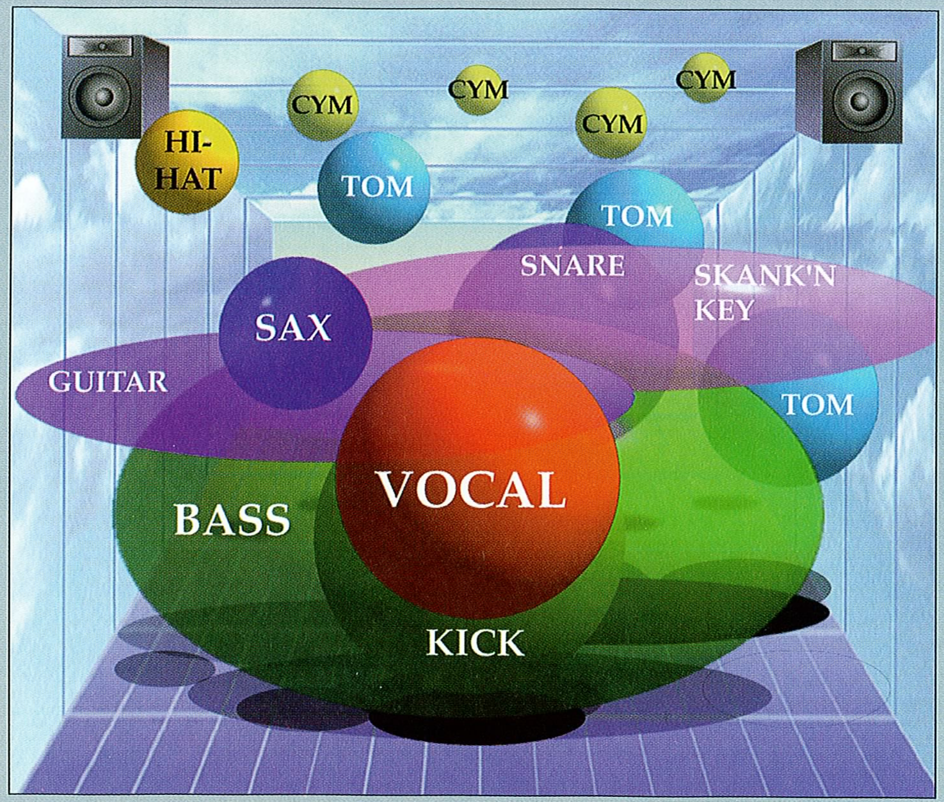
Generally a pretty clean, clear, out front mix. Note the bass is quite loud overall, with the kick drum not far behind. The rhythm guitar, the sax, and especially the lead guitar are right out front. The vocal is set back in the mix a bit, but this is not always the case. The piano is set further back but is spread in complete stereo. The toms, hi-hat, and cymbals are all set back a bit, and the snare is a bit low, which is not necessarily typical of blues.





Visual C.
Rap Mix

The rap mix commonly has the 808 boom boom'n and a loud vocal (though this does vary). The key, guitar, and scratch'n are all spread in stereo with fattening. Note the extremely loud clap and hi-hat; the kick is also right out front. In this mix, the snare is back a little. Also cool is the delay on the synth panned next to it. Finally, note the shaker panned opposite the hi-hat.

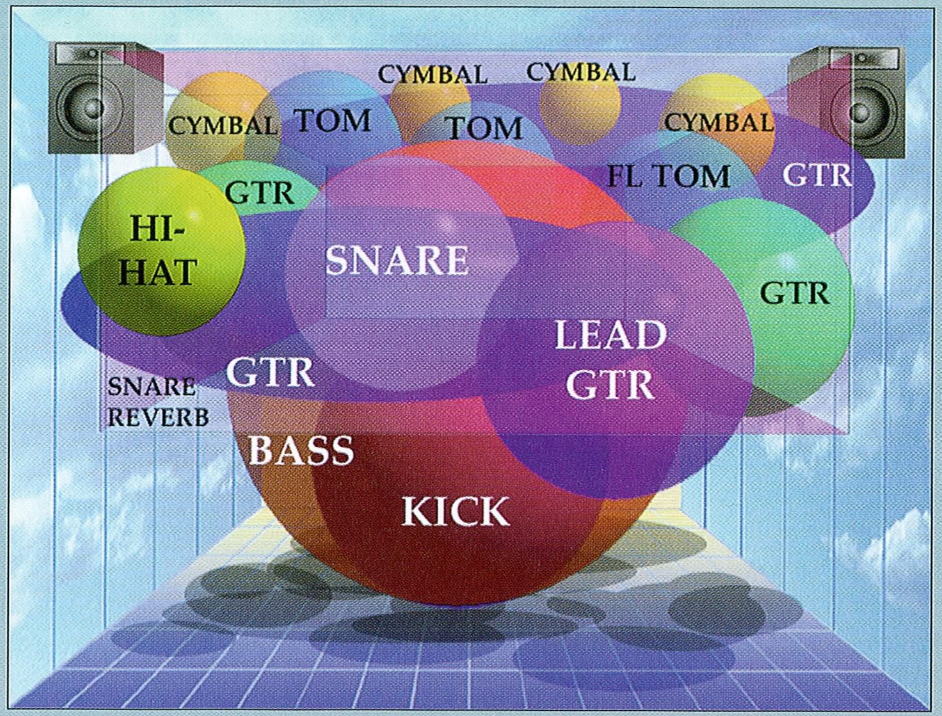


Visual D.
Reggae Mix

These days, reggae mixes tend to have a huge bass with the kick not far behind and the vocal right out front. Note the clarity of the sax. Both the guitar and the skank'n key are spread in stereo so they overlap a large amount. The snare is set back a little but not always, and the hi-hat is right out front.

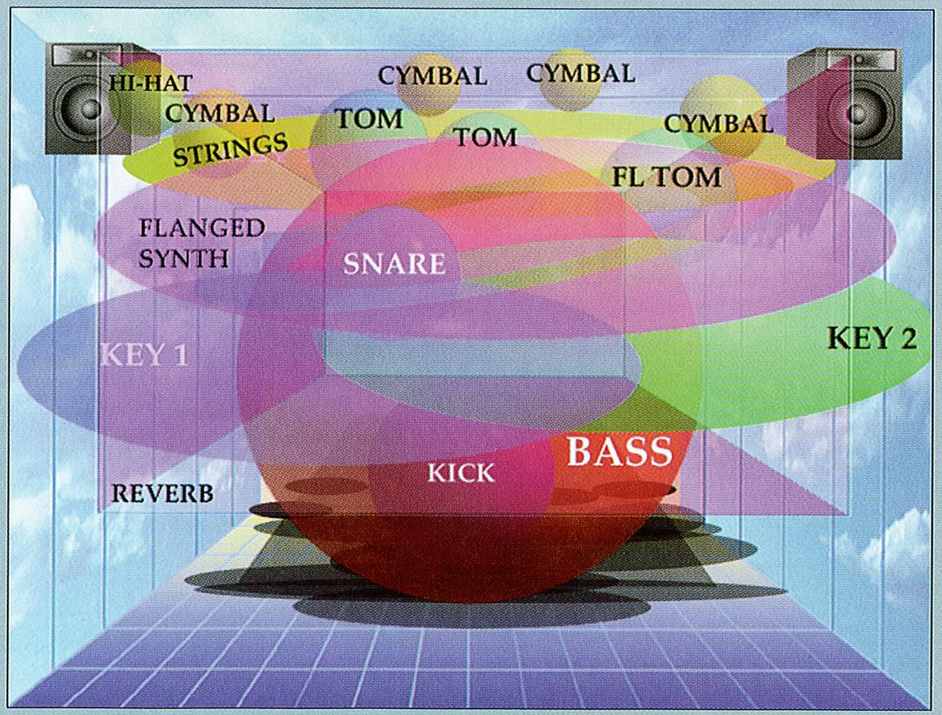
Visual 167C.
Heavy Metal Mix

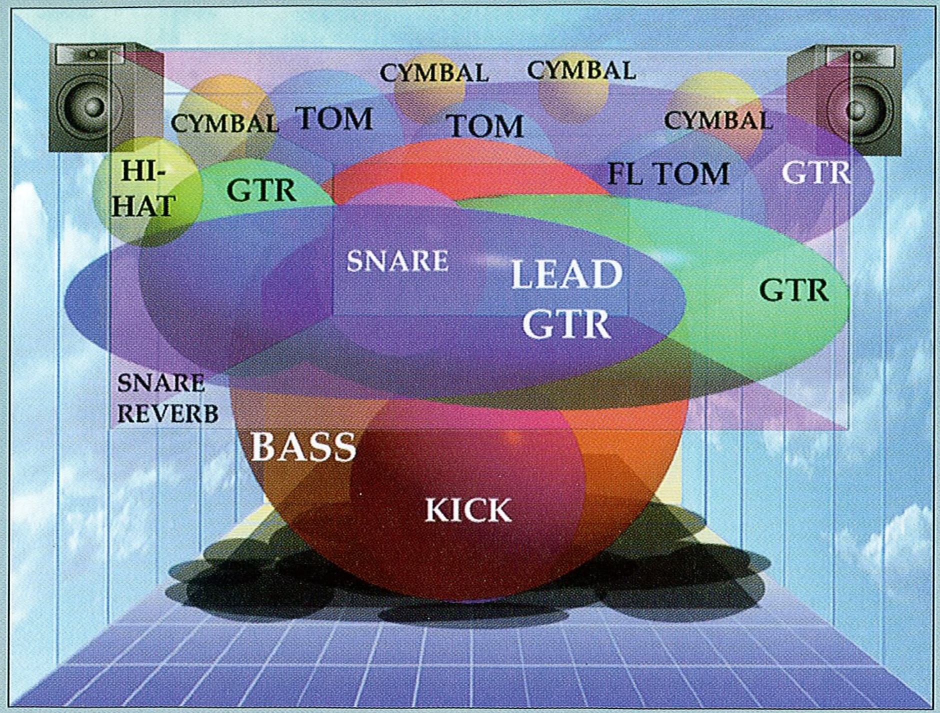
A very full arrangement and mix. Note the clarity of the low end (kick and bass), even though it is an extremely busy mix. The hi-hat, the snare, and especially the lead guitar are right out front. Note the multiple guitar parts with a few panned in stereo. The reverb is present but not so loud that it muddies everything. There isn't much room left for effects unless there is a breakdown section in the song. The overall effect is a massive, powerful wall of sound.



Visual 203C.
New Age Mix

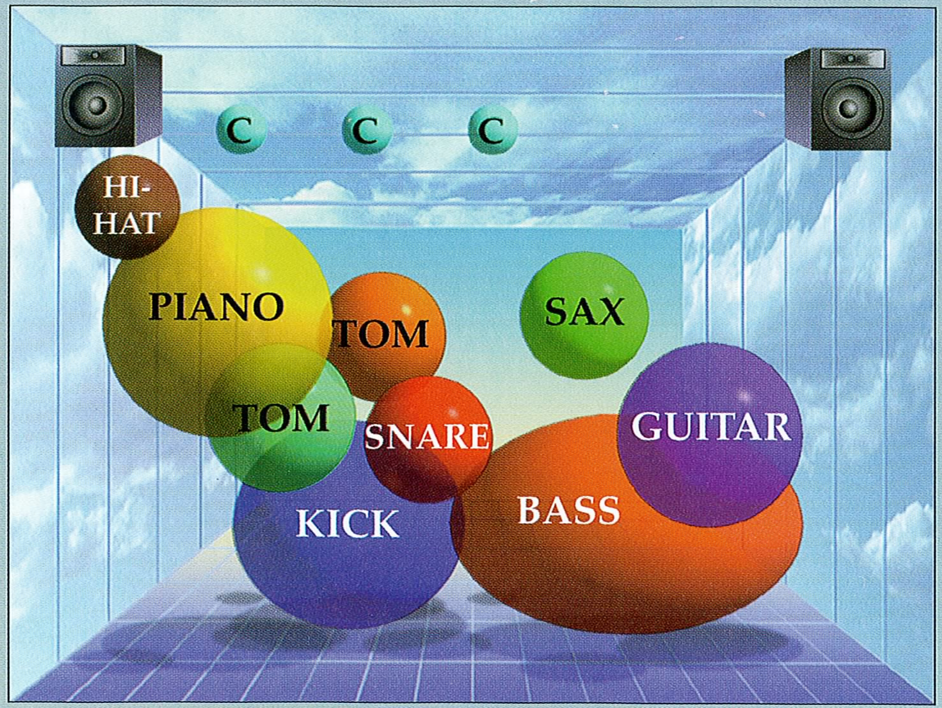
This mix is extremely full with nothing too sharp or cutting (although often individual lead sounds are extremely strong). Note the fattening on the keys and strings, filling out the space. The stereo flanged synth is quite prominent here, and the bass is huge. The low end is kept nice and clean, and the high-frequency cymbals are placed above it all.





Visual 168C. Alternative Rock Mix

Quite full with lots of fattening and overlapping sounds. The lead guitar is spread in stereo with a rhythm guitar right behind it and another stereo guitar in the background. A nice, clean low end, even though the mix is full. The kick and bass are quite strong.

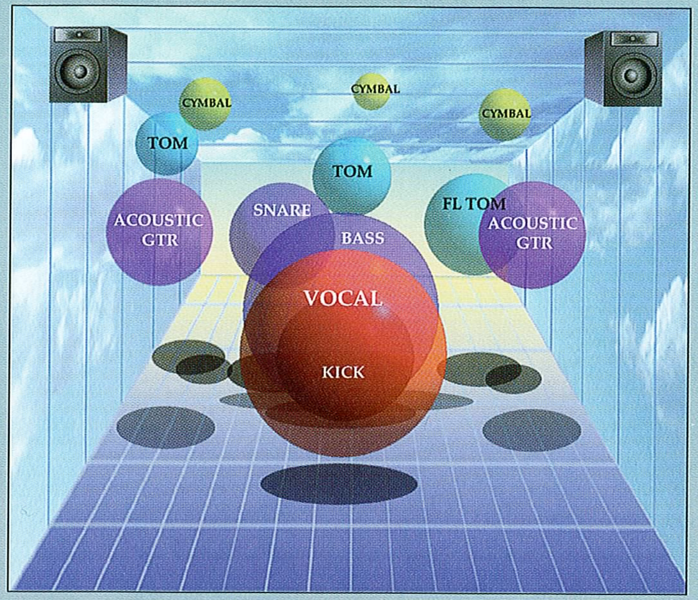


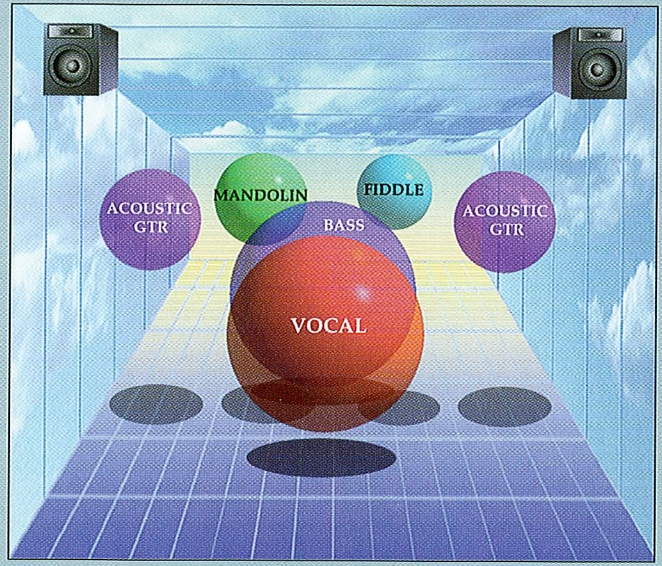
Visual 198C. Acoustic Jazz Mix

Note the incredible cleanliness and clarity of the overall mix. The bass is panned to the right and doesn't have much high end. The guitar is right out front with the piano and the hi-hat. The kick is quite loud, which is not typical.

Visual 199C. Folk Music Mix

This type of music is typically mixed very clean and clear. The vocal is extremely loud. Note the presence and complete left and right panning of the acoustic guitars. The snare is set back, and the bass guitar and kick are not too overwhelming.



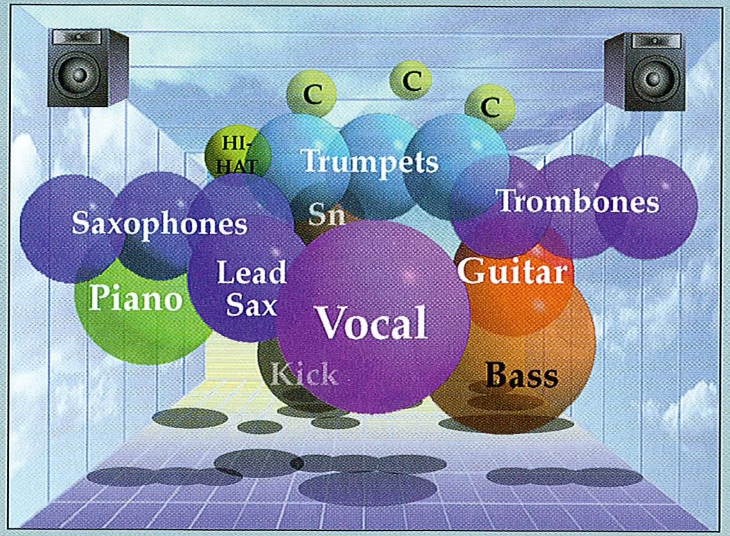


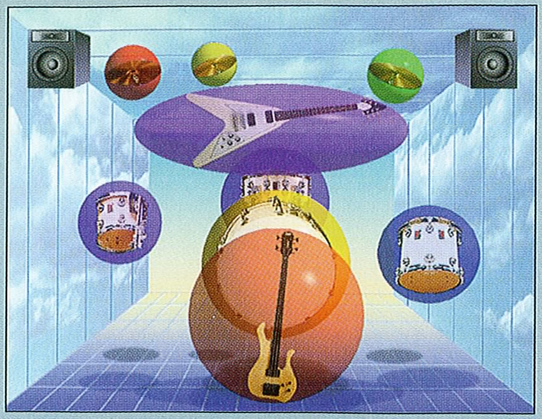
Visual 200C. Bluegrass Mix

Extremely clear and crispy mix. The volume is relatively even in all the instruments. When there is a vocal, it is right out front. The bass is set back and is sometimes panned to one side. The lead solos on any instrument might be bumped up some.

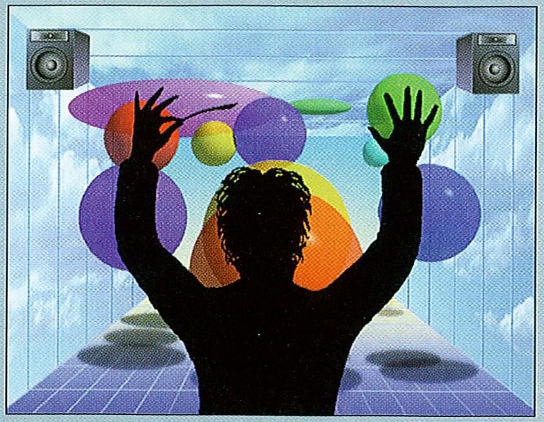
Visual 166C. Big Band Mix

A very clean and clear mix. It's typical for the vocal to be extremely loud. The horns are loud compared to the rest of the band. The hi-hat is sometimes right out front, while the piano, the guitar, the snare, the bass, and especially the kick are often quite low in volume.

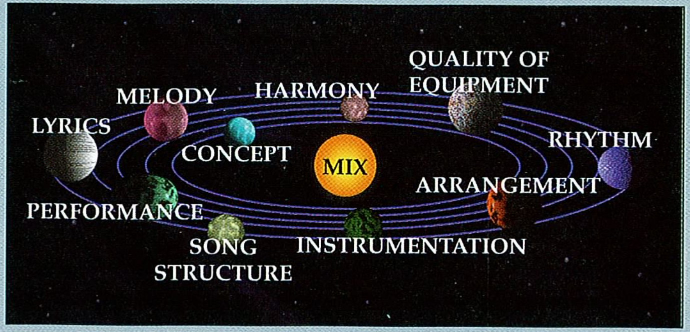




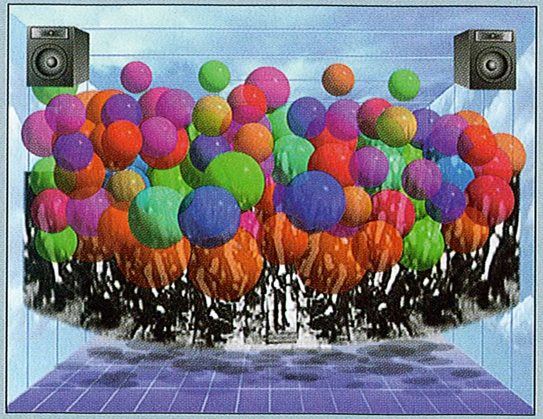
Visual 1C. Sound Imaging



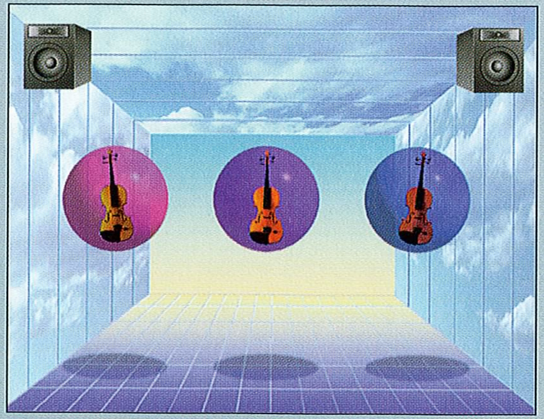
Visual 2C. Structuring Mix



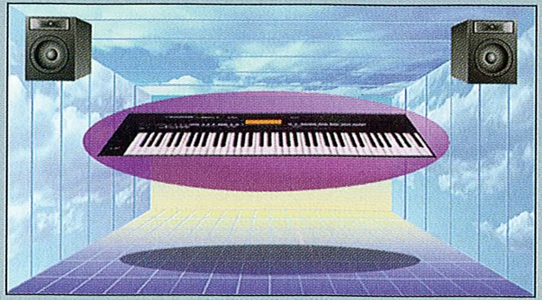
Visual 3C. 11 Aspects of Recorded Piece of Music

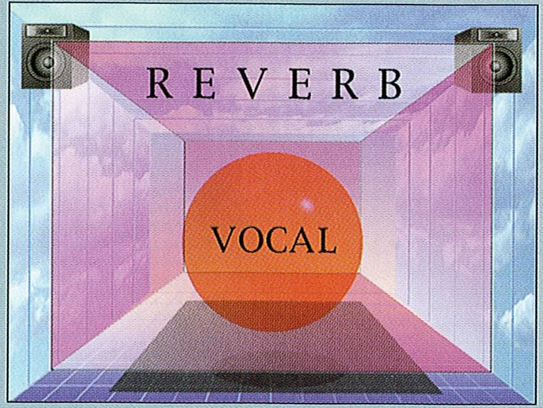


Visual 20C. Large Orchestra Crowded Between Speakers



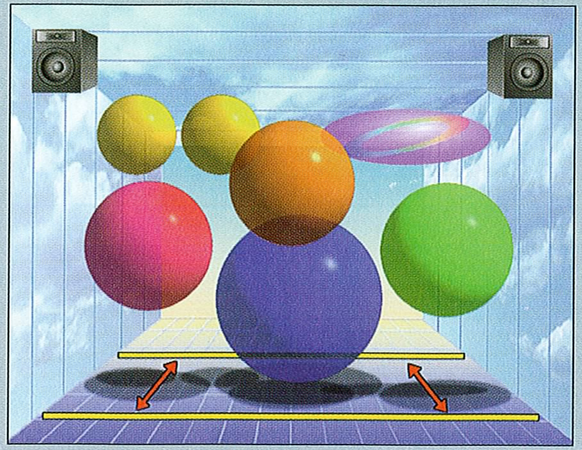
Visual 21C. 3 Violins With Plenty of Space in Between



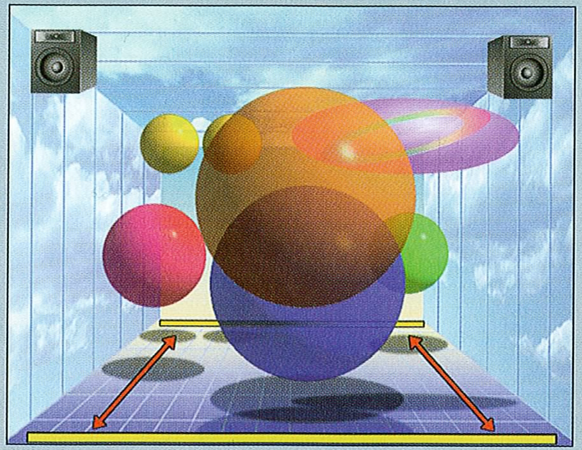


(above) Visual 29C. Fattening: <30ms Delay Time

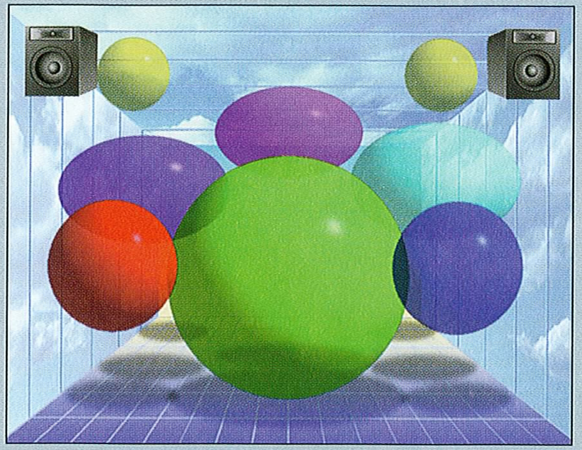
(right) Visual 36C. Stereo Reverb on Sound



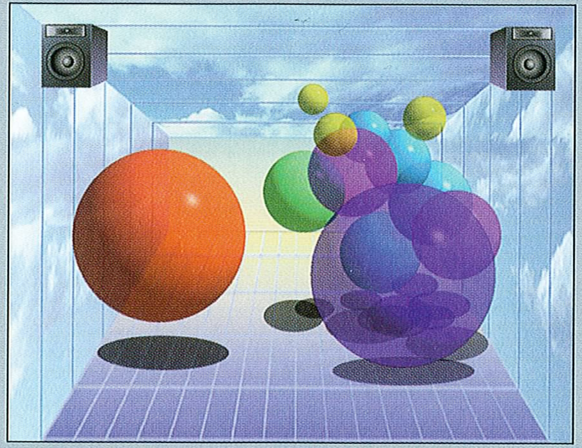
Visual 42C. Even Volumes



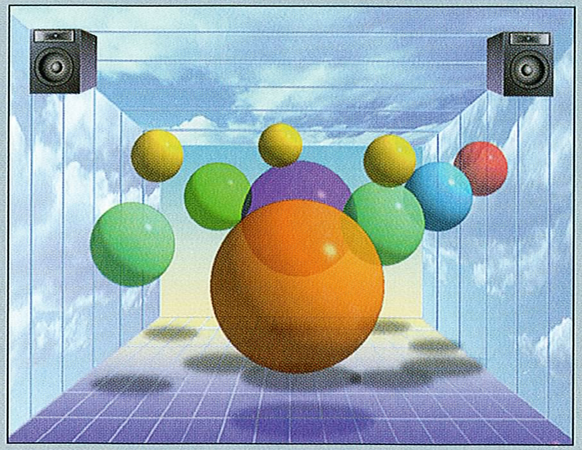
Visual 43C. Uneven Volumes



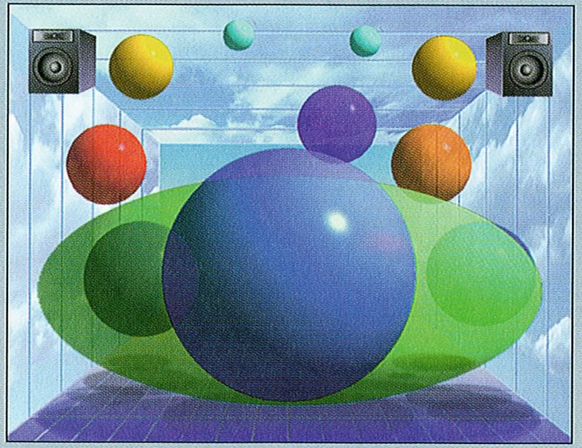
Visual 44C. Balanced (Symmetrical) Mix



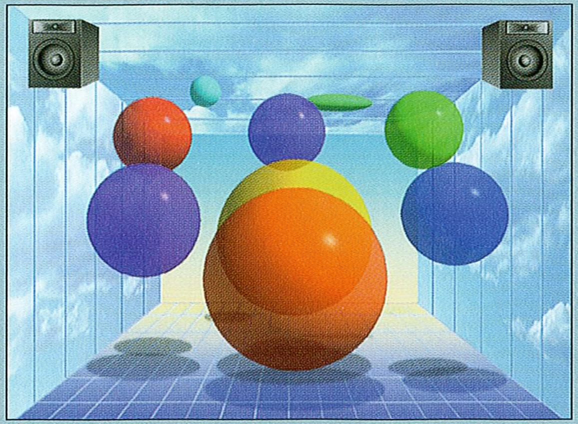
Visual 45C. Lopsided (Asymmetrical) Mix



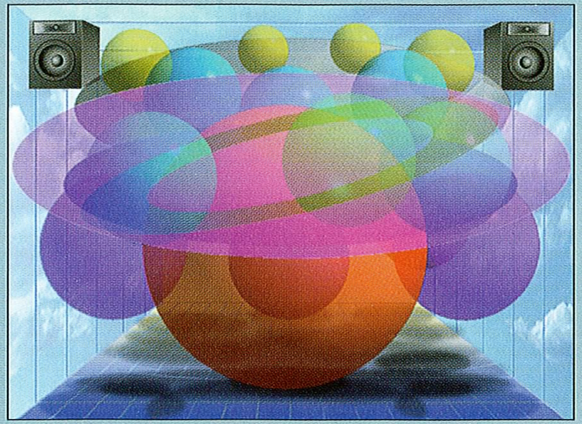
Visual 46C. Natural EQ



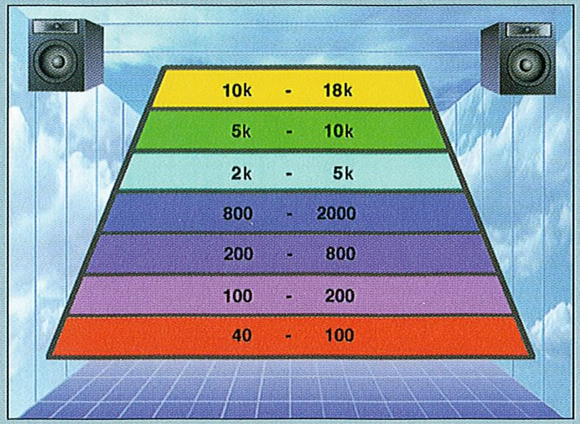
Visual 47C. Interesting EQ



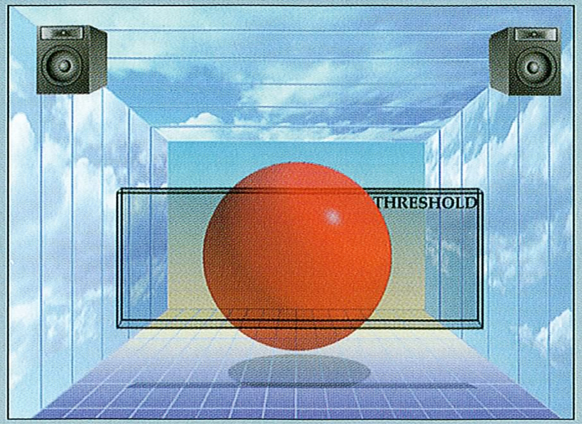
Visual 48C. Sparse Mix



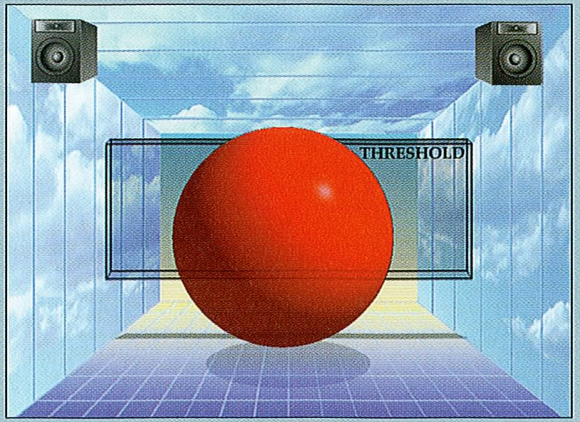
Visual 49C. Full (Wall of Sound) Mix



Visual 50C. Virtual Mixer EQ

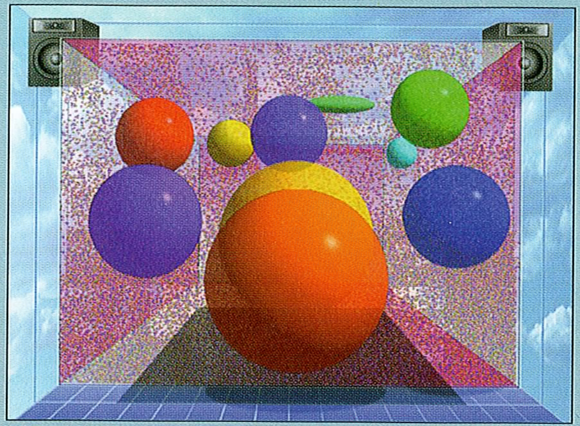


Visual 59C. Sound Smashing Into Threshold on Compressor/Limiter

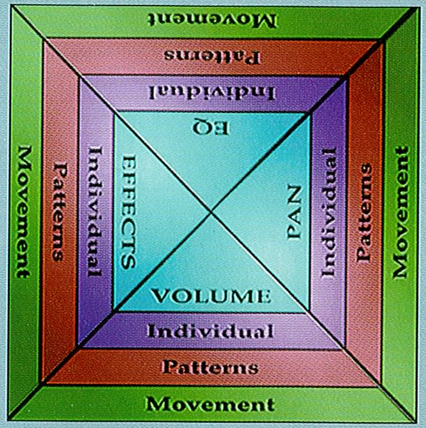


Visual 61C. Sound Fading Out Past Threshold on Noise Gate

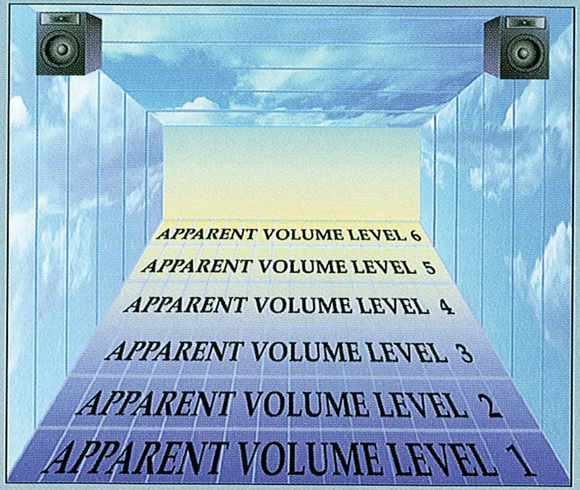
Visual 88C. Virtual Mixer Flanging



Visual 109C. Reverb Filling in Space Between Speakers

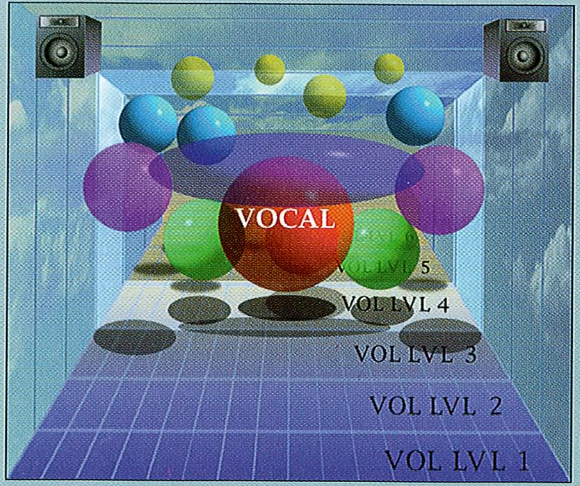


Visual 128C. Pyramid of Tools and Dynamics

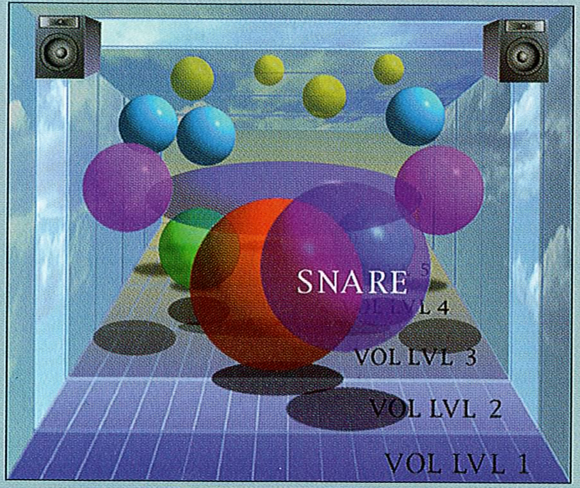


Visual 129C. 6 Apparent Volume Levels

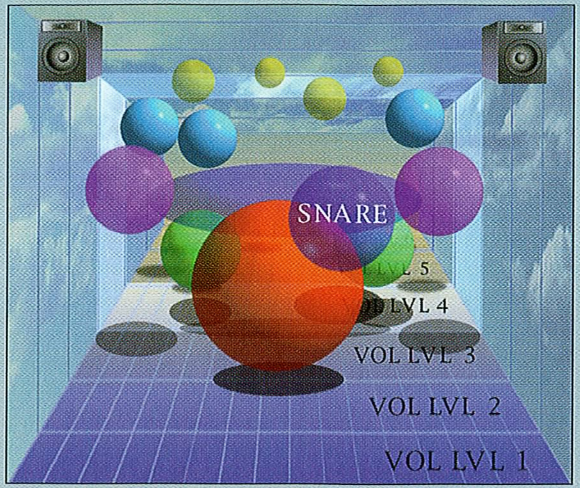
Visual 136C. Apparent Volume Level 2 Vocals



Visual 138C. Apparent Volume Level 4 Vocals



Visual 139C. Apparent Volume Level 2 Snare



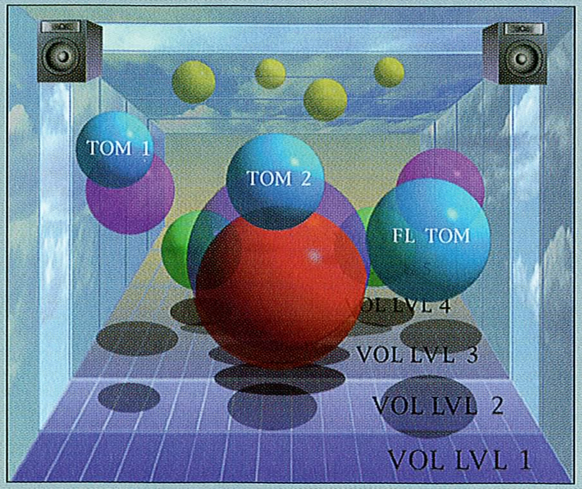
Visual 141C. Apparent Volume Level 4 Snare

Visual 143C. Apparent Volume Level 1 Kick

Visual 146C. Apparent Volume Level 4 Kick

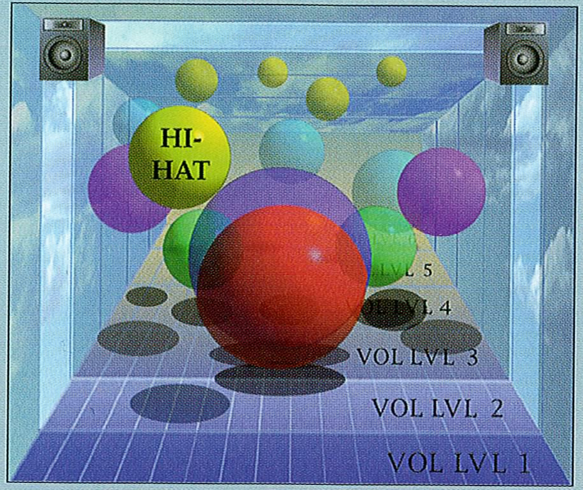
Visual 148C. Apparent Volume Level 1 Bass

Visual 150C. Apparent Volume Level 3 Bass

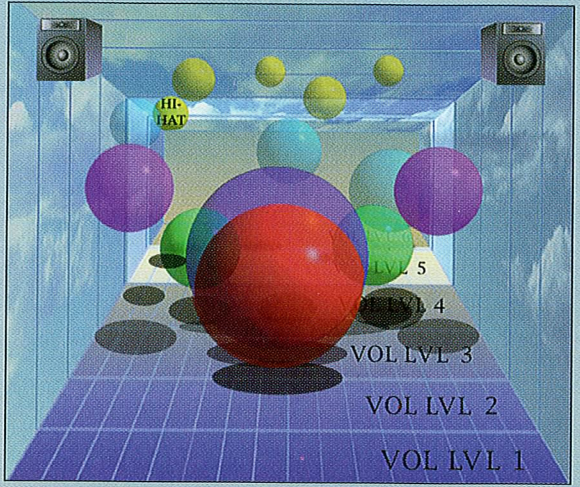


Visual 152C. Apparent Volume Level 2 Toms

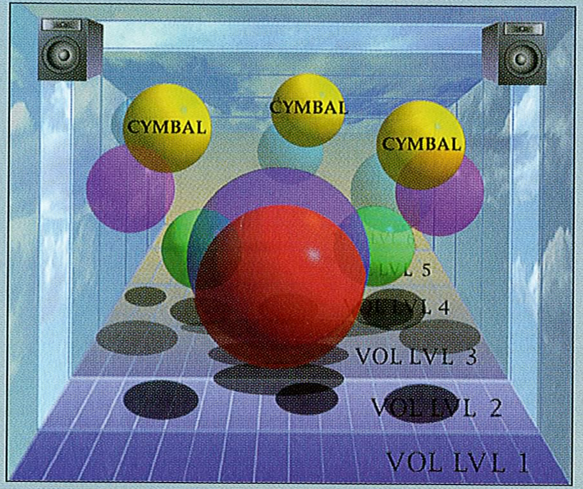
Visual 154C. Apparent Volume Level 4 Toms



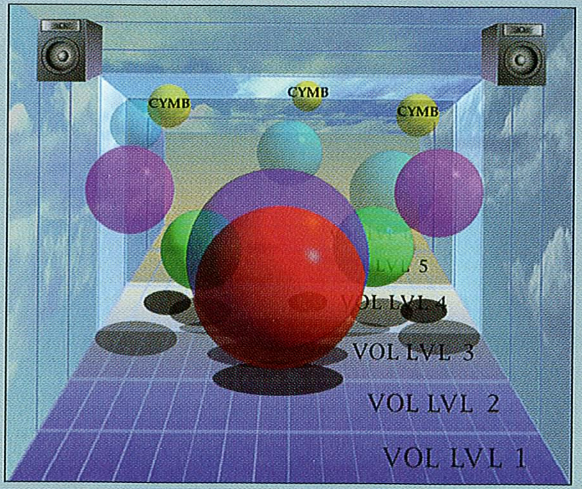
Visual 155C. Apparent Volume Level 2 Hi-Hat



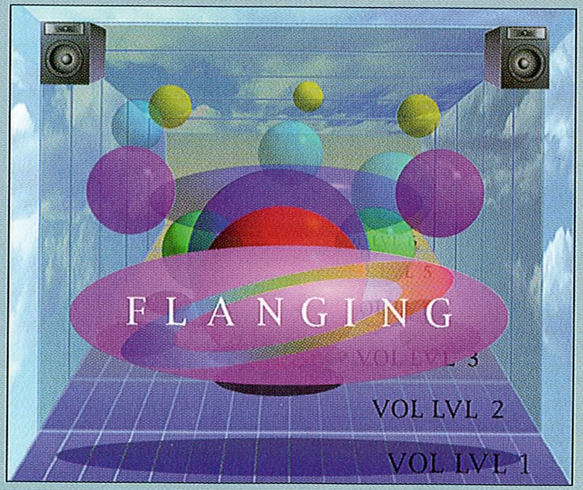
Visual 157C. Apparent Volume Level 5 Hi-Hat



Visual 158C. Apparent Volume Level 2 Cymbals



Visual 159C. Apparent Volume Level 4 Cymbals

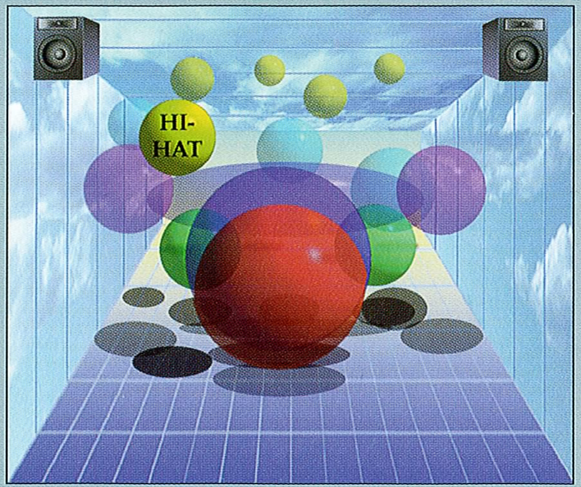


Visual 160C. Apparent Volume Level 1 Effects

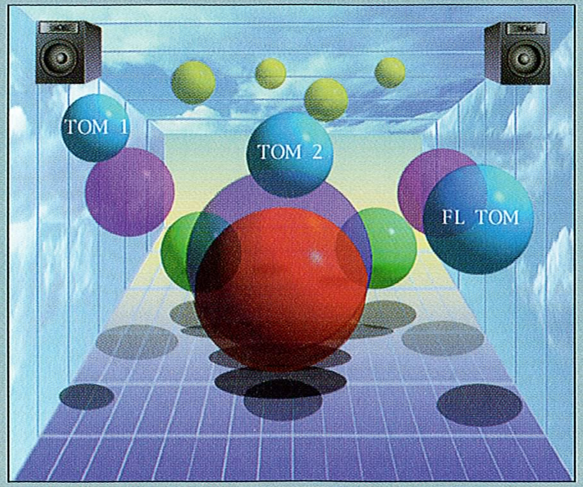
Visual 161C. Apparent Volume Level 2 Effects

Visual 162C. Apparent Volume Level 4 Effects

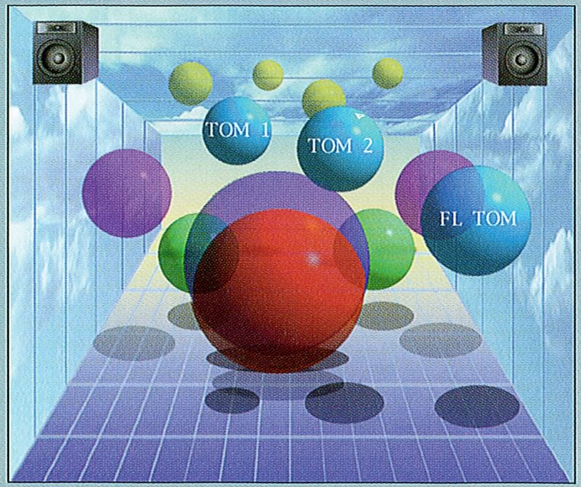
Visual 163C. Apparent Volume Level 5 Effects



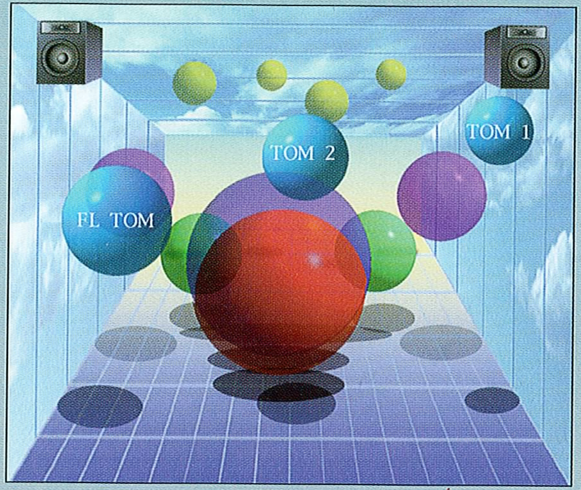
Visual 176C. Hi-Hat Panned Halfway Between Left Side and Middle



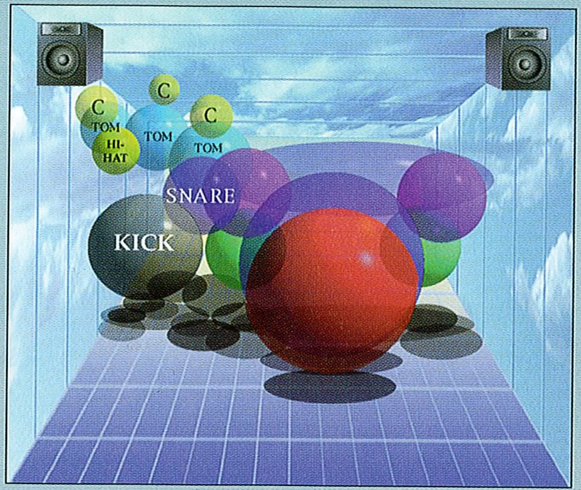
Visual 177C. Toms Panned Completely Left to Right



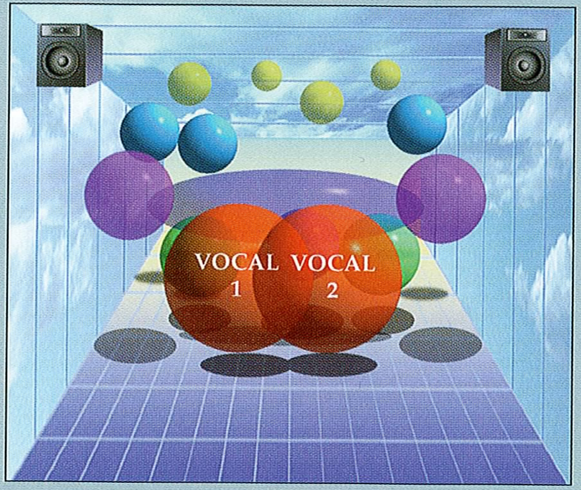
Visual 178C. Toms Panned Same As on Drum Kit



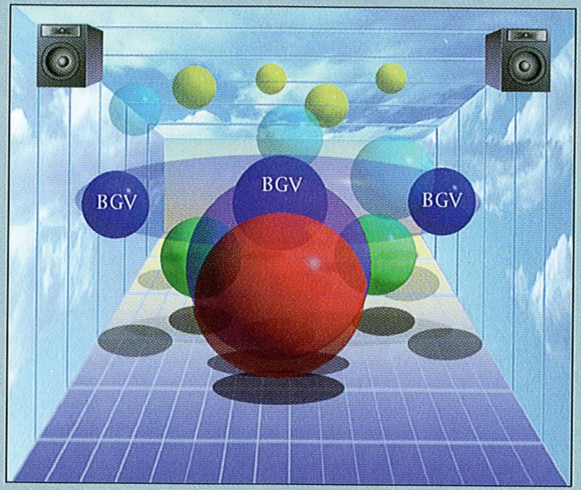
Visual 180C. Toms Panned Right to Left



Visual 183C. Mix With Drums Panned to One Side

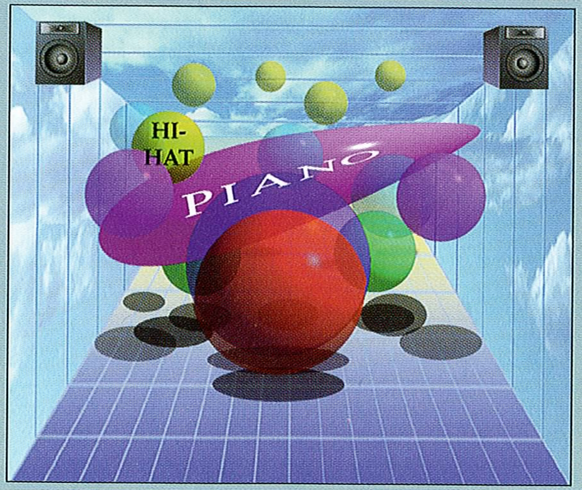


Visual 185C. Lead Vocals Panned at 11:00 and 1:00

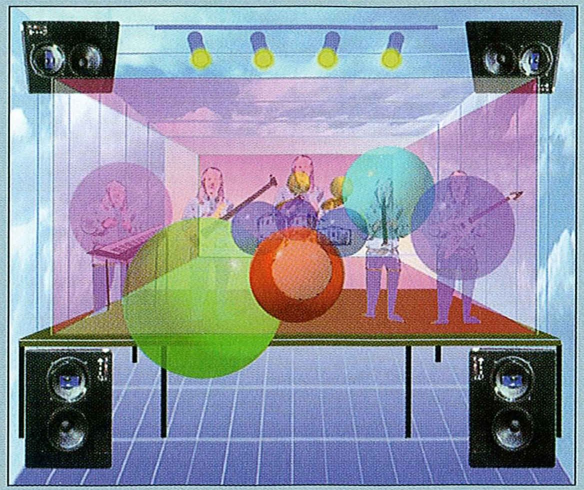


Visual 189C. 3 Background Vocals Panned Separately

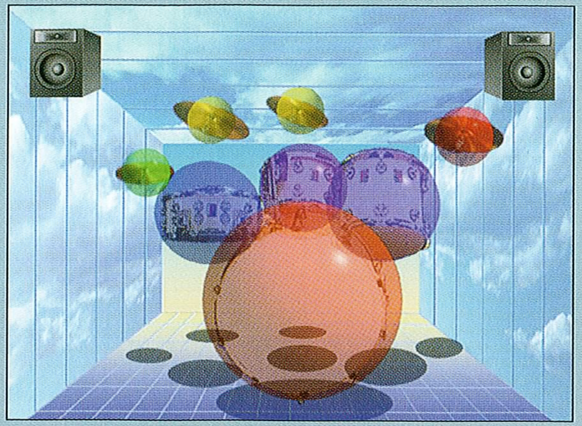
Visual 190C. 7 Background Vocals Panned to 7 Different Places Combined With Variety of Fattening



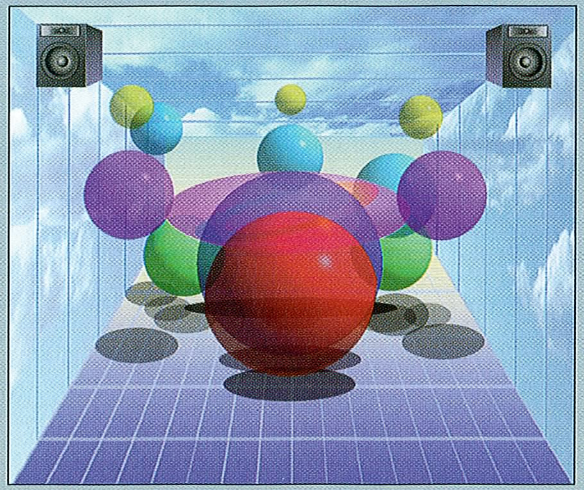
Visual 191C. Panning With High End of Piano on Right and Hi-Hat on Left



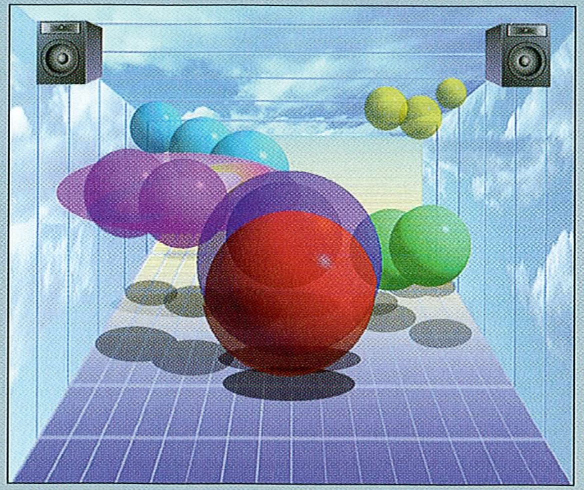
Visual 192C. Panning As If Onstage



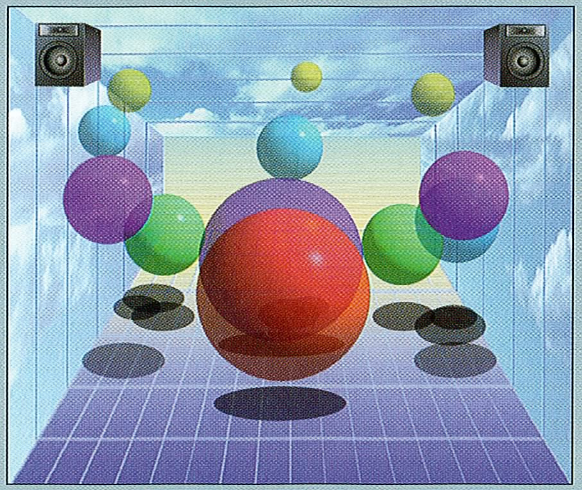
Visual 193C. Natural Panning of Drum Kit



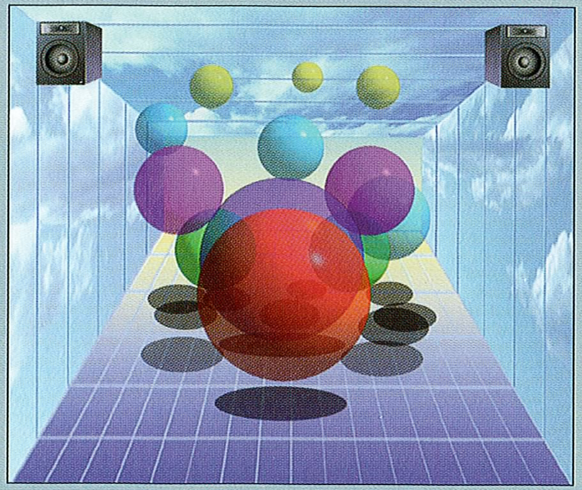
Visual 196C. Mix Balanced at Each Frequency Range



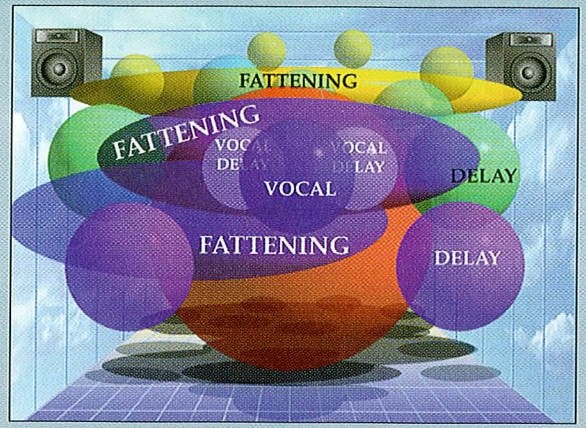
Visual 197C. Unbalanced Mix at Each Frequency Range



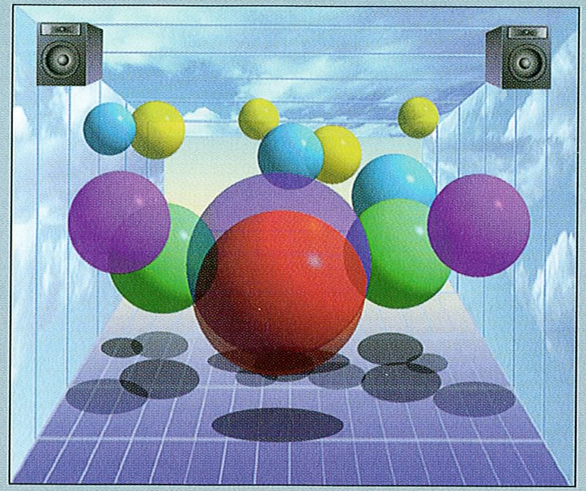
Visual 204C. Mix With Extremely Wide Panning Overall



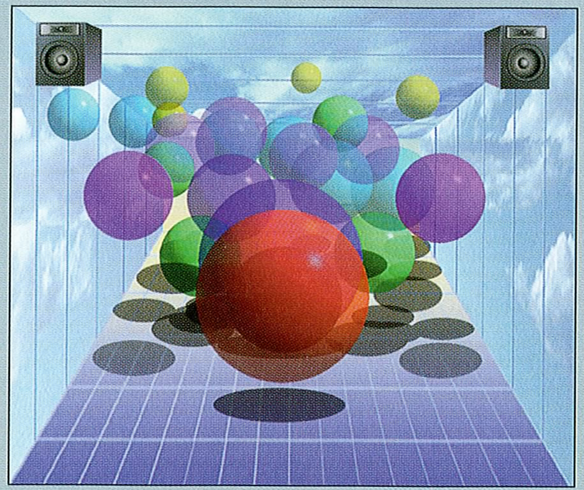
Visual 205C. Mix With Panning Not So Wide Overall



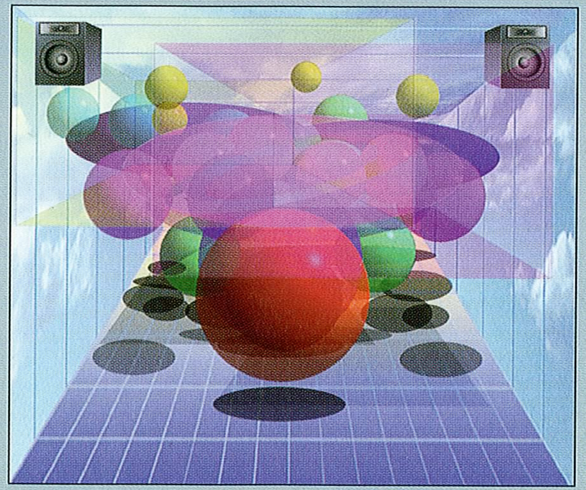
Visual 210C. Mix With Lots of Different Delays Filling Out Mix



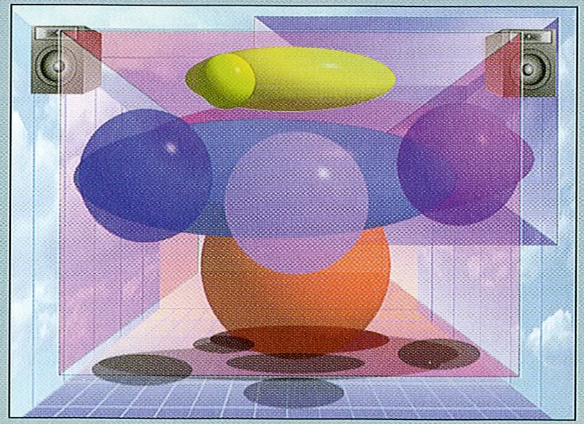
Visual 218C. Clean and Clear Mix



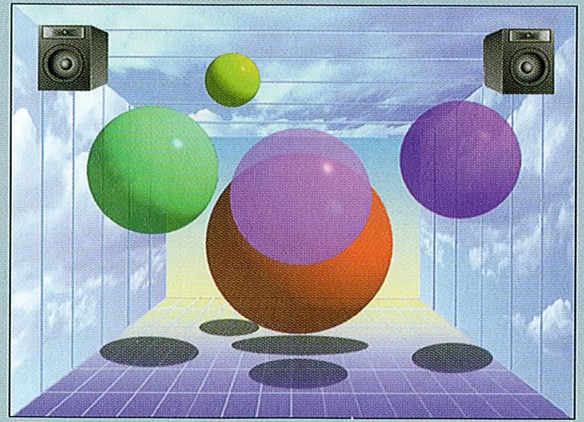
Visual 214C. Extremely Busy Mix With No Effects



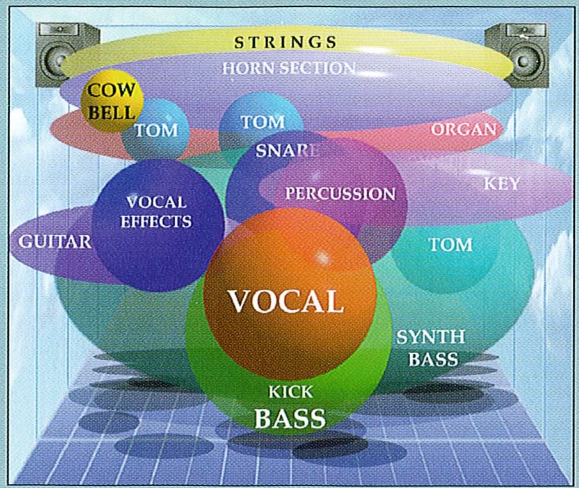
Visual 215C. Extremely Busy Mix With Lots of Effects



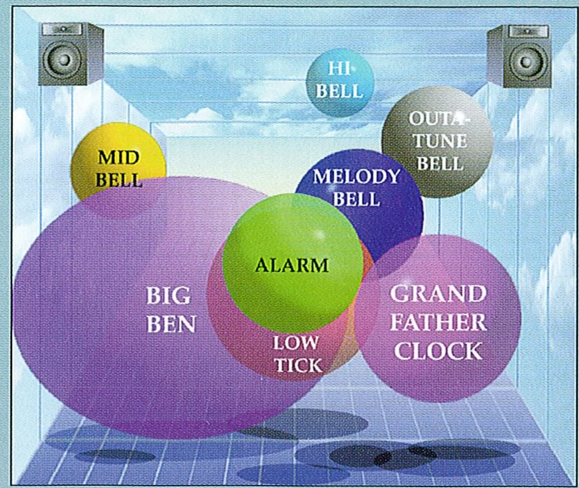
Visual 216C. Extremely Sparse Mix With Fattening and Reverb



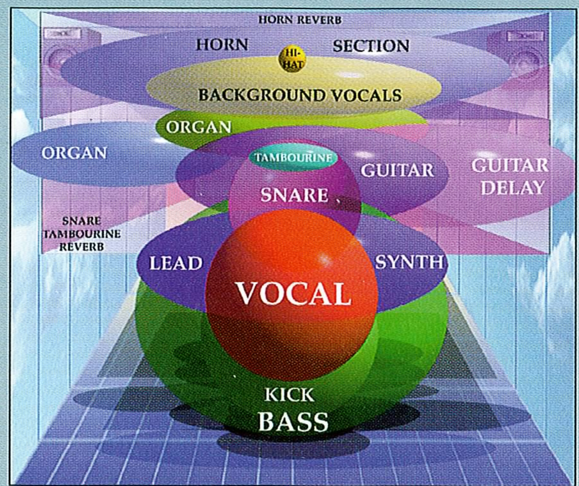
Visual 217C. Extremely Sparse Mix With No Fattening and Reverb



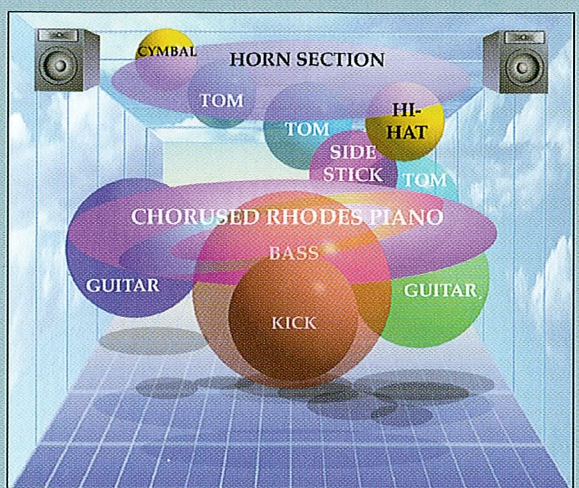
Visual L. "Blinded Me With Science" on Wireless by Thomas Dolby



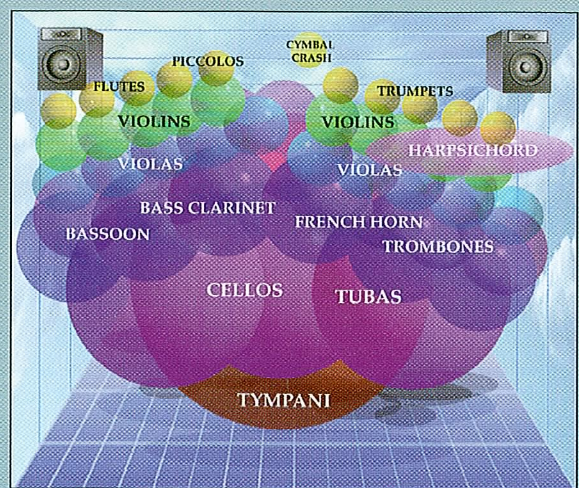
Visual M. The Alarm Clock Section in "Time" on Dark Side of the Moon by Pink Floyd



Visual N. "Sledgehammer" on So by Peter Gabriel



Visual O. "Babylon Sisters" on Gaucho by Steely Dan



Visual P. The Four Seasons by Vivaldi

As distances become smaller and smaller, the distance in feet almost equals the milliseconds of delay. This correlation comes into play when using more than one mic on a sound (e.g., piano, guitar amps, acoustic guitars, horns, or background vocals) and is especially helpful when miking drums. For example, the distance you place overhead mics above the drum set will create a corresponding delay time between the overhead mics and the snare mic (or any of the rest of the mics for that matter). It is also important to note the distance between instruments when miking an entire band live (or recording everyone in the same room at once) since mics may be more than ten feet away from another instrument and still pick it up.

Besides delay time, you must also consider phase cancelation, a problem that happens with extremely short delay times. We'll discuss more about this later.

If you pay attention to the way that something sounds when miked at different distances, you will eventually become aware of what different delay times sound like. Once you become familiar with the way that different delays affect different sounds, you can control their use in a way you deem most appropriate; that is, you can do whatever you want.

There might not be any rules in this industry except one: Gain a perspective so that you know what you are doing. Then, if anyone disagrees, it doesn't matter.

Different Delay Times

Let's define specific delay time ranges, so that you can get to know them and incorporate them into your memory time banks.

More than 100ms

Professional engineers refer to this length of delay as "echo." However, the real world (and my mom) use the term echo to refer to reverb. For our purposes, we will use echo to refer to a delay time greater than 100ms, not reverb.

When setting a delay time greater than 100ms, it is important that the delay time fits the tempo of the song, otherwise it will throw off the timing of the song. The delay time should be in time, a multiple of, or an exact fraction of the tempo. The following chart gives the relationship between tempos and delay times.

Beats Per Minute	Time Between Beats
60 bpm	1000 ms
90 bpm	750 ms
120 bpm	500 ms
150 bpm	437.5 ms
180 bpm	375 ms
210 bpm	312.5 ms
240 bpm	250 ms

Chart 9. Tempo vs. Delay Time

A simple way to set delay times to the tempo without the chart is to put the delay on the snare drum (or some other instrument playing a continuous pattern). You can then hear when the delay is in time with the tempo of the song. Once you have found a delay time that works, any multiple or fraction of that time might also work.

A delay time over 100ms creates a dreamy effect and is most commonly placed in songs with slower tempos where there is room for the additional sound. Therefore, the more instruments and the more notes in a mix, the less often delays are used. This is especially true when there is feedback on a long delay time. The delays take up so much space in a mix that they are often only turned up at the end of a line, where there is enough space to hear the echoes by themselves.

60 to 100ms

You can hear this delay time, commonly referred to as "slap," on the vocals of Elvis Presley and in rockabilly music. In fact, there is about an 80ms delay between the syllables "rock" and "a" in the word "rockabilly."

This effect can be quite helpful in making a thin or irritating sound (especially a voice) sound fuller. It can help to obscure bad vocal technique or pitch problems. In fact, a slap can be used to bury any bad sound. However, you never want to bury anything too deep. Add too much delay on a bad vocal and not only do you have a bad vocal, but you also have a bad mix. On the other hand, a slap can make a vocal seem less personal. If you have an incredible singer, you might forego using any delays. Just put it out there with a little reverb and let it shine.

30 to 60 ms

Put your lips together and blow a raspberry (this is the interactive portion of the book), technically called a "motorboat." The time between each flap of your lips is approximately 50ms. Delay time in this range is referred to as "doubling" because it makes a sound seem like it was played twice, or double tracked. When a part is sung or played twice there will naturally be a time delay ranging from 30 to 60ms (no one can ever sing or play a part twice exactly in time). Therefore, adding a delay of this length makes it sound like the part has been played twice. The Beatles used this effect extensively to simulate more vocals and instruments.

Just like a slap, doubling helps to obscure a bad sound or a bad performance. So it can be used to help bury things in the mix. Likewise, since it does obscure the purity and clarity of a sound, you should use it selectively, depending on the sound, song, and style of music.

NOTE: Although doubling makes a sound seem like it has been played twice, it is a different sound than if you actually doubletrack a sound. In fact, doubling often sounds so precise that it sounds somewhat electronic. This is especially true on vocals and simple sounds. However, if a sound is complex, especially if the sound is a combination of sounds (like a bunch of background vocals or a guitar sound with multiple mics), then you don't notice the precision of the delay. Therefore, when you put doubling on 20 vocals, it sounds like 40 vocals, and it sounds incredibly natural.

1 to 30ms

An unusual thing happens with this type of delay, commonly known as fattening. At this delay time, our brain and ears are not quick enough to hear two sounds; we only hear one fatter sound.

The threshold between hearing one sound or two sounds actually varies depending on the duration of the sound being delayed. Also, the delay times are even shorter when the original sound and the delay are panned separately, left and right. The following chart gives approximate thresholds for some instruments with different durations (actual thresholds will depend on the particular timbre and playing style of the instrument):

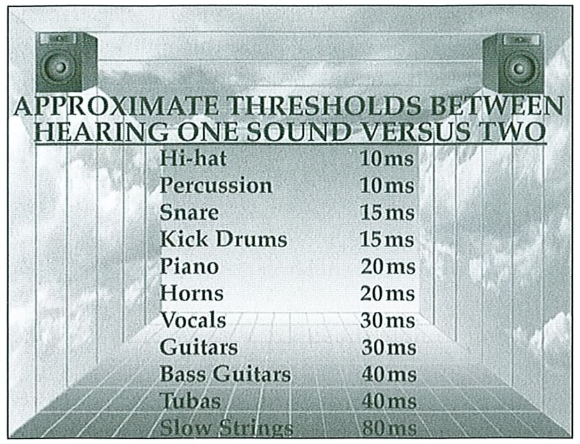
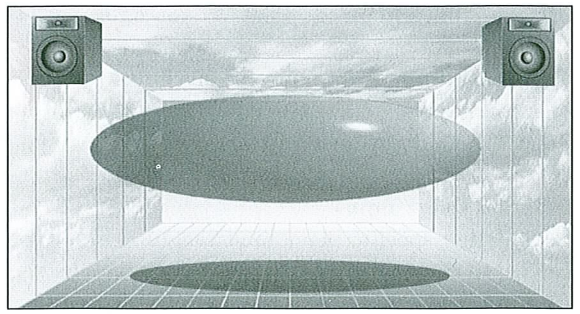


Chart 10. Quickness of Our Brain

Besides reverb, fattening is the most-used effect in the studio, mostly because it doesn't sound much like an effect. Fattening is the primary effect used to make a sound stereo, which has a certain magic to it. When you put the original "dry" instrument sound in one speaker and put a delay less than 30ms in the other speaker, it "stretches" the sound in stereo between the speakers.



Visual 83. Fattening: Delay <30ms

Fattening can make an already beautiful acoustic guitar or piano sound incredible. Fattening is the most effective of all delay times in making a thin or irritating sound fatter and fuller. It also seems to make a sound more present simply because when a sound is in stereo, it takes up more space between the speakers. This is especially effective when you want to turn a sound down in the mix but still have it discernible. On the other hand, because fattening will make a mix fuller and denser, you must make sure there is enough room between the speakers. Therefore, fattening is used most often when there are fewer notes and sounds in the mix.

When you want to create a wall of sound, though, even if the mix is already busy, you can add fattening to make it more busy. (This blows people's minds.) This is commonly done in heavy metal, alternative rock, and some new age music.

0 to 1ms

This sort of a delay time causes phase cancelation. I will address only the critical aspects of phase cancelation here. But keep in mind that phase cancelation is a very serious problem in recording and I highly recommend that you do further research to gain a complete and clear explanation of the problems and detriments of it.

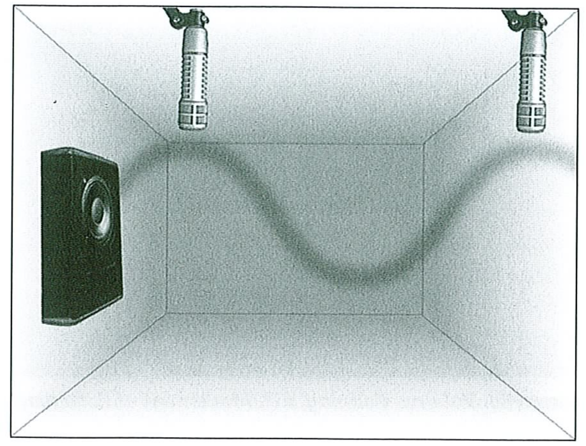
Phase cancelation happens when two of the exact same sound, like those created with two mics or two speakers, are a little bit out of time. A perfect example is when you switch the positive and negative wires on one of two speakers. Now, one speaker is pushing out while the other is pulling in. When a speaker pushes out, it creates denser air than normal. When a speaker pulls in, it creates more spaced out air than normal (rarefied air). When the denser air from one speaker meets the spaced out air from the other speaker, you end up with normal air; normal air equals silence. This means you could have two speakers blasting away and theoretically you could hear nothing.

There are many companies now using phase cancelation to quiet the world. This technology is used in automobiles, on freeways (instead of cement walls on the sides of the freeways), in factories, and even in headphones to cancel out sounds around you. Marriage counselors are selling them by the dozens.

If you have two mics on one sound at two different distances, one mic might be picking up denser air while the other mic is picking up spaced out air. Put the two mics together in the mix and they will tend to cancel each other out, though not completely. The following are common problems when using more than one mic on drums, piano, and guitar.

1. You lose volume when both mics are on, especially when you're in mono (which, by the way, is one of the best ways to detect phase cancelation).
2. You lose bass frequencies, making the sounds thin.
3. Most importantly, you lose the clarity and precision of the perceived image of the sound between the speakers. The sound seems to be more "spacey." Though some people like this effect, most people are addicted to clarity these days. If the mix is ever played back in mono (as on TV or AM radio), the sound will disappear completely.

There are many ways to curb phase cancelation. The primary way is to simply move one of the mics. If both mics are picking up the sound in the same excursion of the wave, there will be no phase cancelation.



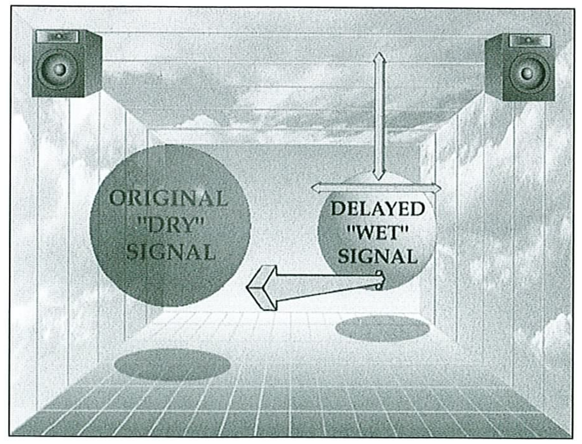
Visual 84. Two Mics Picking Up Sound in Phase

It takes 1ms for a complete wave of 1000Hz to pass by us. If we were to set a delay time of .5ms on a sound, it would put it out of phase. Therefore, we can use a digital delay to put the sound back in time.

Finally, we can remove a large amount of phase cancelation through isolation. Often, the bleed of a sound into a second mic will cause phase cancelation with the first mic. By using baffles or noise gates, we can reduce the bleed in the second mic, voiding the phase cancelation.

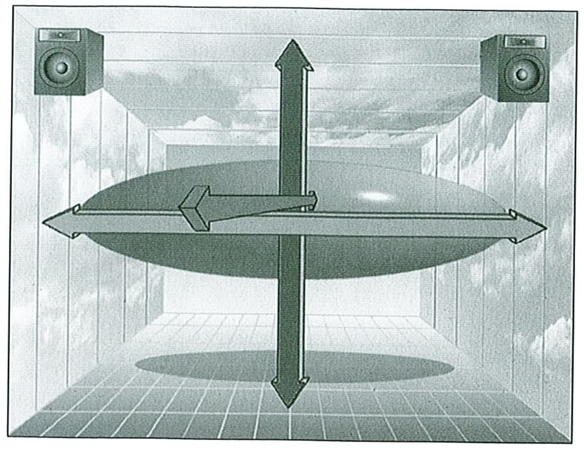
Panning of Delays

When the delay time is long enough to hear two sounds, then the delayed signal can be treated just like another sound and can be placed anywhere in the mix using volume, panning, and EQ.



Visual 85. Volume, Panning, EQ, Movement of Delay >30ms

When the delay time is less than 30ms or so, fattening occurs. We can also place this line of sound anywhere with volume, panning, and EQ.

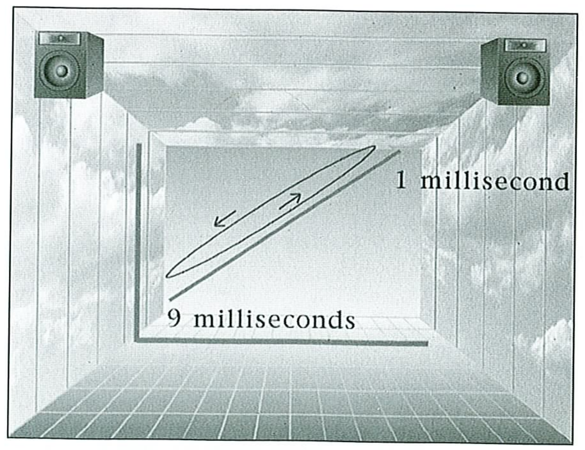


Visual 86. Volume, Panning, EQ, Movement of Fattening

FLANGERS, CHORUSES, AND PHASE SHIFTERS

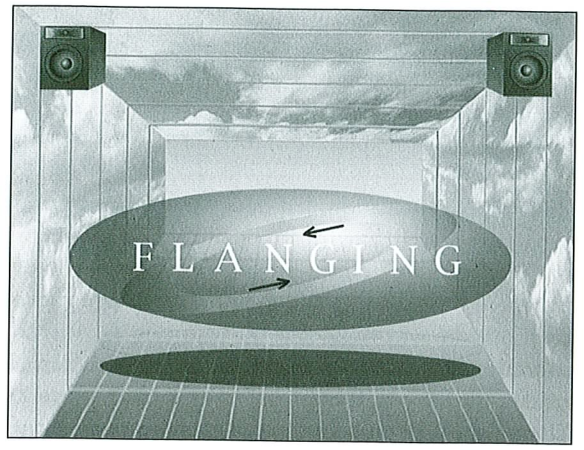
In 1957, Toni Fisher was doing an album and someone accidentally sat on one of the reels on the tape player, slowing down the tape. When they stood up, it sped back up to normal speed. The band went, "Cool, let's put it on the record." They did put it on the record, and thus, flanging was born. The song, "The Big Hurt," went to No. 3 on the charts in 1957.

If you set a digital delay for less than 30ms of delay time and crank up the feedback, you get an effect called tubing (check it out on a digital delay). The interesting thing is that the shorter you set the delay time, the higher the pitch of the tube. The longer the delay time, the lower the pitch of the tube. Now, if you set a clock to sweep the delay time back and forth between, say, 9 and 1ms, then you get the effect called flanging.



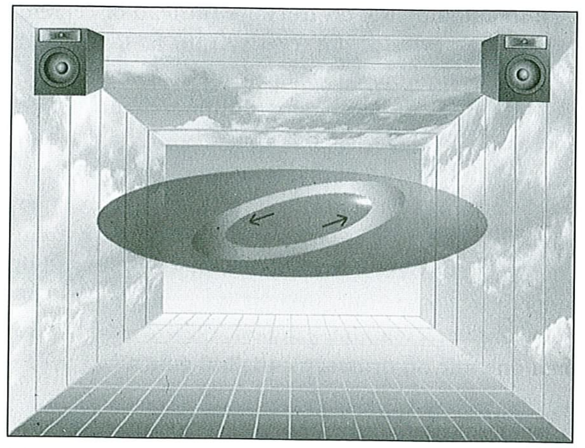
Visual 87. Pitch vs. Delay Time of Flanging

A flange is shown visually like this:



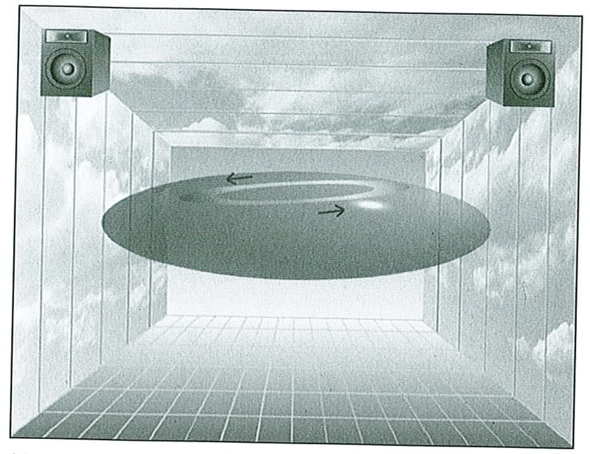
Visual 88. Virtual Mixer Flanging (see color Visual 88C)

If you set the width (depth or intensity on different units) so that the sweep of the delay time is not so wide, you then have the effect called chorusing. (Chorus effects have a delay like doubling or fattening also added.)



Visual 89. Virtual Mixer Chorusing

If you set the delay time so that you are only sweeping between 0 and 1ms, you hear the effect called phasing.



Visual 90. Virtual Mixer Phasing

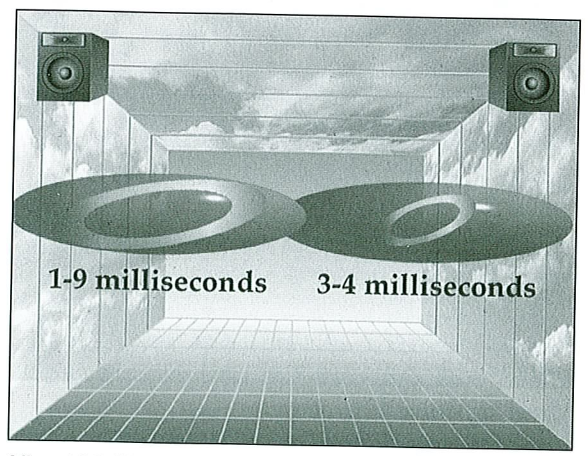
There are various parameters or settings found on flange, chorusing, and phasing units:

Rate, Speed, Frequency

The setting is the time it takes for the delay to sweep back and forth between two delay times. For example, it can be set to take one second to smoothly change from 1 to 9ms and back. The rate of the sweep can be set to the tempo of the song—you might have it rise on one beat and fall on the next beat—or to rise on one chord and fall on the next chord. You could even set it to rise on the first half of the verse and fall on the second half. The rate is often set so slow that it doesn't correspond to any part in the music.

Width, Depth, Intensity

This setting is the range of the delay sweep. For example, a narrow width setting might sweep between 3 and 4ms, while a wide width setting might sweep between 1 and 9ms. Because pitch corresponds to the delay time, this means that the wider (or deeper) the setting, the wider the frequency sweep.



Visual 91. Narrow and Wide Sweep on Flange, Chorus, or Phaser

Feedback

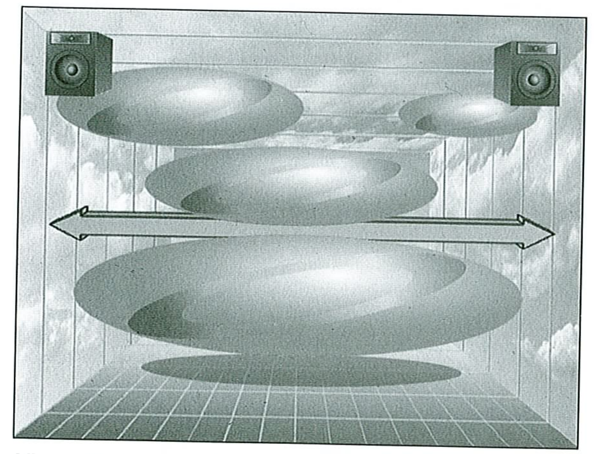
Feedback takes the output of the delay and "feeds it back" into the input. Some feedback is required to get the flange effect in the first place. The more feedback you add, the more intense or dynamic the frequency sweep.

Negative Feedback

Negative feedback puts the signal being fed back into the input out of phase. This generally causes a more hollow tubular type of flange sound.

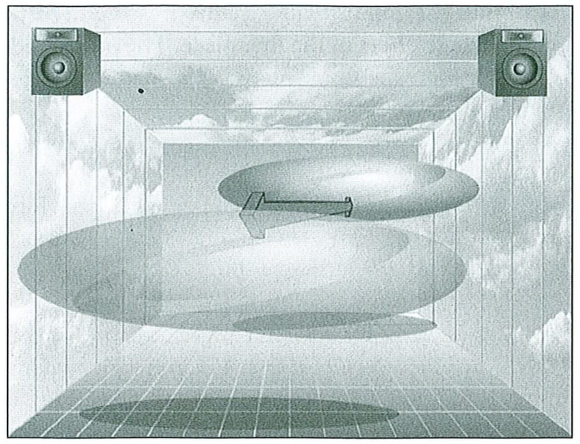
Flanging is used to create a more spacey type of mood, an other-worldly effect. It's great for making things sound like they are under water. Chorusing is often used to simulate a chorus of people or chorus of instruments. Phasing is a very subtle effect—so subtle that when used at Grateful Dead concerts, the crowd often wondered if the effect was actually coming from inside their heads.

Each of these effects can be panned in various ways:



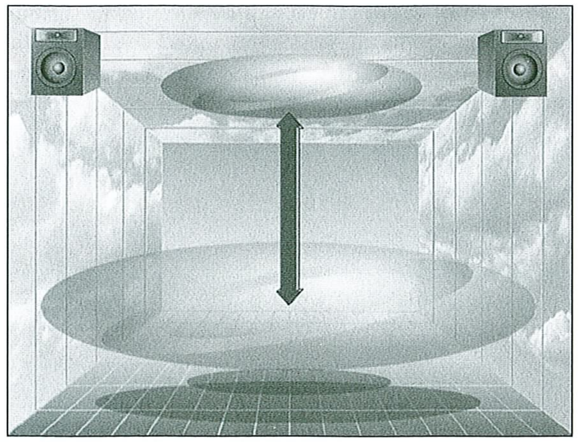
Visual 92. Flanging Panned Various Ways

Each can also be brought out front with volume . . .



Visual 93. Flanging at Different Volumes

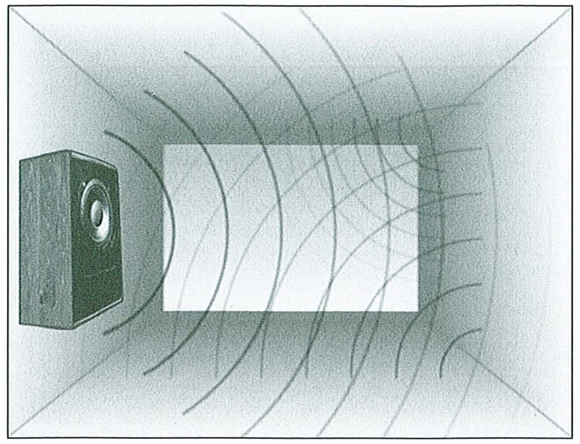
. . . and raised or lowered a little bit with EQ.



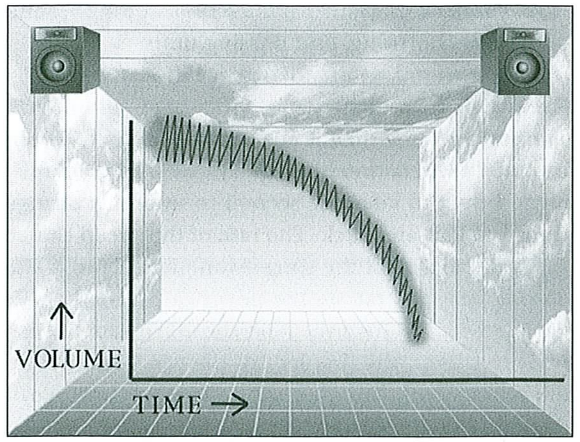
Visual 94. Flanging EQ'd Differently

REVERB

Reverb is hundreds and hundreds of delays. When a sound first occurs, it travels throughout the room at the snail's pace of around 770 miles per hour. It bounces off the walls, ceiling, and floor and comes back to us as hundreds of different delay times. All of these delay times wash together to make the sound we know as reverb.

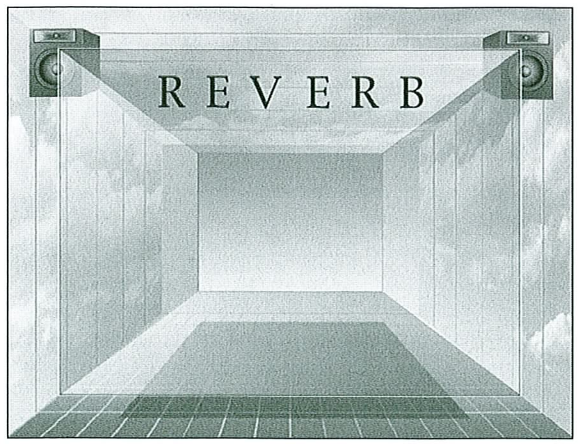


Visual 95. Waves Bouncing Around Room



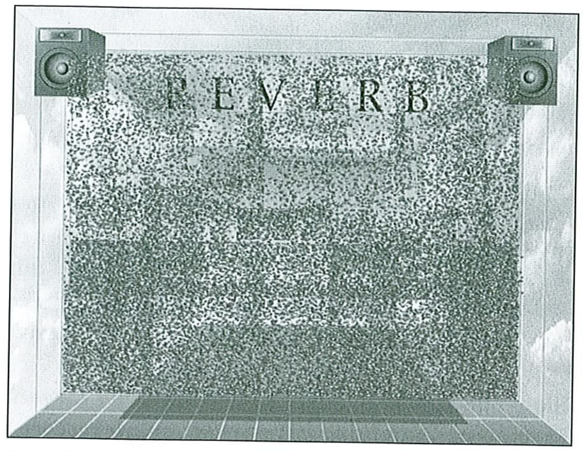
Visual 96. Wash of Reverb

When we place reverb in a mix, it is like we are placing the sound of a room between the speakers. Therefore, I will show reverb visually as a room or cube between the speakers.



Visual 97. Virtual Mixer Reverb

Reverb takes up a tremendous amount of room in this limited space between the speakers. In a digital reverb, all of these delays are panned to virtually hundreds of different places between the speakers. This is why reverb masks other sounds so much in the mix.



Visual 98. Reverb: Hundreds of Delays Panned Between Speakers

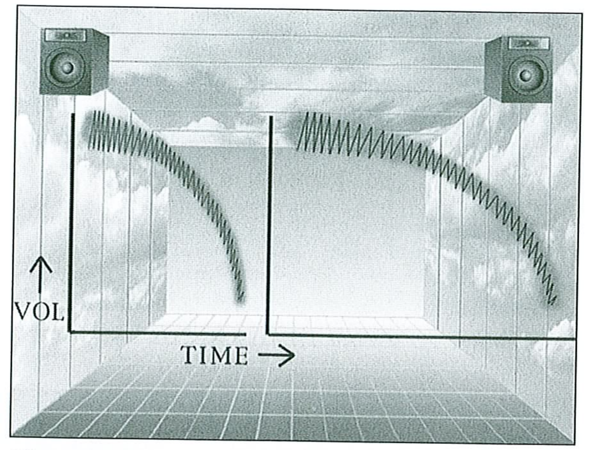
There are certain parameters of control found in units that create reverb. I will explain each setting and show it visually.

Room Types

Modern digital reverbs allow the user to change the "type of room." Imagine different types of rooms between the speakers. There are no strict rules as to the type of room that is used in a mix. Some engineers prefer a plate reverb sound on the snare drum. Some use hall reverbs on saxophones. However, it is important to check the type of reverb while in the mix (with all the sounds on) to make sure it cuts through the mix like you want it to because different types of sounds mask the reverb in different ways.

Reverb Time

You can also change reverb time: the duration or length of time it lasts.

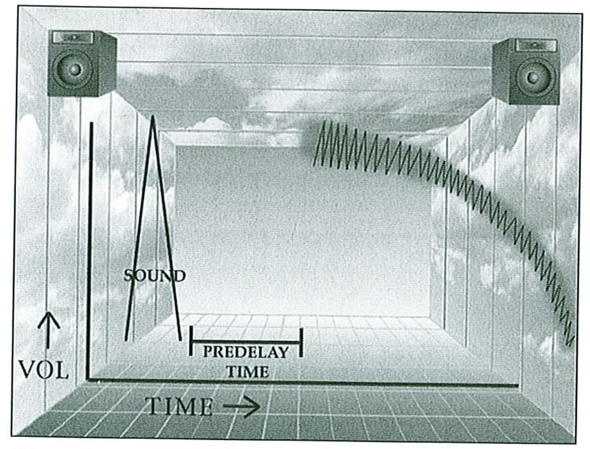


Visual 99. Long and Short Reverb Times

NOTE: A common rule is to set the reverb time on a snare drum so that it ends before the next kick lick; this way, the snare reverb does not obscure the attack of the next kick note, which will keep the kick drum sounding clean, punchy, and tight. The faster the tempo of a piece, the shorter the reverb time. Again though, rules are made to be broken.

Predelay Time

When a sound occurs, it takes awhile for the sound to reach the walls and come back. The time of silence before the reverb begins is called the predelay time.



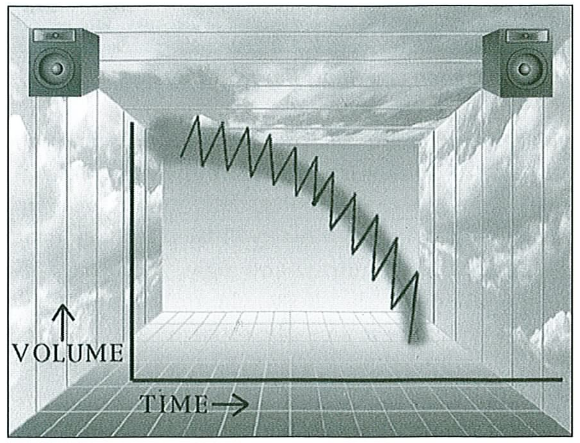
Visual 100. Predelay Time

Different sized rooms will naturally have different predelay times. A medium-sized auditorium has around 30ms of predelay time, while a coliseum might have as much as 100ms of predelay time. Therefore, it is important to have a bit of predelay time if you are looking for a truly natural reverb sound. Most times, when you call up a preset in a reverb unit, someone has already programmed a predelay time. You can adjust this as desired.

The cool thing about longer predelay times (over 60ms or so) is that they help to separate the reverb from the dry sound. With shorter predelay times, reverb will very quickly "mush up" the original dry sound, making it unclear. With longer predelay times, a vocal, for example, will remain clean and clear even with a good amount of reverb.

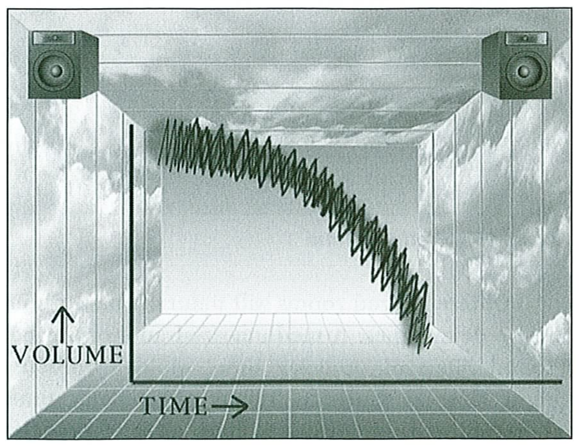
Diffusion

In most effect units, diffusion is the density of the echoes that makes up the reverb. Low diffusion has less echoes.



Visual 101. Low-Diffusion Reverb

You can actually hear the individual echoes in a low-diffusion setting. It sounds kind of like "wil, il, il, il, il, bur, bur, bur, bur, bur, bur." A hall reverb setting is preset with a very low-diffusion setting. High diffusion has more echoes—so many that they meld together into an extremely smooth wash of reverb. Plate reverbs often have a very high-diffusion preset.



Visual 102. High-Diffusion Reverb

There are no strict rules for the use of high- or low-diffusion settings. Some engineers prefer a low-diffusion setting on a snare drum to make it sound more raucous for rock 'n' roll. High diffusion is often used to make vocals sound smoother.

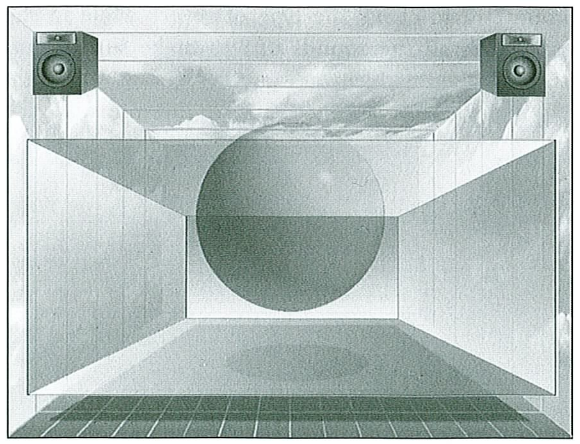
EQ of Reverb

You can equalize reverb at various points in the signal path. First, you can EQ the reverb after the signal comes back into the board (if you are using channels for your reverb returns that have EQ on them). It is usually better to use the EQ in the reverb unit itself. Not because it is necessarily a better EQ, but because in some units you can place the EQ before or after the reverb. Ideally, it is best to EQ the signal going to the reverb. If your reverb unit does not have this capability, you can actually patch in an EQ, after the master auxiliary send, on the way to the reverb unit.

High- and Low-Frequency Reverb Time

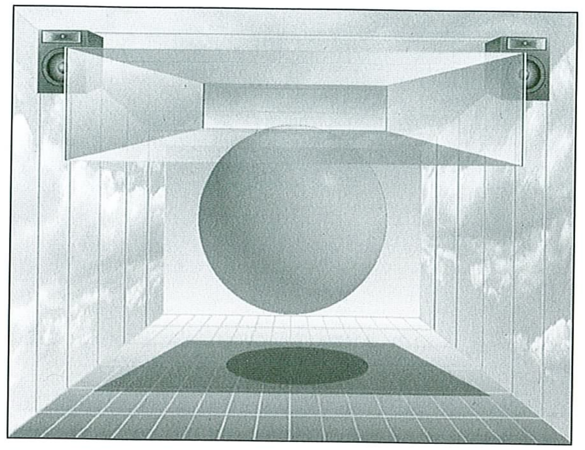
Even better than using EQ on your reverb is setting the duration of the highs and lows. Many reverb units have this setting these days. This is a bit different than EQ, which changes the volume of the frequencies. High- and low-frequency reverb time changes the duration of the frequencies. Using these settings will make the reverb sound more natural than any type of EQ.

Regardless of whether you EQ your reverb or set the duration, there is a huge difference as to how much space it takes up in the mix—and the resulting masking it creates. Reverb with a low-frequency EQ boost takes up an enormous amount of space in a mix . . .



Visual 103. Reverb With Low-Frequency EQ Boost

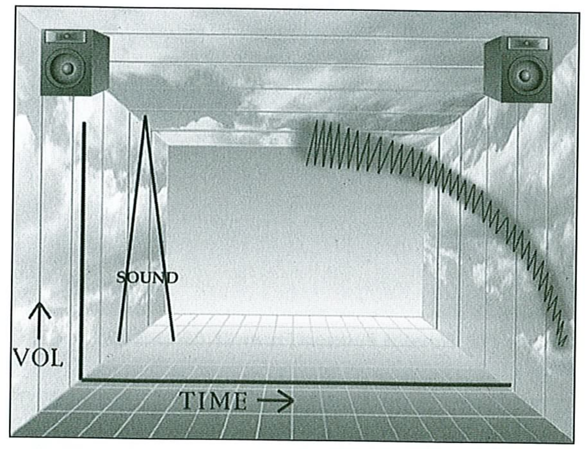
. . . compared to reverb with a high-frequency EQ boost.



Visual 104. Reverb With High-Frequency EQ Boost

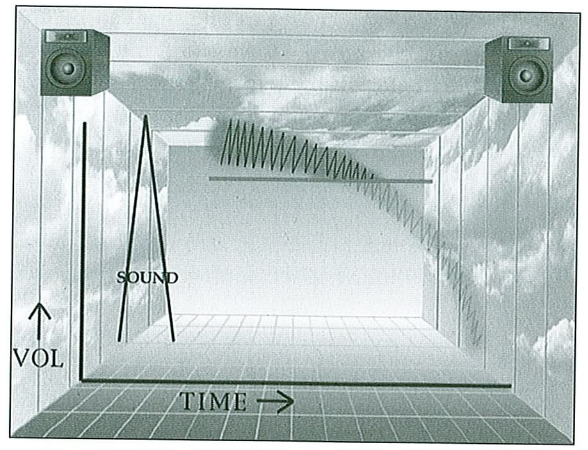
Reverb Envelope

Another setting of reverb is the "envelope"; that is, how the reverb changes its volume over time. Normal reverb has an envelope where the volume fades out smoothly over time.



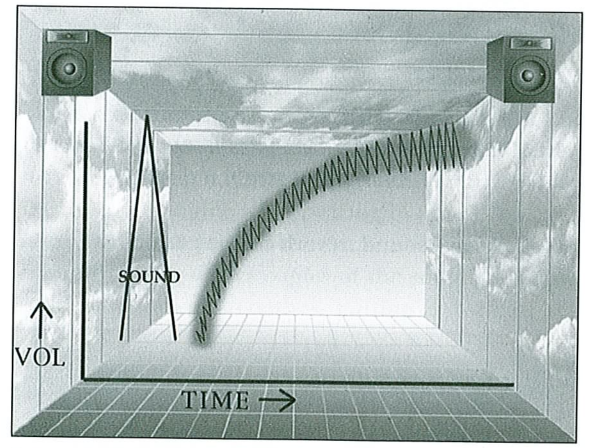
Visual 105. Envelope (Change in Volume Over Time) of Normal Reverb

Engineers thought to put a noise gate on this natural reverb, which chops it off before the volume has a chance to fade out. Therefore, volume stays even then stops abruptly.



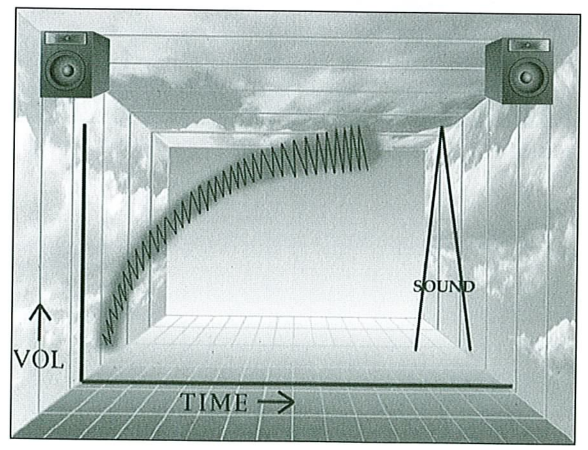
Visual 106. Envelope of Gated Reverb

But it's simpler to use the gated reverb setting on your effects unit. If we turn the envelope of normal reverb backward, reverb volume rises then stops abruptly.



Visual 107. Envelope of Reverse Gate Reverb

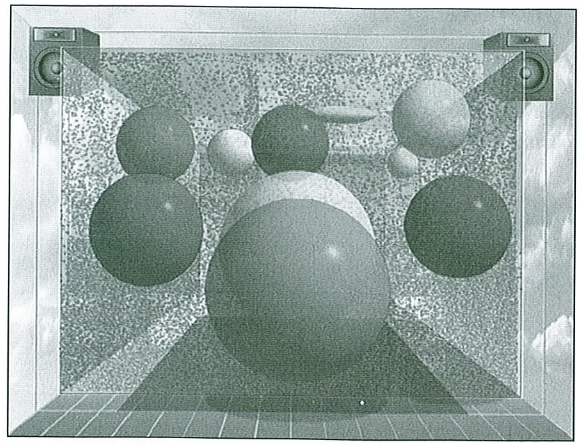
If you take the tape, play it backward, add normal reverb, record it on open tracks on the multitrack, and turn the tape around to run forward, you'd get preverb.



Visual 108. Preverb

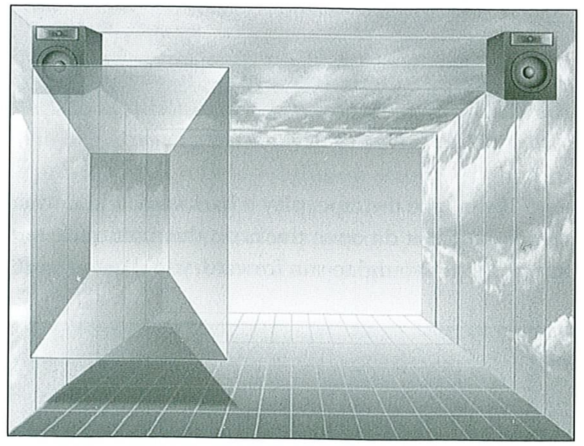
This effect is the most evil one that can be created in the studio; only the devil could put an effect on something before it happens. Furthermore, it has been used in every scary movie made, including *The Exorcist* and *Poltergeist*. And, of course, it is Ozzy Osbourne's favorite effect.

One of reverb's main functions is to connect sounds in a mix and fill in the space between the speakers:

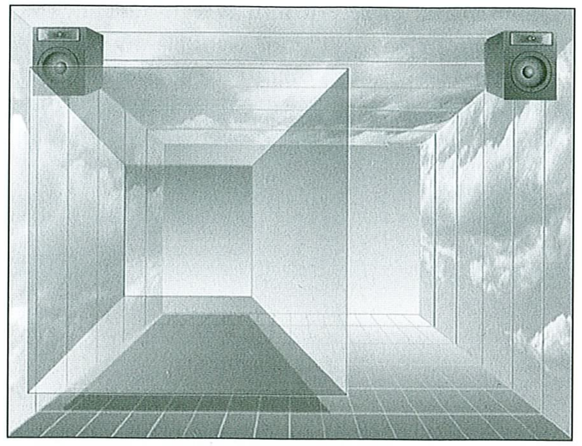


Visual 109. Reverb Filling in Space Between Speakers (see color Visual 109C)

Like any sound, reverb can be panned in various ways:

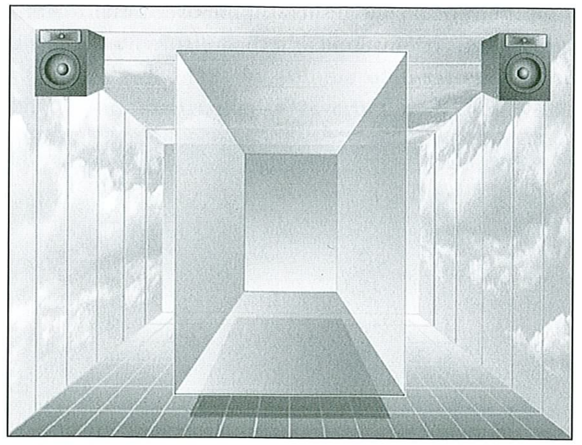


Visual 110. Reverb Panned to Left

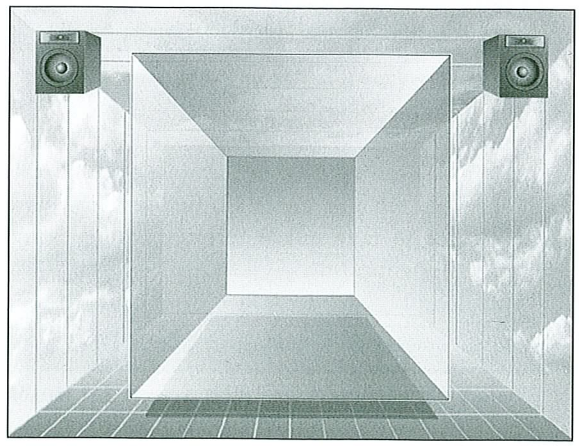


Visual 111. Reverb Panned From Left to 1:00

Just as sounds can be moved left and right with panpots, reverb can be placed left and right between the speakers. Similarly, reverb can be spread to any width.

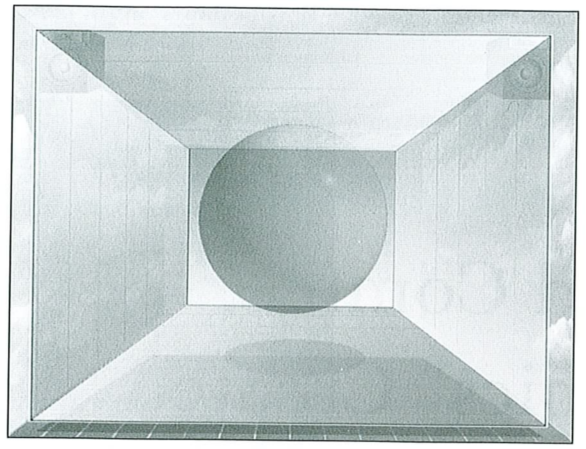


Visual 112. Reverb Panned From 11:00 to 1:00



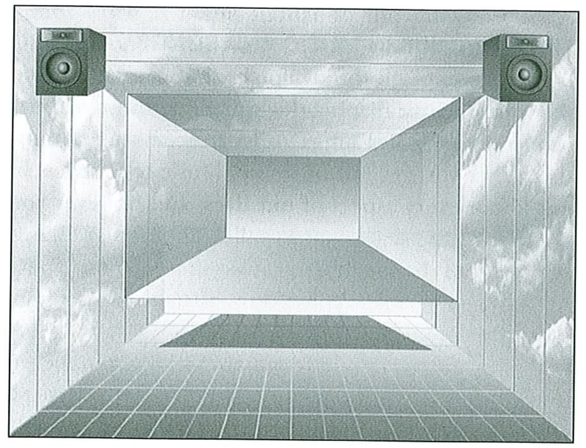
Visual 113. Reverb Panned From 10:00 to 2:00

Reverb can also be brought out front with volume . . .



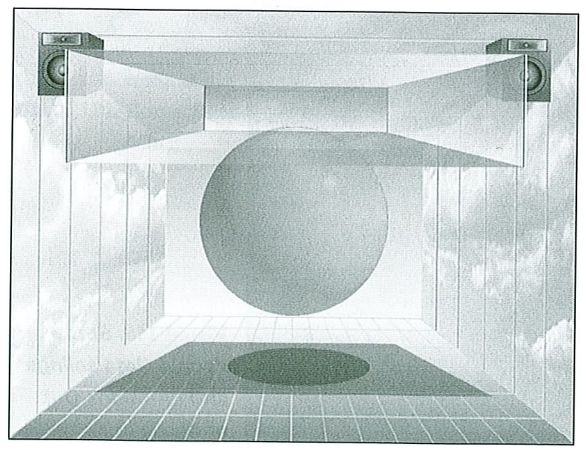
Visual 114. Reverb Turned Up in Mix

. . . placed in the background by turning down the volume . . .

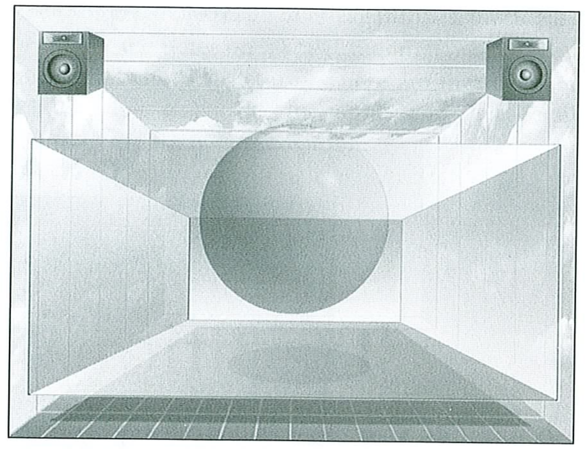


Visual 115. Reverb Turned Down in Mix

. . . or raised or lowered a bit with EQ.



Visual 116. Reverb With High-Frequency EQ Boost



Visual 117. Reverb With Low-Frequency EQ Boost

HARMONY PROCESSORS, PITCH TRANSPOSERS, OCTAVERS

A harmony processor (harmonizer, pitch transposer, octaver) raises or lowers the pitch and then puts it back in time. Usually, when you raise or lower the pitch of a sound, the duration of the sound is either shortened or lengthened. A harmonizer takes a longer, lowered pitch; deletes tiny slivers of sound (individual samples); and then splices it back together to keep it in time. (This means you can have Darth Vader singing a happy song in time.) A harmonizing unit also takes a shorter sound that has been raised in pitch, makes copies of the sound, and then splices them back together, putting it back in real time. Therefore, you can have the Chipmunks singing the blues in time with the rest of the band. Often, on cheaper harmony processors, you can even hear the "glitches" where the sounds have been spliced back together to put them in time.

When you raise or lower the pitch of a sound, it directly affects the amount of space it takes up. The higher the pitch, the less space the sound takes up.

Each and every effect has its own world of feelings that it brings to a mix. The trick is to get to know the feeling it gives you.