CHAPTER 8

Equalization Techniques

I. Introduction

Interaction

Mastering is the art of compromise. It is the art of knowing what is sonically possible, and then making informed decisions about what is most important for the music. The first principle of mastering is this: *Every action affects everything else.* This principle means that we cannot just import practices from elsewhere into the mastering room. Equalization practice is an especially clear case of where a technique used in mastering is crucially different from an apparently similar technique used in mixing. For example, when mastering, adjusting the low bass of a stereo mix will affect the perception of the extreme highs. Similarly, if a snare drum sounds dull but the vocal sounds good, then nine times out of ten, the voice will suffer when you try to equalize for the snare.[1] These problems occur even between elements in the same frequency range: when you work on the bass drum, for example, the bass guitar will more than likely be affected, sometimes for the better, sometimes worse. If the bass drum needs EQ but the bass instrument is correct, it may be possible with careful, selective equalization to "get under the bass" at the fundamental of the drum, somewhere under 60 Hz. But just as often a bass drum exhibits problems in its harmonics, which overlap with the range of the bass instrument. A resonance problem in the bass instrument may be counteracted by

"Mastering is the art of compromise"

dipping around 80, 90, 100 Hz ... but this can easily affect the low end of the vocal or the piano or the guitar. Sometimes we can't tell if a problem can be fixed until we try. We should never promise a client miracles—that way they're delirious when we can deliver them!

II. What is a Good Tonal Balance?

Perhaps the prime reason clients come to us is to verify and obtain an accurate tonal balance. The output of the major mastering studios is remarkably consistent, pointing to their very accurate monitoring. While it is possible to help certain individual instruments, most of the time our goal is to produce a good spectral balance. But exactly what is a "good" tonal balance? The ear fancies the tonality of a symphony orchestra. On a spectrum analyser, the symphony always shows a gradual high frequency rolloff, and so will most good pop music masters. The amount of this rolloff varies considerably depending on the musical style and even the moment in the music, so mastering engineers rarely* use the spectrum analyser display to make EQ judgments.

> *"Practice is the best of all instructions"*
> — Chinese Fortune Cookie

Everything starts with the midrange. If the mid-frequency range is lacking in a rock recording, it's just like leaving the violas or the woodwinds out of the symphony. The fundamentals of the vocal, guitar, piano and other instruments must be correct, or nothing else can be made right. The mastering engineer's job is to make sure that the tonal balance is well within the acceptable range, that things don't *stick out* inappropriately, that the sound is pleasant, warm and clear, and is correct for the song and the genre. Some pieces of music require laid-back cymbals, others are just crying out for an *in your face* treatment; with the right monitors and experience it is possible to know that the EQ is just right.

While we always seek an absolute standard in EQ, a recording can have an intentional color, for example, a brighter, thinner sound, and the ear will "train" itself and learn to accept a slight deviation from neutral.[2] Once the ear has been "trained," if you throw a naturally EQ'd song in the middle of this, it will seem fat and muddy by comparison. The mastering engineer is there to ensure that the deviation from neutral is not excessive because if it is then the sound will not translate adequately on the widest variety of playback systems. We must recognize when a sibilant vocal is acceptable, or must be controlled, for esthetic and technical reasons.[3]

Specialized Music Genres

I try to keep the symphonic tonal balance in my head as a basic reference for most rock, pop, jazz, world music, and folk music, especially in the mid to high frequency balance. This works most of the time. But some specialized music genres deliberately utilize very different frequency balances, and for them the *symphony ideal* is not appropriate. For example, in some styles of music,

* We don't use the spectrum analyser to judge musical balance, but it's useful to have around to reveal problems, e.g. identify noises at discrete frequencies or ultra high or low frequency noise.

'too much' (or 'too little') bass is just right. You could think of Reggae as a symphony with lots more bass instruments whereas punk rock is often extremely aggressive, thin, loud and bright. Punk voices can be thin and tinny over a fat musical background, with the natural fundamental-harmonic relationships completely strained. When this is done for a whole record it can be fatiguing, but it can be interesting and musically special when it's part of the artistic variety of the record.*

Be aware of the intentions of the mix

Equalization (and other processing) affects more than just tonality—it can affect the internal balance of a mix. So a good mastering engineer must be capable of evaluating the mix intentions of the producer/engineer/musicians and be sensitive to the needs of the production team. We must not unintentionally alter carefully-constructed instrumental interrelationships. For example, raising the bass level to get a warmer tonality will inevitably raise the level of, say, the bass instrument compared to, say, the vocalist. Sometimes this is exactly what the producer intended, because it is possible that the lack of warmth will be traced to a monitoring issue in the mix environment, and the same issues that caused a lack of warmth could also be reducing the bass instrument level on an absolute basis. Regardless, when I feel that I am affecting a balance, I always discuss my feelings with the producer to make sure that the balance "fault" which I perceive was not intentional.

* Yes, there are artistic punk rock records! I believe that the musical integrity of the artist determines the worth of a recording, not the style they work in.

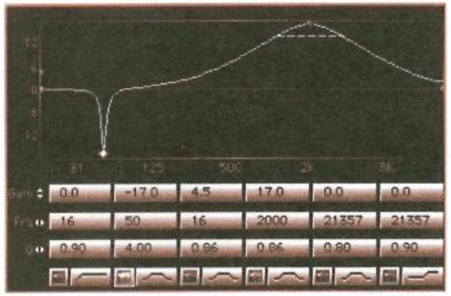
III. Equalization Techniques

Parametric Equalizers

There are two basic types of equalizers — **parametric** and **shelving** — named for the shape of their characteristic curve. Parametric EQ is favoured in recording and mixing. Invented by George Massenburg circa 1967[4], the parametric is the most flexible curve, providing three controls: center frequency, bandwidth, and level of boost or cut. Mix engineers like to use parametrics on individual instruments, either boosting to bring out their clarity or salient characteristic, or selectively dipping to eliminate problems, or by virtue of the dip, to exaggerate the other ranges. The parametric is also the most popular equalizer in mastering since it can be used surgically to remove certain defects, such as overly-resonant bass instruments. A simpler (non-parametric) equalizer has fixed frequency and bandwidth and only the level is adjustable per band.

Q's and Bandwidth

Equalizer Q is defined mathematically as the product of the center frequency divided by the bandwidth in Hertz at the 3 dB down (up) points measured from the peak (dip) of the curve. A low Q means a high bandwidth, and vice versa. The first figure on the next page shows two parametric equalizers with extreme levels for purposes of illustration: On the left, a 17 dB cut at 50 Hz with a very narrow Q of 4, which is 0.36 octaves. The bandwidth is 12.5 Hz. On the right, a 17 dB boost centered at 2 kHz, with a fairly wide (gentle) Q of 0.86, which is 1.6 octaves. The bandwidth is 2325



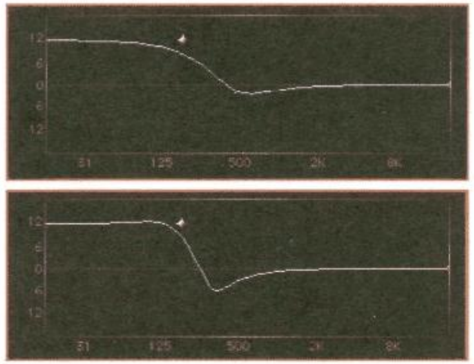
Parametric equalizer with +17 dB boost centered at 2 kHz with a fairly wide bandwidth of 1.60 oct (Q = 0.86), indicated by the dashed white line at the 3 dB down points. A cut of –17 dB at 50 Hz with a very narrow bandwidth of 0.36 octaves (Q = 4).

Hz, represented by the dashed white line.*

The choice of high or low Q depends on the situation. Gentle equalizer slopes almost always sound more natural than sharp ones, so Q's of 0.6 and 0.7 are therefore very popular. Use the higher (sharper) Q's (greater than 2) when you need to be surgical, such as dealing with narrow-band resonances or discrete-frequency noises. It is possible to work on just one note with a sufficiently narrow-band equalizer. I also use higher Q's when I want to emphasize an instrument with minimal effect on another instrument. For example, a poorly-mixed program may have a very weak bass instrument; boosting the bass circa 80 Hz may help the bass instrument but muddy the vocal, in which case I narrow the bandwidth of the bass boost until it stops affecting the vocal. The classic technique for finding a resonance is to **focus the equalizer:** start with a large boost (instead of a cut) to exaggerate the unwanted resonance, and fairly wide (low value) Q, then sweep through the frequencies until the resonance is most exaggerated, then narrow the Q to be surgical, and finally, dip the EQ the amount desired.

Shelving Equalizers

A shelving equalizer affects the level of the entire low frequency or high frequency range below or above a specified frequency. For example, a 1.5 kHz high shelf affects all the frequencies above 1.5 kHz. In mastering, **shelving equalizers** take on an increased role, because we're dealing with overall program material. One interesting variant on the standard shelf shape can be found in the Waves Renaissance EQ and Manley's Massive Passive, very useful mastering equalizers. This *resonant shelf* is based on research from psychoacoustician Michael Gerzon, who believed it to be a very desirable shape. I like to think of it as a combination of a shelving boost and a parametric dip (or vice versa). In the top figure, a low Q (0.71) bass shelf of 11.7 dB below 178 Hz is mollified by a gentle parametric dip above 178 Hz, all controlled by a single band of the equalizer. This is an extreme boost for illustration, but this type of curve can be useful to keep a vocal from sounding thick while implementing a bass boost.



Top: Gerzon resonant shelf with a low Q. Bottom: The same with a high Q. The dip just past the shelving boost frequency is characteristic of the Gerzon resonant shelf.

The bottom figure shows the same boost with a high Q of 1.41.

Shelving equalizers can have low or high Q, with Q defined as the slope of the shelf at its 3 dB up or down point.

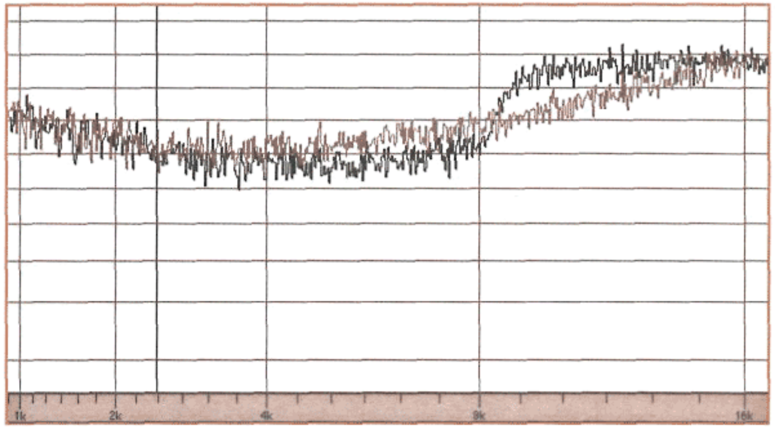
Using Baxandall for air

As I mentioned in Chapter 3, the *air band* is the range of frequencies between about 15-20 kHz, the

* Many equalizers define bandwidth in octaves instead of Q. Appendix 6 contains a convenient table for converting between Q and bandwidth.

highest frequencies we can hear. An accurate monitoring system will indicate whether these frequencies need help. An *air boost* is contraindicated if it makes the sound harsh or unintentionally brings instruments like the cymbals forward in the depth picture. Very few people know of a third and important curve that's extremely useful in mastering: the Baxandall curve, named after Peter Baxandall (pictured at right). Hi-Fi tone controls are usually modelled around the Baxandall curve. Like shelving equalizers, a Baxandall curve is applied to low or high frequency boost/cuts. Instead of reaching a plateau (shelf), the Baxandall continues to rise (or dip, if cutting instead of boosting). Think of the spread wings of a butterfly, but with a gentle curve applied. You can simulate a Baxandall high frequency boost by placing a parametric equalizer (Q= approximately 1) at the high-frequency limit (approximately 20 kHz). The portion of the bell curve above 20 k is ignored, and the result is a gradual rise starting at about 10 k and reaching its extreme at 20 k (see fig). This shape often corresponds better to the ear's desires than any standard shelf and a Baxandall high frequency boost makes a great *air eq*.

Be careful when making high frequency boosts (adding *sparklies*). They are initially seductive, but can easily become fatiguing. In addition, the ear often treats a high frequency boost as a thinning of the lower midrange, which completely changes intended program balance or the mix that was intended. The highs come up, but for example, the cymbals, triangle and tambourine also become louder. Is this consonant with the musical intent? In

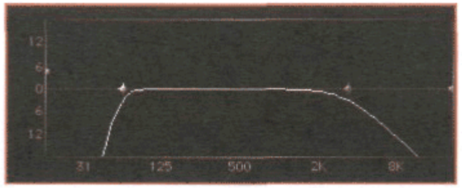


Gentle Baxandall curve (pink) vs. sharp Q shelf (black). Many shelving equalizers have gentler curves and may approach the shape of the Baxandall. Try a shelf with 3 dB per octave slope for this purpose.

accordance with the first principle of mastering, you must pay attention to the instrumental and vocal balance as well as the tonal balance whenever making changes in any EQ range.

High-Pass and Low-Pass Filters

On the left of the figure on the next page is a sharp high-pass (low cut) filter at 61 Hz, and on the right, a gentle low-pass (high cut) filter at 3364 Hz. The frequencies are defined as the points where the filter is 3 dB down. High-pass and low-pass filters are used to solve noise problems in mastering but they can make their own problems as we shall soon see. They're hard to use surgically because they affect everything above or below a certain frequency. High-pass filters are used to reduce rumble, thumps, p-pops and other noises. Low-pass filters are sometimes used to reduce hiss, though since the ear is most sensitive to hiss in the 3 kHz range, a parametric dip may be more surgical than the radical pass-filter solution. I rarely apply a



At left: Sharp high-pass filter at 61 Hz. At right: Gentle low pass filter at 3364 Hz.

standard filter to reduce hiss except for short passages, preferring specialized noise-reduction solutions instead (see Chapter 12).

EQ Yin and Yang

Remember the yin and the yang: Contrasting ranges have an interactive effect. For example...

- A slight dip in the lower midrange (~250 Hz) can have a similar effect to a boost in the presence range (~5 kHz).
- Adding bass will make the highs seem duller and reducing bass will make the sound seem brighter.
- Adding extreme highs between 15-20 kHz will make the sound seem thinner in the bass/lower midrange.
- Warming up a vocal will reduce its presence.

> *"Remember the yin and the yang: Contrasting ranges have an interactive effect"*

Yin and yang considerations imply that you are likely to be working in two contrasting ranges at once to assure that the sound is both warm and clear. Harness the yin and yang when the level is too high—pick the frequency band which you can reduce in level. Harshness in the upper midrange/lower highs can be combated in several ways. For example, a harsh-sounding trumpet-section can be improved by dipping around 6-8 kHz, and/or by boosting circa 250 Hz. Either way produces a warmer (sweeter) presentation, and your choice of which frequency range to work on will be influenced partly by what other instruments are playing at the same time as the trumpets. The next trick is how to restore the sense of *air* which can be lost by even a 1/2 dB cut at 7 kHz, and this can often be accomplished by raising the 15 to 20 kHz range, often only 1/4 dB can do the trick.[5] Never forget the first principle; it's easy to fall into the trap of concentrating on one element while forgetting how it is affecting the rest.

One channel or both (all)?

Most times making the same EQ adjustment in both (all) channels is the best way to proceed as it maintains the stereo (surround) balance and the relative phase between channels. But sometimes it is essential to be able to alter only one channel's EQ. For example, with a too-bright high-hat on the right side, a good vocal in the middle and proper crash cymbal on the left, the best solution is to work on the right channel's high frequencies.

Start subtly first

Sometimes important instruments need help, though, ideally, they should have been fixed in the mix. The best repair approach is to start subtly and advance to severity only if subtlety doesn't work. For example, if the piano solo is weak, we try to make the changes surgically:

- only during the solo
- only on the channel where the piano is primarily located, if that sounds less obtrusive
- only in the frequency range(s) that help, fundamental, harmonic, or both

- only as a last resort by raising the entire level, because a keen ear may notice a change when the gain is brought up

Realize the limitations of the recording

There is only so much that can be accomplished in the mastering and waiting until the mastering stage to fix certain problems usually produces compromise. There is little we can do to fix a recording where one instrument or voice requires one type of equalization and the rest requires another.* For example, rolling off the low end to correct a heavy synth bass is sure to lose the punch of the bass drum. Or brightening a vocal can make the tambourine sound fatiguing. In these cases I often recommend a remix. If a remix is not possible, then we resort to specialized techniques such as M/S equalization or multiband dynamics (compression/expansion) to bring out a weak instrument or hide another, which can produce fabulous results, sometimes indistinguishable from a remix (we explain M/S in Chapter 13). But the better the mix we get, the better the master we can make, which implies that a perfect mix needs no mastering at all! Even so, it is worth the time to get the approval of an experienced mastering engineer working in a neutral monitoring environment, even if she decides that no mastering or polishing is needed.

Instant A/B's?

With good monitoring, equalization changes of less than 1/2 dB are audible. I believe that instant A/B comparisons deceivingly hide the fact that a subtle change has been made, as the change will only be noticed over time.† I will take an equalizer in and out to confirm initial settings, but **I never make instant EQ judgments**. Music is so fluid from moment to moment that changes in the music will be confused with EQ changes. I usually play a passage for a reasonable time with setting "A" (sometimes 30 seconds, sometimes several minutes), then play it again with setting "B." Or, I play a continuous passage, listening to "A" for a reasonable time before switching to "B." For example, over time it will become clear whether a subtle high frequency boost is helping or hurting the music.

"The perfect mix may need no mastering at all!"

Fundamental or Harmonic?

The extreme treble range mostly contains instrumental harmonics. Surprisingly, the fundamental of some crash cymbals can be as low as 1.5 kHz or below. When equalizing or processing bass frequencies, it is easy to confuse the fundamental with the second harmonic. The detail shot of a SpectraFoo™ Spectragram in **Color Plate** ***Figure C8-01*** illustrates the importance of the harmonics of a bass instrument. High amplitudes are indicated in red, descending levels in orange, yellow, green, then blue.

Notice the parallel run of the bass instrument's fundamental from 62-125 Hz and its second and third harmonics from 125-250 and up. Should we

* Bernie Grundman calls this a recording which is "not uniform," as quoted in *The Mastering Engineer's Handbook* (see Appendix 10).

† This is a fundamental part of the see-saw arguments for and against blind testing methods, something which we will not cover in this book.

equalize the bass instrument's fundamental or the harmonic? It's easy to be fooled by the octave relationship; the answer has to be determined by ear—sometimes one, the other or both. To find out which is most important, I use the focusing technique, sweeping the equalizer from the fundamental to the harmonic. But in mastering we may not have the liberty of choice, since the equalizer may simultaneously affect the bass instrument, bass drum, and the low end of the piano, guitar, vocals, etc. It might be necessary to choose the frequency which has the least effect on other instruments rather than the ideal one for the focal instrument. It's also a matter of feel; in a rhythm piece, we can forgo delicacy and make it kick with a general bass boost.*

Bass boosts can create serious problems

Since the ear is significantly less sensitive to bass energy, bass information eats up lots more power (6 to 10 dB) for equal sonic impact below about 50 Hz, and requires about 3-5 dB more between 50 and 100 Hz.[6] This means that our low frequency equalization practice may use up so much energy that it affects the loudest clean level we can give to a song. It also explains why bass instruments often have to be compressed to sound even. Historically, the high pass filter was our best friend when we made LPs, to prevent excess groove excursion and obtain more time per LP side. Digital media do not have this physical problem, but the psychoacoustic problem of the ear's low frequency insensitivity still exists.

One possible way to save "energy" is to use a fairly sharp high pass (low cut) filter somewhere below 40 Hz, which does not significantly affect the energy of the bass drum or the low notes of the bass. I do not make this decision lightly as many recordings sound better flat; the monitor system's woofers must have calibrated, extended response for this judgment. The high pass filter must be extremely transparent and have low distortion. During mastering, I listen carefully, switching a filter in and out to determine if it is helping or hurting. Sometimes a gentle filter is a better choice than a steep one, as when dealing with a boomy bass drum or bass. But subsonic energy, rumble or thumps require a steep filter to have minimal effect on the instruments. When "uncoloring" a resonance, a fairly narrow parametric filter tuned to the offending frequency is also a good choice.

Mix engineers working with limited bandwidth monitors run the risk of producing an inferior product. Subwoofers permit you to hear low frequency leakage problems that tend to muddy up the mix, for example, bass drum leakage in vocal and piano mikes. It's much better to apply selective high pass filtering during the mixing process because mastering filters will affect all the instruments in a frequency range. For example, mix engineers can usually get away with a steep 80 Hz filter on an isolated vocal, but it's extremely rare to see a mastering engineer use one on a whole mix. A mixing engineer should form an alliance with a mastering engineer, who can review her first mix and alert her to potential problems before they get to the mastering stage.

* If that's what the piece needs. I shudder to think that readers may take each recommendation in this chapter literally, and apply it to their work. Mastering engineers do not automatically equalize; we always listen and evaluate first. Many pieces leave mastering with no equalization at all.

IV. Other refinements

Linear-phase Equalizers

All current analog equalizer designs and nearly all current digital equalizers produce phase shift when boosted or cut; that is, signal delay varies with frequency and the length of the delay changes with the amount of boost or cut. Hi-Q filters produce the most phase shift. This kind of filter will always alter the musical timing and wave shape, also known as *phase distortion*. Daniel Weiss says,

> [In contrast] a particular type of digital filter, called the **Symmetric FIR Filter**, is inherently linear-phase.[7] This means that the delay induced by processing is constant across the whole spectrum, unconstrained by eq settings.*

Since FIR filters are expensive to implement in real time, linear-phase equalizers have only recently appeared. Rather than FIR filters, the Weiss uses a complementary IIR technique to obtain linear-phase. This technique seems to avoid one of the downsides of the FIR approach, which can produce weird results at certain frequencies unless they use extreme computing power (MIPS).

John Watkinson believes that much of the audible difference between EQs comes down to the phase response.† I don't think engineers have a good handle on the sonic deteriorations of phase-shift in equalizers; after my first linear-phase experience, it was hard to go back. To my ears, the linear-phase sounds more analog-like than even analog! The Weiss has a very pure tone and seems to boost and cut frequencies without introducing obvious artifacts. Ironically, while mastering a punk rock recording, it proved too *sweet* in linear-phase mode so I had to return to normal mode to give the sound some *grunge*. So clearly much of the qualities we've grown accustomed to in standard equalizers must be due to their phase shift.

Most times I choose linear-phase mode. But both filter designs have their Achilles' heels.

> Whenever you have to equalize, you will alter the signal in both the time and frequency domains (as mathematics requires); there will always be a **time artifact**. In the *analog* style equalizer, which is usually mathematically termed *minimum-phase,* the alteration will be primarily to spread the signal **downstream**, i.e. does not lead the original signal by much, if any. A **downstream modification** translates into different delays at different frequencies dispersing the original signal. In some cases this effect is quite audible. If one uses a *digital approach,* one can either mimic the analog behavior, or use a *linear-phase,* aka *constant delay* filter. This filter will **equally precede and follow the signal;** part of the filter may create a pre-echo effect, modifying the

* Described by Daniel Weiss at the Weiss website, http://www.weiss.ch.

† **Studio Sound** Magazine, 9/97.

leading edge of transients and signal changes. A high Q linear-phase filter can introduce audible pre-echo in the short millisecond range; it's exactly like a floor bounce but without the comb-filtering. Any time that a high Q filter is used, careful listening with both types of equalization may be necessary to decide which choice is best.[8]

Neither approach is fundamentally better. The minimum phase (analog-style) equalizer tends to smear the depth and imaging, and occasionally that artificial smearing produces a pleasantly vague image. The linear phase equalizer can subtly deteriorate transient response. It might be a good idea for manufacturers to allow us to select filter types per band; I might choose minimum-phase for a steep high pass, and linear phase for a gentle presence boost.

Dynamic Equalization

Multiband dynamics processing can also be treated as dynamic equalization, where the time constants or thresholds have little effect on the actual dynamics but rather more on the tonal balance at different amplitudes. Dynamic equalizers emphasize or cut low, mid or high frequencies selectively at either low levels or at high levels. These can be used as noise or hiss gates, rumble filters that only work at low levels (especially useful for traffic control in a delicate classical piece), sibilance controllers, or ambience enhancers. They can enhance inner details of high or low frequencies at low levels, where details are often lost. They can be used to reduce harshness, enhance clarity at high levels or for other purposes, as described in detail in Chapter 10.

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1 We're always seeking techniques (beyond simple equalization) to isolate one instrument from another, and it is possible to greatly improve the impact and clarity of the snare and other percussion instruments without changing the tonality of the vocal, using upward expansion with just the right attack and release times. It's frequently possible to enhance or punch a bass drum without significantly affecting the bass instrument, by using selective-frequency dynamics processing. And so on. See Chapters 10-11.

2 We all believe we have "the absolute sound" in our heads, but are surprised to learn how much tonal variance is tolerable as the ear/brain accomodates. Similarly, the eye accustoms itself to varying color temperatures, which only call attention to themselves when they change. A good photographer can usually identify Ektachrome from Kodachrome, but both look good on their own, and their color difference primarily shows up when you place two slides side by side.

3 Technically, **sibilance** can wreak havoc with the high frequency limiters in FM radio which are there to handle a preemphasis boost. An over sibilant vocal can cause the radio limiters to clamp down and lose definition, in extreme, the sound will bounce and words will be lost at the rate of the radio limiter's recovery time. Thus, overly bright records can sound dull on the air; brightness is self-defeating when it comes to radio processing.

4 In 1967, young George Massenburg began the search for a circuit which would be able to independently adjust an equalizer's gain, bandwidth and frequency. The key word is *independent*, for most analog circuits fail in this regard and the frequency, Q, and gain controls interact with each other. He called this circuit a **parametric equalizer** and his circuit remains proprietary today.

5 Moving coil cartridges sometimes have a dip in the 8 kHz range and a rise from 10 to 20 kHz, which gives them a *sweet sound*, amounting to a tone control in the reproduction system. I prefer my reproduction system to be neutral and to correct problems in the program material itself. But since a lot of older program material was equalized on lower resolution monitor systems, it makes sense to have a tone control in your home playback system.

6 This is dictated by the psychoacoustic *equal loudness curves*, first researched by Fletcher, Harvey and Munson in the 1930's.

7 **FIR** stands for Finite Inpulse Response, and **IIR** for Infinite Impulse Response. Readers interested in a detailed theoretical explanation of the difference between FIR and IIR filters should invest a little time in John Watkinson's **The Art of Digital Audio**.

8 Jim Johnston, in correspondence.