The Cure for the Common Feedback

By Dr. Robert McPeek

As with any discipline, sound engineering has its own language, shorthand, and terminology that non-initiates may find difficult to fathom. Spectral response, polar pattern, signal-to-noise ratio, total harmonic distortion, the precedence effect, Linkwitz-Riley crossovers…the list goes on and on. A conversation between two or more sound engineers can easily lapse into an exercise in esoterica guaranteed to elicit glazed expressions from those who think of sound systems (if they think of them at all) in more utilitarian terms, simply as a means of amplifying music or speech, for either practical necessity, or to enhance the listening experience.

The vast majority of listeners judge the quality of amplified sound using very basic criteria. If these criteria are satisfied, the audience is happy to remain blissfully unaware of the other yardsticks sound engineers employ in their measurements. If we put aside our propeller beanies for the moment and adopt a layman's perspective, we might suggest that the preeminent sound system concerns of a typical audience (concert hall, church, or board room) can be summarized with just three questions:

1. Is the sound loud enough?
2. Is the sound sufficiently clear to communicate the necessary information?
3. Is the sound free from unnecessary or unwanted distractions, such as buzzing, hum, distortion, or feedback?

Of the myriad problems and obstacles encountered in attempting to satisfy these three criteria, feedback—the oscillating scream of amplified frequencies rampaging out of control—is arguably the most disastrous. Feedback impacts all three audience concerns. First, of course, it is a huge, glaring, and painful distraction of sufficient magnitude to have warranted the firing of more than one sound engineer victimized by its sudden appearance. Second, it obscures clarity—covering up frequency bands and diverting the attention of sound engineers away from aesthetic enhancement and creative knob twiddling—towards problem solving and hair pulling. Third, the threat of feedback is often the limiting factor (before amplifier power and system headroom) that determines practical maximum sound system gain.

The prevention of feedback thus assumes preeminent status in the quest for perfect sound. Several well-known rules of thumb used in the design and operation of sound systems can minimize the likelihood of feedback appearing. Inevitably, however, a situation will arise in which such steps prove insufficient—as when venue limitations or performance demands require levels of gain close to or above the threshold of feedback—leaving the audience, the performer, and the sound engineer in need of additional protection. Fortunately, developments in technology (in particular, digital signal processing) over the last ten years provide an elegant solution: automatic feedback control.

Let's examine both common sense steps towards minimizing feedback potential, and the technological solutions to exterminating feedback.

FEEDBACK MINIMIZATION BY COMMON SENSE

As everyone who has ever used a microphone knows, feedback results from the formation of a "vicious circle" of amplification. Sound from a speaker is picked up by a microphone or pickup, which again is amplified and picked up, until some resonant frequency exceeds a gain threshold and begins to squeal. The common, well-known (and perhaps a few not so well-known) methods of reducing the likelihood of feedback listed below are all aimed at somehow interrupting or reducing the "loop path" that generates feedback.

1) Mic/Speaker Proximity. Since the energy of sound propagation obeys the inverse square law, the energy picked up by a microphone from a speaker is reduced to ? the amount when the microphone is moved twice as far away, 1/16 the amount when the microphone is moved 4 times further away, etc. Therefore, moving microphones as far away from speakers as is practical can help prevent feedback by diminishing the energy from speakers that is "fed back" into the microphone.

2) Mic/Speaker Orientation. Since speakers generally throw sound in focused directions, and microphones (except omni-directional microphones) generally are also more sensitive in focused directions, you may set up your system to take advantage of such directionality. House speakers usually are placed in front of stage microphones, facing away, and thus removed from the most direct path of sound radiation. Care should also be taken to avoid aiming speakers at reflective surfaces, which may bounce sound back to a microphone.

Note that unlike house speakers, monitor speakers are by necessity pointed at microphone positions, since they are designed to provide amplified sound for the benefit of the performers. For this reason, monitor speaker amplification is often more susceptible to feedback than house speakers are.

3) Mic/Speaker Selection. Different microphones and speakers may have very different frequency response curves, meaning some frequency bands may be louder for certain speakers, while some microphones may also be more sensitive to particular frequency ranges. In the extreme case, a speaker cabinet may disproportionately produce the same frequency range that a microphone emphasizes in its pickup response, and that frequency will be more susceptible to feedback. A wise choice of compatible microphones and speakers (and other components in the system as well) can reduce the likelihood of feedback. Likewise, choosing cardioid or hyper-cardioid microphones (which are more sensitive to sound in the direction in which they are aimed), as opposed to omni-directional microphones (which pick up sound equally in all directions), can also help prevent feedback through thoughtful microphone positioning. Finally, for acoustic instruments that need amplification, musicians and sound engineers will often opt for using a pickup in lieu of a microphone, which is usually less susceptible (though far from immune!) to feedback problems. Unfortunately, pickup sound quality is generally inferior to that of a microphone.

4) Acoustical Design / Treatment.

When possible, performance spaces should be designed to allow sufficient spacing of microphones and speakers to minimize feedback. Additionally, wall, floor, and ceiling surfaces should be designed so that sound reflections are unlikely to focus or concentrate in areas where microphones will be in use. Unfortunately, in most instances control of the acoustical properties of a performance venue is beyond the power of a sound engineer, or at least not an immediate possibility.

5) Speaker / Musician / Performer Education. Even less controllable in many cases is the human element, i.e., the performer using the sound system. Dangerous levels of gain may be the only option for a sound engineer trying to make audible a corporate speaker uncomfortable with sound amplification and wary of getting too close to a microphone. Even seasoned professionals may ask for unreasonable amounts of gain in monitors. Then there's the problem of potential feedback that can pay a surprise visit when someone with a wireless mic moves to an acoustical hot spot in the auditorium, or inadvertently dips the mic in front of the monitor. To date most performers have proven resistant to my attempts to reorient their stage behavior through a system of rewards and punishments—so I've put away my whips and chairs and accepted harsh reality.

So there are many instances when common sense problem solving proves either insufficient or performer-incompatible, or doesn't provide a comfortable safety margin, or allow enough gain to be heard on three continents. Fortunately, the savvy sound engineer can then resort to the next resource: electronic methods of feedback control.

FEEDBACK CONTROL THROUGH EQUALIZATION (GRAPHIC AND PARAMETRIC)

By far the most common, and probably the most effective, means of electronically minimizing the likelihood of feedback involves equalization. Like any tool, an equalizer offers the potential for misuse as well as proper application. To help understand the difference, let's briefly consider the underlying acoustical conditions and rules that lead to feedback.

UNEQUALIZATION

If equalization can help control feedback, it stands to reason that a state of "unequalization" would therefore increase the likelihood of feedback. Indeed, this is the case: feedback will occur at frequencies that are disproportionately loud in a given sound system/acoustical condition, in relationship to other, quieter frequencies across the audible spectrum. Such unequalization can derive not only from non-linear frequency response of components in the sound system, but also from the geometry of an acoustical space—its dimensions and shape—and the reflective quality of its exposed surfaces. All three play a major role in determining what frequencies are most likely to be problems.

The key to understanding this lies in the inverse relationship between wavelength and frequency. The formula that describes this relationship is as follows:

wavelength = speed of sound/frequency

The range of audible frequencies is generally recognized as spanning 20 Hz (20 complete wave cycles of air molecules per second, a very low bass tone) to 20,000 Hz. Applying the formula above, using the speed of sound = 1119 feet per second, this translates into a range of wavelengths from about 56 feet (for 20 Hz) to a little more than half an inch (for 20KHz), covering the audible frequency range.

At certain frequencies, the dimensions of a room will be exact integer multiples or fractions of corresponding wavelengths. If two or more walls or barriers face each other in a parallel relationship, and the surfaces are sufficiently reflective, the reflections for such sound waves will bounce back and forth along the same pattern as the approach path, if the angle of incidence is perpendicular to the reflecting surface. This will create what is known as a "standing wave." Depending on the position of the listener within the room, the associated frequency may sound louder or quieter, in relationship to the interference pattern of direct and reflected sound waves.

Since the relationship between wavelength and frequency is exactly inverse (one changes up in exact proportion to the other changing down), it follows that the octaves of any frequency producing a standing wave will also produce standing waves. In other words, if 56 Hz produces a standing wave in a room with a dimension of 20 feet (its wavelength), then an octave higher frequency (112 Hz) will also have standing wave problems. Instead of having one wavelength-per-room dimension, the higher frequency will execute two wavelengths over the same span. In both cases a standing wave will be produced. Similar problems may arise with additional octaves above and below 56 Hz (e.g. 28 Hz, 224 Hz, etc.), or for other frequencies in rooms of other dimensions.

All frequencies will combine in complicated phase relationships between direct and reflected sound waves, enhancing some frequencies and reducing others. The resulting spectral measurement is called a comb filter pattern, since different frequencies will be greatly amplified, and others will be diminished, producing a graph that resembles the teeth of a comb.

Those frequencies that are enhanced in the comb filter pattern are of course the ones most "excitable" as regards the potential for feedback. When there is enough source energy (in the audio program) at a problem frequency, at the right positions in the room, the potential for feedback rises accordingly.

With a measuring tape and a calculator, it is possible to mathematically derive the frequencies that are likely to be problematical in a room of a given set of dimensions. However, after considering multiple (first, second, third, etc.) reflections, absorption coefficients of wall treatments (some reflect and absorb to varying degrees), positions of sound source, microphone, speakers, and listeners, and the additional complications of reflections involving other objects in the room, the calculations can become hopelessly complex. In any event, since many people who operate sound equipment do so in blissful disregard to the mathematical relationships of dimensions and frequencies, the more common method of determining problem frequencies, and compensating for them, is by operational testing using an equalizer.

THE GRAPHIC EQUALIZER.

More often than not the equalizer of choice is a graphic equalizer, typically a 1/3-octave variety. Such a device divides the approximately 10 octaves of audible sound into three bands per octave, usually resulting in 31 bands set to fixed center points spaced a third of an octave apart. Graphic equalizers are intuitively appealing and quite simple to use—and misuse.

Misuse stems from the common misunderstanding of what "1/3-octave" actually means. The "1/3" refers to the spacing of the center points of the frequency bands, not to their widths. While the spacing of graphic EQ filters generally follows ISO standards for center points from one manufacturer to another, the width of the filter—and sometimes even the definition of how width is measured—is not so consistent. But the majority of manufacturers define filter width as the span of frequencies surrounding a filter center point that are affected when the filter fader is moved up or down. Typically the width is measured at the -3 dB point for an attenuation, and +3 dB for a boost. **Fig. 1** demonstrates filter width as defined here.

Although widths may vary from one manufacturer to another, the most common width of a 1/3-octave graphic EQ filter is an octave. If that sounds odd or disturbing, rest assured there is a good reason for making filters wider than a third of an octave. **Fig. 2** shows what

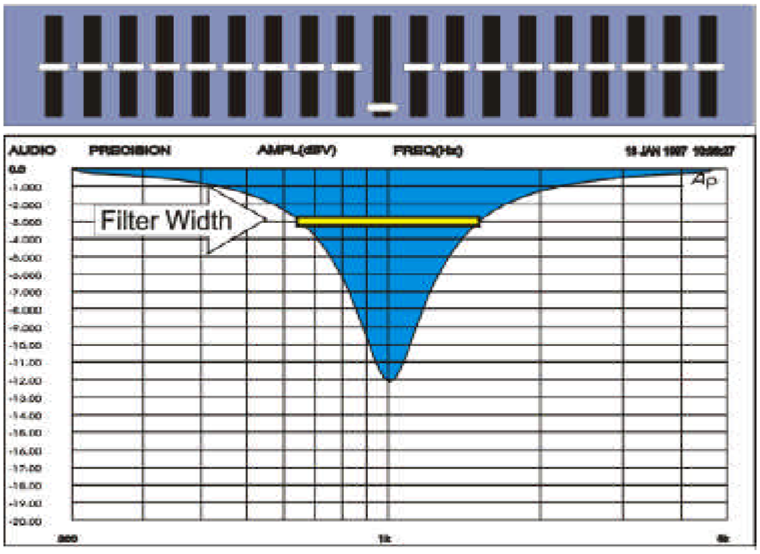

Fig. 1 Width of a one-octave-wide EQ filter.

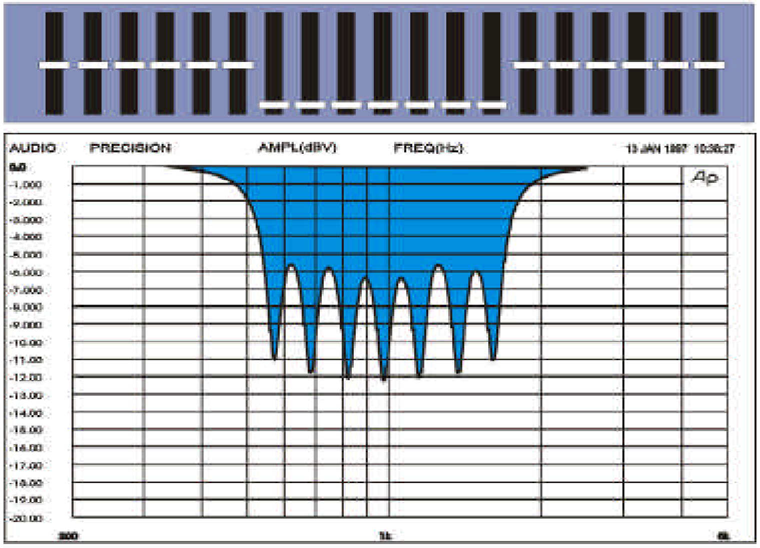

Fig. 2. Theoretical representation of graphic EQ filters that are actually a third-octave wide.

would happen if graphic EQ filters were actually 1/3-octave wide.

If adjacent filters were pulled down, the resulting curve would be an alarming roller coaster ride of variable attenuation. Because graphic EQ filters are shaped with blunt notches that widen outward, such narrow filters would be very deep at the center points, and, with little overlap from narrow adjacent filters, the fringes of such filters would be attenuated very little. For a single filter in place, this translates into accuracy and precision. However, when multiple filters are placed in the signal path, the results are disturbing, as shown in **Fig. 2**. There is a 6 dB variation in amplitude 3 times within each octave—not at all the kind of smooth response curve audio engineers aspire to create.

To achieve such desired smooth response, graphic EQ filters are widened to create greater overlap between adjacent filters. Octave wide graphic EQ filter overlap is demonstrated in **Fig. 3**.

As you can see, the resultant curve of interacting filters is much smoother; however, because there is greater frequency overlap between adjacent filters, the final result of pulling seven adjacent faders down 12 dB is an overall attenuation of 16 dB across the range of the filters. So, the control settings on the panel of a graphic EQ may not indicate the true degree of boost or attenuation—which is deemed a small price to pay in exchange for smooth frequency response.

So much for the theory. What conclusions can we draw for actual practice? Primarily one: graphic EQs are best used for smooth response shaping (such as shaping a room curve), but are NOT well suited for surgical precision. A single feedback event typically calls for such precision, since a given feedback occupies a much narrower frequency band than can be addressed by the broad resolution of a graphic EQ filter. Moreover, feedback, being the unruly problem it is, is not about to conform to the preset center points of graphic EQs, and a graphic filter can often only roughly approximate the correct frequency. The result: you can control a ringing frequency with a graphic EQ filter, but at a great cost to the integrity of the audio program. To subtract the feedback, you also have to subtract a good deal of the surrounding "good" audio frequencies. If you do this repeatedly to different feedback frequencies, you'll kill a great deal of clarity, presence, and ultimately gain in your system. We liken the process to brain surgery with a chain saw—you can remove the tumor, but the consequences to the patient are unpleasant.

THE PARAMETRIC EQUALIZER

A much better solution to controlling feedback is offered by a parametric equalizer. If a graphic EQ is a chain saw, a parametric EQ can be more like a tight laser beam. With a parametric EQ, the user can adjust the width of the filter, the center point, and the amount of boost or attenuation. A skilled user can zero right in on feedback, and perfectly tailor the width and depth of the filter to remove it with minimal consequences to the audio program. If they are narrowly targeted, many filters can be inserted in an audio program with little or no degradation in overall sound quality, affording much more gain.

Fig. 4 demonstrates the difference between filtering feedback that occurs around 1100 Hz with a graphic EQ filter and a parametric filter. The blue shaded area represents the amount of audio program information that is subtracted unnecessarily when pulling down the 1 KHz fader of a graphic equalizer.

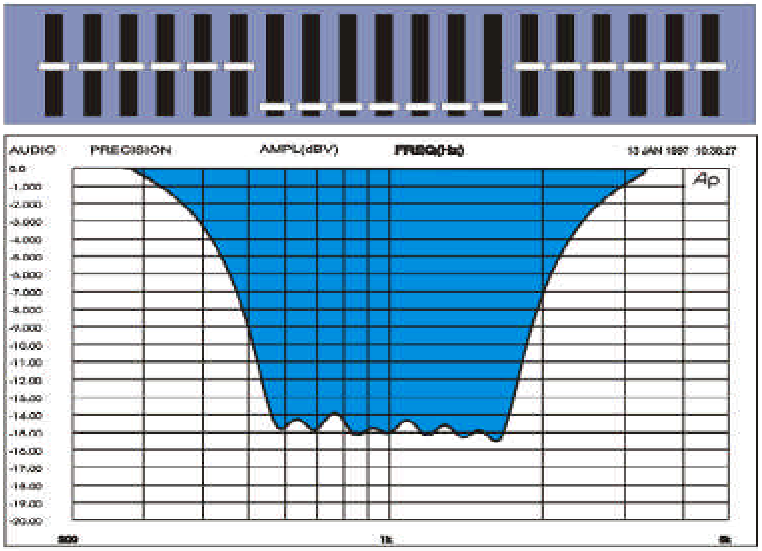

Fig. 3. Octave-wide graphic EQ filter overlap

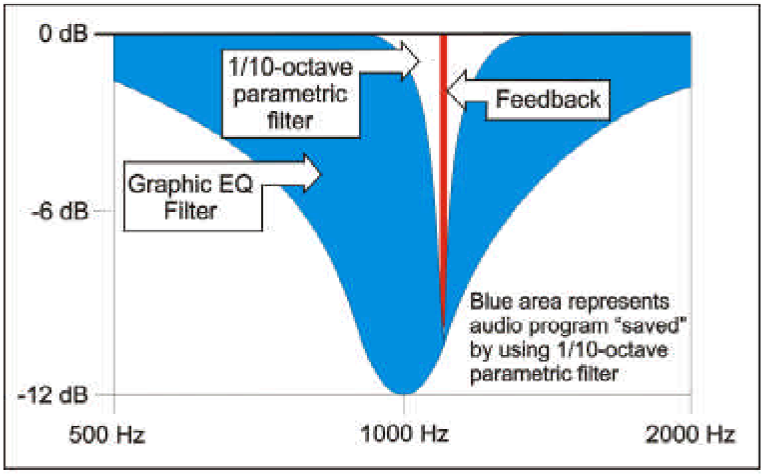

Fig. 4. Feedback filtering comparison between a parametric and a graphic EQ.

The blue area represents the much smaller amount of desirable audio that is removed by the surgical precision of a parametric filter aimed right at the offending feedback. The difference is about 90% less desirable audio removed by the parametric filter (assuming it is 1/10 of an octave wide, compared to the octave wide graphic filter).

So why doesn't everyone toss their graphic EQ in the trash and rush out and buy a new parametric EQ? The best answer is that both units (like chain saws and lasers) have their uses. A graphic EQ's wider filters are very good at broad equalization, such as setting the system response curve of a room. In fact, by using a graphic EQ in such a fashion, you are setting up the potential for the power of a parametric EQ to be used more effectively to solve any remaining narrow-filter problems. Without a graphic EQ, you would have to spend the filter power of your parametric addressing wide-filter concerns.

A second reason for the greater popularity of graphic EQs has to do with ergonomics. Setting up a parametric EQ program can be much more demanding and time-consuming than using a graphic EQ. It requires a higher degree of skill and understanding to operate, and is often a less familiar product (particularly to lay users) than the ubiquitous graphic equalizer.

FEEDBACK AUTO-CONTROL

Fortunately, if the goal of using a parametric equalizer is feedback control, there are products on the market that automate the entire process. A dedicated professional feedback controller will identify feedback the instant it occurs, differentiate feedback from music or speech, and immediately place a parametric filter at the precise frequency, with the precise amount of attenuation of the frequency needed to remove feedback. The whole process takes place instantly and automatically. With the better units available, filters can be limited in number, depth, and width to your liking, allowing you to tailor the operation and scope of the automatic filters to fit your program and requirements. Following proper but simple procedures, a good feedback controller can provide an additional 6-9 dB of system gain before feedback, enough to throw clear, crisp sound to the far reaches of an auditorium or wake up anyone sleeping in the back of the church! All of this can be done with a greatly reduced risk of waking them up with an embarrassing blast of feedback.

Such precise automatic feedback control (dubbed FBX, or feedback extermination) was pioneered by Sabine Musical Manufacturing, Inc. in 1990, and has since become standard equipment in thousands of sound systems around the world, from huge rock concert venues, to small churches and business applications. The past ten years have brought significant improvements in ease of operation and transparency of sound in automatic feedback controllers, and expanded feature sets providing additional layers and varieties of equalization. Thousands of sound engineers, from seasoned pros to unsophisticated knob twiddlers, depend on these units to serve as all-in-one problem solvers and preventers.

The two keys to how well an FBX (or other variety) automatic feedback suppressor works are, first, accurate and quick identification of feedback, and, second, transparency of operation. There are at least three methods by which different manufacturers' suppressors work:

Frequency shifting. These units actually shift the pitch of an audio signal as it passes through. Changing the frequency disrupts the feedback loop and allows additional gain before feedback. This is a very simple and effective solution to some feedback problems. However, the degree of pitch shift sufficient to effectively reduce feedback is enough to cause concern about audio program integrity, particularly for an audio program with musical content. Frequency shifters affect the entire audio program, and do not specifically target feedback. So you might eliminate feedback on an acoustic guitar microphone—but you'll also shift a D-chord sharp or flat.

The other two designs for feedback suppression DO attempt to discriminate feedback from audio program, and to remove the former without impacting the latter.

Level comparison. Since feedback is by definition a problem frequency at an out-of-control level, it's no surprise that one technique to distinguish feedback from program is based on monitoring and comparing program levels. If a certain frequency band rises in volume to a more-or-less consistent or disproportionate level, these units assume that frequency is feedback. Then, a narrow notch filter is placed at the center point of the identified frequency band, in order to reduce the gain at that frequency to a level below the feedback threshold.

This method of identifying potential feedback is not foolproof. Held vocal or instrument notes, for example, may easily be misidentified as feedback, and the unit may end up placing a false filter and reducing the volume of a performance component meant to stand out as a featured solo. There's also a danger that a needed filter might not be placed. If the volume of imbedded feedback doesn't stand out from the rest of the audio program, level comparisons may not vary sufficiently to trigger a filter placement.

Fortunately, the third method of identifying and controlling feedback offers improved discrimination of feedback and audio.

Waveform analysis. Method three is based on the fact that the waveform signature of feedback in almost all instances differs quite significantly from the more complex signature of an audio program (speech or music). With the advent of digital signal processing, audio equipment has become fast enough, and powerful enough, to perform a huge number of very rapid calculations that analyze the wave signature of sounds as they pass through this type of feedback suppressor. In a fraction of a second, the DSP is able to examine a sound and define it as feedback or audio with a high degree of accuracy. If it's feedback, the automatic notch filter if placed, centered precisely on the problem frequency, tailored to be as narrow and only as deep as necessary to eliminate feedback. The process is far faster, and much less detrimental to the sound, than a human engineer pulling down a much wider graphic EQ filter. In essence, this type of feedback controller is an intelligent, automatic parametric equalizer.

FILTER CONTROL

Feedback suppressors that work based on either the level comparison or waveform analysis methods remove feedback by placing such narrow filters. The better quality units offer a high degree of user control over the placement of these filters, which is of essential importance in achieving increased system gain without compromising the quality of your audio program. If you're considering purchasing a feedback controller, here are some important features and qualities you should look for. Remember when testing units from different manufacturers to follow the manufacturers' setup recommendations, in order to assure you are testing under recommended conditions.

Degree of control over setting filters: You should be able to adjust the number of filters that can be set, to lock filters when they are placed (so they don't change), and to tweak the threshold variables that determine when a filter is set. For example, with the Sabine POWER-Q or Sabine GRAPHI-Q, you can adjust filter placement from a very "trigger happy," fast-filter setup to a more cautious analysis, discrimination, and filter placement. You can tailor the setup to match the kind of audio program you are working with, and minimize or eliminate the likelihood of setting unnecessary filters.

A good feedback controller can provide an additional 6 to 9 dB of system gain before feedback.

Filter parameters should be adjustable. You should be able to change the depth of filters, and to adjust the width. For ultimate control, look for the ability to change a feedback filter to a parametric filter, which many units allow. This will allow you to fine tune your filters, or even combine narrow adjacent filters into a single, slightly wider parametric filter that does the work of two or more. Since all feedback controllers offer a finite number of filters, freeing up one or more with this technique will give you additional filters to remove even more feedback. The audience will be able to hear you in Cleveland.

Check how well the unit removes feedback when music is playing at the time of feedback. It's much easier to eliminate feedback when that's all there is to worry about! Testing feedback removal during an audio program will really show how good a unit is.

Also, if you can, test feedback removal at low frequencies. It's a physics reality that low feedback frequencies will take a bit longer to be recognized and suppressed. Find out how much longer.

The ultimate test is to set the full amount of filters a unit offers, then play a CD through the unit and compare the sound quality when the filters are in the signal path vs. when they are bypassed. With a good unit and just a little bit of care, you should be able to place 12 filters or more and hear very little difference. Then put the filters back in and crank up your microphone. How loud can you go before death by feedback ensues?

Common sense sound system setup, while invaluable, is often not sufficient to prevent unwanted feedback from damaging your reputation, hearing, equilibrium, and/or equipment. For an additional safety margin, try a professional feedback controller. It can make a huge difference between a nightmare fight with feedback, and a no-problem installation or event, while giving listeners what they really care about—loud, clear, distraction-free audio.

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