Version 3

Positive Feedback™

The Advantages of FBX Feedback Exterminators®

INSIDE:

SABINE®
ADAPTIVE AUDIO

The Story of Feedback

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Ever since Lee DeForest invented the first vacuum tube, engineers have walked the tightrope between feedback and system gain. The purpose of this guide is to give you the tools to get all the gain you need without the agony of feedback. We'll start with a common-sense discussion of the techniques sound engineers now use to control feedback to get the most gain and clarity out of their sound systems.

Our imaginary work bench

Imagine a mic and speakers set up in a tiny shower room. Clap your hands. The sound reverberates back and forth between the tile walls and floor. Just a touch of the volume fader fills the room with screeching feedback.

Now move our sound system out to an open grassy field. Clap your hands. There is no echo. The speakers are well away from the microphone and there are no reflections, so now we can really crank up the system without a bit of feedback.

Most sound systems have characteristics that fall between these two examples, but examining the extreme cases makes it easier to understand the more common in-between situations.

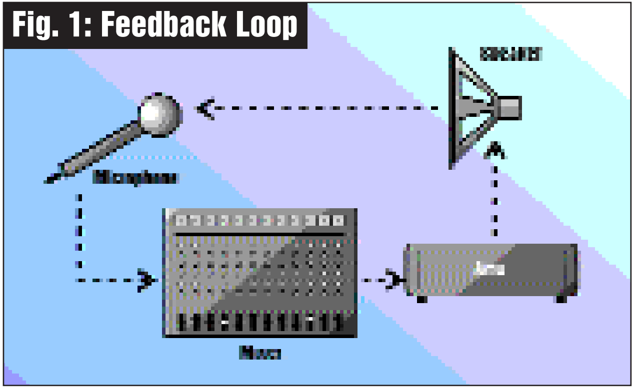


Fig. 1: Feedback Loop

What is acoustic feedback?

Feedback is the loud ringing sound that occurs when the sound leaving a speaker is picked up by a microphone and reamplified again and again. (See Fig. 1.) The cycle repeats until the feedback reaches the system's maximum loudness or until someone turns down the volume. Virtually every sound system that has a microphone and a speaker in the same room is susceptible to feedback.

Which frequencies feed back? All acoustic systems have distinct resonant frequencies. Regardless of where you thump a guitar's top, it always responds with the same tone. This is the natural resonant frequency of the guitar. It is the frequency where all of the instrument's components vibrate naturally as a unit. In sound systems, these resonant points are the frequencies where feedback occurs.

Each of the system's components, including and especially the room itself, has its own set of resonant frequencies. Each component adds together to produce the total system's resonant frequencies. It is almost impossible to predict which frequencies will feed back without first "thumping" the system, but you only have to turn up the amp for them to rudely reveal themselves.

The frequency that feeds back first is the one that requires the least amount of energy to excite the resonance. If you remove the first feedback frequency, the next feedback frequency will be the one that requires the second least amount of energy, and so on.

Controlling feedback

In order for feedback to occur, the amplifier has to be turned up enough so that sound from the speaker reenters the microphone louder than the original sound. In our imaginary experiment, feedback easily occurred in the shower room because the sound leaving the speakers did not dissipate very much before reentering the microphone. But when we move the speakers away in the open field, the sound energy dissipates as it radiates away from the speakers. If there are no surfaces to reflect the sound back to the mic, the sound quickly loses energy, dropping to one quarter the energy every time the distance from the speakers is doubled. By the time the sound finally reaches the microphone, the sound energy is weaker than the original sound, so there is no feedback. From this example we deduce the Prime Directive of Feedback Control:

Keep the sound emanating from the speakers away from the microphones as much as possible.

Here are the most common tricks of the trade for controlling feedback:

- **Stand close to the microphone.** Speak loudly and clearly so that you do not have to amplify the sound too much.
- **Each open microphone has a chance to feed back.** Mute or turn down the gain of any microphone that is not in use. Noise gates can be helpful for this.
- **Mount the microphones in fixed positions.** Moving the microphone around on the stage increases the chances that the microphone and the speaker will form new resonant paths.
- **Use cardioid or hyper-cardioid microphones, and point the mics away from the speakers.** They pick up much less sound from the back side of the mic, which protects against monitor feedback. Be careful not to put your hand on or too close to the microphone's screen, since this can cover the ports that enable the heart-shaped (hence cardioid) rejection pattern.
- **Place the speakers in front of the microphones** so there is not a direct path back to the microphone.
- **Aim the speakers so the sound does not reflect directly off a wall back into the mic.** You can estimate the speaker's dispersion pattern (the area that is directly "sprayed" with sound) for the mids and high frequencies by imagining rays of light radiating out of the speaker's horns. If you can see the center part of the horn, you are probably in the dispersion pattern. Lower frequency sounds tend to radiate out in all directions from all sides of the speakers.
- **Make the surfaces of the room as sound absorbent as possible** to reduce sound reflections. Use acoustical absorbing tiles in the ceiling, put down carpeting, and hang curtains.

In the real world of most performance spaces, you cannot always follow these anti-feedback techniques. Lead singers insist on pointing the monitors directly at the mic. Worship leaders insist on the mobility of a wireless microphone, and night club owners will not likely carpet the dance floor and hang velvet curtains. Even after you've tried all these tricks, you may still not have enough gain and clarity to satisfy the audience. Do the best you can, and then go on to the next level of feedback control: equalization.

Equalization

Equalizers (EQs) are sets of filters, or volume controls, for different parts of the audio spectrum.

Since the earliest days, sound engineers have used equalizers for two distinctly different purposes: 1) To improve the tone quality and balance of the sound, and 2) To control feedback for extra gain and microphone mobility. Some types of EQs are best at shaping the tone and other types are better at controlling feedback.

It may seem paradoxical to add filters to a sound system in order to increase the gain. But if you can use extremely narrow filters to turn down the frequencies that are feeding back, you will be able to increase the gain of all the other frequencies for a total net gain. There are essentially three categories of equalizers: graphic, parametric and adaptive parametric.

Graphic EQ

Graphic EQs are basically a set of volume controls for individual sections of the audio spectrum. The earliest music equalizers were the bass and treble tone knobs. As technology advanced, these filters were narrowed to give more precise control. Today, the industry standard is called a 1/3-octave graphic equalizer, which has 31 individual volume controls spaced 3 per octave.

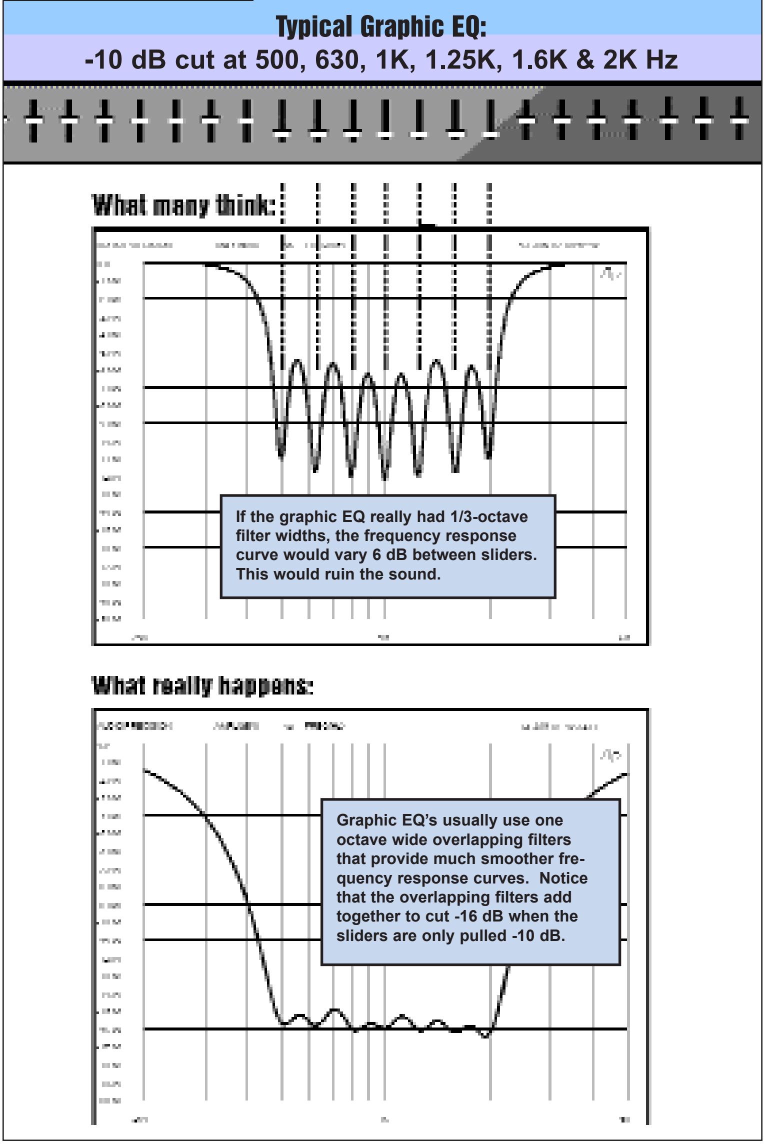
There is a common misconception in the industry about 1/3-octave EQs that is important to this discussion. Many industry veterans incorrectly presume that 1/3-octave EQs use 1/3-octave wide filters. If this were the case, the EQ filters would not be wide enough to create smooth curves. Instead, they would produce a notched frequency response that would make the EQ useless for shaping the sound and useless for controlling feedback frequencies between the sliders. Actually, most manufacturers use 3/4 to 1-octave wide overlapping filters placed on 1/3-octave center points. These wider filters provide the necessary smooth frequency response. (See Fig. 2.) **It's important to understand that the term "1/3-octave" refers to the spacing of the sliders, not the filter width.**

Graphic EQs are excellent for shaping the sound, and they are fairly simple to use. However, using one-octave wide EQ filters to control feedback invariably causes an unnecessary decrease in the gain and fidelity of the program. It's easy to see that if feedback occurs somewhere between the sliders, you will have to pull one of those EQ sliders down pretty far to eliminate feedback. That pulls out plenty of your program, too. On the other hand, you'll get considerably more net gain and much better sound quality if you use wide graphic EQ filters for tone control and insist on narrow filters for feedback control. (See Fig. 3.) That's where parametric EQs come in.

Parametric EQ

In the quest for perfect sound, engineers developed very narrow tuned filters for controlling feedback points in auditoriums. In the early days of sound reinforcement, these filters were custom made to a specific frequency and width for a specific application. Now there are a number of commercially available parametric filter sets that allow engineers to dial in the width, center frequency and depth of the filter.

Fig. 2: Graphic EQ



If the graphic EQ really had 1/3-octave filter widths, the frequency response curve would vary 6 dB between sliders. This would ruin the sound.

Graphic EQ's usually use one octave wide overlapping filters that provide much smoother frequency response curves. Notice that the overlapping filters add together to cut -16 dB when the sliders are only pulled -10 dB.

The problem with parametrics is that they're expensive, they require a good deal of expertise and auxiliary equipment to tune properly, they require constant retuning whenever the room acoustics change, and they are far too slow and cumbersome for catching feedback that occurs during the program.

Adaptive Parametric: The FBX Solution

The Sabine FBX Feedback Exterminator® is the next step in the evolution of feedback control. The FBX is essentially a self-tuning parametric EQ. It constantly monitors the program, searching for tones that have the overtone signature of feedback. Once feedback occurs, the FBX automatically places a very narrow, constant-width filter directly on the feedback frequency and lowers it just deep enough to eliminate the ringing sound.

The FBX out performs other EQs five ways:

1. The FBX finds and eliminates feedback automatically before and during the program.

2. The FBX's narrow filters eliminate feedback without losing the fidelity of the sound.

3. The FBX is fastest. It typically finds and eliminates feedback in less than one second.

4. The FBX gives the most gain. Use wide-filter graphic EQs for controlling the shape of the sound and narrow FBX filters for controlling feedback, and you'll typically achieve a 6 to 9 dB increase in gain compared with using the EQ alone.

5. Increase wireless mic mobility.

What about that 6 to 9 dB increase in gain? Gain increase from equalization really depends on the characteristics of the sound system and the room. Returning to our imaginary system in the shower room, the sound bounces off the hard tile surfaces and reflects back into the microphone with only a slight touch of the volume slider. If you filter the first feedback point, you can only increase the volume fader a touch more before the second feedback occurs at a new frequency. Even if you filter six different resonance points, you may only achieve 1 or 2 decibels of net gain because there are so many low-energy resonant paths.

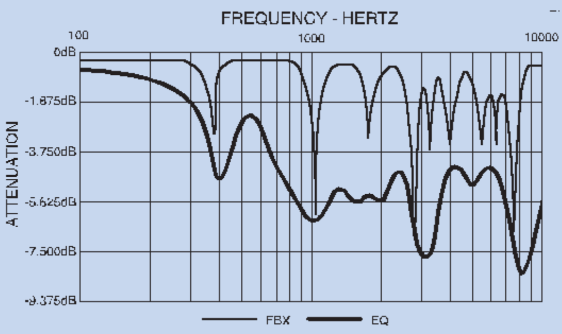
When we set our system in a large open field and the speakers are far away from the microphone, we really have to crank it up before we hear the first feedback. We would need an enormous system to drive six feedback points. In this system, damping six feedback points could easily deliver well over 15 dB net gain!

How much gain do you achieve with the six FBX filters? Six resonance points worth - whatever that happens to be in your unique system. You can maximize your gain by following our anti-feedback directives and by learning more about how the FBX filters work best for your situation.

Microphone Mobility

Mobile karaoke and wireless microphones present a special feedback challenge. It does little good to set a number of filters for a mounted microphone if you plan to carry the mic around the stage to different locations. Each position on the stage has its own unique set of resonant frequencies, so the filters that control feedback in one location will probably not provide much help in other locations.

Fig. 3: FBX vs. 1/3-Octave Graphic EQ



A PA System was set up using a microphone, mixer, FBX, power amp and two speakers. The system's gain was raised until the FBX removed nine feedback points. Next the FBX was replaced with a graphic EQ. The EQ was adjusted while the system gain was raised to the same level achieved with the FBX. The frequency response curves of each device were then plotted.

You are faced with a balancing act. If you insert too many filters in the system, you will hear a degradation of the sound quality. If you set too few filters, you will not have enough mobility or gain.

In this case, it is usually best to walk around the stage area until you find an area where feedback is a particular problem. Then place one or two feedback control filters to take care of that location and repeat the process in the next few areas. FBX filters add less gain to mobile systems than to fixed microphone systems, but they add a significant increase in the usable area while preserving the natural clear sounds.

Feedback Control During the Program

One of the most powerful features of the FBX is that it can eliminate feedback during the program. FBX filters come in two types: fixed and dynamic. Both filters are placed the same way: Feedback is detected, and the filter is placed just deep enough to eliminate it. The difference comes after the filter is placed. Fixed filters remain on the initially detected feedback tone - they do not move. These filters provide the initial maximum gain before feedback and are set automatically during setup. Dynamic filters can release and move to new feedback frequencies and are for adaptive feedback control during the performance. You can change the number of fixed vs. dynamic filters using front panel controls.

Hearing is Believing

To hear the difference for yourself, insert an FBX in your sound system and bypass it. Mount the mics on stands to fix their positions. Remove as much feedback as possible using your normal method with just the graphic EQ. Next, lower the volume, bypass the graphic EQ, and activate the FBX. Now slowly raise the gain of the system until at least six FBX filters have kicked in.

Next, turn down the mics and play your favorite CD through the system. Alternately listen to the system with just the FBX and then just the graphic EQ. You will hear the FBX provides much clearer, brighter and louder sound.

If you do not have immediate access to an FBX, run this experiment with a graphic EQ alone. You will be amazed to

GLOSSARY: Definitions of “tech” terms

What is Gain?

Gain is a measure of the change in power (or loudness) in a sound system. For example, turning up the amp causes an increase in gain, while moving away from the speakers causes a decrease in gain. By convention, gain is expressed in decibels.

ClipGuard™ Adaptive Clip Level Control

Sabine’s ClipGuard™ makes FBX feedback control faster and easier to use, and it adds about 10 dB to the effective dynamic range. Until ClipGuard, engineers manually set the input and output level controls to a compromise setting that causes unnecessary noise during quiet programs and risks clipping overload during high level programs. Now ClipGuard constantly readjusts the FBX’s electronics to match the continually changing program levels.

Another feature of ClipGuard is TURBO mode that cuts the time of the preprogram setup to just a few seconds. ClipGuard is currently a standard feature in Sabine’s FBX-1020P & 2020P Feedback Exterminators, POWER-Q ADF-4000, GRAPHI-Q, DQX-206 parametric EQ/delay and the REAL-Q_2 Real-Time Adaptive Equalizer.

Noise Gate/ Comb Filters

As we mentioned earlier, every microphone creates a potential source of feedback, so it is advantageous to turn off microphones that are not currently being used. Noise gates do this automatically by continuously monitoring the program’s loudness. If the loudness falls below a threshold set by the user, the noise gate automatically turns off the microphone. Once the loudness exceeds the threshold, the microphone channel automatically turns back on.

Noise gates are useful for a number of important sound applications besides feedback control. For example, if a person or instrument is picked up by two microphones placed in different locations, the combined mic signals will interfere with each other, causing a type of distortion called comb filters. Comb filters add gain at certain frequencies and thus increase the chance of feedback. At the same time, they cut the gain at other frequencies, causing the program to sound thin and over-equalized. Gating the unused microphones eliminates this source of comb filtering.

Fig. 4: Loudness in Decibels

Sound Source	Sound Pressure Level in dB
Jet Engine	160
Threshold of pain	132
Riveter	120
Noisy Office	80
Conversational Speech	60
Quiet Residence	40
Recording Studio	30
Leaves Rustling	10
Hearing Threshold	0

Noise gates are often employed in CD players to eliminate noise between songs. They are similarly used in sound systems to mute the hiss of noisy electronic components during quiet periods.

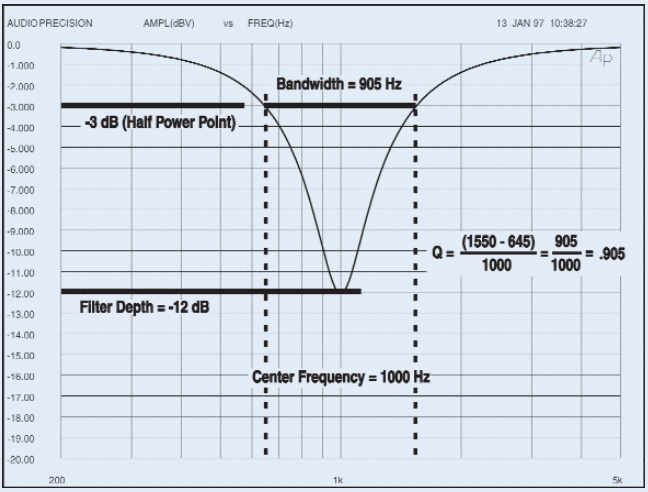
Most Sabine FBX Feedback Exterminators feature user-progammable noise gates.

What are Decibels?

We have the ability to hear an amazing range of loudness. People placed in an absolutely quiet anechoic chamber eventu-

Fig. 5: Typical Frequency Response

1K EQ slider pulled down 12 dB



ally perceive the sound of air molecules hitting their eardrums. On the other hand, people working near jet engines hear sounds a billion times more powerful. Engineers have developed a convention that economizes the calculations of such an enormous range of values. This convention describes these changes in terms of decibels (abbreviated dB) named in honor of Alexander Graham Bell.

Many non-technical people find the different uses of the term decibels confusing because it seems to have so many different meanings. For example, decibels are commonly used to describe the loudness of a sound, the change in loudness (or gain) from one time to another, for changes in signal voltage, and a number of other technical measurements involving the power ratio of large numbers. While we gladly leave these calculations to the engineers, it is helpful to realize that a change of 1 dB is equivalent to a 27 percent change in power. With this in mind, we realize that turning up the system gain by 3 dB increases the power approximately 100% (27% x 3). In other words, turning up the amp from 400 Watts to 800 Watts adds about 3 dB to the system gain.

Wow! Does doubling the power from 400 Watts to 800 Watts make it sound twice as loud? No! A three decibel change sounds only slightly louder. In general, you have to increase the power about 10 times (or 10dB) to make the sound seem twice as loud.

When engineers describe the loudness of a sound in terms of decibels, they are comparing the sound pressure level of a particular sound compared to an international standard. Fig. 4 gives several common reference points.

Frequency Response Curves

A frequency response curve is a graph that shows the gain of a component or a group of components at different frequencies. Fig. 5 shows the frequency response of a typical equalizer with the 1,000 Hz slider pulled down 12 dB. The frequency response curve shows that the biggest cut in power, called the center frequency is at 1,000 Hz, that the filter removes half of the power (-3dB) between 645 Hz and 1550 Hz, the Q of the filter is 1550-645 Hz/1000 Hz (.905), and the maximum depth is -12 dB.

Fig. 6: Typical Frequency Response

Two overlapping EQ sliders pulled down 12 dB

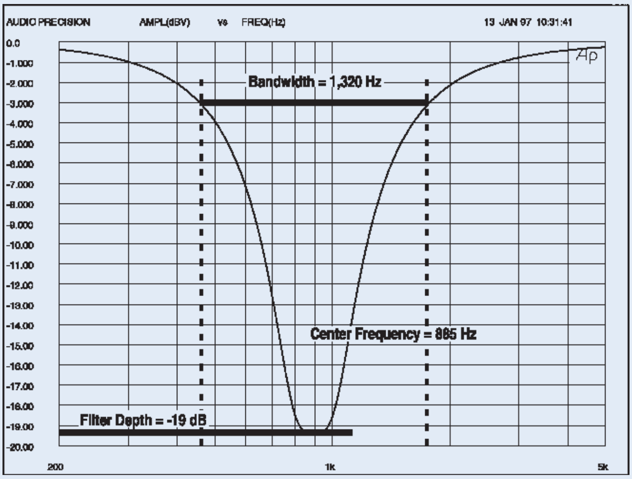


Fig. 6 shows the frequency response of two adjacent sliders pulled down 12 dB. Notice that the center frequency of the two sliders is at 885 Hz. The combined filter width is 1.49 octave and the two filters add together to give a maximum depth of -19.3 dB.

Constant-Q Filters

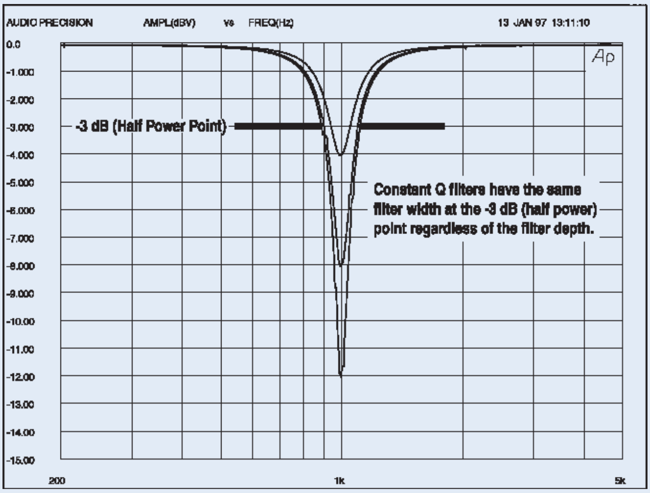
It is common to describe a filter's quality factor, or "Q," as the center frequency of the filter divided by the filter width (in Hertz) measured at the -3dB point. Filters that have the same Q, or width, at the -3dB point regardless of the filter's cut or boost are called constant Q filters. Filters that get wider as the filter gets deeper are called proportional Q filters.

There seems to be a new development in the audio industry. The definition of constant Q is blurring. Many equalizer manufacturers claim their equalizers have constant Q filters, when in fact they get substantially wider as they get deeper. The only way to know for sure if the filters are truly constant Q is to inspect their frequency response curves. (See Figs. 7 & 8.)

Net Gain Before Feedback

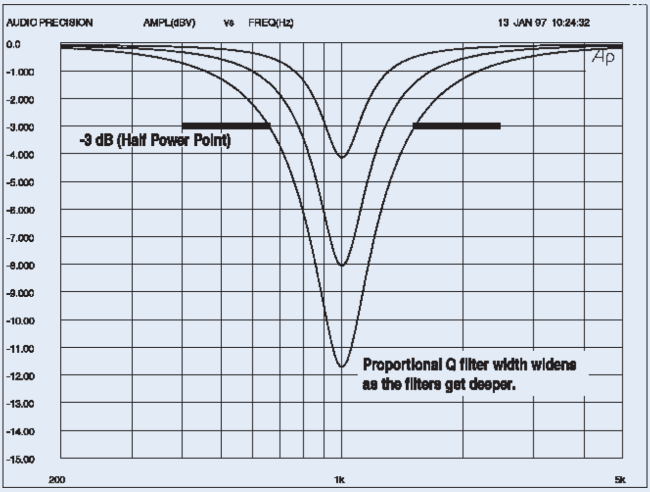
Many people measure their increase in gain by the amount they push up the mixer's calibrated slider. But if adding gain

Fig. 7: Typical Constant Q Filter



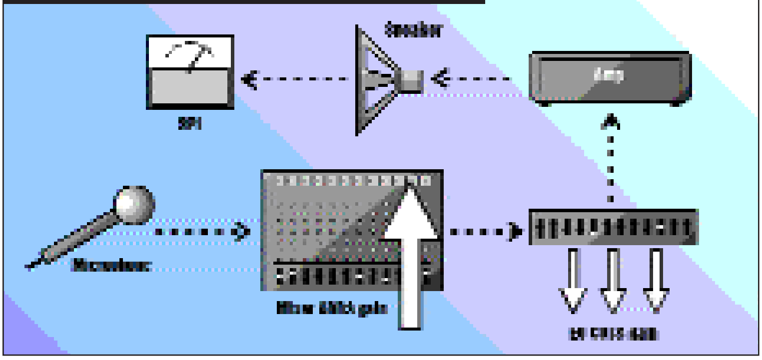
Constant Q filters have the same filter width at the -3 dB (half power) point regardless of the filter depth.

Fig. 8: Typical Proportional Q Filter



Proportional Q filter width widens as the filters get deeper.

Fig. 9: Net Gain = Mixer minus EQ



causes feedback, you will have to cut the gain of the feedback frequency at the EQ in order to add gain at the mixer. A more accurate concept could be called NET gain. It is the amount of gain you achieve pushing up the mixer slider, minus the gain you lose lowering the EQ sliders. NET gain is the gain you realize in front of the speakers as measured by a sound pressure level meter. That is the gain that matters. (See Fig. 9.)

The Frequency Spectrum

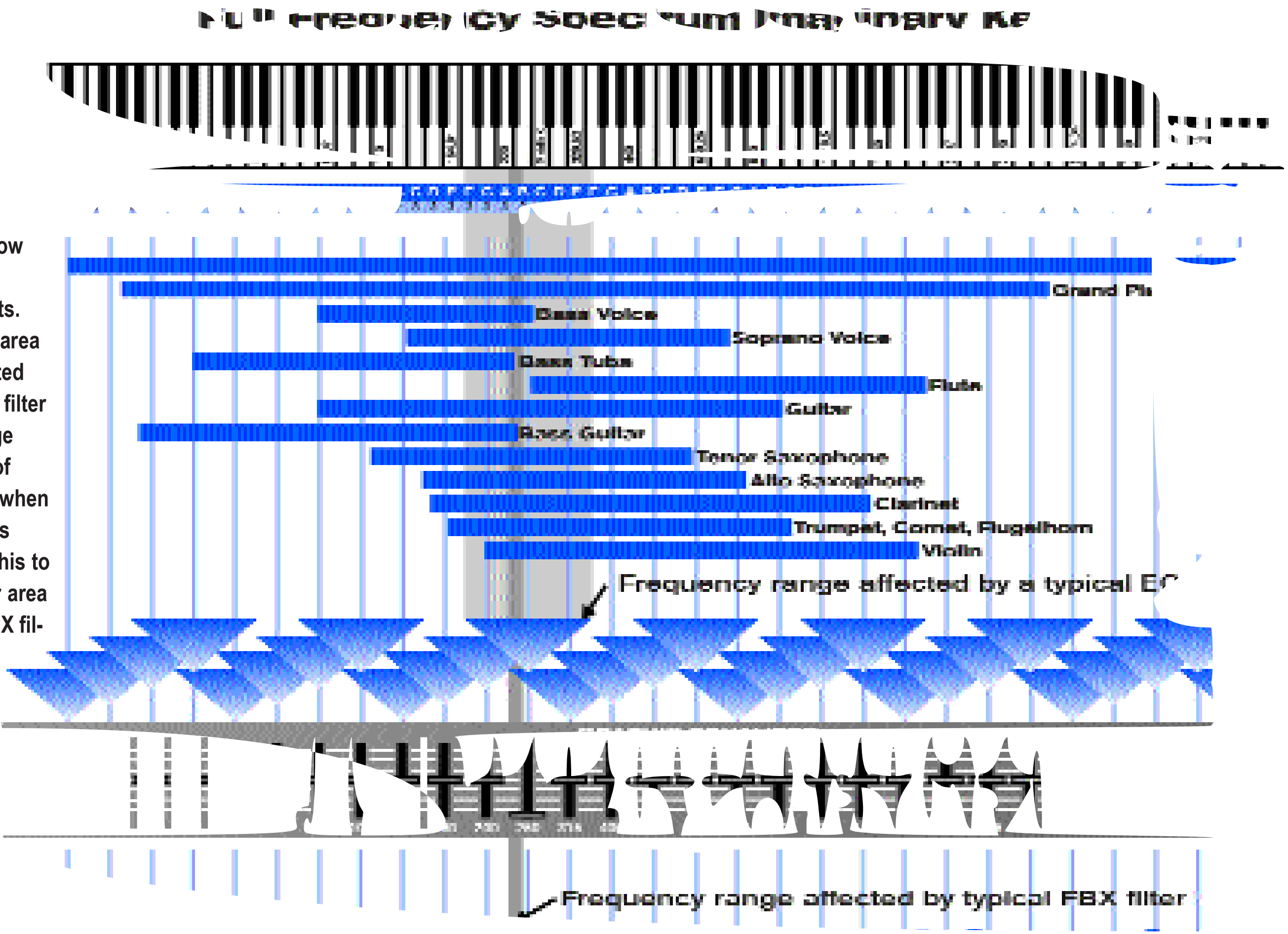
People with excellent hearing can hear frequencies between 20 and 20,000 vibrations per second or Hertz. Fig. 10 shows an imaginary 120 key keyboard that would be big enough to play all the notes that we can hear. The lowest key would play a 20 Hz "E" and the highest key would play a 19,912 Hz "D#." Notice that doubling the frequency raises the pitch one octave. We hear the same one-octave musical interval between 40 and 80 Hz as we do between 10,000 and 20,000 Hertz.

A graphic equalizer is superimposed that shows which sliders affect the notes of several instruments. For example, the chart shows that the 250 Hz slider affects most of the bottom 1/3 of a guitar's range.

The typical FBX filter below the EQ shows the relatively smaller size and effect on sound of FBX filters and illustrates why they cause less tonal change and gain loss.

The nine FBX filters are not preset on any particular frequencies like EQ filters. They are placed precisely where feedback occurs.

Fig. 10: The Frequency Spectrum



Horizontal bars show practical ranges of musical instruments. Light gray shaded area shows tones affected by one graphic EQ filter (250 Hz). Note large gaps in response of many instruments when just one EQ fader is pulled! Compare this to the much narrower area affected by one FBX filter (dark gray bar).