Room EQ???

So what is happening when we put a sound system in a church or in any enclosed room? Why is the sound sometimes clear in one part of the church but only a few seats away we can hardly hear anything at all while another few seats down it is very loud, or if we hear we cannot understand what is being said? Why is it when the speakers were brought in and demonstrated on stands they sounded great on the demo music that was played, but when they were hung up higher and installed, the spoken word just is not easy to understand?

THE ENEMY
When we speak outdoors or hear a loudspeaker outside, the sound is allowed to travel free in any direction. As soon as we put walls around a sound system, the sound starts to bounce off one wall and then to another wall, and so on.

Cancellation and acoustic amplification at certain frequencies can occur. Therefore, much care has to be taken with how the walls are placed, the size of all the structures, what they are made of, the position of the congregation of the church, and so on.

We do not want a church to sound "acoustically" dead for either speech or music. We want a little "warmth" or reverb (not to be confused with echo) but too much can affect speech intelligibility.

Some of the bad things that can happen are:

• Too much reverb smearing the speech.

• If the width, height and length of the room are divisible by the same number, then "standing waves" result. These cause a cancellation of frequencies at a certain area of a room. You may hear everything perfectly where you are sitting but two seats down the pew you can barely hear at all.

• Feedback is that horrible loud squealing sound that can come from your loudspeakers. This occurs when the sound from your speakers re-enters your microphone. Feedback "modes" are very narrow and can be numerous in a church with poor acoustics. They usually occur between 50 Hz and 2500 Hz.

• The room "rings." This is different from feedback. What you hear is a tone-like or ringing sound that seems to be added to almost every word. What is happening is one or more of the frequencies from the loudspeaker are exciting the physical architecture of the room itself. Ring "modes," like feedback modes are also very narrow but different as they do not have the phase characteristics to cause the system to self-oscillate (or feedback).

FIXING IT
You may say to yourself, “well we’ll get an architect to design the church and

that will solve all our problems.” However if the church does not also retain an acoustic consultant at the same stage as the hiring of the architect, trouble could result. Be aware that the architect usually hires the acoustic consultant.

As a result there are far too many cases where his advice was not taken and fundamental acoustic treatment and high quality audio components were left out in lieu of solid brass door knobs, exotic wood furnishings, etc. A lot of designs may look good but if your walls are flat and parallel, worse yet, made out of glass, the room could simultaneously be an architectural splendor and acoustic hell.

Some of the most gorgeous and famous "modern" churches seem to be getting a new sound system every other year (and will continue to do so until the architecture is corrected). My advice would be to have the acoustic consultant work directly for the church with veto power over the architect’s designs as they affect the acoustic performance of the rooms and budget for acoustic treatment and sound system equipment.

Then you might say, "Okay, we will just get an audio equalizer and that can fix all these acoustical architectural errors." The truth is if you have an extremely well-designed acoustical room with high-quality speakers covering their designed areas, then the need for sound system equalization is greatly minimized. And even though I love to sell equalizers, the less you need the better. Besides, of our list of "enemies" above, an equalizer can only help the last two, feedback and ringing.

GOD VERSUS SATAN
And then there is the biggest ongoing conflict seen in churches’ sound systems today. No, it is not God versus Satan. It is the pastor versus the music minister. At many churches I visit, these two people seem to be at cross-purposes. The pastor just wants to be heard clearly. But the music minister wants a "rock ‘n roll" sound system that would rival the Rolling Stones.

One of the problems facing sound system designers is that the equalization curve for a pure speech system is very different for a music system. What we are starting to see in many churches is an idea that came from movie theaters and the Broadway stage.

Designers are using a left-center-right or LCR speaker system. In a true LCR installation the center channel cluster is "equalized" for speech intelligibility, and the two left and right speakers are set up for music. Beware that some mixing consoles are advertised as being an LCR mixer, but are not.

A true LCR mixer pans from left to center and from center to right. Some companies have relabeled their "monaural" or mono output to "center." Do not be fooled, this is not the same. The point is that the pastor’s voice should only come from the center, even if he is singing. In order to minimize comb filter effects, the mix to the left and right speakers need to be treated as stereo, and not a mono program panned to the left and right.

THE FIRST ACOUSTICIANS

So, who were the early practitioners that discovered equalization? We have to go all the way back to the Middle Ages when the first pipe organs were being installed in large cathedrals. The pipe organs were put together in workshops, and then taken apart and reassembled in the cathedral.

When the organ was played they found that certain notes "took off," or blared loudly. These were referred to as "bull" notes (a term still used today). So the pipe organ maker, by ear, would tune that pipe down in level so it would not excite or cause the room to resonate. This was a long painstaking task, since every pitch had to be perfect.

Plus the ornate structures and curves of the ancient churches did a lot to help diffuse acoustic reflections, and intentionally or not, made pleasing acoustic mediums for the organ music. So the art of sound equalization goes back a long way.

Today we have many tools at hand. I will not go into real time analyzers, TEF, SMAART systems, etc. since a discussion of these devices would require their own article. Instead, we will talk about filters. Third-octave and parametric equalizers all use filters, the difference is their shape and characteristics. We found out earlier that feedback and ring modes are extremely narrow, so we just want to rid ourselves of them without affecting the overall sound system.

The most frequently seen EQ is the 1/3-octave equalizer. Prices can range from $200 to $2000. It is interesting to note that a very famous "loudspeaker" manufacturer teaches its sales force to always use the best quality equalizer available, and if you have to cut costs at all in the sound system the last thing you cut is the quality of the equalizer. Bless their hearts

Much goes into the quality of an equalizer. Aside from careful attention to noise and distortion specifications, the first thing the designer needs to choose is the filter's shape or architecture. The two most common today are referred to as constant-Q and variable-Q. The Q in equalization deals with the width of the filter as it is raised and lowered. (See figures 1 & 2)

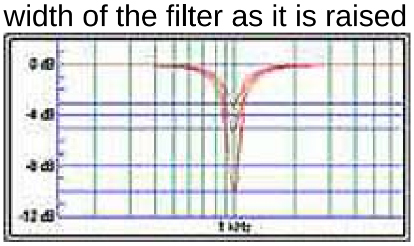

Figure 1

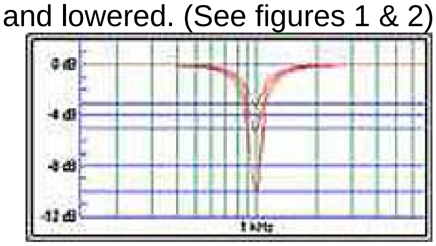

Figure 2

Notice the constant-Q maintains the overall width of the filter as it is adjusted, whereas the variable-Q varies its width as it is adjusted from wide at +/-3 dB to narrow at +/-9 dB. These architectures or "transfer functions" were developed in the 1930's. Companies that use them will make their own minor modifications to how they work, but generally speaking this is how they perform.

Constant-Q was espoused at a time when analog parametric equalizers were very noisy (most still are), to get rid of feedback without taking music out of adjacent frequencies. The fallacy of this claim is that feedback rarely occurs on ISO (Industry Standards Organization) centers. These are the common frequencies everyone has agreed to use for uniformity on equalizers. (See figures 3 & 4)

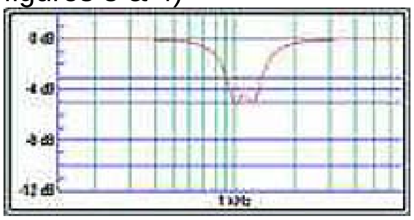
Figure 3

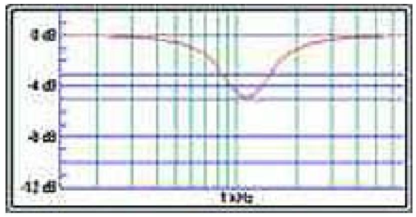
Figure 4

Look at the transfer function of two adjacent frequencies of constant-Q and variable-Q equalizers, and see the gap between the two frequencies on the constant-Q, whereas the variable-Q sums as one. What if the feedback mode was between those two frequencies?

Most professional fixed-installation installers now use 1/3-octave equalizers more for tonal balance, not feedback control. They may move three or four adjacent frequencies up or down a few dB for tone control. (See figures 5 & 6)

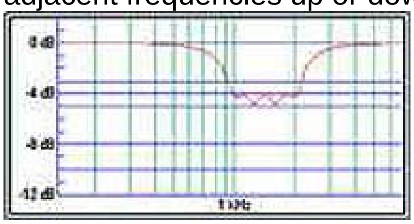
Figure 5

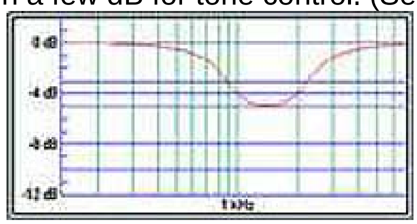
Figure 6

Look at the ripple effect with constant-Q. It looks like the intelligibility distortion known as "comb-filtering." Now look at the variable-Q example and see how the frequencies sum more smoothly. Which do you think sounds better?

Constant-Q is marketed strongly through the "lower-end" of the rock ‘n roll market. One engineer from a well-known audio company has privately said their older equalizers sounded much better than their new constant-Q equalizers but they changed because of marketing pressure. Companies that use variable-Q include Klark Teknik (easily the most visible touring sound equalizer today), Ivie, Micro Audio, Electro-Voice, Altec Lansing, and White Instruments.

Another difference is what is known as designing a "band-pass" type equalizer. In this type of equalizer (which requires extremely high-quality componentry) all the EQ filters are always in the circuit regardless of the position of the faders. This guarantees that the overall noise level will not change much as the filters (or sliders) are moved up or down.

Compare this with companies who make their +/- 0 dB setting a bypass circuit. They do this because their circuitry is not as good (we are talking about much less expensive equalizers now). Every time you move one of their filters, noise is introduced to the system in a cumulative effect. Even when a filter is cut, the noise rushes in. This is probably the major price difference in analog equalizers.

PARAMETRIC EQ & BOWLING

The ideal filter would be a digital parametric filter for control and low noise. Parametric filters let you vary the Q any width you want from many octaves wide, to as narrow as 1/70th of an octave. And, believe it, those feedback and ring modes are even narrower than that.

The idea of ridding feedback with a 1/3 octave EQ in a fixed installation system is like rolling a bowling ball to knock down only one of twenty adjacent toothpicks. You get the one you want but you crush the rest. A 1/3 octave equalizer can diminish the feedback and destroy all the music and sound of the neighboring frequencies. A parametric equalizer can selectively "notch" the offending frequency without harming the overall sound.

We now have much higher quality digital signal processors with built-in parametric equalizers and even stand-alone digital parametric equalizers for reasonable prices that operate very quietly. Fixed installation narrow-banded EQ is available to anyone on almost any budget. Although it still takes a professional to do it correctly.