## Sound Systems and Narrow-Band Anomalies Revisited

By Emory Straus with the help of Ken Dickensheets, Charles Boner, Mark Engebretson and Gifford White

**Where we've been.** Long ago the builders of pipe organs observed that the room in which their organ was installed had a pronounced effect on the sound of the instrument. The practice was to assemble the instrument in their workshop on a device called a *voicing jack*, and do a rough tuning and balancing. After moving the organ into its permanent home, they found that they often needed to re-tune and re-voice the instrument.

The voicing method consisted of regulating the air pressure at the individual pipes to notch or duck the *bull notes* (specific notes that would excite the architecture and cause the room to ring). At the same time adjustments were made to provide smooth power response over the instrument's audible range. A pair of well-trained ears was the only analyzer available to the yeomen who would spend days, if not weeks, doing a craftsmen-like (EQ?) job.

Dr. C. Paul Boner, acknowledged to be the father of equalization, borrowed the voicing concept from the organ world and applied it to sound systems, beginning in the late 1950's. At a 2000 seat auditorium in a small north Texas town, an unfortunate sound contractor was refused payment for his system because feedback limited the usefulness of the system for theatrical productions. Dr. Boner applied the organ voicing methods to the sound system. He, among other researchers, reasoned that if the sound system's electrical response, at or near the feedback frequencies, could be reduced, better gain before feedback could be achieved.

This was a time when there were no equalizers, no audio analyzers and no useful wireless microphones. Compared to the audio components we enjoy today, almost everything manufactured at that time needed some help, especially the frequency response characteristics of the transducer systems. Sounds were measured in a room using white noise carved into bands approaching 1/3 octave by a human walking the sound field with a sound level meter. Dr. Boner constructed his own filters using coils custom-wound with #12 copper around wooden cores. Obviously the Q of these devices was very low and the filters were very broad. After two weeks of this work, by trial and error, Dr. Boner was able to improve the system to a point where the sound contractor was paid.

After this experience "Doc" Boner realized that he would require new filters, which were much more narrow in their response, to achieve the goal of reducing feedback while retaining system response quality. He approached his friend and past physics student, Gifford White, founder of White Instruments, for help. At that time, White Instruments built industrial filters, mainly for instrumentation manufacturers. Together Dr. Boner and Mr. White developed several economically practical, high Q, tapped toroidal coils that were stable and allowed the implementation of filters having various frequency, bandwidth and depth characteristics.

Armed with the new filters Doc Boner began developing an equalization process that always included both broadband and narrow-band correction. The broadband filters, typically 1/10 to three or more octaves wide, were applied to correct for transducer and transducer/room interaction problems. The narrow-band filters, those narrower than 1/10 octave, were used for the room anomalies such as feedback and ring modes.

Very little time passed before Dr. Boner and his consulting firm, Boner Associates, Consultants in Acoustics (founded in 1936) gained a favorable reputation with the public and became famous in the then small world of sound system designers and operators. Among others in the field, Dr. Boner taught or mentored his two sons, Charles and Richard, Bob Newman, the "N" in BBN, Bob Coffeen of Coffeen/Anderson, George Augspurger of Perception, Frank Morris and Ed Jones of LDS Church Engineering and Ken Dickensheets of Dickensheets Design Associates.

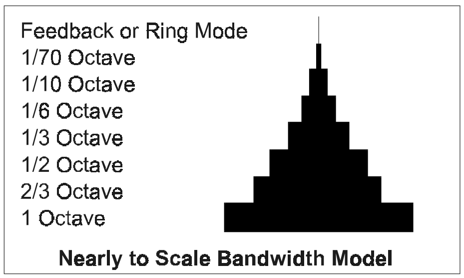

**Figure 1**

**Putting bandwidth into perspective:**
The narrow-band modes causing feedback and room ringing are sinusoidal (infinitely narrow bandwidth). Showing one to true scale in Figure 1 would require a line that is too narrow to display or print. Notching these modes with broadband filters (1/10 octave and broader) also suppresses benign program material between the modes and destroys sound system quality.

For many reasons, this concept of *correct* filter utilization became obscure to all but a handful of audio engineers as the industry matured. The almost universal adoption of the ~~one-third octave equalizer and ana~~lyzer, a compromise approach and largely an economic and marketing decision not firmly grounded in acoustic theory, did more to mask the need for narrow-band equalization than anything else. Aside from the EQ on the mixer (which several consultants had disabled) this convenient and cost effective combination became the sole tool for sound system equalization. The bandwidth of a *combining* one-third octave equalizer's filters was closer to one-half octave than one-third octave. The bad news was that one-third octave equalization, used alone, left the EQ job less than half finished and was easily and often overused to the point of actually making some sound systems sound worse than if they had been left alone in the first place.

In the early 1970's, Charles Boner, Ken Dickensheets and Ed Jones tuned the Marriott Center at BYU, using only narrow-band filters. This experiment proved the significance of the room's narrow-band room anomalies on the system's broadband response. Upon completion of the tuning, the broadband response of the system was measured and found to be nearly flat.

Many audio practitioners at the time were correctly worried about the phase shift and ringing problems associated with narrow-band filters, but the phase shift and ringing problems associated with the acoustic filters was also highly significant and often overlooked. These *acoustic filters* were created by the sound energy that caused the architecture to resonate.

Gifford White came up with an analogy that went something like this: create two narrow-band filters, one that peaks 10dB and its exact reciprocal that notches 10dB (+1 + -1 = 0). Phase shift and ringing can be measured in either filter alone. Put both filters in series and measure the phase shift. You will measure no phase shift in the circuit and it cannot be caused to ring.

Therefore, the practice of correcting a room's narrow-band modes with extremely narrow-band filters does not cause additional phase shift and ringing. In fact, the phase shift and ringing associated with the acoustic filter (the room) is minimized by the introduction of the proper narrow-band electronic filter. While it is true that these narrow-band filters ring for a few milliseconds, they can control room/system ringing that can last for several seconds (or notching reduces the energy to the architecture, thus preventing the ring from building up).

*It should be noted that the handful of designers, who understood and practiced the proper application of narrow-band techniques and broadband equalization, seemed to get the lion's share of the large, high visibility jobs despite all of the confusion in the trade concerning phase, ringing and narrow-band filters.*

Dr. Boner and White Instruments did not ignore other, larger phase problems, however. Their research identified serious response, phase, and impedance problems in the amplifier/crossover/speaker interface (remember that crossovers in the early days operated at speaker level between the amplifier and loudspeaker). First, a *power factor corrector* was developed which operated in conjunction with other manufacturers' crossover networks to correct these problems. Later, the first passive, *low-level crossovers*, which operated as a part of the broadband equalizer at the inputs to the power amplifiers, were manufactured. These crossovers were the forerunners of the integral crossovers, which are a part of today's DSP equalizer systems. Each of these products further reduced the system anomalies inherent in the non-room parts of the sound system.

**Progress.** While one-third octave equalization was having its hey-day, the transducer manufacturers were quietly at work engineering to eliminate the need for corrective equalization. The microphone manufacturers learned to build microphones that exhibited flatter responses and, more importantly, smooth off-axis responses. Many were so flat that they could almost be used for serious instrumentation. Mark Engebretson and John Meyer replaced the generic crossovers with carefully engineered *loudspeaker signal processors*, matched to specific loudspeaker systems. These devices not only kept the individual system components under control by bandlimiting their response (before Mother Nature would do it in a most nonlinear fashion), but also assured a smooth frequency power response from the loudspeaker system. At the same time, the amplifier engineers were increasing their power ratings to adequate levels while lowering the distortion. Acoustical treatment and its materials began being thought of as sound system components by more and more audio professionals who had recently armed themselves with an affordable (and portable) form of Richard Heiser's Time Delay Spectrometry, called TEF™. The net result of these audio milestones was to reduce the necessity for broadband equalizers to straighten out intrinsic component frequency response problems.

A new kind of equalizer, called a *parametric*, was introduced that promised to be all things EQ, to all systems, at all times, and under all circumstances. Some system designers actually left one-third octave EQs out of their specs and began to control narrow-band and broadband problems with the versatile filters the parametric equalizers offered.

Yesterday's parametric equalizers with 6, 12, or even 24 filters were barely sufficient to properly control the room and system's major ring and feedback modes. In order to do an adequate job several parametric units had to be cascaded. This really wasn't a very good solution due to the noise generated by the many active filters.

White Instruments manufactured two passive narrow-band notch filter systems that were later discontinued with the introduction of new, high quality digital signal processor (DSP) units. These DSPs solved the weight, time and expense problems associated with narrow-band equalization. This was a big relief to the many acoustical consultants who carried around the nearly 200 pounds of White's 3900 filter kit and had to keep up with the huge capacitor inventory associated with the tapped inductors. Each new generation of DSPs follows computer industry trends by increasing performance and capability while reducing the cost. It is now possible to apply 100% of the equalization process, conveniently and at a low cost with high quality results using narrow-band filters. This brings the equalization process full circle. It only takes a little time to properly apply the filters.

**So, what's the point?** Dr. Boner had to use a great deal of broadband equalization to straighten out the equipment he had to deal with. After his north Texas experience he never used broadband filters to control narrow-band problems: feedback and room ringing. Today one can specify sound system components that need very little (if any) broadband correction. What hasn't changed is the fact that, as long as we are putting sound systems in rooms that are not anechoic chambers, we still need to deal with the narrow-band problems.

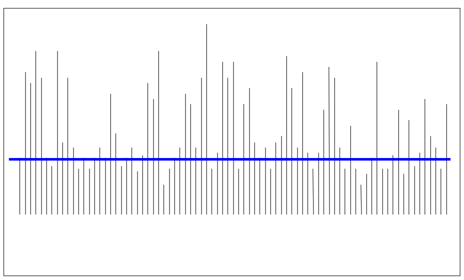

**Figure 2 – T60 Measurements, 1Hz Apart Compared to Average**

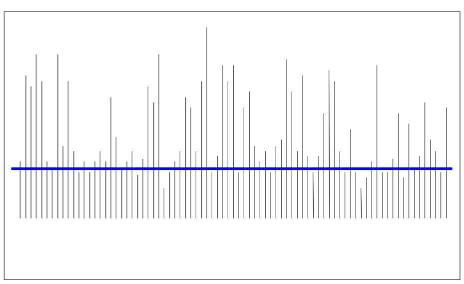

**Figure 3 – Amplitude of Modes Compared to Broadband**

**What are these narrow-band modes?** Feedback and room ring modes are extremely high Q responses caused by a room's architecture resonating from the acoustic energy released by a sound system (or any other pressure source). They are acoustic filters that tend to clump together around a major mode (see Figures 2 and 3). Their Q is so high that it can be thought of as being nearly infinite, making the mode nearly sinusoidal by nature (see Figure 1). Without going into detail, a feedback mode possesses phase characteristics that will cause a sound system with an open microphone(s) to self-oscillate. A room ring mode, on the other hand, does not possess the appropriate phase characteristics to cause the sound system to feedback but is heard as a distinct ringing. Some rooms can be excited with normal conversation and the ring modes can be heard, even when the sound system is turned off. Both modes have amplitude responses that are considerably higher than the average broadband sound pressure level of the sound system. Therefore, the $T_{60}$ decay time measured at either mode's frequency will be markedly longer than the average reverberation time of the room. The effect of a feedback mode is obvious. The lingering room ring masks or confuses the real time signal. It reduces the system's articulation or clarity - music is muddy and speech is not very intelligible. Applying narrow-band notch filters to electronically reduce both modes is close to performing black magic on the sound system, and *is* the direct electronic equivalent of voicing a pipe organ.

Feedback depends on the placement and characteristics of the house microphones and the loudspeaker system and on the distances between them. Ringing depends on the excitation of the architecture. Appropriate specification and deployment of transducers having smooth power response characteristics is paramount. The importance of proper acoustic treatment of the room's surfaces as well as proper design of the room's shape can not be overstated. Both will reduce the occurrence of either mode. After the room design and the sound system have been optimized, more often than not, they will profit from a thorough narrow-band equalization job.

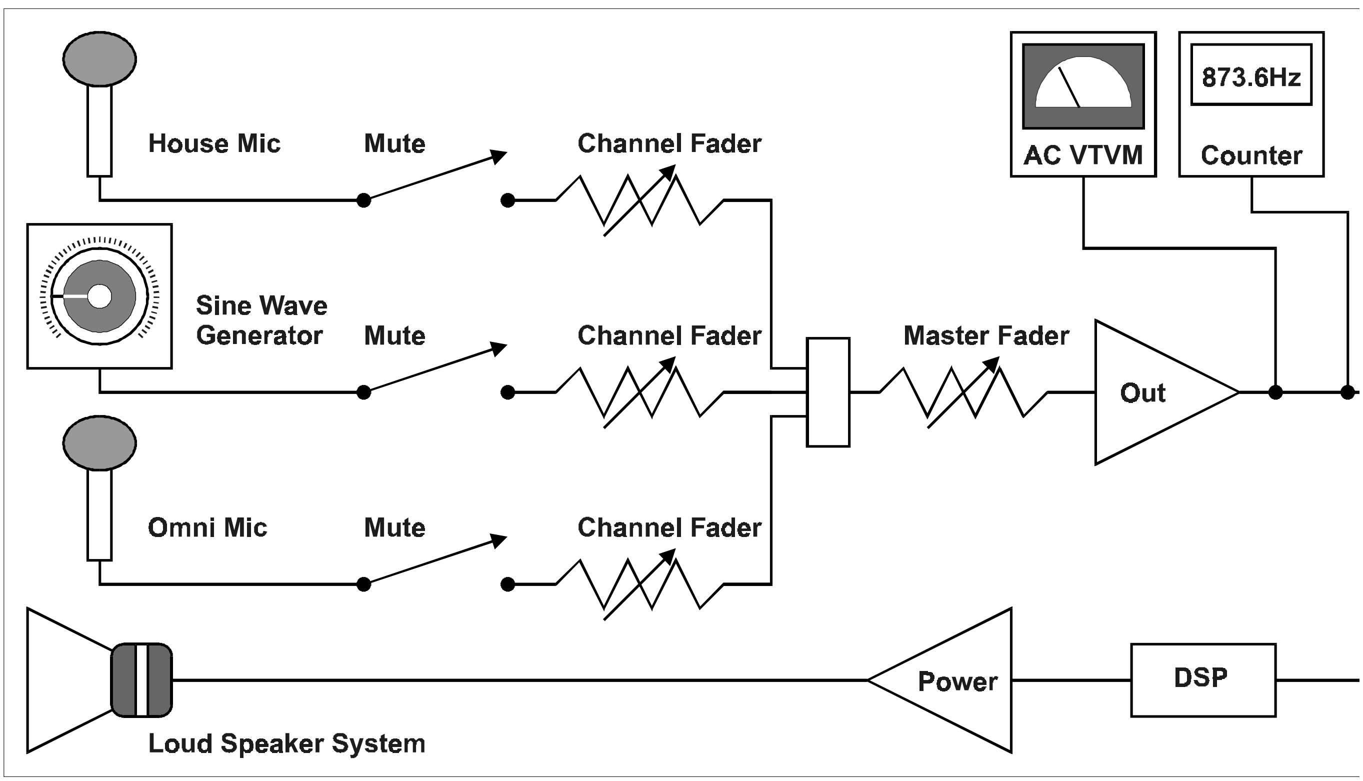


**Figure 4 – Sine Wave Analysis Setup**

**How does one find these menaces?** Figure 4 shows the instrumentation required for sine wave analysis and its hook-up (Ken Dickensheets has written the sidebar for FFT analysis.). For this application the analog style sine wave generator with a large knob for rough setting and a small knob for fine adjustment is best. It allows setting the frequency and doing slow sweeps, back and forth, between two close frequencies. Similarly, an analog AC VTVM with a large meter works best since it shows the peaks better. A frequency counter that displays to the nearest hertz is also necessary. Operation of these instruments is easier using a mixing board with channel mute switches and faders. Make certain that all equalizers on the mixer are either bypassed or set to zero.

**Before you begin - the checklist:**

1. Check that all sound system components are working to spec and are engaged.
2. Check that there is no hum or noise.
3. Check loudspeaker deployment.
4. Check loudspeaker delay settings - engaged.
5. Check loudspeaker component balancing.
6. Check appropriate sound system level.
7. Check sound system high-pass and low-pass settings - engaged.
8. Check any subjective EQ added to the system - engaged.
9. Check microphone deployment.
10. Check the environment. Is it normal?
11. Check that no EQ is being added at the mixer.

**Feedback.** Using the house microphones, one at a time, notch the feedback modes first. The sine wave generator and omni- directional microphone inputs should be muted. Note that the AC voltmeter and frequency counter are attached across the mixer's output. Your ears should be attached to the room.

Open the house microphone and raise its level, using the channel fader, until the sound system takes off into feedback. To isolate a single mode, slowly move up and down the microphone's fader until a single tone dominates. With a little practice a pure sine wave can be isolated long enough to read it on the frequency counter.

Now create a notch filter with the DSP beginning with a Q of at least 20. (At 200 Hz a filter with a Q of 20 will have a bandwidth of 10 Hz; Q of 50 will yield 4 Hz; Q of 100 will yield 2 Hz.)

In theory, you will want to use the highest Q filter possible since you are going after a sine wave. In practice, however, one must expect the frequency of the room modes to drift somewhat with temperature and humidity changes. Experience will be your teacher. Even though White Instruments' new ParaMedic will allow you to use a Q of 100 (about 1/70th octave), such narrow resolution is recommended for use in only the most stable temperature and humidity environments. Research has shown that when a sound system is properly tuned with narrow-band filters, environmental conditions can change substantially yet the system will still operate within the bandpass of the filters. In otherwords, a room can be empty or full of people, humid or dry, hot or cold and the system will still feedback and ring at the same frequencies or within one or two hertz of those frequencies. Filters tuned too narrow will not allow for these variations. Also, filters tuned too broadly will affect too much program material just as the old 1/3 octave filters did.

Most modern DSP units feature multiple memories and their software permits global copying and parameter adjustment. Why not use these features to build experience? Set the filters to a Q of 20 in one memory and experiment with higher and lower Q settings in other memories.

Contrary to popular opinion, the notch depth does not have to be anything like the DSP's capability. Remember: 3dB is half-power and the deeper the notch, the wider it has to be at the top. Try an initial depth of 6dB for the first several feedback modes and as little as 3dB for the last ones. Remember to get the job done while using as little EQ as possible.

Repeat the process to locate and notch additional feedback modes. It will not be uncommon, in a large system, to use one or two dozen filters to achieve adequate gain with lots of headroom. Test all of the house microphones and even try multiple open microphones. If an already notched frequency appears again, add another 2 or 3dB (not 24dB) to the notch depth. Rarely is more than a 9dB notch to suppress a mode needed. If a deeper notch is needed look for a problem like a direct reflection into the microphone or a bad choice for the microphone itself or improper balancing of the loudspeakers. Feedback and ring modes tend to clump together and they are often separated by only a few hertz. Experience has shown that it is better to suppress each mode separately rather than correcting a group of modes with a wider filter.

**Ringing.** Recognizing and notching the ring modes is not quite as straightforward, but like all things, the more you practice, the better you will get at it

Open the mixer channels for the sine wave generator and omni-directional microphone. Slowly sweep between 50Hz and 1000Hz. Sweep very slowly to let the sound energy build in the room. Carefully watch the voltmeter and listen. As the sweep passes through a ring mode the meter will peak and, with practice, you will also hear an increase in level. Stop sweeping when you have found a peak and hit the mute switch on the generator channel. Listen to the room — it will continue to ring at the same frequency. Observe the decay on the voltmeter. The frequency counter will display the ring mode's frequency. Implement a notch filter on the DSP with the same amplitude and bandwidth suggestions as given for feedback.

If equipment to measure articulation is available, an improvement with the addition of each filter can be measured. Dr. Boner did not have portable articulation measurement gear and had to rely entirely on his ears. He often joked that he would keep adding filters until his client ran out of money. In a large, modern sound system it would not be out of the question to deploy as many as three or four dozen filters, so be sure to choose a DSP with enough capability and versatility for the job.

**Conclusion.** If, for example, twenty filters for feedback suppression and another twenty filters for the ring modes were implemented, each with a Q of 50 or about 1/35th of an octave, then only about one octave of program material has been affected for the entire process. Only the troublesome frequencies have been corrected, leaving everything else untouched. Compare this to the loss in a one-third octave equalizer — less overall equalization has been used to make a day and night improvement to the sound system. Less total equalization was used than two adjacent filters of a combining equalizer set to -6dB!