Sound System Equalization

Considerations, Preparation, and a Suggested Methodology for Sound Reinforcement System Equalization

by: John Murray, ProSonic Solutions

TABLE OF CONTENTS

I. INTRODUCTION

Equalization is the process in which relatively narrow band filtering is used to compensate for the efficiency variations & resonances in a speaker system's transfer function, or in English, to fix the fixable problems in its frequency response. It is used to reduce excitation of regenerative resonances (ring modes) caused by the microphone-speaker combination supported by the room. It is also used to compensate for the mutual coupling efficiency of multiple drivers and fractional-space loading by adjacent surfaces. Additionally, it has been used to prevent excitation of room resonances (commonly called standing waves, room modes, or in Europe: Eigenmodes or Eigentones), though they are position-dependent and acoustical solutions such as diaphragmatic absorbers or proper preventative architecture are better, but much more costly, solutions.

A well-equalized sound system exhibits a neutral transfer function, meaning it reproduces, as closely as possible, the exact signal as the sound source at a given listening position. A sound reinforcement system, like a master recording, headphones, or a studio monitor, should be a neutral part of the signal chain.

The creative portion is at the front part of signal chain, microphone selection, mic'ing technique, the mixing console EQ and effects insert sections, and the mix balance. The system EQ, dynamics processing, and crossover sections have only one purpose, to adjust the speaker system for a neutral response and to protect it from damage.

This neutral response electrically appears as a flat frequency response. Acoustically, on an RTA in the near-field, it also appears as a flat frequency response, but in the far-field the highs roll-off at a rate determined by several factors: the power response of the system, the absorption rate of air according to temperature and humidity, the distance between the measurement microphone and the speakers, and the absorption capacity of the room's surfaces.

The purpose of this paper is to give a more accurate method to equalize a sound system than to guess at what this roll-off rate may be for each sound system equalized. The roll-off rate would be different for each one and many equalizable anomalies in the speakers' responses are masked in the far-field frequency response as measured by an RTA.

The following sections describes the acoustical phenomena encountered in the process of equalization, details what cannot be equalized and why, and gives a suggested methodology to prepare for and carry out sound reinforcement system equalization. <u>The topics covered herein should give one a good understanding of why a system's response looks flat on the RTA display, but sounds terrible when instrument-assisted equalization is attempted without proper knowledge of what is involved.</u>

Please remember that the method suggested here is just that: <u>suggested</u>. The author is no absolute authority on audio. However, the science this methodology is based on is not "theory," it is fact. The study of acoustics is not rocket science and its principles have been measured, understood, documented, and implemented in all its variety before most of us were born. Anytime the "theory" doesn't match what is heard, the "theory" is either misapplied, misunderstood, or all its variables have not been considered.

II. ACOUSTICS IS ALL ABOUT WAVELENGTHS

a. The Sine Wave

The sine wave is the basic audio signal, a pure tone developed by a signal generator, flute, or sound system oscillations like acoustic feedback. It is well suited to illustrate the phenomena we will encounter in attempting to equalize a sound reinforcement speaker system. **Figure 1** (courtesy of Yamaha's *Sound Reinforcement Handbook*, p. 2) shows the basic sine wave, its relative phase angles, and its relation to acoustic compression and rarefaction of air pressure. The formula **W = S / F** (W = wavelength, S = speed of sound, F = frequency), enables us to calculate the size of a signal's waveform that we are dealing with.

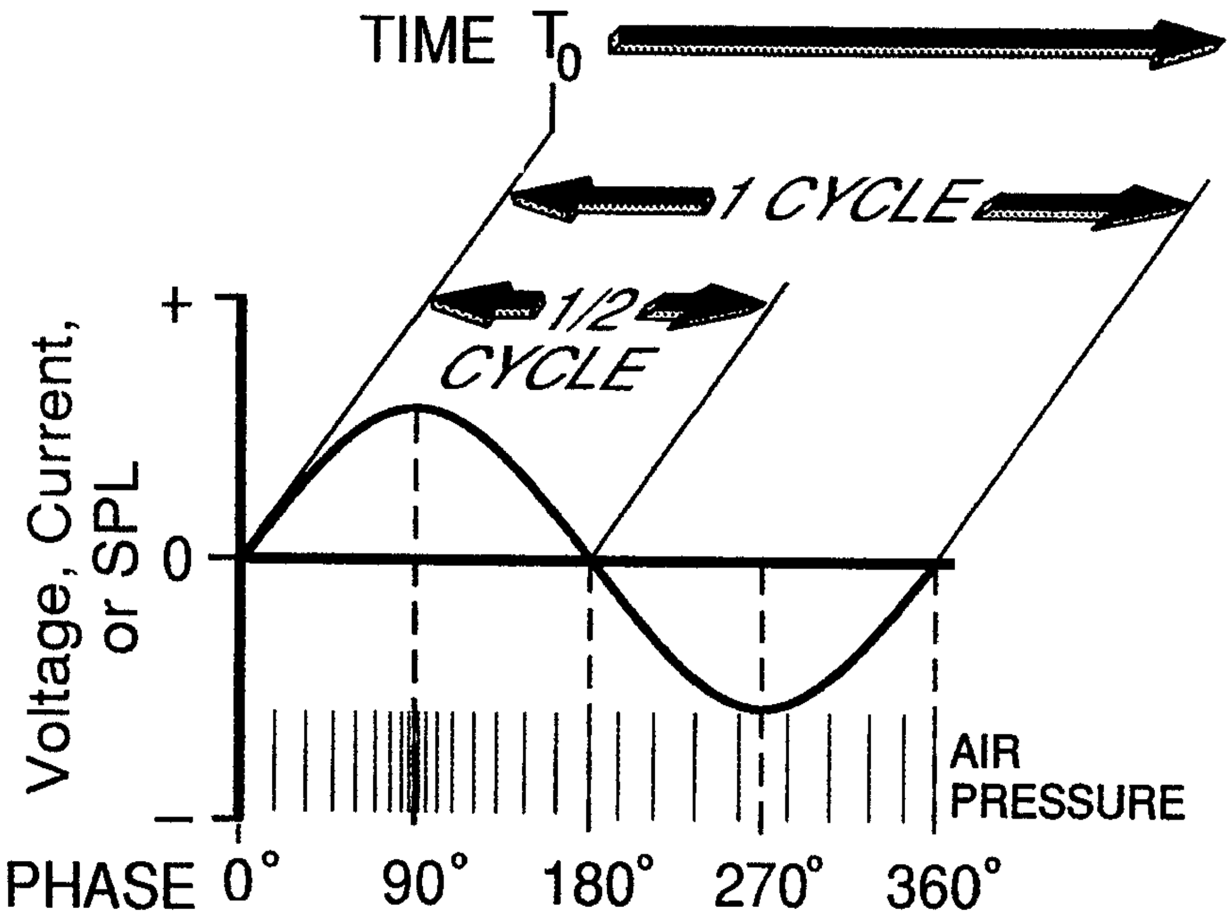


Figure 1-2. Representation of an audio signal (one cycle of a sine wave)

Figure 1.

b. Wavelength

In the realm of audio, we are concerned with signal content from 20 Hz to 20 kHz. This encompasses three decades of bandwidth, far more than most any other discipline of applied technology. Using the above formula, we can calculate that 20 Hz and 20 kHz signals, at sea level and room temperature (S = 1128 ft./sec.), have respective wavelengths of 56.4 ft. and 0.0564 ft. That's a 1000:1 ratio. The way low frequencies and high frequencies are dispersed from speakers and the ways they propagate in a room are very different. This must be kept in mind at all times when attempting system equalization.

c. Relative Phase and Summation

When considering what you are hearing or measuring with an RTA, one must always think in terms of a signal's wavelengths and the relative phase and level of its reflections or other subsequent sources. **Figure 2** (courtesy of Yamaha's *Sound Reinforcement Handbook*, p. 3) illustrates three relationships between two equal signal sources and their resulting combination.

As the relative phase between two sine waves varies from 0° to 180° and back to 0° (360°), the combined waveform goes from being 6 dB larger than one source (0° difference), to 3 dB larger (90° difference), to no signal (180° difference), and back to 6 dB larger again (360° difference). Be aware that in any room, there are many reflections at various levels and phase offsets that affect the direct signal from a speaker. As the distance from the speaker increases, the effect from reflections increases. Simultaneously, more reflections affect the signal, thus causing an averaging-out of this effect on the direct signal.

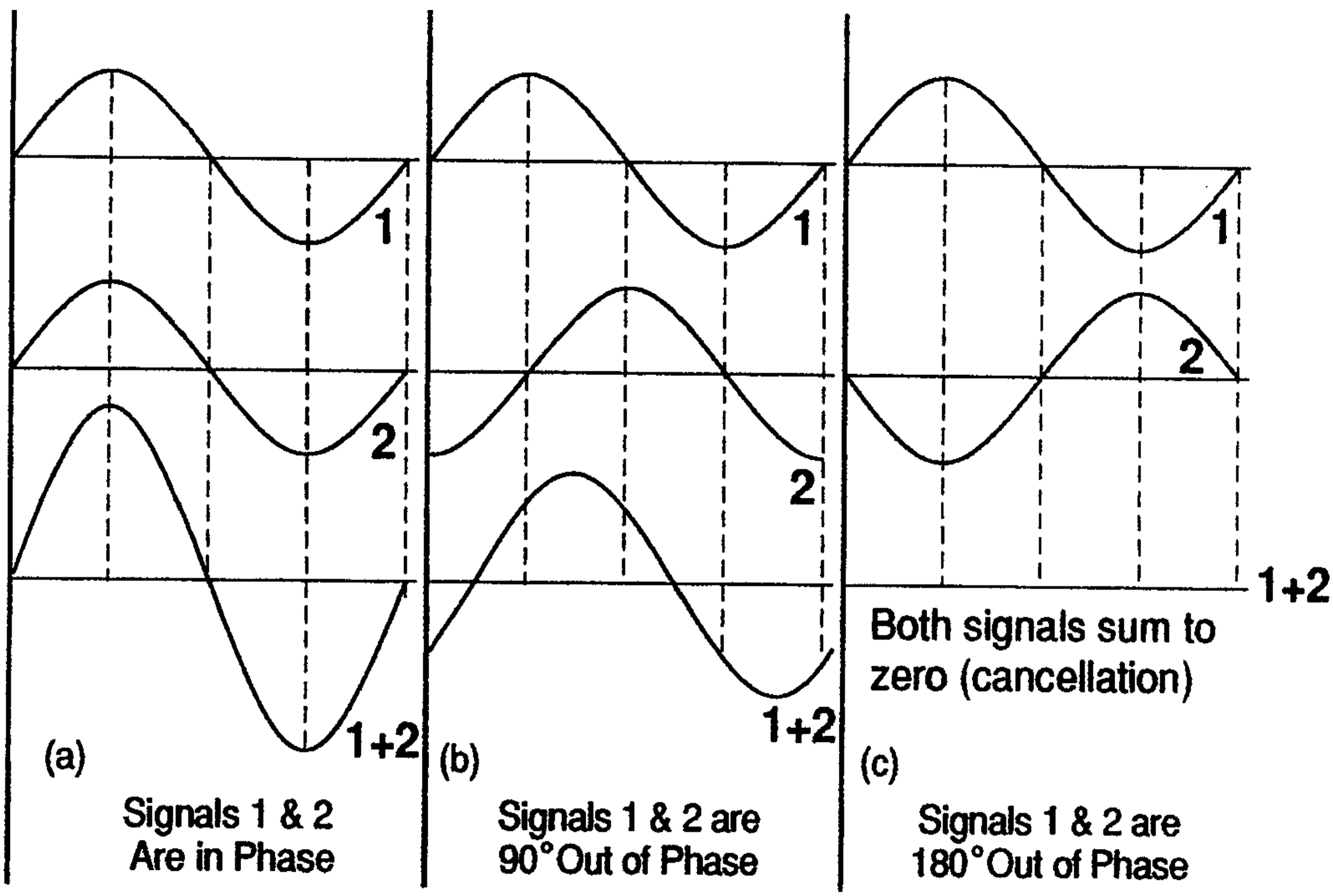


Figure 1-4. Phase affects the way two sine waves add together

Figure 2.

III. WHAT CANNOT BE EQUALIZED (TIME-ORIENTED EVENTS)

a. Non-Optimized Generic Crossover Networks

Unless a crossover network (active or passive) has been optimized for the particular drivers employed in your speaker system, there is generally some amount of misalignment at and around the crossover frequencies. **Figure 3** shows a common symmetrical 24 dB/octave Linkwitz-Riley active crossover driving a two-way loudspeaker system without any compensating change in slope rates, filter types, or signal delay for the drivers employed.

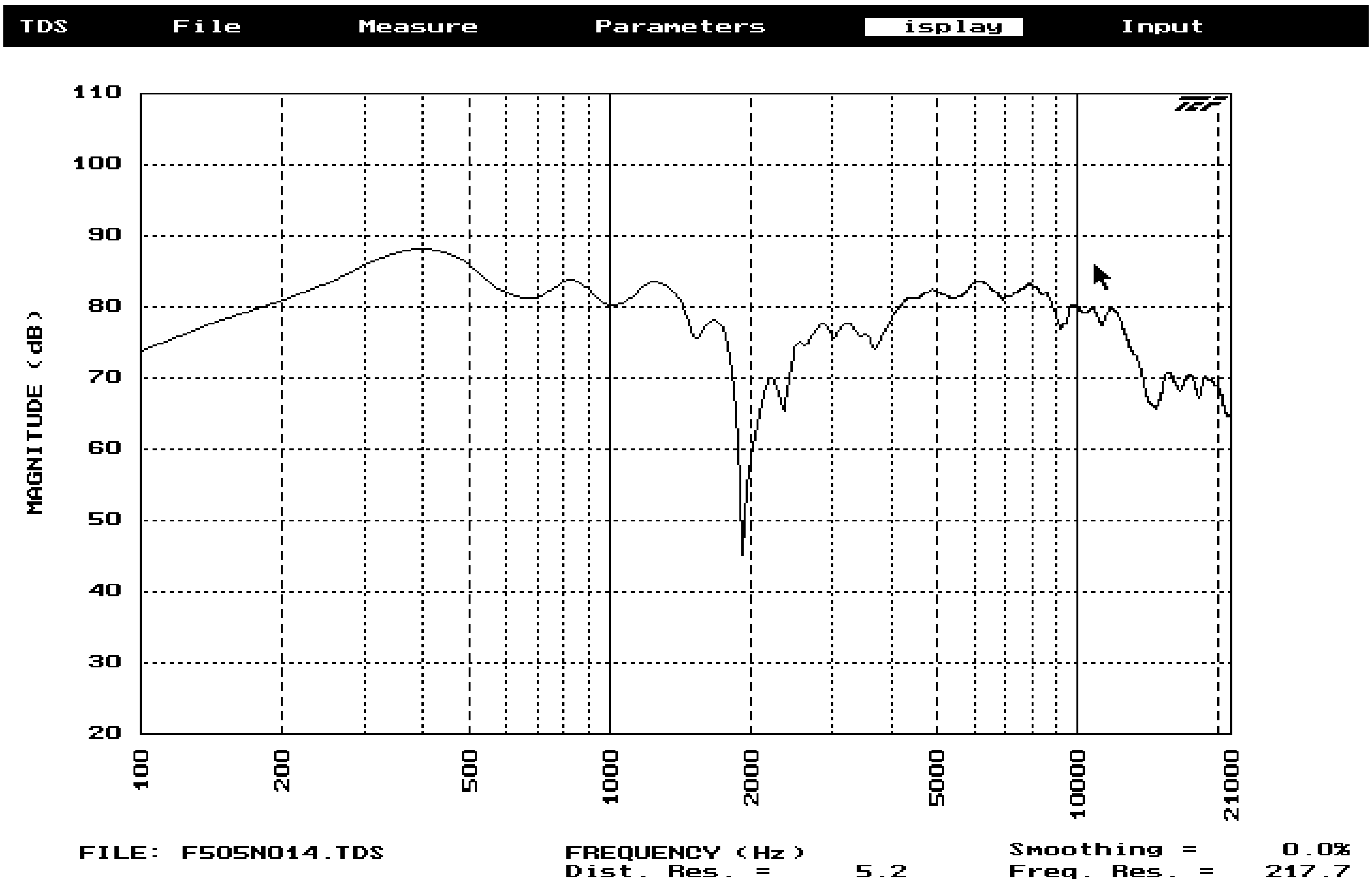


Figure 3.

Figure 4 shows the results of optimizing the crossover of the speaker system. Obviously, this makes the job of equalization much easier; especially considering that what has been improved here could not have been done with an equalizer.

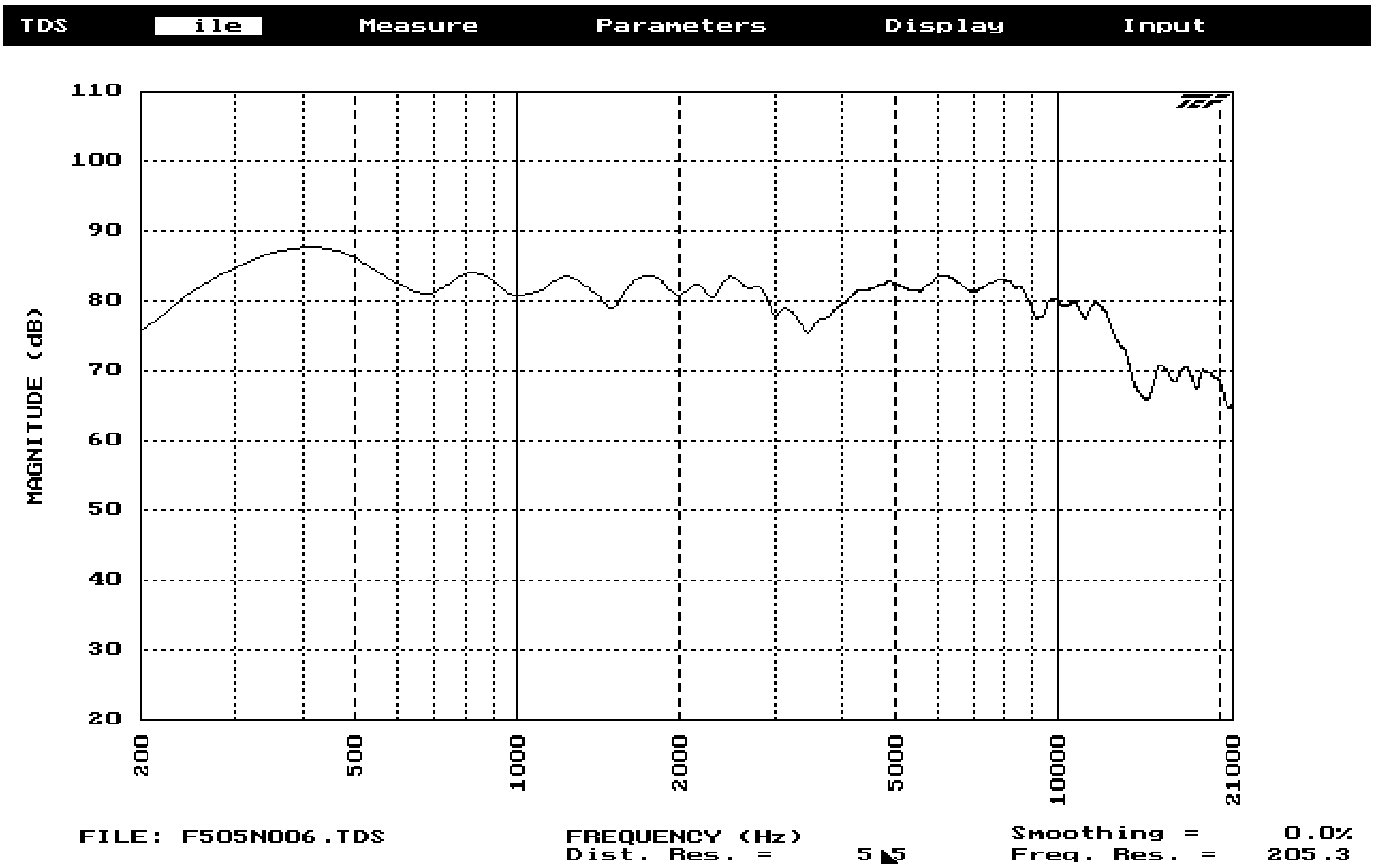


Figure 4.

When a crossover network is fine-tuned for the complimentary drivers, very little equalization may be needed. Most of a dedicated "processor" for a particular speaker system is an optimized crossover with some scheme of compressor/limiting added. High quality analog or DSP-based active crossovers can also be configured for a particular driver combination by using either factory-recommended settings or by careful field tuning (see author's papers: AES Journal Vol. 42, no. 4 & 13th AES Conference Proceedings, Session 2D-6).

To confirm how well a crossover works with a speaker system, use the measurement microphone techniques illustrated later in this presentation for equalization at the appropriate frequency/wavelength. If the frequency response in the region of the crossover is relatively smooth under the proper measurement conditions, the crossover and drivers should be relatively well matched. <u>If the crossover is not optimized and the response through its region has a notch, attempts at equalization at that frequency and an octave to either side may be futile.</u>

b. Multiple Source Interference

Imagine placing a measurement microphone feeding an analyzer between two speakers in an anechoic environment. If the microphone element were positioned just three inches closer to one speaker than the other, **Figure 5** (courtesy of Pat Brown, Syn-Aud-Con) would display the measurement using five different levels of resolution.

The deep notches in the TDS measurement indicate the frequencies where the path length difference causes the two signals to be 180^0 out of phase, thus canceling each other. In between the notches, the "humps" indicate 6dB additions. If one speaker were turned off, the average level would be 3 dB lower but the response would be nearly flat from 80 Hz to 12 kHz. Note that the 1/3 octave display does not have adequate resolution to see the comb filtering contaminating the measurement, nor can it be separated from equalizable problems in the speakers' responses.

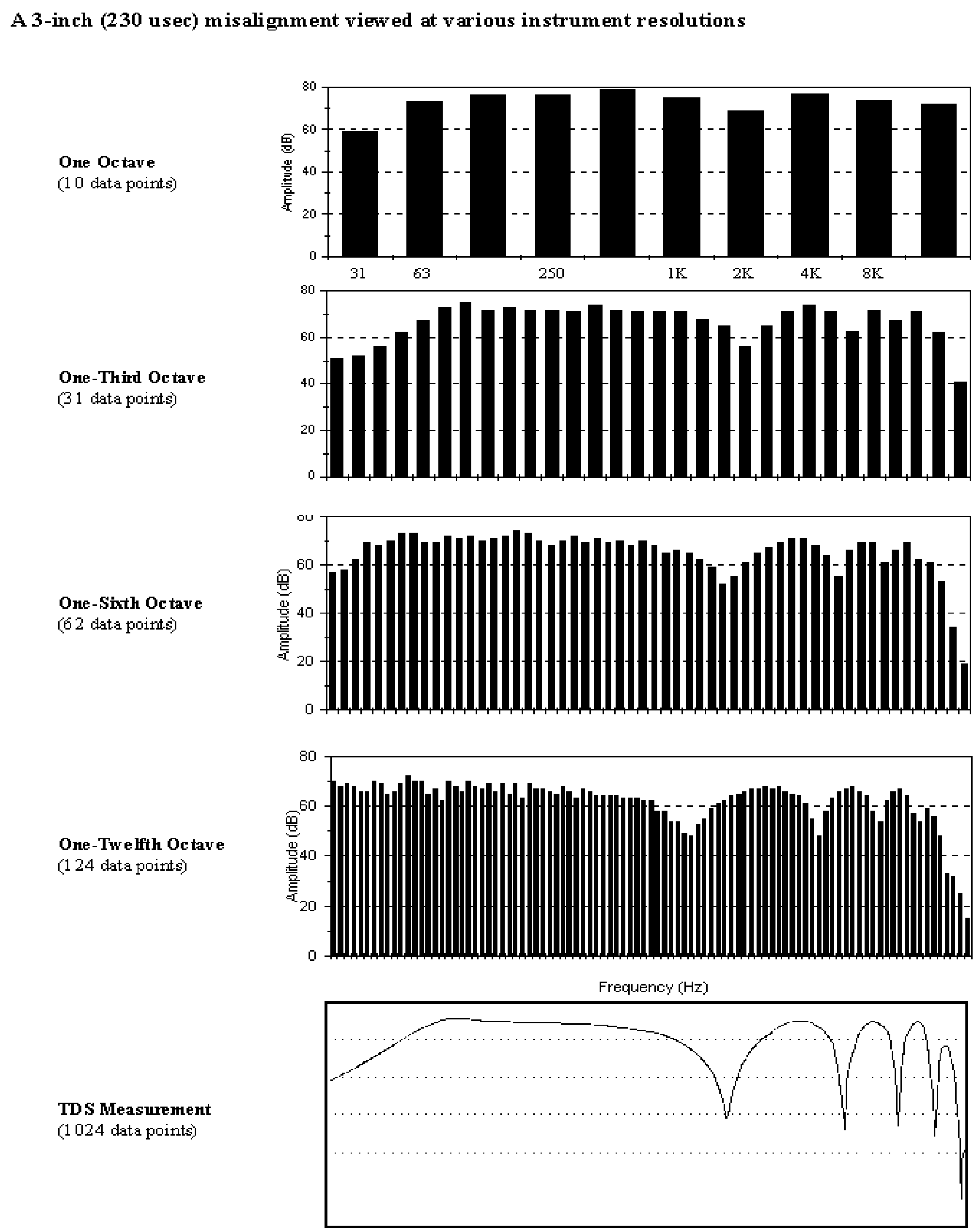


Figure 5.

It should be obvious that one cannot initially equalize a speaker system with more than one device active per passband (crossover section). Multiple-cabinet mutual coupling of low frequency drivers will be addressed as a separate issue later.

c. Strong Acoustical Reflections

In the preceding section (b. Multiple Source Interference), imagine the same measure conditions. But in this case, replace the second speaker with a mirror causing an acoustic reflection from the first speaker to arrive at the microphone 0.23 milliseconds (3 inches) later. The resulting comb filtering would match **Figure 5** exactly. If this difference in time and distance were increased, the notches in the frequency response would move down in frequency and increase in density as frequency rises and wavelengths shorten.

Image now, that the speaker is on the edge of a low stage and the measurement microphone is on a mic stand about 5-1/2 feet above a concrete floor and 30 feet from the speaker. The concrete floor is now our acoustical mirror and would cause a reflection that would have a path length about 2 feet longer than the direct signal from the speaker. To be 180^{o} out of phase, this must be 1/2 of the wavelength of interest, which must be a 4-foot wavelength. With 4 = 1128/X, then X = 1128/4, and X = 282 Hz. This condition would cause deep comb filtering starting at 282 Hz and up, contaminating our measurement. Any hard reflective surface within a few feet of the speaker or measuring microphone can make any attempt to perform equalization at any frequency useless.

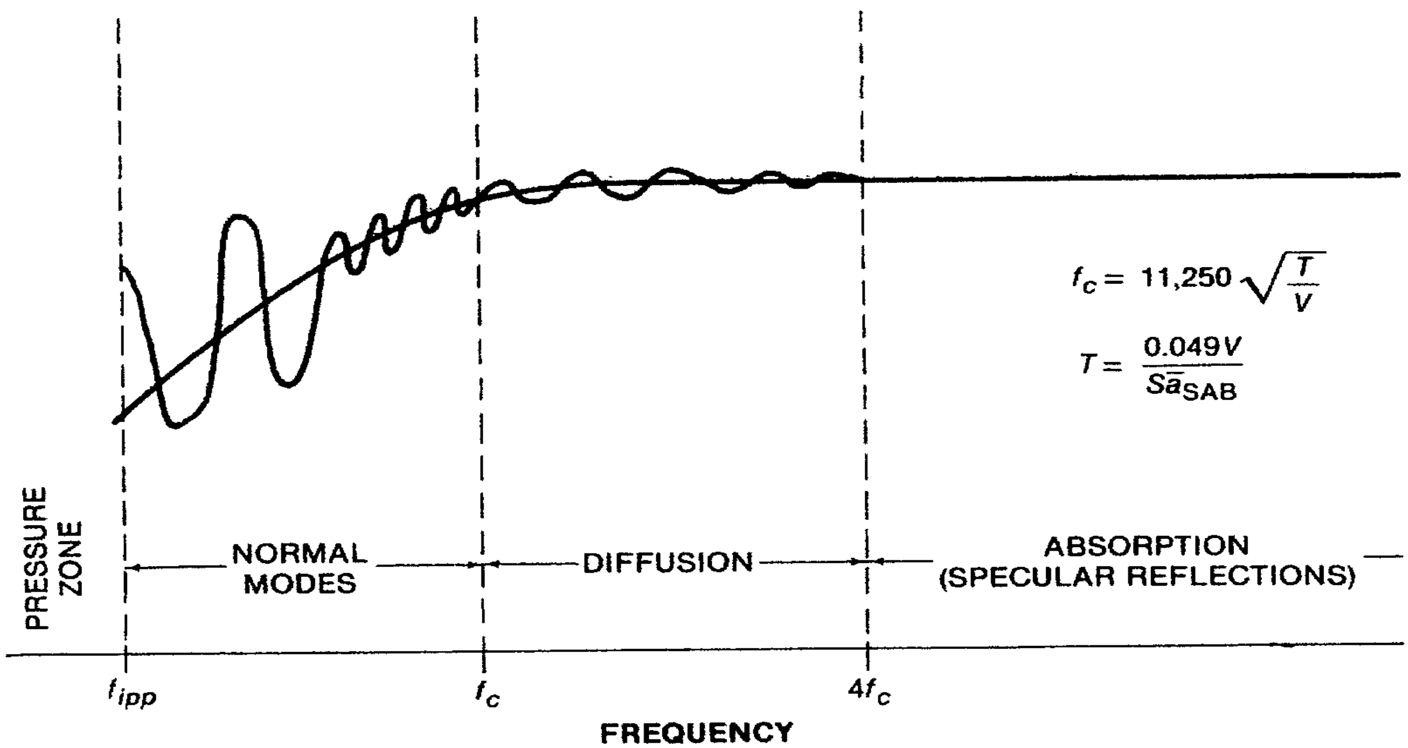


Figure 6.

d. Prominent (Isolated) Standing Waves at Low Frequencies

Figure 6 (courtesy of Bolt, Beranek, and Newman, via Don & Carolyn Davis' *Sound System Engineering, Second Edition* p. 168) plots the results of increasing room mode density with decreasing wavelengths. The four sections can be described as follows:

1) The absorption section (right-most) is where the high frequency response on an RTA's display rolls off in the far field more quickly than the mid's. This is because the high frequency driver's power response declines with increasing frequency and the air and room surfaces more readily absorb shorter wavelengths.

2) The diffusion section (right-center) contains the wavelengths where smooth, even sound fields have their lower limit. These are the frequencies used to aim speakers and determine adequate array coverage.

3) The pressure zone (left-most) section includes those wavelengths that are significantly longer than the room dimensions. In the interior cabin of an automobile, this is a large portion of the bass region, causing the entire enclosure to be pressurized. In other words, a bass note causes the barometric pressure in the cabin to oscillate.

4) The normal modes (commonly called room modes) section (left-center) is the tricky one with respect to equalization. This is the area of wavelengths where the room's dimensions cause drastic differences in the level of a signal with respect to position in the room. If the measurement microphone is at a position where the room causes a null, no amount of equalization boost will help. Likewise in a summed position, reducing level will make the overall average bass in the room lacking. The simplest formula to determine upper end of this tricky area is as follows: $\mathbf{F_c = 3 \times S / RSD}$ (F_c = critical frequency, S = speed of sound, RSD = room's smallest dimension). The bottom of this area is where the wavelength is about 4 times the room's longest dimension, an impractical consideration in all but the smallest rooms.

As an example, let's say we have to equalize a sound system in a room that is 130 ft. long by 75 ft. wide with a 24-ft. ceiling. The critical frequency would be $F_c = 3 \times 1128/24$, and F_c = 141 Hz. The beginning of the pressure zone would be 4 x 130 = 520 ft. or 2.16 Hz. Obviously a pressure zone is not a consideration in this room! Therefore, trying to equalize the sound system via an RTA at any frequency below the 160 Hz 1/3 octave center in this room is not feasible. <u>Once the critical frequency for a room has been determined, no attempt should be made to equalize below this wavelength via an acoustic analyzer</u>. The author recommends walking the room, listening, and tune to the best compromise by ear, or using Don Keele's very-nearfield method mentioned in section V. b.

e. The Room

In the late 1960's Syn-Aud-Con's Don Davis, then with Altec Lansing, coined the term "Room Equalization." When asked recently "What one thing would you take back if you could?" He answered "The term "Room Equalization" because <u>a room cannot be equalized, only the direct sound of the speaker can be equalized."</u> When an empty room fills with people, its temperature and, more importantly, humidity changes. Then the air absorption rate of high frequency energy changes also. But changing the equalization to accommodate that is still tuning the direct sound of the speaker, not the room. Once the sound leaves the speaker, the effect the acoustic environment has on it cannot be separately altered by any piece of electronics. Acoustical problems must be solved acoustically. Have you ever noted that a good quality processed speaker system needs very little EQ, regardless of the room it is in? It's because the controller has properly equalized the direct sound.

IV. SYSTEM PREPARATION

a. Signal Chain Gain Structure

Before equalizing a sound system, the gain structure must be optimized. Please refer to the author's separate paper to set up proper signal-chain gain structure.

b. Signal Chain Electrical Frequency Response

An RTA is quite adequate to confirm that the frequency response of each electronic device's outputs are flat and that the crossover sections have the proper passband for the speakers they are driving. If this is not the case, the problems must be corrected before system equalization can be begun.

c. One Device per Passband

Only one cabinet of each speaker type (mid-high, sub, etc.) will be analyzed and only one driver per crossover output will be energized. This is to avoid the contamination of the measurement by multiple source interference as detailed in Section III b. If, for example, a mid-high cabinet has two 12-inch drivers wired in parallel, disconnect the wiring to one of them. Be aware that if these drivers are used to a relatively low frequency, the lowest 1/2 octave of their passband will be 6 dB lower than when both are connected. If one driver cannot be disconnected, be sure that both are equidistant from the measuring microphone during equalization. If this is not possible, cover one of them with a massive structure that will block its output. This is not a perfect substitute, but is better than trying to equalize with comb filtering masking the true response.

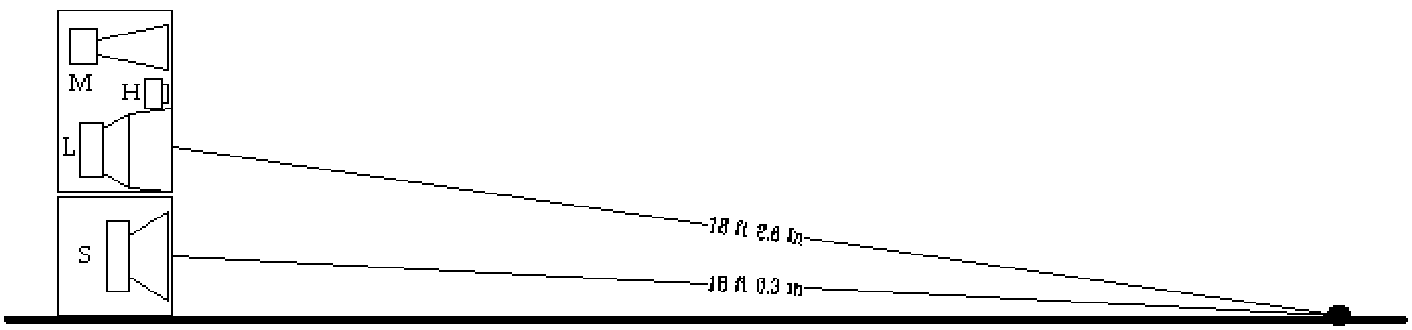
V. BASIC EQUALIZATION FOR A NEUTRAL FREQUENCY RESPONSE

a. General Guidelines

The function of a crossover network, signal alignment delay, and equalization is to develop the inverse transfer function of the speaker. This means to correct the problems the speaker introduces into the signal from the console. The purpose of basic equalization for a neutral frequency response is to provide a flat direct response from the speaker system that matches the output of the console as closely as is possible.

The positioning of the measurement microphone is crucial in order to avoid the problems associated with acoustic reflections as described in Section III c. **Figure 7** shows three measurement microphone positions for equalizing different passband sections of the crossover's outputs. These positions can be done in the sound company's shop to establish a basic neutral tuning for that particular mix of drivers. These positions are also useful in analyzing how well the crossover is matched to the drivers employed. The distances shown will work well with all but the largest of horns (those are 3 ft. square or larger). The recommended absorption material should be 4-inch Sonex or equivalent to be maximally effective.

Measurement microphone position for subwoofer-to-low frequency equalization



Measurement microphone position for low frequency-to-mid frequency equalization

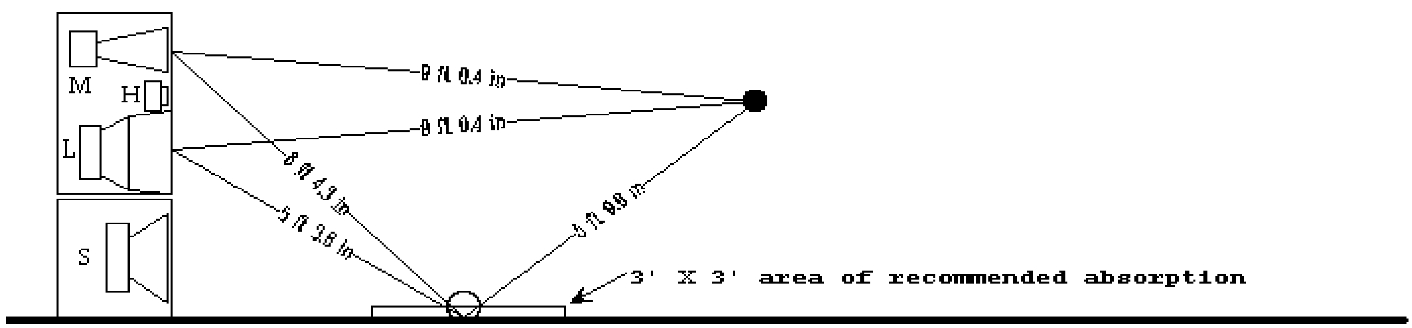
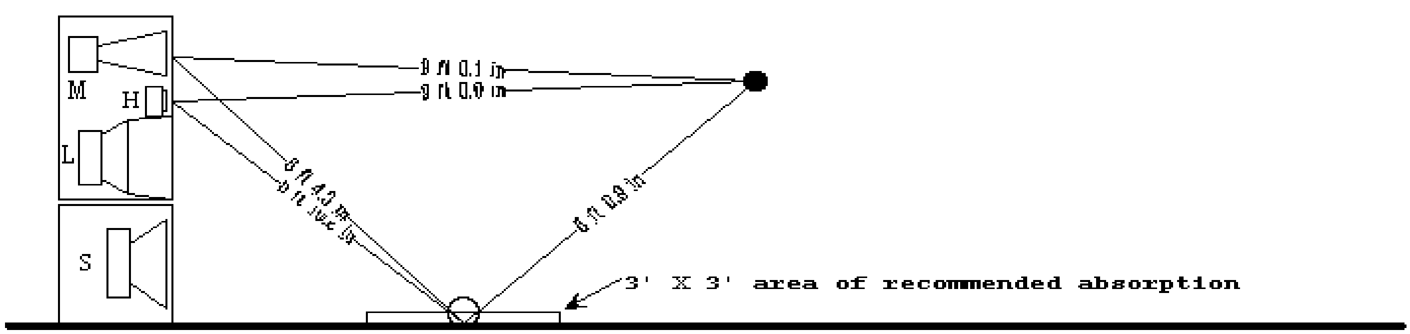


FIGURE # 3: Measurement microphone position for mid frequency-to-high frequency equalization



H indicates the high-frequency driver

M indicates the mid-frequency driver

L indicates the low-frequency driver

S indicates the subwoofer driver

● indicates the measurement microphone position

○ indicates the *approximate* floor reflection point

Figure 7.

Once the microphone has been positioned for each passband of interest, the level of the appropriate amplifier is to be turned up until it matches the adjacent lower passband with a pink-noise source. If possible, use filter cut rather than boost to flatten the response. <u>Any adjustments requiring more than 6 dB of cut or boost at one filter compared to adjacent filters are not equalizable problems. Look for the items described in Section III, structural vibrations and transmission paths, or electrical oscillations in the system's wiring as the source of the problem.</u>

At these distances, the system's response should be tuned to be flat within a few dB from approximately 200-300 Hz to 10 kHz. Above 10 kHz, drastic boosting should generally be avoided, but CD horn EQ (rising boost from about 2 kHz to 20 kHz, starting at 0dB to 15 or 20 dB) is an option the system technician can choose. Any loudness contouring (rock-n-roll bass haystack, etc.) or multiple driver bass coupling will be dealt with later.

b. Very-Low Frequency Equalization

For subwoofer-to-low frequency equalization, the measurement microphone, optimally a small diaphragm omni-directional condenser, should have its slotted grille place directly on a hard, smooth surface like a concrete floor or large finished plywood sheet. No large flat surfaces like walls, ceilings, or stage-fronts should be within twice the distance between the speaker system and microphone. This keeps their potential interference at a minimum. Equalization via a 1/3 octave or parametric filter set may now be done from the middle of the low frequency driver's passband down to the critical frequency as determined in Section III d. 4). Below critical frequency, avoid radical EQ boost or cut, walk the room, and tune by ear using a variety of familiar sources. To be more exacting, one may refer to D. B. Keele's paper on very near-field low frequency driver equalization, *Low-Frequency Loudspeaker Assessment by Nearfield Sound-Pressure Measurement*, published in the Journal of the Audio Engineering Society, April 1974.

c. Low-Mid Frequency Equalization

For low frequency-to-mid frequency equalization, the microphone is about half the distance to the speaker system as above, and centered between the drivers of interest. The recommended absorption and the acute angle of reflection work together to attenuate the acoustic reflection. This is usually sufficient to minimize its effect on the measurement. This position should also be used to check the crossover's smoothness between these two passbands.

d. Mid-High Frequency Equalization

For mid frequency-to-high frequency equalization, vertically adjust the microphone once to be centered between the drivers of interest. If a notch is detected near a fairly high crossover frequency (say 4-5 kHz or above), this indicates a driver misalignment that is very small and very position-dependent. A slight repositioning of the microphone in the proper direction should move this notch upward in frequency until it becomes academic. A notch that does not change or is significantly lower in frequency than the crossover point indicates some problem with the crossover that cannot be equalized. Be sure that a reflective surface nearby is not the culprit! Once again, above 10 kHz, do not radically boost any filters and tune by ear for a pleasing sound using a variety of familiar signal sources.

VI. FULL ARRAY FINE TUNING ON SITE

a. Multiple Driver and Adjacent Surface Mutual Coupling

Mutual coupling is when multiple drivers produce relatively more output at the low end of their response curve than a single driver. This occurs when the drivers are close enough together to have less than 90^o of path length difference for the wavelengths of interest at a given listening position. This also occurs when adjacent surfaces produce a reflection that is close enough in phase to sum with the signal produced by the driver or drivers in the system. Sometimes this is referred to as fractional-space loading (half-space loading is the most common reference).

These phenomena are position-dependent, and are, in fact, what causes the high level portion of lobing. At shorter wavelengths (higher frequencies) and listening positions farther off-axis of the drivers, the phase difference becomes destructive which results in the nulls of lobing. Even with drivers that are touching, mutual coupling over a wide listening area is virtually always below 300-500 Hz.

Mutual coupling can become quite strong when large numbers of drivers or several surfaces are involved. The author has seen mutual driver coupling in very large sound systems, like Woodstock '94 at the north stage, where 15 dB of mutual coupling at 125 Hz had to be accounted for. Tall vertical stacks of 12" or 15" drivers or a long row of 18" subwoofers underneath and across the stage exhibit excellent examples of mutual driver coupling. Coupling in the 150-to-250 Hz range sounds rather "muddy" and is almost always undesirable. It may be that one may choose not to attenuate this coupling in the range below 150 Hz as it sounds pleasing (the rock-n-roll bass haystack). Additionally, at these very long wavelengths, drivers at each end of a large array are so far apart that they no longer couple with respect to certain listening positions.

The author tuned a four-way system with only one driver per passband in Section V. Since multiple drivers are used in sound reinforcement systems, and since large surfaces are almost always adjacent to the system subwoofers, one must adjust the effects of mutual coupling in the array to be used. Since this occurs over a wide listening area only at low frequencies, one must compare the relative overall levels of the mids compared to the lows to determine how much mutual coupling is being dealt with. To look at this with an RTA, the measuring microphone must be positioned on-axis to the stack and on the floor as shown in **Figure 7** for very low frequency equalization. But it must but in the far field at a much greater distance than earlier measurements to see the power response of the system under test. Mutually coupled wavelengths will show a greater power response than those that are not. Once again, be sure there are no hard surfaces (other than the floor that the microphone is positioned against) near the speakers or microphone.

Suppose there is seating that cannot be removed between the speakers and microphone. **Figure 8** depicts a method that avoids the interference of the seating surfaces by using the large flat surface of a 4 ft. by 8 ft. by 3/4" or 1" piece of plywood or similar material as a "PZM" surface. It should have the longest dimension aimed at the speakers. Note that an 8 ft. surface (141 Hz wavelength) will exhibit a 6 dB shelving effect that begins dropping the level at wavelengths of 4 ft. and longer.

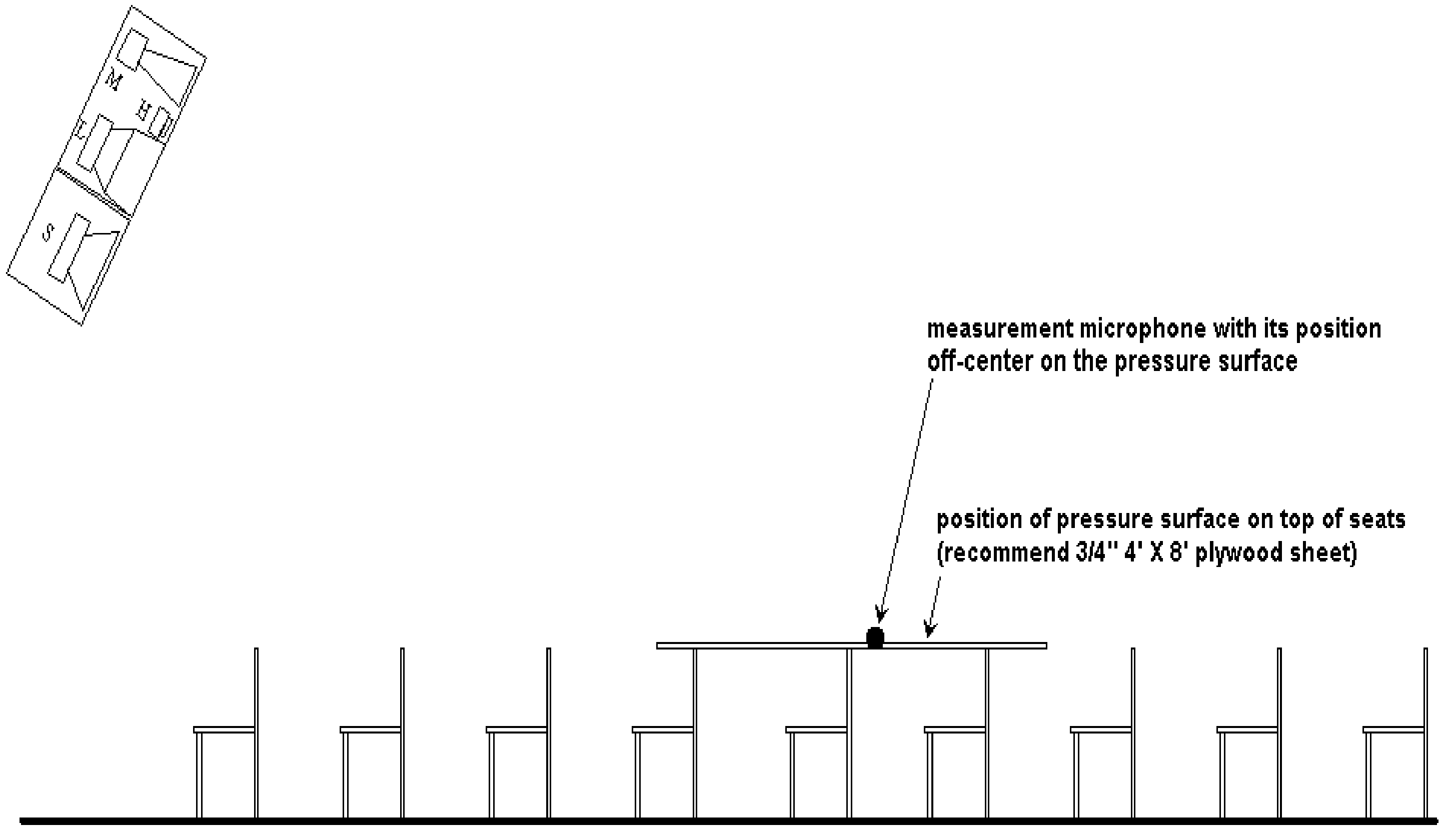


Figure 8.

b. Ring Modes
Supposing a transducer (mic or speaker) in the sound system has a small bump in the response that may only be a few dB higher than adjacent frequencies. It may not even appear on an analyzer's display because it is a very high-Q undamped resonance (a very narrow peak). A ring mode (regenerative frequency) may be detected. These ring modes do not necessarily cause feedback easily, but they do ring, or hang on in time after the original signal source has decayed in the room causing coloration and degradation in the overall sound quality.
It is the opinion of the author, that these ring modes are what is being tuned out of the sound system when a system technician shocks the system with a signal impulse. This is the classic "check, one, two." The technician then reduces the gain at particular 1/3 octave centers to remove these colorations from the signal decay in the room. If one has a TEF-20, or similar FFT-based analyzer, narrow-band sweeps of the passband of interest can be made and the offending frequency range can be visually detected and reduced appropriately, or, without a TEF, sound technicians can keep on doing what works and what they've always been doing: check, one, two......

c. High-Level Driver Distortion Products
For systems that are used at very high levels, like concert sound systems, some equalization may be needed to attenuate frequency ranges where a transducer's distortion components occur most strongly. You may find that these are occurring at the same frequencies that the ring modes in section **b.** above and may be related.

Starting with a system that has had the gain structure properly set first, play the system with a clean, undistorted source like a CD at higher and higher level. Note any frequency ranges that begin to be irritating that were not when playing the same passage of music at a lower level. Using a single EQ filter, attenuate the band you think is the right one until you find it. Attenuate this band slightly (1-3 dB). Be sure that this does not negatively impact the sound quality at the standard operating level for the system.

Be sure not to over-use this technique. You may find more than one area of distortion and must keep in mind that at some point, you will be approaching the limits of the system transducers and will begin to attenuate so many filters that you will essentially be turning the overall volume level down. Obviously, this exceeds the usefulness of this type of adjustment.

d. Air Attenuation of High Frequencies
If the measurement mic is many feet from the drivers during equalization, the frequencies above roughly 5 kHz will begin rolling off with the rate dictated by the mic's distance. One must account for speaker to measurement mic distance and the amount of HF attenuation due to air absorption of the direct sound level. This issue deals with the fact that the human ear/brain finds a flat direct response to very high frequencies may sound very intimate or "in your face" but unnatural to some listeners if the source's distance produces an acoustic character that would dictate less HF content.

You must choose at this point to either allow the roll-off to occur, maintaining a natural long-distant source spectrum with less HF content, or to boost the HF to realize a less natural, but more "hi-fi" sound quality.

e. Program Equalization
Finally the process of program equalization, which creates a pleasing tonal balance of an artistic rather than technical nature, is to be discussed. This is the process where the loudness contouring done in the studio to a recording is applied to a live performance. This is also the place where the "rock-n-roll" bass haystack at the low end, or that airiness or high-end sizzle is tuned into the system's response.

Since this is attempting to produce an effect that originates in the studio, it ought to be done in the same manner. It should not matter if the console is feeding a multi-track recorder, DAT machine, studio monitor speaker system, or FOH (front-of-house) sound reinforcement speaker system. The sound quality and feel of the music should be the same. Therefore, it is the opinion of the author that <u>most of this effect should be done on the mixing console via the input EQ and levels of particular sources rather than at the main front-of-house speaker system's equalizer.</u> For example, if the kick-drum needs to be quite strong, turn its level up and adjust its EQ appropriately for the desired sound, don't boost the entire low-end and then have to re-EQ all the vocal mics. Try a variety of sources, the kick-drum, instrument mics, vocal mics, etc. and adjust their mic'ing technique and input EQ until it produces the desired sound quality. Avoid grabbing for the FOH speaker system equalizer unless there is a problem that is common to <u>all</u> sources. Play familiar CD's and note that virtually no EQ should be needed for it to have the sound quality nearest what was originally intended when it was recorded.

The author believes the reason most FOH mixers don't do it this way in the first place is because so few of the sound systems that a mixing technician encounters are adjusted properly to start with. They are attempting to adjust the speaker system's response and add program equalization simultaneously. Using the methods described in this presentation, this should not be necessary. At Woodstock '94, the author noted that once the north stage sound system was properly adjusted, more than two dozen mixing technicians used the system and none of them adjusted more than a few filters a couple of dB on the FOH equalizers in four days. Virtually all their program equalization was done on the mixing console where it should be done.

Once the program equalization has been done and (hopefully) little has been adjusted on the FOH equalizers, we are ready for the final step.

f. Restoring Unity Gain

Since the initial adjustments for system gain structure, the speaker system's equalizer (1/3 octave or similar) has been adjusted to provide a flat response for the direct (anechoic) sound of the speakers, to compensate for mutual coupling, to suppress ring modes and high level distortion products. Largely cut filtering rather than boost should have accomplished these adjustments. Additionally, CD horn boost EQ above 2 kHz may have been employed as well. Some gain change may be necessary to restore the proper headroom as done in setting the system gain structure before equalization. This is known as restoring unity gain.

To do this, a pink noise source is run through the system with the equalizer bypassed and the overall broadband level noted on an RTA. The measurement microphone's location is not critical, just be sure that the ambient noise level is not significant. Reinsert the equalizer into the signal chain and note any change in the broadband level and adjust it until it matches the bypassed level. If the level needs to go down, adjust the equalizer's input gain. If it needs to go up, adjust the equalizer's output level. If it only has one gain control, use it in either case. Unity gain has now been restored. Note that if there has been any radical equalization, especially narrow-band boost or CD horn compensation, one must take care that the broadband unity gain setting does not allow easy clipping at a highly boosted frequency center. In this case, the sine wave signal at the most boosted frequency may have to be used to set the final gain positions.

The process is now complete and your sound system's performance should exhibit the best frequency response and dynamic range that it is capable of producing with the particular driver compliment employed.