SYNERGETIC AUDIO CONCEPTS

TECH TOPICS

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Editor's Note: Ronald P. Genereux of Cambridge Signal Technologies prepared a paper for the May 1990 AES 8th International Conference, Washington, D.C. - "Adaptive Loudspeaker Systems: Correcting for the Acoustic Environment." Dr. Girzon from England has written glowingly of the SigTech product in recent issues of Studio Sound. An article entitled "Room Remover" in the April 1993 Audio Magazine, had a picture with the SigTech DSP with this lead-in sentence: "The SigTech DSP unit can make many of the acoustical problems in a room disappear?...Fortunately for you, DSP can now provide another solution for about $5,000, which is considerable less than a remake of the room itself would cost." Consider the above in light of the information exchange between George Augspurger and Dr. Patronis! We are grateful to them for sharing. George Augspurger points out in an addendum on page 4 that R. Genereux, ". . .has taken a rational approach. I think he should be given credit for understanding the situation."

Acoustic Comb Filters and Practical EQ

By George Augspurger—Perception Inc.

By now, largely through the efforts of Syn-Aud-Con, audio engineers are aware of the nasty comb filter effects produced by misaligned sound sources or unwanted reflections. All too often we have to deal with a single, strong reflection from a nearby wall or ceiling.

A home music room, for example, may have stereo speakers in front and a couch at the rear where you, the listener, sit. If you lean back, your head is about 14 inches from the rear wall and the mix of direct and reflected sound results in a comb filter with its first null around 250 Hz. If you lean forward for better listening, your head is about 30 inches from the wall and the null slides down to 115 Hz or so. The effect is real and easy to hear using pink noise as a program source.

Any discussion of minimum phase properties or "distortionless" digital room equalization is pointless because a small change in ear location produces a drastic change in the room curve.

But consider a second real-life example: a motion picture dubbing theatre. Acoustic treatment controls wall and ceiling reflections. However, unlike an exhibition theatre, there are no seats to scatter the floor reflection. In many dubbing theatres the result is a big notch around 200 Hz. In this case, however, the notch frequency is relatively stable throughout the working area. Moreover, higher frequency notches are largely suppressed because they fall into the range of the directional high frequency horn.

For various reasons, the offending floor surface may have to remain as is. We are stuck with brute force equalization as the only practical way to smooth out bass response. From a theoretical standpoint, will old-fashioned, minimum-phase analog EQ make things better or worse?

We might expect that it would make things worse. After all, a delay-induced comb filter is symmetrical on a linear frequency scale whereas ordinary analog boost and cut filters are symmetrical on a logarithmic scale.

A simplified computer simulation is shown in Figure 1. At the mixing console the floor reflection is assumed to be 3 dB lower in level than direct sound. The attenuation results from absorption (floor carpet) and some scattering from the console. The 12 dB notch at 250 Hz is worse than a lot of dubbing theatres but the general shape of the curve is all too common.

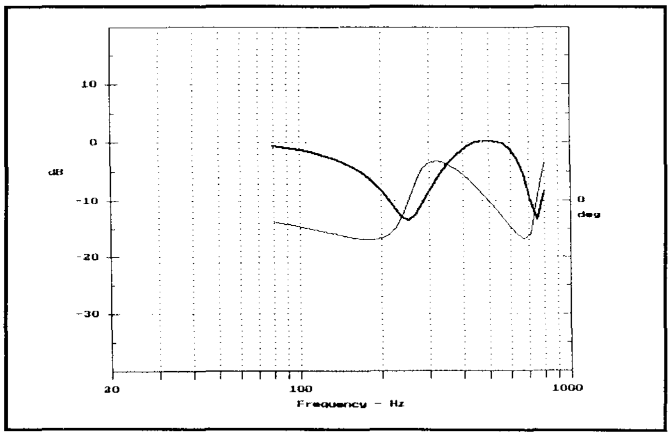


Figure 1

In deference to the limitations of woofers and power amplifiers we might be reluctant to introduce a full 12 dB of equalization, but eight or nine dB may be worth a try. Figure 2 shows a first-try EQ curve. Its magnitude and phase are not exact complements of the room curve, but they seem to head in the right direction.

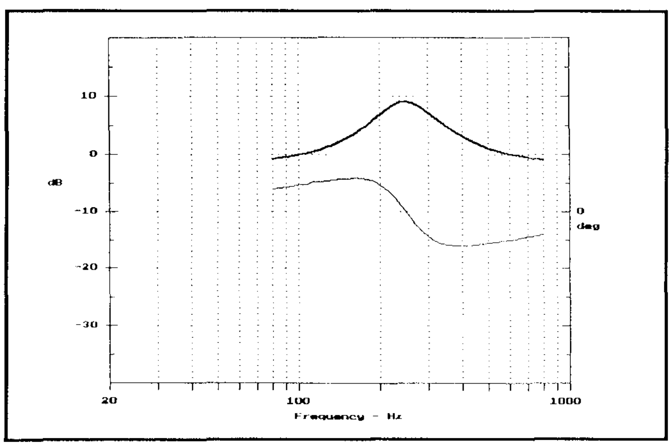


Figure 2

When the two transfer functions are combined (Figure 3), it is obvious that real improvement has been achieved. Magnitude and phase are both much flatter. The next notch of the comb filter at 750 Hz is clearly evident but most movie loudspeaker systems cross over at 500 Hz.

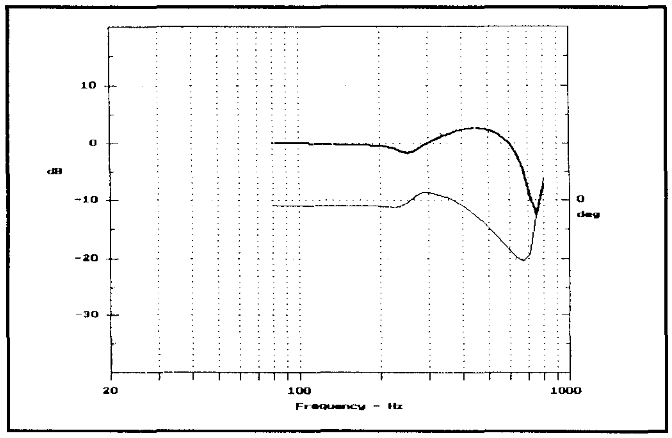


Figure 3

Are these good results, just lucky coincidence? If we repeat the process at higher and higher notch frequencies will one comb filter simply be replaced by another? Not so. For the first two or three octave at least, minimum-phase filters can theoretically equalize a comb filter to near-flat response, as in Figure 4.

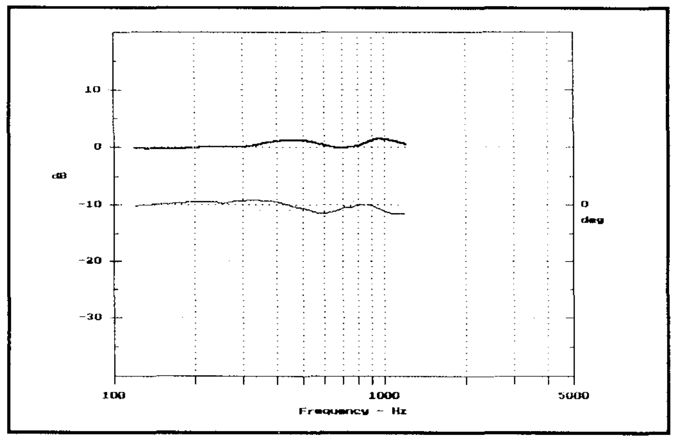


Figure 4

Assuming that my computer model is valid, the conclusions seem to be:

1. If an interference notch is reasonably constant throughout the listening area, minimum-phase equalization can be used to improve response, in both the frequency and time domains.

2. Conversely, brute-force electrical boost strains loudspeakers and eats up amplifier power. You should first do as much as possible with the acoustical tools available and then use electronic equalization as sparingly as possible.

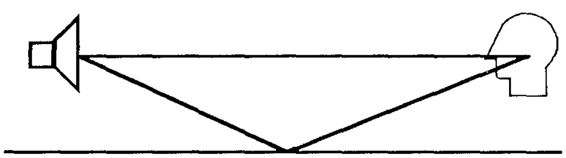
Can a Single Boundary Reflection be Equalized?

By Eugene Patronis, PhD—Georgia Tech, School of Physics

This appears to be a simple straight forward question for which one might think there exists an equally simple straight forward answer. Wrong!! If the question is enlarged to be, can a reflection from a single boundary be equalized for all listeners in a space independent of their specific locations? The answer is an unqualified no. If the question is diminished to be, can a single boundary reflection be equalized for a listener at a specific point? The answer is maybe. All of the foregoing deserves an explanation and such will be attempted in what follows.

In general, equalizable anomalies are those which display minimum phase behavior. Mathematically this means that the zeroes as well as the poles are located in the left half of the complex S plane. Airborne reflections always occur after the direct sound and consequently the transfer function describing the reflection alone involves a delay which is not minimum phase. The combination, however, of a single reflection with the direct sound at a single listening position may still be minimum phase. The author was unaware of this special circumstance until it was pointed out to him by G.L. Augspurger.

The sketch below illustrates the physical situation under consideration.



The listener in the sketch only has the use of his left ear. He foolishly spent too many hours on the firing range without benefit of hearing protection. The coaxial loudspeaker is actually mounted in an appropriate unvented enclosure but I didn't take the time to draw that. I want to spend time on the firing range myself along with the benefit of adequate protection of course. The distances are such that the attenuated reflected sound arrives two milliseconds after the direct sound having undergone a broadband attenuation of 3 dB. We will be very generous in describing the loudspeaker response by giving it a second order Butterworth high pass and low pass characteristic with -3dB points at 20 Hz and 20kHz. This is much easier to do on paper than it is in practice.

We will start by describing the transfer function for the direct sound alone.

$$j = \sqrt{-1}$$

$$S = j\,\omega$$

$$\omega = 2\pi f$$

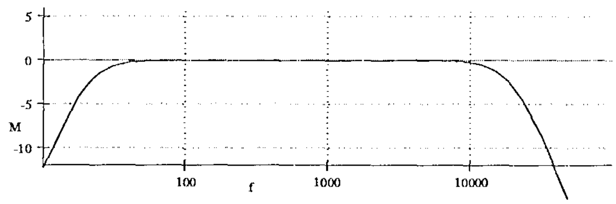
$$f_0 = 20$$

$$\omega_0 = 2\pi f_0$$

$$H = \frac{10^6 S^2 \omega_0^2}{\left(S^2+\sqrt{2}\,S\,\omega_0+\omega_0^2\right)\left(S^2+\sqrt{2}\,S\cdot 10^3\omega_0+10^6\omega_0^2\right)}$$

$$M = 20\log\left(|H|\right)$$

The amplitude response is displayed in the depicted graph.



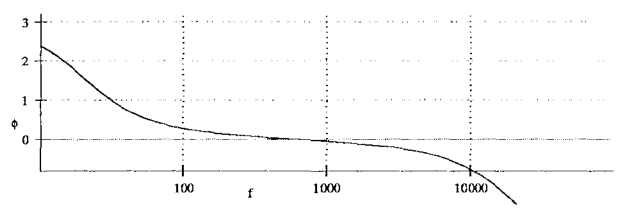
In addition to this idealized amplitude response, we also need the attendant phase response to complete the picture.

$$\phi_1 = \pi$$

$$\phi_2 = -\arctan\left(\frac{f-\frac{1}{\sqrt{2}}f_0}{\frac{1}{\sqrt{2}}f_0}\right) - \arctan\left(\frac{f+\frac{1}{\sqrt{2}}f_0}{\frac{1}{\sqrt{2}}f_0}\right)$$

$$\phi_3 = -\arctan\left(\frac{f-\frac{10^3}{\sqrt{2}}f_0}{\frac{10^3}{\sqrt{2}}f_0}\right) - \arctan\left(\frac{f+\frac{10^3}{\sqrt{2}}f_0}{\frac{10^3}{\sqrt{2}}f_0}\right)$$

$$\phi = \phi_1+\phi_2+\phi_3$$



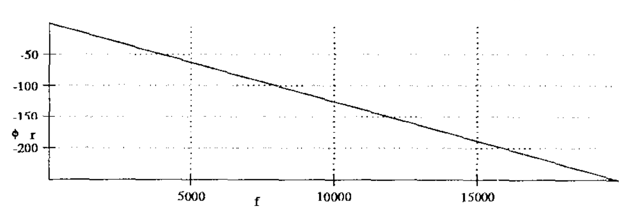
The reflection alone has a transfer function

$$R = \frac{1}{\sqrt{2}}e^{-S\cdot 2\cdot 10^{-3}}$$

and an associated phase response

$$\phi_r = -2\pi f\cdot 2\cdot 10^{-3}$$

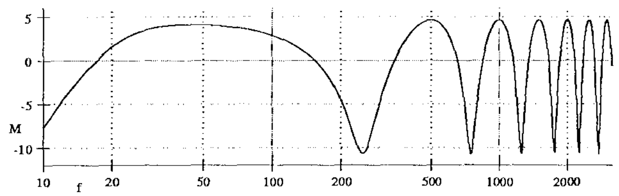
The amplitude response is singularly uninteresting because it is just a horizontal line at -3dB. The phase response is worthy of note and is depicted in the next graph.

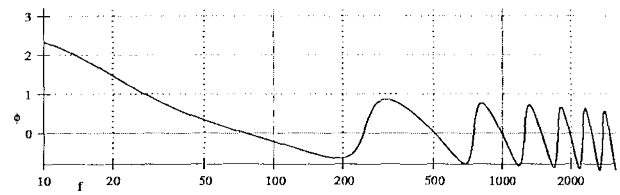


Now for the combined response. The transfer function describing the combination of the direct and reflected sound appears as

$$H = \frac{10^6 S^2 \omega_0^2}{\left(S^2+\sqrt{2}\,S\omega_0+\omega_0^2\right)\left(S^2+\sqrt{2}\,S\cdot 10^3\omega_0+10^6\omega_0^2\right)}\left[1+\frac{1}{\sqrt{2}}e^{-S\cdot 2\cdot 10^{-3}}\right]$$

The amplitude and phase responses associated with this transfer function are given in the next two graphs.

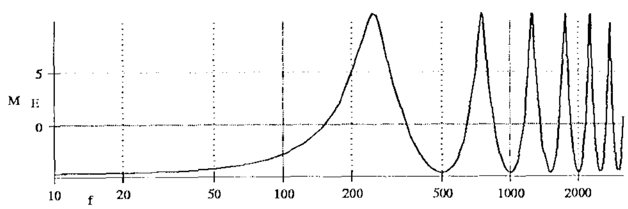




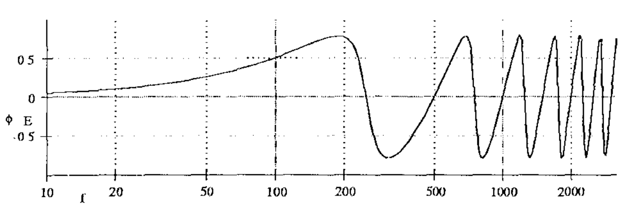
The comb filtering brought on by this combination is clearly evident in both graphs but the phase response is still minimum phase and hence is capable of equalization. The equalizer response necessary to do this has a transfer function which is the following expression.

$$H_E = \frac{1}{1+\frac{1}{\sqrt{2}}e^{-S\cdot 2\cdot 10^{-3}}}$$

The amplitude response of this is given by



and the phase response appears as



If one examines the overall transfer function including loudspeaker, reflection and equalizer it is found that the original direct sound is restored at the listener's single ear.

The question now becomes whether or not the required equalizer can be constructed and what demands it will place on the loudspeaker.

When one examines the pole zero diagram of the required equalizer transfer function, it is found that there are no zeroes and that the poles are complex and are located at the points $[-500\ln(2^{1/2}) \pm jn500\pi]$, where n is 1,3,5,7,...all odd integers. In principle this is physically realizable as all of the poles have negative real parts and hence lie in the left half plane. The fact that an infinite number of poles would be required for exact performance can be relaxed in practice since only a finite bandwidth, i.e., the audio spectrum is to be dealt with. The actual number required is quite large, roughly 80, especially when viewed from the vantage point of the man who must construct and adjust the equalizer. Additionally this equalizer will be boosting the loudspeaker drive signal by about 10 dB at certain frequencies while cutting the drive signal by about 5 dB at other frequencies. This in itself may place unreasonable demands on the amplifier loudspeaker combination.

A more modest equalization than that required for exact correction may indeed, however, be both reasonable and advantageous to apply. In that event, the techniques embodied in ACE would be the way to proceed.

At the outset, it was assumed that the listener only had one active ear. This was necessary as this single reflection could only be equalized at one point. Furthermore, it was assumed that the reflection operated independent of frequency which is hardly ever the case.

This example, if nothing else, points up the necessity for precision measurement both of amplitude as well as phase and the almost absolute necessity for determining if measured anomalies are indeed of a minimum phase character. Additionally, I will not be surprised if we soon include two channel measurements with dummy heads in our assessments of sound system performance as well as adjustment in the professional arena.

Addendum: Ronald P. Genereux himself has taken a rational approach, emphasizing that the important thing is to develop methods of analyzing and averaging which correlate with subjective impressions over a reasonable listening area. I think he should be given credit for understanding the situation. however, a number of patents have been issued for similar processes (mostly to the big Japanese audio manufacturers) and many of these suggest that "perfect" correction of the loudspeaker-room interface is now possible.

I was amused by Gene's single-ear example. I have long complained about the techniques used by many practitioners in equalizing monitor loudspeakers; in my view their efforts would only be appreciated by a one-eared mixing engineer with his head locked in a brace.

For that matter, why bother with equalization at all? You may remember that Amar Bose used a computer program to "prove" that the response of his first loudspeaker design was subjectively indistinguishable from the original recording session.

G.L. Augspurger, 4-16-93