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Quasi-Anechoic Loudspeaker Measurement Using Notch Equalization for Impulse Shortening

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ABSTRACT

The length of the impulse response of a typical piston driver is largely determined by the characteristic second-order high-pass response of the driver. This time response makes anechoic (i.e. gated) measurement difficult in non-anechoic environments, as reflections must be suppressed to returns of 30 ms. or longer.

This paper outlines a quasi-anechoic frequency and phase response modification technique using a tuned notch, or band-cut, equalization that shortens the impulse response and allows correct full-range speaker measurement in moderately sized non-anechoic rooms.

1. INTRODUCTION

It is desirable to measure loudspeakers in an environment in which reflections are not easily controlled. Swept sine wave systems using windowed impulse responses are often used to measure speaker response. Their measurement window must enclose sufficient time to allow all significant and relevant energy to reach their analyzers, but not be open so long as to include reflective contamination.

A typical low-frequency loudspeaker produces a ringing response to impulsive stimulation. As this ringing energy corresponds to a particular frequency response, the majority of this energy must be captured for accurate measurement.

Papers and experiments by Fincham, Bachman, Benjamin, et al, have featured methodologies to address the shortening of the duration of this ringing [1 – 4]. These works are discussed in more detail later.

A new methodology for performing quasi-anechoic measurement is outlined herein. This methodology uses an open gate time as short as 6-7 ms. This short gate time allows low-frequency measurement of either an enclosed speaker system or a speaker on a baffle of reasonable size in a non-anechoic room.

In its first instance, the subject methodology involves equalizing a band-cut (notch) response into the speaker amplifier's stimulus signal. This equalization also appears in a second reference channel derived from this amplifier signal. These two signals connect to a dual-channel analyzer. The reference signal is then subtracted from the speaker response signal (in dB) to obtain a corrected anechoic characteristic.

This is a response-modification process similar to those seen in prior art with perhaps some implementation and accuracy advantages.

2. SIMULATED TYPICAL SPEAKER MEASUREMENT

A somewhat typical loudspeaker's low-frequency response was simulated in SPICE by implementation of a second-order hi-pass filter design. The filter's response is shown in Figure 1. This hypothetical speaker has a resonant frequency of 32.5 Hz and a Qts of 0.68.

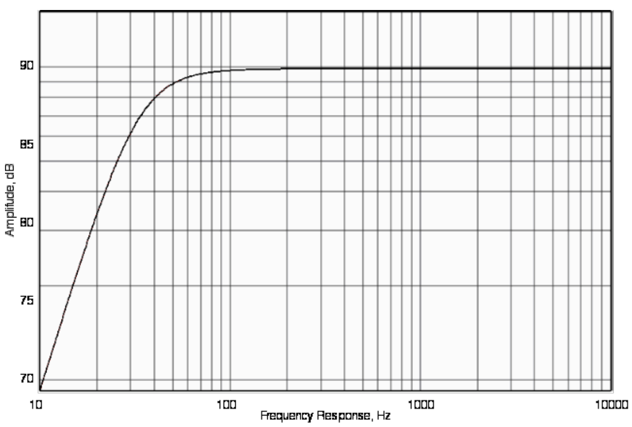

Figure 1. Simulated anechoic speaker response

A speaker like this would produce a transient response like the one shown next in Figure 2. For this example, the pulse width is 2 ms. and the period is 89 ms. The driving pulse height is 5 Volts.

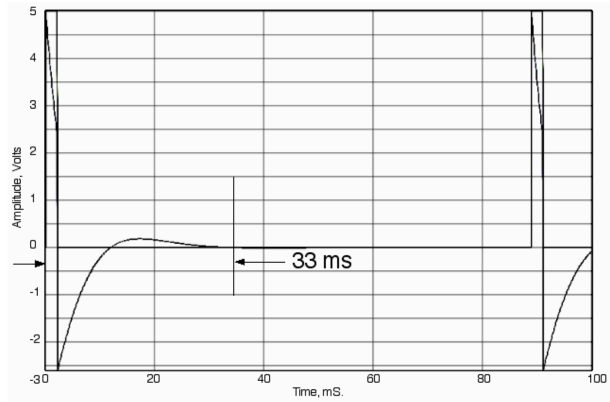

Figure 2. Transient response of simulated speaker with Qts of 0.68

Note that there is ringing from the high pass filter with significant energy for nearly 33 ms from the start of the impulse. This ringing can be much longer for a filter of higher Q, perhaps 75-100 ms or more.

Valid speaker measurement in a non-anechoic room would not allow the use of gate times of 30 ms or more. Shorter gate times, without some signal processing, would introduce error due to a failure to capture all of the significant radiated energy.

3. PRIOR ART IN QUASI-ACOUSTIC MEASUREMENT

There have been excellent papers proposing methodologies for shortening a loudspeaker's impulse response for accuracy in gated measurement.

Laurie Fincham proposed adding a high-pass filter to the system response prior to analysis, gating the response after the now-shorter transient "tail" [1]. This added filter has a corner perhaps 10 times that of the speaker under test, and thus a much shorter transient response tail.

The response of the speaker under test is corrected in post processing. The author has a concern that LF information may be lost when the response is restored after gating has removed possibly critical information.

Juah Backman proposed flatting the response to near DC with a compensating filter [2]. This filter eliminates the high-pass corner and extends the response rolloff to very low frequencies. This stretches the transient ringing to almost infinity, but the energy level of this now very low-level ringing can be ignored.

The Backman method can produce accurate data, but because of the large signal boost at low frequencies, a very quiet measurement environment is required.

A very extensive look at these two methods is seen in a paper from John Vanderkooy and Stanley Lipshitz [4].

4. REFERENCE CHANNEL SPEAKER MEASUREMENT

The prior art and the subject methodology use a frequency modification that must be removed to obtain final data. Additionally, the subject methodology depends on phase modification to fully shorten the impulsive return.

The author uses a parametric notch equalization to shorten the impulse response, and a reference channel to correct acoustic path data. A setup that would demonstrate this system is shown in Figure 3.

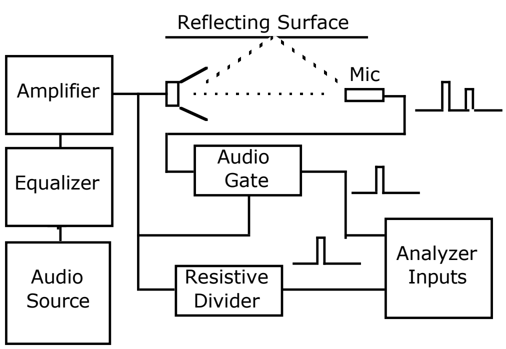

Figure 3. Block diagram of reference channel measurement setup (impulse with FFT)

In the above diagram, the impulse signal is generated and passed to a parametric equalizer. This equalizer is carefully tuned to shorten the acoustic path's impulse response, which is defined not only by the high-pass characteristic, but also other loudspeaker spectral anomalies. This equalizer presents its signal to the speaker's driving amplifier.

The microphone picks up the acoustic signal from the speaker, and then from the room. An audio gate suppresses the room's return. This gate does not affect the reference channel's signal. Note again that this gate is often provided by the analyzer's software.

The dual channel analyzer produces a frequency domain result and the reference channel result in dB is subtracted from the acoustic signal, also in dB.

If the gate has not closed on any substantial signal directly from the speaker to the microphone, the resulting response is that of an anechoic measurement.

5. SUPPRESSION OF RINGING WITH PARAMETRIC EQUALIZATION NOTCH

Now that a reference channel is set up, any change made with our parametric equalizer affects both the acoustic path and the reference channel path.

Figure 4 shows the responses of a high pass filter simulation with Q=0.68. The reference path and the anechoic path responses each show a 2nd order equalization notch of approximately 20.6 dB, all simulated in SPICE. The notch shown is used to suppress the long response tail.

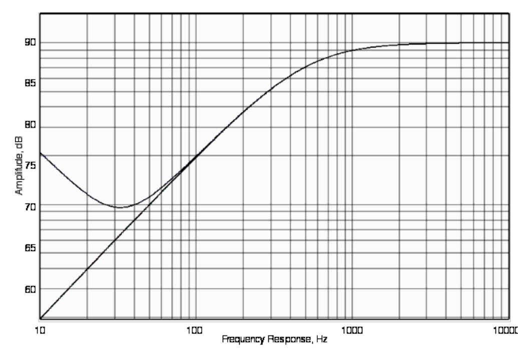

Figure 4. Reference and acoustic path responses

Note that the resulting response looks much like that of the Fincham method. The subject method, however, uses a tuned notch matched to the high pass filter's order resonant frequency, and Q with simple observations of the transient response. No prior knowledge of the measured high-pass filter's (i.e. speaker's) characteristics is required.

If amplitude suppression were the only mechanism involved, any processing using "cut-gate-restore" would not produce a net gain. However, the ringing is suppressed much more than notch depth of 20.6 dB. We will see that additional suppression on the order of 15-20 dB or more is obtained by properly tuned cancellation. This is due to the most important phase cancellation that comes with notch equalization, discussed in detail later.

The transient response simulation shows the suppression provided by the tuned notch equalization. Three signals can be seen in Figure 5. The tallest peaks are that of the reference channel. The shortest peaks are those of the acoustic path response.

Please note that the 2ms impulse with an 89 ms. period does not enable high resolution, full range frequency response testing. This impulsive signal is usable for illustration and, more importantly, for real-time adjustment of notch equalization.

Ringing of the high pass filter is suppressed by careful “tuning” of the parametric notch. This allows a gate open time of approximately 5.5 ms. Looking at Figure 5, it would almost appear that the high pass filter’s ringing has been “moved” to the reference channel.

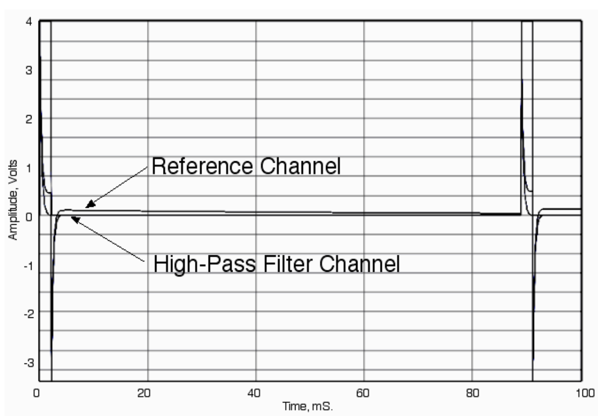

Figure 5. Transient response of acoustic and reference paths with parametric notch

The reference channel now has the long transient “tail” but as this is only an electrical signal, there is no need to put a short window around its data.

Figures 6 and 7 show this attenuation more clearly, showing only the “acoustic” high-pass path response. The images are that of a 2 ms. impulsive signal that has passed through a 2nd order high-pass filter and a notch equalization. As before the equalizer has been carefully tuned to flatten the transient response beyond 5.5 ms.

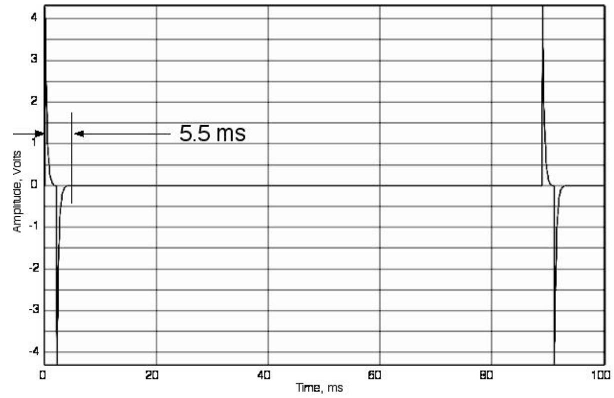

Figure 6. Transient response of notch equalized acoustic path (5.5 ms. gate possible)

The reader could be concerned that when this notch equalization has been corrected for final data presentation, this 20 dB of amplitude reduction could be hiding a suppressed transient tail that would invalidate the process.

To evaluate this possibility, the peak voltages in the SPICE simulation shown below was “diode clamped” to allow the simulation to expand the vertical scale by a factor of 100, or 40 dB. As one can see, energy beyond the 5.5 ms. period has been virtually eliminated even at this low amplitude.

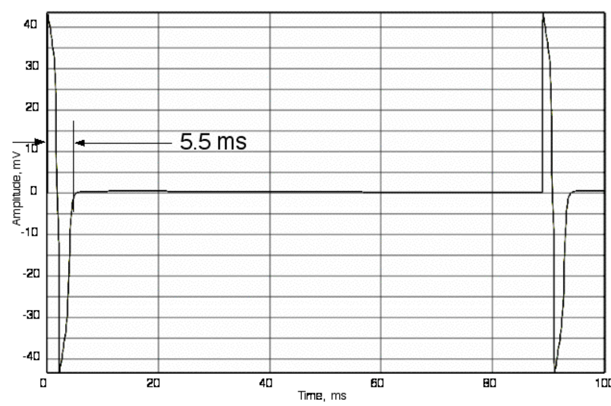

Figure 7. Transient response of high-pass filter with notch equalization, Q of 0.68, vertical scale expanded 100X

Next is shown the transient response of a new filter with the same resonance as before but with a Q of 2.04. The author has never seen a speaker with a Qts of 2.04, but this example was tested to see if ringing of this duration could be suppressed by the subject methodology.

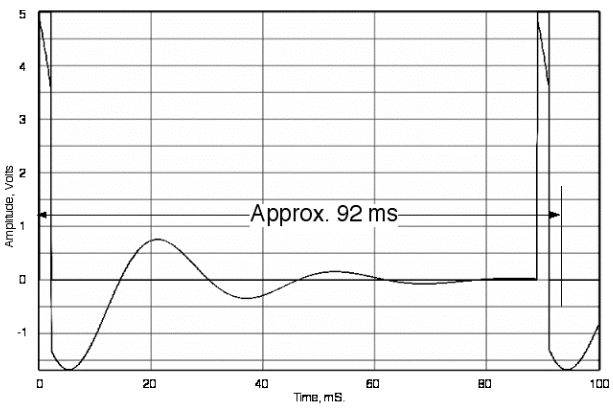

Figure 8. Transient response of filter with Q of 2.04

In Figure 9 below, this ripple energy has been virtually eliminated by tuned notch equalization. For this case, the notch depth is approximately 30 dB, with only the equalization's depth and Q changed from those of the previous example.

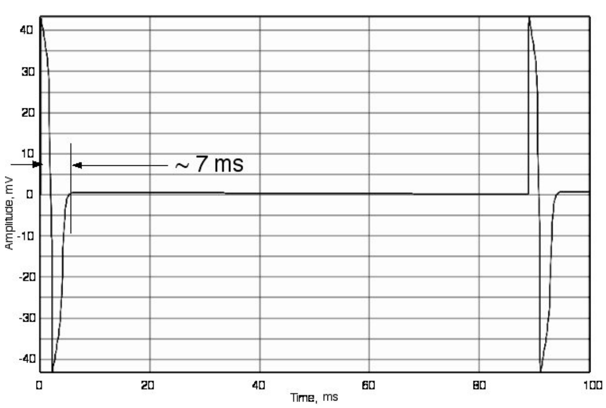

Figure 9. Transient response of filter with notch equalization, Q of 2.04, vertical scale expanded 100X

One can see that the impulse "tail" has been shortened from approximately 92 ms to about 7 ms.

6. PHASE COMPENSATION

Smoothing of the transient response baseline to this degree happens because the notch can be tuned to smooth the phase response of the high-pass filter around its resonance. Eric Benjamin included phase compensation among suggested methods to shorten transient response [3]. The author has found notch filters ideal for this purpose.

The phase property of a notch filter is unique among common analog filters. Its phase properties near its center frequency can rather well compliment the phase response of a high-pass filter

Simulations the author has tested show that the optimum notch equalization for phase compensation matches the resonant frequency of the high-pass filter and has a Q of a value that closely approximates that of the filter's Q (simulated speaker's Qts).

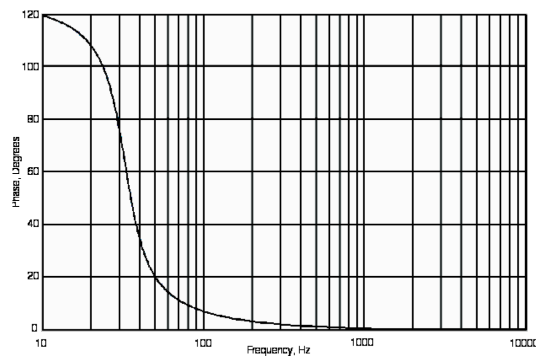

Figure 10. Phase response of a 2nd order high-pass filter with Q=2.04

Figure 10 shows the phase response of the high pass filter. Please compare its response around resonance with the phase response of the notch filter, shown in Figure 11 on the following page

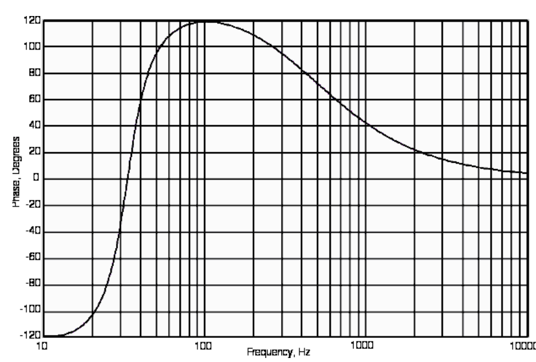

Figure 11. Phase response of 2^{nd} order notch filter with Q=1.99

Note that the notch filter's response around resonance demonstrates a similar, but opposite phase characteristic (the scales of the preceding two figures are not precisely matched, but this is a matter of gain, which is to be adjusted for best null).

A smooth frequency response is produced by the combination of the two filters. Also note in Figure 12 that the phase response (starts high in Figure below) varies smoothly throughout the frequency range. Note that the logarithmic plot bends the phase curve for this Figure.

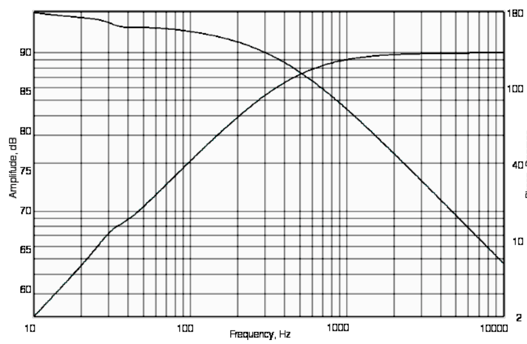

Figure 12. Amplitude and phase response of tuned system with high-pass Q= 2.04

The phase plot shows let than 180 degrees of variation throughout the range from 10 Hz to 10,000 Hz. This phase response represents very short impulse times.

As the speaker's Q is increased, a notch of greater depth is required to provide the best phase-compensation effect. As seen with the Q of 2.04, illustrated on the previous page, a notch depth of approximately 30 dB was needed.

7. DUAL EQUALIZER SETUP

This equalization notch method does not boost VLF to a great degree and would be thus less sensitive to moving air that some similar methods. However, because the acoustic path is attenuated about 30 dB at 35 Hz (for this high Q example), the measurement room must be quiet in that region.

Inserting the notch filter later in the signal stream can practically eliminate the notch depth as a noise concern. If the user can implement a dual-equalizer setup, or use a suitable two-step process, the room noise rejection can be significantly improved compared to that of a non-equalized system.

Figure 13 shows a dual-equalizer setup that performs the response modification after the microphone signal is obtained, but prior to signal gating. Using this setup, the room noise is reduced just as the signal is attenuated. Note that the noise performance of the Fincham method would be likewise improved by this method.

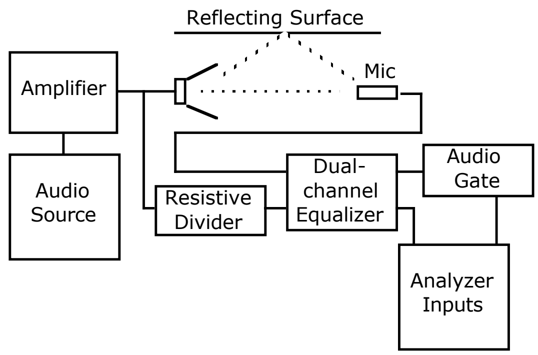

Figure 13. Block diagram of reference channel dual equalizer measurement setup (gate may be implemented in analyzer)

If the two equalizers are sufficiently matched, there is little functional difference between the data set produced by the two approaches.

Additionally, in the single equalizer approach, the speaker source voltage is reduced to a low level at resonance. With a low source voltage, suspension thixotropy could prove problematic. This problem is also eliminated by use of the dual equalizer approach.

For every filter's (speaker's) resonant frequency and Q, the notch parameter is adjusted to null the unwanted transient response that is found on the acoustic signal path.

Shorter gate times can be realized with tunings that use greater notch depth. However, using very deep notches could reduce the user's ability to differentiate between simple suppression and actual tuned cancellation.

Because high Q speakers require a high Q null tuning, the reference path response will exhibit a ringing transient tail that extends 90 ms or more. The significant energy of this reference signal's impulse response must be fully captured. If one is using an impulsive stimulus with an FFT to measure the reference channel, one must utilize a square or nearly square window approximately equal in width to the period of the impulses.

If the acoustic measurement and reference channels are captured simultaneously using an impulsive stimulus-dual channel FFT setup, the reference channel path should include a signal delay to match the delay of the acoustic path. This helps optimize the likely simultaneous FFT windows for best accuracy.

Response data acquisition methods using post-processing are recommended for response testing once the equalization is tuned. This allows one to separately window the desired impulse responses of the measurement and reference channels. Also, methods using post-processing typically provide significant signal to noise performance improvements over impulsive stimulus-FFT measurement method.

8. THE SPICE SIMULATION

It is instructive to "tune" the equalizer using the SPICE simulation. Toward this end, this simulation is now discussed

Figure 14 represents the SPICE network implemented by the decks shown in the appendix. The simulated speaker (a 2nd order high pass) is implemented as C1, L1 and R2. The notch equalization is implemented as R1, R3, C2 and L2.

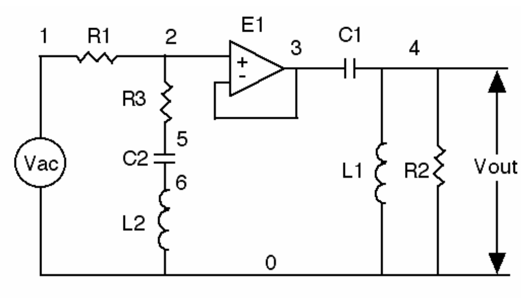

Figure 14. SPICE network used for simulations

The reader is invited to implement this circuit in a simulation program and manipulate its values. The author finds it most interesting to watch the residual "tail", see it virtually disappear and then change phase as the notch circuit is tuned.

9. ACTUAL LOUDSPEAKER MEASUREMENT

There are substantial differences between testing in simulation and testing a real speaker in the author's basement. A vertical flat baffle, moderately larger than the smallest IEC speaker baffle, is mounted below an 8 foot ceiling. The floor and ceiling produce signal returns delayed from the direct signal by less than 7 ms.

High-frequency reflections are managed by absorption and deflection. Low frequencies are slightly absorbed and are somewhat dispersed by their spherical nature. More importantly for lower frequencies, the vertical baffle presents the edge of a dipole speaker to the floor and ceiling, reducing the energy going directly at, then reflected back from these surfaces.

More troublesome is that longer gate times allow the effects of room modal energies to influence data. It is these modal effects that have proven very detrimental to the author's speaker measurement.

An automotive 12-inch speaker was mounted on the baffle. The measurement microphone was positioned 0.5 meters above the front surface of the baffle. The speaker has a specified resonance of 29.2 Hz and a Qts of 0.63. Parametric tuning would suggest that the speaker's actual resonance was just above 40 Hz. The resonance of the new speaker, still in an initial state of break-in, was measured after measurement to be 33.5 Hz.

Given the quiet conditions of the author's basement, a setup as shown in Figure. 3 was used for testing. There was some difficulty finding a suitable parametric equalizer, as the Q adjustment must be tunable to match the Qts of the speaker. The equalizer chosen was a DSP device with just enough Q adjustment range for this speaker. Please note that it is necessary to use a minimum-phase equalizer for this tuning process.

Impulses of 1.5 ms and 0.2 ms. widths with a period of 100 ms were used for equalizer tuning. The wider impulse is used to stimulate the speaker's resonance region and the narrow pulse is used to check for higher-frequency ringing.

High-Q response dips as well as peaks can produce troublesome speaker ringing. Tuning may involve bandpass as well as band-cut tuning.

Shown below in Figure 15 is the impulse response of the speaker and measurement system before equalization. Note the large amount of energy that is present beyond a preferable gate time.

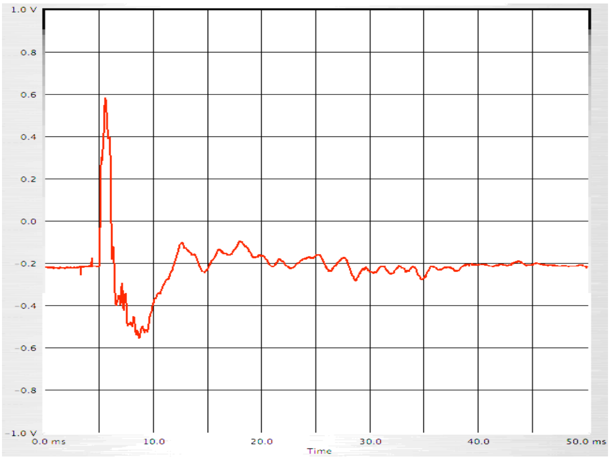

Figure 15. System impulse response without EQ, 0.5 meters from baffle (5 ms per division)

One can see that the impulse return features much information delayed well beyond a desirable gate period. We will need to remove all real speakers ringing, and reject the train of reflection clutter that follows.

Essentially, tuning the parametric equalizer to minimize the transient ringing can separate what the speaker is doing from what the room is doing.

In the author's small space, it is difficult to discern from the signal what is speaker and what is room. A second microphone was placed in the very near field of the speaker to emphasize just what the speaker was doing. This allowed equalizer tuning that looked much like that of the simulation.

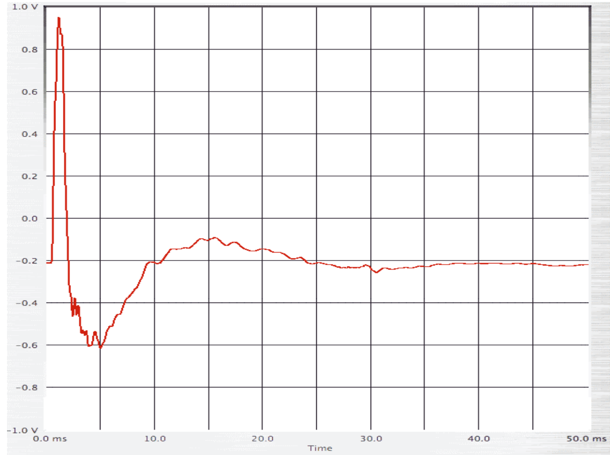

Figure 16. Close mic transient response before equalization (5 ms/division)

Note in Figure 16 above the expected transient ringing from the speaker's high-pass nature. Also not the presence of other ringing frequencies.

After tuning the equalization notches, the impulse response was as shown below in Figure. 17. This could allow a gate time of approximately 6-7 ms.

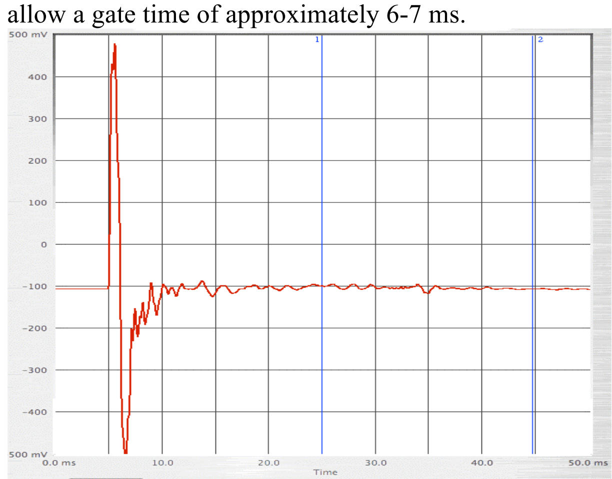

Figure 17. Close mic transient response with tuned notch equalizations (5 ms/division)

Equalization included not only tuning to manage the high-pass effect ringing, but also three other notch tunings at approximately 400 Hz, 630 Hz and 1.6kHz.

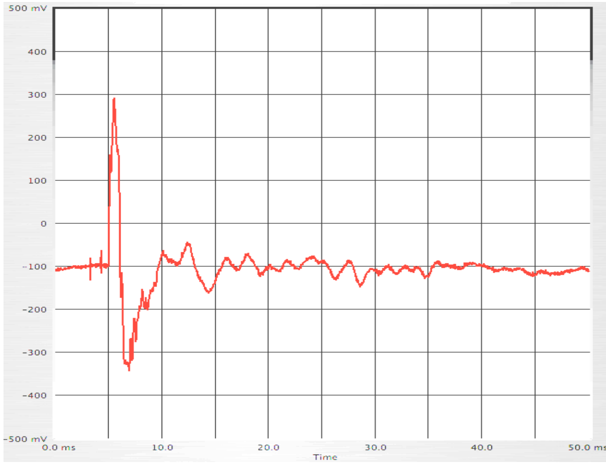

Figure 18. One-half meter transient response (with EQ, 5 ms/division)

Frequency response measurement was done with Fuzzmeasure [5] software. This uses a swept sine measurement method similar to that of the Time Delay Spectrometry method pioneered by Richard C. Heyser [6]. The reader is also invited to see 1) an exhaustive treatment on this subject by S. Müller and P. Massrani. [7], and 2) a paper from A. Farina on advances in sine sweeps [8]. These papers have many illustrations and numerous references.

Windowing was done post-measurement and a half-Bingham window was used. The half-Bingham has a rectangular leading edge, which is useful as there is no relevant data preceding the first impulse arrival. It also has a significant flat-topped region and a very steep trailing window edge. The latter features are useful for separating valid data from closely following room reflection data.

Figure 19 shows the target frequency response derived from close-mic measurement. Because we are in the pistonic frequency range of this speaker, this is a valid way of showing the shape of the speaker's high-pass related frequency response [9].

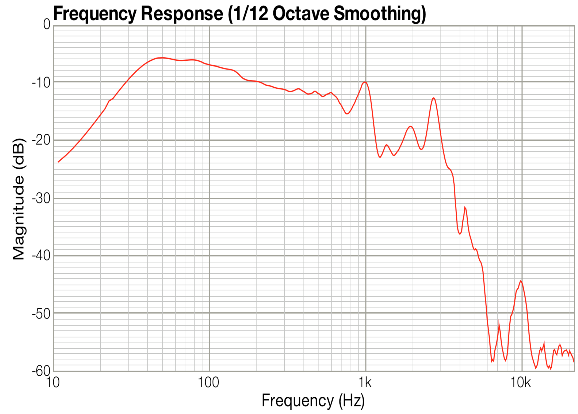

Figure 19. Close mic frequency response measurement (no equalization)

Figure 20 shows an ungated measurement with almost certain contamination of early reflection and room reverberation.

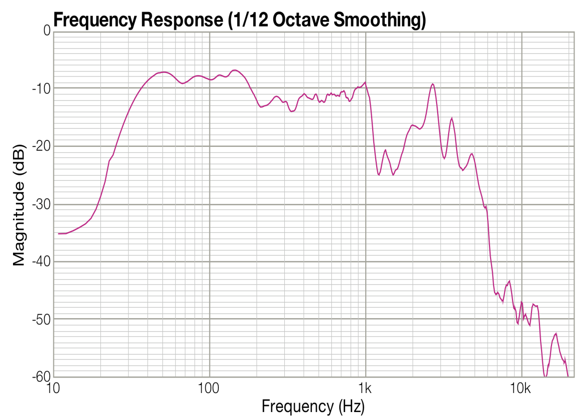

Figure 20. Ungated measurement at 0.5 meters

In figure 21 on the following page, we can see the result of gating without notch equalization tuning. Note that the bass region's transient ringing has convolved with the trailing window's edge. At a minimum, the bass measurement is thereby made invalid.

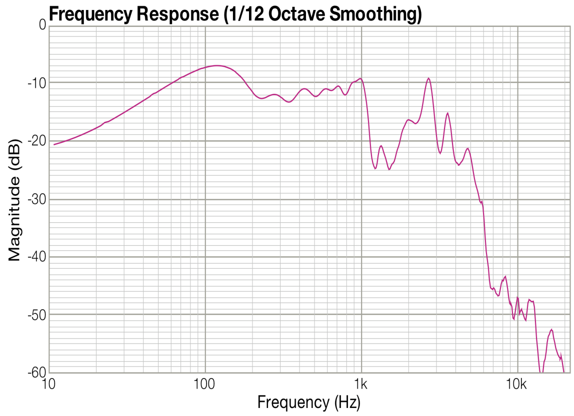

Figure 21. Half-meter measurement, gate of 10.45 ms without equalization

The gate timing of 10.45 ms is arbitrary: any such short gate times without impulse management will produce a similar unusable result.

With equalization, a 6.8 ms. gate time can be applied, and the result is shown in Figure 22. With all of the room disturbances visible in the impulse response, having the close mic impulse and low frequency response data at hand improves the process of gate timing selection.

One can see that the response data somewhat matches that of the close-mic data (shown here offset for comparison). The solid line is the speaker's response and the dotted line is the close-mic response.

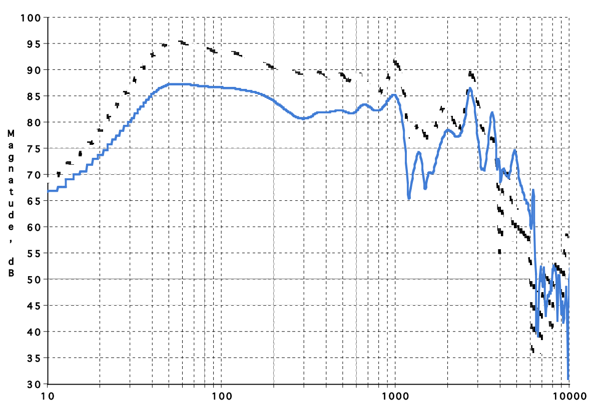

Figure 22. Gated measurement with equalization

The match is not exact. One possibility is that the author's room is just too small to allow reasonable gating. The other is that this is a 12-inch speaker with a relatively large dust cap. On this cap is deeply embossed lettering. For this speaker, these factors may make the close-mic assumption less valid at frequencies in the mid-bass range.

It was observed that as the gate timing was being adjusted downward, the two curves of Figure 22 were rather consistently different from 200 Hz and above.

Shortening all of the ringing in many speakers requires substantial, careful tuning. Very short windowing times require much effort. Recommended is that those doing speaker measurement find a location that would allow windowing times of 10 ms or more.

10. MEASURING SPEAKER ENCLOSURES AND OTHER SPEAKER SYSTEMS

The author would expect enclosed speakers to be measured much like one would do in an anechoic space, except that the unit under test would not need to be away from surrounding objects farther that windowing time would permit. One would expect the order of the compensating notches to match the orders of the measured system's needs.

The impulse property of a higher order system such as a tuned port configuration can likely be managed by the subject methodology. Since complimenting phase compensation is needed, one would expect a 4^{th} order system to require two 2^{nd} order parametric notches.

In multi-speaker systems such as woofer-midrange-tweeter setups, notch tuning would likely be needed for the high pass effects of midrange unit as well as those of the woofer. Tweeter hi pass effects could present themselves, but the author would expect ringing at higher frequencies to be shorter in duration than probable windowing times.

Note that if two notches of perhaps 20 dB were needed at a similar frequency, the user should make sure the electronics and digitizing noise floors are properly managed.

The author would likewise expect that planar and other speakers would be measured just as they would be in an anechoic environment, but again, suitably spaced away from nearby objects in a non-anechoic environment.

11. CONCLUSION

Both in simulation and in measurement of a typical loudspeaker, proper notch tuning has been shown to shorten acoustic path impulse response duration and allow useful gated measurement of loudspeakers.

Gate times can be reduced such that the contaminating reflections of moderately sized rooms can be managed without significant compromise to data.

12. ACKNOWLEDGEMENTS

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I would also like to thank John Mulchay, manager of HomeTheaterShack.com for his support and for his thinking toward implementing an automated method for implementing the subject methodology.

Finally acknowledged is Chris Lisco, creator of Fuzzmeasure software. Chris was most helpful in bringing the author quickly up to speed on the use of his measurement software.

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14. APPENDIX

The SPICE decks used for simulation testing emulate a second order filter (in this case, Q=0.68) using notch equalization. The circuit used is shown earlier in the paper.

Your SPICE program may not use the same syntax, so some modifications may be required.

The first deck implements the transient response simulation.

```
Quasi Sub Meas Transient Response
.TRAN 0.0001 0.1
VPulse1 1 0 PULSE(0 5 .00001 0.0001 0.0001 0.002
.089)
R1 1 2 10k
E1 3 0 2 3 10k
C1 3 4 400u
L1 4 0 60m
R2 4 0 8.3
R3 2 5 1057
C2 5 6 6.85u
L2 6 0 3.5
.print tran V(1) V(4)
.end
```

The second deck displays the frequency response of the simulation. Component values in this deck should match those of the deck above:

```
Quasi Sub Meas Frequency Response
.AC OCT 50 10HZ 10000HZ
Vac1 1 0 DC 0 AC 1
R1 1 2 10k
E1 3 0 2 3 10k
C1 3 4 400u
L1 4 0 60m
R2 4 0 8.3
R3 2 5 1057
C2 5 6 6.85u
L2 6 0 3.5
.print AC V(1) V(3) V(4)
.end
```

The reader is invited to adjust C2 and R3 and watch the characteristics of the ringing go from an underdamped response, through a “flat” transient tail, to an overdamped response.

To see the “anechoic” response of the “speaker”, adjust R3 to a very high value. To adjust speaker Q, adjust R2. To watch the effect of notch depth on the transient response, adjust R1.

Upon request, the author will be happy to supply SPICE decks for other simulations used in this paper.