Back

More on the **Sound Strobe**

With the aid of a speaker analyzer and woofer mod, this author gets his speaker system in sync.

By Ed Simon

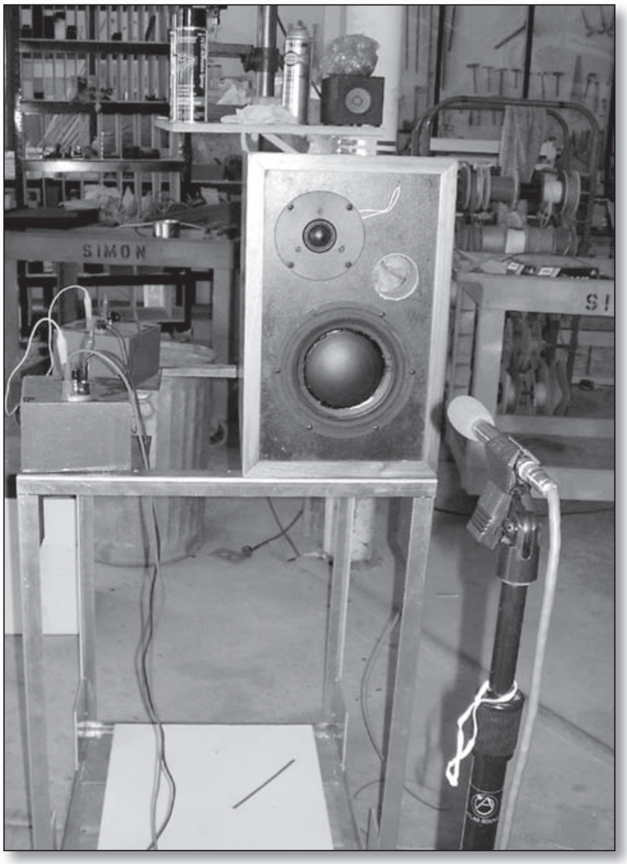

PHOTO 1: Test setup.

One of the rules of DIY audio is that the best-sounding loudspeaker is the one you build yourself. That rule holds true until you start making measurements.

LATE ARRIVALS

A while back I tried building a two-way loudspeaker by ear using just the Sound Strobe. One of the things I did not mention is that when I voiced the crossover by ear, there was a 2dB dip at the crossover versus where I would have set it using just the frequency response measurements. With a bit of measurement and some thought, it became clear that the devious Sound Strobe impulse from the woofer came after the impulse from the tweeter.

When you look at a frequency response curve, you can often see a discontinuity in the graph at the crossover point. Some curves hide this by using a smoothing function. It is possible in theory to get a perfectly flat frequency response curve for a single microphone position with just about any combination of well-tweaked drivers. That would work, if you have only one functioning ear and want to hold your head in one exact position.

The frequency response in theory is the reciprocal of the time response (1/f = T). When listening to the pulses, my ear sensed that a slight ducking of the crossover region gave a cleaner sound. The final arbiter of this would, of course, be extended music listening tests, the results of which would be open to debate because listeners have different preferences. The solution is, of course, to make sure the impulse from the tweeter and the woofer arrive at the same time.

In order to achieve that solution, some loudspeaker designers slope the front and move the tweeter back to delay its arrival time. Others put more elements in the tweeters' crossover path to add delay.

I decided to try a different technique. I thought I would see whether I could get the sound out of the woofer sooner.

TEST SETUP

Although I have a reasonable assortment of test equipment, it seemed the simplest way to test this theory was with a microphone, an oscilloscope, and that infernal Sound Strobe driving both the scope and a small amplifier.

For the amplifier I used my single-ended solid-state amplifier, which lives on my test bench. The microphone was an ACO Pacific 7012 capsule with a 4012 preamp powered by a Bruel and Kjaer 2801 modified power supply. Of course, just about any microphone would do for this application. I just keep that setup on my bench. The oscilloscope advertises itself on every picture! Almost any oscilloscope or computer emulation would do for this test.

The setup was quite simple: I placed the loudspeaker on a stand with the woofer 36″ off the ground and the tweeter 42″ up. I placed the microphone 39″ high and 19″ away. The Sound Strobe fed channel one of my oscilloscope and the microphone channel two.

MEASUREMENTS

Figure 1 shows my results. The top trace shows the pulse provided by the test signal. This pulse has a sharp rise time and a slower exponential decay. Some internal oscilloscope noise shows up on this trace.

Channel two shows there is a delay of 1.525ms before the microphone gets the signal. At standard temperature and pressure (STP) the delay should be about 73.6μs per inch. So there is about 125μs of other delay. This comes from the crossover, the transducers themselves, and smaller amounts from everything else in the chain. The first pulse is from the tweeter, then there is a delay of 338μs until the woofer reaches maximum output.

The tweeter pulse is a classic lazy S curve. (For those of you who do not read brands, lazy means the letter is on its side!) The tweeter has some mass, although very small. It cannot move out instantly as the input pulse would require. To do that would require an infinite force (at just a few watts at most, the amplifier falls a bit short). The top of the pulse rounds a bit as the tweeter piston slows down when the input stops. The tweeter does not cover the low frequencies, so its work is done. Because the piston moved out, it now must move back. That is why you see a negative-going spike.

If the amplifier provided better damping to the tweeter, the negative-going spike would be smaller. That is because as the tweeter moves back in it generates a voltage—a short circuit across the voltage would increase the resistance to this motion. To some extent this could be helped by a Zobel network, a low value shunt resistor, or an amplifier with higher damping factor at high frequency. The last sounds the best, but because loudspeakers are voiced with normal amplifiers, such a design would sound dull.

In **Fig. 1** the woofer begins to rise 70μs after the tweeter. It takes an extra 268μs for the cone to fully move. This corresponds to an upper frequency response of about 1800Hz, which is not too bad for a woofer. The woofer has an interesting flattened top. I could guess why, but there are instruments that could prove my guess wrong, and because I never make misteaks, I won't guess.

That the output curve does not match the input signal may come as a surprise to some, but these are real-world results. The woofer, as mentioned, cannot rise instantly, so it lacks the sharp front edge. The time delay because the tweeter is mounted closer to the microphone than the woofer means that the spike from the tweeter is not riding on top of the woofer so the combined results only crudely approximate the input signal. That is why time response, which should be the reciprocal of frequency response, is an approximation that does not hold up well for real (finite bandwidth) devices.

SUB MOD

Now it is time to play. The motion in the

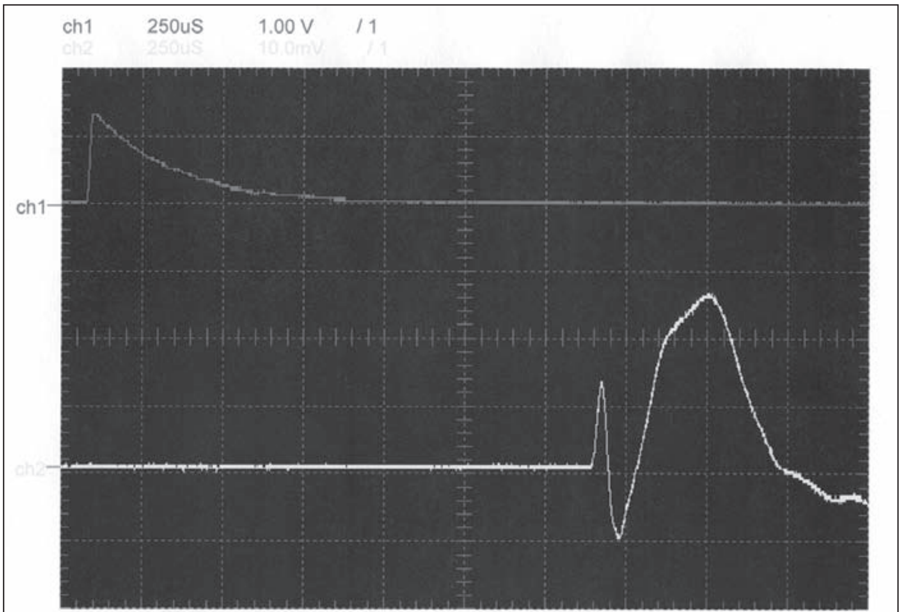

FIGURE 1: Large cap.

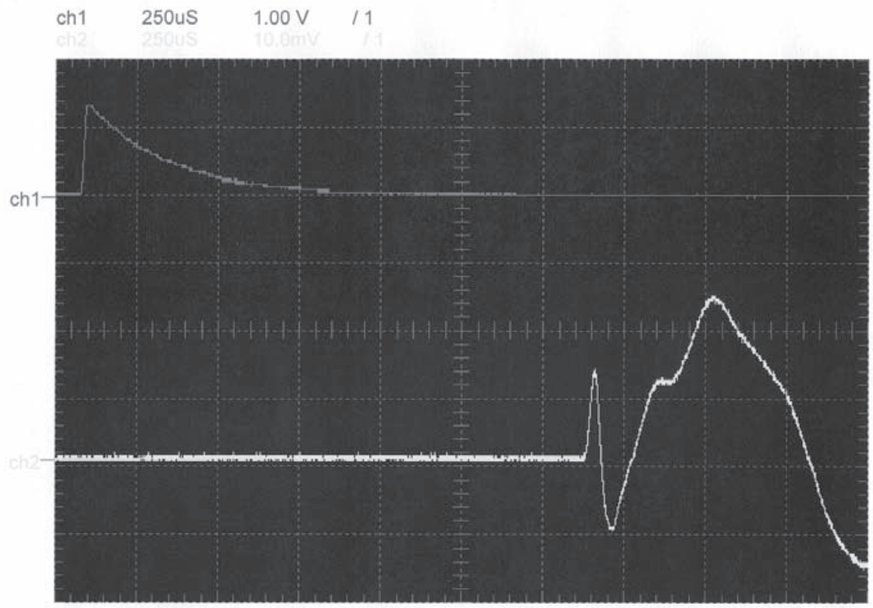

FIGURE 2: Small cap.

woofer is started by applying a current to the voice coil. This moves the coil and sets the attached cone in motion. The cone then moves the air. The velocity of propagation in the cone is faster than in the air. I decided to move the area of the cone forward. I chose to couple to the existing cone about three-quarters of the way to the front edge of the cone.

A few years back my friend Bill bet the combination center of two loudspeakers was based at the voice coil, as many believe. I picked the closest edge of the cone based on computer simulations I had done before on my then new PDP8. We measured the -6dB beam width of a dual 15″ loudspeaker. This point would be 45° off the center of combination. On 15″ loudspeakers this would be far enough apart to give a good indication of the center.

It was not at the voice coil but actually only about three-quarters of the way out on the cone. We suspected this may be due to cone break-up.

I later concluded that it may not have been due to cone breakup. That is because in my computer model the time for the motion of the cone to reach the edge was zero. In the real world it would take some time.

My estimate of the cone density for this loudspeaker based on the dust cap I removed was .42 ounces per cubic inch. That is very, very light, but denser than air. My seat-of-the-pants estimate is that this would give me a velocity of propagation only four or five times that of the air.

If I break the cone up into three parts—the dust cap, the cone middle, and the edge—I can simplify my model of how the pressure is generated. The air by the dust cap is first pushed forward, then the air coupled to the middle of the cone, and finally the edge air. However, the air from the edge reaches the ear first because it is the closest, then the air moved by the middle, and finally the air put into motion by the dust cap. As a result, I do not get a clean pulse but a stretched pulse!

To get the loudspeaker to couple three-quarters of the cone distance closer to the front edge was not too hard. I did this by cutting out the original dust cap and replacing it with a larger one. I used a 5″ dust cap that I cut down to about 3¾″.

The trick in cutting a dust cap is to place it on a flat surface and rotate it against a pencil to get a uniform cut line. Place the pencil on a spacer to get the diameter you want.

The increase in cone weight from removing the smaller dust cap and adding a larger one was 2.58 grams. This would change the response of the woofer and require a new frequency response tuning for the loudspeaker, but I wanted to see what it would do to the impulse response.

For Digital Recorders
Introducing
For USB A/D Systems
PHANTOM POWER
7052PH
Measurement Mic System
7052H Type 1.5TM
Titanium Diaphragm
3Hz to >20 kHz
<20 dBA > 140 dBSPL
MK224 (<14 dBA to >134 dBSPL) Optional
4048 Preamp
Superior
IEC 1094 Type 1
Long-term Stability
Temperature and Humidity
Performance
Now in Stock
Phantom
to IEPE/ICP
Adaptor
Supplies 3-4 mA
Power
Accelerometers
Microphones
ICP1248
ACOustics Begins with ACO
TM
MATTtm
Family
MATT TM
Mic
Attenuator
Handle Higher Sound Pressure Levels
ACO Pacific, Inc., 2604 Read Ave., Belmont, CA 94002
Tel:(650) 595-8588 FAX: (650) 591-2891 E-Mail: sales@acopacific.com
Web Site: www.acopacific.com

Figure 2 shows the new response. The test signal remains the same!

The easy way to notice that the woofer begins to move forward in time a bit sooner is there is less of a negative spike from the tweeter. This decrease in the woofer's initial response lag is about 25µs.

This corresponds to moving the woofer 1/3″ closer to the microphone, which is just about one-half of where the new dust cap sits. (Leading me to suspect the velocity of propagation in the cone is twice that of air; however, the motion in the dust cap is mostly in the transverse mode!)

I tried using a larger closer dust cap, but the increase in mass and cone breakup prevented that from working well.

Looking at **Fig. 2** you should also notice that the woofer now has a sharper spike and a slower rise time. In addition, the pulse width now is longer. These results are probably due to the increase in mass resulting in a lower frequency response. Of course, the modified woofer now has a much nicer lazy S curve.

I did not get as much time delay as I would like. I could try a longer voice coil form for the tweeter. A deeper tweeter or an all-pass filter would also be solutions. Also, it is important to note that even with the time set optimally, there still would not be as sharp a rise time as the input signal. The woofer does not talk to the tweeter to split up the workload. Each does its own thing. The combination will not be perfect. Better is the goal.

HOW IT SOUNDS

Of course, to be fair for listening tests, I now must re-voice the crossover to account for the change in the woofer. But my preliminary result was that a noticeable sibilance in the loudspeaker on some female vocalists was now reduced.

I expect that sibilance is caused by upper midrange energy hanging around too long. In this case the tweeter is clearly before the woofer, although the cancellation of the rebound tail may be what I am hearing.

Most folks would argue that I cannot hear the 25µs time change, that it must be my imagination or something else. But then they probably have not tried this.

The important question is, "Does it sound better?" The stock answer is, of course, "If you try this it will sound better to you!" If you wish to try this yourself, be safe, use removable glue.

The result of all this is that when I build a loudspeaker system I will first need to see the time delay introduced by each proposed component and then pick and choose my drivers. This is an interesting extra step because this data is not given on datasheets. *aX*

REFERENCES

1. "A Solid-State Single-Ended Power Amp," by Ed Simon, Apr. '06 *aX*, p. 24.
2. "The Sound Strobe," by Dennis Colin," March '06 *aX*, p. 17.
3. "A Combination Horn You Can Build," by Ed Simon, Jan. '07 *aX*, p. 10.
4. "The Sound Strobe," by Ed Simon, Apr. '07 *aX*, p. 6.


Ohm my.
Eminence Professional Series Speakers
EMINENCE®
THE ART AND SCIENCE OF SOUND
WWW.EMINENCE.COM