PRODUCT REVIEW

The Sound Strobe

Discover what this sound system analyzer can do. . . or at least some of what it can do.

By Ed Simon

I looked for Dennis Colin's picture at the Post Office, but I couldn't find it.

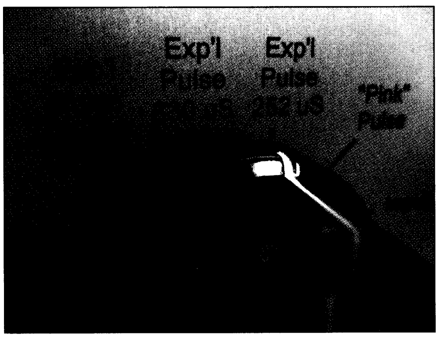
He had the audacity to come up with the Sound Strobe (*aX*, March '06, p. 17). When I first read the article I thought to myself, Colin (possibly with some help) came up with a new twist. The idea of using a simple 555 timer to produce a voltage spike that shoots up and down very quickly with a second or so between pulses certainly is not new.

CLICKING SOUNDS

This is known as the unit impulse. A perfect unit impulse response is a jump from 0V to some fixed level instantly and back again. There is an infinite rise time, zero time at the top, and a perfect fall time. So, of course, it only exists in theory. A practical impulse has a rise and fall time at least ten times faster than the system you are testing. The time at the full voltage should last just long enough to allow you to take good measurements.

How a system responds to this impulse can tell you just about all you need to know. If the frequency response does not go high enough, you get a rounded top instead of a sharp square edge. If the response does not go low enough, the top slopes. If the unit has too much energy storage, the pulse does not fall fast enough. Getting the pulse size right is a bit of an art. . . until now.

The unit impulse generator is the old method we pros used to time-adjust large sound systems with audio delayed loudspeakers. If you heard two clicks you knew the delay was off. With a bit of skill you could listen until there was one click that was not "fat," and when it became neutral you knew you had the delay dialed in.



Of course, when Smaart software came out, almost everyone started using it instead. It was reasonably cheap (today you can get started for $695 plus a computer, a mike preamp—$100-$600, and a microphone—$400++) and showed you not just the unit impulse response but also let you take a reading to determine how far out you were. Like all good measurement systems, if you use them and listen, you train your ear and can often do without the gear after a while.

When I was putting a system into the L.A. Forum, the old road dog Ted Leamy from JBL stopped by. There was an issue of how much delay to put in the side fills; he offered to set up a Smaart rig to measure it. I walked into the zone and listened, used my experience to dial in a number, and we both agreed that was close enough. Because there was only a zone of coverage where the timing would vary anyway, there being no absolute answer, good enough was the right answer. You could save a bit of money if you used the Sound Strobe to set up a delay, but you would not have all the bells and whistles of a full-blown rig.

SPEAKER DEMO

When I first got a Sound Strobe, I plugged it into the office loudspeakers (*aX* 1/07, "A Coax Horn") using a single-ended class A amplifier. After a nice clean click, I tried a few of the settings and heard nothing new. . . until I tried the "frequency" rate knob.

In my article on this loudspeaker, you may remember I had a problem measuring the low-frequency response. Although my measurement microphone is rated –1dB at 9Hz, the noise floor of my testing space rises as frequency drops so that I was unable to get a reasonable low-frequency plot. The Sound Strobe pulse began to behave differently below a 17Hz repetition rate! This was enough evidence for me to make a few simple listening checks to tweak the speaker's position in the corner, something I could not do with my other gear.

The next day I showed the unit to a client who came in to pick up a repaired speaker, and to one of my guys. We hooked up the speaker, waited until it clicked, and demonstrated that the speaker worked. We got a nice clean click out of it until I twisted the waveform knob. At the "Exponential Pulse 830μS" setting, we all heard two clicks coming out of the loudspeaker, indicating the crossover point was well out of time adjustment. This loudspeaker is a widely copied product, but even with the knockoffs it still sells. Last I knew, about $15 million a year worth of product in the pro market.

I mentioned the double click to Bill (he can hear pretty well), who was one of the guys responsible for the speaker design. His response was, "Where do I get a Sound Strobe?" He was as surprised as I was, because the normal measurements and user responses did not show this problem.

ON THE ROAD

One of the common misconceptions in audio is that the voice coil is the center of the time origin for loudspeaker driver components. Actually, it is not. Sound travels about ten times faster through a cone than through the air, which is why for time origins determinations where the forward edge of the piston couples to the air requires a rough guess. There is usually considerable argument when you try to explain this to a "True Believer," unless you can actually demonstrate this. Of course, to really do it right required complex test equipment. . . until now.

This is where that Colin fellow becomes downright dangerous. There are six settings on the "Waveform" knob, each of which does something useful. The timings he picked for the exponential decay are pretty much dead on to where they are useful. Just twiddle the knob and something

jumps out at you! Lucky guess, I suppose.

I took the tiny box with me on a few system checkouts and tuneups. After a few hours of adjusting a small church (600 seats) system, I put a few impulses through the system and it confirmed what I fixed, and what was wrong with the room was still wrong with the room. No new information, just a last-minute check to be sure I had not overlooked anything that would possibly embarrass myself.

At a bigger joint (18,000 seats) it told me the good seats were good and the bad seats were acceptable. I could hear the pulses "fatten" where the sound was not as good, but the STI meter—although it dropped—was still in the acceptable range.

SETTING THE TRAP

Although it was a great double-check box, most *aX* readers probably won't use it to check out a stadium or even an auditorium. Because I had a nice woofer left-over, I thought I would build a pretty standard two-way bookshelf speaker to give Colin's box one last chance to show that it could not do everything.

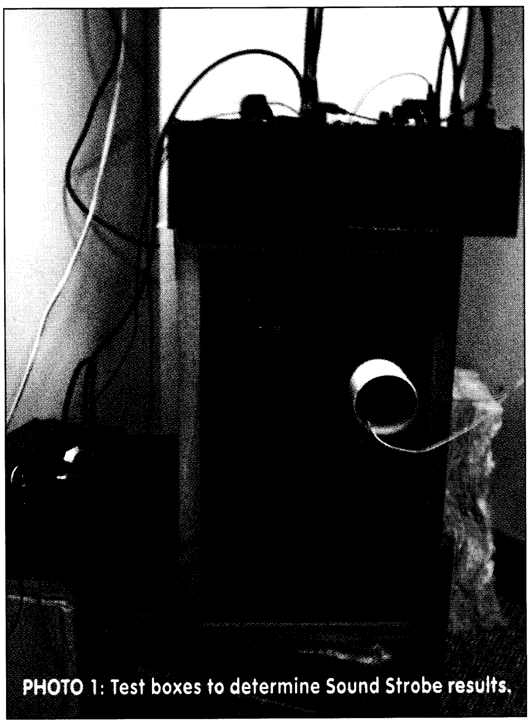


PHOTO 1: Test boxes to determine Sound Strobe results.

While talking to the woofer supplier about another project, I asked for a suitable tweeter and I bought a pair based on his suggestion. This is one of the pro sound secrets: Ask the guy selling the audio gear what he recommends. If it works out, he knows what he is talking about, so you can trust him again. If it doesn't, find another place to shop.

I put the speaker parameters into one of my box programs to determine box size. I then visited a local woodworker's store and picked up some walnut sticks on sale for $5 a board foot. I needed to cut the boards into smaller pieces consisting of a side and top length, plus allowance for planing.

After planing, edging, and ripping, I built a small sled to crosscut the pieces. Even though I had just planed the wood, I still did not cut a 45° joint and flip the board over to save wood. After I cut it to final length, the wood still cupped enough that the seam would not close nicely, so I kept all the outsides out, and the cup in. I used an old mailing tube for the port, one that fit snugly into the hole left by my 2″ hole saw (**Photo 1**).

I put together an Excel spreadsheet to calculate the frequency response of a crossover and estimated what component values I would need based solely on the datasheets for the loudspeaker components. I then built some capacitor, inductor, and resistor boxes. The capacitor boxes use 1, 2, 4, and 7μF capacitors with a two-pole 12-position switch to give a value between 1 to 9μF. I used a meter to assemble reasonably accurate values out of my junkbox assortment of film capacitors.

I wound my own tapped inductors with 18-gauge wire for the inductor box. I made a form from a dowel 1″ by 1½″. I found that the first tap at 99 turns gave me 250μH. The next 45 turns brought the total inductance up to 500μH. Then 39 turns to 750μH and 33 more to 1mH. In case you want to go further, try 31, 29, 24, 22, 18, 16, and 16 turns to get you to a nicely stepped total of 2.75mH.

For the resistor box I used just ten 1Ω 2W film resistors and, of course, a 12-position rotary switch—single pole. I now had all the makings of a do-it-yourself adjustable crossover kit. I used to have a single box for this, but after I lent it to Bob, I never saw it again.

I hooked up the test boxes to give me an LC filter into the woofer and a CLCRR into the tweeter (**Fig. 1**). I turned on the dastardly black box and adjusted the frequency to a 2Hz pulse rate and listened. At first I thought I had made a wiring mistake because all I heard was bass. After disconnecting the woofer, I traced out the tweeter wiring and found I had set the "Waveform" to low-frequency pulse.

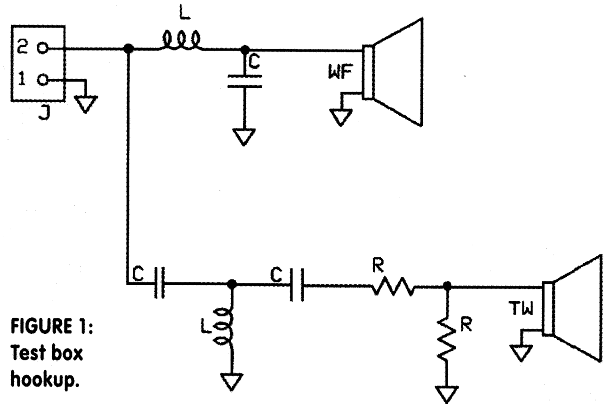
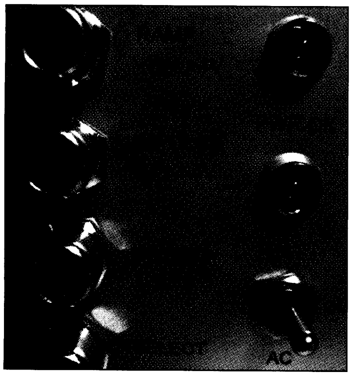


FIGURE 1: Test box hookup.

I set it to the 252µS exponential pulse and then adjusted the tweeters' crossover components for the cleanest click I could get. I then hooked the woofer back up and changed to the 830µS weighting on the pulse. After setting the capacitor in the woofer circuit to the middle of its range, I closed my eyes and adjusted the woofer inductor for the cleanest click. Then I tweaked the capacitor.



THE FINAL TEST

I then set up my field test kit—an Earthworks M30 microphone and an M-Audio USB preamp to my Dell laptop running Smaart. For the signal source I used a Sony SCD595 SACD player, my custom built Holco resistor attenuator, and my Class A amp. The curve was within ±3dB from 150Hz to 20kHz. The limitations were the driver, box, and location!

With my field kit I could calculate the adjustments to the crossover to get even better response and then try them out. But the Sound Strobe demonstrated that even technically blind it would allow me to get the best possible settings with just one box! Playing with the crossover topology and values, the Sound Strobe would take just a bit longer.

The question that remains is, can you use music to make these same adjustments? One of the concepts that is currently exploited in audio measurement systems is to use processing power to compare a reasonably simple signal such as pink noise and to the loudspeakers' output and compute what the difference is.

The other school of thought is to make the test signal complex and the measurement gear simple. Dennis Colin has not just done the second method, but he has nailed the variations in waveform to maximize the results. You could use music, but it would take much longer.

The Sound Strobe can do even more! I am not going to reveal all its capabilities. It takes two days for me or my competitors to tune up a big sound system, and when we are done we really have tuned only half of the things that should be adjusted. This unit will let me tune the other half and I am not about to disclose in print what my competition does not know. You will have to find out for yourself! *aX*

The Sound Strobe is available in kit or assembled units from **Old Colony Sound Lab**, PO Box 876, Peterborough, NH 03458, 888-924-9465, e-mail: custserv@audioXpress.com.