

# Practical Sound System Time Alignment

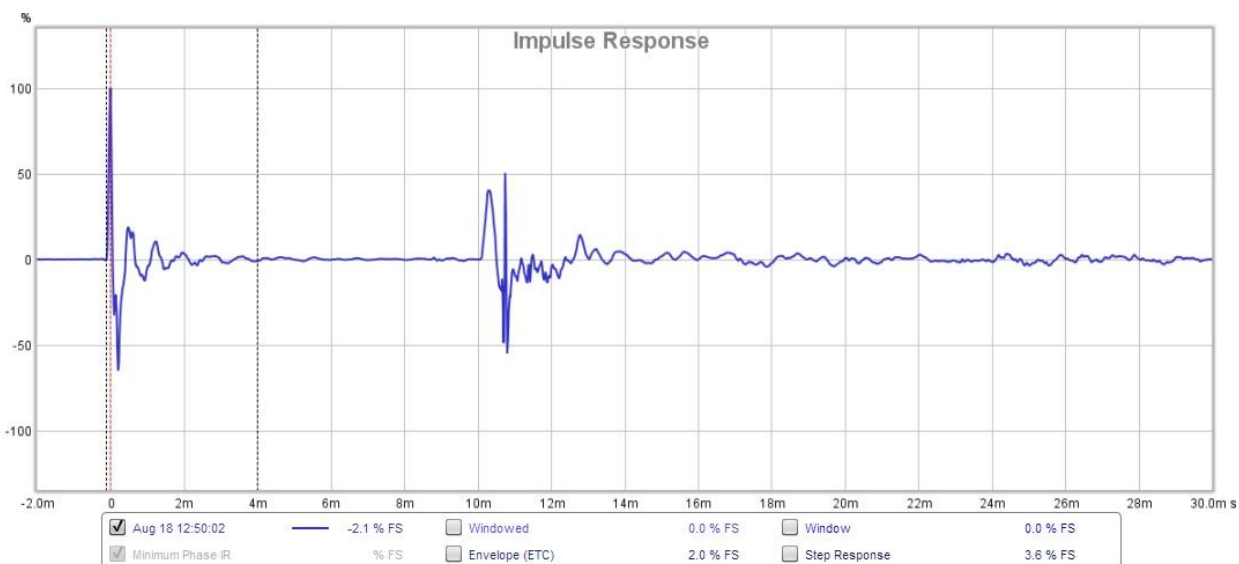
Or

“Parking your DeLorean in the right spot...kinda.”

By

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## Introduction

Sound system time and phase alignment is hard, you have to have a pretty wide knowledge of signal processing and linear systems, and the correct test equipment. We're hoping this document makes the whole process a tad bit easier. Without getting too detailed into loudspeaker measurement systems....there's ways to do it using systems that have an absolute time reference (LAUD, CLIO, PRAXIS, REW (depending on how you set it up), SoundEasy, SMAART and others...and ways to do it without an absolute time reference. REW (depending on how you set it up), Omnimic and others don't have a time reference.

One thing you *\*need\** to have is the ability to measure the impulse response of the speaker. A spectrum analyzer or RTA will leave you blind. Seriously, they are pretty useless for this, it's like saying you can drive a car north with a blind fold on. You might get lucky, but most of the time you'll think you're doing okay when really you're just making a mess out of things.

We're going to show you one of the least expensive ways to do this, using REW and a uMik-1 from MiniDSP and an extra speaker driver.

(There's another way with an analog microphone and a loopback reference...but that's for another paper.)

## REW Setup.

A step by step guide on where to click and what to do is about 50 pictures and screen caps too many and already covered in the manual.

For the absolute best results you'll want to get your REW working so that:

1. The soundcard is calibrated.
2. The microphone is calibrated.

Neither are mandatory for time alignment, but the next step (not covered here) is the final adjustments for phase and frequency response and well...they are mandatory for that.

Basically you want to get to the point where you are doing a frequency response sweep. But instead of looking at the frequency response you'll want to look at the time domain which is under the impulse response tab.

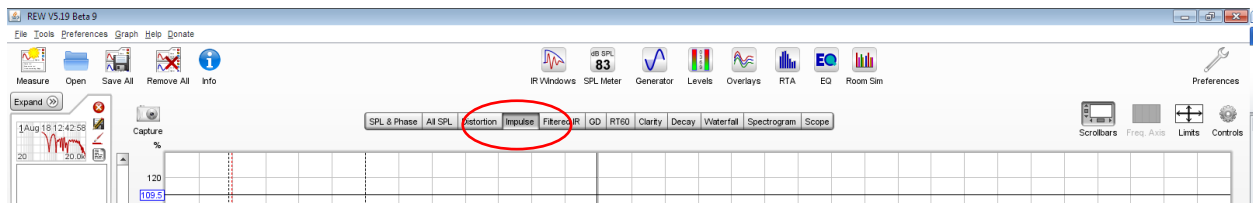


Figure 1 Impulse Tab in REW

Next, I (Scott) prefer looking at the time domain response using % full scale instead of dB scale which is the default in REW. In the top left corner of the impulse graph you can set it as %FS or dB.

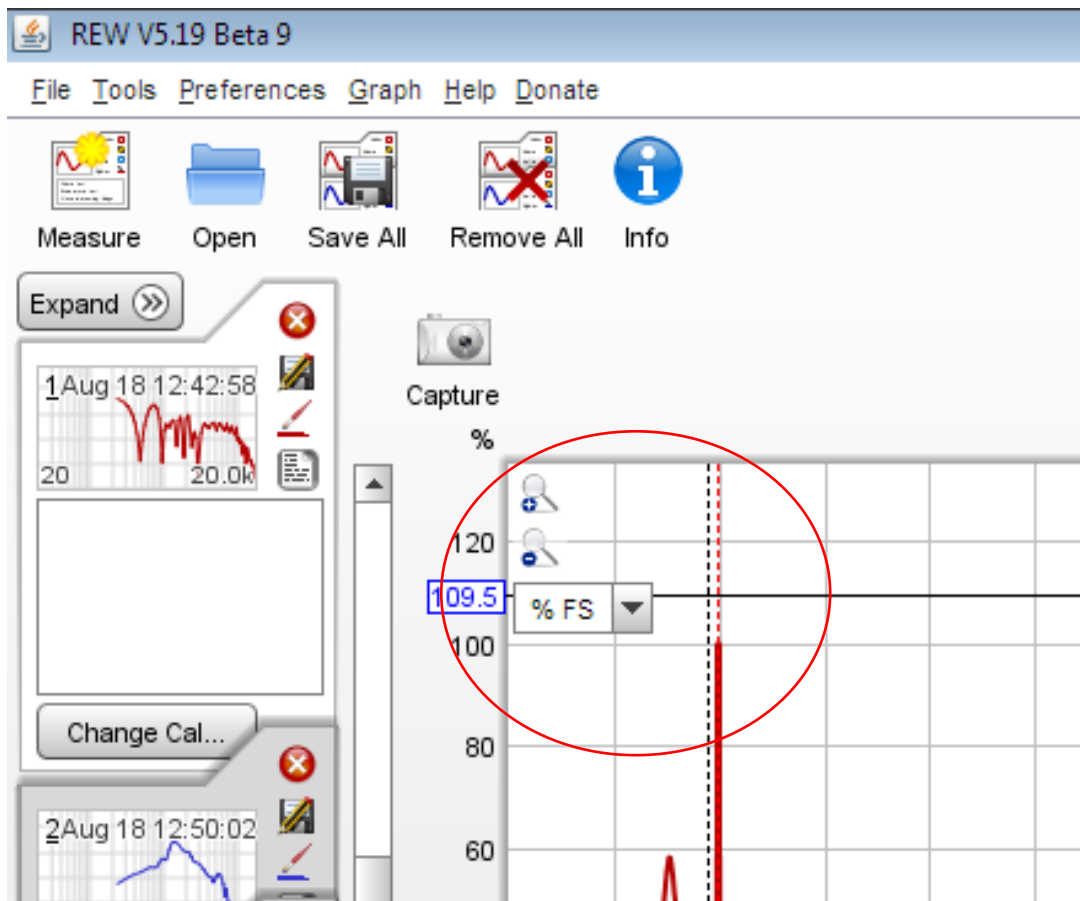


Figure 2 Select Vertical Scale

Once that is done set the time scale using the limits button to around -2 to 30mS or so.

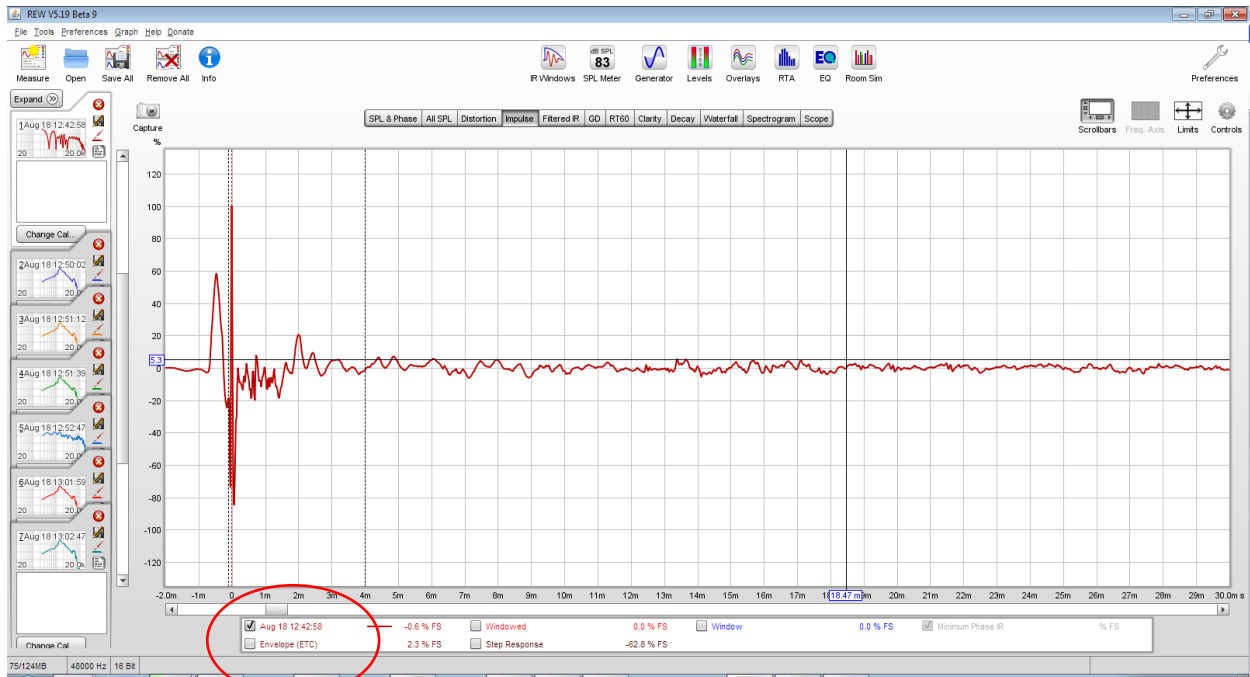


Figure 3 Select impulse response

Keep in mind that sound travels about 1.1 feet in a millisecond, so you'll want to zoom in a bit. The default scale for the impulse response is about 500 milliseconds which is useful for room acoustics (Room EQ Wizard) but not real useful for speaker alignment. Next you'll want to make sure that you're looking at the impulse response, not the window or step response. That's the top left hand check box, circled above.

Eventually you might find the ETC useful, and if you want to do frequency response measurements I know you'll find windowing useful, but let's stick to one thing for now.

## Measurements

Now the impulse response in Figure 3 represents a double 15 box and a two way speaker with a 6" mid and custom horn driver. But...it's kind of hard to tell what's going on. If you turn one of them off, since REW is operating in an open ended mode where it doesn't know exactly when it sent the signal to the amplifier, and it doesn't know when the signal came out of the amp with any possible processing latency...and it doesn't know how long it took it to get to the microphone, or how long it took to get out of the microphone (processing latency) it sets time 0 at the biggest portion of the pulse. That's not particularly helpful.

However...you can give yourself another reference. Easily. You just need a small wide range

Practical Sound System Alignment

-An 88MPH ticket is how much?

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reference driver. For this test I used a 5" woofer that's got reasonable response to about 5kHz.

I put that driver in parallel with everything else...and I put it close to the microphone. With a bit of care in placement and scaling, that becomes a fixed time reference. Too close or too loud and it might overload the microphone. Too soft or too far away, it might not be the largest signal.

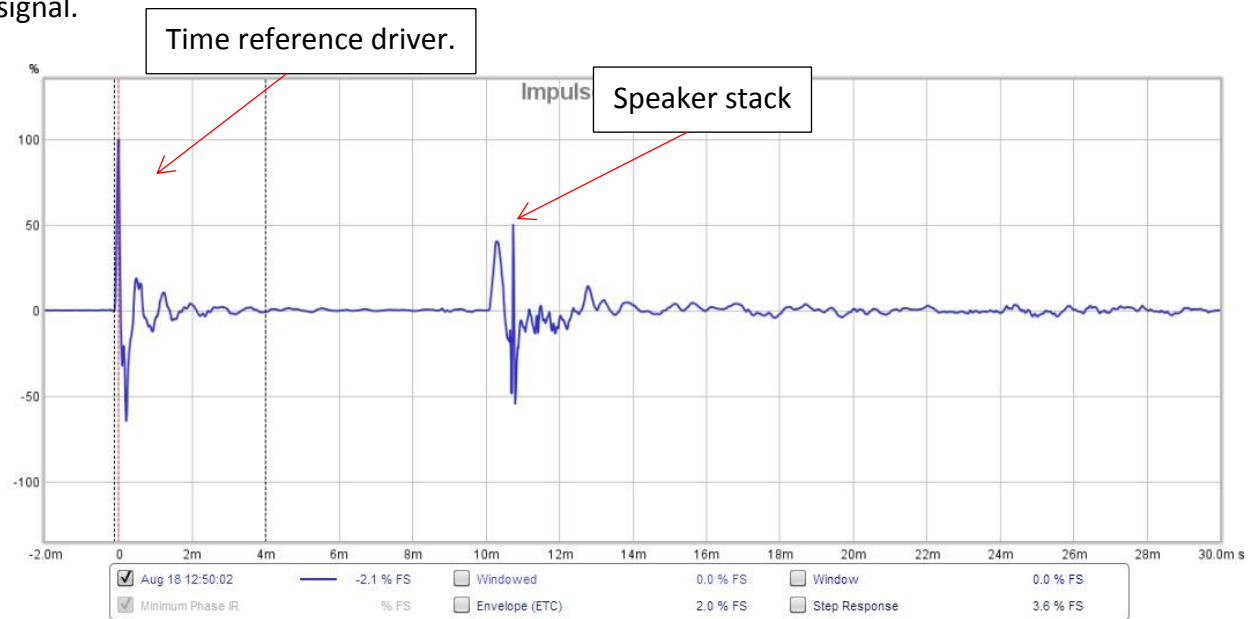
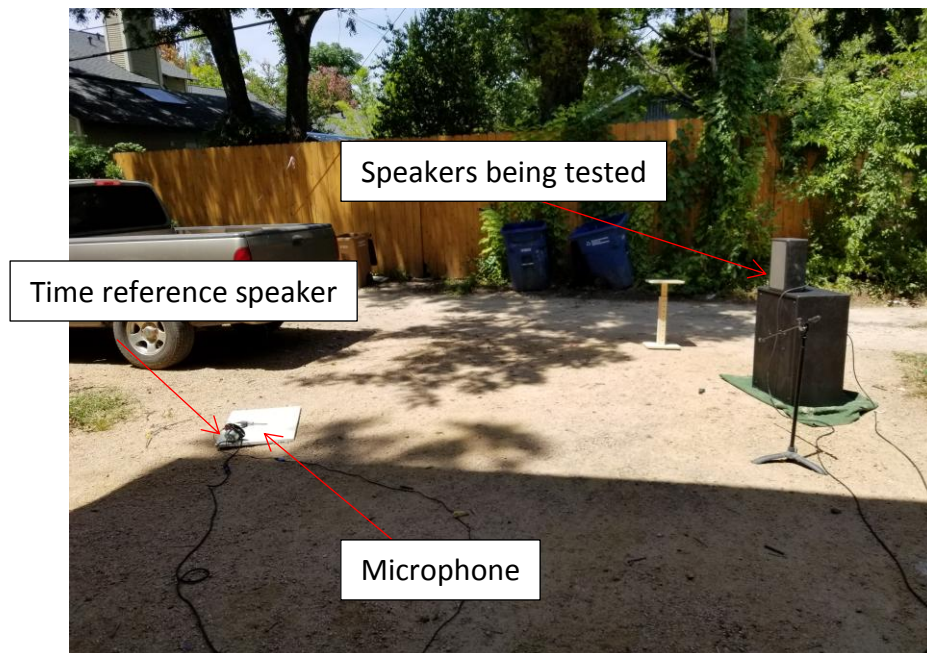


Figure 4 Impulse response with new driver....close to microphone

Figure 4 shows the result of measuring with this extra driver added close to the microphone.



*Figure 5 Basic Measurement Setup*

Figure 5 shows the basic measurement setup. The important part to remember is don't move the speaker close to the microphone, or the microphone. That way your time  $T = 0$  doesn't change.

Next we ran two sweeps, one with just the woofers and one with just the mid/high top. There's going to be a big blank space so we can get those two...along with the original, all on the same page.

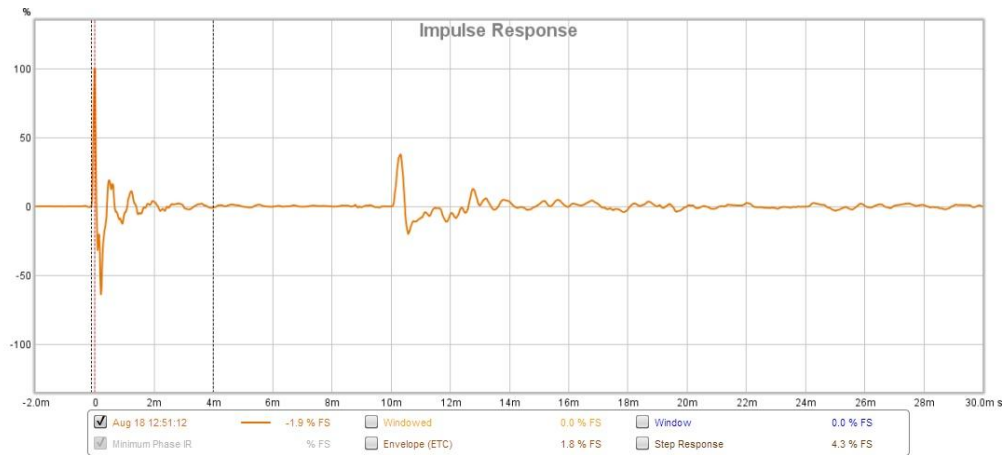


Figure 6 Woofer Only

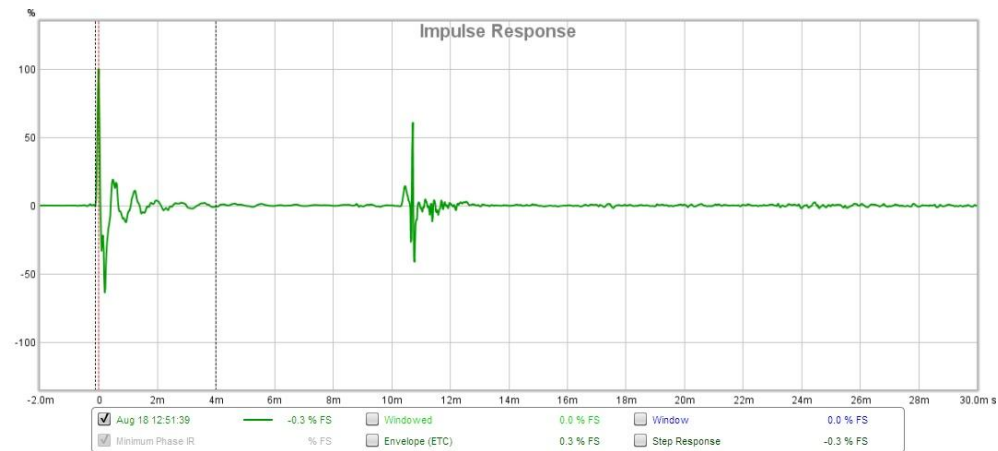


Figure 7 Mid/High Only

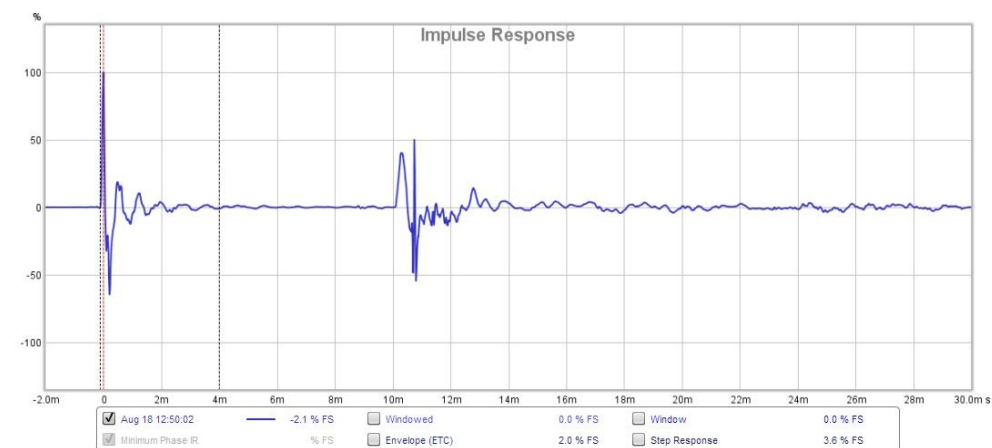


Figure 8 Both

For Figure 6 through Figure 8 notice how, if you study them you can see the individual shapes of the woofer and high speaker in the combination plot. The sharp spike is in the same place, and



the slower spike is in the same place. When both speakers are on...both spikes overlap each other a bit.

Now, this brings into question what is time aligned, or more importantly how do you get multiple speakers to be both time aligned and work well together in phase. That's a bit more than we're willing to bite off for this 8 (and counting) page paper. We will refer you to the publications sections of Charlie Hughes' website at Excelsior Audio for further reading, until we can get to it in the future. The method we're showing here will help you align the energy peaks of each speaker, but won't necessarily guarantee good phase tracking through a crossover region. However...it can help you get a lot closer than you would have been without doing this at all.

First let's move the top to someplace pretty ridiculous and measure.



*Figure 9 Top moved forward roughly 4 feet.*

Figure 9 shows the measurement setup for the top moved forward about four feet.



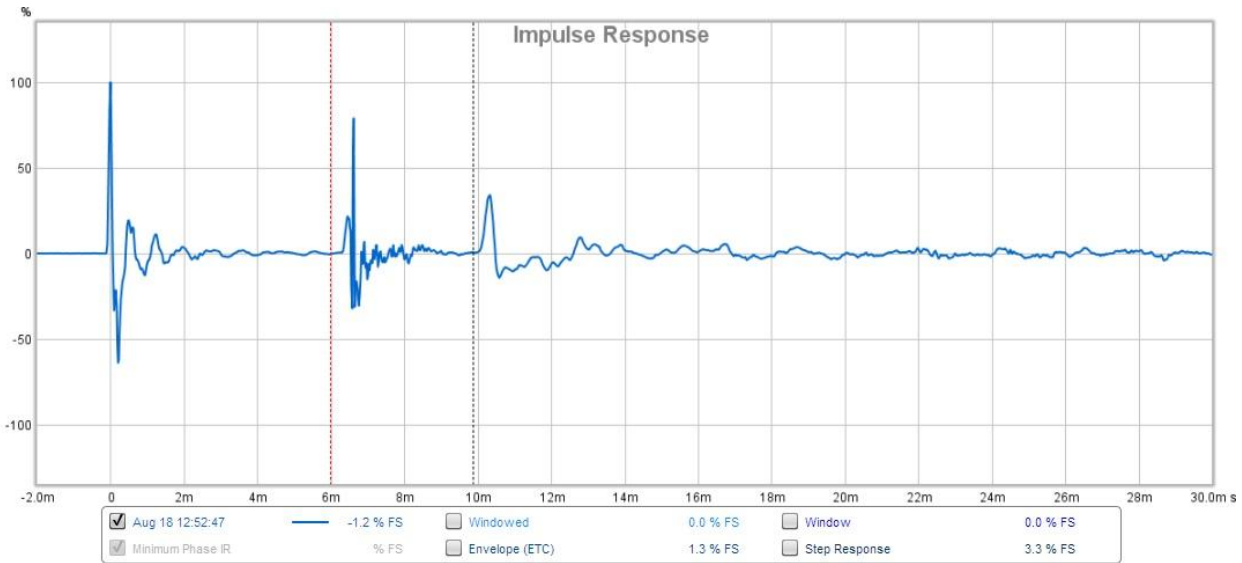


Figure 10 Impulse response, top moved ~4 feet.

Notice how the impulse response of the mid/high speaker is now earlier in time, in fact it's about 4mS forward. It's also gotten a bit louder since it is closer to the microphone than before.

Next Patrick and I moved the mid/high speaker to a spot to better align the impulses. Typically this would be done using the processor settings, but we didn't have time to move a DSP unit over to this test setup.

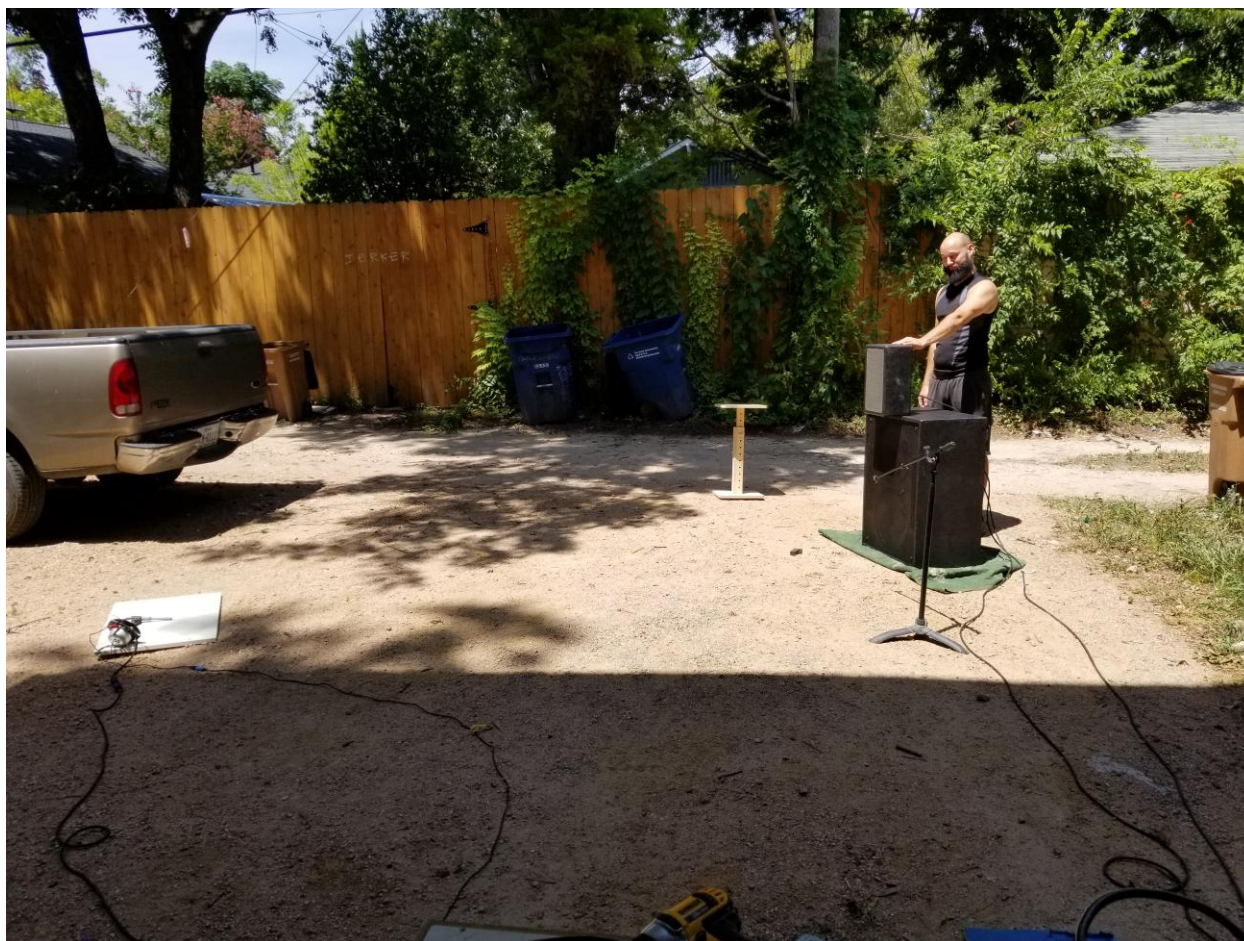


Figure 11 Better Time Alignment

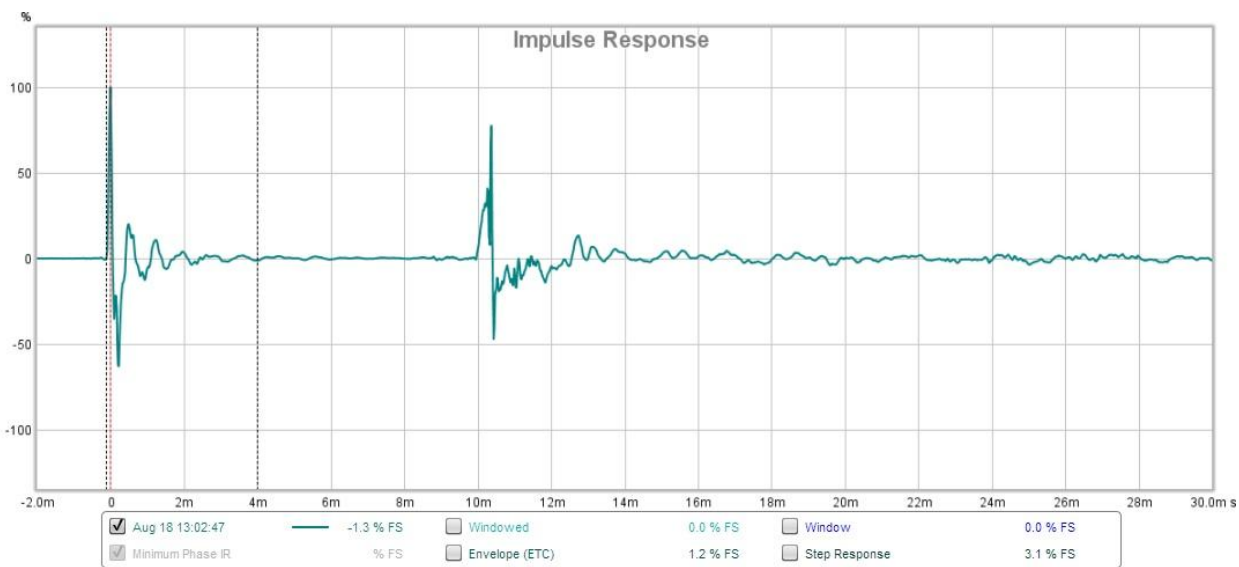


Figure 12 Time Domain Improved Alignment

Now, to be clear there are several pitfalls to this measurement. Since the speaker sources are separated by a distance....if you measure somewhere else, the distance difference between the speakers will change. That means these speakers are time aligned for only one point in space. Only coincident source speakers are effectively immune to this.

Additionally this method doesn't ensure phase tracking for your crossovers through the crossover frequency...so you might still have some pretty gnarly frequency response deviations. Improving that is for another writing session.

However, we've found that many stacks can start out with mids/highs several mS of difference between them and other sources, more if there are horns with a long path for a low cutoff frequency. If you start out with this method, and then start adjusting frequency response flatness by adjusting crossover point and slope, you can go a long way towards improving the sound of a system.