Time Alignment by Sine Wave Convolution - example

(revised 12/18/2016)

Introduction: It may help to confirm the microphone is equidistant from left and right channel. Measure interchannel time differences using Microphone alignment in Acourate in the logsweep menu.

Quit Acourate Convolver if not already quit. Make sure your clocking is correct and the interface connected to acourate is locked to the converter which has the microphone. Only if the interface incorporates a microphone input or takes the input from a microphone preamplifier, then it can be on internal sync.

For front left and front right, set microphone alignment as follows:



The output channels may vary depending on your system patching. 1 or 2 samples delay is very acceptable. That's only .04 ms at 48 kHz sampling.

Sinewave Convolution to determine alignment of subwoofers with the mains:

1. Create a temporary three way XO:

Crossover 3-way Speaker	
Crossover:       C:\\LF-RF48 - NT LP xo\4 inches higher         60       0         f lowest (Hz) Delay 2       Butterworth         4000       0         f low (Hz) Delay 3       Neville-Thiele         0       0         2       Filter order	er Filter taps 65536 V DelAIIXO Crossover filter file format Single precision = .pcm Oduble precision = .dbl
f high (Hz) Delay 4 0 f highest (Hz)Delay 5 0 0 0 0 0 0 0 0 0 0 0 0 0	Calculate and Save

For the lowest frequency use the intended crossover frequency for your subwoofer. Then in Explorer delete XO2. Rename XO3 as XO2 for both channels. This turns it into a two way XO without the midrange as explained in the PDF "Time Alignment of Drivers by Sine Wave Convolution". Save a multiway XO from these two elements with the name "multiway XO no mid.wav".

Then with both the subs and mains turned ON record an Acourate Logsweep with this no mid crossover:

(((acourate®-LSR))) - ASIO LogSweep Recorder	
Soundcard         Driver:       ASIO Lynx         Input Channel:       Lynx AES16e Record 01         Output Channels:       Lynx AES16e Play 01	ASIO Control
Logsweep Data 0.5 10 24000 0.1 0 Fade In [s] Sweep Start [Hz] Sweep End [Hz] Fade Out [s] Sweep Slope [dB] S	60 Sweep Time [s]
Samplerate     Channel Selection     Target Length / Lead-In     PeakOpt     I       48000     V     Stereo     65536 / 6000     Add Dirac     F	nvert Pulse Polarity Freeze Gain
-40 -35 -30 -25 -20 -15 -10 -5 0 [] Recording Level [dB]	
Progress	2:1
Mic Calibration Filter 4 channel multiXOnomid.wav	
onvolver_user\Documents\Acourate\Surr work parts 48k 1 Output Volume: 0.08 equals -22 dB	2 8 16 Dynaudio\Ll
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Then go into explorer and rename the two resulting pulses as:

Pulse48L no mid.dbl, etc. Or save the pulses with the new names from within Acourate.

# 2. Create an NT 2nd order linear phase XO. In this example I used 60 Hz:

<u>K</u>	(((acourate)))® - Crossover Ger	erator 🗕 🗖 🗙				
Crossover						
2-way Speaker Crossover:	2-way Speaker Crossover:					
60 0 f lowest (Hz) Delay 2	Filter Type Butterworth Linkwitz Neville-Thiele	Filter taps 65536 ♥ ♥ DelAIIXO Crossover filter file format				
f low (Hz) Delay 3 0 f high (Hz) Delay 4 0	2 Filter order Phase O minimum Inear	double precision = .dbl     Calculate and Save     Calculate and Save				
f highest (Hz)Delay 5		Exit				

Then save a multiway XO.wav from these XO files as

multiXO no offset.wav

**3.** Acourate Logsweep recorder with subs turned off:

<u>R</u>	(((acourate®-LSR))) - ASIO LogSweep Recorder	- 🗆 ×
Soundcard Driver: AS Input Channel: Lyr Output Channels: Lyr	SIO Lynx /nx AES16e Record 01 /nx AES16e Play 01 v	<ul> <li>✓</li> <li>✓ ASIO Control</li> <li>✓</li> </ul>
Logsweep Data 0.5 Fade In [s]	10     24000     0.1     0       Sweep Start [Hz]     Sweep End [Hz]     Fade Out [s]     Sweep Slope	60 e [dB] Sweep Time [s]
Samplerate 48000 v	Channel SelectionTarget Length / Lead-InPeakOpStereo<	t 🔄 Invert Pulse Polarity ac 🗌 Freeze Gain
-40 -35 -40 0 · · · · 0 · · · ·	-30 -25 -20 -15 -10 -5 0 0 0 0 0 0 Recording Level [dB]	
Progress		
Mic Calibration Filter 4 channel	ItiXO no offset.wav	
Output Volume: 0.08 e	Please check if driver and channels are p equals -22 dB	properly selected!
<		>

Save these two pulse files with the names:

Pulse48L no sub.dbl, etc. Do this from within Acourate with save or in explorer rename the files.

#### 4. Prepare to convolve.

Since the left channel no sub is already in slot 1, load the "no mid" into slot 2 of Acourate. Amplitude display looks like this:



### 5. Generate a sinewave at the XO frequency

Switch to slot 3. Generate sine wave. The amplitude plot should look like this:



#### 5. Convolve the no sub pulse with the sinewave into slot 5.

Convolve the no mid pulse with the sinewave into slot 6:

📶 (((acourate)))® - Signal Convolution 🛛 – 🗖 🗙							
Convolution							
Time domain		Time domain		Result Curve			
○ Pulse48L no s		○ Pulse48L no s		○ Pulse48L no s			
Pulse48L no r		O Pulse48L no r		⊖ Pulse48L no r			
◯ Sinewave		Sinewave		◯ Sinewave			
Series4	*	Series4	=	⊖ Series4			
⊖ Conv1_3		⊖ Conv1_3		⊖ Conv1_3			
⊖ Conv2_3		O Conv2_3		Conv2_3			
Convolution							
				🖌 🖌 Exit			

Acourate now looks like this, in the time domain, zoomed into near the beginning of the pulses. Note, if you see dashed traces on your screen, it's pixelation. Just increase your zoom and they will turn into lines and curves.



Zoom out. Then zoom into the center area of the graph. In my case the no mid (which contains the sub) has a higher amplitude than the no sub. Adjust the appropriate gain to make the amplitudes more equal, to make it easy to compare phase:



No surprise, we see the sub is delayed.

Zooming in further on the "steady state" area we see that the peak of the main is at 76339 samples and the peak of the sub is at 76537 so the left sub is delayed by 198 samples in the steady state area:



6. We rotate the sub left by 198 samples and it then appears to coincide with the mains in this center steady state area.

We next add 6000 leading zeros to the sine wave so we can see the optimal start point when we zoom in at the transition zone. It's just a visual aid, you don't have to look at the sinewave start but it will help identify that the transition zone begins just before 6000 samples. Because the two impulses in our system begin at 6000 (in the PDF example, the impulses begin at 12000, use which ever value is appropriate).

We then zoom in on the transition zone and we see:



The sub becomes just a little bit early in this zone, after rotation. So either we decide to make a compromise, or we decide to favor the steady state zone or the transition zone. I chose to make the steady state zone exact, since psychoacoustically transients come from high frequencies, and bass notes are appreciated largely from their sustain. Since the filter takes time to settle, and the ear reacts more to the sustain of the bass frequencies, it seems to me psychoacoustically logical to use the steady-state zone. So we note that the left sub must rotate left 198 samples in this case.

Here is what the steady state zone looks like after proper rotation:



The two pulses are so perfectly aligned they look like one.

## **7. Repeat the above procedure for the right sub** and note the rotation required.

### 8. Clear the traces. Then load XO1L and XO1R into Acourate slots 1 and 2. Rotate each XO by the required amount.

Save the two rotated low pass sections. Save a multiway filter. I include the offset amount in the file name to make it clear what was done:

multiXO -198, -204.wav

**9. Logsweep record with that multiway filter and go through the macros**. Save CPVs for acourate convolver. Enjoy the enhanced listening!

N.B.: Uli reports that this kind of driver alignment does NOT change after excess phase correction because with the aligned subs and mains, Acourate sees pulse responses for complete speakers (like a speaker with a one-way driver). So the excess phase correction is applied in common to all drivers. There is no longer a separate phase shift for sub and speaker. This is another advantage of the sine wave convolution method.

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