SECTION 5 - ACOUSTICAL SYSTEM TESTING
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SECTION 5 - ACOUSTICAL SYSTEM TESTING

When building prototype crossovers for the final acoustical (ear) testing, low impedance in the connections is extremely important, as it more accurately resembles the final soldered connections. Avoid alligator clips, if possible, and consider terminal blocks instead.

Note: After measurements are taken, save a copy of the .frd files with an alternate name so that they do not get overwritten. You will be using these files to generate Distortion Measurements such as the Waterfall plot.

BOX RESPONSE MEASUREMENTS
(Section Index)

1. Mount the woofer, flush to the baffle, if possible. Attempt to seal all sources of leakage including the woofer mount (you might consider using door/window insulation rubber).

2. The connections remain as they were for impedance, capacitance, and resistance testing except that the woofer is now the object of the test.
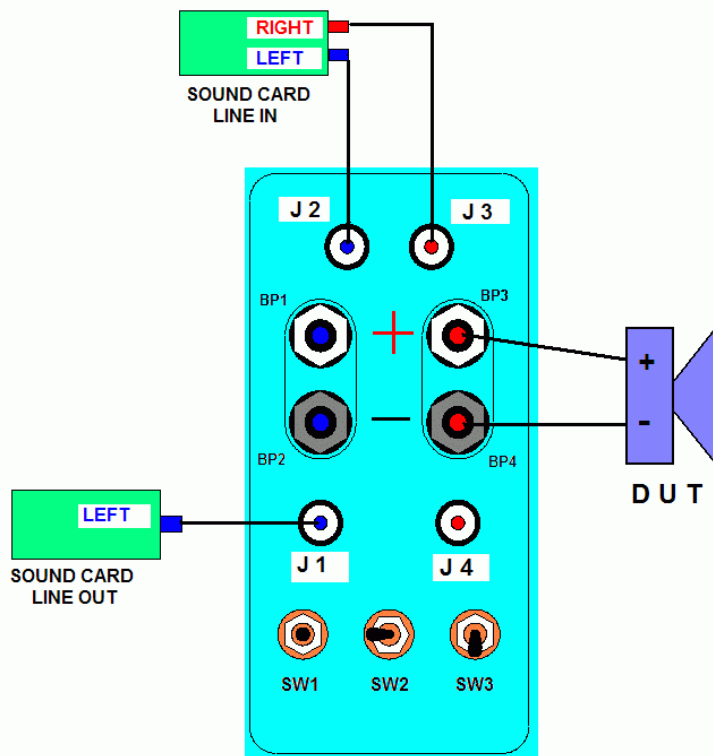
3. Leave the volume and sampling frequency as they were set for impedance testing.

4. **Check** to assure that the repeat count is as large as you can tolerate as the larger this is the more accurate the measurement is likely to be (recommend 10 or more).

5. **Double Click** on 🔊 /Measure/Impedance

6. **After** the measurement ends, there will be a new file created called SpeakerA.Impedance.

7. **Right Click** on Speaker A.Impedance to view the chart.
    a. For a **Closed Box** the Peak is Fc and is larger than the Fs (free air) measured earlier. It will be inversely proportional to the box volume.
    b. A **Ported Box** will display 2 peaks: F_L is lower than Fs and F_H is higher than Fs. The lowest point of the dip between F_L and Fs is approximately the tuning frequency of the box, F_B (the exact value is equal to the geometric mean of F_H and F_L). The phase should be at zero at F_B.
        i. To find F_B exactly
            1. Select (activate) the chart.
            2. Resource/Export

3. Name and save the file with an easily identifiable name and location.
4. Open the file in Notepad or WordPad.
5. Look for the zero phase point between F_L and Fs.

a. *An alternative way of measuring F_B is as follows:*
   i. After measuring and obtaining F_L and F_H:
   ii. Seal off the ports using wood and a sealing material.
   iii. Double Click on 🔊 /Measure/Impedance
   iv. This measurement will give an Fc value (closed box value).
   v. $F_B = (F_L\wedge2 + F_H\wedge2 - Fc\wedge2)\wedge(1/2)$.

8. You can use the Data Check Spreadsheet (written by Claudio Negro) to calculate F_B and other parameters like Q.

"data check.xls"

9. Vary the port length until the desired box tuning frequency is obtained (a longer tube decreases F_B assuming a constant port diameter and box volume).
   a. The lower you tune the box, the more similar it becomes to a sealed box.
   b. Use the zero phase crossing point between the two peaks of impedance to tune the box. Do this by zooming in on the impedance plot, place the mouse over your target and read the frequency.

10. Dimensions of standing waves can be observed as small peaks on the impedance graph. Expected frequencies at which this will occur are at c (speed of sound)/(2*any box dimension) (you would expect to see it for each box dimension). A spreadsheet is provided to calculate this for you. Simply use either Imperial or Metric units:

"frequency of standing waves1.xls" **OR** "Room Mode Calculator1.xls"

which was written by Allan Devantier from Harmon International and is meant for room response but will work for in box mode estimations. One would want to target moving the resonance to as high a frequency as possible in order to allow for damping by stuffing and damping material or to spread out the modes as much as possible in order to minimize the energy in any region of frequencies. One other spreadsheet that is somewhat more comprehensive that is also targeting room design is called the Control Room Calculator (copyright Chris Whealy,

"Control Room Calculator V2.62 XL9" 2004), which demonstrates other ratios other than the Golden Ratio that can be quite effective in spreading out modes in a room.

11. If the peaks are large, the addition of stuffing may decrease this at the expense of box efficiency (Q loss).

ACOUSTIC MEASUREMENTS
(Section Index)

This type of measurement requires an external amplifier and a measurement microphone. Speaker Workshop uses the Right Channel for data and the Left Channel as reference by default (however you can select which channel is used for data and which is for reference if so desired).

Though most solid-state high fidelity amplifiers will introduce little distortion, one might want to calibrate the amplifier and insert this measurement into Speaker Workshop. Speaker Workshop will work without this calibration and if you choose to bypass this.

With Jig II, the reference channel is on the left channel and the microphone response is on the right. Options/Preference/Acoustic should be checked to assure it is set up this way.

If during your measurements, you notice that one channel has a significantly greater amplitude than the other, there is a problem. If there is a significant difference, it may be that one channel is getting no signal other than cross talk. To test:
1. *Connect the line in to the line out on the sound card*
2. *Resource/New/Driver*
3. *Select this folder to make the driver active*
4. *Measure/Frequency Response/Nearfield*
5. *Select Reverse channels*
6. *If the volume levels are now reversed, the sound card is okay and the problem is with the measurement set up (microphone or cables).*

After taking the initial acoustic measurements, you may want to evaluate the speaker for accuracy. Impulse measurements are done to assist with this. Tests such as the Waterfall (Cumulative Spectral Delay) and Step response will require that you have saved all of your original acoustical testing to use for these procedures. It is recommended that you save them to a file that is easily retrievable at a later time.
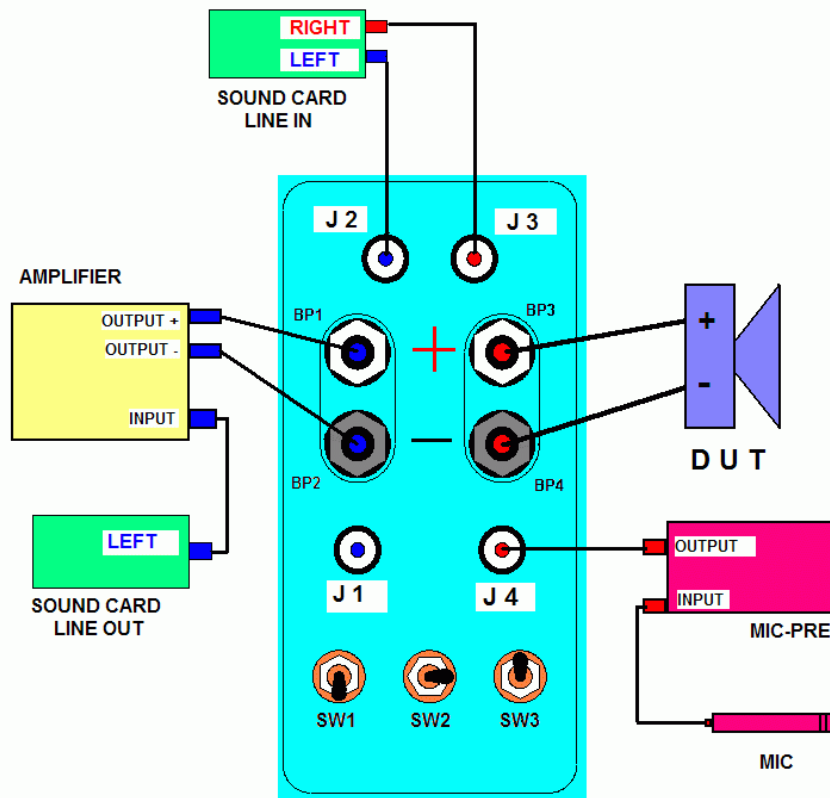
AMPLIFIER CALIBRATION

(Section Index)

If you choose to calibrate the amplifier, be very careful because there is a risk to overloading the soundcard. Prior to allowing any direct connection between the amplifier and the sound card, first hook up your DMM to the output to assure that no more than 0.5 to 0.75 volts are being used to test (most soundcards can tolerate at least 0.75 volts).   If you are going to use an external amplifier, this is recommended for linear frequency response (though this is not generally an issue with most hi fidelity equipment) and to assure a "non colored" phase measurement.

Connections are made from Line Out to the Amplifier Line In and then Amplifier Line Out to Sound Card Line In Rt and Line In Left.

Another option is to use the Wallin jig, which by the nature of its voltage divider will decrease the chances of burning the soundcard. Hook up with the Wallin Jig would be as follows:

Acoustic Measurement and Amplifier calibration set up.   See text for attenuation settings.

1. Note that the switches are in loop mode.
2. Set the amplifier volumes all the way down.
3. Do a sound card record of an MLS wave (using a repeat count of 10) and slowly increase the amplifier volume until you reach 20-22K on the SW Vu meter.
4. Options/Preferences/Select Measurement.

5. Make certain that the sample rate slider is all the way to the right and the volume is set to 100.
6. The sample size should be set to at least 32K to assure adequate frequency response.
7. Check the "Use Pre-Emphasis" if you want to use this for the frequency response measurements.
   a. Pre-emphasis acts as a low pass filter system which increases the signal to noise ratio for lower frequency tests but it may cause a sudden roll off of high frequency response if selected. <u>If the amplifier calibration is not used, this should not be checked.</u> Even if amplifier calibration is used, pre-emphasis does not need to be checked as it will make the sound more tolerable during testing however it will decrease the accuracy of results.
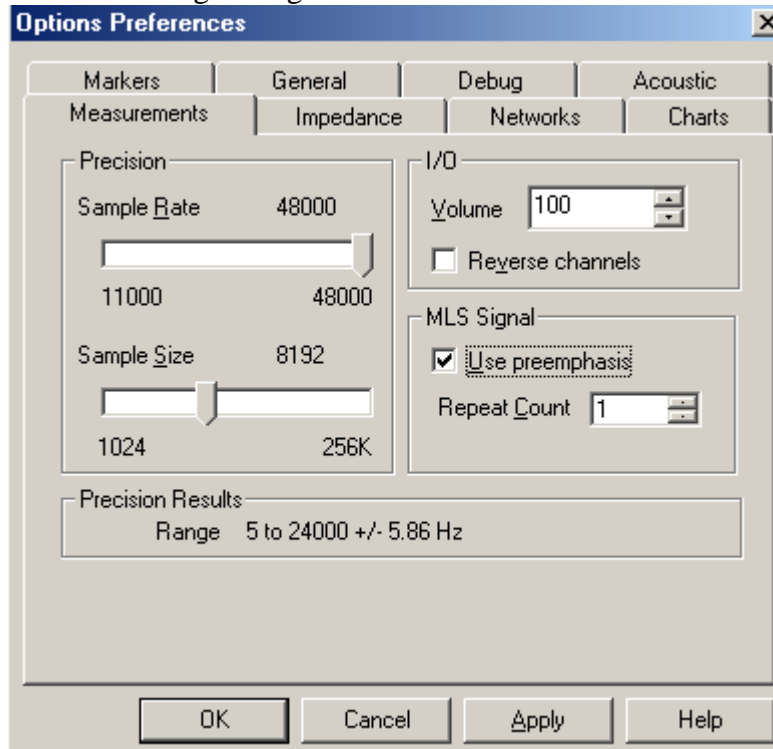
8. Click "OK".
9. Options/Calibrate/Amplifier/Response/Measure
10. Go to the System file/Measurement/Impulse
11. Click Calculate/Frequency Response
12. A new file will be created called Measurement.Impulse.Frequency
13. Double click to Open Measurement.Impulse.Frequency to view the Amplifier Frequency Response.

THE MEASUREMENT MICROPHONE
(Section Index)

The Measurement Microphone must be an Omni Directional Condenser Type. This type of Microphone requires a pre-amplifier with phantom power. The most commonly used microphones are the Mighty Mike, the Behringer ECM 8000, or DIY microphones (generally made using either the Panasonic Capsule WM 60 or WM 61).

One can import a calibration file for the Microphone as well though this is also not required in Speaker Workshop.

To Import:

Option/Preferences/Calibrate/Mic Calibration/Browse/choose the mic calibration file
(this may be either a generic file or one specific to your microphone)/Okay/Import

Many people will send their microphone or capsule to Kim Girardin for calibration. He may be contacted at kmgrdn@luminet.net or by fax at 507-4545-8844 or by mail at 1400 Homer Rd, Suite 2, Winona, MN 55987.

One could also calibrate the preamp in a similar fashion to the amplifier. The Behringer UB802 mixer/pre-amplifier, which is often used with Behringer Measurement Microphones, is quite flat and introduces very little noise. It has a balanced input and phantom power. Make certain that with any mixer type pre-amplifier, that the controls are set so that the tone controls are at mid point.

The Behringer UB 802 pre-amplifier is hooked up as follows:
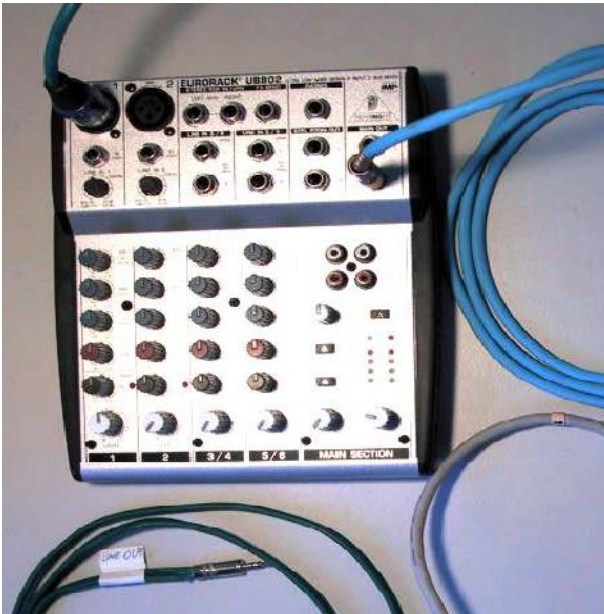
MEASURING ACOUSTIC RESPONSE

For testing with the Wallin Jig 2, set the jig up in loop mode for acoustic testing as follows:

.

The overall acoustic measurement set up is as below if you are using the cable (Claudio Negro jig) set-up:

There are 4 stages to acoustic measurements:
   a.   Nearfield

b. Gated Farfield
c. Splicing
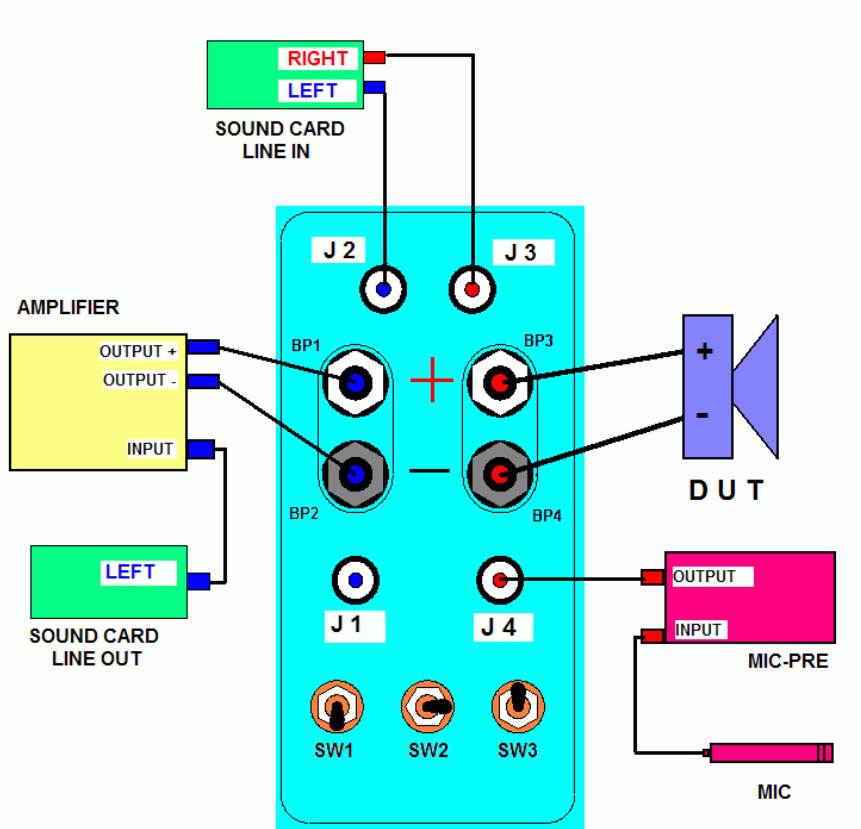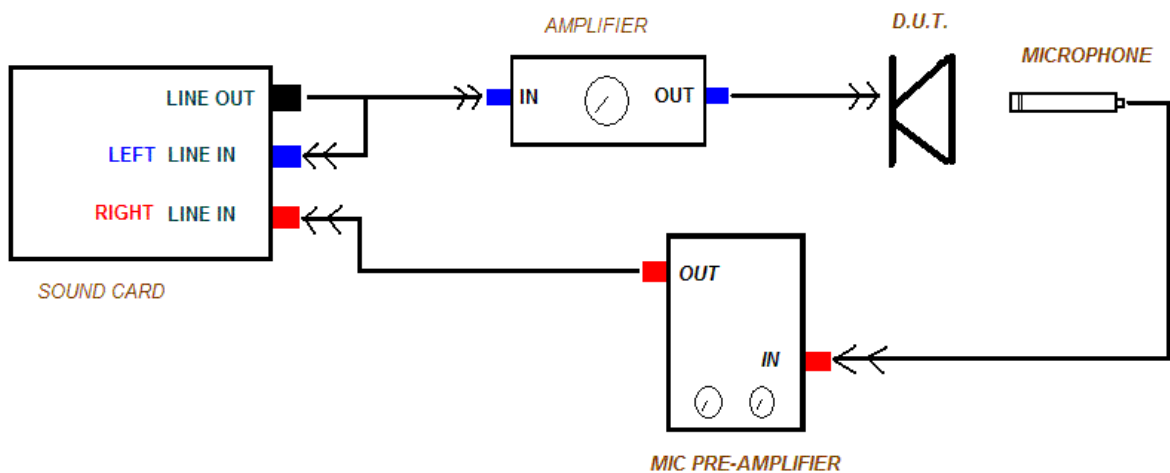d. In room system evaluation using Non-gated Farfield.
    i. This includes:
        1. in room frequency response with reflection measurements
            a. both on and off axis measurements
            b. Professional loudspeakers use a measurement called Beamwidth Delimited Power Response (BDPR); this averages the frequency response over the radiation direction that is of interest; i.e., the radiation pattern that is necessary for the audience. In a living room, this may be limited to a 20 or 30 degree off horizontal axis range. A vertical axis may be limited to a range even smaller than this. The Beamwidth is commonly defined as the angular distance between two points on either side of the primary axis where the sound pressure is 6 dB below the maximum pressure. The maximum amplitude is often found on the primary axis but not always depending upon the transfer function. The smoothness with which the decrease in amplitude off axis is seen is the most important parameter of this measurement (aside from knowing the actual usable beamwidth).
        2. and tests to evaluate for linear distortion (Impulse Testing in the time domain which contains implicit information regarding frequency response, phase response, step response, CSD or Waterfall plots) and non-linear distortions (Total Harmonic Distortion and Intermodulation Distortion).

SETTING THE VOLUME TO MEASURE ACOUSTIC RESPONSE
(Section Index)

1. With Jig II, the reference channel is on the left channel and the microphone response is on the right. Options/Preference/Acoustic should be checked to assure it is set up this way

2. Play an MLS signal. (Right click /properties/MLS/OK/ and set for a loop)

3. Look at the VU meter in SW software to assure that the Left and Right Channels are approximately equal.

4. Adjust volumes lower than the clipping reference level (you could use your earlier setting from QuickMix).

5. If you set the volume too high on the amplifier, you will have to use attenuation on the jig (10 to 20 dB), which may make the reference level too low.

6. Hook up your DMM to the input of the amplifier and adjust the volume until you are putting out approximately 0.5 to 0.75 volts. This is the voltage that will later go to the input of the Sound Card.

7. Hook the DMM up to the output of the amplifier.

8. Continue to play the signal and adjust the volume of the amplifier until you get approximately 1 volt at the output. Caution must be taken to not allow too much output voltage as the crossover network is not yet in place to protect the tweeter.

NEARFIELD MEASUREMENTS
(Section Index)

By measuring close to the woofer, the effects of low frequency standing waves, reflections, and diffraction (baffle effects and beaming effects) are minimized or eliminated. This is effective only at frequencies at which the diaphragm behaves like a piston. Higher frequency signals, however, have shorter wavelengths and therefore experience phase interference leading to erratic measurements at higher frequencies. The effective upper frequency limit of this technique is defined by the effective diameter (De), which is defined as the diameter of the cone plus approximately ½ of the surround. A useful way to measure this is to measure from the attachment of the surround to the frame on one side to the attachment of the surround to the cone on the other. The effective upper frequency limit of this technique (Fmax) is approximately equal to: c/(PI()*De) where c is the speed of sound and De is the effective diameter (the diameter of the driver plus 1/3 to ½ of the surround). D'Appolito notes that this limit is actually somewhat lower if you are not working with an infinite baffle.

The microphone must be placed very close to the center of the dust cap to prevent environmental influences. For speakers, the near field is generally considered distances which are less than lambda/(2*pi) from the driver (combining with the above formula, you get De/2). Beyond this point, the pressure & velocity are related by free-space radiation. To be unambiguously in the near field you probably want to be less than 1/10th of this distance (to account for field radiation variations and the inset of the acoustic center or De/10) . D.B. Keele suggests that near field measurements can be adequately measured at a distance of 0.055*De (1/18th of the De) or less from the dust cap of the driver being measured. It would be wise to pad the microphone stand to avoid or decrease reflecting waves as much as possible.

The following spreadsheet written by Claudio Negro and myself. It uses Keele's estimates for near field and may be of assistance with far field measurements, as well. It

"Calc phase and freq
dip from floor bounce

also includes phase, dip, and peak information:

Procedure:

1. Run an MLS test signal in SW to assure that the Volume in the software is set to record at your maximum non-clipping level (usually 20-25K).
2. Assure that your sample rate and sample size of at optimum for your sound card as tested before in the SW Set Up Section.
3. Option/Preference/Repeat Count and verify that the repeat count is set to at least 5 (the higher the number, the longer the test will take but the more accuracy you will achieve).
4. Check the "Use Pre-emphasis" only if you did an amplifier calibration (Note: Pre-emphasis acts as a low pass filter system which increases the signal to noise ratio for lower frequency tests but it may cause a sudden roll off of high frequency response if selected. <u>If the amplifier calibration is not used, this should not be checked.</u> Even if amplifier calibration is used, pre-emphasis does not need to be checked as it will make the sound more tolerable during testing however it will decrease the accuracy of results).
5. Double click on 🔊 /Measure/Frequency Response/Nearfield
6. Assure that values for the Right and Left Volume Meters in SW are almost identical and in the ideal range as determined previously (usually 20-25K).
7. A new file will be created: SpeakerA.Nearfield.
8. Open this file and look at the graph.
9. If you are working with a closed box design, move on the Far Field Section.

10. If you are working with a Ported Box Design:

11. There will be a dip around the vented system tuning frequency (about 38 Hz in the example below) since this is the frequency at which the tuned box/port combination puts its maximum load on the woofer diaphragm.

12.

13. Leave all settings as they were for the Driver Measurement.
14. Place the microphone in the center of the port but without breaching the port boundary.

Note: Port output generally provides a broad peak. A more accurate measure of the system tuning is to look at the near field response of the woofer in a vented system.

The output minimum of the woofer is the tuning point of the system. This is an even more accurate measure than the zero phase crossing point (D'Appolito).

The only place that you can measure and combine the port output and driver output is at the tuning or resonance of the system as all other points include unknown relative phases of the two. The combination formula is 20*LOG(effective port diameter/effective woofer diameter).

15. Measure/Frequency Response/Port Response.
16. A new file will appear named Speaker A.Port
17. If you have more than one port, rename this file SpeakerA.Port-1 (or any identifying label) and then measure the next port following the same procedure.
18. Combine multiple port measurements (if applicable) as follows:
    a. Highlight the first port
    b. Calculate/Combine
    c. Choose Add
    d. Select the 2nd port file (in the other window)
    e. Click OK
    f. The first port file will change. Rename this SpeakerA.port combined
19. Combine the port and the driver results as follows:
    a. Double click SpeakerA/Driver/Merge Port Response
    b. A new window appears
    c. Select the appropriate nearfield driver and port responses.
    d. Look at the Piston Area and verify that Sd (piston area) is equal to that in the driver properties.
    e. Enter the total area of the ports (internal area). Each port area = PI * (.5*internal diameter)^2. Remember to enter the sum of all of the port areas if you are using multiple ports.

Note: The measurement for the vent has to be corrected for the vent surface area by 20*log (speaker radius/vent radius); Speaker Workshop does this for you. Port correction for an MTM = 20*LOG(port area/(sum of the woofer areas))^.5 This is the amount that you would subtract from the near field

port response in addition to (near field – far field response) in order to align the near field and far field responses. (Figures 5.25a&b from D'Appolito).

If you use a 2 port system, the output of the two ports will be 6 dB more than one port. The correction would be added to this. The correction would be = 10*LOG(sum of port areas/sum of woofer areas).

    f.   Click OK.

    g.   A new file will be created called SpeakerA.Merge Nearfield.

    h.   Remember that this response is useful only up to a frequency equal to: c/(PI()*De)

TUNING THE PORT(S)

1. Do an impedance measurement with the woofer in the box.
2. Examine the plot.
3. The lowest point between the two peaks of the saddle is your port tuning frequency.
    a. Use the zero phase crossing point between the two peaks of impedance to tune the box. Do this by zooming in on the impedance plot, place the mouse over your target and read the frequency.
4. Corroborate this by making certain that the phase is zero at this point.
5. The target is to obtain the lowest Impedance or zero phase point at the enclosures resonance frequency by adjusting the port and the stuffing.
6. A longer port tunes the box lower however it will require a larger diameter opening to prevent air turbulence.
7. The maximum length of the port is of concern in that longer ports are more prone to act as Helmholtz resonators so limiting the port length to less than $1/10$ to $1/20^{th}$ of a wavelength of the frequency to which it is tuned is recommended. This drives the resonance of the tube up to higher harmonics that are less likely to be audible.
8. The lower you tune the box, the more similar it becomes to a sealed box.
9. Using Unibox from FRD Consortium at http://www.pvconsultants.com/audio/frdgroup.htm, you can easily and accurately model different vent lengths and diameters and immediately see a predicted graph.
10. When evaluating a port, the ultimate output will be 20log(radius of port/radius of driver) in dB compared to the driver output. This should be calculated when combining nearfield port/driver response.   Of course, far field measurements already measure for this however an estimate of output can be made in near field measurements by using this correction. Speaker Workshop calculates this when combining port and driver responses.

GATING AND MLS OPTIMIZATION
(Section Index)

An understanding of MLS measurements is important for acoustical measuring. If you set the gate length visually with SW, the following clarification may not be necessary, however it is good to understand.

The further away the microphone is from the driver under test, the more room reflections there are that are likely to be picked up. Since sound travels approximately 1130 ft/sec, a 10 ms gate will record 11.3 feet of sound, which is roughly the equivalent of 100 Hz (1130 ft/sec /11.3 = 100 Hz) so anything below this frequency will not be measured. A 50 ms gate increases the sound wavelength to 56.5 feet or down to 1130/56.5 or 20 Hz. Another way to look at this is that 1000 divided by the gate length in ms is equal to the lowest usable measurable frequency. A further explanation may be found at http://members.aol.com/_ht_a/pjay99site/MLSmain.htm.

An excel spreadsheet gives the values (simply plug in the approximate temperature and humidity [use 50% for humidity if you are not certain]):

"Frequency vs wavelength1.xls"

The lowest usable frequency with a gated measurement can be calculated by the inverse of the gate duration, i.e. a 5 ms gate = 1/.005 = 200 Hz. The following spreadsheet may

"calculating MLS gate2.xls"

be of assistance:

Far field measurements become corrupted by reflections, which lead to standing waves. Near field measurements must be taken close enough to avoid diffraction and room effects. These produce a true 2pi response. The effective frequency is up to where the driver stops behaving like a rigid piston. At higher frequencies, the sound waves from various parts of the diaphragm arrive at the microphone in different phases causing signal augmentation and cancellation. The upper limit for near field measurements is equal to the speed of sound divided by the effective circumference of the driver or = c/(pi*D) (D is the effective diameter). If the microphone is placed less than 0.11 x the driver radius away from the driver it will yield less than a 1 dB error (Keele).

To convert near field into far field measurements (assuming equal input voltages), you would subtract 20*LOG(x*d/r) from the near field measurements where x is either 2 if the far field measurement is into a 2 pi space or 4 if into a 4 pi space, d is the distance to the driver acoustic center and r is the effective radius of the driver (the measurements must all be in the same units).

In the far field, driver pressure drops off with the square of the distance or by 6 dB each time the distance is doubled. Line sources are the exception to this as they drop off by 3 dB for each doubling of the distance because they are not acting as a point source unless you are 10-20x the array length away from the drivers. According to D'Appolito, far field measurements should be taken a minimum of 3-5x the driver diaphragm diameter away from the driver.

To combine near field and far field data, find the highest effective range of near field and the lowest effective range of far field and the combining point will be between the two.

Ground plane measurements eliminate floor reflections and if far enough away from other reflecting surfaces will yield a true anechoic response raised by 6 dB. Ideally, the speaker should be tilted slightly so that the driver is on axis with the microphone. The angle would be =Arctan (height to the center of the driver from the ground/distance of the base of the driver from the microphone). As long as far field conditions are met, placing the driver at 2 meters for measurement provides the benefit of yielding a true 1 meter sensitivity due to the natural 6 dB boost with this measurement. Ground plane measurements can be used to measure the vertical axis as well by simply lying the speaker on it's side and taking measurements at different angles at exactly the same distance from the driver in a semi circle.

Ground plane techniques make the speaker appear to have a baffle twice as wide as it actually does, thus it is effective for resonance measurements but not for diffraction effects. Because of the close spacing of the microphone to the ground, the only interference effects will be at very high frequencies (frequencies representing wavelengths that are smaller than the microphone diaphragm).

A limitation to the Ground Plane Technique is that to measure the stand, the speaker must be lifted off the ground and you are then measuring the stand resonances, as well. This could be desirable. One must keep in mind that the measurement is an off axis one unless the loudspeaker is tilted towards the microphone though the lower frequency waves are fairly omni-directional at this point.

Indoors, one could put the microphone at the junction of the floor and the wall and then angle the loudspeaker at 45 degrees to the wall to avoid reflections from the wall to the

loudspeaker and back to the microphone. From http://stereophile.com/features/405time/:
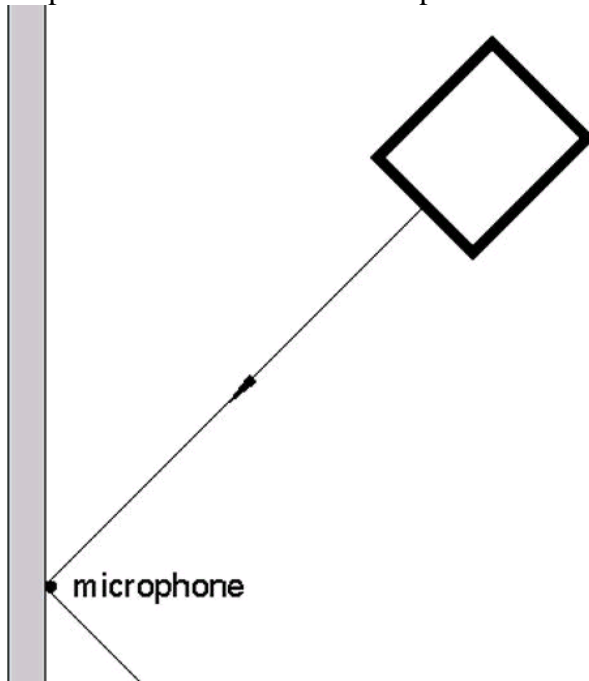
Doing this requires that both walls be fairly equal in their reflective properties. One could do this by placing the microphone at the junction of three surfaces, as well. An example of a two plane measurement system from http://stereophile.com/features/405time/:

Ideally, any reflecting surfaces should be at least 5x further away from the DUT than the microphone so that reflection errors will be at least 20 dB lower than the main signal. The signal decrease is calculated by 20*LOG(2*distance).

FAR FIELD GATED MEASUREMENTS
(Section Index)

You can take far field measurements with or without gating.

Gated measurements set limitations for the time of the collected information in an effort to minimize environmental influences. The industry standard is usually based upon one meter measurements though line arrays and ribbon tweeters are often measured at two meters. At issue is the attempt to see the source of sound as a point source. Too short a distance, and measurements do not reflect a point source. As the distance increases, so does the Signal to Noise ratio. Some have reported using as an estimate 5 times the driver effective diameter to a maximum of 1 meter. One might consider doing measurements at several distances until the curve looks consistent though in most cases, 1 meter is likely to work out well.

With gated measurements the time interval between the start marker and the stop marker will determine the low freq. cutoff of the measurement as well as the resolution of the measurements. For example, the time window must be around 5.0 msec to obtain meaningful measurements down to 200 Hz (1/.005=200 Hz).

Increasing time gate gradually does not change resolution gradually. For example, going from 6ms gate to 8ms gate does not make any difference for 44100Hz. The table below demonstrates this.

sampling rate * 1000	data points	16	32	64	128	256	512	1024	2048
11	from (msec)	1.45	2.91	5.82	11.64	23.27	46.55	93.09	186.18
	to (msec)	2.90	5.81	11.63	23.26	46.54	93.08	186.17	372.35
12	from (msec)	1.33	2.67	5.33	10.67	21.33	42.67	85.33	170.67
	to (msec)	2.66	5.32	10.66	21.32	42.66	85.32	170.66	341.32
22.1	from (msec)	0.72	1.45	2.90	5.79	11.58	23.17	46.33	92.67
	to (msec)	1.44	2.89	5.78	11.57	23.16	46.32	92.66	185.33
24	from (msec)	0.67	1.33	2.67	5.33	10.67	21.33	42.67	85.33
	to (msec)	1.32	2.66	5.32	10.66	21.32	42.66	85.32	170.66
44.1	from (msec)	0.36	0.73	1.45	2.90	5.80	11.61	23.22	46.44
	to (msec)	0.72	1.44	2.89	5.79	11.60	23.21	46.43	92.87
48	from (msec)	0.33	0.67	1.33	2.67	5.33	10.67	21.33	42.67
	to (msec)	0.66	1.32	2.66	5.32	10.66	21.32	42.66	85.32
96	from (msec)	0.17	0.33	0.67	1.33	2.67	5.33	10.67	21.33
	to (msec)	0.32	0.66	1.32	2.66	5.32	10.66	21.32	42.66

You would need to go beyond 11.61ms for satisfactory resolution (512 points and more). Therefore, the larger the gate, the better the resolution, however you will need as close to an anechoic chamber (outside measurements) as possible as a larger gate causes noise immunity to decrease. A larger sample size and a slower sample rate are more prone to noise effects. Low frequency measurements require a larger sampling size and a slower sampling rate in order to get a reasonable amount of data per octave (resolution).

Tweeter measurements do well with smaller sampling sizes and faster sampling rates as they have many more frequency points in each octave so, in order to provide the best noise immunity, you will want to raise the SAMPLE RATE and lower the SAMPLE SIZE for tweeter measurements.
.
You can check the number of data points (resolution) by right clicking in the dataset and choosing Properties/Specifics tab. You can change this number to attempt to improve the graph but once you repeat a measurement, the program will reset this number according to its algorithm.

Unless you are measuring outside or in a very large room you will be including some room reflections in the marker settings. The only way for SW to get more data points using the gated measurement is using a card with a higher sampling rate (96K) or getting clean measurements with a longer time interval. Speaker Workshop limits the resolution of the data looking at the sample rate and the time gate. It selects whichever is lower. The difference between two gate points is what counts. So it helps to move the starting gate at 0 or 0.1.

An MLS size of :
1024 gives 512 points
2048 gives 1024 points etc.

To get low frequency data, the normal procedure is to do Nearfield measurements and then splice into the gated measurement somewhere above 1/(time window in the gated measurement in seconds) Hz (or for 5 milliseconds, you would get data to 1/.005 or 200 Hz)..

Measurements may be done in a room (as described below), using "ground plane" measurements outside (microphone on the ground at least one meter from the speaker, speaker tilted forward, and the nearest reflecting surface at least ½ times further than the lowest frequency wavelength of interest; usually 20 Hz or 29 feet), or in an anechoic (non reflecting) room. This is best done with sound absorbent material on the ground between the measurement microphone and the Speaker.

One other issue regarding the setting of the start marker is the phase. If you set the starting marker just prior to the impulse, you will minimize the travel time in your measurements and decrease the amount of phase rotation. As you get further away from Minimum phase, the phase will wrap more frequently making it more difficult to visually inspect and choose a crossover point. Setting too long a period prior to the first pulse may result in a "zig zag" type phase curve. This is due to having excess delay (too much time in flight that is included which results in a lot of phase rotation). You can remove the excess delay by Transform/Delay. This can be done manually (inserting your estimated time of flight and remove the check from *Remove excess* delay) or by checking *Remove excess delay* and letting SW do it. The caveat to allowing SW to do it is that SW may not place the starting marker at the same point for each driver so, though it may be

close, it will not be the same as if you set the marker and leave it for relative phase measurements.

Note the saw tooth phase pattern when one does not remove excess delay:
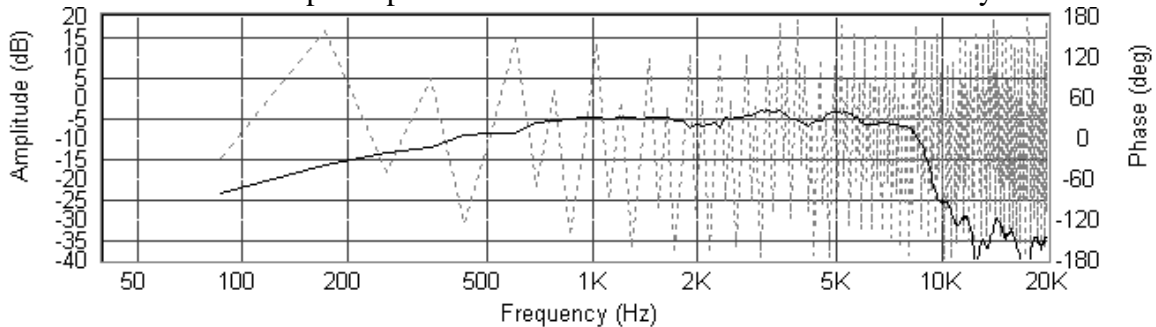

whereas removing excess delay, one can see a much improved phase response:
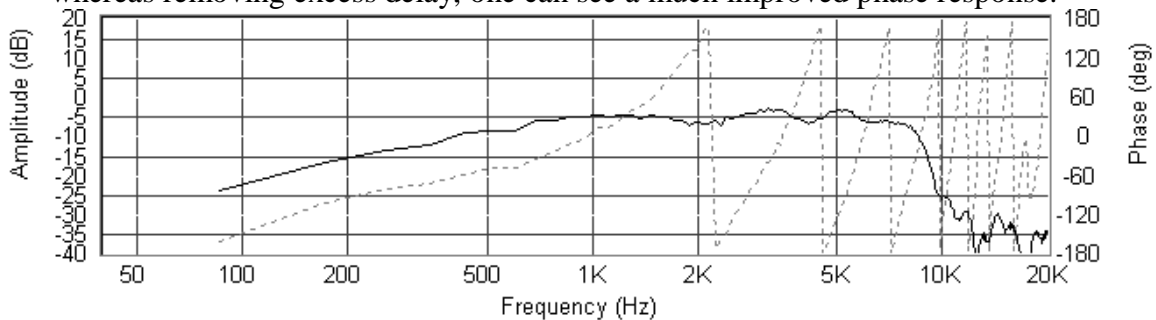
Having the pulse closer to the Impulse will allow for easier manual phase integration though software can often compensate for this through its summation ability utilizing amplitude and phase. Some advocate the importance of phase graphs in order to assure that phase is not rapidly rotating at the point of crossover, which will result in acoustic problems.

As you set your marker further from the Impulse spike and closer to zero there will be better resolution as noted in the table above. The advantage of improved resolution is that it will result in a smoother spliced graph as the nearfield graph will be very smooth. In addition, the improved resolution will make it such that you are less likely to miss dips and peaks (particularly those with a high Q). If phase is not wrapping too much with a marker closer to zero, you get the best of both worlds, resolution and the ability to manually observe a region of good crossover integration.

The most important thing is to keep your marker constant for all measurements in order to compensate for driver offset. One thing that you can try if you have problems obtaining a consistent pulse is to increase the MLS warm up time (Highlight the measurement signal then Options/Preferences/Debug/set the warm up time for a longer period than the 100 sample default).
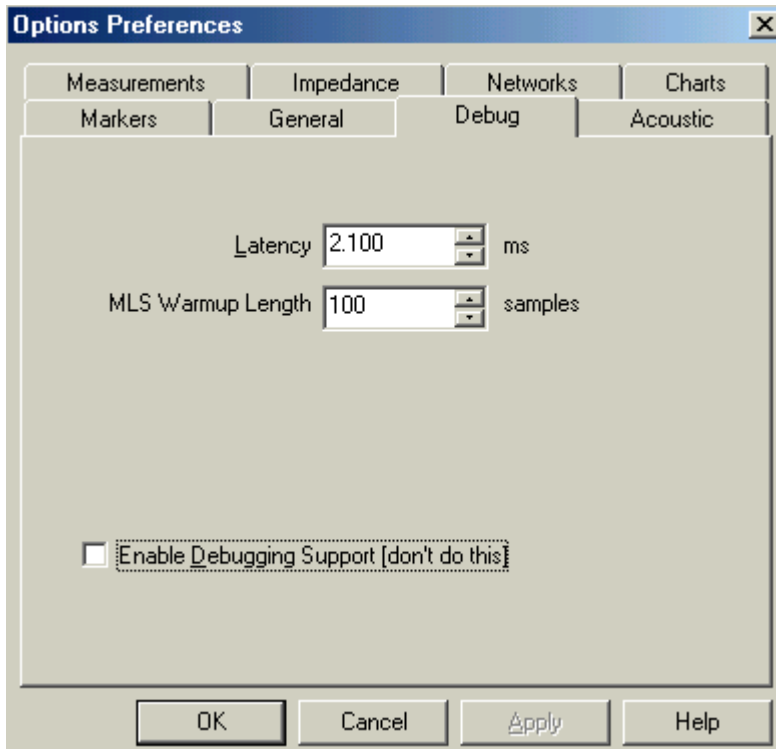
If you have problems obtaining a consistent pulse, Claudio Negro, Bruno Dalle Carbonare, and Valerio Russo have developed a method to use the Minimum phase though you will need to manually enter driver offset then during calculations for crossover design. This is described under Gated Measurements for Crossover Design 12.

One can estimate the amount of Phase shift at a certain distance through the formula: Phase Shift = 360*f*d/c for degrees and 2*PI()*f*d/c for radians where f is frequency, d is distance and c is speed of sound if distance and speed of sound are in the same units.

GATED MEASUREMENTS FOR CROSSOVER DESIGN

SW does not calculate the phase in crossover design from amplitude response (from a Hilbert Transform) like many programs do but rather uses actual measured phase. If you take the following precautions, you will not have to worry about offset as it is taken into account by the measurements:

1. Attempt to place the speaker equidistant from all parallel walls.
2. Put sound absorbent padding between the speaker and the microphone on the floor and pad the microphone stand as well.
3. Maintain the same cables and hardware set up that you used for Nearfield measurements.
4. Measure each driver separately.
5. Measurements are taken On Axis from the listening level (generally the level of the tweeter) unless you are modeling for an off axis response, in which case you would take all measurements off axis but at the same position.
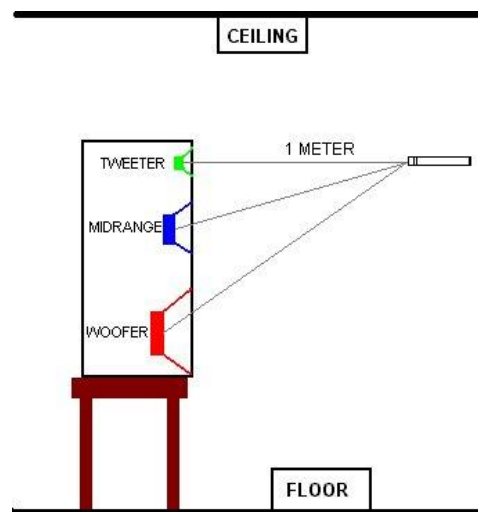
6. Both LF and HF pulse responses must be within the gate (check this by looking at dataset system/measurement.in.r or l depending upon which you use as reference).
   a. Remember that crossover simulations utilize farfield measurements only unless the crossover point is too low for farfield to be accurate. Spliced measurements are for the purpose of observing full scale frequency response of the LP driver or of the entire system.
7. Do not change the gate during the process of observation or measurement.
8. The lay out of the drivers during testing on the baffle is the one you will use in the final box. Testing in the final box is ideal. If you alter the speaker layout after taking measurements, you will have to make corrections in the driver offset.
9. The microphone remains in the exact same place for measurement, preferably the listening position relative to speaker placement.
10. Make sure that the reference pulse is identical for all measurements used for filter design; look for maximum amplitude and the timing with which it occurs. This is an

indication of latency. Aim for the value that occurs most consistently and reject other measurements that do not correspond to this value. Some problems in this regard has been identified.

- a. You can verify the degree of latency as follows
    - i. Connect the Line In and Line Out of the Sound Card with a loop cable
    - ii. Resource/New/Signal
    - iii. Double click on the Signal
    - iv. Measure/Frequency Response
    - v. Verify in the VU meter that you are not clipping
    - vi. Double click Measurement.in.l (in the system folder)
    - vii. Right click on the chart that opens
    - viii. Add/ Select Measurement.in.r (in the system folder)
    - ix. Zoom in on the start of the Impulse to observe that both impulses start at the same time.
    - x. If they do not, repeat this several times
    - xi. If you can get them to the same starting point or if the difference remains consistent, you are okay
- b. Compare both drivers by checking the reference impulse response starting point in the Measurement.in.l for each driver. If the pulse is not at the same position, you can attempt to repeat this measurement until you get a measurement with a pulse at the same point and then use that data.

11. If your pulse position is consistent, you can accurately measure phase from the same point of reference which allows you to use the On Axis Measurement phase to calculate driver offset; it is advisable to allow Speaker Workshop to use measured phase rather than attempting to calculate the driver offset and use this function or the remove excess delay as it will be more accurate.
12. If you do use the Acoustic Offset features, note that negative offset means into the baffle; positive is closer to the listener. (This is the opposite of the FRD consortium programs designations!) If you have problems obtaining a consistent pulse position, Claudio Negro along with Bruno Dalle Carbonare and Valerio Russo have developed a procedure to calculate the Minimum Phase for each driver, which will overcome the inconsistent results of the impulse-phase response. This approach is only necessary if the pulse position changes from measurement to measurement, a problem noted by some with Speaker Workshop. In using this approach, you would need to accommodate for driver offset as it is working with minimum phase calculated at the driver's acoustic center.

Note: Acoustic Center is a theoretical consideration. It is defined as the position of the point from which it appears that all spherical wave fronts diverge. It's location will vary depending upon the method you use to locate it but can generally be isolated to a point within one particular approach (using phase techniques, distance [uses the inverse square rule along with amplitude measurements], or reciprocity methods [compares transfer functions between two microphones as transducer and receiver]). An excellent discussion of this can be found in "A Note on the Concept of Acoustic Center" by Finn Jacobsen, et. al. published in J Acoust. Soc. Am. 115 (4), April 2004.

a. Remember that for Driver offset, a negative value means that the acoustic center is recessed into the baffle and a positive value means that it is closer to the listener (sticking out from the baffle). This is accomplished by converting the SW file into a logarithmic scale using

"Converte file SW in file Logaritmici.exe"

the following program and then exporting the response from SW into FRC (Frequency Response Combiner freely available for download at http://www.pvconsultants.com/audio/utility/frc.htm).

   1. Open Logarithmic Scale converter
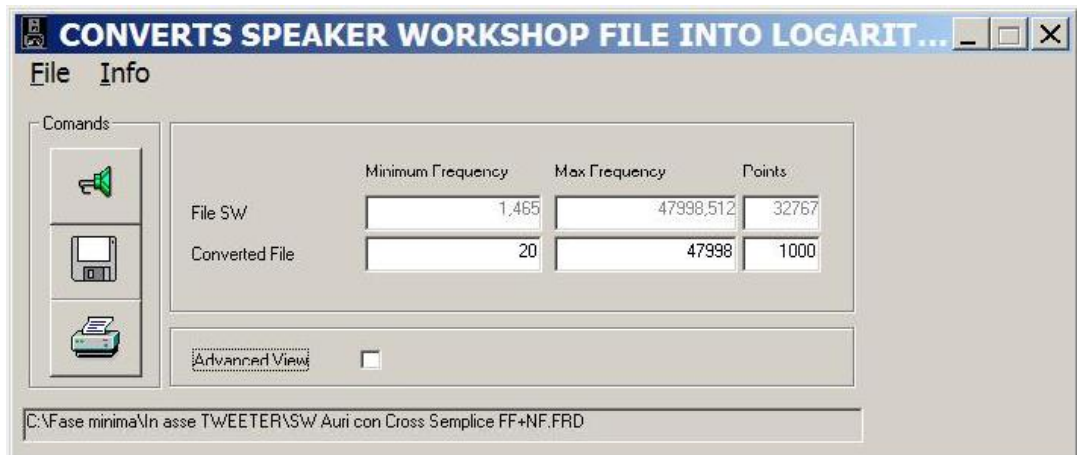
   2. Click on
   3. Read the imported file limit
   4. Enter the minimum and maximum frequency desired. Choose a frequency band where the measurements are without noise.
   5. Enter the number of resolution points (around 1000 is reasonable).
   6. Click and the logarithmic converted file is saved in the same directory where the original file was.
   7. Open Frequency Response Combiner
   8. Click Execute to load the logarithmic file. It will also load the file frequency limits and the number of points in this column.

a. Select 1Drv Base e Precise Phase and Group Delay
b. Click on Extract Minimum Phase which will take a few seconds to calculate depending upon the speed of your computer (it does this through the Hilbert Transform)
c. Then click on the Load button just above the Execute button and it will change to save.
d. Now click on Execute just below and it will ask you where you want to save the file
e. Use as complete a response as possible for the drivers in order to minimize phase errors (See David L. Ralph's papers, "Hilbert Transform Generated Minimum Phase & Errors" and "Driver Model Accuracy and its Impact on Phase".
   i. For the woofer, you must use the combined response of nearfield plus farfield
   ii. For the tweeter, attempt to extend well to a frequency of over 20kHz, if possible
   iii. If you have a 48 kHz soundcard, you can only sample up to 24kHz. Use the Normalizer function to extend the response in both the low and in the
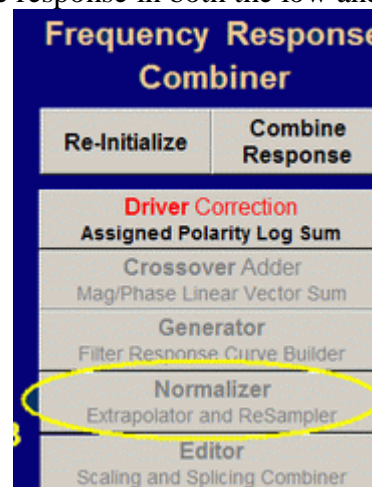

high region.

12. You do not have to repeat any of the nearfield measurements if you have not adjusted any settings since making the nearfield measurements.
13. Run an MLS test signal in SW to assure that the Volume in the software is set to record at your maximum non-clipping level (usually 20-25K).
14. Assure that your sample rate and sample size of at optimum for your sound card as tested before in the SW Set Up Section.
15. Option/Preference/Repeat Count and verify that the repeat count is set to at least 5 (the higher the number, the longer the test will take but the more accuracy you will achieve).
16. Check the "Use Pre-emphasis" only if you did an amplifier calibration (Note: Pre-emphasis acts as a low pass filter system which increases the signal to noise ratio for lower frequency tests but it may cause a sudden roll off of high frequency response if selected. If the amplifier calibration is not used, this should not be checked. Even if amplifier calibration is used, pre-emphasis does not need to be checked as it will make the sound more tolerable during testing however it will decrease the accuracy of results).
17. The microphone pre-amplifier settings will need to be set higher to compensate for the increased distance from the sound source.
18. If you don't see 2 vertical red lines (markers)
19. Click Options/Preferences/Markers.
20. Under Time click Visible.
21. Click on Markers to make sure that all three choices are visible.
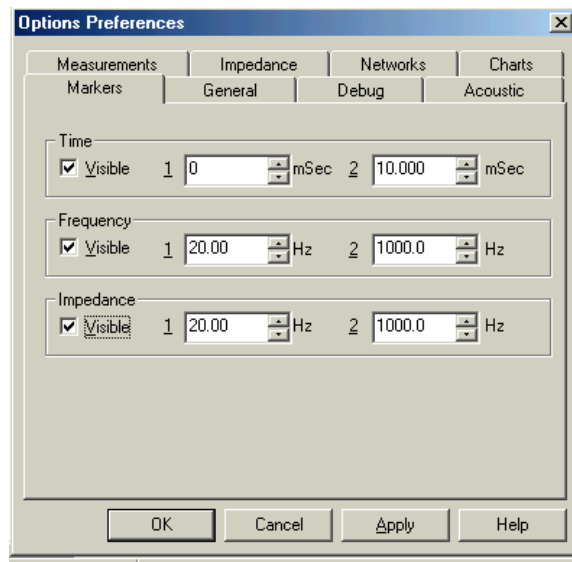
22. Double click on   /Measure/Frequency Response/On Axis
23. Assure that values for the Right and Left Volume Meters in SW are almost identical and in the ideal range as determined previously (usually 20-25K).
24. A new file will be created: SpeakerA.On Axis.
25. Open chart properties to hide the phase as it is not useful here without a common reference point.
26. Check the Y-axis Min/Max.

27. Set Y axis Major grid lines to 6 dB.
28. If the data needs to be smoothed, select the window then Transform/Smooth/ 1/3 octave/OK.
29. Once the volume settings are right, click Measure/Pulse Response.
30. Repeat the test (adjusting volume if necessary) until you observe data that makes sense.
31. A new file will be created: SpeakerA.Impulse (this file is not yet useful as the markers have not yet been set).
32. Double click on SpeakerA.Impulse.
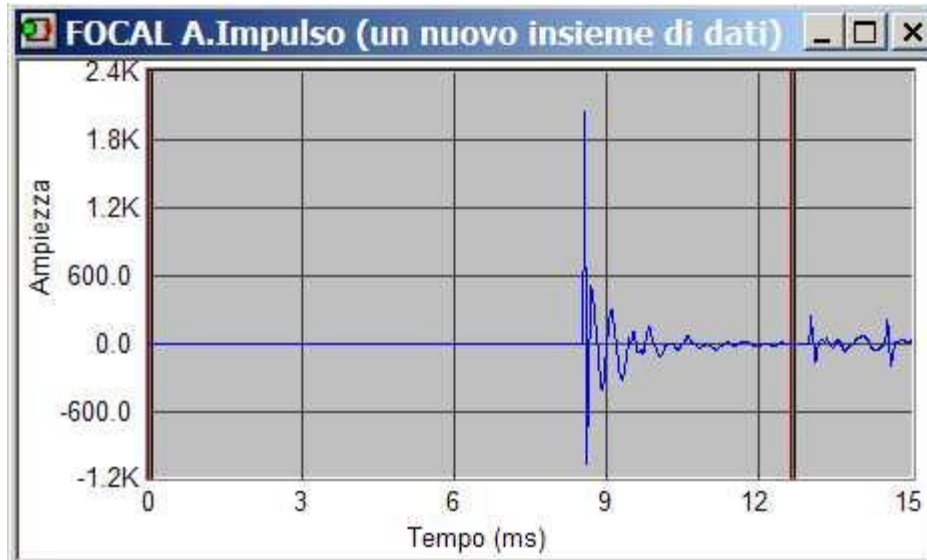
33. The graph demonstrates pulse points (large spikes).
34. The first impulse is the signal and the second impulse is the first reflection. The distance is approximately 1/3 of the duration of time between the first pulse and the second pulse in meters.
35. The markers are moved by pointing to the target marker with the mouse, left clicking and holding down the button, and moving it to the desired location. The zoom option may aid in visualizing the chart.
36. Each time that you do a pulsed measurement, check the Measurement.in.l file (in the system folder) to make certain that the impulse remains at the same reference point and that it's amplitude is the same. If this is inconsistent, measurements will not be useful.
    a. Set the first marker based upon your preference (after reviewing the discussion under the section entitled Farfield_Gated_Measurements) and the second marker just prior to the second impulse (which represents the first reflection). As a review, there has been some discussion regarding the setting of the first marker. Many advise setting it just prior to the impulse to decrease some of the phase wrapping that occurs if it is set too far from the impulse. Setting the marker closer to the impulse makes the phase somewhat easier to work with. On the other hand, Speaker Workshop sets

the number of data points based upon the duration of the window and the sampling rate as in the following table:

sampling rate * 1000	data points	16	32	64	128	256	512	1024	2048
11	from (msec)	1.45	2.91	5.82	11.64	23.27	46.55	93.09	186.18
	to (msec)	2.90	5.81	11.63	23.26	46.54	93.08	186.17	372.35
12	from (msec)	1.33	2.67	5.33	10.67	21.33	42.67	85.33	170.67
	to (msec)	2.66	5.32	10.66	21.32	42.66	85.32	170.66	341.32
22.1	from (msec)	0.72	1.45	2.90	5.79	11.58	23.17	46.33	92.67
	to (msec)	1.44	2.89	5.78	11.57	23.16	46.32	92.66	185.33
24	from (msec)	0.67	1.33	2.67	5.33	10.67	21.33	42.67	85.33
	to (msec)	1.32	2.66	5.32	10.66	21.32	42.66	85.32	170.66
44.1	from (msec)	0.36	0.73	1.45	2.90	5.80	11.61	23.22	46.44
	to (msec)	0.72	1.44	2.89	5.79	11.60	23.21	46.43	92.87
48	from (msec)	0.33	0.67	1.33	2.67	5.33	10.67	21.33	42.67
	to (msec)	0.66	1.32	2.66	5.32	10.66	21.32	42.66	85.32
96	from (msec)	0.17	0.33	0.67	1.33	2.67	5.33	10.67	21.33
	to (msec)	0.32	0.66	1.32	2.66	5.32	10.66	21.32	42.66

    b. : You can calculate the gate range for the number of data points by 1000*number of data points/sampling rate = lower end of gate in milliseconds. It goes up in a step wise progression to establish the range. Increasing the time gate by moving the marker closer to zero will increase resolution. You can start by setting the marker just prior to the Impulse; if the resolution is poor you can gradually move the marker further away (towards zero) as long as it is kept at the same place for all drivers.

37. See the spreadsheet for calculating expected floor bounce impulse appearance based upon the microphone distance, the expected impulse length and the useful lower frequency as a result. The further from reflecting surfaces you are, the longer the window of test you will have. Sound absorbent matting will decrease the impact of the bounce signals. A longer test window allows for more resolution as SW allows for a resolution of 128 points(with a 48K sampling rate) up to a time slice of 9 milliseconds and then it doubles with each doubling of the time window. By increasing the window to just over 9 milliseconds, you significantly enhance resolution.
    a. To check the number of data points: Right Click the Response Chart/Properties/Specifics.
    b. You can change this number and the graph will look slightly better for crossover modeling, but once you repeat a measurement SW will reset this number according to its algorithm.
38. Repeat the On Axis Measurement with the markers in place:
39. Double click /Measure/Frequency Response/On Axis.
40. Check SW to assure that the Left and Right Volumes are approximately equal and at the optimum volume range (usually 20-25K).

41. Open .On Axis.
42. You can smooth your on axis measurements (usually with 1/10 or 1/12 octave smoothing as seen in magazines) but never save them as the file you will use for crossover design as it will interfere with the phase computations.
43. Each time the microphone to baffle distance changes, pulse measurements must be re-done and you must re-set the markers.
44. The lowest usable frequency may be read from the spreadsheet noted above or approximated by 1000/window gate duration(ms) in hertz so if the window gate is 5 ms, the lowest usable frequency would be 200 Hz.

Alternatively:

1. Place the microphone at a distance of 1 meter on axis from the tweeter **_or_** at listening height.
    a. Note that one meter is a standard used in the industry however John Atkinson asserts in Stereophile Magazine (January, 1999) that a true far field measurement should be taken at a distance of no less than that of the smallest baffle dimension and preferably twice this dimension to allow for full integration of the signals from the drivers.
2. Connect all drivers as they will ultimately be connected, either in series or in parallel. (to this point we have only measured individual drivers).
    a. Remember that a series connection would be + to – to + to – while a parallel connection will be + to + and – to -.
3. Since the tweeter is not yet protected by the crossover, it is best to start with a low volume
4. Double click ![icon] /Measure/Frequency Response/Gated.
5. Assure that the right and left SW volumes are approximately equal.
6. A new file appears called **SpeakerA.Gate** but since all drivers were combined, rename this **SpeakerA.Gate Combined**.
7. Now hook up only the woofer but leave all other settings and placements as they were.
8. Double click ![icon] /Measure/Frequency Response/Gated.
9. Assure that the right and left SW volumes are approximately equal.
10. Rename this file **SpeakerA.Gate Woofer.**
11. Now hook up only the tweeter but leave all other settings and placements as they were.
12. Double click ![icon] /Measure/Frequency Response/Gated.
13. Assure that the right and left SW volumes are approximately equal.
14. Rename this file **SpeakerA.Gate Tweeter.**
15. You will now have one more file than the number of drivers that you are measuring for this speaker.
16. You can repeat these measurements with the crossover in place to evaluate how the crossover impacts each driver however unless the setup is exactly the same, there will be some differences in measurement due to the different set up.
17. Evaluate the charts. By looking at the overlap in frequency where the response is relatively flat, one can narrow down a range for choosing a crossover frequency. Look carefully at the off axis comparison, however, as this will prevent drop out in the off axis response. When looking at a normalized off axis measurement of polar response (on axis measurement subtracted from each of the off axis curves), look at the region around the crossover. A dip prior to and a peak subsequent to the crossover point would suggest a rapid transition from the woofer to the tweeter. The reason that this is suggested is that as the frequency increases such that the wavelength is closer to that of the driver diameter, you lose off axis dispersion. The woofer's loss of off axis dispersion will become evident if the

crossover point is too high. Naturally, one would target a flat off axis response. The further off axis you can evaluate, the less likelihood of off axis drop out.
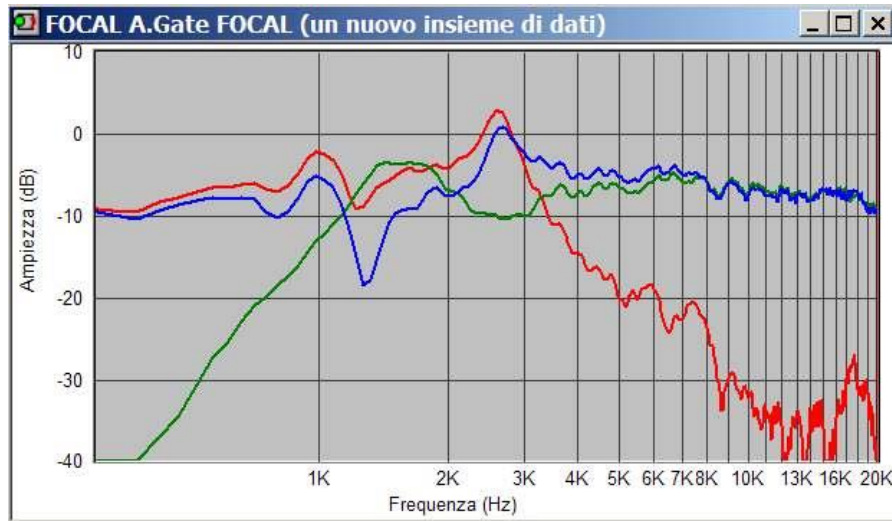
On Axis with the Acoustic Center:

Tweeter   **Woofer**   **Both**

18. The chart below indicates that neither the woofer nor the tweeter phases are rotating between approximately 1.8 and 3.8 KHz but they are quite out of phase during this time. This is something to take into account during the crossover modeling. Choosing a crossover point where none of the drivers are rotating phase will aid in phase coherence.
19. If you choose a crossover point that is too low, you will have more tweeter IM distortion at mid to high listening levels (raspy sound); if you choose a crossover point that is too high, the sound will be more forward and aggressive at moderate listening levels and will break up at high listening levels.
20. Lower crossover points will decrease the acoustic offset issue at the crossover as it will be a smaller percentage of the wavelength with less phase distortion.
21. You don't need drivers to be in exact phase to achieve a flat response. Sometimes the offset will be beneficial, sometimes overlapping slopes or asymmetrical slopes will be of assistance in lining up the response.

# SPLICING THE RESULTS INTO A SINGLE GRAPH

This is done in order to display frequency response of the LP driver or of the entire Network. It is not the graph that should be used for Crossover determinations; use only the farfield graphs. Nearfield measurements will not have the same phase characteristics as the Farfield measurements. Remember that mathematically, if you provide the same input voltage, you would just add 20*LOG((4*Far field measurement distance)/(Effective Diameter/2) in dB to the Far Field Graph. Doing this visually is probably accurate enough for our purposes.

Splicing the Nearfield and Gated Farfield results will allow for a continuous graph of the acoustical response. The results will be much more smooth if you don't smooth the curves prior to splicing. If you do, you will likely see a stepped response in the lower frequencies due to less resolution.

1. Double click on the Nearfield Measurement to open it.
2. Open Chart Properties.
3. Click on Add Gated Farfield Response (On Axis with Acoustic Center) measurements with all drivers working.


4. Evaluate the graph for a place between the usable farfield lower limit and the usable nearfield upper limit (as calculated before in those sections) for a reasonably flat section on both graphs where the splicing can occur.
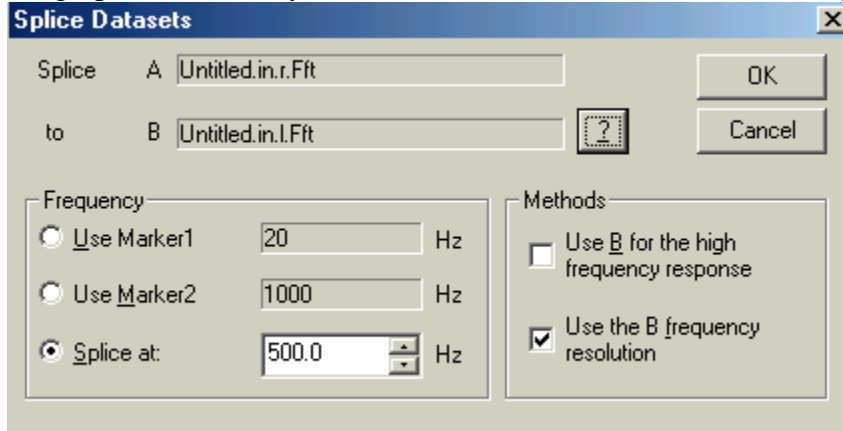5. Click Transform/Scale.
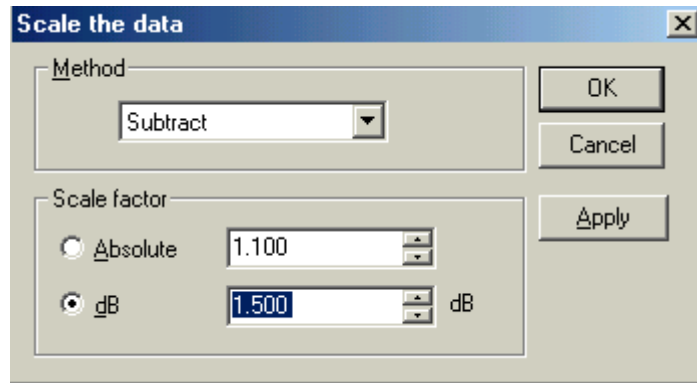
6. Use the Zoom option on the graph to make it easier to see the area of interest.
7. Since Nearfield has a higher amplitude due to proximity of the measuring microphone, estimate the dB separation
    a. Choose Subtract and fill in the estimated amount.
8. Click Apply.
9. Keep repeating this procedure until it appears that the graph is likely to be continuous.
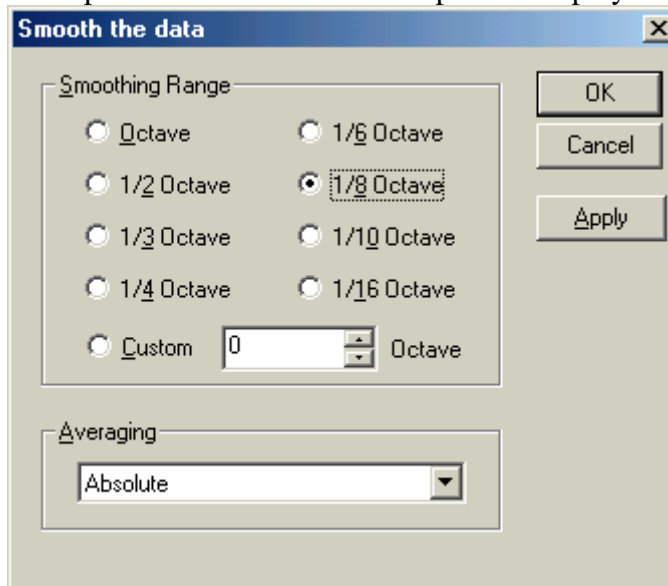10. Click OK.
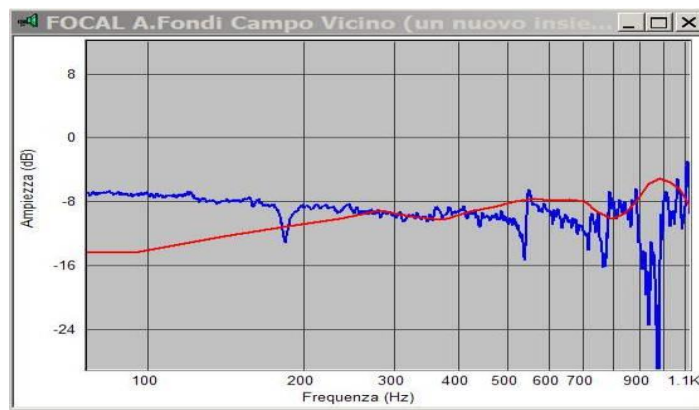
11. Eliminate the Farfield response from the Nearfield chart.
12. Close the Nearfield chart.
13. Open the Gated Farfield Response SpeakerA.Gated Combined.
14. Click Calculate/Splice.
15. The A is the source dataset (the dataset currently being viewed) and is not editable.
16. The B is the near-field response, click on and select the file.
17. Select the frequency at which the splice takes place.

18. Under Method, choose the graph to be used for the higher frequency portion of the graph. Check B if you are to use this; leave this unchecked if you are to use A.

Splice Datasets

Splice	A	Untitled.in.r.Fft
to	B	Untitled.in.l.Fft

OK
Cancel
[?]

Frequency
- Use Marker1 — 20 Hz
- Use Marker2 — 1000 Hz
- Splice at: 500.0 Hz

Methods
- ☐ Use B for the high frequency response
- ☑ Use the B frequency resolution

19. Select OK to perform the splice.
20. A new file called SpeakerA.Gate Combined. Splice is displayed.

Smooth the data

Smoothing Range
- ○ Octave   ○ 1/6 Octave
- ○ 1/2 Octave   ● 1/8 Octave
- ○ 1/3 Octave   ○ 1/10 Octave
- ○ 1/4 Octave   ○ 1/16 Octave
- ○ Custom 0 Octave

OK
Cancel
Apply

Averaging
Absolute

21. Click Transform/Smooth to smooth the response. Generally 1/8 Octave is adequate so check this box and evaluate by Clicking Apply.
22. You can repeat this step until you are satisfied. Then Click OK to make the changes permanent.

MEASURING INDIVIDUAL DRIVERS IN THE ENCLOSURE

1. Attempt to place the speaker equidistant from all parallel walls.
2. Put sound absorbent padding between the speaker and the microphone on the floor and pad the microphone stand as well.
3. Maintain the same cables and hardware set up that you used for Nearfield measurements.
4. Measure each driver separately.
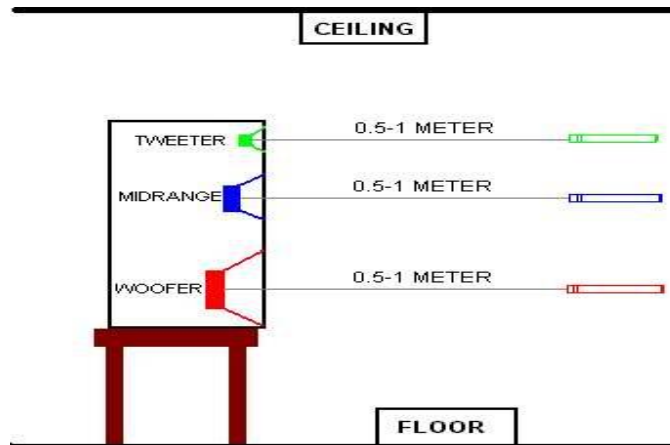
5. Each driver will be measured with the microphone in line with the driver (being measured) center
6. All drivers must be measured at the same distance measured from the front baffle. Because the microphone is not consistently set at the same place for these two measurements and because the gating markers are likely to be different, phase measurements cannot be used from this type of measurement.

Procedures 7-10 were done in the Nearfield set up, they do not have to be repeated if you have not adjusted any settings since making the nearfield measurements:

7. Run an MLS test signal in SW to assure that the Volume in the software is set to record at your maximum non-clipping level (usually 20-25K).
8. Assure that your sample rate and sample size of at optimum for your sound card as tested before in the SW Set Up Section.
9. Option/Preference/Repeat Count and verify that the repeat count is set to at least 5 (the higher the number, the longer the test will take but the more accuracy you will achieve).
10. Check the "Use Pre-emphasis" only if you did an amplifier calibration (Note: Pre-emphasis acts as a low pass filter system which increases the signal to noise ratio for lower frequency tests but it may cause a sudden roll off of high frequency response if selected. If the amplifier calibration is not used, this should not be checked. Even if amplifier calibration is used, pre-emphasis does not need to be checked as it will make the sound more tolerable during testing however it will decrease the accuracy of results).

11. The microphone pre-amplifier settings will need to be set higher to compensate for the increased distance from the sound source.
12. If you don't see 2 vertical red lines (markers)
    a. Click Options/Preferences/Markers.
    b. Under Time click Visible.
13. Click on Markers to make sure that all three choices are visible.
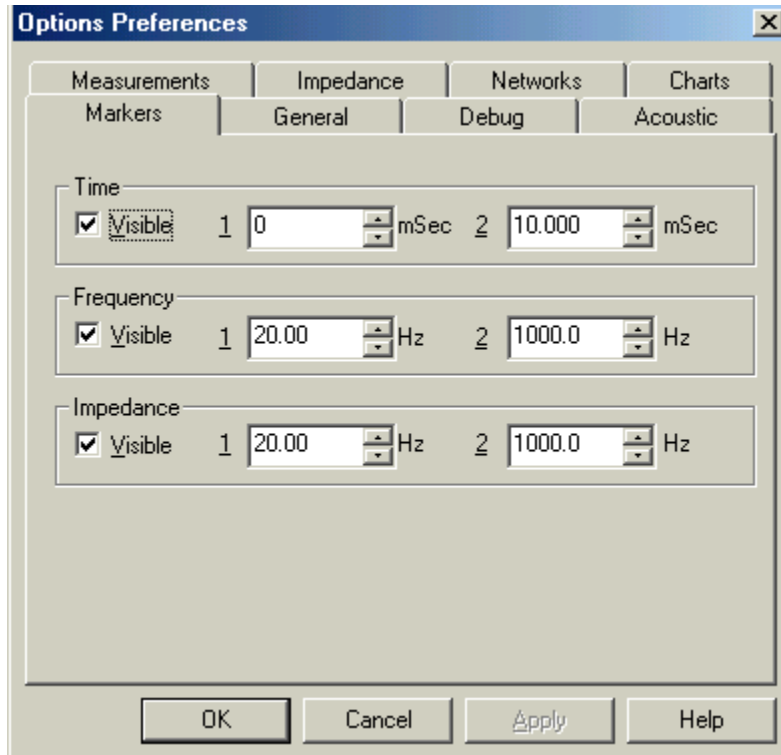
14. Double click on ◀ /Measure/Frequency Response/On Axis
15. Assure that values for the Right and Left Volume Meters in SW are almost identical and in the ideal range as determined previously (usually 20-25K).
16. A new file will be created: SpeakerA.On Axis
17. Open chart properties to hide the phase as it is not useful here without a common reference point.
18. Check the Y-axis Min/Max.
19. Set Y axis Major grid lines to 6 dB.
20. **If the data needs to be smoothed**, select the window then Transform/Smooth/ 1/3 octave/OK.
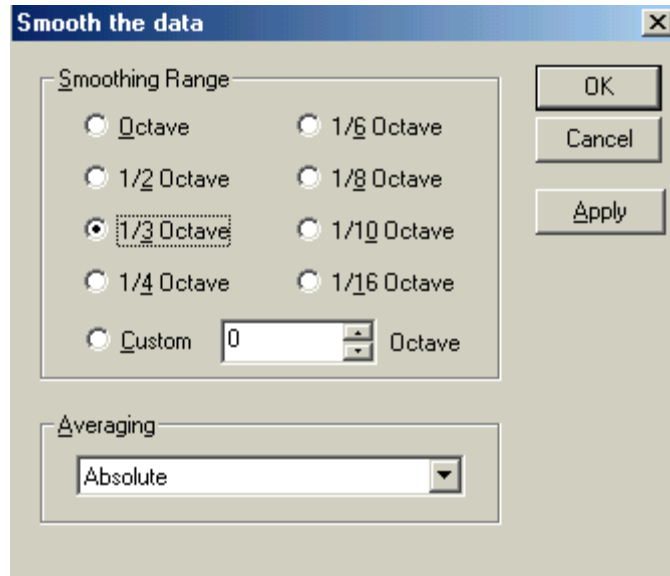
21. Once the volume settings are right, click Measure/Pulse Response
22. Repeat the test (adjusting volume if necessary) until you observe data that makes sense.
23. A new file will be created: SpeakerA.Impulse (this file is not yet useful as the markers have not yet been set).
24. Double click on SpeakerA.Impulse
25. The graph demonstrates pulse points (large spikes)
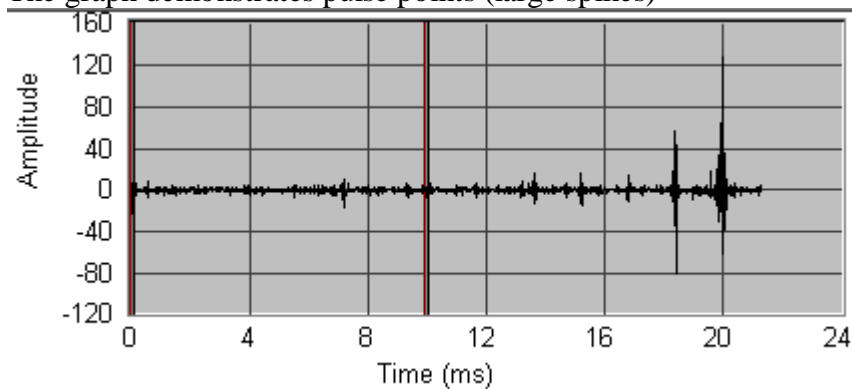
   a. Note the first major pulse here at 18.4 msec and the second at 19.9 msec. The first impulse is the signal and the second impulse is the first reflection. The distance is approximately 1/3 of the duration of time between the first pulse and the second pulse in meters (so here, it would be about 1.5 meters or about 4.9 ft).
26. The markers are moved by pointing to the target marker with the mouse, left clicking and holding down the button, and moving it to the desired location. The zoom option may aid in visualizing the chart.
27. Set the first marker at zero and the second marker just prior to the second impulse (note that this method is providing the best resolution with the potential for more phase wrapping).
28. *See the spreadsheet (originally written by Claudio Negro with a slight modification by me) for calculating expected floor bounce impulse appearance*

based upon the microphone distance, the expected impulse length and the useful lower frequency as a result (see [Floor Bounce Calculator](#)).   The further from reflecting surfaces you are, the longer the window of test you will have. Sound absorbent matting will decrease the impact of the bounce signals. A longer test window allows for more resolution as SW allows for a resolution of 128 points(with a 48K sampling rate) up to a time slice of 9 milliseconds and then it doubles with each doubling of the time window. By increasing the window to just over 9 milliseconds, you significantly enhance resolution.

      a. To check the number of data points: Right Click the Response Chart/Properties/Specifics.

      b.   You can change this number and the graph will look slightly better for crossover modeling, but once you repeat a measurement SW will reset this number according to its algorithm.

29. Repeat the On Axis Measurement with the markers in place:

      a. Double click /Measure/Frequency Response/On Axis

      b. Check SW to assure that the Left and Right Volumes are approximately equal and at the optimum volume range (usually 20-25K).

      c. Open **.On Axis**

30. You can smooth your on axis measurements (usually with 1/3 octave smoothing) but never save them as the file you will use for crossover design as it will interfere with the phase computations.

31. Each time the microphone to baffle distance changes, pulse measurements must be re-done and you must re-set the markers.

32. The lowest usable frequency may be read from the spreadsheet noted above or approximated by 1000/window gate duration(ms) in hertz so if the window gate is 5 ms, the lowest usable frequency would be 200 Hz.

MEASURING ACOUSTIC OFFSET

*Note that this is generally not necessary if you did the acoustic measurements from the same point of reference as offset is incorporated into the data. Though using the Acoustic Offset command may "clean up" the phase of the individual driver, it will alter the relative phase measurements between drivers which is the more important aspect of phase measurement.*

Only use offset values under two conditions
➢ *If you have taken the driver measurements with a constant set up on the final baffle and later decide to recess a driver. Note that you will be ignoring diffraction effect changes.*
➢ *If you are importing Minimum Phase Data from another application (.frd files). Minimum Phase Data is derived from the Hilbert-Bode Transform Method and is more portable as SW is Speaker Specific and it will wrap around a few times since it is not Minimum Phase Measurement (but rather true phase). To use this data in SW, you would have to enter the Minimum Phase Measurements of each driver along with the relative offset that will exist in your design. Negative offset is into the cabinet.*

1. Evaluate the charts
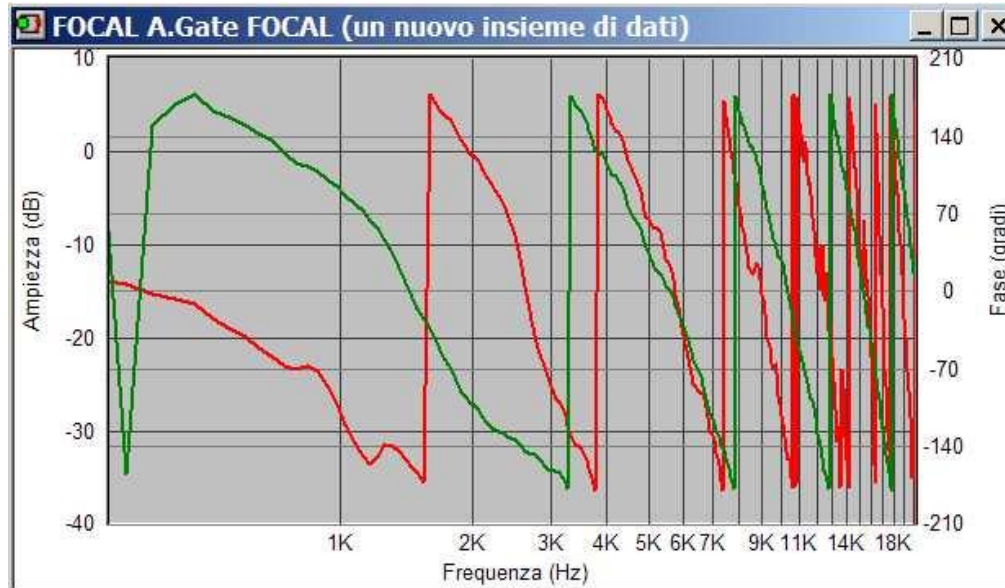
2. Obtain the actual phase by adding 360 Degrees for each phase revolution to the reading
3. Delay time (sec)=phase difference(between the drivers)/(360*frequency)
4. Path length difference(millimeters)=delay time*344,000

    OR

    Path length difference(inches)=delay time*13543.3

5. Note that acoustic centers vary based upon the frequency being produced; the target is to attempt to align the acoustic centers either physically or electronically at the crossover point.

The following spreadsheet (from www.diyspeakers.net/Projects/Tools/offset.xls) written by Carlos Lorenzo may be of benefit if you wish to calculate offset:

"driver offset
spreadsheet.xls"

In addition, phase will change as you move away from the minimum phase point. Here is a spreadsheet to calculate how the phase will change with distance away from this point:

"phase shift
calculator.xls"

OFF-AXIS MEASUREMENTS

Leave the markers where they were and do not change the volume or the position of the microphone.

1. Turn the speaker at a 30º angle to the microphone.
2. Double click /Measure/Frequency Response/30 Degree.
3. A new file will be created called **Speaker A.f30**.
4. Repeat steps 1-4 but instead turn the speaker to a 60º angle and the new file created will be **Speaker A.f60.**
5. *Optional: If you want to, you can do other angles such as a 45º angle:*
   a. *Turn the speaker at a 45º angle to the microphone.*
   b. *Double click* */Measure/Frequency Response/30 Degree*
   c. *Rename the file* **Speaker A.f45**
   d. *Open the* **SpeakerA.On Axis** *file and add to the chart files for the additional off axis gated measurements.*

EXCESS DELAY

If you need to modify signals to remove excess delay: Transform/Delay/Remove Excess Delay. This portion of Speaker Workshop can be used if you have taken measurements from varying reference points or if you feel that you cannot take all measurements from exactly the same place. It is not recommended to use this as if you take all measurements from a reference point, it will already be adjusted for in SW. Though using the Delay command may "clean up" the phase of the individual driver, it will alter the relative phase measurements between drivers which is the more important aspect of phase measurement.

To use this:
1. Uncheck "Remove Excess Delay"
2. You can stop here if it is unimportant to you to observe the *individual driver* phase response.
3. Enter the Delay time to just before the first pulse measurement to start your gate.
4. If you work with Delay times, you must make certain to enter the exact delay times for all drivers or else your crossover phase calculations will be incorrect.