

## **SECTION 7 – SYSTEM TESTING**

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## **SECTION 7 – SYSTEM TESTING**

*[\(Section Index\)](#)*

### **TESTING A COMPLETE SPEAKER**

1. Resource/New/Driver
2. Set up as if far field acoustic testing
3. Data will be usable from the lowest usable far field frequency
4. You could do a nearfield measurement of the woofer or use that which was already measured and splice it to the far field for a complete system frequency response graph.
5. Both Frequency Response and Impedance data can be gathered of the complete system but be careful to place it into a new folder so as not to corrupt the original data.




## **NON GATED FAR FIELD MEASUREMENTS**

### *(Section Index)*

This is used to measure the performance of the driver or system in a room including reflections and environmental influence. Acoustic measurements might show peaks where the wavelength of the frequency is equal to the distance from the driver center to the edge of the box (likely tweeter or possibly midrange). If you see this, it suggests that additional baffle compensation work needs to be done. Baffle step losses in a 4pi space would lead to a gradual 6 dB drop in the bass frequency response centered at  $f_3 \sim 4560/W_b$  where  $W_b$  is the width of the front baffle in inches and  $f_3$  is calculated in Hz.

It has been suggested by John Dunlavy (multiple postings on rec.audio.highend) that Distortion Testing as well as Phase Response, Impulse Responses (including Waterfall, Step Response, and Energy Time Curves) be measured at a distance of 10 feet or from the listening position to obtain a more accurate account of how this will impact listening.

It can also be used to assist with speaker placement to maximize the response in a room. One could also use a Real Time Analyzer such as True RTA or Spectra RTA however with Speaker Workshop, this is unnecessary. Speaker Workshop can also use a Warble Tone (Signal/Properties), which is a tone that sweeps forwards and backwards through a range repeatedly. Warble tones permit you to assess low frequency response with reduced interaction from standing waves as compared to sine waves, which makes microphone placement for informal measurements taken under non-anechoic conditions much less critical.

1. Place both speakers into the room at the positions they will be in for listening.
2. Place the measuring microphone at ear level at the listening position.
3. Use either the monaural switch (mono) of the amplifier (if present) or use a Y cable to connect the sound card output to both the amplifier left and right inputs.
4. All other connections will be as they were for the gated far field set up.
5. 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).
6. Assure that your sample rate and sample size of at optimum for your sound card as tested before in the SW Set Up Section.
7. 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).
8. Check the "Use Pre-emphasis" only if you did an amplifier calibration.
9. Double click on  /Measure/Frequency Response/Far Field
10. 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).
11. A new file will be created:  .Farfield
12. Open  .Farfield and evaluate the graph.
13. If the graph appears to be noisy: Transform/Smooth/ 1/3 Octave/ Apply and review the result. OK will make this change permanent while Cancel will

eliminate the smoothing. You can also right click the chart and use the “undo smooth data” to return to the original chart.

*Note: During farfield measurements, the start gate is active and sets the beginning of the measurement. This should be set close to zero if you want to include the entire room response or you can shift it forward to eliminate as much of the pulse as you want in order to look at only the reflections and room cancellations and reinforcements.*

## NETWORK TRANSFER FUNCTION

*(Section Index)*

The Transfer function of a loudspeaker is generally defined at low amplitudes (Small Signal Domain). For Small signals, the loudspeaker displays a near linear system response. As the amount of signal is increased (Large Signal Domain), Thiele Small parameters are less descriptive of the system and it demonstrates a non-linear response.

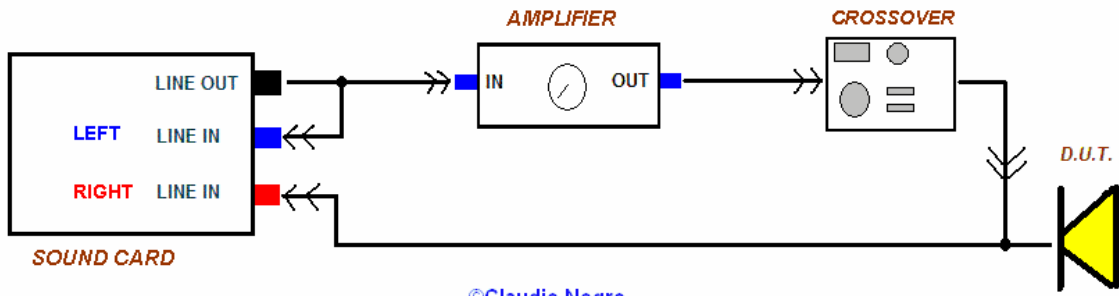
1. Make a copy of the Calculated Frequency Response plot for the network
2. Rename it to a recognizable name.
3. Select the calculated response for the network of interest.
4. Select CALCULATE/COMBINE
5. In the Pop-Up box, select DIVIDE
6. With [?] choose the Frequency Response data set for the driver, which is used in the network.
7. Select OK!
8. The frequency response plot will now be the network transfer function for network of interest.

*Note: You may have to re-scale the SPL scale so that the response is visible, but it is really easy and fast.*

Another method of measuring this is described by Claudio Negro as developed by Guglielmo Caioli and studied by Valerio Russo. The set up is as if you are doing a frequency response measurement. The amplifier output is connected to the filter input and then the filter output is connected to both the driver and to the **Right** Line In of the soundcard (this is acting as the data input). The **Left** Line In can be connected to either the Amplifier output or to the Sound Card Output.

Of the two methods:

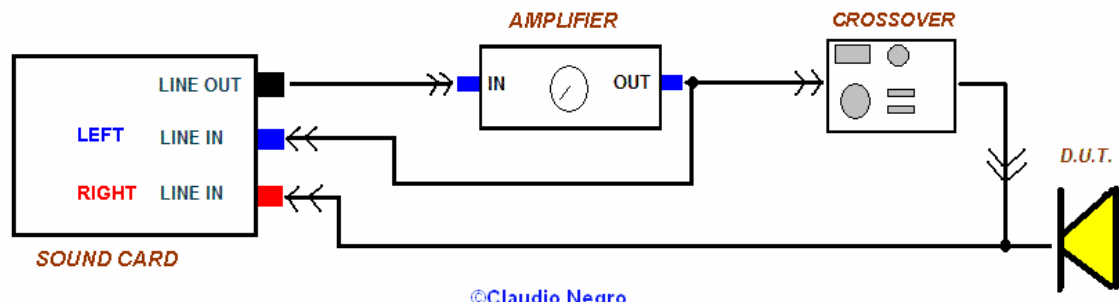
- A. If you choose to use the Amplifier Output approach, you have to ***be careful not to allow the amplifier to send over 500mVolts output to the sound card*** lest you potentially burn out your card. The advantage to this method is that Speaker Workshop measures the voltage difference between the Data and the Reference Channel and since this is bypassing the amplifier gain, it will not be taken into account leaving only the Network Transfer Function and zero dB of attenuation where the filter is not active. Thus, if you sum the Network Transfer Function with the Driver Unfiltered Response, this would be equal to the filtered driver response.
- B. The limitation on using the Line Out as your reference is that the phase is not likely to be as accurate at the extremes of the measurement frequencies.



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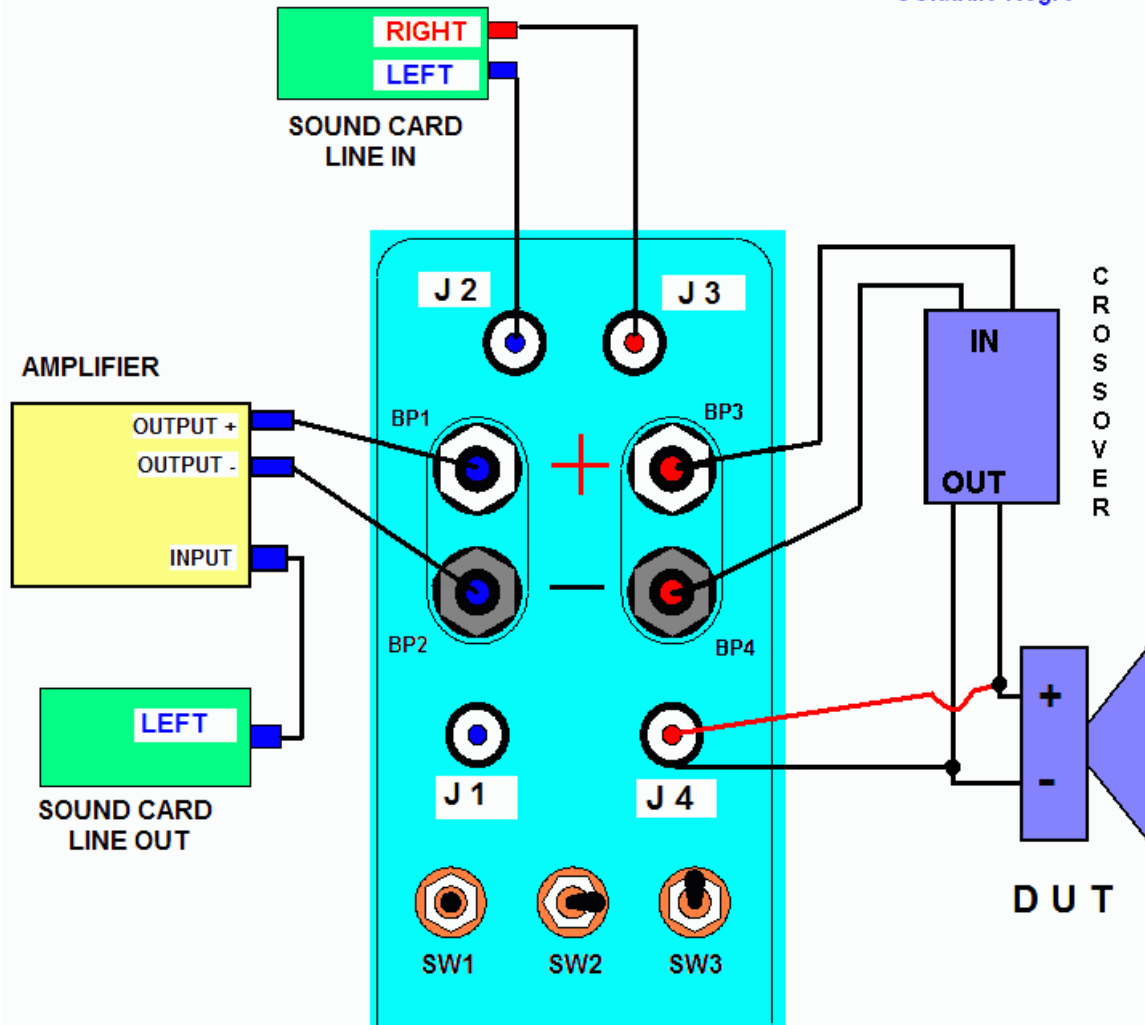
Cables using LINE OUT as reference

or




©Claudio Negro

Cables using Amplifier OUT as reference



Once the set up is in place:

1. Set the amplifier volume to it's minimum
2. Connect a voltmeter to the output
3. Resource/New/Signal and create a new signal
4. Double click on the signal
5. Measure/ Frequency Response
6. Increase the amplifier volume and repeat readings on the voltmeter until you get a reading of 300-500mV (AC)
7. Once the volume level is okay, repeat the measurement which measure the network (this is a measure of the filter only if you are using the amplifier output reference approach)
8. Open the newly created file to see the Network Transfer Function
9. If you have used the amplifier output as a reference, you can sum the Network Transfer Function and the Unfiltered Driver Response to get the filtered driver response
  - a. Open the Network Transfer Function Chart
  - b. Click Calculate/ Combine

- c. A window will appear
- d. Choose Multiply
- e. Click  and select the unfiltered measured driver response file (this would be the gated far field measurement so it will not be useful at lower frequencies where nearfield measurements would be necessary)
- f. Click on “OK” and you will see the calculated driver frequency response chart (obtained by summing the Network Transfer Function and the Unfiltered Driver Response)
- g. This method allows us to check the validity of measured results. There should be fairly good agreement between calculated and measured results



## **MAKING SpL MEASUREMENTS (dB/2.83 volts @ one meter)**

*(Section Index)*

*Note: In order to establish the true meaning of dB readings in SW, one must first calibrate by using a known mic sensitivity, amplifier, and volume control level. One method would be to use a dB level meter and then use Transform/Scale to scale the dB chart to get the right amplitude value.*

*Since there is no voltage calibration in SW, it appears that what SW does is that it compares the input signal to the output signal and if equal, 0 dB will appear. This is the reason to attempt to start with the values as close to equal (rt and lt volumes in SW) as possible. By using a Sine wave signal, you can adjust the signal to through the software to read 0.7 Vrms on a voltmeter read at the point where the signal would be input to the amplifier. Then connect an 8-ohm resistor to the amplifier and adjust the amplifier until 2.8282 volts are produced at the output.*

*When measuring frequency in loudspeakers, it is almost impossible to truly measure 1-watt response as power is load dependent. In the case of the loudspeaker, the Impedance represents the load. The Impedance is not constant over the frequency range. What is measured in practice is actually at a constant voltage of 2.828 volts; this happens to be 1 watt at 8 ohms, the Nominal Impedance of most speakers. The manufacturers commonly use the nominal impedance to determine what input voltage should be used to deliver 1 watt equivalent ( $P=E^2/R$ ). Amplifiers are constant voltage sources, not constant power sources. At lower impedances, the amplifier will attempt to deliver more power due to the constant voltage nature of the output. If the amount of power to be delivered exceeds the capability of the amplifier, the amplifier can be damaged.*

*If you have a calibrated microphone and you know your pre-microphone gain, you can calculate the actual SpL through the formula derived by Valerio Russo (based upon the standard that x number of mV/PA would be used to obtain 94dB):*

*SpL=94 + dBsw – Gain + 20\*LOG(Vref/Sens)*

*dBsw - dB read in the SW response chart*

*Gain - pre-microphone gain in dB*

*Vref - Vrms in the reference channel (Left line IN) in millivolts*

*Sens – microphone sensitivity mV/PA*

Method:

1. Connect everything up through the jig to the soundcard
2. Make sure that the mic is one meter from the baffle
3. Insert a power resistor (as close to the DUT DCR as possible) into the + output of the amp and attached to the + lead of the DMM. Attach the negative lead of the DMM to the (–) output of the amplifier.
4. Do a sound/play of an MLS signal through the speaker terminal.

5. Adjust the play-time to provide a long enough signal for the DMM to measure or alternatively use the “loop record”. If using a DMM, it must be capable of measuring True RMS voltage.
6. Using a DMM, set the volume control for as close to 2.83V as possible for the load.
7. Remove the resistor and connect to the DUT.
8. Power up the microphone pre-amplifier and set the jig for acoustical testing.
9. Set up the pulse and gate timing for the on axis measurements.
10. Measure on axis.
11. Then measure 30, 45 60 degree off axis all at a one meter distance.

## **MEASURING TOTAL RADIATED SOUND POWER**

*(Section Index)*

This cannot be done with Speaker Workshop exclusively but requires the additional use of the Frequency Response Combiner from the FRD consortium in order to calculation minimum phase. See a brief review of how to use this elsewhere in this document.

Many feel (such as Floyd Toole or Harmon International) that this measurement is the single best predictor of how a loudspeaker will perform in multiple environments. It takes a weighted average of on and off axis measurements and combines them. Total sound power is used to represent the sounds that arrive at the listener's ears after encountering more than one room boundary. Though Sound Power would be used to capture the entire acoustic power going into the room around a 360 degree circle, unless you are working in an anechoic environment or a very large room and sitting a good distance away from the source, Ian Colquhoun of Axiom Audio feels that on axis to 90 degrees off axis in each direction is probably adequate (the front hemisphere of the loudspeaker). The forward 90 degree off axis measurements are the ones that are most useful in identifying how a speaker will sound in double blind listening tests. It is probably adequate to take acoustic measurements at 15 degree increments on a horizontal plane and perhaps one or two vertical plane 15 degree measurements. John Atkinson asserts that the power response can be reasonably estimated from 7 measurements taken at 5 degree increments taken across 30 degrees on a horizontal axis (it appears that he is assuming a bilaterally equal response) in *Stereophile Magazine*, January 1999. You would want to extract minimum phase using the Frequency Response Combiner and attach this phase information to each .frd curve. The Minimum Phase is necessary for accurate vector summation of the curves. This information could then be imported as .frd files and summed/averaged in Speaker Workshop. To weight the average, you would subtract or level shift the sum by  $20 \cdot \log(\text{number of measurements})$ .

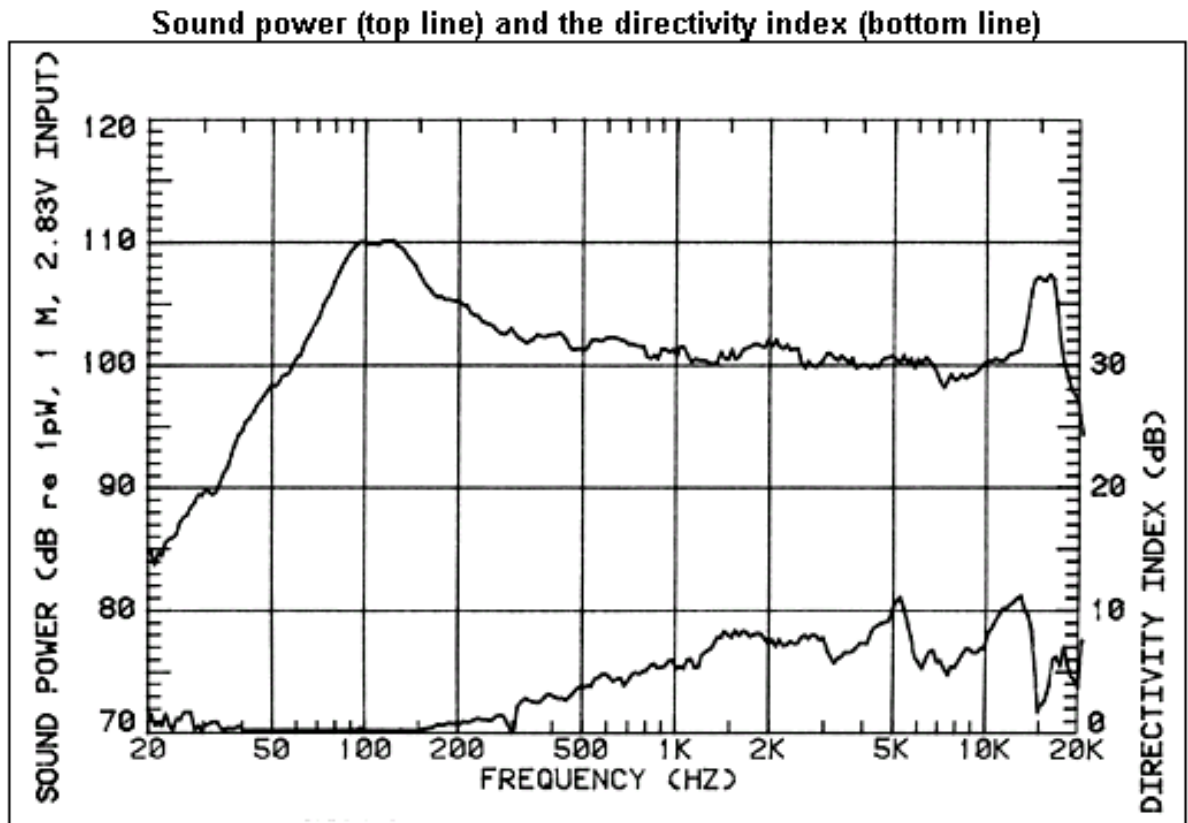
Due to the directivity of sound, a good sound power curve should start to fall off at around 150 Hz (the greatest power will be seen in the lower frequency bands due to the omnidirectionality of lower frequencies) and drift downward by about 10 dB by the time the measurements become primarily 2 pi. It should level out beyond that point until somewhere the 10 kHz mark depending upon tweeter directionality. The upper and lower curves will be influenced by the off axis response of the loudspeaker. Probably the most crucial aspect of this curve is that it transitions downward fairly smoothly as this suggests that the reflective and reverberent sound fields will sound similar to the direct sound and this will also reflect improved imaging.

The presence of absorbent room furnishings will dramatically increase the downward slope of the power response.

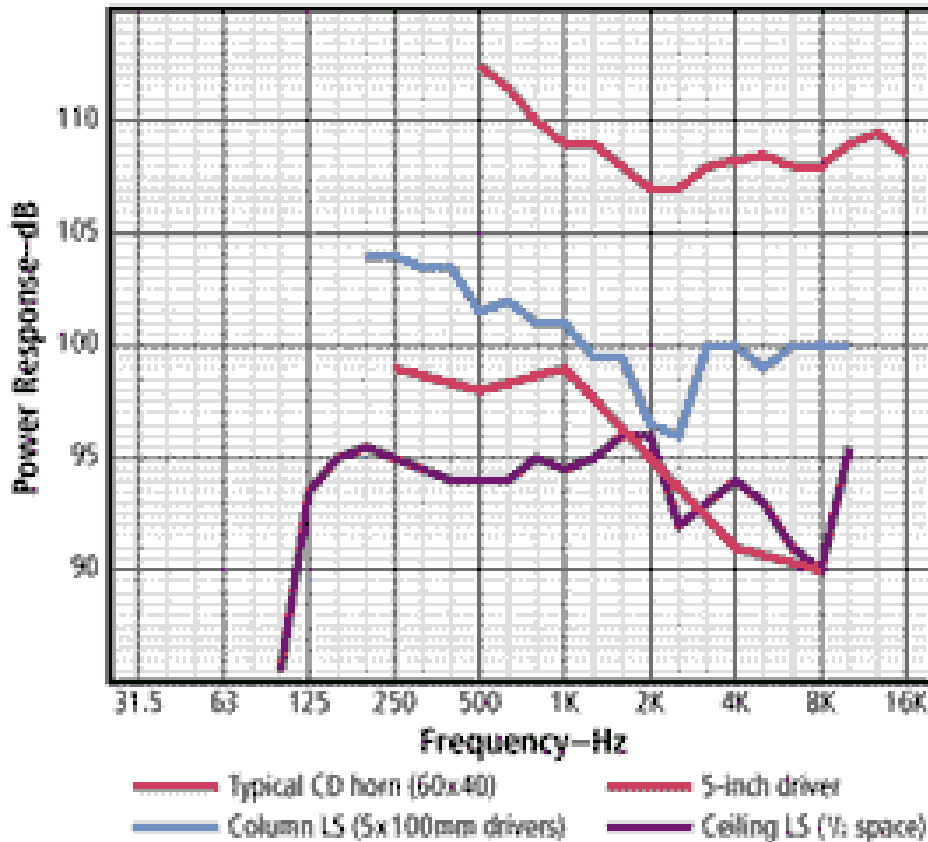
When the measurements for the "total radiated sound power" curve are taken, directivity index, or DI for short can also be calculated. The directivity index is the decibel difference between the sound power radiated by an imaginary, perfectly omnidirectional loudspeaker having a frequency response equal to the on-axis response of the test loudspeaker, and the sound power actually radiated by the test loudspeaker. In other

words, a directivity index of zero indicates the test loudspeaker is perfectly omnidirectional. Taking the on axis measurement and subtracting the average off axis measurements will provide you with a fairly good representation of this. If graphed against frequency, a flat line with very little variation with frequency suggests a wide dispersion at all frequencies.

From <http://www.soundstage.com/gettingtechnical/gettingtechnical200502.htm>:



From [http://www.svconline.com/mag/avinstall\\_sound\\_power/](http://www.svconline.com/mag/avinstall_sound_power/), note the ceiling (2 pi measurement which maintains more of a flat power response and then the other measurements which demonstrate the drop off until the tweeter's response becomes primary and there is little off axis loss of power at the frequencies measured. The horn response is uniform due to the high degree of directionality throughout the entire frequency range measured and demonstrates how multiple measurements must be evaluated together (such as the off axis frequency response graphs) in order to get a true picture of what is going on. One can see how both the reflected and reverberent fields play a crucial role in the final response and they may act as a superior indicator of speech intelligibility and that all must ultimately be considered when designing a loudspeaker.



## **CHECKING FOR PHASE RESPONSE AT THE CROSSOVER FOR EVEN ORDER CROSSOVERS**

*(Section Index)*

For Even Acoustical Order Crossovers, one can check the phase by looking at the reverse null response. By reversing the phase of the tweeter in even order crossovers, a reverse null that is wide, symmetric, and deep (20-30 dB) suggests good phase tracking through the crossover region. The lower the order of the crossover, the wider the null should be to cover the entire range. There are times that you will not see a dramatic null if you are designing for a better off axis response and not for on axis flat response. This deep null will not be seen in odd order crossovers.

Remember that for crossover simulations, we are using far field measurements unless the crossover point is too low for farfield to be meaningful. Spliced responses are for the purpose of displaying the total frequency response of the driver or loudspeaker.

## **ACOUSTIC TWEAKING (HINTS AND SUGGESTIONS)**

### *(Section Index)*

1. One tweak you can make after the crossover is done is to unwind the bass inductor a little. If the speaker appears to be too laid back, unwinding a couple of turns will raise the frequency at which the bass driver is rolled off which results in an increase in the summed response at and just above the crossover. The speaker will sound harder and brighter with voices and drum strikes coming more forward.
2. If your speakers are too harsh, adding a couple of turns of magnetic wire to the inductor may soften the response.
3. One tip for metal and ceramic dome tweeters is to put a resistor of 20-50 ohms parallel to the tweeter. This dampens unwanted oscillations and may give a smoother sound. Where this helps is that the crossover prevents damping of unwanted resonance so the amplifier can't use its damping factor. Under other conditions the amplifier could otherwise prevent or limit this oscillation. A roughly 47 ohm metal film 1 watt resistor is recommended for 8 ohms and a roughly 22 ohm resistor is recommended for 4 ohms.

## **SYSTEM IMPEDANCE MEASUREMENTS**

*(Section Index)*

To calculate the final impedance curve of a speaker system:

- a. Set up for an impedance measurement by connecting the speaker to the jig
- b. Open a driver file or a test signal
- c. Use Measure/Impedance
- d. When complete, rename the data file

Impedance Calculations from imported data:

1. Create a Driver File by Resource/New/Driver
2. Right Click on the chart
3. Select Properties/Data

**Driver properties**

General Parameters **Data**

Impedance [?] X

Frequency Response [?] X

Gated Response [?] X

Response at 30 deg [?] X

Pulse Response [?] X

Nearfield Response [?] X

Port Response [?] X

Response at 60 deg [?] X

For Thiele Calculations

Impedance in Free Air [?] X

Use this DC Res 8.000 Ohms

Impedance in Sealed Box [?] X

Sealed Volume 35.31 cu ft

Impedance with Added Mass [?] X

Added Mass 35.27 m oz

OK Cancel Apply Help

- 4.
5. Click [?] next to Impedance and select the appropriate .zma file
6. Click [?] next to Frequency and select the appropriate .frd file
7. Do this for each driver that you are importing data for
8. Build the final network in SW
9. With the network open, Right click on the chart and select Properties/General
10. Right click on the network and put a check in the "Calculate Network Impedance"
11. Select Calculate/Response



# **DISTORTION MEASUREMENT**

*(Section Index)*

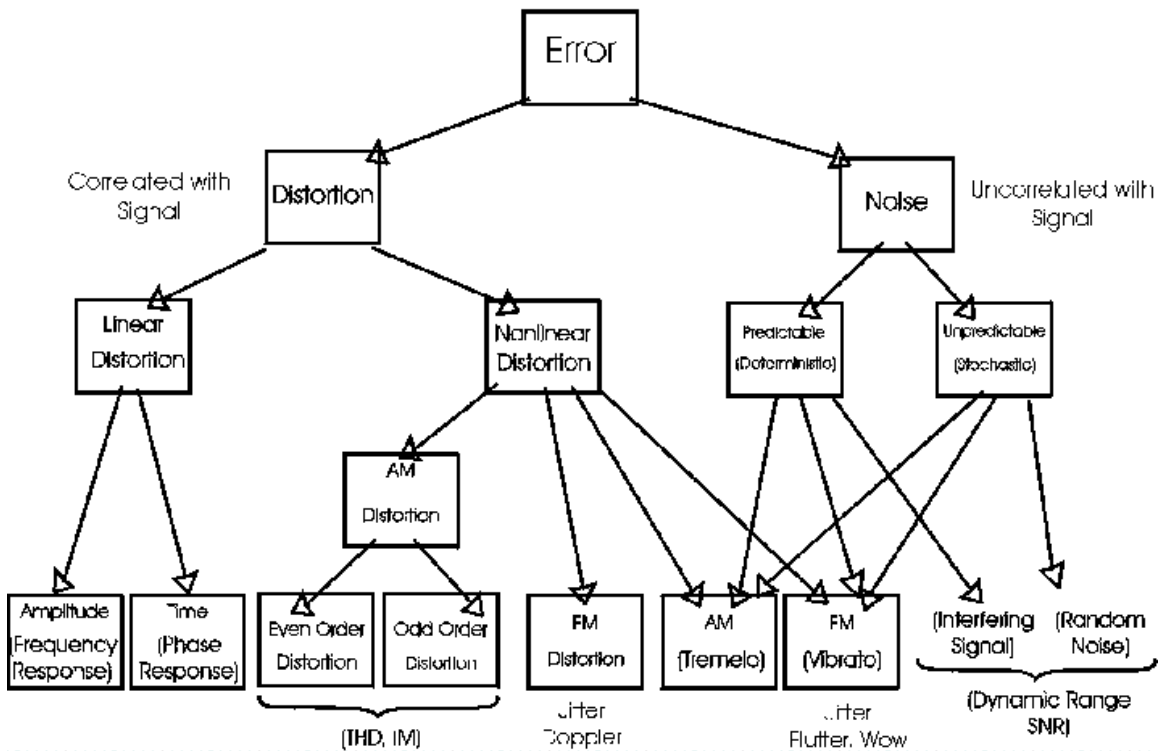
Speaker Workshop has the ability to measure certain linear (Impulse Based) and Non-linear types of distortion. To evaluate transient response, record an impulse and then measure. You can learn how to listen for different types of distortion at <http://www.pcabx.com/index.htm>. This site has freeware to teach how to listen for tone quality, loudness, noise distortion, distortion, and articulation.

The diagram below summarizes types of acoustic distortion/error and is from:

[http://www.pcvtech.com/soundcards/techtalk/dist\\_noise/index.htm](http://www.pcvtech.com/soundcards/techtalk/dist_noise/index.htm)

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## Categories of Error



Linear distortion measurements include the Frequency, Phase Distortion, and Transient Response. Linear Distortion changes the relationship of the size and timing of the frequencies that compose the input signal but it creates no new frequency signals beyond those that are already present in the input signal. It changes the tonal balance of program material. Linear Distortion is stored energy, which is associated with irregularities or resonances in the frequency response. This might be best visualized using the Cosine Wave of an infinite number of frequencies of waves. An impulse is about as pure a transient as you'll find. Impulses can be gathered directly or by taking an autocorrelation of an MLS excitation. By starting all of these waves at a value of 1 at time zero to the system, a decrease in amplitude of any of these FM waves (measured using a FFT or Fast Fourier Transformation) will show up as a decrease in frequency response at that

wavelength; so in actuality, the frequency response of a system demonstrates one form of linear distortion. Frequency response measurements were covered earlier in this manual.

Another Frequency related linear distortion is Group Delay. This is the change in phase that occurs in relationship to the change in frequency after an Impulse is fed through the system. One would expect that an impulse applied to a linear phase system would preserve the waveform shape. If the magnitude is not evenly delayed throughout the frequency spectrum, waveform distortion will appear and this is called Group Delay.

There are also linear distortions that occur in the time domain. Excess phase delay is the difference between the loudspeaker's actual measured phase and its theoretically minimum phase as calculated from its frequency response. This is done in the time domain so that this is measured in seconds; i.e., how far you would have to shift a waveform to the left on a typical oscilloscope plot in order to get it to align with the input reference waveform.

Transient response may be evaluated by looking at impulses or steps in the time domain. The Step Response and Waterfall Plots measure these. Waterfalls are more useful for spotting resonance, internal box reflections, and stored energy (though frequency plots may also demonstrate this). Waterfall plots are derived from the impulse response. The acoustical reflections from the floor, ceiling, walls, etc. of the listening room add another set of "time and frequency domain distortions" onto those of the audio equipment, though more delayed in appearance (a few milliseconds later). See [Fusion Time](#) below. If the difference in the arrival of the signals to the ear (and brain) are spread apart enough, the brain can differentiate the reflected signal from the original and does not interpret the sound as a distortion.

One property of hearing known as the "fusion time" (the interval of separation between short transients required to perceive whether one or two transients are present), permits a listener to distinguish between room reflections (normally arriving more than about 5 msec after the direct sound) and time-domain distortion created within audio components; thus fusion time helps us to become familiar with and ignore most reflections from room boundaries, while letting us discern time domain distortions produced by equipment/loudspeakers as a blurring/smearing of musical transients or an alteration of spectral balance, etc (paraphrase from John Dunlavy, 1997).

Earl R. Geddes and Lidia W. Lee have found that:

1. Distortion by products that are created upward in frequency are less likely to be perceptible than those that are created downward from the stimulus signal.
  - a. Higher order non-linearities are more likely to be heard than lower order non-linearities.
2. Distortion by-products that lie closer to the excitation are less likely to be perceived than those that lie further away (masking).
3. Distortion signals are more likely to be perceived at lower signal levels than at higher signal levels.

## **LINEAR DISTORTION TESTING**

### **Waterfall Plots**

*(Section Index)* Commonly used forms include:

1. Cumulative Spectral Decay (CSD)
2. The Wigner distribution (not supported by SW)
3. The Energy-Time-Curve (ETC) or spectrogram. (which is not supported by SW)
4. Frequency-File-Display (FFD) (not supported by SW).

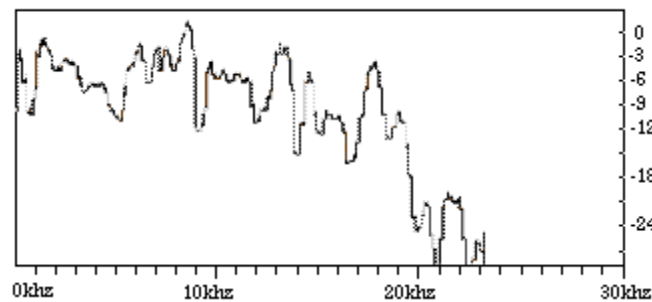
The Cumulative Spectral Decay is intended primarily for anechoic evaluations (or with gating, if necessary) and always uses the z-axis as time. The cumulative spectral decay plot, for any specific frequency, shows the decay of the system output for a pure tone input that is suddenly turned off; you can observe loudspeaker cabinet or other resonance which persist after the excitation ceases – ideally, the loudspeaker’s impulse response should die away instantly. Real drivers store energy and have inertia so this causes a delay. Another way of interpreting a CSD plot is as a spectrogram of the tail of the selected impulse response segment. An impulse response that decays quickly will have a CSD waterfall that also decays quickly and visa versa. The advantage of the CSD over just the impulse response or, an energy-time-curve (ETC) display of the decay is that the CSD plot shows you the frequency-decomposed decay while the ETC lumps together all frequencies. For Waterfall plots, the more vertical the wall, the better the transient response. A resonance will show up as a slowly decaying ridge along the time axis (often referred to a ringing). You can see where there is ringing (energy storage) by a slower drop off of total system energy at a particular frequency or group of frequencies. Well-defined ridges that are parallel to the time axis represent a non-uniform decay mode often suggestive of some sort of resonance (this could be driver non linearities, port “organ pipe” resonance, crossover resonance, etc.). Ridges that are perpendicular to the time axis represent a uniform decay mode. The closer the ridges are together, the more rapid the decay and the less energy storage there is. Beyond 2 or 3 milliseconds, low frequencies and long delay times in a CSD plot are less reliable as the number of points collected is linearly related to the frequency; this means that the highest octave has half of all sample points and the lowest octave has a very low resolution. The rise-time represents how fast the impulse tone is turned off. Small rise time values improve time resolution at the expense of frequency resolution while larger values improve frequency resolution at the expense of time resolution. This is a tradeoff of CSD analysis. Typical values for the window rise time range from 0.01 to 0.35 msec. A non-resonant driver will show a fast and even decay response.

ETC produces a more accurate psycho acoustic measure of in room response than RTA. It is derived from the impulse response. The ETC is primarily intended for room acoustic evaluations. The z-axis always represents time. This is not supported in SW. The ETC is the envelope of the analytic signal made up of a real part (the impulse response) and a calculated part (the Hilbert Transform of the impulse response). It does not really represent sound energy. It never goes negative so it removes confusion that may be caused by negative going ripples in the Impulse Response.

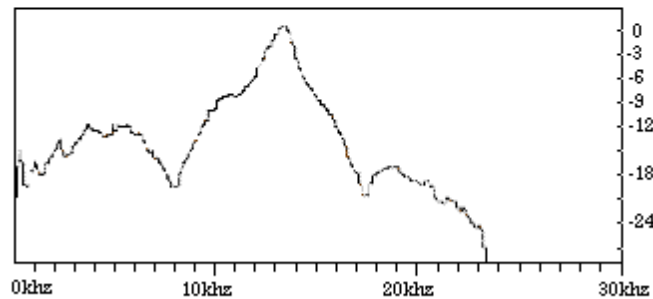
The Wigner distribution is primarily intended for anechoic loudspeaker evaluations. The Wigner distribution is always plotted on a linear vertical scale. The Wigner distribution floor level is always zero and negative values are plotted below the floor level. The z-axis always represents time. This is not supported in SW and provides little more useful information than that of the CSD.

The FFD mode offers a general purpose means of plotting multiple frequency domain data files and finds its primary application in plotting loudspeaker polar response data. This is not supported in SW.

Resonance or Ringing can sometimes be more clearly seen when graphing a frequency response in a linear rather than a logarithmic manner such as below where multiple resonances are noted:



Or one can sometimes identify a single significant resonance as below:






*While most "musical transients" probably do not generate as broad a spectrum of frequencies as a short-duration rectangular pulses used for measurement purposes, their accurate reproduction usually requires a bandwidth extending from an octave below their lowest frequency component to an octave above the highest component. Over this bandwidth, the entire reproducing system must exhibit an excellent "amplitude" response in both the frequency-domain (amplitude Vs frequency) and the time-domain (amplitude Vs time), to reproduce them with little or no audible waveform distortion. This requires a "system", including the loudspeaker that exhibits what is frequently referred to as "time-coherent" performance - along with a flat "frequency response".*

*John Dunlavy, 1997.*

## **Waterfall Measurements**

### *(Section Index)*

*Note: Waterfall plots can only be calculated on time data sets.*

1. Double Click  Measurement.in.l and a graph opens
2. Go to Menu/Calculate/Waterfall
3. For the source, click on “?” and locate and insert System/ 
4. For the reference, click on “?” and locate and insert System/ 
5. For Windowing, you can try various types however Blackman and Hamming are the most commonly used for this type of measurement (however Uniform or no windowing is the default). (See [Windowing And FFTs](#) in the appendix).
6. Sample Time: This is the amount of decay time you want to capture, usually between 1 and 5 milliseconds.
  - a. Make certain that the sample time is slightly less than the gate time (this is the sample size or the range)
7. Number of Samples: Set this between 20 and 50.
8. Smoothing: 100.00m
9. Sample Size: You can choose to use the markers as they were when you measured the initial pulse or you can manually set the range
  - a. The points will be automatically calculated based upon your choice from the sample size divided by the number of samples.
10. Select OK
  - a. A Waterfall Plot will appear titled *Waterfall Plot of Measurement.in.l*
  - b. This creates many new datasets (one per plot line), a chart and a folder to contain the datasets.
11. Right Click on the Chart
12. Select Chart Properties.
13. Set Y axis limit at  $-40$  to  $-50$  dB to start
  - a. if you don't limit the graph enough, you may not see the Waterfall plot well
14. Set all of the individual data sets to the desired colors on the chart.
15. Make sure that all of the data sets are the same color
16. Right click the chart area and use the undo create waterfall if you need to change the sample time or number of samples or if you want to apply smoothing to the data set.
17. To apply smoothing
  - a. Undo the Waterfall
  - b. Menu/Transform/Smooth/Choose Amount of Smoothing (usually 1/10 Octave is ample here)
  - c. Menu/Calculate/Waterfall
18. When you are satisfied, Right click the chart area.
19. Select make chart default
20. All of the Waterfall plot lines will then show up in the color selected.

## Calculate/Waterfall/for Datasets

*(Section Index)*

Waterfall plots can only be calculated on Time datasets.

### Fields

<i>Source</i>	The source time data to be analyzed. By default this is the current dataset.
<i>Reference (optional)</i>	The reference time data. This should be the actual reference data captured during the pulse capture.
<i>Windowing</i>	The FFT windowing applied during the analysis. Default is Uniform (none).
<i>Sample Time</i>	The amount of time to calculate the waterfall for. This is the decay time you want to examine (often 1-5ms). Note: the sample time must be less than the Sample Size (time range).
<i>Number of Samples</i>	How many different plots to generate for the waterfall. They will be spaced sample time / number of samples apart.
<i>Smoothing</i>	You can apply smoothing in pieces of an octave (e.g. .3 = 1/3 of an octave). Usually 1/10 of an octave (0.1) is plenty.
<i>Sample Size (Time Range)</i>	Select the range of data to be analyzed You can choose to use the markers or set the range manually..
Actions	
?	Press the ? button to browse for a reference or source response.
X	Press X to clear the reference response.
OK	Click OK to perform the calculation. This creates many new datasets (one per plot line), a chart and a folder to contain the datasets.

Click OK to calculate the waterfall. Right click the chart area and use the undo create waterfall if you need to change the sample time or number of samples.

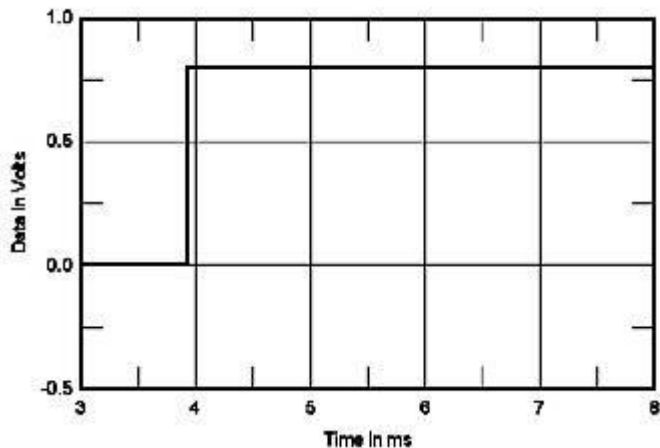
## Step Response

[\(Section Index\)](#)

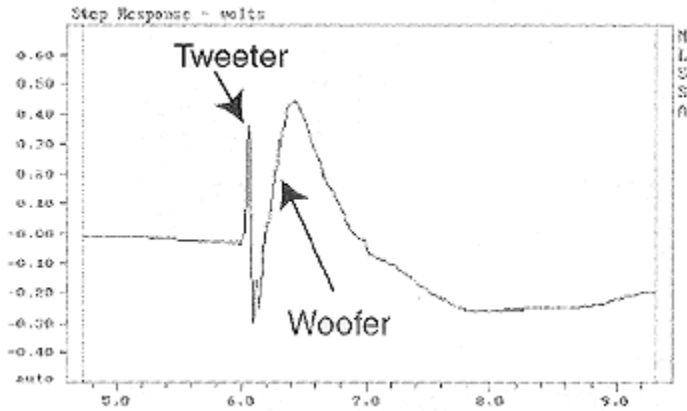
Step Response is a useful qualitative measure of the systems time coherence. It is an integration of the Impulse Response. The first positive going spike is the tweeter moving forward, the second is the midrange or the mid-woofer, and the third is the woofer. As a general rule, because of differences in the timing of the response of the different drivers, Step Response measurements would need to look at tweeter responses from 0-2 msec, midranges from 0-20 msec, and woofers from 0-60 msec. If any of the drivers are connected with opposite polarity, their spike will be negative. The recovery wave is opposite in polarity to the step peak. If there were to be perfect time coherence, there would only be one peak. Using the step response, you could read the differential temporal arrival of the sound waves at the measured (usually crossover) frequency, calculate the difference in acoustic centers, and physically re-align the drivers (you would still need to adjust the network for phase, which would impact the acoustic center further complicating matters).

The following information is primarily taken from John Atkinson published in *Stereophile Magazine Measuring Loudspeakers, Part Two*, December, 1998:

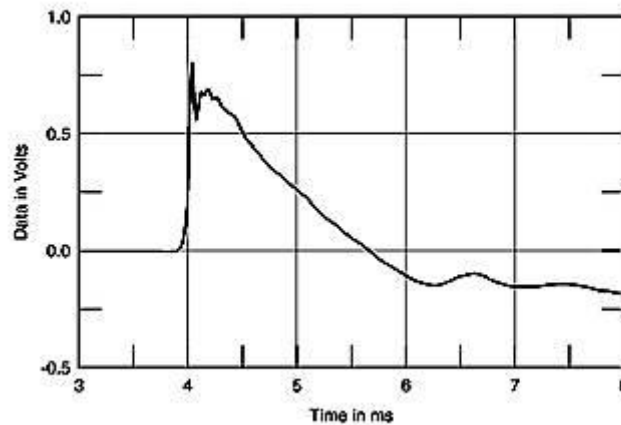
An ideal Step Response would look like this with an immediate rise and continuation of response as long as the duration of the pulse (Note that an impulse lasts briefly so ideally you would see an immediate drop after the impulse is completed):



A good step response though not so time coherent step response is as follows:



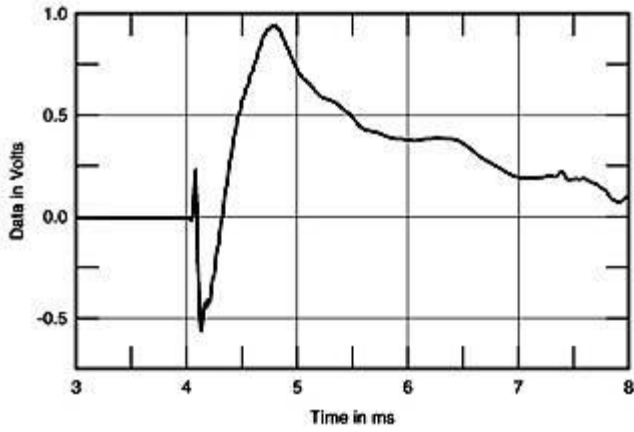
The following is an example of an excellent Step Response Plot with fairly good time coherence. It demonstrates an initial rapid sharp peak from the tweeter and the subsequent slower rise and



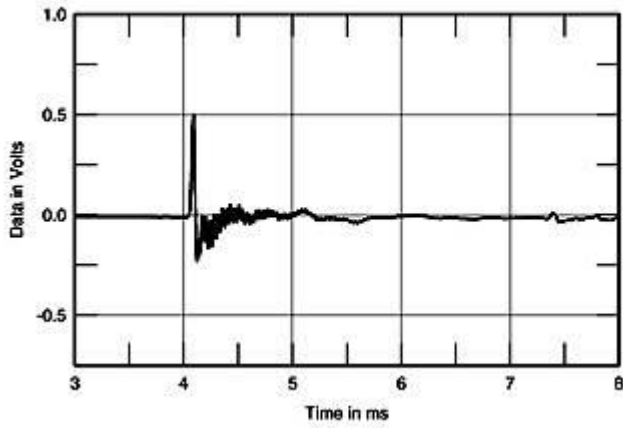
fall of the woofer.

The following diagram demonstrates a typical step response of a three-way loudspeaker. This time there are actually three step responses apparent in the graph: a narrow, positive-going step response from the tweeter; the next, negative-going step is the midrange unit (it's connected with opposite polarity to the tweeter); with finally a slow, wide positive pulse from the woofer. The tweeter's and woofer's step responses initially depart from the time axis in the positive direction, meaning that both units are electrically connected with positive absolute polarity: a positive electrical voltage results in a positive acoustic pressure; however, the midrange unit's step response initially moves away from the time axis in the negative direction, showing that this drive-unit is electrically connected in inverted absolute polarity.

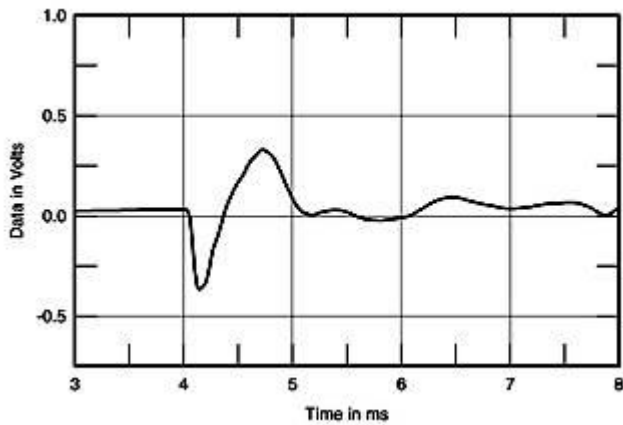




Dividing out the different responses, you see first the tweeter step response (below):



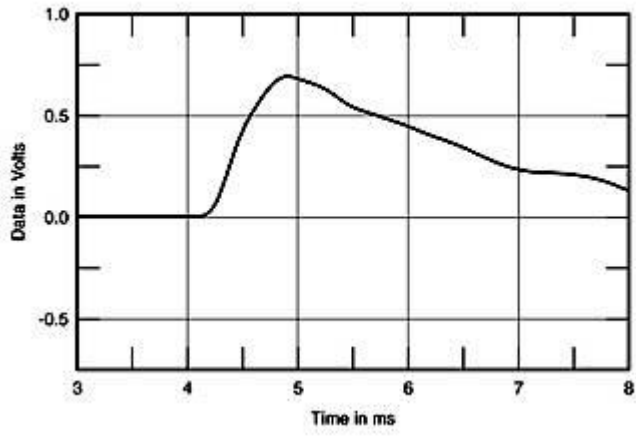
then the midrange response (below)



and finally the woofer's step response (below).

Note that the woofer's step response features a very slow rise time as well as a slow decay. A woofer's rise time is dominated by the crossover low-pass filter, which discards the high-

frequency information associated with a quick move away from the time axis.



## **Measuring the Step Response**

[\*\(Section Index\)\*](#)

Unlike most of the measurements in SW (which use an MLS), step response is done with a pulse. To measure step response

1. Create a new generator
2. Set properties to type = Pulse
3. Go to the pulse tab and set the pulse duration you want
4. Select Sound / Record
5. Set the time duration of the recording and whatever your input channel set is
6. The data will be in myGenerator.in.l and .in.r

*If one of your channels is reference you should see a clean pulse on that channel and the system step response on the other.*

An alternate way would be as follows:

1. Open a test signal and have it set for an MLS signal
2. Measure/time response
3. Look for the response in the .l and .r data sets.

## **Group Delay**

[\*\(Section Index\)\*](#)

$$\text{Group Delay} = - (\text{Phase @f2} - \text{Phase @f1}) / (f2 - f1).$$

Group Delay is derived from the slope of the phase response in the frequency domain. Group delay really refers to the relationship of phase of a small group of frequencies in a narrow range. It reports the data based upon a central frequency in this range but is looking at the change in phase vs. change in frequency in this range.

In a perfect world, there should be no delay from one frequency to the next so the slope would be zero. In general, the group delay decreases with increasing frequency. A flat Group Delay curve assures that all frequencies are arriving simultaneously. Group Delay provides some information about the Q of the filter action and hence the ringing of the loudspeaker at Fs. Up to Q values of about 0.6, there is no peak and therefore no ringing is noted (A Qtc of about 0.5 would be considered Transient Perfect; note that lower Qtc's will have a better transient response but will sound more "thin" as a general rule. Increasing the Q will lead to more peaking with a result of more ringing. It may sound more "full bodied" than the original signal source (which some people like though it is not accurate reproduction). Speakers with a high peak in the Group Delay (at low frequencies) tend to sound "boomy".

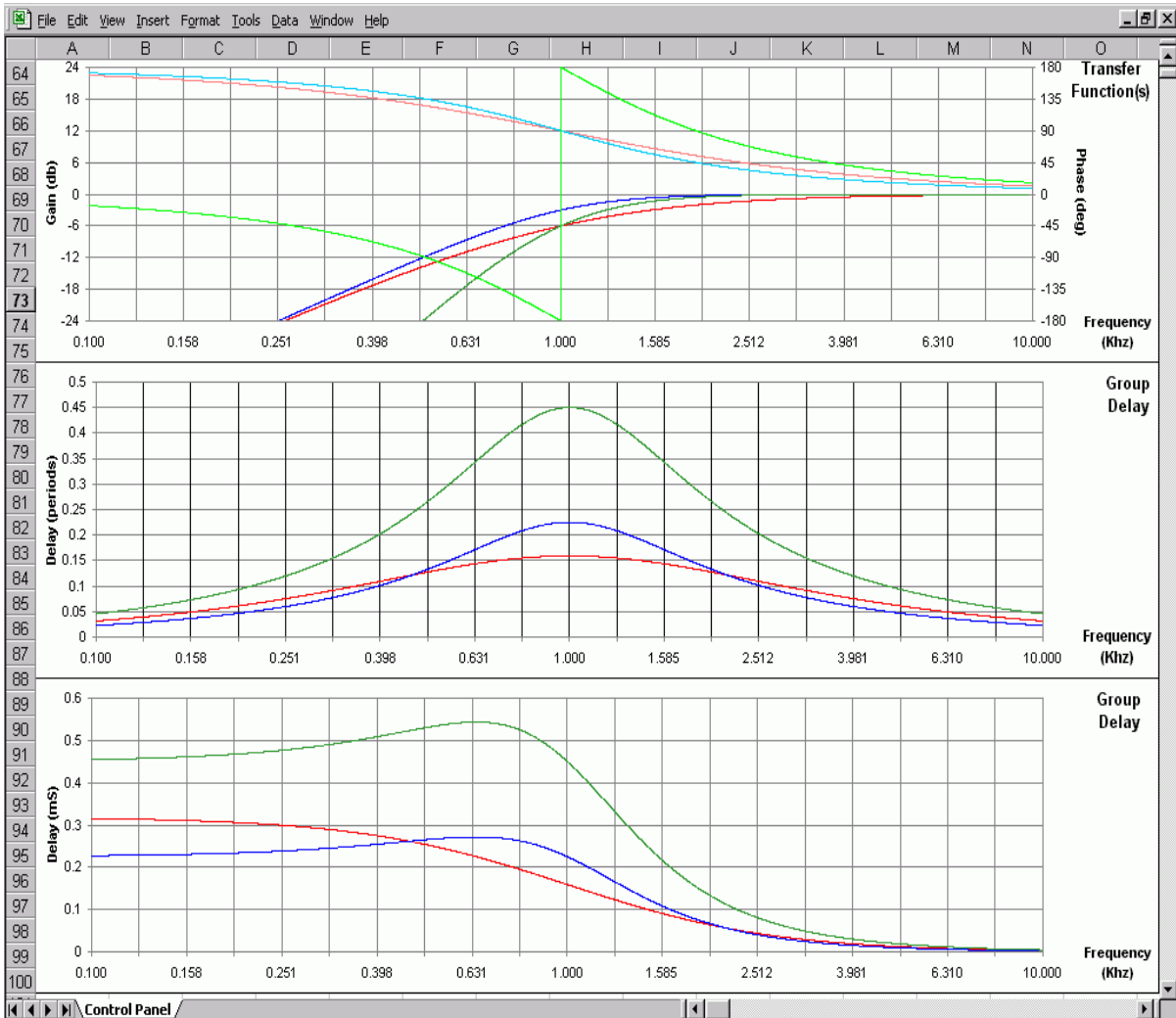
If Group Delay is large enough it will be audible Blauert and Laws [May 1978] report threshold levels in the range of 2 msec @8kHz; 1 msec @ 2kHz; and 3.2 msec @ 500 Hz). John Murphy has taken this information one step further. He "normalizes" Group Delay into cycles of delay calculated by dividing the group delay by the period of the frequency at each point (or the frequency times the number of seconds of delay); he feels that this may be a more accurate comparator. He notes that below 2000 Hz, 1 - 2 periods of Group Delay (or group delay \* frequency products of less than 1000 to 2000) are likely to be inaudible or barely audible. This would equate to 1 ms at 2000 Hz or (through extrapolation) 10 ms at 200 Hz. The audible threshold reported by John Dunlavy [1997] is about 30 degrees of phase shift per octave of Group Delay (the product of Group Delay and Frequency would be about 167 or a cycle of delay of .167 periods). Dan Wiggins (Adire Audio) recommends targeting a Group Delay and Frequency product of less than 400 (or a "cycle of delay" of 0.4). Other authors recommend that the Group Delay should be down to about 10 ms by 100 Hz. For ported systems, the maximum Group delay occurs around the box tuning frequency. For a given woofer and box size, lowering the Fb will increase the peak value of the group delay but this will also reduce the Group Delay at higher frequencies. .

A large group delay will provide a very low deep base that will rumble however the response is slow and not sonically pure. A smaller the Group Delay results in greater sonic purity (from this perspective). A larger box and lower tuning tend to favor a greater group delay. Closed box systems are more likely to have a lower Group Delay than that of Ported systems.

The perception of Group Delay is related to frequency and you cannot compare the group delay of two separate systems if they have different frequency responses, distortion characteristics, and out of band influences as these response differences will have much

more of an audible impact than will the Group Delay. Group Delay from crossovers will be more of an issue as your ears are much more sensitive in the mid frequency ranges to it. A group delay of .2 milliseconds at 2000 Hz will be as audible as one of 2 milliseconds at 200 Hz.

From Daryl Patterson (<http://www.villagephotos.com/pubgallery.asp?id =144711>) at the Madisound Board: If a sealed system has a  $Q_{tc}$  of .7, the Group delay will peak at .23 periods at the  $F_{tc}$  no matter what the  $F_{tc}$  is. The Group Delay of a Linkwitz Riley 4 crossover will be .45 periods at the crossover frequency while a Linkwitz Riley 2 will be .17 periods at the crossover point and a Butterworth 2<sup>nd</sup> order crossover will be roughly .23 periods. Note that the peak delay in Group Delay time occurs about 2/3 of an octave below the crossover point for LR4 and for BW2 (roughly .55 msec for LR4, .28 msec for BW2) and will be roughly 2 octaves below the crossover point for a LR2 at .3 msec. At the crossover point, the time delays will be roughly .45 msec for LR4, .22 msec for BW2, and .16 msec for LR2. Below LR2 (red), BW2 (blue) and LR4 (green) using an arbitrary 1000 Hz crossover:



## **Measuring the Group Delay**

[\(Section Index\)](#) Group Delay Dialog

### 1. Calculate/Group Delay/for Datasets

Group delay can only be calculated on Frequency datasets. It essentially finds the slope of the phase curve at each frequency.

2. Frequency Range    Restrict the frequency range of the calculation by selecting a min and max frequency. When the low frequency information is jittery set the minimum frequency above the jitter.

3. Click OK            This calculates group delay with a new frequency dataset with a Delay suffix.

## Interpreting Square Wave Responses

[\*\(Section Index\)\*](#)

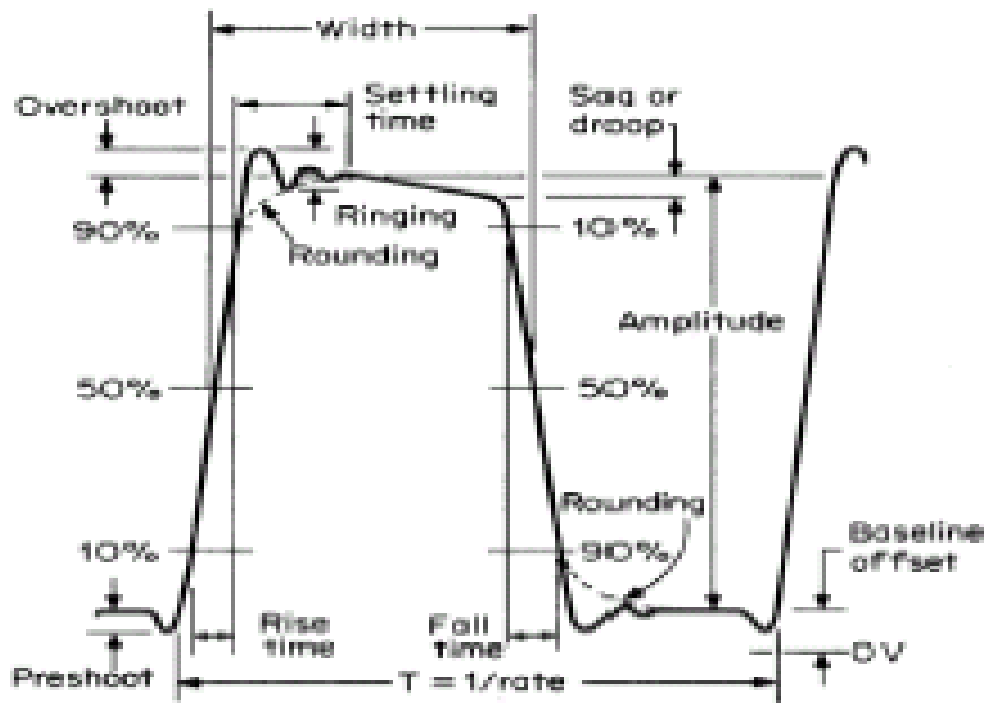
Though Square waves do not occur in real music, a square wave signal can be very revealing of the system's ability to accurately reproduce the signal entering it. From: Interpreting square waves (from Tremaine, H. M.: *Audio Cyclopedia*, Howard W. Sams, Indianapolis, IN, 1975.): Definition of rise time, overshoot, ringing, and droop. From Time Domain Audio Measurements by Dr. Richard C. Cabot, P.E Bruce Hofer, Robert Metzler:

“The overshoot of a device is the amount by which the peak of the square wave exceeds steady state positive or negative amplitude, as shown. Large overshoots are indicative of peaking or excess phase shifts in the high-frequency response of the device under test. The maximum overshoot is normally the parameter specified, but the subjective appearance of the overshoot is also of interest. Well-behaved devices will have smooth overshoot, symmetrical on both positive and negative peaks. Overshoot normally occurs only on the leading edges of a square wave. However, if the device is linear-phase, as are many digital filters, there will be symmetrical overshoot on both leading and trailing edges.

Ringing is the tendency of band-limited square waves to oscillate on the peaks. This is largely a subjective measurement with the results stated as the severity of ringing. The greater the high frequency peaking, the greater will be the ringing.

Tilt is a measure of low-frequency behavior. As low frequencies are filtered, phase shifts are introduced which cause the leading edge of the square wave to rise and the trailing edge to fall. This produces a tilt to the top and bottom of the square wave. The tilt is usually expressed as a percentage of the peak amplitude of the square wave.

Slew rate is a measure of how fast a signal changes from one instantaneous value to another. The ideal square wave can switch from one amplitude extreme to the other instantly. Practical devices cannot keep up with this transition and will often have a maximum speed with which they can change. This speed limitation will result in a tilted straight-line portion of the square wave edge. This should not be confused with the exponential rounding of square-wave edges, which results from a high-frequency roll-off (the effect measured by rise time). The effects of bandwidth limiting are linear; the effects of slew-rate limiting are not. A simple way to determine whether a signal is slew-limited is to increase the signal amplitude. A bandwidth-limited signal will make the transition in the same time; a slew-limited signal will take longer.”

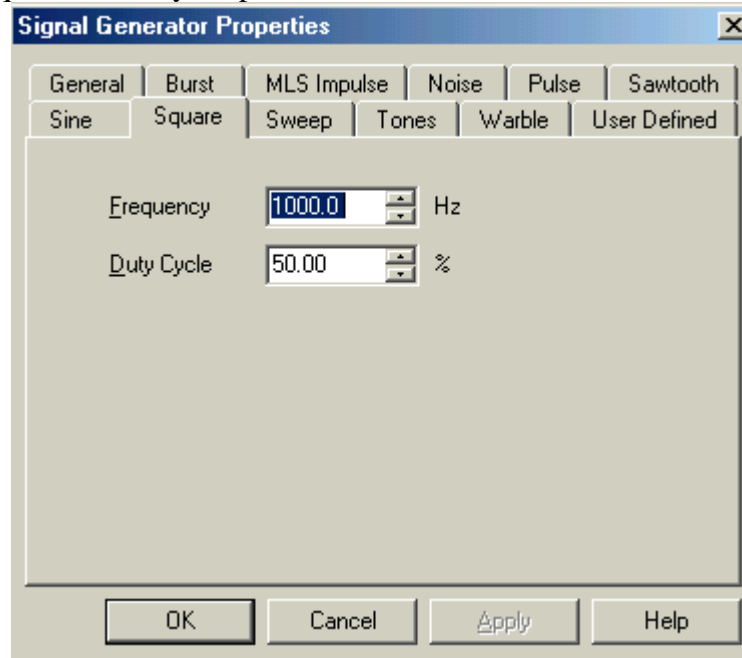




## Measuring Square Wave Response

[\(Section Index\)](#)

1. Resource/New/Signal
2. Name the Signal "Square Wave"
3. Click OK
4. Right click on the chart/Properties
5. Select Square and set your parameters:



6. Go to the General Tab and select the Square Signal Type along with the desired Sampling Rate.
7. Exit out of the dialogue box and the signal generated with the next command will remain as the last signal that you have chosen (a square wave in this case).

## **NON LINEAR DISTORTION TESTING**

*(Section Index)*

Right Mark Audio Analyzer ([Volume Check with RMAA](#)) is another piece of freeware that has the capability of measuring some forms of distortion. It may augment the abilities of Speaker Workshop in this area. This will be described after the SW portion is described. I have done my best to walk through this program but I have not tested it for accuracy.

Non-linear distortion measurements that are supported include Total Harmonic Distortion (THD) and Intermodulation distortion (IMD). Siegfried Linkwitz feels that IMD, especially using multiple tones along with evaluation of energy storage (see Waterfall, CSD, etc.), is a major limiting factor in loudspeaker transparency. Comparisons of distortion should be made at equal volumes and frequencies. Some recommend that it should be tested near the power limit of the driver (i.e. 20 watt output for a 30 watt driver). Distortion will change based upon the power fed to the driver and then registered by the microphone. Harmonic Distortion is usually measured under either anechoic conditions or under near field conditions to attempt to minimize contamination via reflections.

Distortion occurs as a result of non-ideal conditions in a dynamic loudspeaker. The main non linear causes of distortion arise as a result of suspension compliance ( $C_{ms}(x)$ ), force factor ( $Bl(x)$ ), and inductance ( $L_e(x)$ ) nonlinearities. The magnetic field in the region where the voice coil travels (the magnetic gap) is not uniform due to the presence of the cone, which prevents a symmetrical magnet array. As a result of this, the coil experiences different magnetic fields traveling forward and in reverse. The field is attenuated as the coil travels off center with an increase in attenuation as it gets further from the center. The spider and the surround impact the cone similarly. This effect alters both the positive and negative pulls on the coil and leads to even order harmonic distortion since the resistance is symmetric. Temperature changes as the voice coil heats up will modify the amount of distortion. In addition, construction inconsistencies may cause irregular resistances with the cone movement. These issues affect the forward and reverse directions differently leading to odd order distortions in the loudspeaker. Even-order distortion products are caused by asymmetrical driver system nonlinearities while odd-order distortion products are caused by symmetrical driver system nonlinearities.

There are several confounding variables that can cause distortion measurements to be inaccurate or inconsistent. Room acoustics can cause problems due to reflected tones and their impact on the originally generated tones. Microphone distortion is likely to occur at volumes significantly lower than that of the speaker. In most cases, the testing level for distortion will be below 100 dB; this is below the point at which most testing microphones begin to distort. You need to make sure that the microphone and the pre-amplifier can handle the testing conditions without introducing clipping distortion. Preferably, the test should be conducted in anechoic chamber or at least in "half-space" acoustic environment.

Comparisons of distortion should be made at equal volumes and frequencies. Non-linear distortion tends to be level-dependent; this is one argument for the use of multiple drivers. The amplitude of the non-linear distortion products increases more quickly than the primary signal with increases in voltage. Some recommend that it should be tested near the power limit of the driver (i.e. 20 watt output for a 30 watt driver). Distortion will change based upon the power fed to the driver and then registered by the microphone.

The following chart from

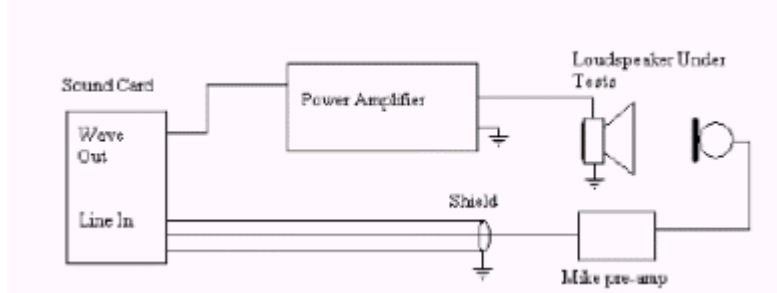
<http://www.audioholics.com/techtips/roomacoustics/HumanHearingDistortion.php>

indicates the results of studies looking at the audibility of non-linear distortion:

Item	Study/Report Author & Date	Subject	Results/Conclusion	Misc. Notes																				
1	James Moir/1981	Just detectible distortion (JDD) levels	JDD "level can be no lower than 1%"	Assumed non-linear distortion was the primary type of distortion involved. JDD levels dropped as listeners learned to recognize distortion.																				
2	D.E.L Shorter/1950	Sound quality of systems with known quantities of harmonic distortion	Just perceivable distortion values of 0.8% to 1.3%	Multiplying harmonic amplitudes by $n^{2/4}$ ( $n$ = harmonic order) before rms summing produced better correlation between objective measurement and subjective assessment.																				
3	P.A Fryer/1979	Listening tests for intermodulation distortion	2% - 4% distortion detectable in piano music, 5% in other types of test signals	Distorted test material using 1st order IM distortion products																				
4	Von Braunmühl & Weber/1937	Distortion sensitivity at selected frequency bands	1% - 2% at frequencies $>\sim$ 500 Hz	Noted that at lower frequencies JDD levels could go much higher																				
5	Harry F Olson/1940	Just detectible distortion (JDD) levels	JDD level of .7% using 40 Hz to 14 kHz bandwidth test system	Restricting system to 4 kHz doubled the JDD level																				
6	M. E. Bryan & H. D. Parbrook/1960	Just detectible amplitude of 2nd – 8th harmonic in the presence of a 360 Hz fundamental (f1)	<table border="1"> <thead> <tr> <th></th> <th>2nd</th> <th>3rd</th> <th>4th</th> </tr> </thead> <tbody> <tr> <td>52.5</td> <td>44</td> <td>52</td> <td>52</td> </tr> <tr> <td>60</td> <td>52</td> <td>57</td> <td>61</td> </tr> <tr> <td>70</td> <td>47</td> <td>62</td> <td>67</td> </tr> <tr> <td>76</td> <td>-</td> <td>54</td> <td>59</td> </tr> </tbody> </table>		2nd	3rd	4th	52.5	44	52	52	60	52	57	61	70	47	62	67	76	-	54	59	Chart at left shows just discernible amplitude in dB for the 2nd, 3rd, and 4th harmonics below (f1) levels shown in left column
	2nd	3rd	4th																					
52.5	44	52	52																					
60	52	57	61																					
70	47	62	67																					
76	-	54	59																					

In interpreting the % data above, remember that every 6dB drop equates to a halving of distortion while every 20 dB drop equates to 10% of the higher output. If the distortion level is at 40dB and the primary signal is at 80 dB, you are at 10%\*10% or 1% distortion. If the primary signal were at 86 dB, it would be half of 1% distortion or .05% distortion.

The distortion measurements are done with a set up as you would for acoustic

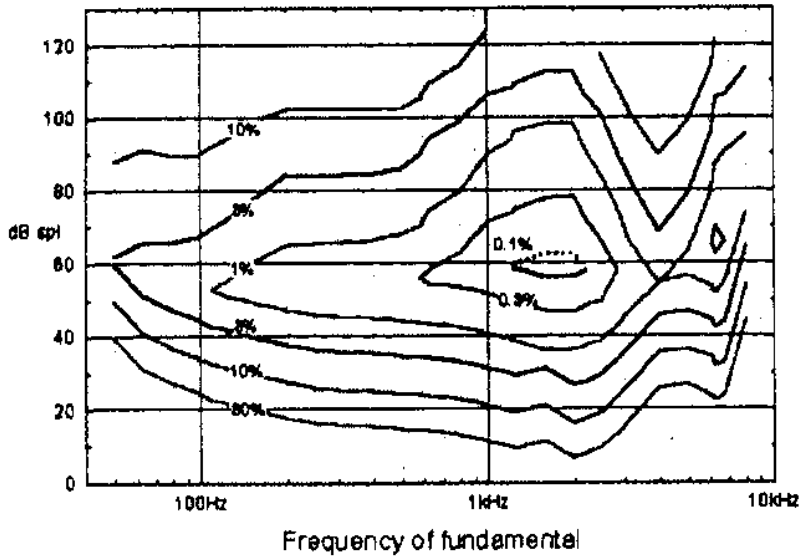


measurements:

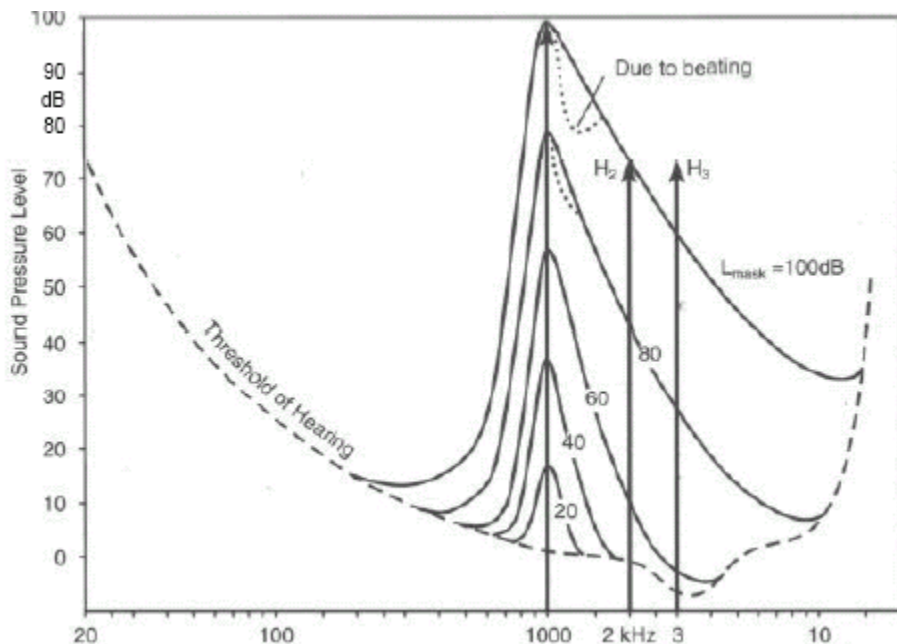
*You may be able to minimize the microphone's contribution to distortion by following these guidelines though as noted above, most measurement microphones introduce only a small amount of distortion at measurement levels:*

1. *Do a distortion measurement nearfield with the microphone exposed to a volume that would distort it but the driver volume should be low enough so that it barely distorts (i.e. the distortion curve would essentially be that of the microphone).*
2. *Then move the microphone farfield and adjust the volume so that the driver/speaker is being exposed to this same volume (SpL) that the microphone was exposed to when closer to the driver. To obtain the same SpL, with the microphone further away, the power to the speaker would have to be increased somewhat increasing the amount of likely Speaker distortion.*
3. *This measurement may need to be gated to eliminate room effects.*
4. *Combine the graphs and do a Calculate/Subtract leaving the driver distortion measurements.*

The ability to hear a harmonic depends on the frequency and amplitude of the fundamental, as well as the percentage of distortion. The ear is more sensitive to harmonics of fundamentals at medium frequencies (1-2 kHz) and medium levels (80 dB). In the plot below, a second harmonic smaller or equal to a given percentage is audible if it is outside the corresponding contour curve of constant distortion (i.e., 0.1% in the case above). In the following graphic, in order to be audible at 1 kHz and 60 dB, distortion would have to be over 0.3% or over 5% if the sound is at 100 Hz and 40 dB. Another thing of interest to notice on the graph is that as volume increases, masking will occur due to other causes.



Another issue is masking. The closer that a fundamental is to its distortion frequency, the less likely you are to hear distortion. At lower frequencies, harmonics are closer to one another and to the fundamental. When you combine this with the effects demonstrated by the previous graph at lower frequencies, you can see how low frequency distortion may be difficult to distinguish. The graphic below demonstrates a masking curve where the threshold for a pure tone in the presence of narrow band noise is centered at 1 kHz (Zwicker, 1975). For a masking tone of 100 dB SPL, the 2nd Harmonic is masked for levels below 70 dB and the 3rd Harmonic is masked for levels below 60 dB SPL.



Distortion audibility is also a function of sound duration. The ear has a finite time resolution. Clipping of a 4-millisecond tone burst reached about 10% before it was detectable, but increasing the pulse length to 20 milliseconds reduced the “just detectable” distortion point to around 0.3% (Moir).

Finally, one must keep in mind that the freedom from distortion of the sound card and of the reference signal must be greater than that of the device being measured if meaningful results are to be obtained. The purity of the signal generated is dependent upon the THD of the sound card. This is not generally an issue with regard to loudspeakers but can be if you use SW or any other software to measure distortion in other electronic equipment. RMAA can be used to measure the distortion of the sound card.

## **Total Harmonic Distortion**

*(Section Index)*

*THD or Total Harmonic Distortion* measurements can be accomplished by feeding a single high purity sine wave into the driver and then separate the sine wave from its harmonics at the output. The new frequencies are called *harmonics* because they exist at multiples of the input signal. Even Ordered harmonics are called *Even Order Distortion* while odd ordered harmonics are called *Odd Order Distortion*. Total harmonic distortion (THD) is then defined as the ratio of the geometric sum of all of the rms voltages of the harmonics to that of the fundamental component. This is usually expressed as a percentage but sometimes also expressed as a decibel.

$$\text{THD} = \sqrt{(E_{2f}^2 + E_{3f}^2 + \dots + E_{nf}^2)} / \sqrt{(E_f^2 + E_{2f}^2 + E_{3f}^2 \dots + E_{nf}^2)} \times 100$$

where

THD = percentage of total harmonic distortion

$E_f$  = amplitude of fundamental voltage

$E_{2f}$  = amplitude of second harmonic

$E_{nf}$  = amplitude of nth harmonic voltage

The audibility of the harmonic depends upon its frequency and its amplitude compared to the fundamental tone. The closer two tones are to one another, the more difficult it will be to hear one tone distinctly. In the case of harmonic distortion, the fundamental masks the 2nd harmonic component more than the 3<sup>rd</sup> harmonic and very little for the higher harmonic components. The masking threshold widens at the extremes of the audible frequency bandwidth (the fundamental will mask a wider range of harmonic frequencies) and with higher sound pressure levels as noted in the graphs above. Instruments that produce fewer low frequency harmonics (such as a flute) will sound more pure while those that produce more low frequency harmonics (bass) will sound more “warm” or “rich”. Even order harmonics coincide with octave harmonics and may have a “richer” sound due to the apparent increase in harmonics. Odd order harmonics will make sound appear to be more “grainy”.  $X_{\text{max}} = (\text{Voice Coil Length} - \text{Air Gap Height}) / 2$  and  $X_{\text{max}} * 1.15$  generally correlates with a 3% harmonic distortion. As excursion increases, 3rd harmonic distortion products increase. The voice coil drive current along with the reactive back current caused by the current induced motion of the voice coil (EMF) modulates the magnetic gap field; this tends to be asymmetric so it can cause 2nd harmonic distortion. A shorting ring at the base of the pole piece can decrease this distortion.

The ear is more sensitive to frequencies in the 1kHz-5kHz range and amplitudes over 80 dB . THD generally requires measuring harmonics at higher frequencies than the fundamental tone applied. Once these frequencies exceed roughly half that of the sound cards’ sampling rate, these tones are often filtered out or dropped leading to a misleadingly low estimate of THD. The measurement of a 10 kHz frequency fundamental for THD will yield false low estimates due to the inability to measure anything beyond the 2<sup>nd</sup> harmonic (at 20 kHz).

Something to keep in mind is that if your ear's sensitivity to a sound is 20 dB down at 20 Hz as it is at 40 Hz (see the masking graph above from Zwicker, 1975), that a harmonic of the 20 Hz signal at 40 Hz may be heard at equal sound levels if the harmonic is 20 dB down. This is not unrealistic and becomes a concern for a driver that is expected to drive frequencies that it is not adept at driving.

Since THD is going to be dependent upon which frequency range is measured, the number of harmonics calculated into THD, its level and gain conditions, these must be specified in the report. For general comparison, THD is usually measured with a fundamental tone of 90 dB SpL and at 1kHz frequency. From "Audio Specifications", Dennis Bohn [RaneNote 145, written 2000; last revised 1/03],

Of note, measuring individual harmonics accurately is difficult so often results are reported as THD + N which includes a noise factor. THD + N measures everything that comes out of the unit that isn't the pure test signal (such as harmonics, hum, noise, RFI, buzz, etc.). The remainder contains harmonics as well as random noise and other artifacts. THD + N will always be higher than THD alone. SW does not appear to measure THD + N however RMAA does appear to be capable of measuring this.



## Measuring Harmonic Distortion

[\(Section Index\)](#) Use the Harmonic Distortion Command to measure the harmonic distortion of a circuit or a speaker. This command generates different output signals depending upon which distortion option is selected in the Harmonic Distortion dialog. This command creates or replaces a dataset named mysignal.Harmonic. It may also change a number of datasets in the System folder.

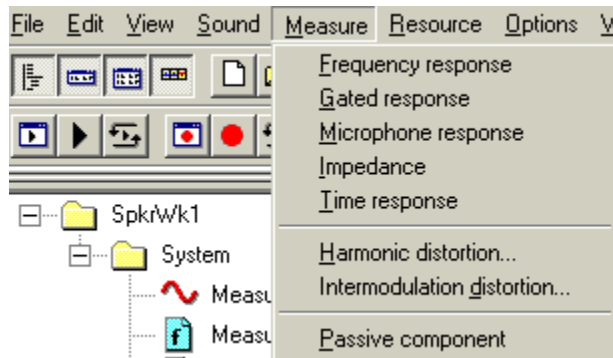
Speaker Workshop does measurements in 3 different realms - **Spectrum at 1 power / 1 frequency**, **Frequency at 1 Power**, and **Power at 1 Frequency**. **It should be noted that there is an error in the programming in this portion of the program. The distortion measurements for Frequency at 1 Power and Power at 1 Frequency vertical scales are in percentages and not in dB as is labeled.**

Spectrum at 1 power / 1 frequency provides a full spectrum view of the results of a sine wave stimulus and the resulting harmonic responses. These responses are seen as spikes along the frequency axis. This measurement will demonstrate 2<sup>nd</sup>, 3<sup>rd</sup>, and 4<sup>th</sup> order harmonics.

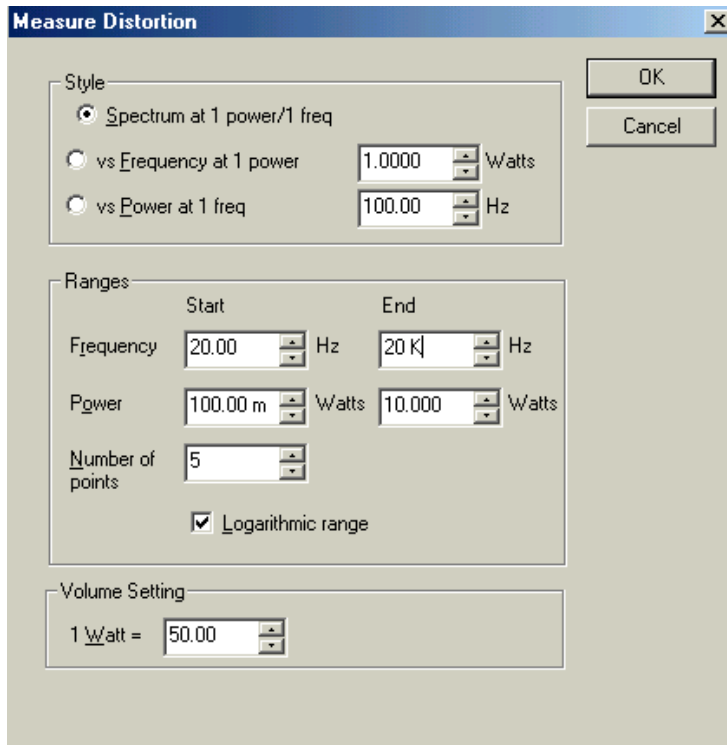
Frequency at 1 Power results in a frequency spectrum view of each harmonic measured as a function of the fundamental frequency. Here, a frequency sweep is done as the stimulus and the response will be a graph of the order of harmonic chosen as a function of the fundamental. In other words, for a 1 kHz frequency fundamental, the 2nd harmonic will be at 2 kHz but will be displayed at the 1 kHz mark under the 2nd harmonic file. The 2 kHz portion of the 2nd harmonic file is really representative of the 2nd harmonic of a 2 kHz signal. This graph demonstrates a frequency dependent distortion. By measuring a frequency sweep at one power, you can identify distortion products defined by both the frequency and power input.

Power at 1 Frequency results in a frequency spectrum of distortion that is representative of the power used to provide the stimulus and the resulting distortion. When THD starts to rise significantly at high powers, clipping is suggested while when it starts to rise at low power, amplifier linearity problems at the crossover are suggested (bias problems).

Go to Measure menu, for Signals



Fields Shown:



## 1. Style

### a. Spectrum at 1 power / 1 frequency

- i. The spectrum at one power is a single sine wave that is set in the style portion.
- ii. Select this to generate a sine wave at 1 frequency. The result is the power spectrum of the input - showing distortion.
- iii. Method:

1. Open a test signal
2. Select measure/harmonic distortion.
3. In the Style section of the dialog box, select the frequency at which the test will be performed.

*Note: If the test signal is at too high a frequency, then many harmonics may be filtered out during the process of handling the signal. This leads to misleadingly low estimates of nonlinearity.*

4. In the volume setting portion of the dialog enter the volume level that equal a 1 watt output.

Note: This volume level corresponds to the I/O volume level in the signal generator dialog and under Options/Preferences/Measurements. This volume setting will be the maximum test level.

5. In the Style section of the dialog box, select the power level at which the test will be performed.
6. Select okay. The dialog box will close and the test will be performed.

- a. During the Spectrum at 1power/1frequency measurement SW will generate a Sine wave at the selected freq. and power level.
7. A new file will be generated my test\_signal\_name.Harmonic Distortion when the test is completed.
8. Open the chart
  - a. The chart will show the harmonic distortion content starting from the selected frequency out to the limit of the cards frequency response. It appears that the chart will show the combination of both even and odd order harmonic distortion content.
  - b. You can calculate Total Harmonic Distortion with the following spreadsheet by inserting the dB level of the fundamental tone and then each harmonic.



"THD and IMD and Doppler1.xls"

#### **b. vs. Frequency at 1 Power**

- i. Select this to generate a sweep over the selected frequency range at one power. The result is a graph showing 2nd, 3rd, and more harmonic distortion.
- ii. Method
  1. Open a test signal
  2. Select Measure/Harmonic Distortion.
  3. Select vs. frequency at 1 power.
  4. In the style section, select the power level at which the test will be performed.
  5. In the Ranges section
    - a. select the frequency range over which the test will be performed.
    - b. select the number of (sample) points to be performed during the test.

***Note:** The minimum number of points should be set to at least 10 during the test. A larger number of points will give you more sample points between the start and end frequency points. This will give more accurate results, but the test is slower due to the increase in the number of play/record cycles. 10 to 20 sample points seem to work well. The frequency test interval will be determined by the number of points selected. For best accuracy, limit the frequency range to be tested while using a larger number of points across this range.*

- c. select logarithmic range to have the test performed on a log scale. Un-selected for the test to be performed on a linear scale.

*Note: At this time you may not be able to select the scale.*

6. Select okay and the test will be performed. Depending on the number of points and the speed of the computer this test will take some time. SW will generate test tones based on the number of points from the start frequency to the end frequency.
7. When the test is complete several new file will be generated under the “my test signal” folder.
  - a. These new files will appear as time data files and will be labeled:
    - i. my test signal.2<sup>nd</sup> harmonic
    - ii. my test signal.3<sup>rd</sup> harmonic
    - iii. my test signal. 4<sup>th</sup> harmonic
    - iv. my test signal.upper harmonic.

*Note: These files are actually frequency files and will appear as such sometimes during the test. They may actually be more clearly viewed as linear charts (right click on the chart then select chart properties/x axis/select linear scale). The Charts show dB on the Y-axis scale but are actually % distortion. When performing chart/properties/add these files will appear as freq. files.*

**c. vs. Power at 1 Frequency**

- i. Select this to generate a power sweep (1 frequency, increasing power). The result is a graph showing 2<sup>nd</sup>, 3<sup>rd</sup>, and more harmonic distortion.
- ii. Method
  1. Open a test signal
  2. Select Measure/Harmonic Distortion.
  3. Select vs. power at 1 frequency.
  4. In the style section, select the frequency at which the test will be performed.

*Note: If the test signal is at too high a frequency, then many harmonics may be filtered out during the process of handling the signal. This leads to misleadingly low estimates of nonlinearity.*

5. In the Ranges section
  - a. select the power range over which the test will be performed
  - b. select the number of (sample) points to be performed during the test.

**Note:** The minimum number of points should be set to at least 10 during the test. A larger number of points will give you more sample points between the start and ending power levels. This will give more accurate results, but the test is slower due to the increase in the number of play/record cycles. 10 to 20 sample points seem to work well. The test will occur at power levels that are divided by the number of points between the maximum and minimum levels. For best accuracy, limit the power range to be tested while using a larger number of points across this range.

- c. select logarithmic range to have the test performed on a log scale. Make certain that the linear scale is not selected.

**Note:** At this time this function is not performing correctly and you may not be able to select the scale.

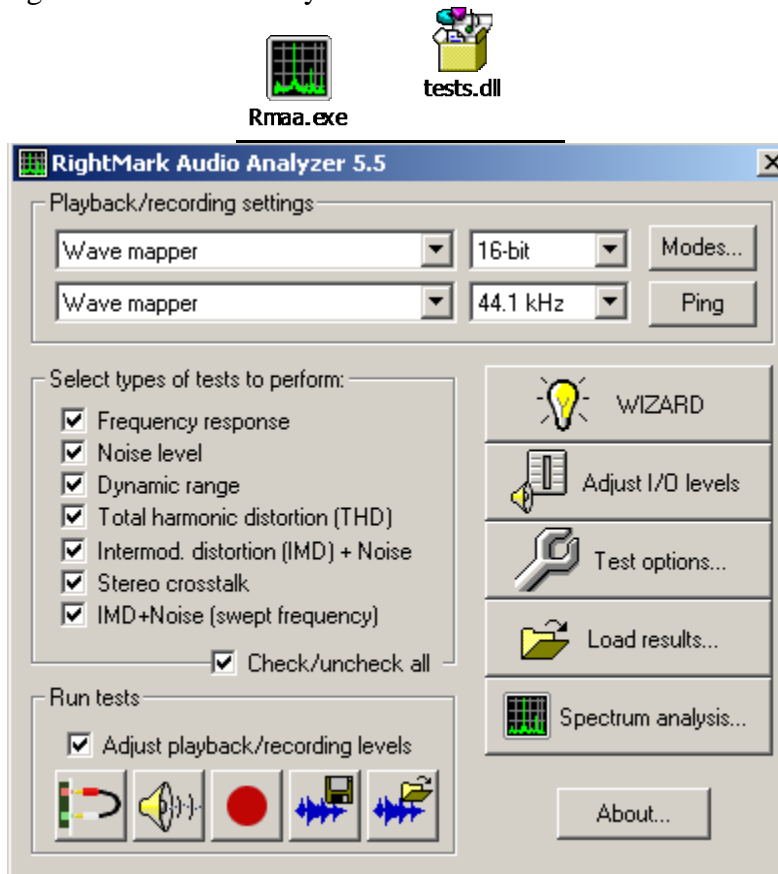
6. Select okay and the test will be performed. Depending on the number of points and the speed of the computer this test will take some time. SW will play and record the selected frequency test tone over the power levels based on the number of points from the starting watts to the ending watts.
7. When the test is complete several new file will be generated under the “my test signal” folder.
  - a. These new files will appear as time data files and will be labeled:
    - i. my test signal.2<sup>nd</sup> harmonic
    - ii. my test signal.3<sup>rd</sup> harmonic
    - iii. my test signal. 4<sup>th</sup> harmonic
    - iv. my test signal.upper harmonic.

**Note:** These files are actually frequency files and will appear as such sometimes during the test. They may actually be more clearly viewed as linear charts (right click on the chart then select chart properties/x axis/select linear scale). The Charts show dB on the Y-axis scale but are actually % distortion. When performing chart/properties/add these files will appear as freq. files.

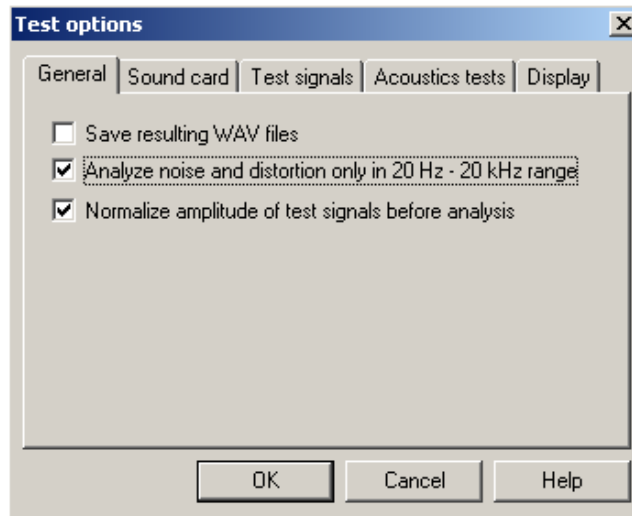
## Testing THD with RMAA

From <http://audio.rightmark.org/download.shtml>  
([Section Index](#))

6. Set up your system as you would for acoustic measurements
7. Open Right Mark Audio Analyzer

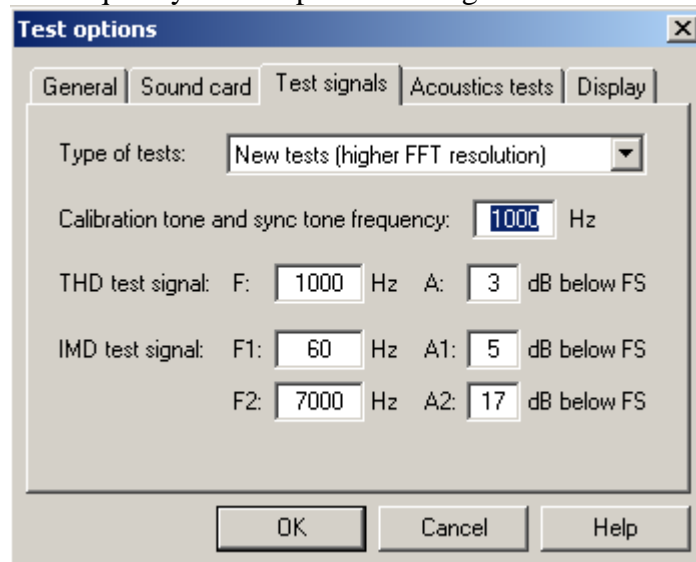


8. Select Playback/recording settings that match your card.
  - a. If you are unsure, select Modes and the program will tell you the compatible sampling rates and number of bits that your card supports.
4. Select Wizard and then Loop back Test.
5. Select Next (assuming that you are still set up as you would for acoustical testing).
6. Adjust volumes per RMAA.
7. Return to main menu and select Test Options

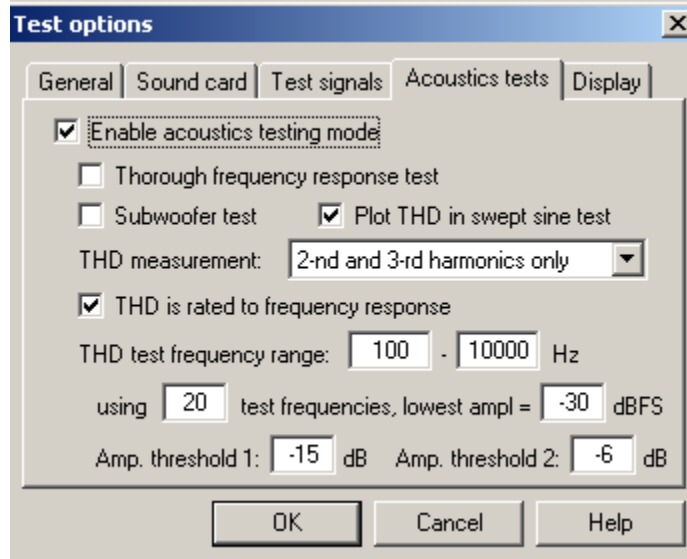


7.

- a. General Tab : Select Analyze noise and distortion only in 20 Hz to 20 kHz range.
- b. Sound Card : Leave at default settings
- c. Test Signals Set THD settings to
  - i. New Tests
  - ii. Set desired frequency of interest
  - iii. Set frequency and Amplitude settings as desired



- d. Acoustic Tests : Check enable acoustic testing mode and select THD measurement (either 2<sup>nd</sup> and 3rd harmonics or THD + Noise).



- e. Set your display as desired.
8. Run the test and observe results.



## IMD or Intermodulation Distortion

*(Section Index)*

IMD or Intermodulation Distortion measures the size of each of the new frequencies created by the distortion source. If two separate frequencies exist together in a non-linear device, sum and difference frequencies are also produced in addition to the harmonics. This is the result of a multiplication process between the two original frequencies and hence the two new frequencies are called products. If the two original frequencies are  $f_1$ ,  $f_2$  (and they are non-harmonically related) and the highest frequency is  $f_2$ , then we can expect two other components (or products) of  $(f_1+f_2)$  and  $(f_2-f_1)$ . Since there are harmonics of  $f_1$  and  $f_2$ , then there will be sum and difference products between all of the harmonics and the fundamentals and between each other. If there are more than two fundamental frequencies there will be many other products, as well. The new frequencies are called *beats*. The beats at the sum and differences frequency are called *Even Order Distortion* and are caused by errors that are not symmetrical. *Odd Order Distortion* is caused by symmetrical (or multiples of the differences of the two tones in the input signal). Intermodulation products are infinite in number; however, we are only really interested in those of such a level that they might deteriorate the quality of our signal beyond an acceptable level. Since IM does have to distinguish harmonics (or multiples) of the signals at frequencies that are many times that of the input tones, it good for measuring non linearity at high frequencies. The weakness is that it sometimes does not clearly distinguish symmetrical vs. non-symmetrical error. The IM distortion audibility increases at low frequencies.

We are only interested in those harmonics and intermodulation components that fall closest to the fundamental frequency and within the frequency range of our speakers. As can be seen from the following table (taken from Intermodulation Performance and Measurement of Intermodulation Components by Lloyd Butler published in "Amateur Radio" August 1997), the third order components are the closest and also usually the highest in amplitude. Because of this, they are usually the products that are most commonly reported.

1st Order	$f_1$ ,	$f_2$	100 kHz	101 kHz
2nd Order	$f_1+f_2$ ,	$f_2-f_1$	201 kHz	1 kHz
3rd Order	$2f_1-f_2$ ,	$2f_2-f_1$	99 kHz	102 kHz
	$2f_1+f_2$ ,	$2f_2+f_1$	301 kHz	302 kHz
4th Order	$2f_2+2f_1$ ,	$2f_2-2f_1$	402 kHz	2 kHz
5th Order	$3f_1-2f_2$ ,	$3f_2-2f_1$	98 kHz	103 kHz
	$3f_1+2f_2$ ,	$3f_2+2f_1$	502 kHz	503 kHz
Etc.				

3rd Order	$2f_1-f_2$ ,	$2f_2-f_1$	99 kHz	102 kHz
5th Order	$3f_1-2f_2$ ,	$3f_2-2f_1$	98 kHz	103 kHz
7th Order	$4f_1-3f_2$ ,	$4f_2-3f_1$	97 kHz	104 kHz
9th Order	$5f_1-4f_2$ ,	$5f_2-4f_1$	96 kHz	105 kHz
Etc.				

Dennis Bohn gives the clearest explanation of the two common forms of measurement of IMD that I have seen in "Audio Specifications", Dennis Bohn [RaneNote 145, written 2000; last revised 1/03] at <http://www.rane.com/note145.html>. The following is an excerpt from that article:

### ***IMD -- SMPTE. Intermodulation Distortion -- SMPTE Method***

*What is tested? A more meaningful test than THD, intermodulation distortion gives a measure of distortion products not harmonically related to the pure signal. This is important since these artifacts make music sound harsh and unpleasant.*

*Intermodulation distortion testing was first adopted in the U.S. as a practical procedure in the motion picture industry in 1939 by the Society of Motion Picture Engineers (SMPE -- no "T" [television] yet) and made into a standard in 1941.*

*How is it measured? The test signal is a low frequency (60 Hz) and a non-harmonically related high frequency (7 kHz) tone, summed together in a 4:1 (12 dB difference with the lower frequency tone being louder) amplitude ratio. (Other frequencies and amplitude ratios are used; for example, DIN favors 250 Hz & 8 kHz.) This signal is applied to the unit, and the output signal is examined for modulation of the upper frequency by the low frequency tone (the lower frequency tone is filtered out to examine the higher frequency tone). As with harmonic distortion measurement, this is done with a spectrum analyzer or a dedicated intermodulation distortion analyzer. The modulation components of the upper signal appear as sidebands spaced at multiples of the lower frequency tone. The amplitudes of the sidebands are rms summed and expressed as a percentage of the upper frequency level.*

*[Noise has little effect on SMPTE measurements because the test uses a low pass filter that sets the measurement bandwidth, thus restricting noise components; therefore there is no need for an "IM+N" test.]*

*Required Conditions. SMPTE specifies this test use 60 Hz and 7 kHz combined in a 12 dB ratio (4:1) and that the peak value of the signal be stated along with the results. Strictly speaking, all that needs stating is "SMPTE IM" and the peak value used. However, measuring the peak value is difficult. Alternatively, a common method is to set the low frequency tone (60 Hz) for +4 dBu and then mixing the 7 kHz tone at a value of -8 dBu (12 dB less).*

*Correct: IMD (SMPTE) less than 0.01%, 60Hz/7kHz, 4:1, +4 dBu*

*Wrong: IMD less than 0.01%*

*Note that Speaker Workshop does not appear to use this method.*

### ***IMD -- ITU-R (CCIF). Intermodulation Distortion -- ITU-R Method***

*What is tested? This tests for non-harmonic nonlinearities, using two equal amplitude, closely spaced, high frequency tones, and looking for beat frequencies between them. Use of beat frequencies for distortion detection dates back to work first documented in Germany in 1929, but was not considered a standard until 1937, when the CCIF (International Telephonic Consultative Committee) recommend the test. [This test is often mistakenly referred to as the CCIR method (as opposed to the CCIF method). A mistake compounded by the many correct audio references to the CCIR 468 weighting filter.] Ultimately, the CCIF became the radio communications sector (ITU-R) of the ITU (International Telecommunications Union), therefore the test is now known as the IMD (ITU-R). (Note: this type of IMD testing is also called “Close Thirds” and has better immunity to echo contamination however it is more demanding on the frequency resolution ability of the system. It is advisable to keep the stimulus spacing to at least 10x that of the resolution. For example, at a sample rate of 48 kHz with a sample size of 8192, the frequency resolution is 48000/8192 or 6 Hz so you would need to space the sample frequencies at least 60 Hz apart).*

*How is it measured? The common test signal is a pair of equal amplitude tones spaced 1 kHz apart. Nonlinearity in the unit causes intermodulation products between the two signals. These are found by subtracting the two tones to find the first location at 1 kHz, then subtracting the second tone from twice the first tone, and then turning around and subtracting the first tone from twice the second, and so on. Usually only the first two or three components are measured, but for the oft-seen case of 19 kHz and 20 kHz, only the 1 kHz component is measured.*

*Required Conditions. Many variations exist for this test. Therefore, the manufacturer needs to clearly spell out the two frequencies used, and their level. The ratio is understood to be 1:1.*

*Correct: IMD (ITU-R) less than 0.01%, 19 kHz/20 kHz, 1:1, +4 dBu*

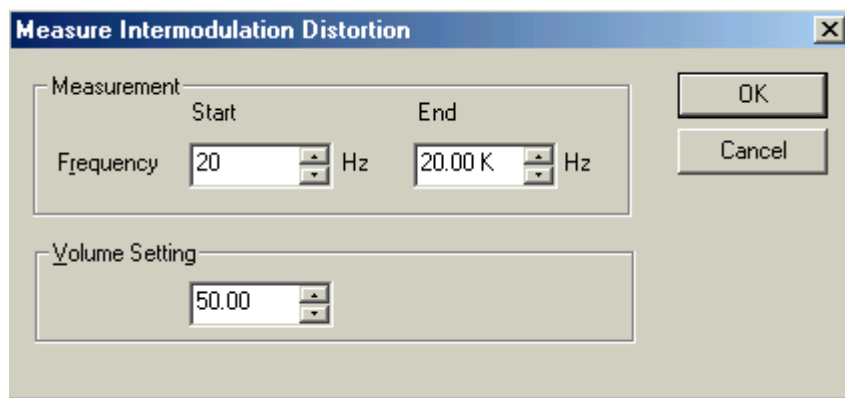
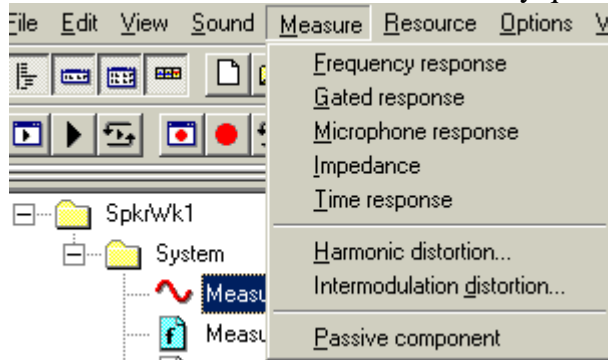
*Wrong: IMD less than 0.01%*

Speaker Workshop utilizes the ITU-R (CCIF) method of measuring IMD.

## Measuring Intermodulation Distortion

[\(Section Index\)](#)

Use this command to measure the intermodulation distortion of a circuit or a speaker. It appears that Speaker Workshop uses the ITU-R (CCIF) method. This test uses a sine wave as its output signals regardless of which distortion option is selected in the Intermodulation Distortion dialog. It appears that the accuracy is best if the starting and ending frequencies are closely spaced. This may be due to noise issues. The measurement environment has to be very quiet or it will show noise in the lower ranges.



This command creates or replaces a dataset named `mysignal.Intermod`. It may also change a number of datasets in the System folder.

### Method:

1. Open a test signal
2. Select measure/ Intermodulation Distortion.
3. In the Measure Intermodulation Distortion dialog select the starting and ending frequencies to be tested. Note: The frequencies selected should be closely spaced for best results, i.e.; 10K & 11K, 200 & 300, 15K & 16K, 2K & 2.5K, etc.
4. Select the power level at which the test is to be performed.
5. Select okay and the test will be performed. SW will generate a test tone, which is a combination of the start frequency tone and the end frequency tone at the selected power level.

6. When the test is complete, SW will create a file labeled “my test signal name”. Intermodulation Distortion. The chart will show the % level of intermodulation distortion up to the start frequency range.

*Note: The Chart shows dB on the Y-axis scale, which is actually % intermodulation distortion.*

*A spreadsheet to calculate IMD may be of some assistance can be used: ([THD, IMD and Doppler Distortion](#))*

*You will note that it can also assist in calculating Doppler Distortion in a driver. Doppler distortion is described more fully in the next section.*

## **DOPPLER DISTORTION**

*(Section Index)*

*From Siegfried Linkwitz:*

Doppler distortion occurs when two tones of a largely different frequency are radiated from the same driver. It is a linear form of distortion (which means that the magnitude is dependent upon the amplitude of the lower frequency tone). While the diaphragm is moving towards the listener, sound is compressed into a smaller distance so the frequency increases. Sound undergoes rarefaction (decompression) while the diaphragm is moving away so the effective frequency decreases. The lower frequency modulates the upper frequency since it's displacement greater than the upper frequency though the effect that the upper frequency has on the lower frequency will increase as the frequencies are closer together; however, as frequencies are closer together the Doppler effect is decreased because of the similarity in the amount of cone movement that they cause. The presence of Doppler distortion may be another reason to crossover at lower frequencies as it will decrease the maximum span between low and high frequencies and the wavelength ratios are much greater for the low pass frequencies than they are for the high pass frequencies.(assuming constant sound pressure levels, the diaphragm displacement decreases with the square of the frequency). Since the lower frequency is the major determinant regarding Doppler effects, music with more bass in it will result in more Doppler distortion.

The audibility of Doppler distortion depends upon the

1. Frequency Modulation threshold of hearing
  - a. The FM threshold is about .35% for frequencies above 500 Hz (so at 500 Hz, it would be  $.0035 * 500$  or 1.75 Hz). .
2. Masking by the lower tone
  - a. Masking is dependent upon the relative levels and the frequency separation of the two tones. The 3 dB bandwidth (BW) is equal to  $2.2/RT60$  [sec] while the slope of the filter curve is about  $6 * \text{dB}/(\text{BW})$
3. Frequency modulation to amplitude modulation conversion due to room resonance.
  - a. The Doppler distortion of a driver is observed on a Spectrum Analyzer by observing the amplitude of the frequency or phase modulation sidebands ( $A_s$ ) relative to the carrier ( $A_H$ ).  $A_s / A_H = 0.5 * \text{change in higher frequency} / \text{lower frequency} = \text{PI} * \text{higher frequency} * \text{displacement of driver} / c$ .
  - b. The sidebands occur at the same frequencies as the 2<sup>nd</sup> order intermodulation products of the two frequencies. The difference between Doppler products and Intermodulation products is that the Doppler products change by an equal amount of dB to changes in signal levels whereas 2<sup>nd</sup> order IMD sidebands change by twice as many dB.
  - c. If the sidebands are not equal to one another, it is due to the FM Doppler distortion as IMD products are in phase with one another while FM sidebands may be out of phase.

## **IMAGING INDEX**

*(Section Index)*

Imaging index developed by William Duddleston in *An Engineer's Perspective* (<http://www.legacy-audio.com/engineer/imaging-index.pdf>) : At a given point, measure dB from the Left and then from the Right Speakers. Do this by

1. orienting the microphone normal to the plane of the speakers at the listener's seated height
2. Set the volume to approximately 85 dB
3. play a pink noise signal (monophonic) through both loudspeakers with the right connected in reverse polarity
4. Sweep the microphone to about 18 inches on either side of the listeners position
5. Note the minimum position and mark this with tape
6. Reconnect the Right Speaker in polarity and re-measure the SpL (which should be higher)
7. The Imaging Index is In Phase Acoustic Sum – Out of Phase Acoustic Sum.
8. A good set up will provide an Index of about 6 while a poor set up will be closer to 0.
9. If your system is incorrectly polarized, the index will be a negative number