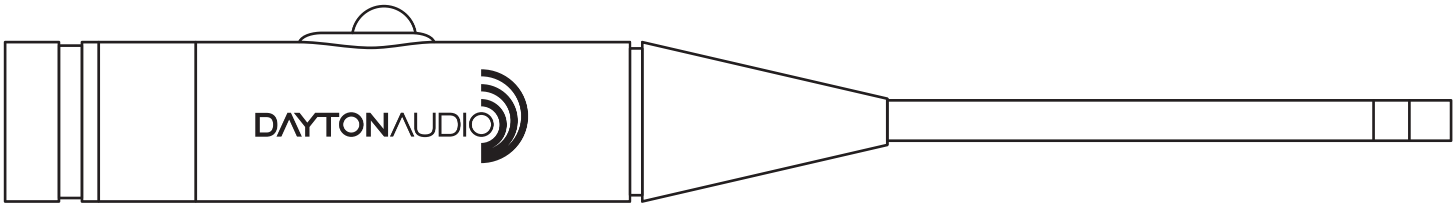
OmniMic
OmniMic 40k
User Manual

OmniMic software version 6 for OmniMic and OmniMic40k



OmniMic is simple to use.

While supporting sophisticated measurements with extensive built-in Help, you can still be up and making measurements in *minutes*.

Here are the initial steps:

- **Plug the OmniMic into a USB port** of your Windows computer.

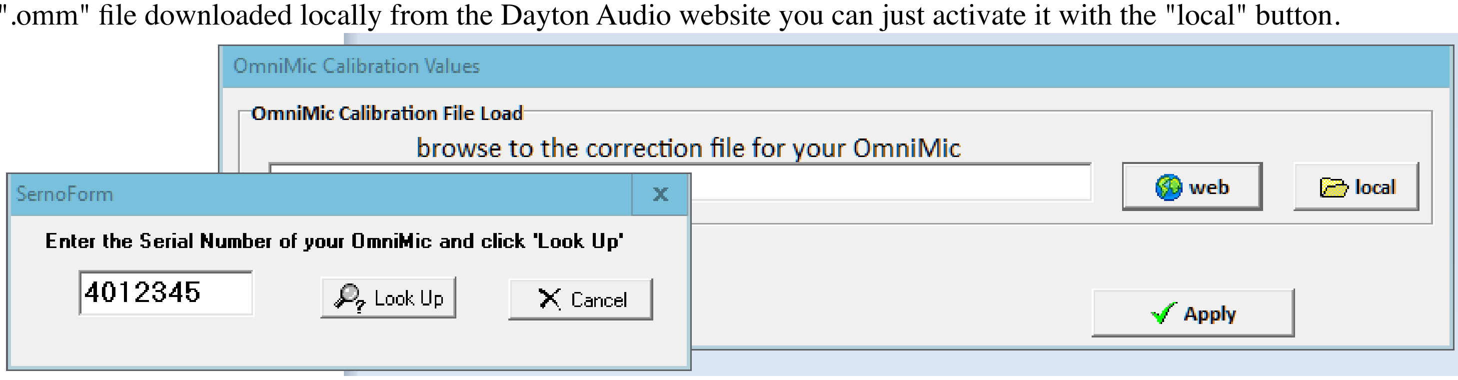
FOR FIRST USE ON THIS COMPUTER

(with *original OmniMic hardware* *only* -- *not needed with OmniMic40k*):

>> If necessary, get to the Desktop screen by clicking on the "Windows" button of the keyboard

>> Right-click on the small loudspeaker icon (usually on the bottom right corner of your screen).
>> In the menu that pops up, select "Recording Devices". A form will pop up, click on the "OmniMic" entry, and then on the "Properties" button.
>> Click on the "Advanced" tab and for "Default Format", select 2-channel **"DVD Quality" (48kHz)**. Then click "Ok" (twice).
>> *You should repeat all the steps above with the OmniMic plugged into each USB ports you plan to use with OmniMic on your computer.*

- **Select "Config>MICROPHONE serno"** from the menu at the top of the OmniMic screen. If your computer is connected to Web, OmniMic software can download the associated calibration file for you. If you already have the ".omm" file downloaded locally from the Dayton Audio website you can just activate it with the "local" button.



USING OMNIMIC

- Choose the desired measurement type tabsheet by clicking on its labeled tab on the OmniMic screen.
- If the title bar at the top specifies test signal tracks, play the file for that track through your audio system. Track files can be downloaded from the www.DaytonAudio.com website or can be easily generated from within OmniMic software. Or, if your computer soundcard is connected to your audio system, OmniMic can play the required test signal directly.
- Position the OmniMic to pick up the sound. The graphs and meters will graph the measurement.
- Adjust the display panel to the desired size and format the graphs as you desire.

what's new in OmniMic?

For help and information about each measurement type, select:

SPL Meter/Spectrum

Frequency Response

Oscilloscope

Harmonic Distortion

Reverberation

Bass Decay

...or read any of the Help topics listed in the bar to the left, including:

How to...

Making test signal WAV files

Troubleshooting

Frequently Asked Questions

What's New

New in software version 6

The OmniMic software version 6+ also supports the original OmniMic microphones hardware functions

- "shift-click" (rather than right-click) is now used to select the beginning point of impulse responses for frequency response measurements
- All audio test signal generated in the software can be easily saved to WAV files for playback on external audio/visual devices.
- OmniMic generated sweep signals (for Frequency Response and Distortion) can be bandlimited in software and levels can be finely adjusted. A 55Hz tone can be easily generated for use with an AC DVM to calibrate for desired voltage drive levels.
- support for PhotoLink for drive level adjustment for all OmniMic and OmniMic40k microphones
- support for photosync optical loopback (OmniMic40k microphones only, and a PhotoLink sender) for true time of flight and phase measurements
- Automated impulse response window edge determination.
- support for 96ksps sampling (OmniMic40k microphones only)
- drive level management for "dBSPL/m" measurements of loudspeakers and drivers
- on-trace persistent markers on frequency-domain graphs. Just Click on the curve at the desired frequency and press one of the number keys 1-9. Key 0 removes all persistent markers
- harmonic distortion graphs displays curves as percent (%) distortion (use the "show as %" checkbox on the graph)
- Measurement of Spectral Contamination intermodulation distortion
- Measurement of loudspeaker Compression
- various bugfixes and graphics enhancements

Also see: OmniMic40k hardware improvements and changes over Original OmniMic

OmniMic Adjustments

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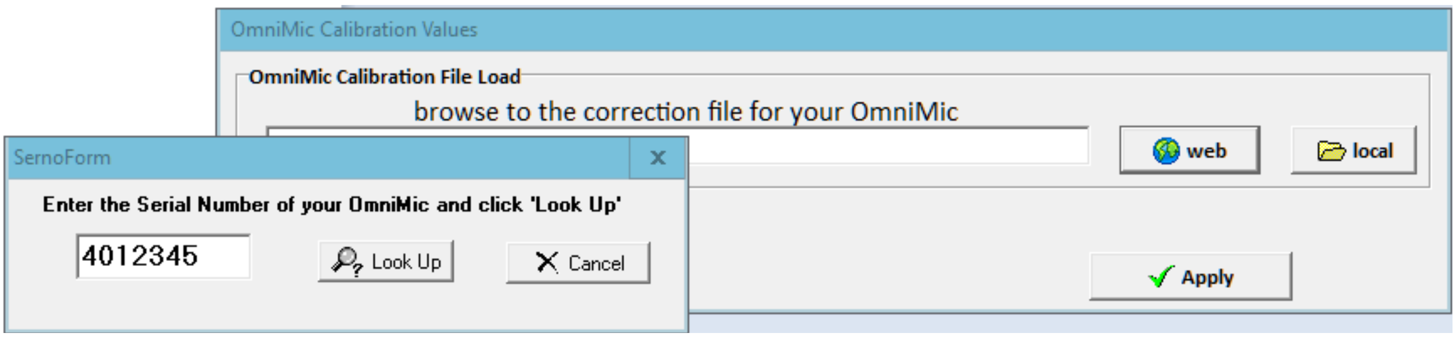
The accuracy of each individual OmniMic is enhanced in software to account for frequency and sensitivity variations as determined during the production calibration process. This is done using data in an individual file corresponding to each microphone serial number. Omnimic hardware with serial numbers starting with "40" are Omnimic40k units.

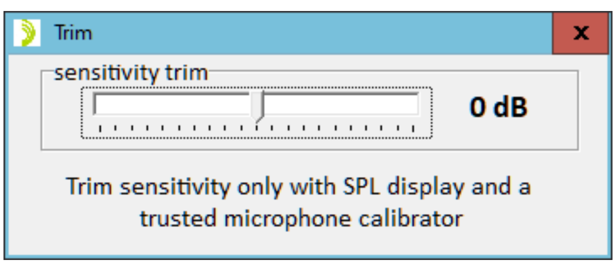
Calibration Files:
At installation: If your computer is connected to the interent, click on the main menu "**Config>MICROPHONE serno**" within the OmniMic software, then click the "Web" button, and enter your serial number and Omnimic will do the rest.

Calibration data can also be manually downloaded from the Dayton Audio website, according to the serial number that is printed on the microphone. The file (which has an extension type ".omm") should be saved onto a folder on any computer that the microphone will be used with.
To activate that calibration file for Omnimic software to use, select "**Config>MICROPHONE serno**" from the menu at the top of the screen. Click the button showing the yellow folder and browse to where you stored your ".omm" file. Then click "Apply".



Sensitivity Readjustment
Over time, the sensitivity of microphones could possibly drift to some degree. Provided reasonable care is taken to avoid exposure to high temperatures or humidity, their frequency response shapes typically remain stable.

The overall sensitivity of the microphone can be checked with a trusted microphone calibrator and body adaptor, and the OmniMic **SPL/Spectrum** display. If it is found to be in error, the sensitivity can be adjusted by up to several decibels with the control that appears when you select "**Config>Adjust**" from the menu at the top of the screen.



Using Multiple Microphones
The OmniMic software is able to work with several OmniMic microphones, though only with one at a time. Also only one instance of the OmniMic software can be properly run on a computer at a time. But an OmniMic microphone cable can be removed from the USB port of the computer while the software is running, and a different OmniMic can then be plugged in to go on working. There will be a short pause while the program recovers operation with the microphone and soundcard hardware.

But different OmniMics use different calibration files. So there is now an option in the Config menu for a "Microphone selector". This is a form that contains buttons and title blocks for up to 5 OmniMics. To set up for each, click on its title block (the round button at its left will then turn red) and then click on the "configure" menu in the selector form. You'll be prompted for a name or description of the mic, which could be something like "On Axis", "30 degrees", "nearfield" or the like. Then you will be given a file box for finding the calibration file for this microphone (which needs to be already on your computer).

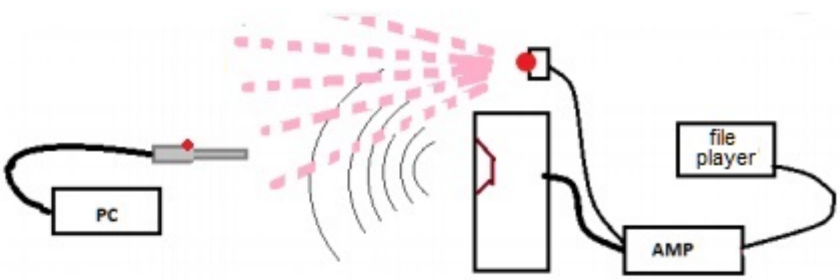
After this is setup, you can quickly change between mics. Unplug the mic previously being used and plug in the new one. Then click on the name or button for that mic on the selector form and its calibration will be loaded for new measurements from that microphone.

User IDs
You can assign your initials, name, email address, etc. to the "User ID". This will be attached to some files gnerated by OmniMic to identify who created them.

OmniMic40k Microphone Enhancements

The OmniMic40k compared to OmniMic V2:

- OmniMic40k, with the PhotoLink box, provides "time of flight" or "loopback" delay responses. There is an infrared photodetector on the body of the OmniMic40k which detects the timing of stimulus signals from applied from an audio power amplifier.



- Uses 96ksps sampling and measures out to 40 kHz *-- double the bandwidth of the OmniMic V2. Provides for harmonic distortion measurements (second order) with applied frequencies to 20kHz.
- With internal high voltage circuitry, it can handle much higher SPL levels - up to 145dBSPL or higher using its 24dB attenuator switch. Allows for high linearity testing at extremely high sound pressure levels, for prosound, nearfield distortion or compression testing.
- Smaller capsule and narrow tube diameter, results in a more "omni" response at the highest frequencies.
- Simpler USB integration with Windows -- the proper hardware configuration is automatic, simply plug into a USB slot.
- Rugged, stainless steel body and tube.

***Note:** CD audio and MP3 files and most TOSLINK optical interfaces lack adequate bandwidth to enable OmniMic40k frequency response or distortion measurements beyond about 20kHz-- flac or WAV files must be played through a 96ksps file player, or generated from the sound output of the computer running OmniMic (see the "Config->Play from Soundcard" menu option). A Dayton Audio `DAC01 USB converter` device is a good option.

48kHz tracks, DVDs, or CDs ***are*** still usable with OmniMic40k, but useful data only up to about 22kHz maximum frequency can be obtained.

Operating to 40kHz

OmniMic40k is capable of measuring at frequencies up to 40kHz, a full octave above the generally recognized limit of normal human hearing. But most audio hardware isn't capable of this, so here are some tips on successful operation to these frequencies:

Distortion Measurements:
While many hardware devices do not operate or operate predictably to 40kHz, they can still generate distortion products to these higher frequencies. Measuring harmonic distortion with OmniMic40k still has the advantage of being able to reveal second order (H2) distortion products with applied frequencies to 20kHz, third order (H3) products with to 13.3kHz, forth order (H4) with up to 10kHz and fifth order (H5) at up to 8kHz.

With non-40k OmniMics (V2), harmonic products can be detected to frequencies of half these values (i.e., H2 with up to 10kHz, etc).

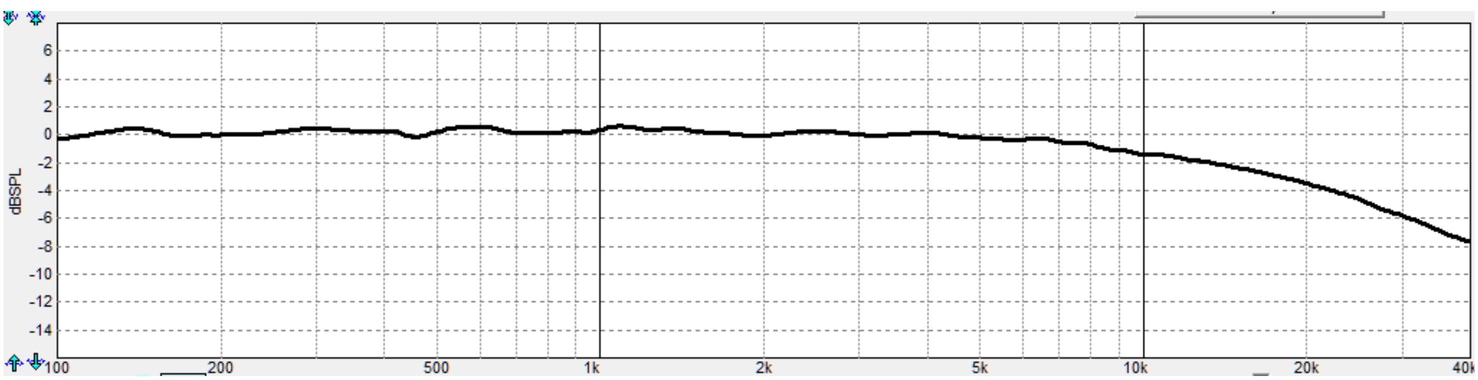
Frequency Responses with Power Amplifiers:
Many class A or class AB power amplifie rs are usable to 40kHz, though many don't list that capability in their specifications. Still, even when usable to 40kHz, many of them are already rolling off significantly in their frequency responses. Frequency response files are provided in the OmniMic40k program installation (see the C:\Users\Public\OmniMic folder) for two power amplifiers from Dayton Audio, the APA150 and the A400. These files can be used to correct measurments made with these power amplifiers by using the Menus: Main Curve->Normalize, then select the response file for the power amplifier being used.

In most cases, class D type power amplifiers are not adequate for use in measuring loudspeakers to 40kHz, as their output filters cause significant effects at such high frequencies, and the amplifer response will often vary according the loudspeaker impedances at supersonic frequencies. There are exceptions of course and some class D amplifiers work pretty well for this. But if you start seeing strange results above 20kHz, the amplifier's behavior should be the first thing you suspect of being the issue.

Frequency Responses with DACs and Digital playback interfaces:
Modern DACS and interfaces typically have no problem with reproducing beyond 20kHz, using the 96ksps or higher rates. But choose a technically good DAC, rather than an exotic "audiophile" DAC, to avoid strange behaviours and frequency response abberations.

Loudspeakers beyond 20kHz
Relatively few loudspeakers have significant measured output up to 40kHz, and it is arguable whether there is any reason in terms of music reproduction for them to support it. But for those speakers that can reproduce these frequencies usefully, keep in mind that the output from speakers will tend to be very directional ("beamy"), and careful aiming of the tweeter at the microphone will likely be needed for reproducable results.

Like all microphones, the OmniMic40k unit itself also begins to be more directional with sensitivty reduced in the higest octave (by about 3.5dB at 20kHz) when the microphone is pointed perpendicularly to the sound source. This is the plot of the response to the OmniMic 40k at 90 deg, as compared to at 0 deg on axis.



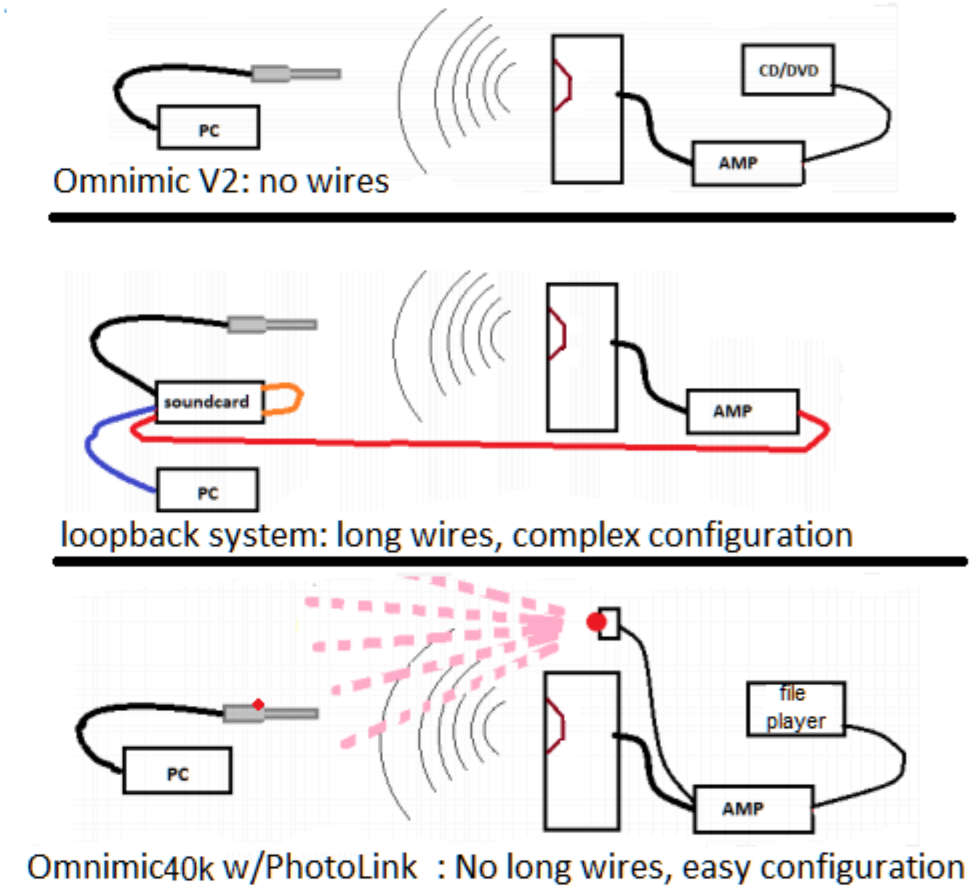
Using the OmniMic PhotoLink device

The PhotoLink has two main functions, time synchronization and setting of nominal drive levels.

The first function is to transmit a timing Optical Syncrhonization signal via modulated infrared light to an OmniMic40k microphone. Note that "Photosync" refers to the technique of syncronizing timeing via a light signal, while "PhotoLink" refers to the device that is used to accomplish this. It is primarily intended for measuring drivers for speaker design with true phase delay included and to account for time-of-flight effects of sound through the air to the microphone. In multi-way loudspeakers, the way that various drivers interract is affected by their relative distances from the listener/microphone and the "starting point" of drivers varies with their construction. When making FRD frequency response data files with optical syncronization for crossover design, the included delay should be compensated per the measured distance of the microphone tip to the mounting baffle surface the drivers are on. Also, the measurements should be made in "dB sensitivity mode" to account for driver sensitivities (see below). This will keep driver measurements compatible with others done the same way to allow you to investigate different combinations with a crossover simulator.

Optical synchronization can also be useful when measuring subwoofers should the speaker output lack sufficient higher frequency energy for OmniMic's automatic acoustic synch to align with the stimulus. In such cases, the PhotoLink device should be driven by full bandwidth signal through a separate amplifier rather than by the signal that is presented to the subwoofer terminals.

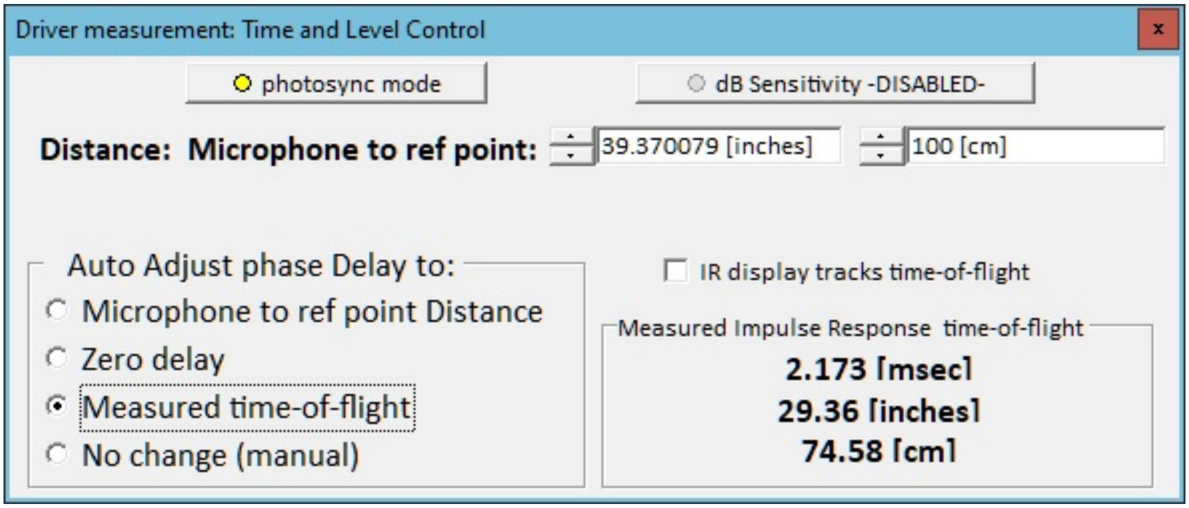
Optical syncrhonization (or "**photosync`**") is similar to the wired "loopback" technique used with some measurements systems, except Photosync still allows for OmniMic's wireless advantage that lets you quickly and conveniently measure without stretching cables around the room. Optical synch requires an OmniMic40k type microphone and PhotoLink device.



To use optical syncrhonization, the leads from the PhotoLink should be clipped to the speaker terminals or the amplifier output, with the PhotoLink placed and oriented so that its infrared LED can be "seen" by the black sensor that is on the OmniMic40k body. This can work up to about 20 feet away, and usually even just being exposed to the reflection of the infrared signal off of a nearby wall or ceiling will be enough for the OmniMic40k to syncronize with.

The OmniMic version 6 software can be used without photosync or with. Photosync is only used when in Frequency Response mode with "Sine Sweep" stimulation, and when the photosync button on the OmniMic Frequency Response screen is enabled. Photosync is not required for most measurements, but can simplify obtaining FRD files to be used for loudspeaker design and for more accurate polar data collection. Frequency response measurements (such as for room EQ or completed speaker evaluation) that don't require absolute phase delay information can and usually should be made without Photosync.

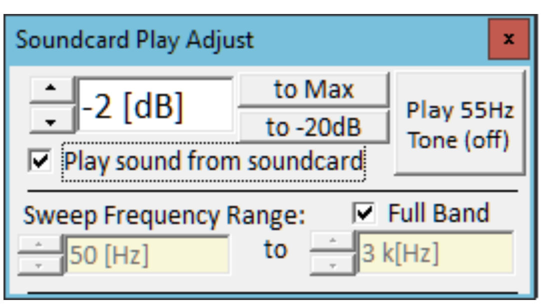
When using optical syncrhonization, you can select how this affects the applied delay to the viewed response using the Time and Level Control. For example, an impulse response that includes acoustical 'time of flight' would naturally show effects of significant delay, which could cause the displayed phase response to be difficult to interpret. Taking out delay according to the time of flight or to known distance from the speaker can be used to account for that.



The second Photolink function is to indicate to the user when the voltage level of a sweep signal applied to a speaker or driver is at the standard **2.83 Vrms** for proper sensitivity measurement of a loudspeaker or for setting levels for a compression test. This second function **will work with either OmniMic or OmniMic40k** hardware. Remember that only sweep type signals (not noise or tones) will giver proper indication of drive level.

Sensitivity measured to a common standard (equivalent to 2.83Vrms drive with SPL measured at 1m distance) is a requirement when maintaining or sharing a library of driver curves for use with a crossover simulation program like Xsim. The scale of graphs measured this way can be displayed as "dBSPL/1m" rather than "dBSPL". OmniMic software can also account for the effects of distances other than 1m from the loudspeaker, assuming a theoretical falloff rate of 6dB per each doubling of distance.

The PhotoLink has two visible LEDs, one **GREEN**, one **RED**. The **GREEN** lights when there is **ENOUGH** signal level (for both target level and for Photosync function), the **RED** lights when there is **TOO MUCH** signal above the target SPL level (but Photosync will still work with red). When setting sweep levels for 2.83Vrms (equivalent to 1W at 8 ohms or 2W at 4 ohms), aim to adjust to where the **GREEN LED blinks** with each sweep but the **RED does NOT**. When using the computer's sound output or DAC to generate the sweep signal, you can use the "Soundcard Output Level" from the Config menu to fine adjust the level.



The Sweep Frequency Range should be "full band" and the signal at the loudspeaker terminals must be from a flat un-equalized setting when using the PhotoLink device for signal level setting. Equalization or bandlimiting of the sweep signal seen by the PhotoLink will result in incorrect determination of the applied signal level.

Further, **only the OmniMic "short sweep" test signal** (as used for frequency response measurements) **should be used for setting the signal level with the PhotoLink.** A continuous sine-wave or noise type signal **cannot** accurately be used for this. However, once levels are set using the Short Sweep signal, the same level will be correct if you change to one of the other test tabs such as Distortion, SPL, or Reverb. You can limit the Sweep Frequency Range after setting proper level with the "Soundcard Output Level" first set to "full band", and the levels over the specified range will be at the determined level.

For optical time synchronization, the PhotoLink can also be operated at levels well above where the RED LED lights, up to about 350W/8 ohms.

What if you want to set to a level other than 2.83Vrms?

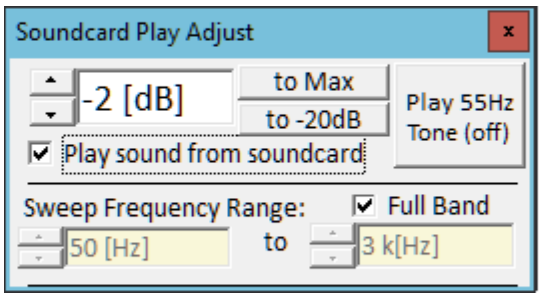
If you, **for example**, wanted to set to a level that is 3dB higher than 2.83Vrms:
You would first set the Soundcard Output Level to a lower value (-3dB or lower).
Adjust the system (amplifier or Windows sound control level) and use the PhotoLink device so the sweep shows the green but, not the red, LEDs.
Then set the Soundcard Output Level to 3dB higher than you had it. The output voltage would then be set to 2.83Vrms+3dB, which is 4Vrms.

Soundcard Play Adjust tool

OmniMic software can generate needed audio test signals using the computer's soundcard, or can generate WAV files for playback from external audio/visual devices. An OmniMic or OmniMic40k must be connected for this.

When using the computer to generate test signals, you can use the Soundcard Play Adjust tool to control or fine tune the relative output level and modify the effective sweep frequency range of measurement stimulus signals.

The tool can be brought up using the "Config --> Soundcard Output Level" menu.



The Sweep Frequency Range can be restricted by unchecking "Full Band" and changing the start and stop frequencies. Note that only the levels of different frequency ranges and not the actual sweep range is changed. Beyond the selected frequency range, output at frequencies below the range is attenuated steeply. At frequencies above the indicated range, output is attenuated by 12dB so that the software can still frame the timing of the measured signal sequence.

Important: Only when the Sweep Frequency Range "Full Band" box is checked should a PhotoLink device should be used for setting stimulus levels, as the PhotoLink requres a full, flat, unequalized sweep to correctly identify a 2.83Vrms operating level.

An expanded implementation of the Soundcard Play Adjust tool is also used for setting up and beginning comprssion tests.

55Hz Tone Activation

A button at top left of the tool panel marked "Play 55Hz Tone" is also provided. This can be used to temporarily change the software generated computer output to a steady 55Hz tone for use as an alternate way to set stimulus test levels. ***DO NOT USE THIS WHILE UNPROTECTED TWEETERS ARE DIRECTLY CONNECTED TO THE POWER AMPLIFIER!***

To use the 55Hz level calibration tone, connect a voltmeter capable of measuring AC RMS voltage levels across your power amplifier's output terminals, disconnect your speaker and configure the meter to measure 2.83Vrms (1W at 8 ohms) or to your desired stimulus power level. Most digital voltmeters (DVMs) can measure 55Hz AC sine wave voltages with acceptable accuracy. Then adjust the power amplifier's gain control and/or the software control at the top left of the Soundcard Play Adjust tool panel to provide the desired voltage level. When the 55Hz tone is removed (by clicking the button again or by closing the Soundcard Play Adjust tool) and the speaker is reconnected, sweep tones Omnimic plays from the computer soundcard will remain at the arranged voltage level.

*Note that this will only be true when the power amplifier's frequency response is flat and no external equalization is being applied.
*Also, note that this applies only to sweep tones, not to noise or other signals, for which voltage output level is not simply defined.

Adjusting Input Gain and Auto-Level

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For best results, the input sensing gain of OmniMic should be adjusted approximately for the sound level. See Dynamic Range Considerations. The input gain slider control is at the top right of the OmniMic screen.

- The slider should be set so that sensitivity is not too low (which could increase noise) and not too high (which could overload OmniMic's electronics). OmniMic40k units also have a physical switch which can be used to attenuate the sensitivity by 24dB. The attenuation is in effect when the switch is set toward the back of the microphone. **Be sure to also check the box marked Attenuator** at the top right of the program's form so OmniMic software can know to compensate readings for the applied attenuation.
- When a sound level is sensed that is too high for the OmniMic, a warning message will appear -- if you see this, move the slider to the left and/or use the switch (OmniMic40k) to reduce the gain until the message disappears. With OmniMic40k, the sensitivity slider should normally be all the way toward the right unless the "Input Overdrive" warning message appears. **BE CAREFUL OF VERY HIGH SOUND LEVELS WHICH CAN DAMAGE YOUR HEARING -- USE HEARING PROTECTION WHEN TESTING AT HIGH LEVELS.**
- With the orignal OmniMic, the sensitivity can instead set to adjust automatically by the OmniMic software if you temporarily put a check in the small box next to the slider. Expose the OmniMic to the sound level that will be used for several seconds, and the slider should adjust to a compatible level. OmniMic40k hardware is more tolerant of high levels and uses much fewer slider steps avoiding need for automatic level.

It is best NOT to leave the auto gain button checked during measurements since the slider movement can interfere with ongoing measurements. After the proper level has been found, uncheck the box.

Frequency Response

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Use the OmniMic Frequency Response analyzer to measure the frequency response or impulse response ("**IR**") of a sound system.

Doing a Frequency Response measurement with OmniMic is easy -- essentially, you just play one of the proper Tracks, as indicated near the top of the form, set the microphone to pick up your system's sound, and the graph is shown on the screen. The OmniMic software also allows many adjustments which you can use to customize the graph or measurement. For more Advanced functions, see the help section on that topic.

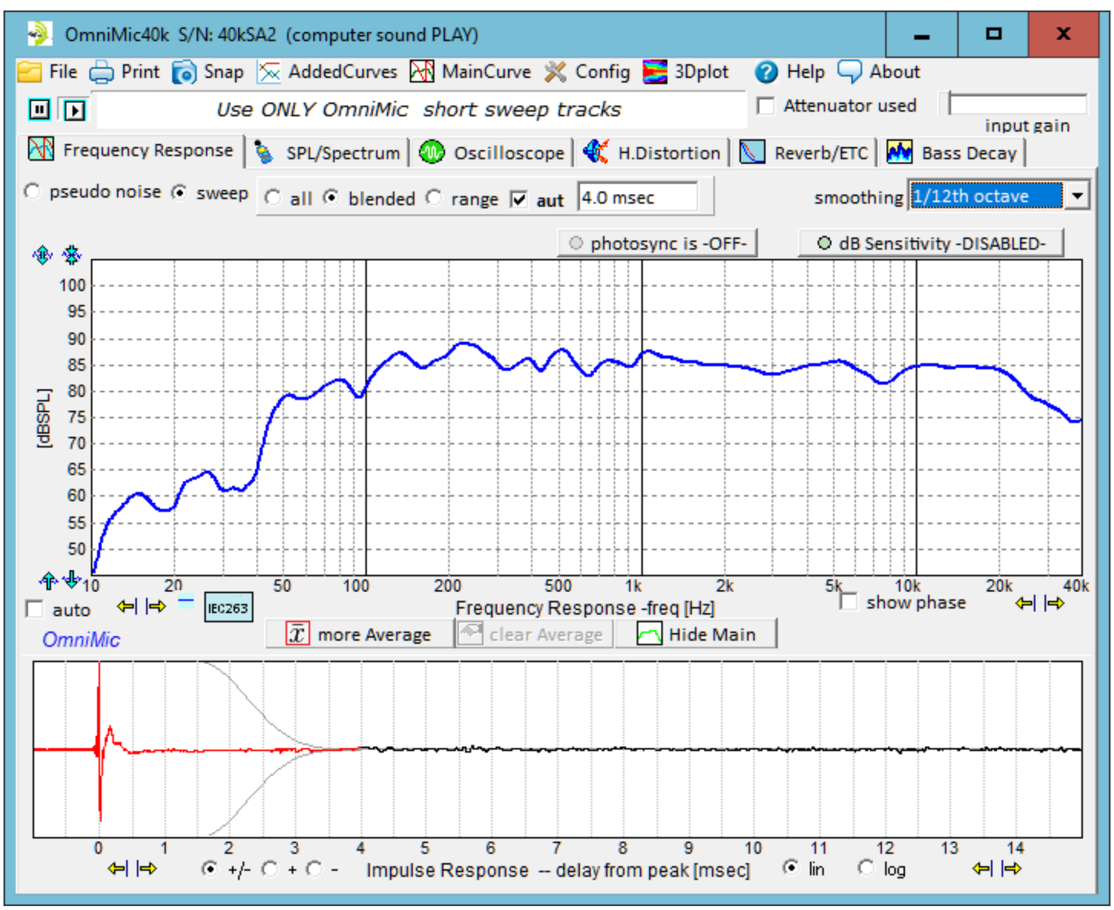
Important! To measure Frequency Responses (or Harmonic Distortion, Reverb, Distortion, or BassDecay) properly you ***must*** be playing the specified audio file. This can be a downloaded WAV or FLAC file from the Dayton Audio site, a stimulus WAV file you can save from the software itself, an OmniMic test track CD, or a test track DVD.

Or the signal can be output by the OmniMic program itself through your computer's soundcard output or an attached DAC device.
---> With OmniMic40k hardware, be sure to configure the playback soundcard on your computer to operate at 96ksps or higher!

For best operation, a high quality USB DAC, such as the **Dayton Audio DAC01** can be used for this purpose. Beware of "audiophile" DACS, many of which fail to exhibit flat frequency responses! **48kHz sweep test files as used with original OmniMic, or a 48kHz DAC can also be used with an OmniMic40k, but proper results can only be obtained up to about 22kHz with those signal sources.**

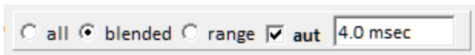
The needed audio file content is indicated at the top of the program window. The measurement is synchronously ***matched*** to the test signals provided on the specified tracks, and will be unable to lock onto other test signals. You can choose to use one of two types of signal by selecting between the "pseudo noise" and "sine sweep" buttons above the graphs.
- Sine Sweep signals ("Short Sine Sweep") provide the cleanest and most accurate measurements, as well as being able to drive speakers at specific SPL levels. This is the preferred choice for frequency response measurements and should be used for all high-frequency measurements
- "Pseudo-noise" test signals, that sound like Pink Noise are easier on the ears for extended sessions. The accuracy using pseudo-noise at highest frequencies, however, can be degraded by sample clock tolerances, so this type should only be used for rough measurements where sweeps may disturb others in the area. Pseudo noise tracks must match the microphone type, 48kHz for original OmniMic microphones and 96kHz for OmniMic40k microphones.

Frequency Response measurements will **not** operate correctly if you try to use other Pink Noise test signals or any signal other than the one specified on the program window. The exact timing and spectral content of the signals are critical for proper operation of OmniMic's precise frequency response measurement system.



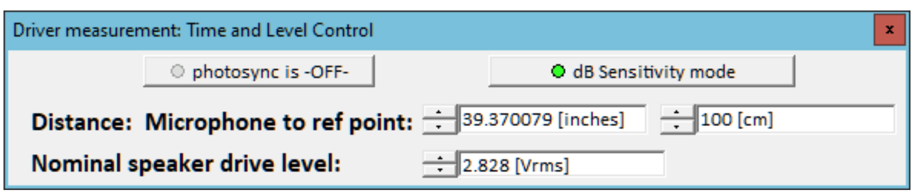
About Frequency Responses

- **Frequency Response is a curve that shows how strongly an audio system reproduces different parts of the frequency range**. The curve will be higher on the graph at frequencies where the system plays louder, and lower (perhaps showing only varying background noise) at frequencies where the system plays weaker or not at all.
- **A perfect Frequency Response curve would look like a flat line over all frequencies**. A *real-world* one will have variations of some decibels ("dB") over most of the frequency range, dropping off at very low bass or at high treble frequencies. Ambient noise or dynamic range limitations may also cause irregular variation at the frequency extremes.
- You can measure near individual speakers (about 1 meter away is best), or out in the room at various listening positions. It is best to **have only one speaker at a time** playing to prevent interference and cancellations between audio waves.
- **The frequency response of a loudspeaker will be different at different places in the room.** You can also use OmniMic to make an ongoing "Average curve" over multiple positions (advanced) to give a typical frequency response curve.
- **A frequency response is related to its "impulse response"** (the pressure signal that a system would make if it were fed by an extremely short pulse). OmniMic calculates effective impulse responses mathematically by comparing a system's output with the signal it was stimulated with (either a swept tone or a noise-like tone). OmniMic shows impulse responses, and you can select how much of an impulse response you want the analyzer to look at ("windowing") when comverting back to a frequency response. **Click on the impulse response (IR) graph (below the frequency response) at the latest part of the impulse response you want OmniMic to include when it computes frequency response. Shift-click on the impulse response to select where you want the IR analysis to begin. The selected portion will appear as a red trace, the rest will remain black.** The shape of the 'windowing' that tapers down the impulse response is shown in green or gray.
- **OmniMic can automatically set the edges of the windowing** region, by checking the checkbox marked "aut".



. The automatic windowing will be turned off if you set the edges by clicking on the IR graph.

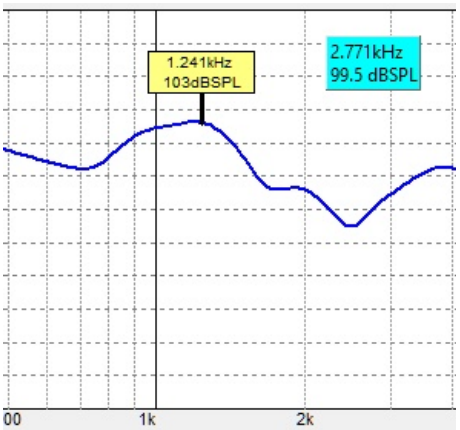
- **Reflections of sound in a room will appear as abrupt features in later parts of the impulse response** (starting usually after about 3 to 10 milliseconds from the main impulse peak). The response of the speaker without room reflections is smoother and less varied than when room reflections are included. Removal of room reflection effects is possible only at higher frequencies -- at lower frequencies, typically below 200Hz, the reflections happen before even one full cycle of a longer low frequency wave has completed so they cannot be separated.
- Because all audio frequencies are played at once in the **pseudo noise** signal, you cannot determine that the speaker is playing at any specific sound level at any individual frequency, so measurements with the pseudo noise show only relative response flatness. Those curves are shown as values given in units of "**dB**". Also, minute variations in clock frequencies of different players can cause the response at higher frequencies can appear to vary when using pseudo noise. When high frequency accuracy is important, use the sine sweep signal. (Pseudo noise stimulus is intended for use only in situations where the sweep tones would be too disturbing by others in the room).
- To evaluate loudspeaker how frequency response changes at specific sound volumes and for best accuracy at high frequencies, use the **sine sweep** to display curve values as "**dBSPL**" (sound pressure level, as detected at the microphone). With an **OmniMic Photolink**, you can easily set the drive level of the sweep to the required 2.83Vrms for true accurate sensitivity measurements. There is also a "dB Sensitivity" button on the Frequency Response screen, which can be used to configure the display for speaker sensitivities when driven at the standard 2.83Vrms level. The control this button brings up enables the program to compensate for measurement at different distances than the standard "1 meter".



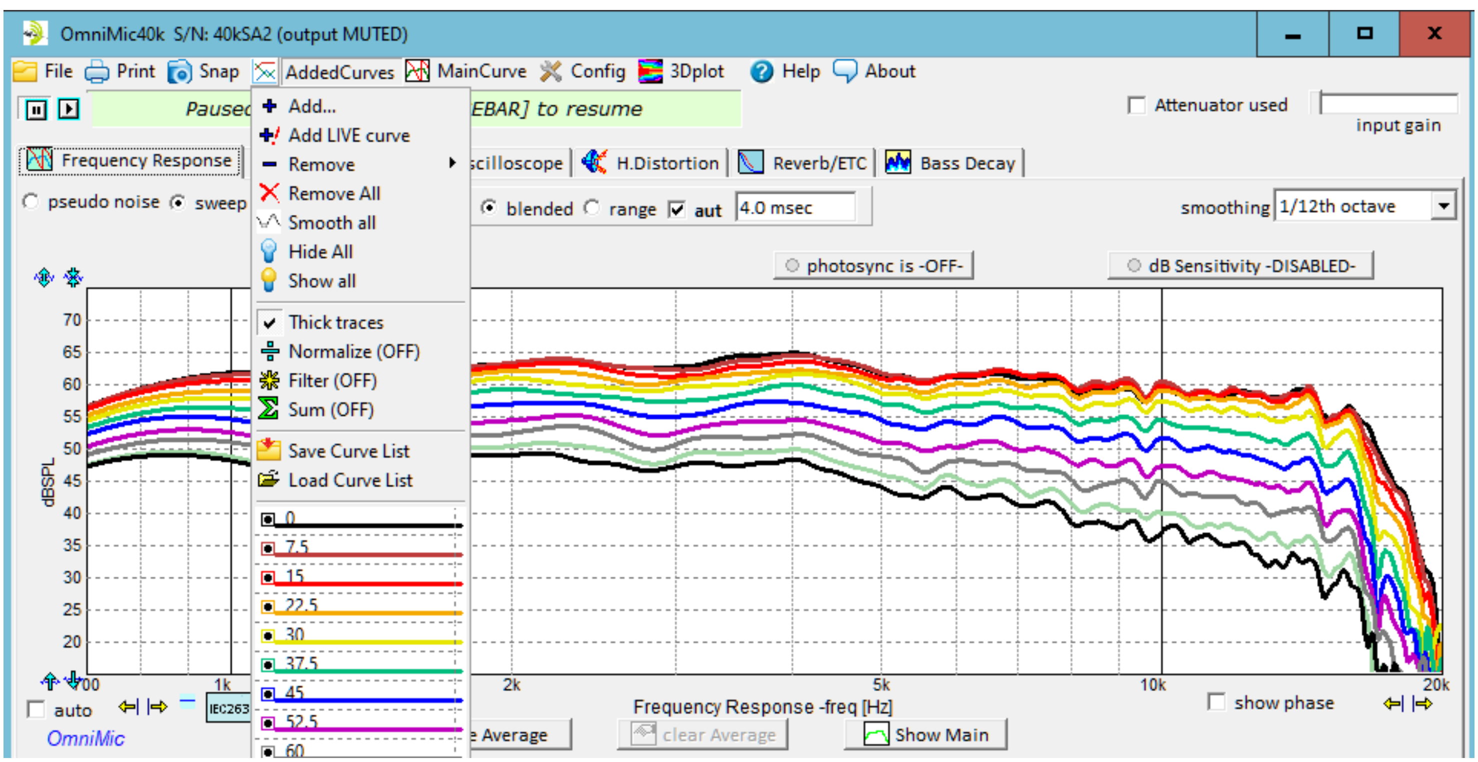
Frequency Response Options:

- **"all":** this setting shows a frequency response that includes all room echoes and reflections. In other words, the entire impulse response. In this view, impulse response itself is not shown.
- "**only to**": for suppressing reflections. This is calculated only from the impulse response within the time selected. To select a different ending time, click the mouse at the desired time point within the impulse response graph. Impulse response information after the selected time will be excluded from the frequency response calculation. Select this time to totally exclude later reflections visible on the impulse response plot. Lower frequencies can't be measured using this option, limited by the length of the time selected. This mode works best when the OmniMic is relatively close to a loudspeaker.
- "**blended**": blends the "only to" impulse response with a time/frequency dependent filtered version. In other words, this mode removes echoes when it can but gradually allows more of the IR time to be included at lower frequencies when it can't. Energy toward the left side of the impulse response is given priority.
- **smoothing**: You can choose to smooth the frequency response graphs over 1/2 octave to 1/96th octave regions, to vary the amount of detail shown. An unsmoothed frequency response will be very ragged except when echoes are windowed or the measurement is taken very close to the loudspeakers. An additional "ERB" smoothing option simulates the sensitivity of human hearing to response irregularities for estimating the sonic importance of frequency response irregularities.

- **markers:** When the frequency response display has the mouse's "attention", a blue text box will read out the frequency and level related to the location of the mouse cursor. If you then type one of the keys 1 through 9, a persistent white "floating" marker will attach to the live (currentlybeing measured) trace at the freqeuncy related to the marker. Click again on the persistent marker to clear it, or type the 0 (zero) key to clear all of the persistent markers.



- **Frequency Response measurement curves can be saved i**n FRD format using the ["File, Save" menu](). You can save either or both of the current "live" curve (currently produced by the microphone) or any displayed Average curve. These can then be reloaded using the "**Added Curves**" menu so that **multiple Added Curves can be shown on the same graph** along with the "live" curve and the Average curve. When Frequency Response graphs that contain Added Curves are printed or saved in a "Snapshot", a list of legends can be included. In snapshot files, a text note can also be added.
- Up to 40 saved "**AddedCurves**" Frequency Response curves (previously saved in "FRD" files) can be displayed on-screen simultaneously with the currently measured curve trace -- use the "AddedCurves -> Add" menu.



- The full set of AddedCurve filenames, along with applied offsets or delays can be saved into a **Curve List** for easy retrieval to the OmniMic screen.
- You can alter the **thickness** of the displayed "live" Frequency Response curve by toggling (**left-clicking**) the small button that is just to the right of the "OmniMic" logo at bottom left of the graph. **Right-clicking** the button will change the thickness of the saved or [average]() curves.
- There are also a considerable number of additional [features for advanced users]() available including phase responses, color polar displays, compression plots and waterfall decay plots.

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Frequency Response - Advanced Functions

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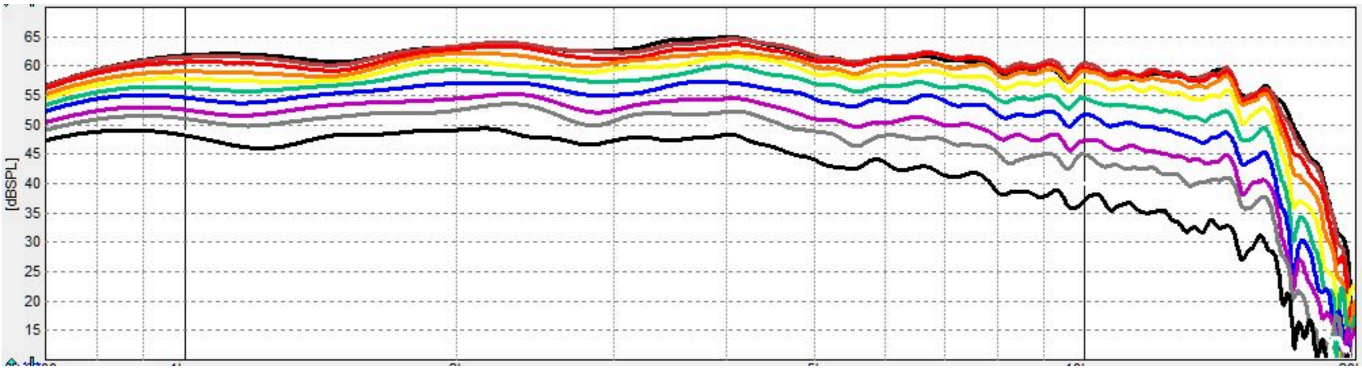
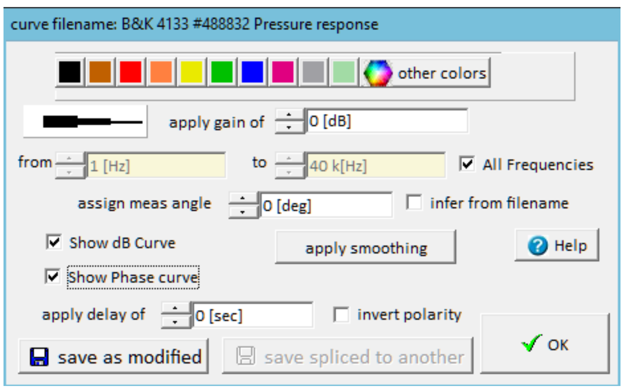
Frequency Response analyzer measurements and features for users who need to go into more depth in their measurements.

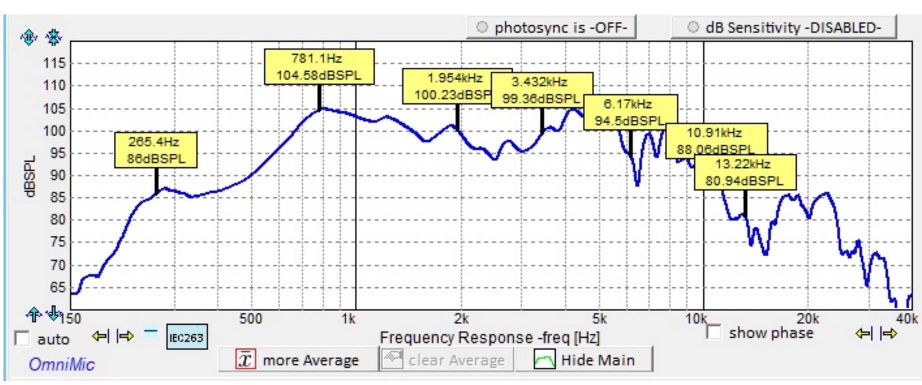
Advanced users should read the following and try the features described below:

- Up to 40 saved "**AddedCurves**" Frequency Response curves (previously saved in "FRD" files) can be displayed on-screen simultaneously with the currently measured curve trace -- just use the "AddedCurves -> Add" menu. Each **can be individually offset in decibels, delayed in time, inverted in polarity, further smoothed, or have its frequency range limited** for display or resave. The curves, as adjusted with those changes can be further saved as an additional FRD file.





- **Repeated Plot overlays** - the "Added Curves" function can be used to quickly ovelay plots from a series of measurements. Just go to the "Added Curves" and click on "**Add Live Curve**". The currently measured ("live") curve will be saved to a directory (which you select when the first Live Curve is added) and will be displayed on-screen as an Added Curve. Curve colors will be automatically changed on each Add, but can also be easily changed by selecting the measured curve in the "Added Curve" menu.
- Up to 9 persistent **"floating" markers,** which will move up and down with the trace, can be placed on a live trace. Position the mouse cursor over the graph at the frequency you want the marker placed at, and tap one of the keys for "1" through "9". Tapping "0" will allow you to clear all floating markers. Right-clicking on a floating marker will allow you to delete it individually. Such markers can also be placed on Harmonic Distortion graphs or



RTA/Spectrum graphs.

- a set of **Math** functions are available in the MainMath menu to apply when measuring live responses. (Some of these functions can also be applied to all "Added" curves, see that menu).

-- as compared ("**Normalized**") to a saved file curve; This reverses the response effects of the curve normalized with. If you normalized a curve by itself, you'd get a flat line.

-- use file curves as a filter (like and eq curve) during measurement ("**Filter**"). If you Filtered a flat line with a Filter curve, you'd get the Filter curve.

-- "**sum**" (vector style, including phase) with a previously measured response from a FRD file. This could be used, for instance, for combining port and woofer measurements (with proper weighting).

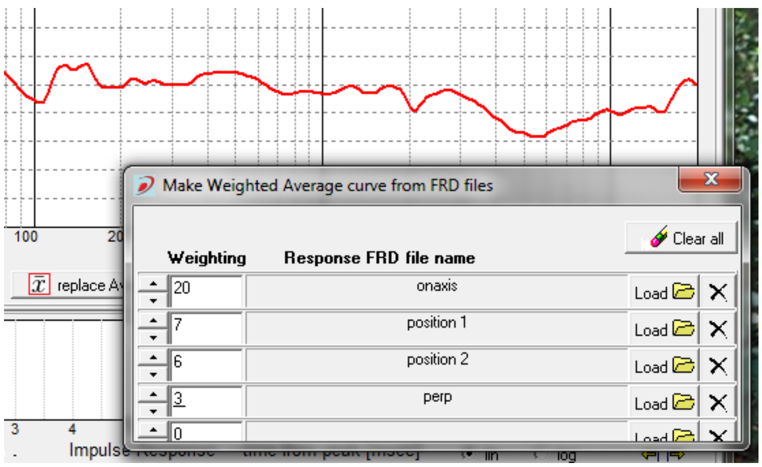
-- "**offset",** that is, moved up or down by a specified amount (in dB)

-- with the response "**flipped**" (so that peaks are replaced with troughs and vice-versa). Could be used to generate equalization targets.

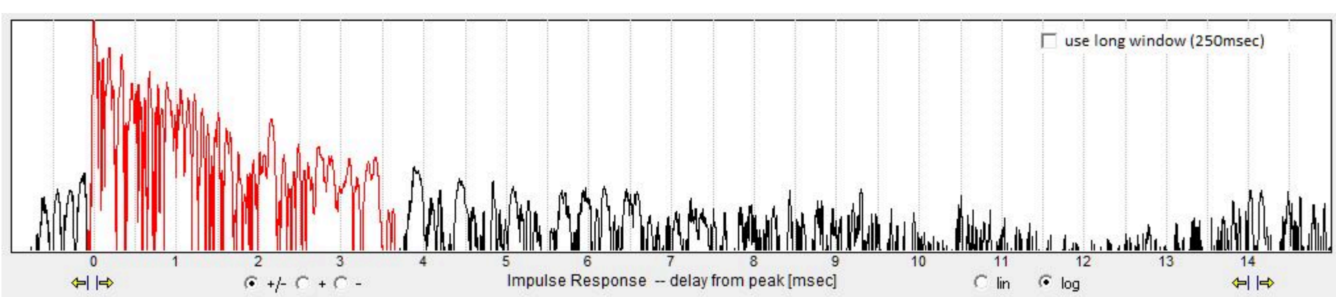
-- or have the OmniMic evaluate whether the displayed portion of a measurement falls entirely within the area between the first two specified file curves ("**Evaluate within**"). This allows for easy pass/fail production testing of loudspeakers for frequency responses. Offsets applied to the curve files during display are utilized for the evaluation (so you can use the same file for both limits, one offset upwards and the other offset downwards). You can also select whether to allow the program some vertical shifting to get the measured curve to fit (good for when you care about shape more than absolute sensitivity match).

Some of these functions utilize ".FRD" type text files as generated by OmniMic or other programs. These files can also be easily created or edited using any text editor such as Windows' "Notepad", or with a spreadsheet program (export in "text" format). FRD are simple text files, with each line containing values for Frequency, dB, and (optionally) phase. Frequencies must be in ascending order with each successive line, and lines at the beginning starting with a quote (") will be skipped (ignored).

- measured responses can be arranged to be inverted in shape or offset vertically in dB.
- **Averaging**: Left-click the "**New Average**" button to freeze the current frequency response curve on the graph (alternately, your keyboard's space bar can be used for this function). New live measured curves will be shown along with the frozen one. If you left-click it repeatedly, each current live curve will be averaged into the frozen curve (the button label will change to "**more average**" and indicate the number of averages included so far). This function is very useful for determining the average response curve over a range of listening positions in a room.
- Click "**Clear Averages**" to erase a frozen or averaged curve and show only the live curve.
- If you want to see only the frozen Average curve, click on "**Hide Main**".
- You can save an Average curve for later recall using the "File" menu, or recall a previous saved FRD file to replace the Average curve.
- To **average in a response curve that has been previously saved to a file** via the file menu, first "Add" the curve to the display using the "Curves" menu. Then **Right-click** on the "New Average" or "more Averages" button to include the last Added file to the average curve.
- **Weighted Averaging**: An alternate and more flexible way to generate an Average curve is to save as FRD files all of the responses you wish to average into one. Then select the menu "File->make Weighted Average" and load each of the desired files using the buttons provided, along with the weight you would like to apply to each. The result on the Average curve will be updated in real time as files are added or reweighted.



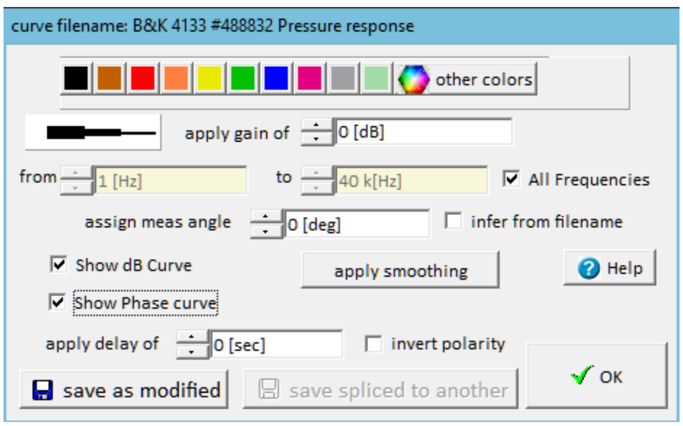
- An option to show the **impulse response plot in "logarithmic"** magnitude form rather than the usual "linear" form. Logarithmic form hides polarity but makes it easier to look at the decay shape of an impulse response curve. With logarithmic display it is often easier to identify discrete reflections in the impulse response, which can be particularly helpful when making Waterfall measurements. More detailed and quantified log IR graphs can also be measured using the tools on the "Reverb/ETC" tab.



- Two styles of color **3D Polar displays**, for displaying responses in sets of saved files in formats that show how the frequency response of a speaker varies as it radiates into different directions. To use these, there must be **at least three added curves** in the main frequency response plot, each with different measurement angles assigned. See: Polar displays.
- Three different styles of **Waterfall graphs** can be made from impulse responses. When the Waterfall button (just to the left of the smoothing control) is pressed, the OmniMic form shows the impulse response being measured at the bottom of the screen and a Waterfall plot above it. For more about making Waterfalls, see: Waterfalls.
- You can also display the **phase response** of the loudspeaker by selecting the checkbox shown at bottom right below the frequency response graph. Phase display works with the "only to" or "blended" options. You can use the **delay adjustment** (lower right part of the screen) to specify an amount of delay you

wish applied to the live measured phase display. (Note that phase response is lost when Averaging is being used). You can also configure to show phase response for any of the "Added Curves" (see above).

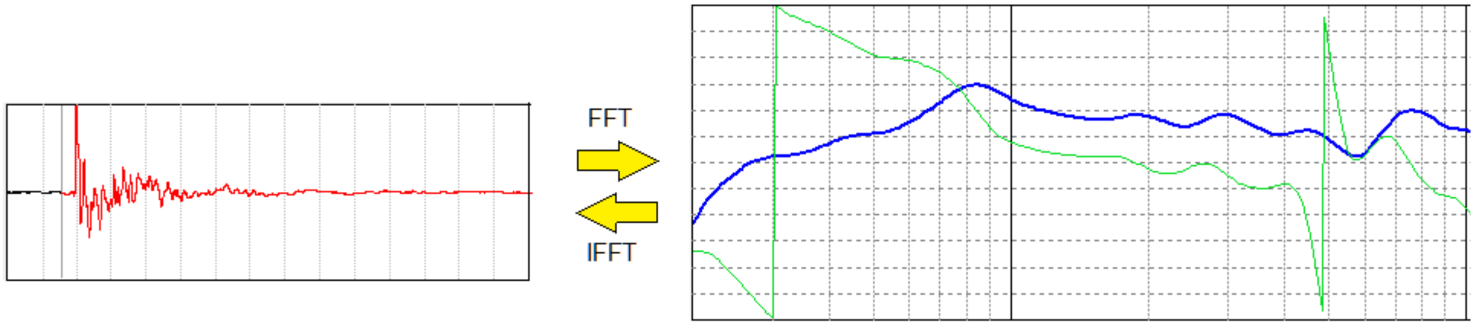
- You can choose to **invert the polarity** of the signals seen by OmniMic, using the "Main Math -> invert capture phase polarity" menu checkbox. This will turn measured Impulse Responses upside-down, and shift Frequency Response phases by 180 degrees.
- **Time references**: Without photosync, OmniMic aligns "0 ms" to occur at a maximum peak in the impulse response. With photosync, 0 msec meaning depends on the setting in the Driver Measurement Control.   Normally, the frequency response calculates from this point forward, and the peak of the response will be the time reference plane. Left click to define the end of the selected IR range, Shift-click can be used to modify the start point.
- When you are not using photsync, there are buttons at the lower left of the impulse response to choose whether to synch to the larges **positive [+], negative [-], or largest of either [+/-]**. This can be useful when a positive peak and a negative peak are similar in level, to prevent synch jumping back and forth between the two peaks.
- You can also select to have OmniMic start calculation from a different first point.   If you wish to see only the response resulting from a later peak (and if the first "highest" impulse response has decayed sufficiently by then to not interfere), **Shift-click** the mouse on the impulse response at the starting point of the portion you wish to include. (In previous software versions, the right-click was used to do this).
- **AddedCurves** can be combined mathematically, can have another response from a saved file applied to further shape them ("**Filter**") or to unshape them ("**Normalize**"). Normalizing a curve by its own file will give a flat line.
- **AddedCurves** can also be **associated with a "measurement angle"** (and will try to infer this angle from a numeric value in the file name). This is to allow easy configuration of polar response plots.
- You can also select the color of an **AddedCurve**, whether to show phase or dB (or both).
- An **AddedCurve** can be resaved with these alterations (offset, delay, frequency range, polarity) to a new FRD file.
- When exactly two **AddedCurves** are used on a plot, you **can splice them together** and save them to a new FRD file. This can be done from the form that appears when you add or edit a curve, from the AddedCurves menu . This feature can be used to easily **splice nearfield and farfield measurements together**. Phase responses (if displayed) will be spliced also. See "Splicing". If the frequency ranges shown for the curves overlap, the responses will be blended together to make a smooth transition. If there is a gap between the frequency ranges, the region between will be attached with as smooth a curve as possible.
- To **edit an already visible AddedCurve**, just click on its name near the bottom of the list on the AddedCurves menu.



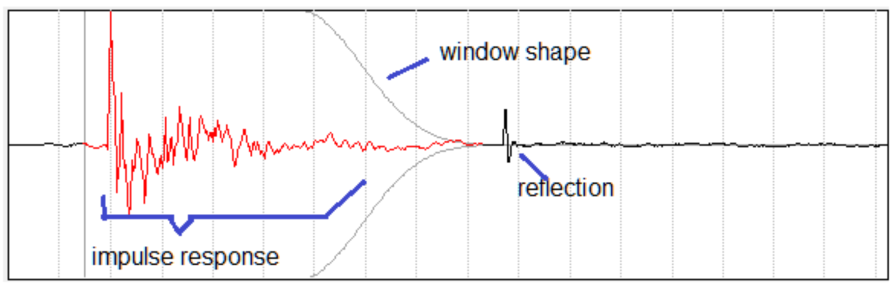
- The full set of **AddedCurve** filenames, along with applied offsets or delays can be saved into a **Curve List file** for easy retrieval to the OmniMic screen at a later time. The first two curves specified can be used to define frequency response limits for (QC) testing (see above).
- **Generation of parametric equalizer values** while simulating their effect on an Average curve. Files can be directly loaded into some versions of Mini-DSP. See MiniDSP Equalizer Tuning
- With most computers, you can play the frequency response test signals (the pseudo noise or the short swept sine) out **from a soundcard** in the computer rather than playing the needed signals from a file player, CD or DVD. Use the menu in the Config menu. 96kHz sweep files (or OmniMic sourced signals from the computer's sound output) and an OmniMic40k microphone must be used for obtaining data beyond 20kHz.
- When a soundcard in the computer is used to generate sweep test signals, a **Compression Test** can be run to find how a speaker output compresses at increasing SPL volume or voltage drive levels. Use the Compression Test menu item under the Config menu.
- See also ~~"Test Signal Files"~~ for ways to play the test signals for making measurements.

Setting Impulse Response Window Edges

There is a direct correspondence between an impulse response (IR) and a complete frequency response (with both magnitude and phase), and the two data sets can be directly converted between them using the well-known Fourier (or Inverse Fourier) Transforms.



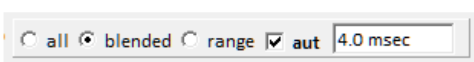
Having the data in the form of an impulse response is particularly useful as it can be edited to exclude or attenuate easily seen effects from reflections in the room, in effect giving the advantages of an anechoic chamber. The region of the impulse response that is free of reflections is chosen and then transformed back to frequency response form.



The start (left side, earlier end) of the window can be chosen by holding the Ctrl button while clicking the mouse where you want the selection to start. The end (right side, later) position is selected by simply left-clicking where you want it to end.

Automatic Window Edge Setting

In most situations you can simply choose to l**et OmniMic find the window edges**, by putting a check mark in the checkbox marked



"aut":
Then, OmniMic will choose a starting position that is before the largest peak found in a following impulse response and choose an ending position before the first amplitude that is more than a certain fraction of the maximum peak's amplitude. You can configure how the edges are to be found by Ctrl-clicking on the box that shows the chosen window length (just to the right of the "aut" checkbox). This will prompt you for three numbers:

- the **minimum window time length,** which is how many seconds to the right of the maximum peak to go before looking for a large enough reflection.
- the **reflection height threshold**, the fraction of the maximum peak's amplitude to look for to consider it a large enough reflection.
- the **star before peak** time, which is how far to the left of the maximum peak to set the start of the window.

Included window length affects how low in frequency reflection-free frequency response can go, so the **window time length parameter** should be long enough to assure reasonable data to at least upper midrange frequencies. It also should be long enough that any of the multiple peaks in the first impulse response are not interpreted as being reflections.

The only reason to not make the **start before peak** be all the way back at the left-most area is to analyze the energy found in a reflection. Also, slightly more noise can get into the measurement than if you move the edge up close to the start of the activity. With some impulse responses, though, the largest peak isn't actually the largest, so in that case you should certainly have the impulse response start before that so the earliest activity is included.

See also: Phase Response and Compensation

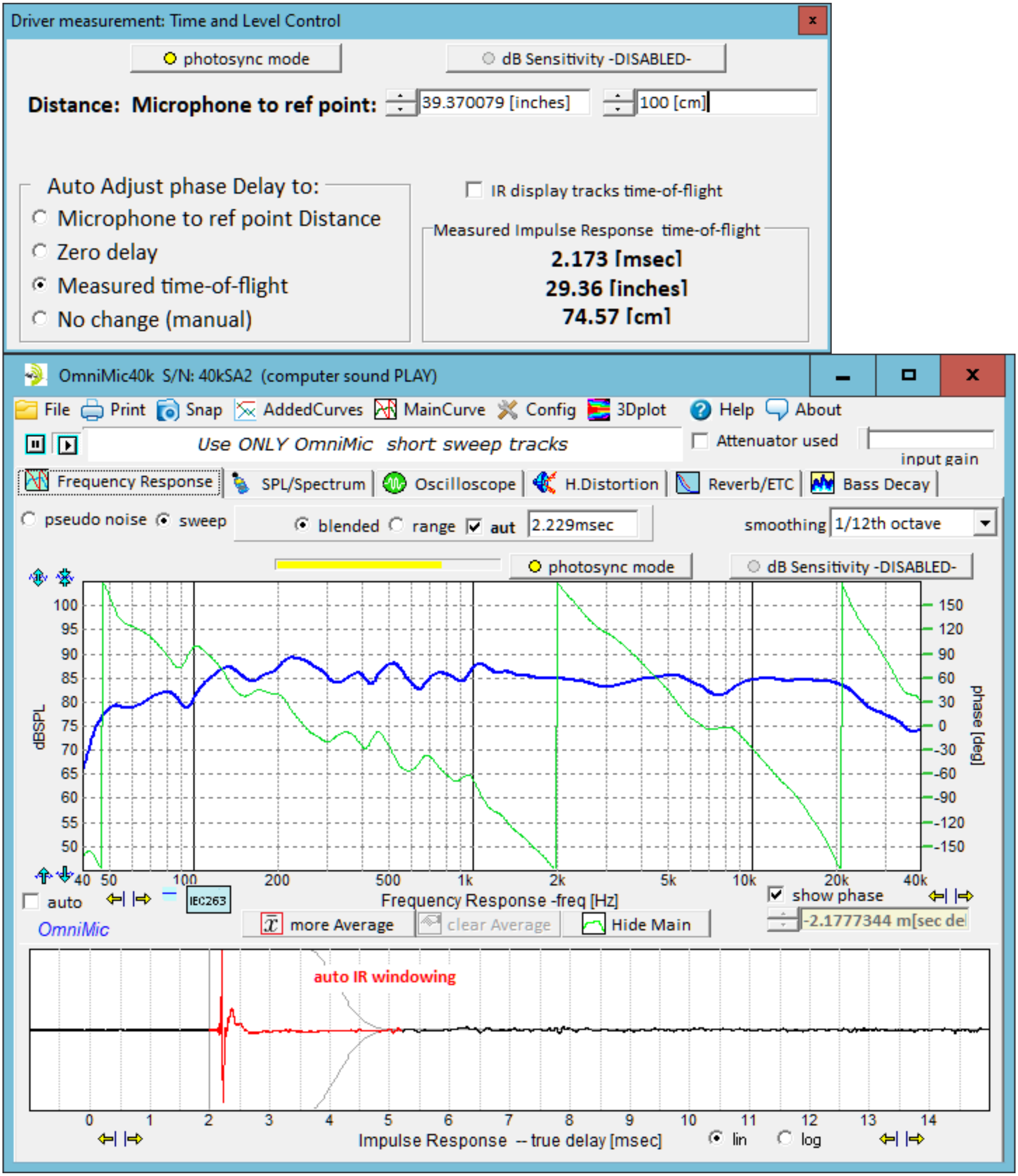
Distance from Microphone to Loudspeaker

During measurement of frequency response or distortion, the distance of the OmniMic from the loudspeaker or driver will affect both the sound pressure level it detects and with PhotoSync, the phase response. The phase response effect is actually an advantage, because it includes true information about the travel time delay of the sound on its way from the apparent origin of the driver's output and the microphone, which is important for measurements such as when making frequency response data files for crossover network design using simulation tools.

Sound pressure drops off with distance. The ideal value is 6.02dB drop each time the distance doubles though it can be as much as 3dB less than that for some driver types.

When not using PhotoSync, OmniMIc processes phase data based on the largest peak in the impulse response, approximately as if there were no distance between driver and microphone, so the phase curve shape is more gradual and easier to interpret. However, without PhotoSynch it doesn't account for distance from the acoustic source point.

Using PhotoSynch does visually complicate the curve so the delay should be compensated to minimize the phase "wrapping". If you are making files for crossover design, use the "Microphone to ref pt Distance" setting with the correct physically measured microphone to mounting baffle distance. If not, then use "Measured time-of-flight" and the program will compensate delay based on the time it determined for the impulse response peak to arrive (with similar phase result as you would get with PhotoSync turned off).

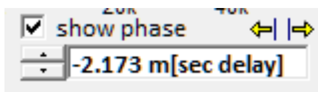


Driver measurement: Time and Level Control
photosync mode
dB Sensitivity -DISABLED-
Distance: Microphone to ref point:
39.370079 [inches]
100 [cm]
Auto Adjust phase Delay to:
Microphone to ref point Distance
Zero delay
Measured time-of-flight
No change (manual)
IR display tracks time-of-flight
Measured Impulse Response time-of-flight
2.173 [msec]
29.36 [inches]
74.57 [cm]
OmniMic40k S/N: 40kSA2 (computer sound PLAY)
File
Print
Snap
AddedCurves
MainCurve
Config
3Dplot
Help
About
Use ONLY OmniMic short sweep tracks
Attenuator used
input gain
Frequency Response
SPL/Spectrum
Oscilloscope
H.Distortion
Reverb/ETC
Bass Decay
pseudo noise
sweep
blended
range
aut
2.229msec
smoothing
1/12th octave
dBSPL
phase [deg]
Frequency Response -freq [Hz]
auto
IEC263
show phase
OmniMic
more Average
clear Average
Hide Main
-2.1777344 m[sec de
auto IR windowing
Impulse Response -- true delay [msec]
lin
log

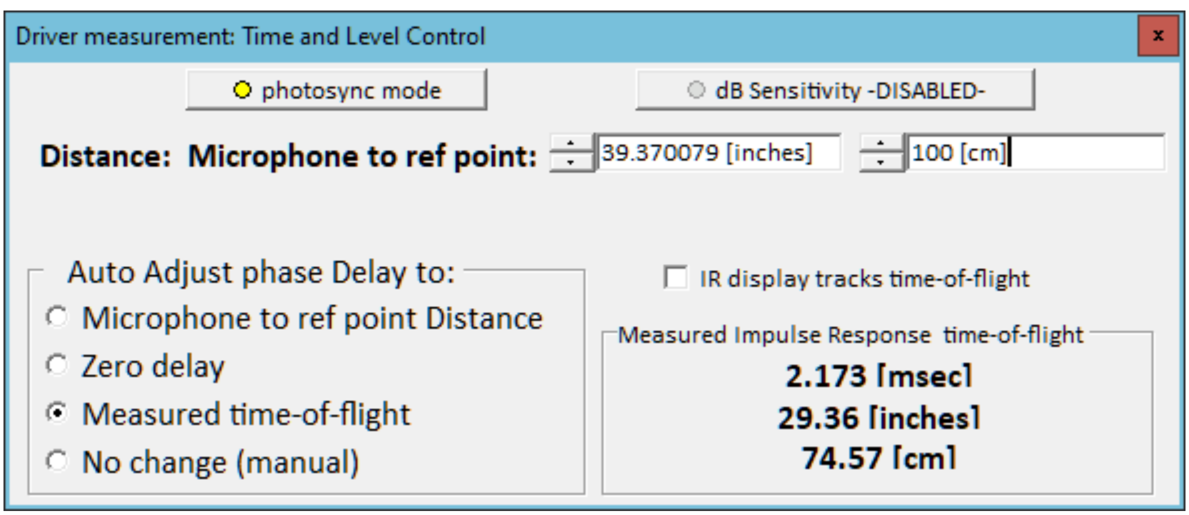
Phase Response and Compensation

The choice of window edges around the impulse response does not generally affect the phase response that shows in the resulting frequency response graph, provided that the impulse response does fall within the window and that interference from reflections included don't corrupt the overall response. How far to the right of 0 seconds the actual acoustic impulse response begins represents delay which DOES strongly affect phase.

This is important to keep in mind particularly when using Photosync with the Photolink device and OmniMic40k to include true distanc travel in the measurement. In that case, the impulse response will be shown at some distance to the right, depending on the "time of flight" or how long it took the sound to get from the loudspeaker to the Omnimic due to the finite speed of sound travel. The frequency response magnitude will not be affected, but its phase response will be strongly effected because time delay increases downward phase shift proportional to the frequency of each data point. That will usually make the phase graph too busy, with phase repeatedly crossing the +/-180 degree wrap-around limits*. So to make it more comprehensible you can compensate the phase response with a negative delay value in the control at the bottom right of the frequency response graph below

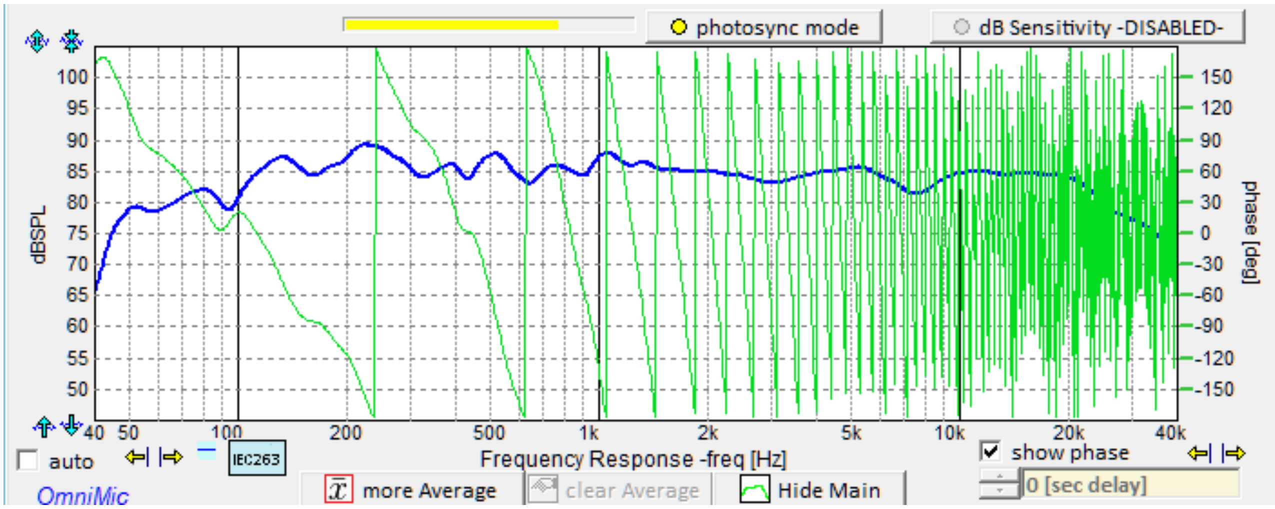
the "show phase" checkbox. 

When you enable Photosync, a control box will appear labeled "Driver measurement: Time and Level Control". On it, there is a section "Auto Adjust Phase Delay To:" that you can use to configure how you want the delay control to get its value.

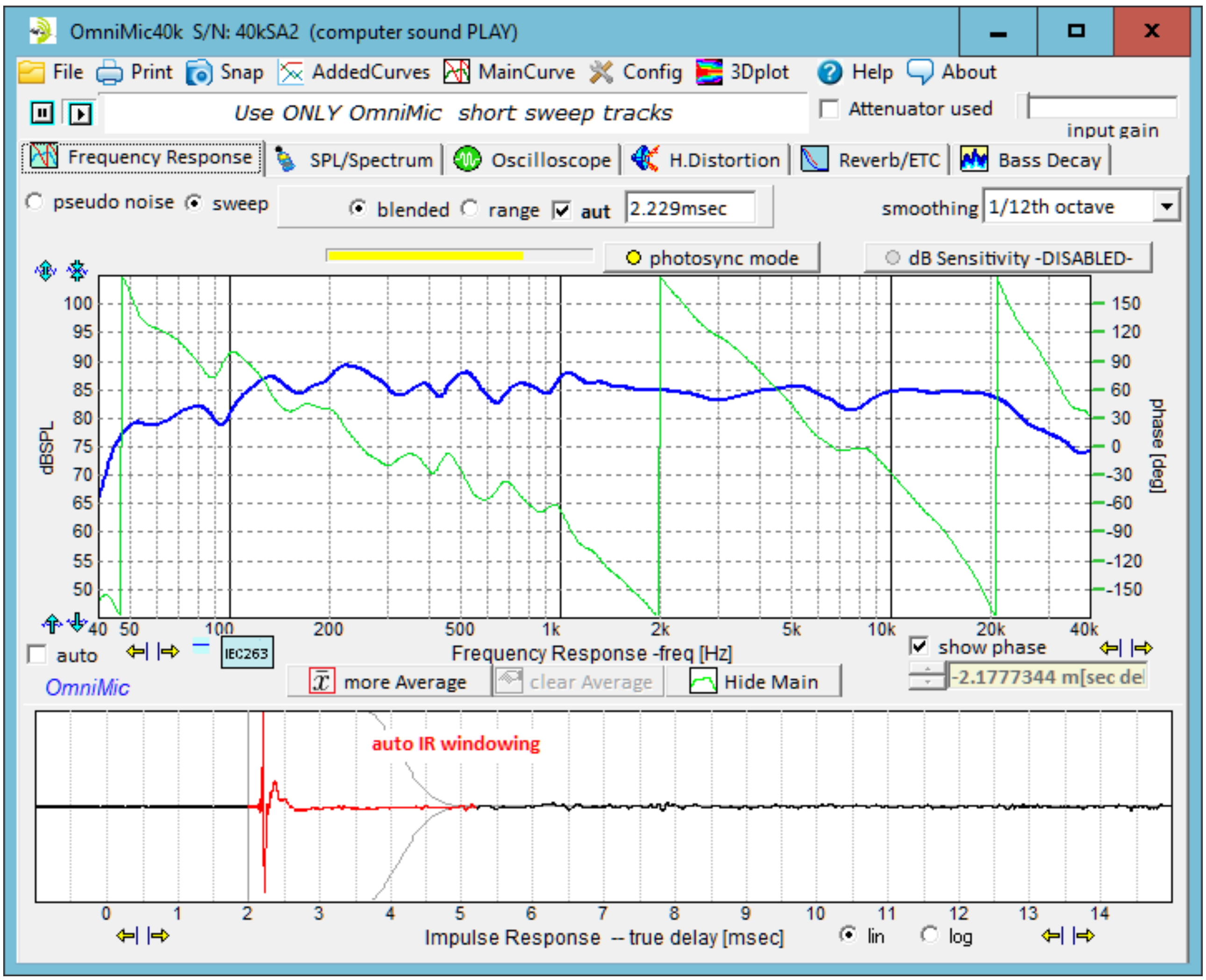


- Microphone to reference point distance: compensates for the delay caused by time sound takes to travel the distance manually entered at the top. Use an accurate measure of the distance. This is the method to use when generating data files of drivers for crossover network design.
- Zero delay.... just the number 0 seconds.
- Measured time-of-flight: how much time passed between when the signal appeared at the speaker terminals as sensed by PhotoLink, and when the largest peak of its impulse response arrived at the OmniMic40k. This will automatically track if you change the microphone position.
- No change (manual). Use the delay control at bottom right of the frequency response graph to set the value manually (for a known delay or per the desired appearance of the phase curve).

Delay uncompensated:



Delay Compensated for Measured Time-of-Flight:

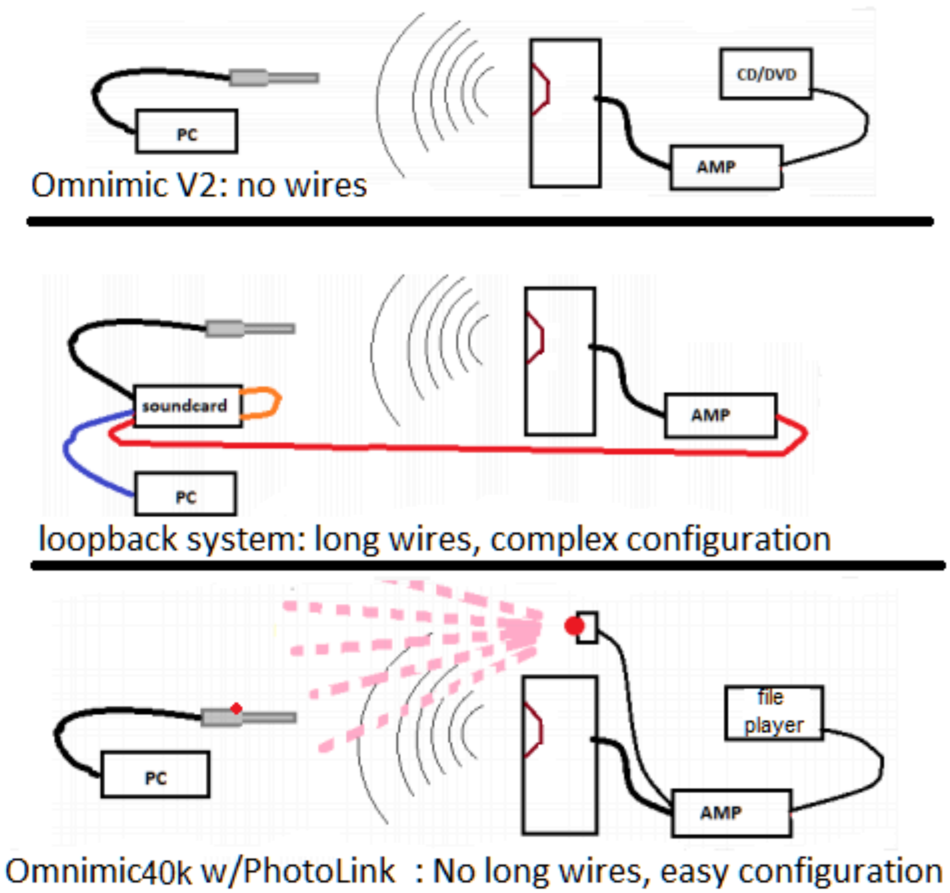


*Phase is periodic. For instance at any one point, +180 degrees = -180 degrees = *540 degrees = -540 degrees, etc. Mathematically, any phase number plus or minus 360 degrees is exactly the same. The graph will wrap around the phase curve within -180 to +180 degrees as that is all that can be determined at any one frequency. Expressing as an unwrapped phase curve is possible, but necessarily assumptions must be made about how the result was reached from one frequency to the next --- did the true curve wrap around the +/-180 border to get there, or was it direct?

Optical Synchronization

Optical Syncrhonization (or "PhotoSynch") is a way for OmniMic40k to provide benefits of "loopback" measurement of frequency responses of loudspeakers, such as true delay and phase measurement. This is accomplished while still maintaining OmniMic's easy operation and without requiring wired connection between the microphone/computer and the audio system. See "Using the OmniMic Photolink device " for details on connection and use of the PhotoLink for drive level setting and for PhotosSync timing synchronization. (Photosynch operates only with OmniMic40k hardware, but the PhotoLink device can be used for level setting of sweeps with when using OmniMic V2 microphones).

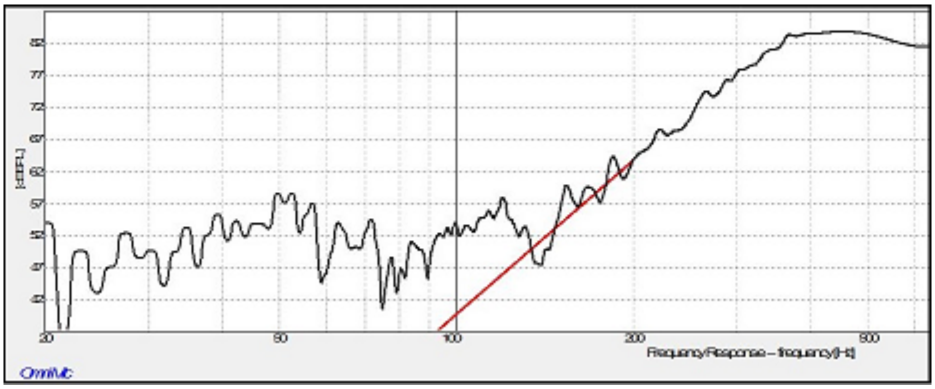
Photosync operates by transmitting an infrared signal from the PhotoLink sender to an optical sensor mounted on the microphone body. The timing signal arrives at the microphone at light speed with essentially zero delay and the OmniMic software can then determine when the test signal appeared at the terminals of the loudspeaker being measured.



Cleaning up Frequency Response Curves (or calculating Minimum Phase responses)

Index

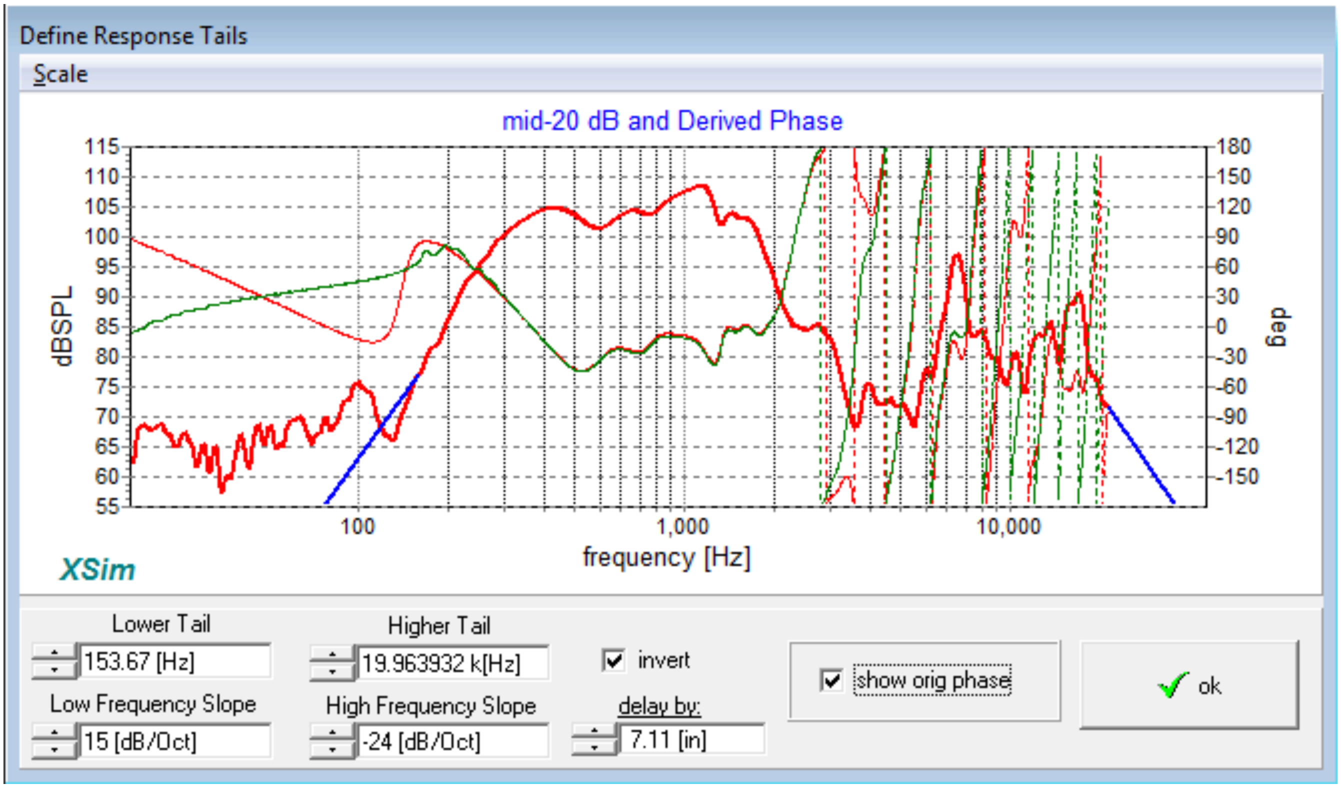
A reality of any physical measurement is the finite degree of dynamic range that can be practically obtained. Environmental noise, hardware limitations or mathematica processing can all limit the dynamic range of any collected data, and can result in portions of a measured curve in which data is corrupted, incorrect, or meaningless. A typical case would be at lower frequencies, where the frequency response in lower-level regions may show mostly the noise present at the time of the measurement.



This could cause errors in later interpretation, processing, or simulation using the data. Usually, the lower levels involved minimize the error caused in, for example, a crossover network simulation done in an application such as XSim, as the poor data will be swamped by energy from other signals in the same band. Phase data in that frequency range, however, will usually be completely incorrect, possibly even seeming random. And it certainly makes the data curve look poor.

Often the actual response shape in the lower level regions is known. For instance, at low frequency extremes, the response of a sealed box loudspeaker is known to roll-off at a 12dB/octave rate; a ported box rolls off at 24dB/octave. OmniMic provides a feature to allow you to fix these portions of a response curve, and to calculate the correct phase response shapes in these regions. NOTE THAT THIS PHASE CORRECTION WILL ONLY WORK ON SINGLE DRIVER (i.e., MINIMUM PHASE DEVICE) CURVES. For the calculated phase response to be accurate, the magnitude (dB) levels must be known even at frequencies below and also to much higher frequencies than the measured data.

In the Frequency Response section of OmniMic, use the "**File -> Phase Restore an FRD File**" **menu** to bring up an editing form in which you can select the points where to attach the estimated rolloff curves and the slopes at which they roll up (or down). You can also adjust the polarity and effective delay (relative to minimum phase) so that phase response will match over the strong parts of the curve. This operations can be performed on **saved** frequency response (FRD) files, and the result is saved onto another (or the same) FRD file, as you choose.



To use the feature, use the menu item to load from the presented browser the FRD file you want to fix. Click on the box labeled "show orig phase", set the Lower Tail and Upper Tail frequencies, and if necessary, the Low Frequency Slope and High Frequency Slope. As you make these settings, the calculated "**minimum phase** response" curve (the thin green line) will update. For cleaning up data without affecting the phase response in the strong signal area, adjust the applied delay ("delay by") and polarity along with the rolloff slopes, if necessary, to match the phase of the measured curve in the area where the magnitude is strong. (If you are only looking for a minimum phase curve for the data, set the delay to zero (0) and the rolloffs to the best estimate of the device rolloff in the noisy or out of band area).

Then click 'Ok' and OmniMic will prompt you for the file name to save the new results under.

Frequency Response: Waterfalls

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Waterfall plots are used by driver and loudspeaker designers for driver selection, to identify resonances or reflections, and to view driver and waveguide behavior.

The Waterfall feature is accessed via the "3DPlot" menu. Waterfalls are calculated from an impulse response while measuring.

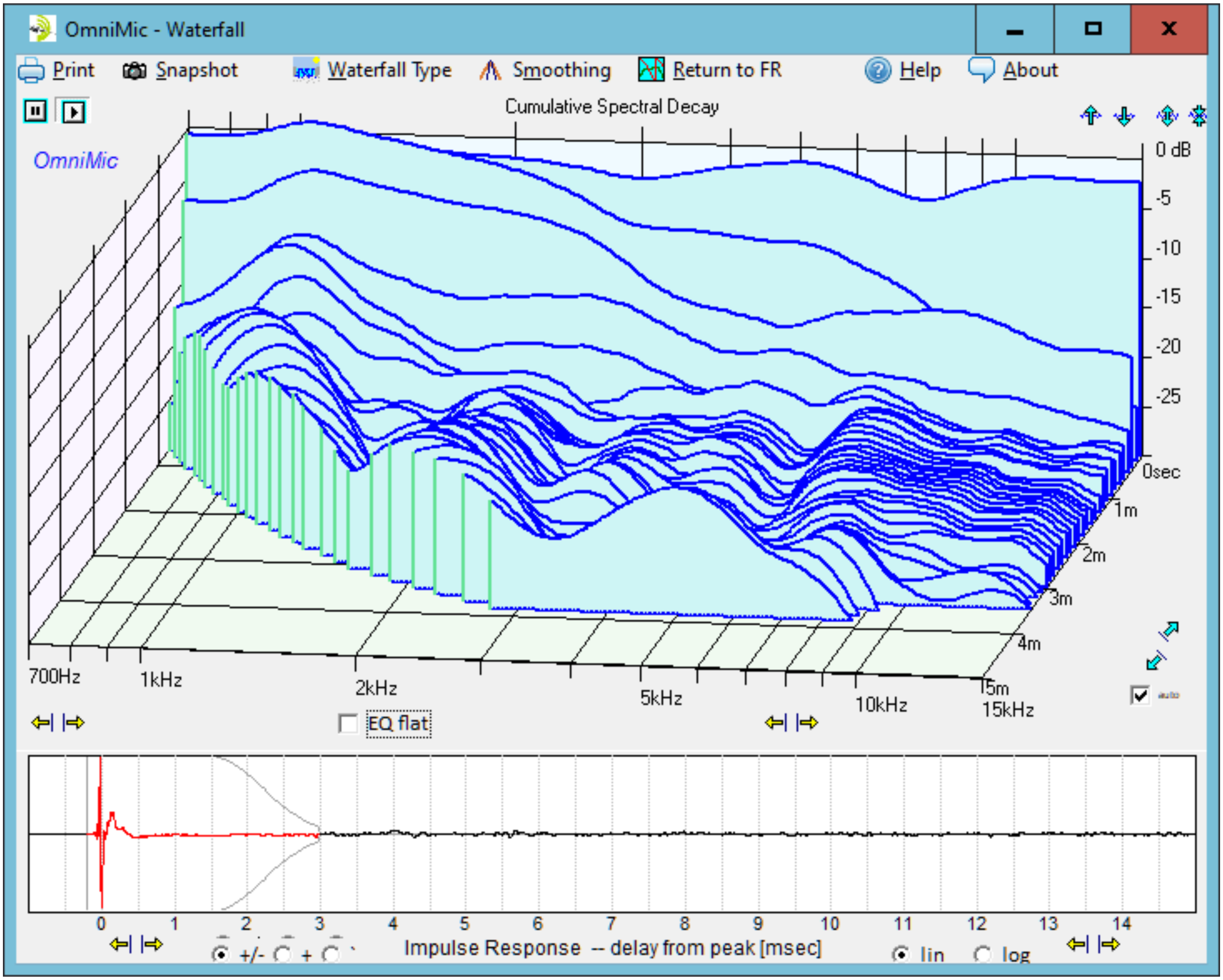
What does a Waterfall mean?

A waterfall is an attempt to illustrate on a 3-D graph how the energy decays or is radiated over a range of frequencies. OmniMic includes three different styles of waterfall processes, selectable via the "Waterfall Type" menu.

A "**Cumulative Spectral Decay**", or "CSD" waterfall shows a series of time slices approximately indicating the contribution to the total response that is made after the time instant shown in the axis going into the screen. When a loudspeaker is driven with an electrical impulse, the pressure it creates should ideally also represent a pressure impulse. But loudspeaker drivers aren't ideal so they also generate resonances -- pressure waves that decay more slowly at various frequencies. The effects of echoes can hide the resonances in a CSD waterfall, but at higher frequencies the echoes can be removed by "Windowing" the calculation to only include the part of the Impulse Response that occurs before the first reflection (from a surface such as a wall or furniture) reaches the OmniMic. Careful choice of positioning within the Impulse Response is critical, because the effects of any reflections included within the selected portion will contaminate all regions of the graph ***up to*** that point on the time axis. Below some frequency determined by where the Impulse Response is clicked and how far along on the time (depth) axis a trace exists, meaningful calculation cannot be done. The graph curve is chopped off at those points on the waterfall display.

The CSD waterfall calculation process introduces some spurious side effects, so the graph should be viewed in general terms. Exact values along the curves of waterfalls are not usually reliable, rather, the positions and sizes of decaying forward-approaching ridges on the graph indicate frequency and relative intensities of resonances.

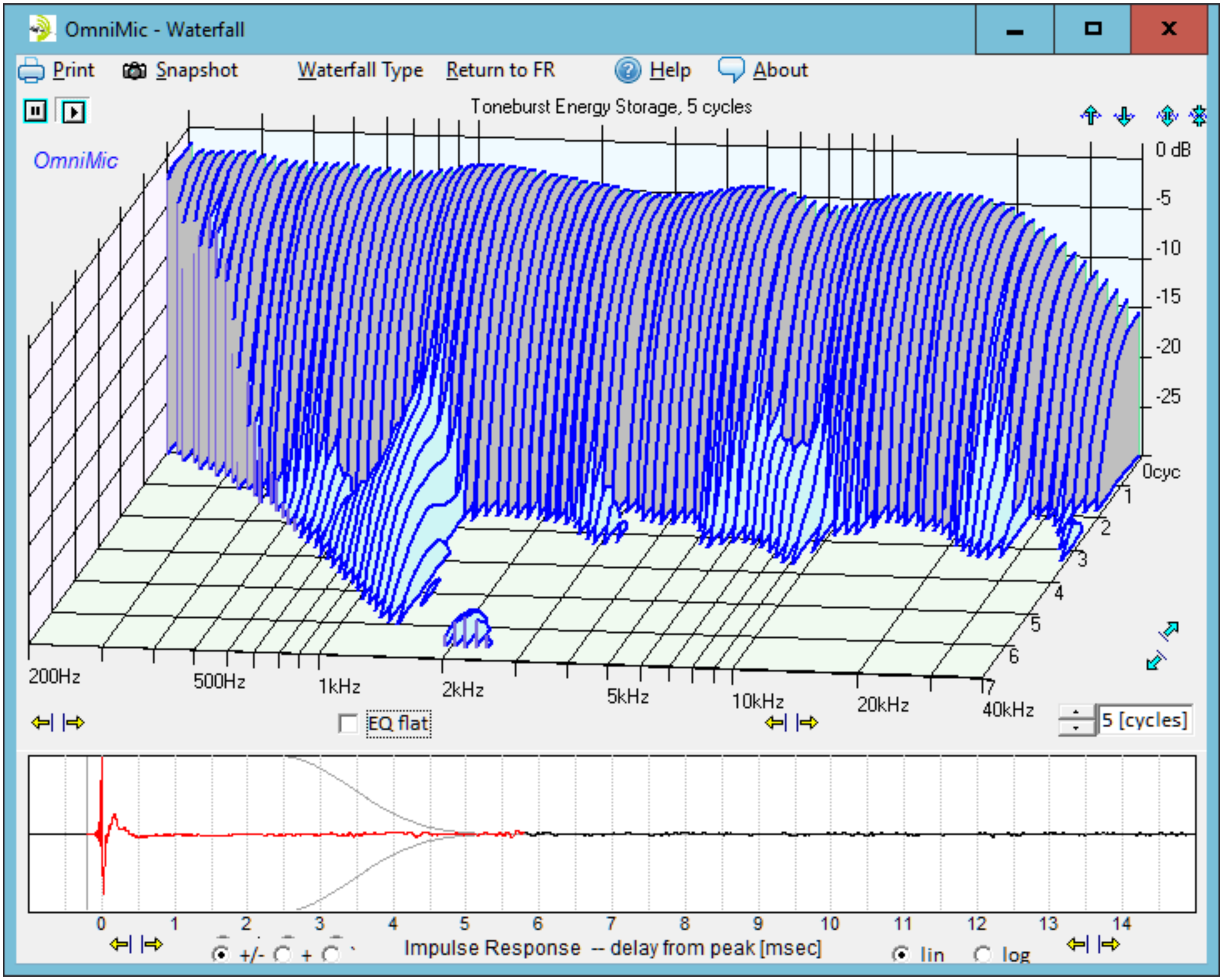
CSD waterfall curves can now be shown with different degrees of smoothing, and can also be used with long time lengths (to 150 milliseconds) for viewing effects of room reflections.



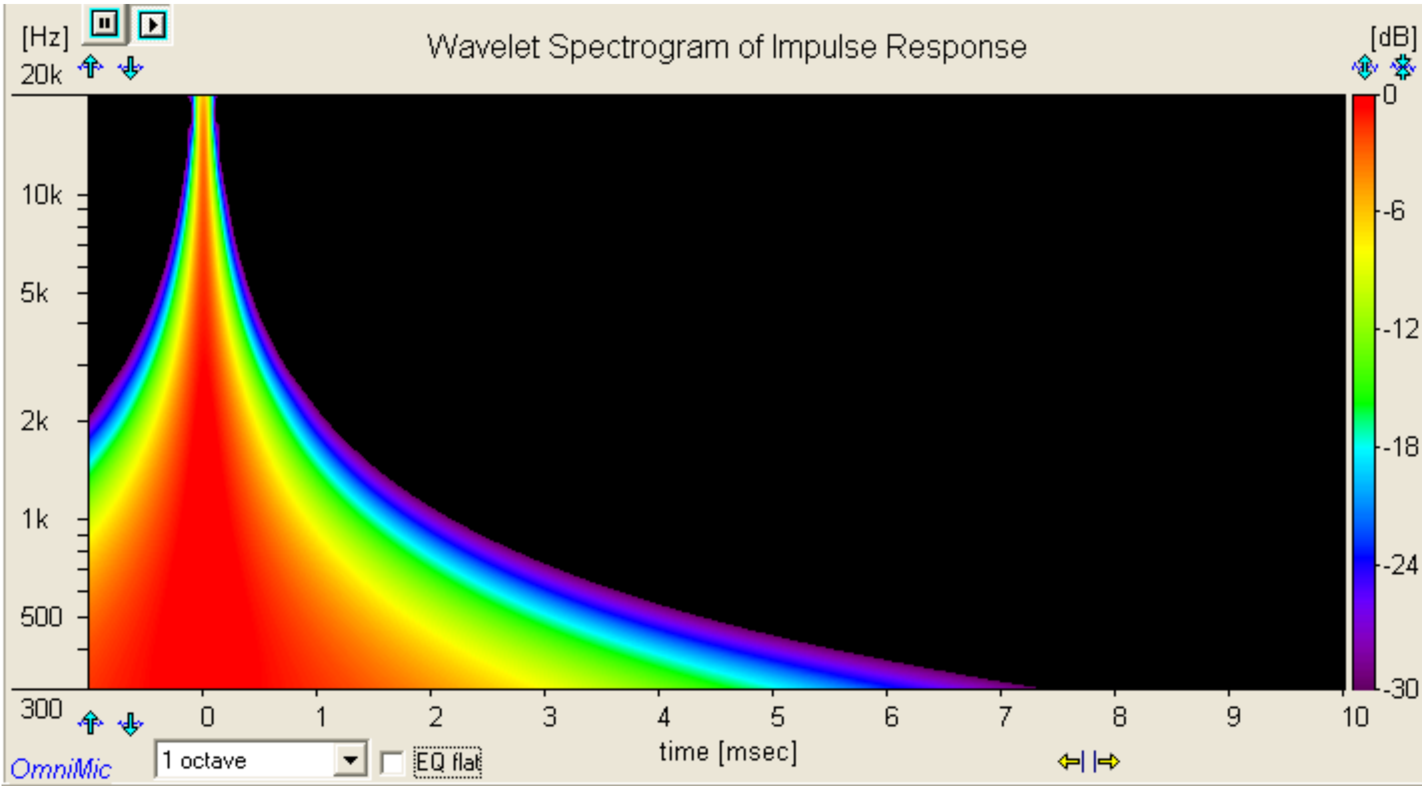
"**Toneburst Energy Storage**" (TES) shows the effect that would occur if the loudspeaker were driven by short tonebursts of energy one at a time, concentrated near each test frequency. The speaker output would ideally end after the toneburst ended, but real world devices will continue to ring as the energy stored within dies out. This is similar to a test devised by Linkwitz using special hardware, but OmniMic can calculate it from impulse responses. The number of applied toneburst cycles can be selected using a control at the bottom right. Like the CSD waterfall, the impulse response can be windowed to remove effects of reflections.

The gray area shows the energy that is expected if there were no storage or hangover. The light-blue area near the floor of the plot represents the stored (delayed) energy at the indicated frequency caused by the driver.

The CSD and Toneburst Energy Storage waterfall plots are useful identifying moderate to high Q resonances in a driver's frequency response. The audibility of the features easily identified in these waterfall plots is somewhat controversial, with some research (see Toole) indicating that the higher Q resonances seen in waterfall displays are significantly less audible than low-Q resonances that do not stand out in CSD or TES waterfall displays. In any event, it should be remembered that waterfall data (and also frequency response data) are simply alternate presentations of information contained within impulse responses.

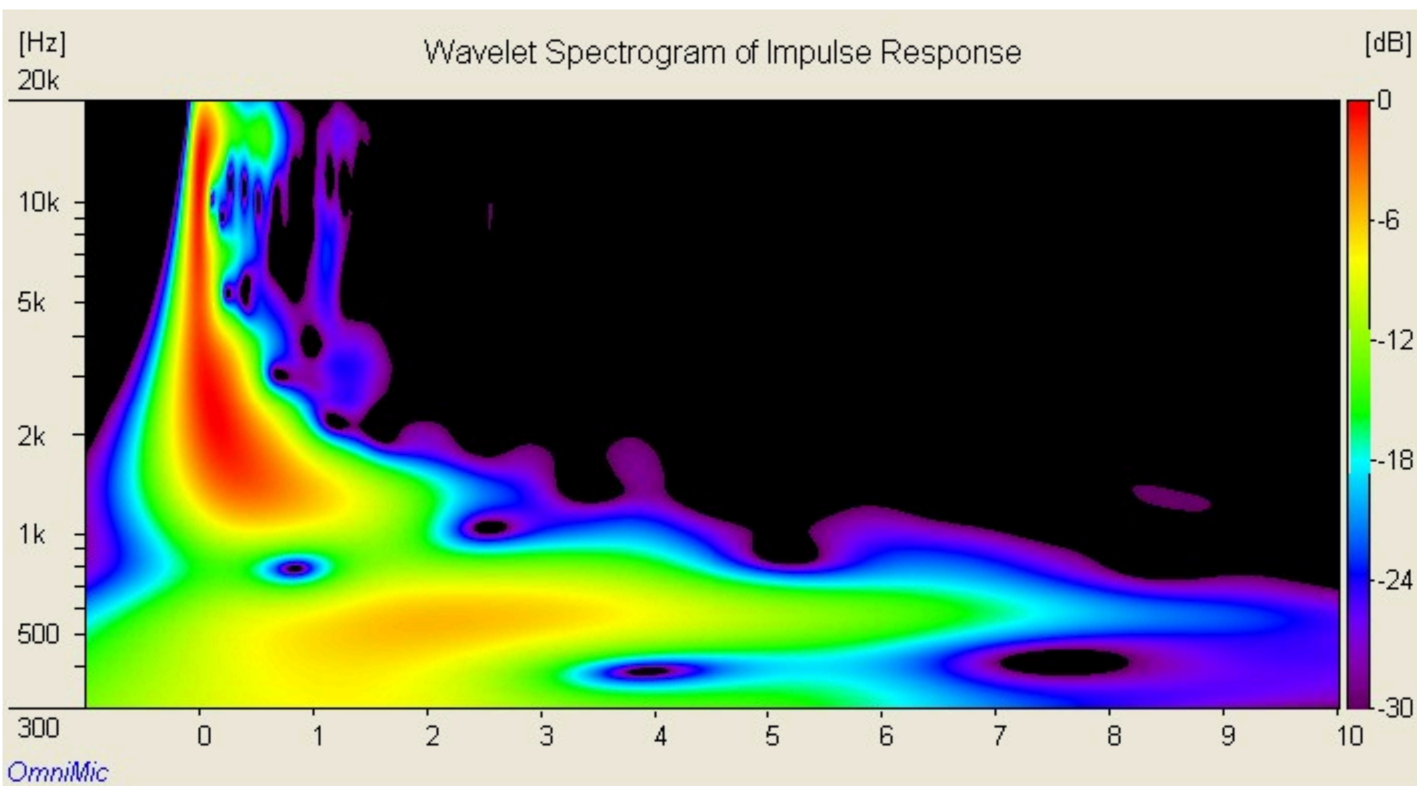


"**Wavelet Spectrogram**" shows a combined time/frequency representation of the impulse response. The Wavelet Spectrogram in OmniMic uses a very fast algorithm that allows the display to occur in real time. In all time/frequency displays there is a mathematical "uncertainty principle" which limits the degree of time resolution that can be obtained for a given frequency resolution, and vice versa. In other words, the more detailed the time character of the display, the less detailed will be the frequency character. The Wavelet Spectrogram shows the optimized presentation, giving as much combined resolution as possible. The horizontal axis is time, the vertical axis is frequency, and color shows the relative intensity (in dB). You can select the octave resolution, in a control below the plot, to determine the desired resolution trade off. An ideal wavelet spectrogram (flat response, no resonances or reflections) will look like a vertical tapered horn, like this:



The time resolution is more detailed at higher frequencies than at lower frequencies (because there are more "Hz" in an octave at high frequencies than at lower frequencies).

A typical loudspeaker will show a less clear graph, with smearing at various frequencies and additional color features appearing at later times where reflection or diffraction occur.



If you spread the time axis out to full length, you can also use the Wavelet Spectrogram for viewing room reflections and the frequency ranges over which they predominate.

Features of the Waterfall Displays

- For CSD and TES type waterfalls, the top of the screen reference line is set by the largest feature over the selected frequency range. Both types also allow selection of an "EQ flat" function that adjusts gain at each frequency, as if an ideal equalizer were applied. Time is shown on the "depth" axis. You can click at the end of a line trace on the labeled axes (time for CSD, frequency for Toneburst) to highlight the single line in a waterfall plot for easier reading.
- For CSD and TES, the position where you click within the Impulse Response graph below determines the length of the waterfall calculation, starting from 0ms..
- For the Wavelet Spectrograms, the red color indicates the highest decibel level in the frequency range. The "EQ flat" button can be used so that red instead indicates the highest level at each frequency (in effect, what would be obtained if the speaker could be equalized flat without affecting its phase). Impulse response windowing will ***not*** have an effect on Wavelet Spectrogram displays.
- The three dimensions (intensity, frequency, and time) of the graph can be adjusted as desired for display using scaling controls similar to those on the other OmniMic graphs.
- As with the rest of the OmniMic graphs, there are buttons provided for both taking Snapshots of graphs or for sending copies of the screen display to a printer.
- Often selection of the Log format display of the Impulse Response graph below will allow for easier location of strong reflections.
- To return to the normal Frequency Response page of OmniMic, click on the "Return to FR" menu button.

Room Equalization with OmniMic

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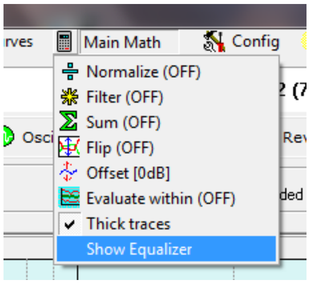
OmniMic enables you to measure the frequency response of your room and speaker system at any number of listening positions, weight the emphasis you want at of the positions relative to the others, and generate the required response curve for your equalizer. The result can be the curve graph itself, or a set of parametric equalizer settings you can enter by hand into parametric equalizers, or which can be loaded as a file to MiniDSP equalizers. These functions are provided by the **Equalizer Configuration form**, which you can reach while on the Frequency Response page, by clicking the "Main Math>Show Equalizer" menu.

Automatic equalizers, such as are built into most modern receivers, do a poor to decent job of equalization, but better results can always be obtained with a parametric equalizer and a human brain involved.

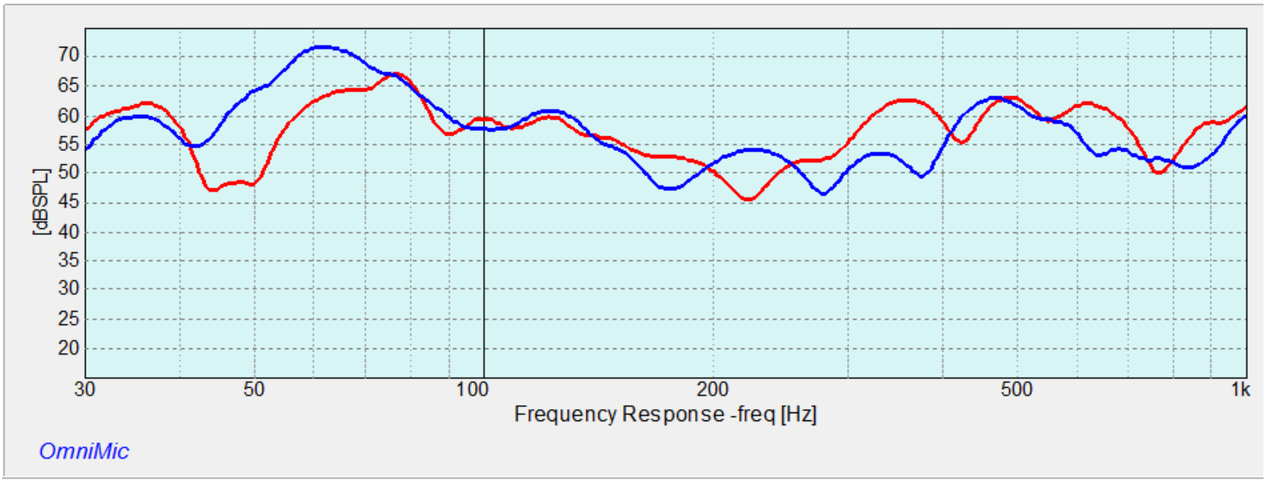


The Equalizer Configuration form can't be shown unless there is an "Average" curve available in the Frequency Response plot, which you can obtain from measurements as discussed below, or by loading an FRD file using the "File>Load to Avg Curve" menu.

About Equalization

The frequency response of an audio system is not a constant, but will be different at each individual seat. Ideally we would like the response to be flat (or some other target curve shape) at all seats from each speaker. But that isn't possible in real rooms with real speakers because of the strong effects of loudspeaker directivity and sound reflections in a room. So the best approach is to adjust the overall equalization to best smooth the response for all of the seating positions, perhaps emphasizing the response at certain critical seating positions that are more often used. With OmniMic, this is simple to accomplish, using its curve Average functions.

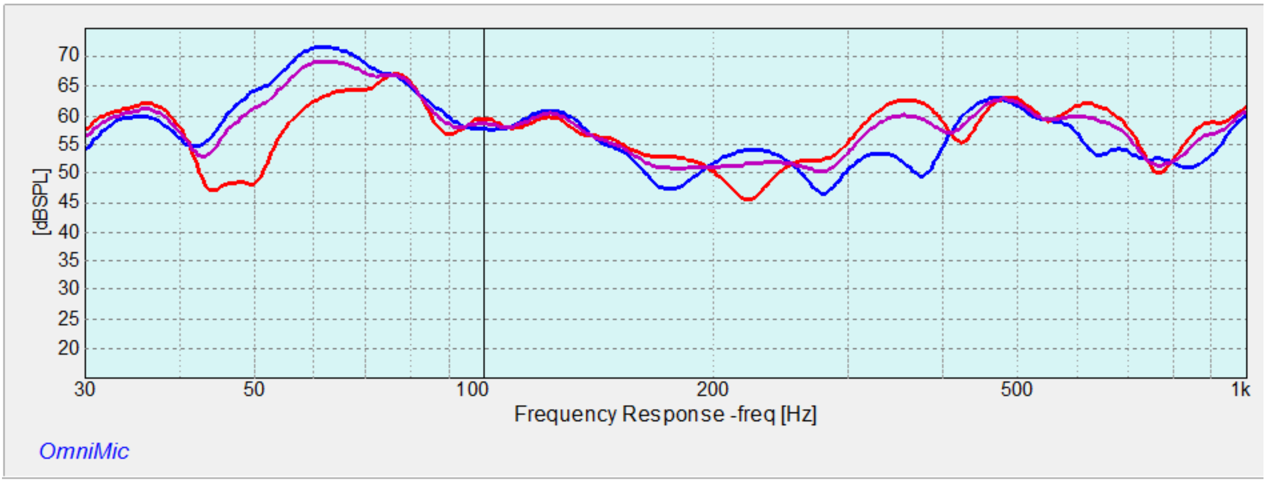
For instance, here is a graph with two responses from two different positions in a room:



Red is at Seat 1, Blue is at Seat 2

If we equalized the response for flat at the Red position, then at 100Hz, that would be fine for both Red and Blue around 100Hz. But Blue would end up with a big peak around 50Hz if it were bumped up as much as would be needed to make the Red flat.

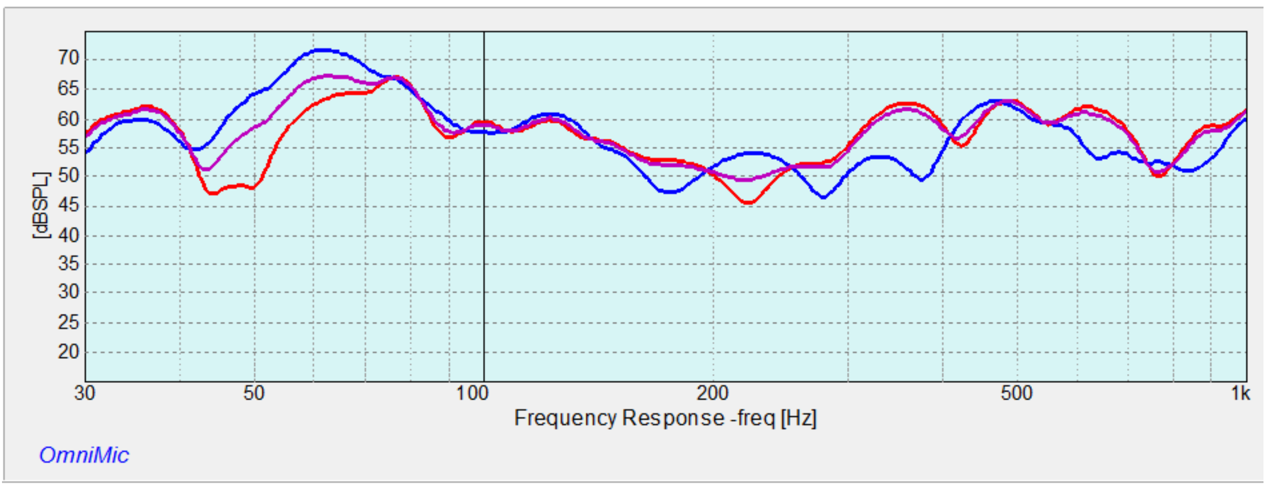
If we equally weighted the responses from both, we would get the Violet curve as shown below:



Violet is power average of Red and Blue curves, equal weighting

Averaging in OmniMic is "power averaging", meaning that peaks shown in dB carry more weight than do dips. This is good, because peaks in response are more bothersome to sound than dips, so we want those to show stronger in the curve we plan to correct from. Remember, if there is a peak in our Average curve, then during EQ we will be working to pull the response down there with the equalizer's response -- peaks in room response become dips in equalizer response to compensate.

But what if the Blue position is hardly every used, or is usually occupied by someone who doesn't care much about the sound balance? In that case, we can just use more weighting for the Red curve as we develop our average curve. That is simply done by introducing more averages from the Red position as for the Blue position. For instance, here is what the Violet "Average" curve becomes if we used three averages from Red but only one from Blue:



Now Violet has Red weighted 3 times as strong as Blue.

You can see that the Violet curve now becomes a little more like the Red and a little less than the Blue.

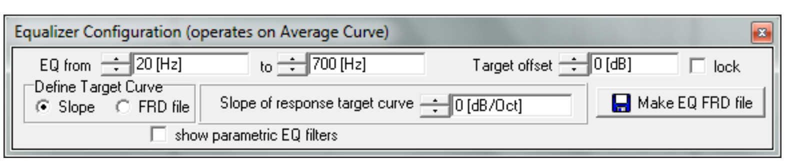
A more versatile method of creating an average curve for equalization is to save each of the measured curves to a file, and then use the "Make Weighted Average" function in the File menu. This brings up a simple table that lets you load and average-in any saved FRD curve file and select the weighting of each when you add it.

Obtaining your Typical ("Average") Response Measurement

To make up your average curve, play one of the "Short Sine Sweep" responses from the speakers while measuring with the Frequency Response tool. Set the "*all, blended, only to*" buttons to "all" (which will include all reflections in the results) and the smoothing to 1/6th or 1/12th octave. Put the microphone at the first listening position, measure the response and save it as an FRD file (using the File menu). Then repeat for all seats of interest. In a large theater, you can save time and just do this for a sampling of seats in different general areas. Then, use the "make Weighted Average" menu (also under File) to load the desired curves (up to 22) and apply a weighting value to each according to their relative importance. The red curve on the graph that appears is the "Average" response curve that you will want to equalize for. You may want to save this average response curve to disk using the "File>Save Avg Curve" menu, which will allow you to reload it later should you wish to do so.

Getting the Equalization Curve

Now, bring up the Equalizer Configuration form by clicking the "Main Math>Show Equalizer" menu. A small form will appear:

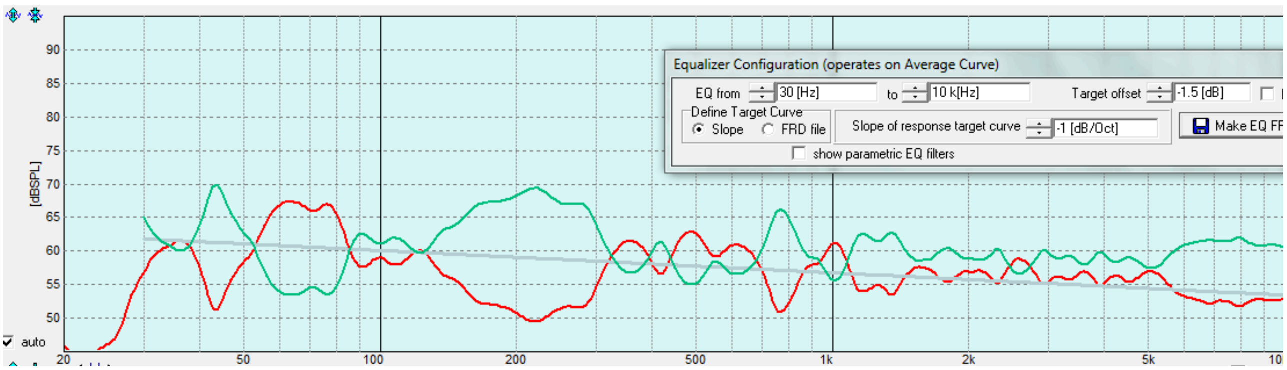


In the form, you choose:

- the frequency range over which you wish to equalize
- the type of target curve: a flat line with a specified slope; or a curve from a text FRD file (which you can select or even create on the fly from an included text editor). The target curve will appear on the Frequency Response form in a thicker gray line.
- the amount (in dB) to offset the target curve. You would normally set this one (visually, watching the gray line and the red Average curve) to the position that requires the least work of the equalizers to approximate the target curve.
- The offset will normally be based on the response within the optimization range. You can also choose to "lock" it so that frequency range selection doesn't further affect it or so that the same target can be used for different speaker channels.

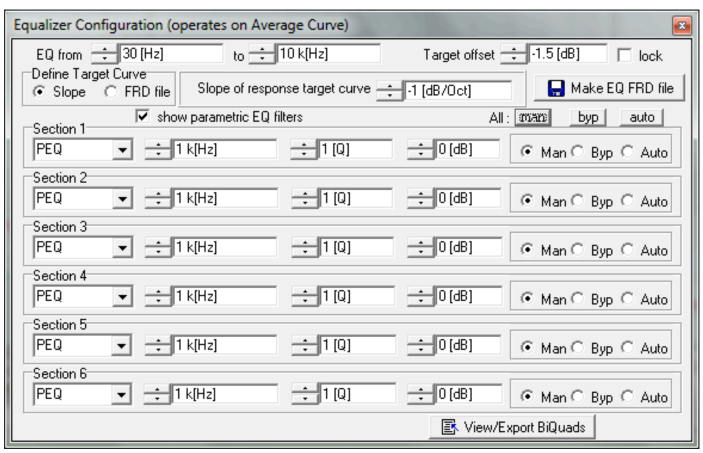
You would typically equalize an in-room response to approximate some "target" curve, which is the response shape you wish the system to have. In many cases, the target will be a "flat" response, but possibly you may find that using a "house curve" or a response with a slight downward slope will sound better with recorded music or video programs. OmniMic makes this easy to achieve and to vary as desired. Equalization may only be desired at bass frequencies in some systems.

If you need to generate an FRD or text file of the response curve, then at this point just click the "Make EQ FRD file" button and tell the program where you want to store the file. This would be the approach to use if you have only a "graphic" type equalizer available. Use the file to see the response your equalizer should ideally be set for; approximation of this curve is usually sufficient, great detail in setting is not needed and is seldom recommended.



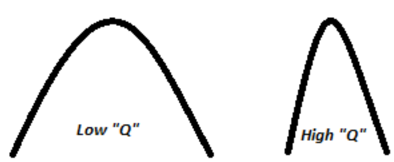
The red curve above is an in-room measured curve, and the green curve is a generated EQ frd file (loaded as an "Added Curve" and offset 60dB for display).

To obtain parametric equalizer settings, put a check in the box labeled "show parametric EQ filters" and the form will expand to show:



Six sets of EQ section controls appear. Each section can be set to PEQ (parametric equalizer), High Shelf, or Low Shelf, and can be set for manual mode, bypass, or automatic mode. You can select the effective frequency, the "Q" (the ratio of center frequency to bandwidth), and the dB intensity of each. The result of applying all the filters will appear as a gold curve on the graph.

For example, for each of 6 sections, you can apply a parametric filter to cut or boost a selected amount (in dB), at a selected frequency, and with a selected "Q". Q is related to bandwidth. A lower-Q setting means that the boost or cut will be effective over a wider range of frequencies, while a higher-Q setting will cause the effect to act over a narrower range of frequencies with a sharper curve.



If you set any of the EQ sections to Auto, then the form will further expand to show buttons and limit settings that the OmniMic program will use to automatically optimize the filters to approach the target response shape. Two buttons are at the bottom of the expanded form, "Reset Auto values and Opt" and "Optimize from current values". You would use the first if you want OmniMic to distribute the available filters over the range and start a new optimization. You can use the second when the current settings (from a previous optimization or from manually set values of certain sections) are to be used as the starting point for further optimization. You can interrupt optimization at any time by using the "Stop Optimizing" button that will appear during optimization. The "err" value is a number that shows the progress of the equalization, so you can tell whether the optimizer is "stuck" (and may need some manual intervention) or is still making good headway. Often a better solution will be found if you stop optimization and then restart with the "Optimize from current values" button several times.

The filter settings can be applied manually to parametric equalizer units (you may need to convert Q to Bandwidth using the formula BW=Frequency/Q). For equalizers such as MiniDSP which accept biquad filter coefficients, use the "View/Export Biquads" button to export a file which can be directly imported. See MiniDSP Equalizer Tuning.

MiniDSP Equalizer Tuning

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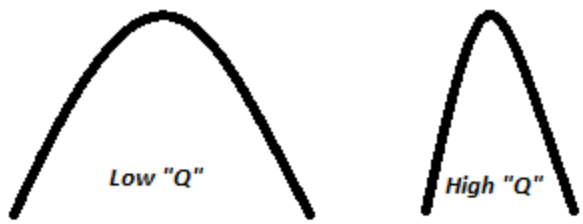
The OmniMic MiniDSP Equalizer function allows you to configure parametric filter settings, manually and/or automatically, and view the result as applied to a Frequency Response in the **Average Curve**. The settings you determine by this method can then be exported in a file from OmniMic, and then imported into the MiniDSP's control program for downloading to MiniDSP hardware. You can adjust up to 6 of the MiniDSP PEQ or shelf filters at a time, and then generate a parameter file for use by MiniDSP. Additional filters can be added, if desired, in sets of 6 (before or after the crossover filters, with MiniDSP's "2-way Advanced" crossover, and some other plug-ins -- see MiniDSP's documentation for details).

For example, for each of 6 sections, you can apply a parametric filter to cut or boost a selected amount (in dB), at a selected frequency, and with a selected "Q". Q is related to bandwidth. A lower-Q setting means that the boost or cut will be effective over a wider range of frequencies, while a higher-Q setting will cause the effect to act over a narrower range of frequencies with a sharper curve.



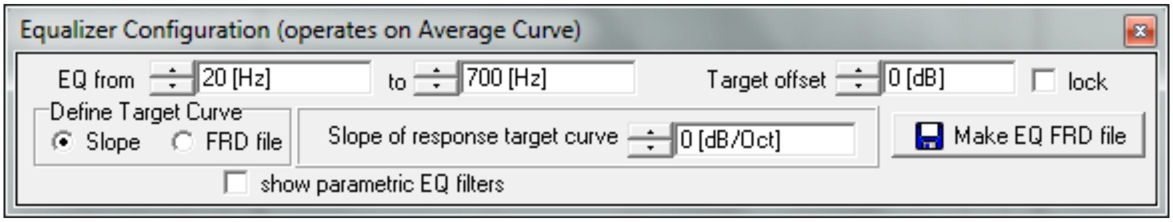
You would typically equalize an in-room response to approximate some "target" curve, which is the response shape you wish the system to have. In many cases, the target will be a "flat" response, but possibly you may find that using a "house curve" or a response with a slight downward slope will sound better with recorded music or video programs. OmniMic and MiniDSP make this easy to achieve and to vary as desired. When you adjust the settings for any of the filter sections, its effect is immediately shown in an additional yellow curve on the plot, along with the unaltered "Average" curve.

When equalizing a room response, one of the errors that users often make is to "over-equalize" -- making tedious narrow adjustments to flatten each dip and peak as seen at a very specific listening position. The problem with doing this is that the response of a speaker and room is extremely position dependent. Ideally, you want an equalizer correction that provides the closest approach to the desired response for ALL likely listening positions (as well as all sitting positions of the listener!). This is why OmniMic has you adjust the equalizer based on the shape of an Average Curve. You can obtain an overall typical frequency response measurement (an "Average"), as determined at all the listening seats, by measuring the response at each and clicking the "New Average" or "More Average" button to include that data into the ongoing Average curve shape (which will show in a red line). Or you can load any FRD file as the "Average Curve" using the File menu.

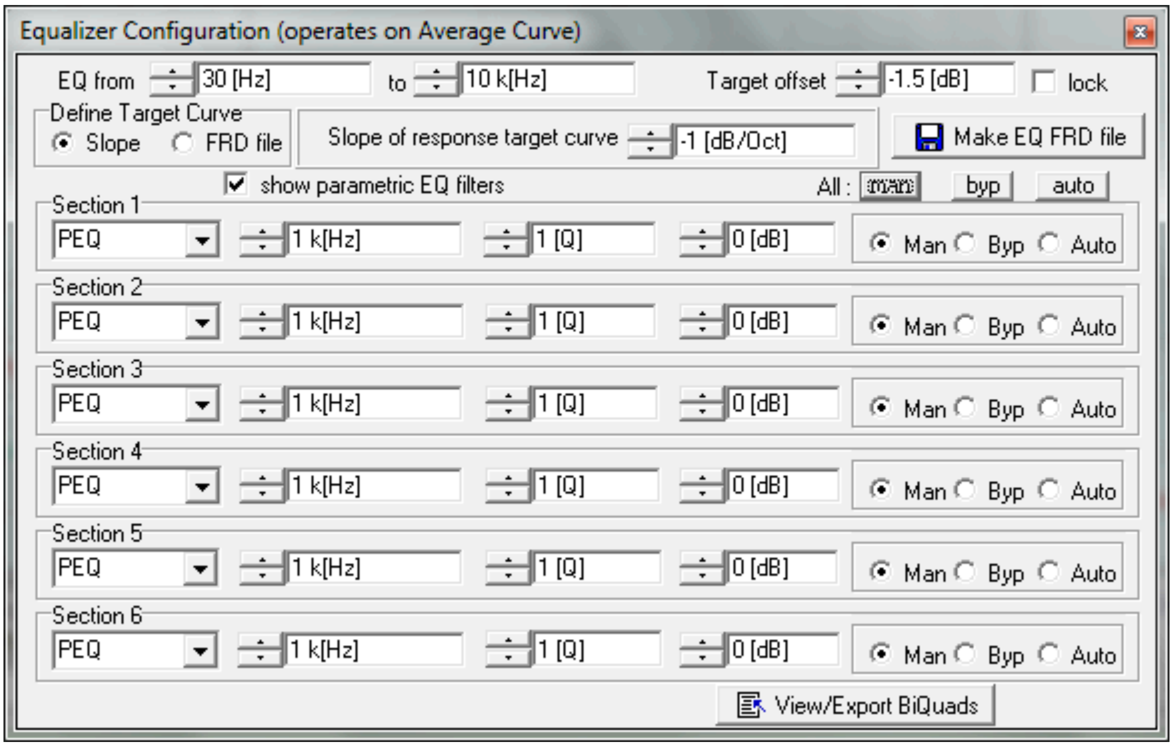
You can also take a collection of measured FRD response curve files, and weight the effect of each to make an Average Curve to equalize from. Select "Make Weighted Average" in the file menu. Operation is simple and intuitive.

If you want to emphasize the response at any particular seat (such as where you normally sit) to correct better for that position while still considering the others, click the "More Average" button several times after measuring at that position. You can also average in the most recently added "Added" curve by right-clicking (rather than left-clicking) on the "More Average" button. Or, you can load a previously saved Average (or standard) response curve using the File Menu ("Load to Avg", "Save to Avg"), this overwrites the existing Average curve (rather than just averaging into it).

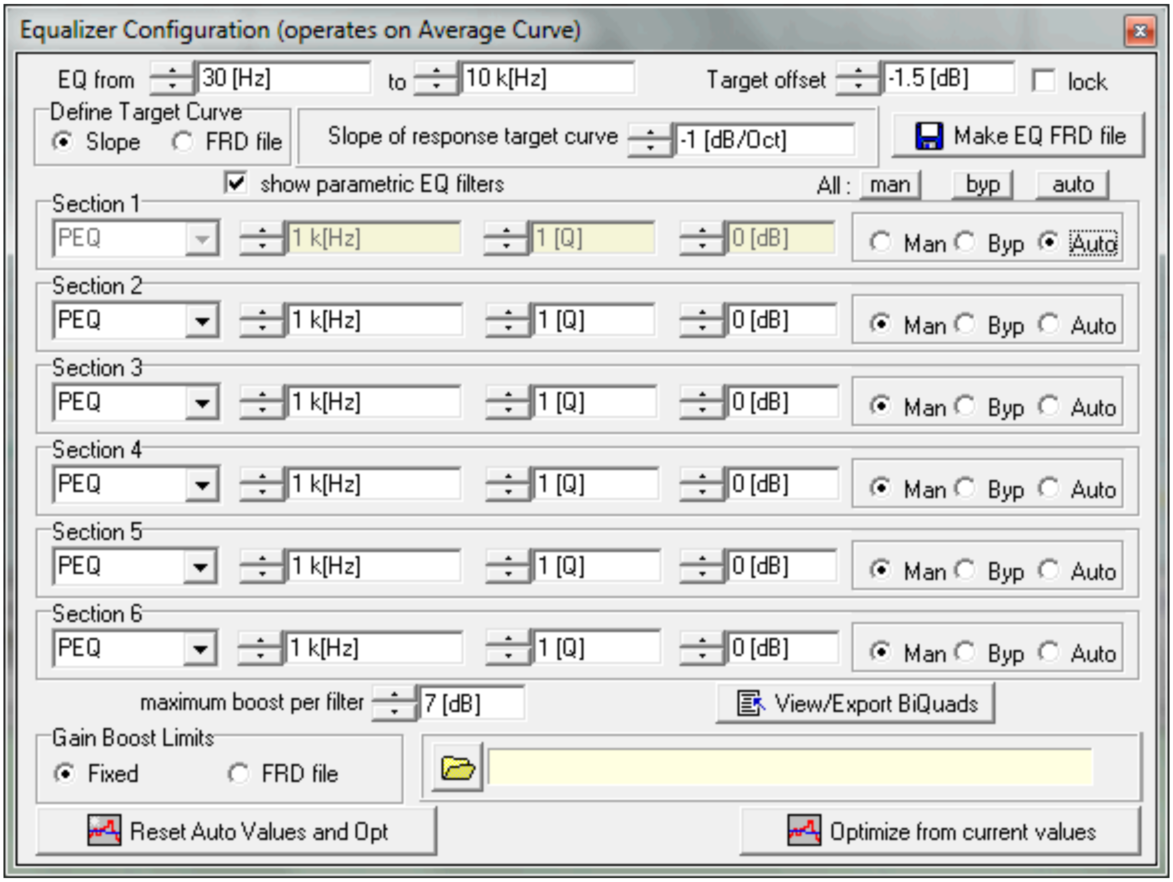
To bring up the equalizer control form, use the "Main Math">"Show Equalizer" menu from the Frequency Response screen. You must have an Average Curve showing to bring up the Equalizer form, which initially looks like this:



Put a checkmark in the box labeled "show parametric EQ filters" and the form will expand to show:



Each Equalizer section can be set for "Manual" or "Auto" operation, or can be bypassed altogether. You can set the mode of all six sections by using the master ("All") buttons at the top. In order to select the frequency, intensity, or Q of a filter section, it must be in manual mode. You can adjust all the filters by eye, if you wish, or have the OmniMic software automatically place and optimize the filters. Or you can set some of the filters only and let the rest be automatically set. When you select any filter to use the automatic optimization, the form will expand to show additional optimization controls:



In the form, you choose:

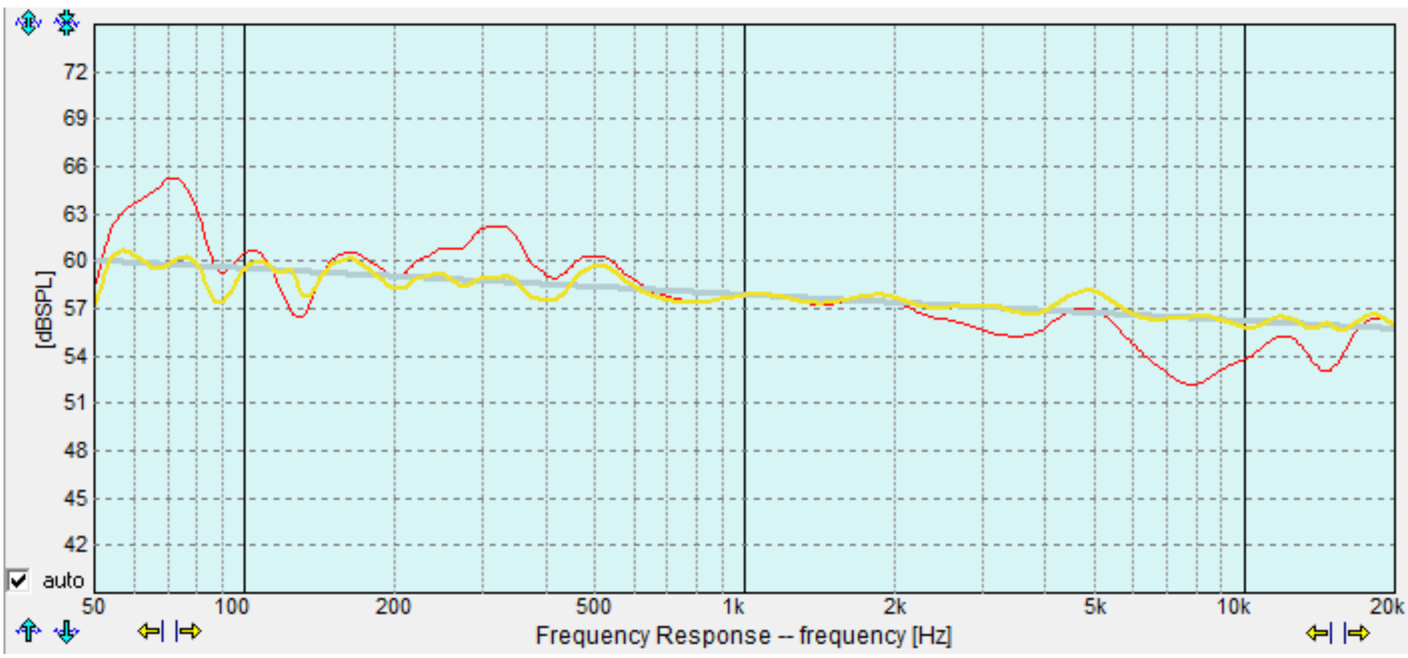
- the frequency range over which you wish to have automatic optimization occur
- the type of target curve: a flat line with a specified slope; or a curve from a text FRD file (which you can select or even create on the fly from an included editor). The target curve will appear on the Frequency Response form in a thicker gray line.
- the amount (in dB) to offset the target curve. You would normally set this one (visually, watching the gray line and the red Average curve) to the position that requires the least work from the equalizers to approach the target curve. The offset will normally be based on the response within the optimization range. You can also choose to "lock" it so that frequency range selection doesn't further affect it or so that the same target can be used for different speaker channels.
- the maximum amount of boost you want the optimizer to attempt to use, to not overstress power amplifiers or drivers in case of a severe response suckout. The limit can be specified as a fixed value maximum boost (for all filters combined at any frequency) or as an FRD curve from a file (which you can again load or create on the fly).
- you can also specify the maximum boost you want any individual filter section to use.

When exporting the parameters to download to a DSP equalizer, make sure that the **DSP sample rate** (as shown at the bottom of the Equalizer Configuration Form) matches the sample rate used by your DSP equalizer. At present, 48kHz and 96kHz equalizers are available from MiniDSP. There is also an option for 192kHz for future expansion.

There are two buttons at the bottom of the form, "Reset Auto values and Opt" and "Optimize from current values". You would use the first if you want OmniMic to distribute the available filters over the range and start a new optimization. You can use the second when the current settings (from a previous optimization or from manually set values of certain sections) are to be used as the starting point for further optimization. You can interrupt optimization at any time by using the "Stop Optimizing" button that will appear during optimization. The "err" value is a number that shows the progress of the equalization, so you can tell whether the optimizer is "stuck" (and may need some manual intervention) or is still making good headway. Often a better solution will be found if you stop optimization and then restart with the "Optimize from current values" button several times.

So, in summary, the steps for MiniDSP room equalization are:

- 1) Measure the frequency response (probably without windowing, using the "All" radio button above the plot) and perhaps 1/6th or 1/12th octave smoothing. Make a number of measurements at different location and average them into the Average curve. See: Room Equalization.
- 2) Open the Equalizer and use shelf sections to manually adjust the upper or lower ends of the response, and possibly to fix obvious peaks. Set the other filter sections for Auto.
- 3) Set the frequency range for the automatic optimization
- 4) Set the target curve shape as desired (a -0.25dB/Octave downward sloped curve is usually a good-sounding choice), and adjust the offset for best fit.
- 5) Set any boost limits you may desire. A 6 or 7dB limit (or lower) is usually a good idea. Also make sure the DSP sample rate at the bottom of the form matches your DSP equalizer hardware.
- 6) Click "Reset..and Opt" or "Optimize.." to start the process.
- 7) Stop when you are satisfied, or when equalization has run out of gas, or to change some settings before optimizing some more with the "Optimize..." button.
- 8) When you are done, click on the "View/Export Biquads" button to bring up a display of the values that will be exported to a file for MiniDSP. Save it to a file, and then import it into MiniDSP for loading to the equalizer hardware.
- 9) Check the response at various seats to see the general improvement, then do some listening.



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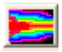
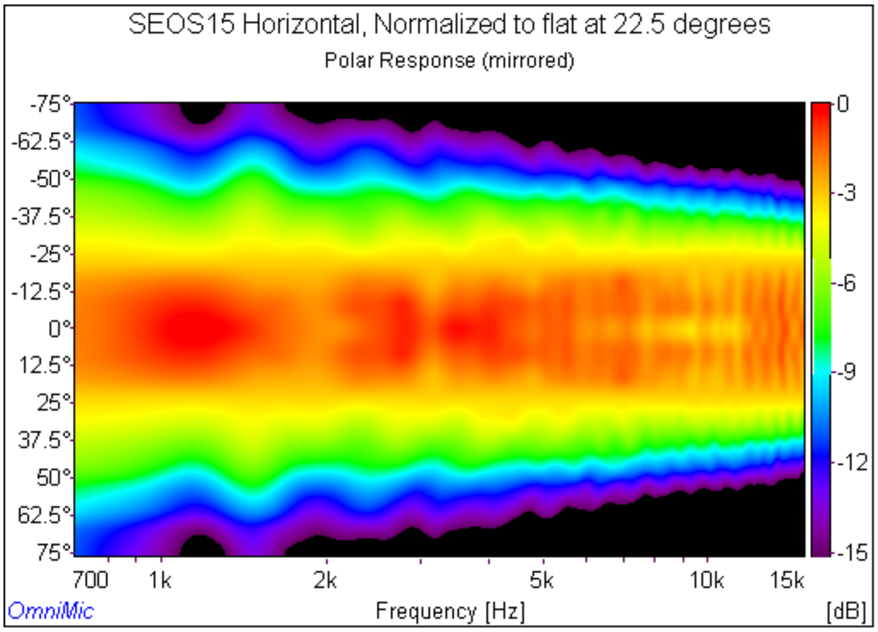


Polar displays are provided by a post calculation using multiple saved frequency response data files ("**AddedCurves**") added the OmniMic Frequency Response screen. The files must each have angle values assigned to them, and there must be at least three "Added" files present with different angles assigned in order for a calculation to be possible. In general, seven or more files should be used for good results. When you enable polars with the **"3DPlot-->Polars" menu**, the OmniMic system will pause (halt incoming live measurements) as the Polar Display calculations are very intensive and would interrupt live processing.

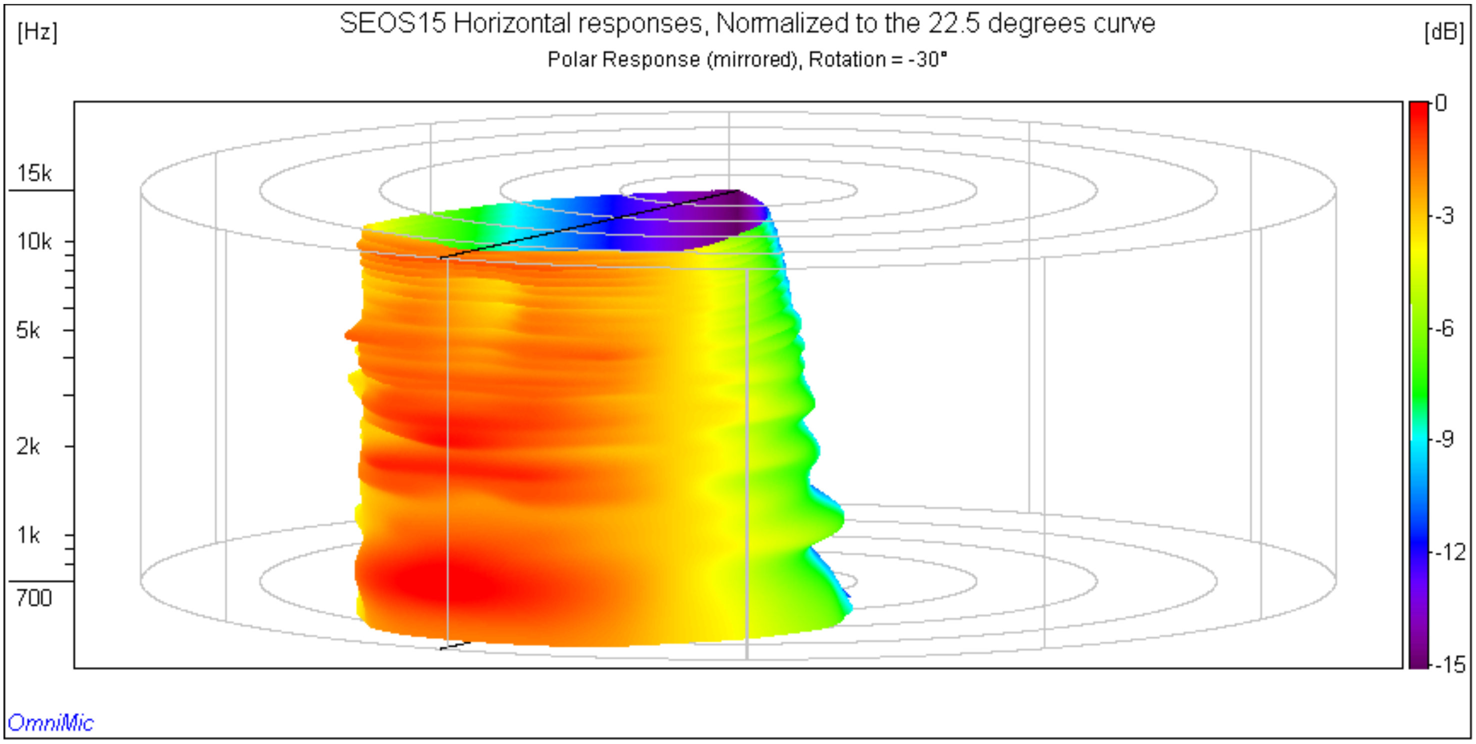


The purpose of a Polar Display is to reveal how the frequency response of a loudspeaker varies with horizontal or vertical angles from the baffle. Speaker designers generally design loudspeakers for a specified (usually, more or less flat) frequency response at a position on-axis of a speaker and at some assumed distance from the baffle. But such a response is not what a user actually hears in a real room -- though some sound that projects at off angles isn't initially aimed at a listener, that doesn't mean that its effects won't be heard. If you point a speaker away from you, you will still hear it very nearly as loudly than as if it was pointed at you. You may hear it a few milliseconds later, but it will certainly not be insignificant. Usually, most of the energy you hear from a speaker actually wasn't initially directed at you, but is reflecting from around the room before reaching you.

Research indicates that users generally prefer that the spectrum of reflected sounds should resemble a flat (or smoothly decreasing) response relative to that from the directly arriving sounds. This has generated interest in the polar radiation patterns of loudspeakers, and in designs intended to address these patterns -- dipoles, omnidirectional, bipole, arrays, or waveguides.

To see the response magnitude (dB) varying with both frequency and radiation angle, a 3-dimensional graph is required. OmniMic provides two versions:

- A "flat" format, as shown above. In this format the horizontal axis is frequency, the vertical is radiation angle, and the *color* represents corresponding dB level. An index relating color to dB levels is shown to the right of each plot.
- A "cylindrical" format, in which frequency is the vertical axis, the angle around the projected cylinder is radiation angle, and both color and distance from the cylinder axis represent dB level. The colored region of the graph can be rotated using provided buttons to give a more intuitive view of the response shape than is obtainable by colors alone. This format can be time consuming to process, so the "density" can be selected to trade-off graphic quality versus time. Smaller form sizes for the plot also take less time to calculate, so you may wish to drag the Polar form to a small size.



The graphs assume that the highest level in the included frequency range (of all included curves) is displayed as "0dB" (red). If you are investigating a driver (or horn, or waveguide) that hasn't been equalized (or voiced in a crossover), it is best to select the "**Curves>Normalize**" menu and choose one of the curve angles to reference the others from. The result would then be the pattern you could get were you to perfectly equalize the response as seen from that angle. Typical normalization angles are for 0 degrees or 22.5 degrees (for toed-in waveguides).

Normalizing the responses, however, can result in some apparently large peaks at frequencies where the reference curve has low output. Such peaks would dominate the "0dB" value, so you may want or need to adjust the frequency range of the polar plot to avoid ranges where these peaks appear.

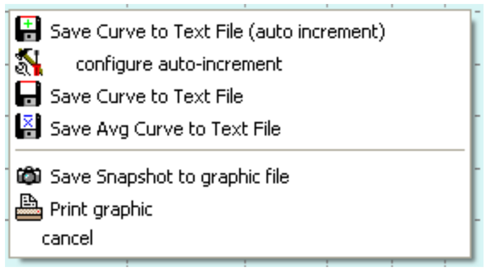
There is a set of example FRD files which can be loaded all at once by going to the Frequency Response menu "*Curves>LoadCurveList*", then browsing to ***C:\Users\Public\OmniMic\SEOS15 Examples*** and loading "**All Curves**". After loading this, click on the Polar button to see the effect. Then try normalizing the files by going to "*Curves>Normalize*", then browsing to the same directory and choosing one of the curves (the 22.5 degree curve is used for some of the illustrations on this Help page). Adjust the frequency range of the Polar Display and note the effects -- if the lower frequencies below 700Hz (which are full of artifacts in the example), or the frequencies above 17kHz (also full of normalization artifacts) are included, the artifacts will dominate the display's 0dB level. Adjust the frequency ranges to exclude these. The result should be approximately the effects of (hypothetically) equalizing the speaker's response to be flat at the normalization angle in that frequency range.

Making Measurements for Polar Displays

For good quality Polar Displays, echoes should be minimized in the measurements. Set the speaker out in a clear area so that reflected signals are delayed as much as possible. This will allow them to be avoided at higher frequencies (see "Only To" and "Blended" in Frequency Response). Either the microphone stand or the speaker can be moved to arrange for each angle for measurement. Steps of approximately 7.5 degrees or less are preferred for good detail. The program will interpolate between the steps, and if only positive (or only negative) angles are given, will mirror the measurements to the opposite side (this will be accurate, of course, only for symmetric speakers or drivers). Try to measure out to at least 75 degrees from the baffle axis on each side -- unmeasured positions will not be represented in the plot. Dipole or Bipole speakers should be measured a full 360 degrees.

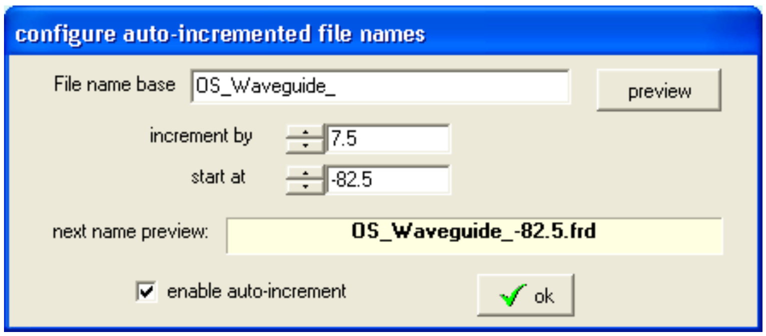
Name each file so that you can identify the angle it was measured from -- if you include the number (-180 to 180 degrees) in the file name, the OmniMic software will try to infer the angle from the file name when you later bring in ("Add") the files to a frequency response page.

Tips:
If you save the files by right-clicking on the graph and then choosing "Save Curve to Text File (auto-increment)" then the program



will automatically name each file by incrementing a number in the file name.
You can also reach this menu by pressing the [Enter] key when the mouse is within the frequency response curve area -- then, saving a successive curve after each speaker or microphone move is a simple matter of pressing [Enter] three times!

Initially you should configure the base file name and the increment value (and starting value), as well as the folder to which you wish



to save the FRD files, by choosing "configure auto-increment".

Be sure to save the collection of Added files to a File List after assembling them, for easy future retrieval of the set!

Is is best to collect all these files with Smoothing turned OFF. Smoothing can be added to each curve when it is loaded in as an 'Added Curve', or to all loaded curves at once by clicking "Smooth All" in the Added Curves menu.

You can add many Added Curves from a common folder at once by clicking on the first, the hold the Shift key while selecting the last. Each will be automatically assigned a different color (based for the most part on the old standard electronic resistor color code sequence!).

A grid/protractor tool to assist in arranging angular measurements can be printed from the Help page at Polar Protractor.

Polar Protractor

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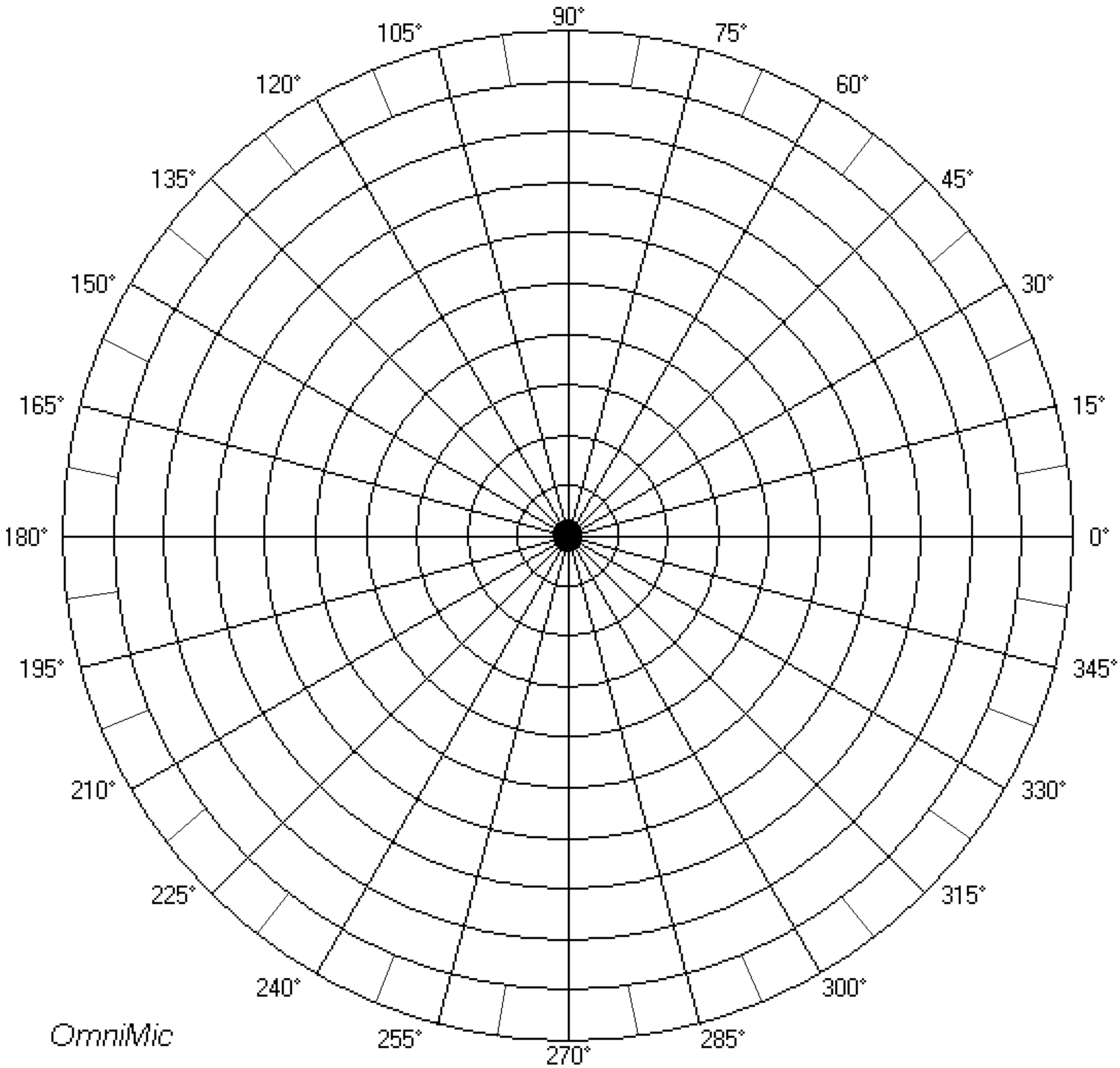
This protractor image is provided to assist in arranging OmniMic and loudspeakers when collecting responses for Polar Displays. Print this to a sheet of paper and place at the speaker position.



SPL/Spectrum

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Use the SPL Meter/Spectrum Analyzer type to see sound levels from any acoustic source. The Generator (in the top menu) can be used to generate one or two tones, or the Spectral Contamination tool can be used to generate larger numbers of tones. Either signal can be recorded to WAV files which can optionally be played on most modern audio/visual equpment). This tab page provides a graph of level vs frequency and a meter indicating broadband level.

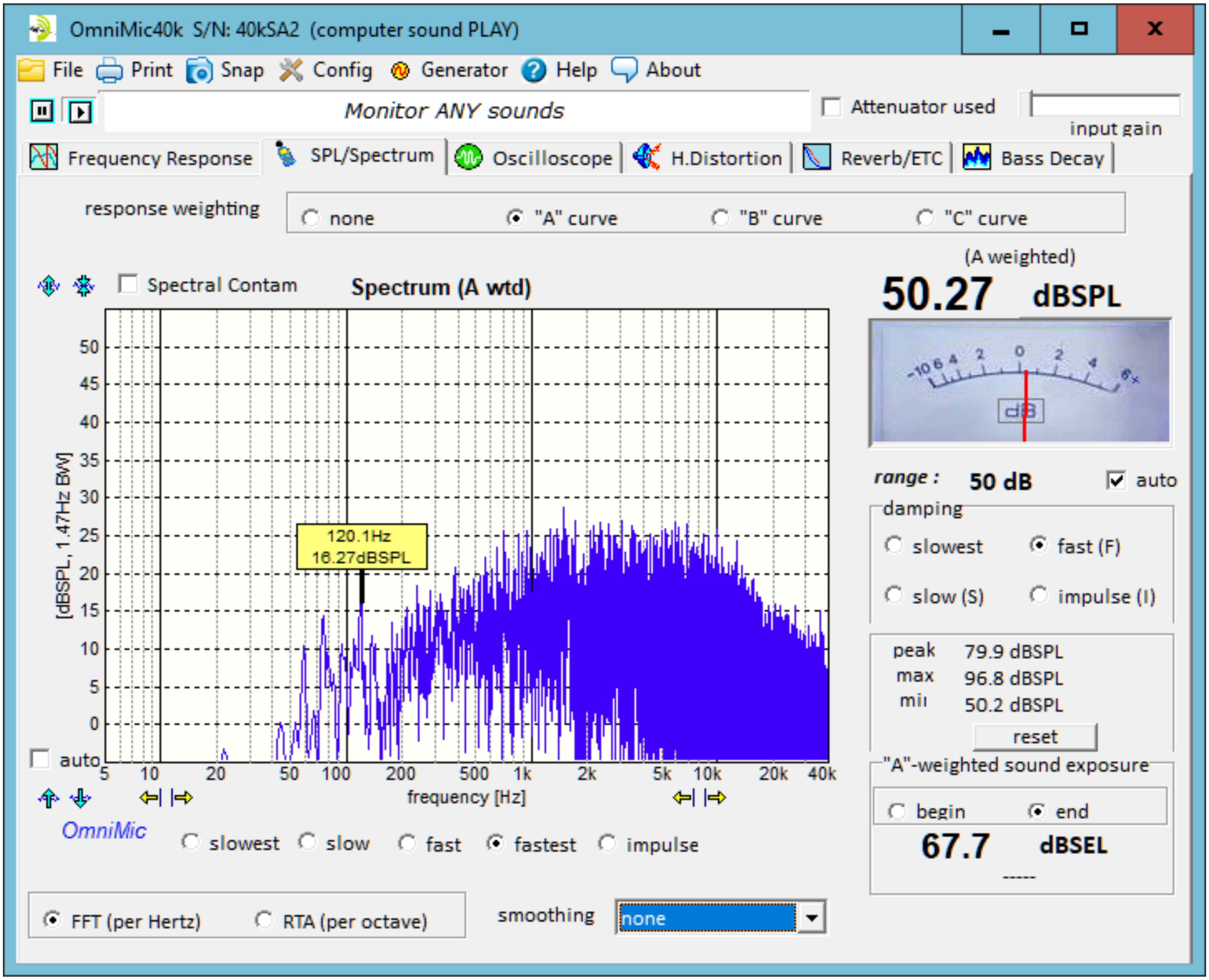
The level of any sounds on the SPL meter face.
Options:

- Select the meter damping type: Impulse, Fast, Slow or Slowest. Damping determines how fast the needle moves in changing sound levels. With Impulse type, there will be some error for lower frequencies, so if those are of interest, choose Fast instead of Impulse.

- Read the Peak, Maximum or Minimum values sensed since the Reset button was last clicked
- Choose the response weighting to use at the top of the tab page: A, B, C, or None.
- When A weighting is selected, read the cumulative Sound Exposure Level (SEL) over time spans starting from a click of the Begin circle to a click of the End button. The time span is shown on the readout along with the Leq (equivalent level).
- In the Config menu, use the option "Show Large SPL" to change to a full screen display of the SPL level (only), with the settings pre-selected above, in large characters for easier reading at distance for level monitoring applications.

The spectrum of any sounds, on the Spectrum Analyzer graph.
Options:

- Choose the FFT format for display in terms of 1.47Hz bandwidth. This format allows you to choose the amount of smoothing applied.
- Choose the RTA format is more traditional and simple, for display in terms of 1/6th octave frequency bandwidths.
- Choose the response weighting shape to use for either at the top of the tab page: A, B, C, or None.
- **Don't use the Spectrum Analyzer to measure frequency response** -- the Frequency Response analyzer is much better for that function!
- Select the damping type for either analyzer graph from: None, Impulse, Fast, Slow or Slowest. This determines how quickly the trace rises and settles in presence of changing levels.
- The FFT analyzer has two options "Noise" and "Tones" when smoothing is being used. These correspond to the types of signals being measured. If a set of fixed frequency tones are being displayed (such as for sine waves or their distortion products), use "Tones". If broadband signals, use "Noise". This is because an analyzer must measure signal within some bandwidth, and pure tones have zero bandwidth. Without smoothing, each point shows the energy within each 1.47Hz bandwidth and separated tones, though at correct level, will appear as barely visible lines. When smoothing is used each tone appears wider, but each point on the trace is an average of levels over nearby frequencies. If tones were input with a smoothed plot, it would cause those averaged levels to appear lower than the tone level because of low energy in most of that bandwidth. The "Tones" mode compensates approximately for that (but noise in the plot will appear higher).
- The FFT Spectrum Analyzer is also used in Spectral Contamination measurements.
- Up to 9 **"floating" markers** can be placed on an RTA or FFT trace. Position the mouse cursor over the graph at the frequency you want the marker placed at, and tap one of the keys for "1" through "9". Tapping "0" will allow you to clear all floating markers. Right-clicking on a floating marker will allow you to delete it individually.


OmniMic40k S/N: 40kSA2 (computer sound PLAY)
File
Print
Snap
Config
Generator
Help
About
Monitor ANY sounds
Attenuator used
input gain
Frequency Response
SPL/Spectrum
Oscilloscope
H.Distortion
Reverb/ETC
Bass Decay
response weighting
none
"A" curve
"B" curve
"C" curve
(A weighted)
Spectral Contam
Spectrum (A wtd)
50.27
dBSPL
[dBSPL, 1.47Hz BW]
120.1Hz
16.27dBSPL
auto
frequency [Hz]
range :
50 dB
auto
damping
slowest
fast (F)
slow (S)
impulse (I)
peak 79.9 dBSPL
max 96.8 dBSPL
min 50.2 dBSPL
reset
"A"-weighted sound exposure
begin
end
67.7
dBSEL
OmniMic
slowest
slow
fast
fastest
impulse
FFT (per Hertz)
RTA (per octave)
smoothing
none

Oscilloscope

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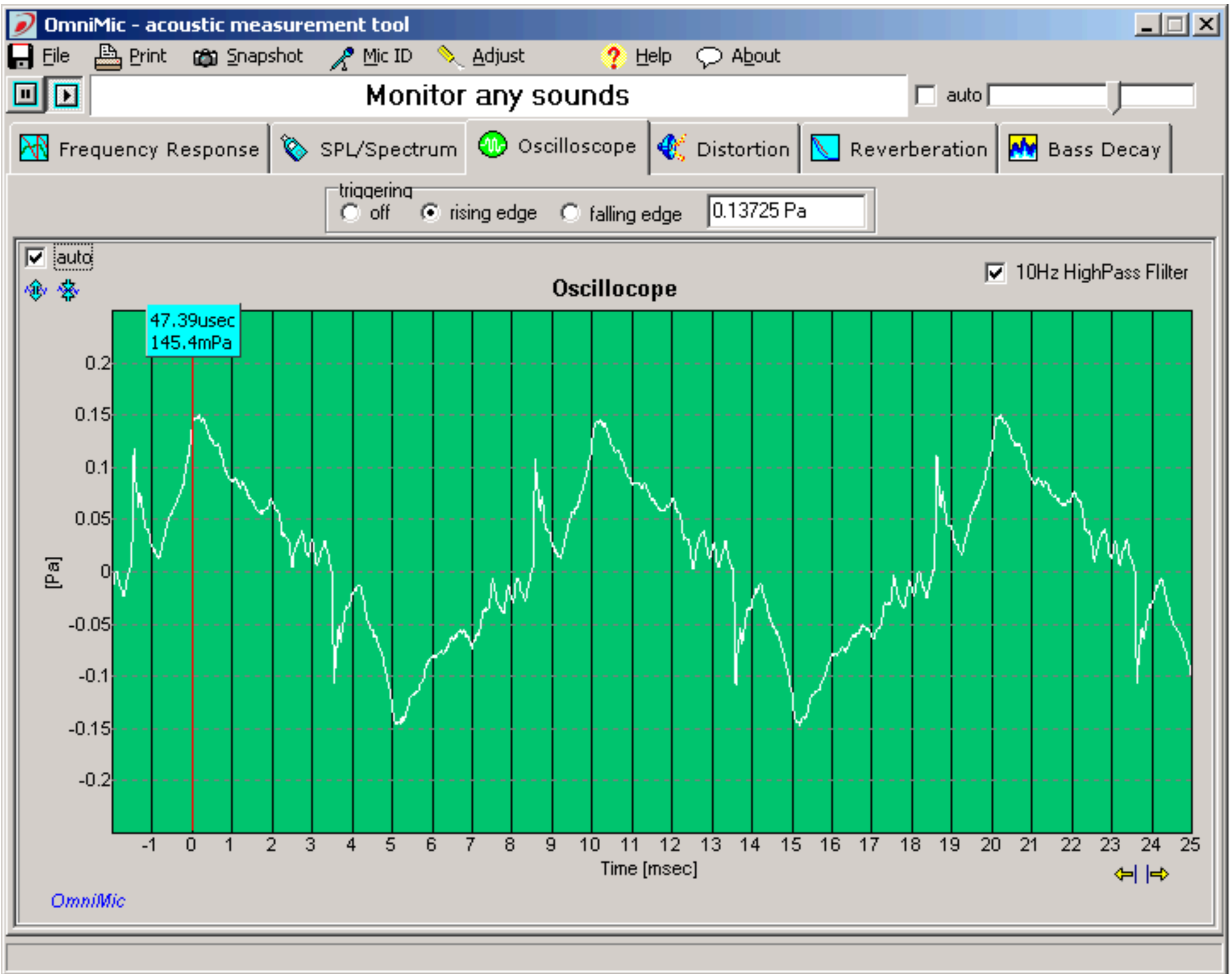
Use the Oscilloscope to view any sound waveforms. These might include music, your voice, or waveforms played by loudspeakers.

You can choose to trigger each "sweep" from the point where the acoustic pressure level rises or falls past a selected acoustic pressure levels. To select, click the mouse on the graph at the desired level. This makes display of repeating waveforms easier to see. To make the trace free-run, click triggering to "off".

- You can freeze the oscilloscope display to better examine a captured waveform by using the Pause button (two vertical bars) near the top left of the OmniMic screen. Click the Play button (forward arrow) to begin the display again.
- If the waveforms being examined are cluttered with subsonic noise in the room, select the "10Hz High Pass Filter" checkbox above the graph to remove the lower frequencies from the display.



Signal Generator

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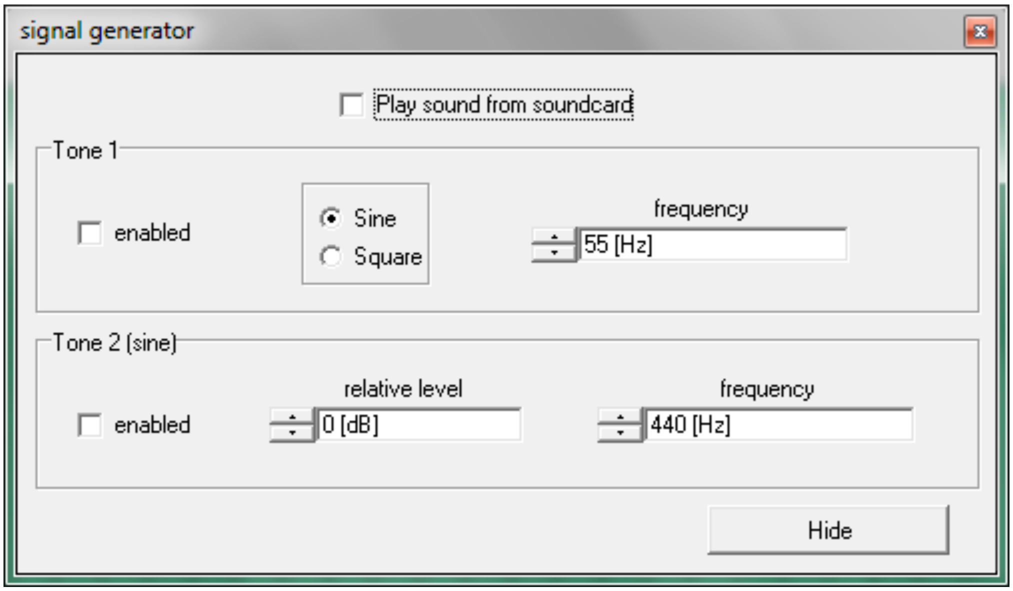
With most computers, OmniMic can be used to directly play test signals such as the sweeps for Frequency Response or Harmonic Distortion. The signal is generated from the computer's soundcard output, which can be connected via cabling to other amplifiers or audio systems.

When OmniMic is used in the SPL/Spectrum mode or Oscilloscope mode, it can also generate sine wave or square wave test tones. To use this function, select the "Config">"Generator" menu and this form will appear:



There are two tone generators included, either (or none) of which can be used simultaneously.

Tone1 can be either a sine wave or a square wave of any audio frequency and is always at a fixed level -- this level is the same (when a sine wave) as the level at which sweep tones are output from the soundcard when sound is played out in the Frequency Response, Distortion, Reverb or Bass Decay.

This provides a handy way to provide known signal level to loudspeakers under test. With Tone1 set to about 55Hz sine wave (from the Generator or the button on the Souncard Play Adjust form,) and applied to a power amplifier which is in turn connected to an AC Voltmeter or DVM set to AC Voltage mode, adjust the amplifier's volume control to the desired voltage level. For loudspeaker sensitivity measurements, this is normally a voltage level of 2.83Vrms (equal to 1W into an 8 ohm load). Then change over to the Frequency Response page, connect the speaker, and with the soundcard again providing the test tone measurements will be at the 2.83V standard level.

Square wave mode with Tone1 can be used to view (on the Oscilloscope) the response of speakers to a square wave. Be aware, however, that very few loudspeakers can produce a recognizable square wave over any range of frequencies. Also, as the measurement system operates between 5Hz and 20kHz or 40kHz (and square-looking waves require from 1/10 to 10x the square wave frequency), only square waves between about 50Hz and 2kHz (or 4kHz) could approach ideal appearance. There is little if any evidence that ability to reproduce a square wave is audible, but the characteristic can still be interesting.

Tone2 is always a sine wave, and both its level and frequency can be adjusted as desired. Its output level (in dB) is relative to the level of Tone1, that is, when the relative level is set to 0dB, then Tone2's level is the same as Tone1's level. Application of the two tones simultaneously can be used to conduct frequency-pair intermodulation tests of loudspeakers, viewing the level of intermodulation product frequencies on the FFT Spectrum Analyzer (in the Spectrum/SPL tab page). When doing this, however, be aware of complications from sound reflections in the room -- such tests are best done with the microphone close to the speaker (if the total SPL level is less than about 110dBSPL or 145dBSPL with OmniMic40k) or outdoors where reflections can be avoided.

Test signals generated by the OmniMIc software can also be saved to WAV files for playing from external audio hardware.

About Distortion

A perfect system could reproduce sound waveforms -- exactly -- in only one way. But there are infinite ways a real workd system could reproduce sound imperfectly, which is the subject of distortion. Overall distortion could be detected using something like an "`Audio DiffMaker`" tool, which can determine whether or not some particular signal has been changed at all by what you are testing. But that is an extremely stringent test and doesn't provide much informatiom about causes or what someone might do to prevent the distortions -- or whether it might even be needed to correct them at all. Distortion can be categorozed as either **linear or non-linear** distortion and, to be practical, refers only to what a system does when presented with some sort of test stimulus (you can't play ***music*** at 0.1% THD, which would be a meaningless concept -- only test signals).

Linear distortion in a system refers to its volume scale, frequency response, time response, or phase response effects, and these can be investigated with OmniMic using the Frequency Response tabpage. Basic frequency response is the most common property of a system that OmniMic is used to find, and is done by playing either a sweep (a.k.a. "chirp") signal or a pseudo-noise signal and displaying the resulting curve on OmniMic's main graph.. You can display phase response characteristics on the same graph. Various types of combined time-and-frequency response characteristics can be displayed as waterfall graphs. Frequency responses can vary toward different directions from a loudspeaker, and you can create and view that information in the form of a polar display. Since a loudspeaker and room will always modify the overall system's frequency and time response, desirably or not, OmniMic software provides additional tools for Reverb/ETC echoic room analysis.

Non-linear distortion

When people refer to distortion, though, they usually mean some kind of non-linear distortion. These are (mostly) changes to sound signal than can't be corrected beyond frequency, scale (volume), and simple time related changes. OmniMic software provides for three types of non-linear distortion testing.

Harmonic Distortion (HD), is measured by applying a stimulus of a singe-frequency tone (or more conveniently, a log-sweep) to the system and seeing what other tones at frequency multiples of the applied "fundamental tone" get created. Total or individual harmonic distortion is familiar to most people as it is simple to understand, repeatable, and quantifiable. Numbers can be assigned to results for comparasons, though except in extreme cases, they seldom are found to much relate to perceived sound quality. However the simiplicity and standardisation of HD tests can show that ***something*** is wrong (and sometimes suggest **what** is wrong). Harmonic distortion is particularly helpful as an aid to determining or specifiying amplitude capability -- you can state how much SPL or how many watts can be produced before some (usually gross!) amount of HD starts to appear and compare that to results from other power producing devices such as power amplifiers.

Intermodulation distortion (IM) characterization is similar to harmonic distortion, but it uses two or more tones applied and looks for multiple sum and difference frequencies of those to be created. There are far more variations possible for intermodulation distortion measurement, both in the relationships between the frequencies and what product frequencies to look at. There is still not significant correlation however between simple intermodulation distortion numbers and what listeners hear. A more extensive and general type of intermodulation distortion is **Spectral Contamination**, in which very many (usually logarithmically spaced) frequency tones are applied at once to the system and the comnbined level of energy resulting at any other audio frequencies is analyzed. There are of course numerous possibilities for the frequencies of applied tones with Spectral Contamination, but at least there's just one result characteristic being collected for all the possible orders of intermodulation products!

While HD, IM, and Spectral Contamination all look at energy produced at frequencies **other than** those being applied to the device being tested, a **Compression Test** is a non-linear distortion test that looks at only how output at the **same** applied frequency changes as level is increased. A compression test can only be done in terms of output level changes in comparison to expected levels.

Quantifying most non-linear distortion results will be complicated by the frequency response of what is being tested, including the effects of the room on the actual frequency response of loudspeakers. A frequency response valley or a room null that occurs at a harmonic frequency of interest could make the harmonic distortion of a loudspeaker look far better than it actually is. Omnimic's ability to easily measure over wide frequency ranges helps with this, but when testing at low frequencies in particular, room effects must be avoided. This can be done outdoors (though noise levels become difficult to suppress in most areas) and/or by close-mic'ing within an inch or so of a loudspeaker driver. Then, of course, the SPL levels at the microphone will be much higher than out in the room so it is important that the microphone not be overdriven or you'll only see **its** distortion. OmniMic40k hardware is capable of operating well above 140dBSPL to avoid such problems.

Non-linear distortion behavior typically **gets worse as signal level (SPL or volume) is increased**, and so the related level applied will be a part of most meaningful non-linear distortion results. For example, stating "0.2% total harmonic distortion" really doesn't mean very much, but "0.2% total harmonic distortion at 135dB SPL at 40Hz and 1 meter" could be relevant. So, when measuring harmonic or spectral contamination distortion, you typically want to find out how the subject behaves as you increase the volume rather than make just one curve. The OmniMic40k hardware is particularly well suited for non-linear testing because of its inherent high linearity and SPL handling ability.

In general, keep in mind that non-linear distortion tests will require that background noise levels be low enough that distortion products of interest are detectable. That is, the noise level will deternine how well you can verify something acoustic isn't distorting.

Harmonic Distortion

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see also: About Distortion

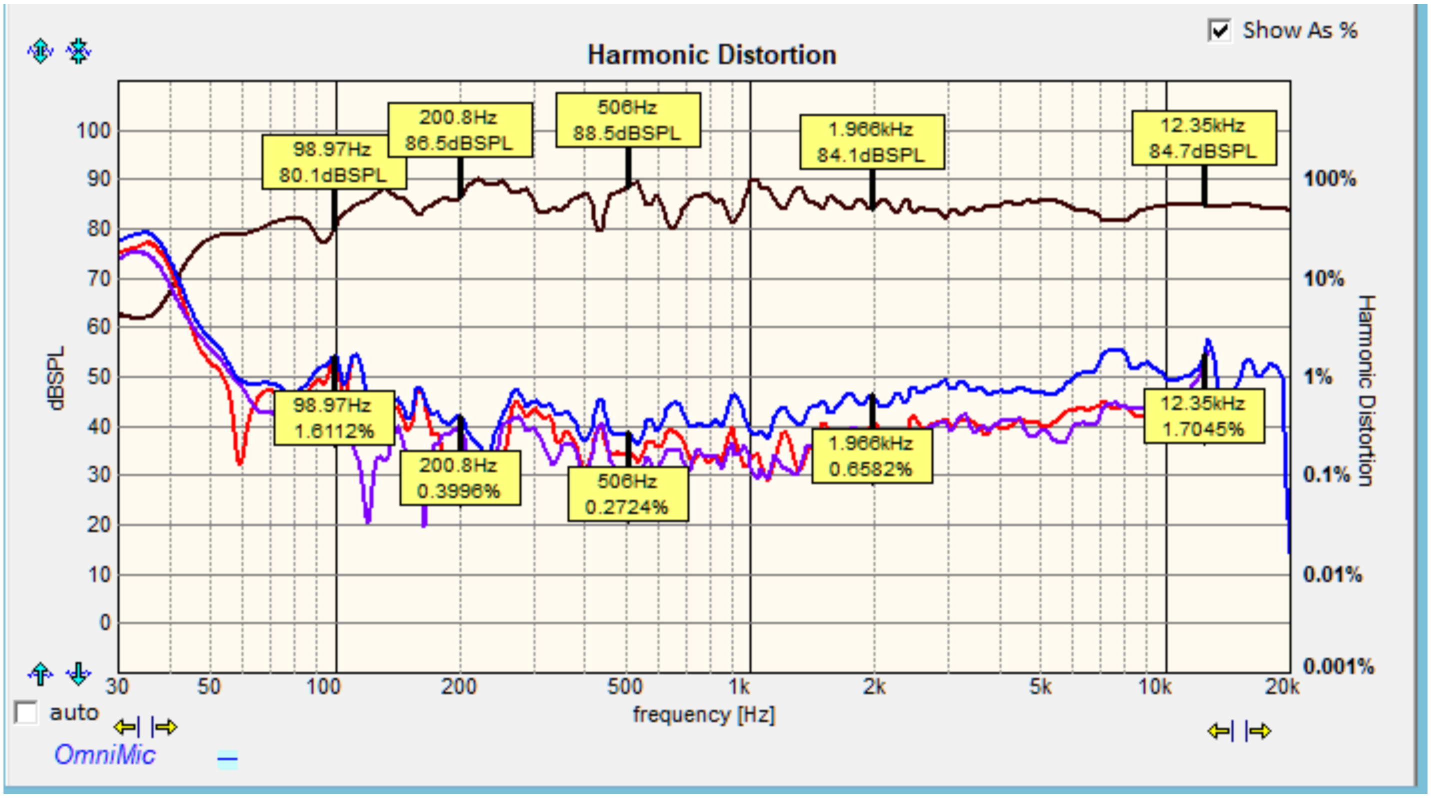
To properly measure Harmonic Distortion with OmniMic, measure only while the sound system is playing **the "Long Sine Sweep"** provided by a downloaded or generated WAV test file, the OmniMic Test Track CD or DVD, or as output from the soundcard of your computer (see the selection for that in the Config menu at the top of the OmniMic form). Use of other sound signals than this type will ***not*** provide meaningful results.

- This measurement will work best when the OmniMic is positioned relatively closely to the loudspeakers, so that sounds coming directly from the speaker are much stronger than those coming reflected from elsewhere. Room reflections are detrimental to measurement of harmonic distortion.
- The overall SPL level and the distortion products or rms Total, chosen via the checkboxes, will be shown in traces on the graph.
- The microphone and speaker should be held still over the length of several of the test sweeps (approximately 5.5 seconds each) before each graph update
- Harmonic distortion typically increases with increasing overall SPL level, and a stated number is only meaningful if expressed at an SPL level and distance from the driver.
- At very high levels (>120dBSPL for the original OmniMic, >145dBSPL for OmniMic40k), appreciable distortion may be generated from overdriving of the OmniMic itself. On OmniMic40k microphones, an attenuator switch is provided for highest signal handling so that the micropone can be placed very near woofer cones when measuring distortion at high levels. **Make certain that the "Attenuator used" checkbox (near right-top of the screen) is checked** when the attenuator is used so that sensed SPL levels will be shown correctly. The slide switch will be toward the microphone tip when the attenuator is **NOT** in use. **WHEN TESTING AT HIGH SIGNAL LEVELS, WEAR ADEQUATE HEARING PROTECTION -- DAMAGED HEARING DOES NOT HEAL.**
- You can place a persistent floating marker by typing one of the numerical buttons 1 through 9 to place on-trace markers on the SPL and the THD (if shown) traces, based on the position of the mouse cursor when the key is pressed. Click the mouse on the marker readout of the SPL level to remove an on-trace marker.
- **Distortion can be displayed as** individual distortion product SPL levels or as **percent (%) of the total output** at each frequency, using the "**show as %**" checkbox that is near upper right of the distortion plot. The enabled distortion component traces (2nd through 5th, and THD) will be shown as percentage, per the right axis of the graph, while the fundamental will still be shown as dBSPL per the left axis of the graph.
- You can freeze the graph to read multiple positions by using the Pause button near the top left of the OmniMic screen.

The graph will always display the frequency response curve in dark black, indicating the effective SPL level sensed at the position of the OmniMic. In addition, the graph can be configured to display

- 2nd Harmonic Distortion level or percentage
- 3rd Harmonic Distortion level or percentage
- 4th Harmonic Distortion level or percentage
- 5th Harmonic Distortion level or percentage
- 2nd through 5th Harmonic Distortion levels or percentage (Total Harmonic Distortion, THD, for the first 5 products; for undamaged loudspeaker distortion these harmonics normally dominate).

See also Compression Testing, another form of distortion test.

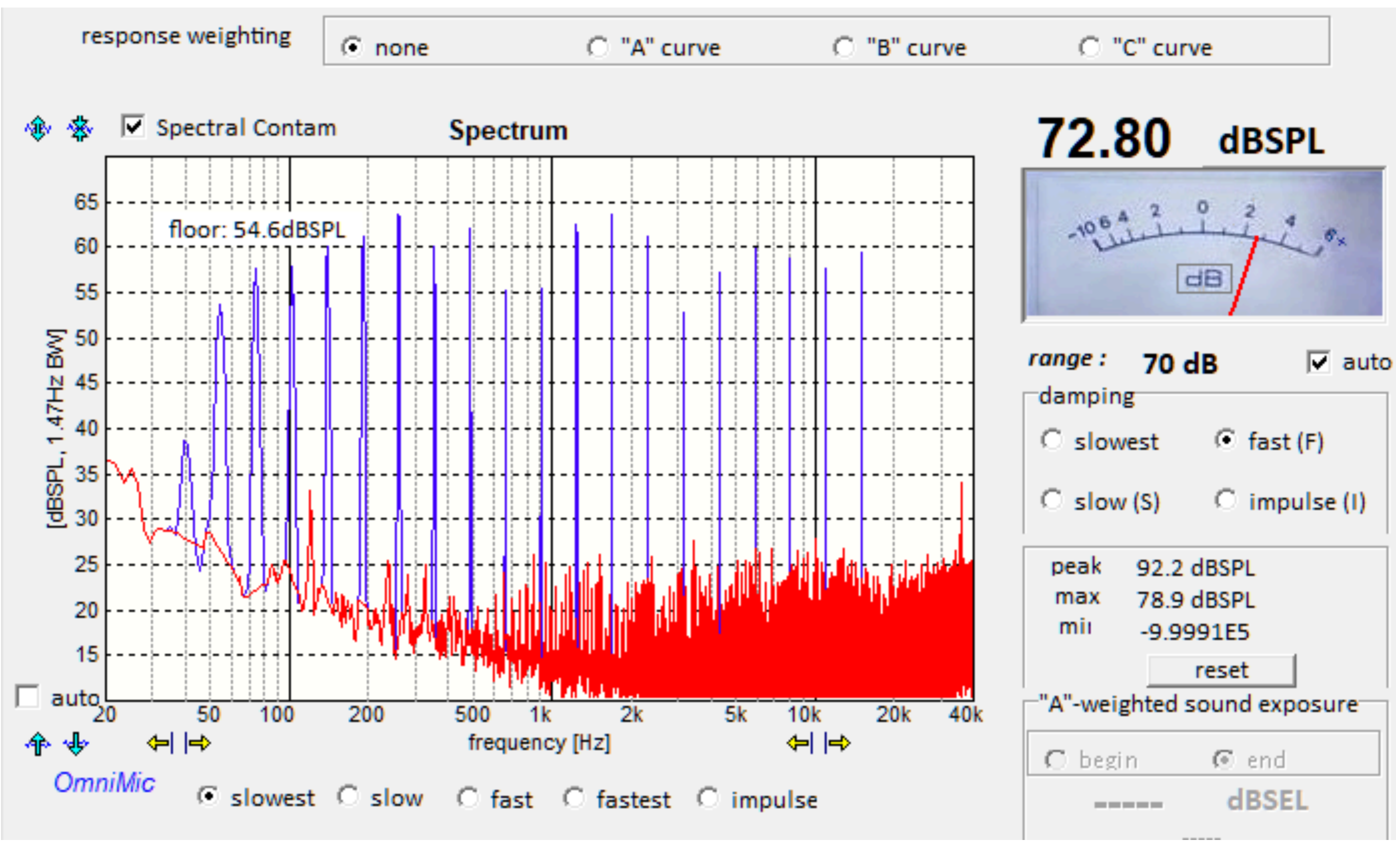


Show As %
Harmonic Distortion
98.97Hz
80.1dBSPL
200.8Hz
86.5dBSPL
506Hz
88.5dBSPL
1.966kHz
84.1dBSPL
12.35kHz
84.7dBSPL
98.97Hz
1.6112%
200.8Hz
0.3996%
506Hz
0.2724%
1.966kHz
0.6582%
12.35kHz
1.7045%
dBSPL
Harmonic Distortion
frequency [Hz]
auto
OmniMic

Spectral Contamination

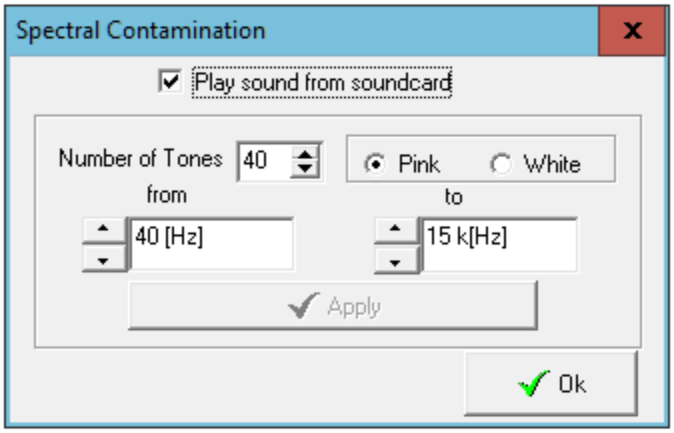
Spectral contamination refers to a measurement of intermodulation distortion induced by numerous simultaneous tones ("multitone"), typically appearing as an increase in the noise floor well below the levels of the tones, shown in Spectrum Analyzer curves. The tones used in OmniMIc software are log-frequency (constant multiple) spaced to minimize the likelihood of harmonic and intermodulation distortion products falling at the frequencies of applied tones.

The software shows the spectrum of the output in two overlapping traces: in **BLUE**, containing the overall output from the measured audio system. And in **RED**, which has energy in bandwidths near the applied tones removed and the calculated noise floor displayed numerically. The numerical values with and without the tones being applied can be compared for an indication of how much the noise floor is being raised or what it is being raised to.   The SPL meter will show the overall SPL level, including tone energy, being sensed. The noise level increase will be more marked as volume levels are increased, but be careful of increasing volume too much to avoid damage to equipment or your hearing!



There will be some variation in noise floor calculation because content in bands around the tones must be excluded and estimated from the nearby non-excluded regions. Changes of only a dB or so are probably not meaningful. The A, B, or C weighting curves can be applied to the microphone pickup to effectively shape or limit the frequency range being analyzed.

The tones for direct output or for WAV file generation can be configured with a configuration form which appears when Spectral Contamination is selected. The tones can be adjusted with a slope matching a white (flat) or pink (-3dB/octave) spectrum. Too closely placed tones at low frequencies should be avoided, as the exclusion bands are about 12.5Hz wide and will start to cover most or all of the frequency space between the tones.



The test can be used to indicate when a system is having trouble reproducing the multitone, however it is limited in many cases to the level of the existing noise floor in the room. For example, if the noise floor were at -55dBSPL, then induced distortion levels near or below that cannot be clearly detected. Acoustic noise sources in the area should be reduced if possible, and the setup should be arranged for maximum dynamic range.

A statement of a measurement result could be

"With [*specified multitone configuration*] and a level of [*SPL meter*] dBSPL at [*distance*] meters, a noise floor of [..] dBSPL was induced."

This result would only be significant, though, if the background noise was low enough that the test showed a significant increase in the noise floor when the system was driven.

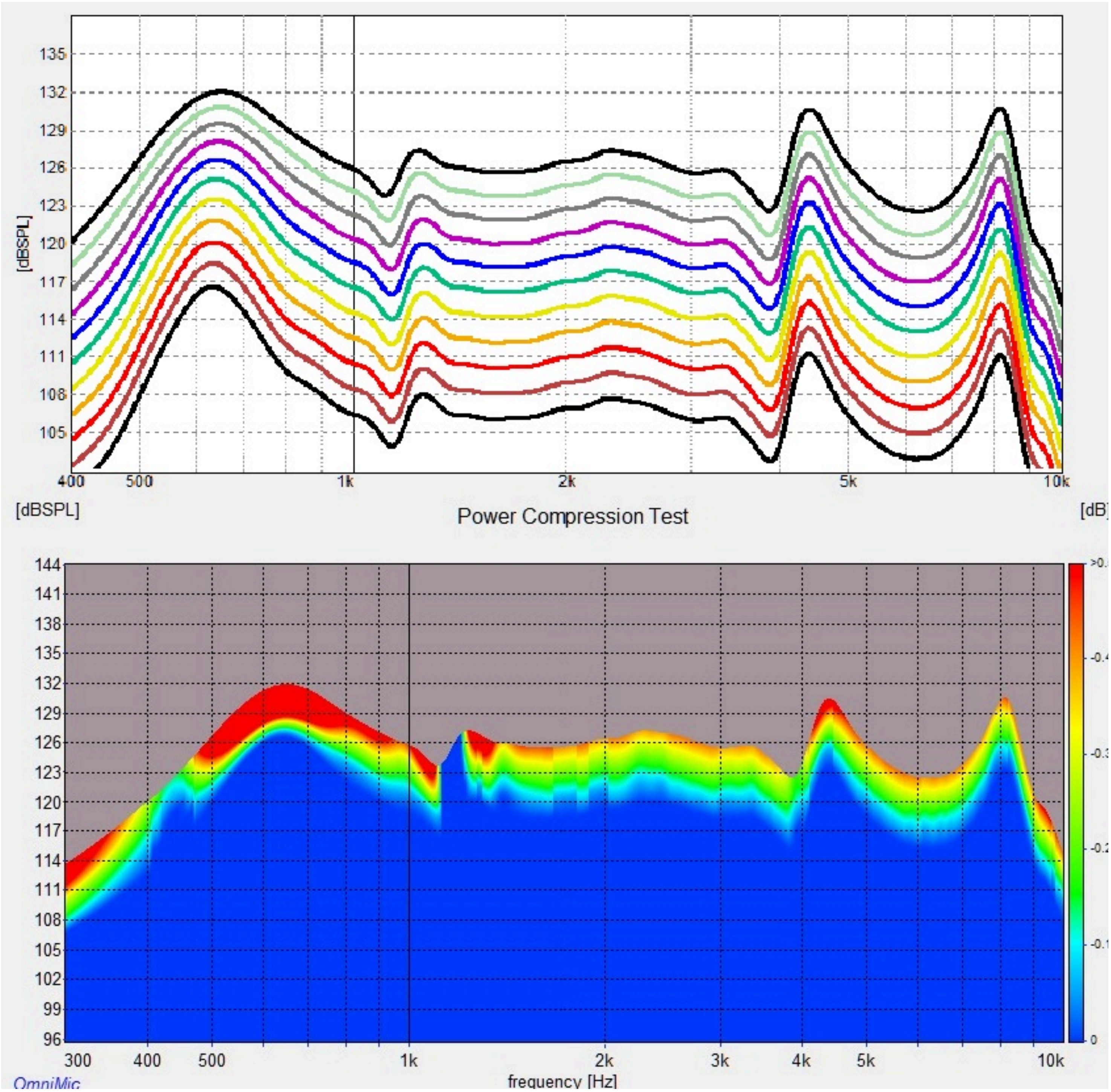
Compression Testing

When a well-behaved transducer or amplifier's upper signal level handling is approached it begins to compress waveforms. (When not well-behaved, the results of overdriving -- like crackling sounds, damage, even fire! -- aren't difficult to detect, and so are not addressed here).

This compression effect is frequency dependent and could be subtle (fractions of a decibel) or drastic (such as when a power amplifier clips). These overdrive effects could be investigated through Harmonic Distortion testing, particularly by noticing sudden increases in 2nd or 3rd harmonic output as level is increased.

OmniMic compression testing senses change from expected output level at ***fundamental*** applied frequencies. This can happen in loudspeakers, power amplifiers (particularly low-feedback types) or even in microphones.
[So it is important that the OmniMic being used is not being subjected to SPL levels higher than it can handle quite linearly: about 120dBSPL for OmniMic V2, or 145dBSPL for OmniMic40k. If necessary with a V2 microphone, move the microphone further from the loudspeaker].

In OmniMic compression tests, the **test stimulus must come from the computer** or from an external DAC or soundcard driven by the computer. After the highest level of a series of sweeps (as you can configure, in terms of nominal power or drive voltage) is reached, the family of response curves is analyzed to determine how much and at what frequency the result was found to fall short of the expected output. You can view the result in terms of equivalent applied power at 4 or 8 ohms, applied voltage, or resulting sound pressure level. This is from a test of a small earphone coupled to an OmniMic40k, shown in terms of output SPL:



The color bar at the right of the plot shows how many decibels of compression happened at frequencies and levels shown in that color: dark blue for no compression, dark red for 0.5dB or greater compression in the example. Notice in the example that there appears to be no compression at all seen near 1300Hz at the drive levels applied - it is possible that "negative compression" can also occur when nonlinearities cause resonances to move in frequency (such 'negative compressions' aren't quantified in that graph). At other frequencies, you can see the compression amount increase from zero through 0.1dB, 0.2dB, etcetera. Around 450Hz, you can see that compression has only barely begun at the applied drive levels. Usually, you want the display to assume the output level is monotonically increasing, even though measurement noise or other issues may cause that not appear to be the case. In the Options menu of a compression display you can experiment with showing both upward and downward apparent compression.

Loudspeaker compression behavior can also be even more complex, such as when it results from heating over some time duration and so varies by how long a test is run. In the OmniMic compression tests, the drive power is applied over less than half a second length, repeated every 683 milliseconds and is applied increasing from low levels, so that higher power levels where compression begins to occur should not yet begin to generate much heat in loudspeaker drivers. Still, if you are repeating a compression test on the same loudspeaker several times, give it some minutes of rest between runs.

These tests can be ***LOUD***, **stressful**, or **damaging to equipment and the hearing** of humans nearby. When the compression test is about to begin, you'll see a (maybe annoying) **warning**, and you <u>do</u> need to **take** that warning ***seriously***! So it is also given here:

********CAUTION*******

These tests can involve VERY HIGH SPL levels. Exposure to such levels is hazardous to equipment as well as to your ears!

Make sure that anyone within the area is equipped with hearing protection such as earplugs or protectors!

Be prepared to stop the process if you notice the test appoaching dangerous levels.

**********************WEAR HEARING PROTECTION**********************

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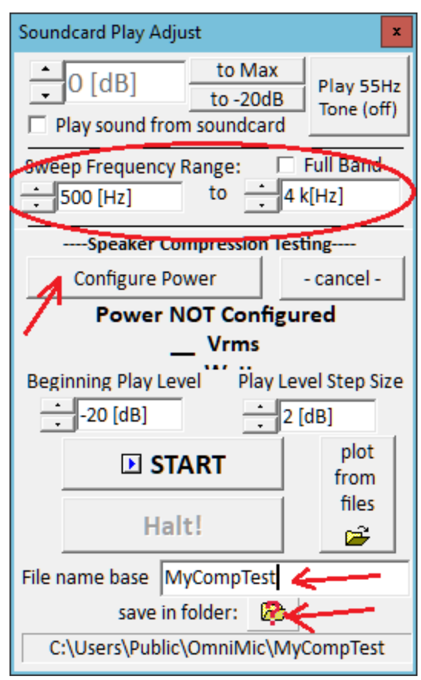
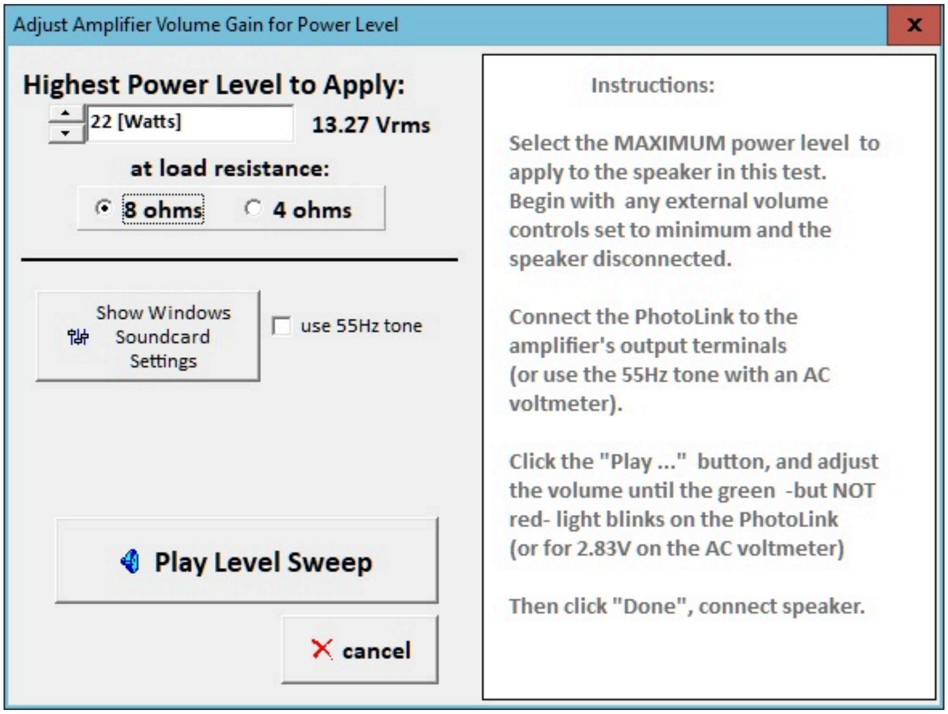
To arrange a compression test, use the "Config --> Compression Test" menu when you are on the Frequency Response tabsheet. In this form you must select the directory and a file heading under which the compression test measurement data is to be saved. The effective **test frequency range** can and (usually should) **be limited** to help avoid damage to loudspeaker components -- and to the ears of anyone in the test area! For example, so that excessive low frequency power doesn't get applied to a connected tweeter or a person within earshot.

Here, you also select the sweep step size and how far in decibels (below maximum) you want the sweeps to start. Then, you must configure the power level for the test.





On the form that appears when you click the "Configure Power" button, select the test power level (in equivalent 4 or 8 ohm levels)and then click "Play Level Sweep". The program will apply a swept signal at a level some dB below maximum allowing for you to arrange for the needed voltage drive.
You are to turn up the gain (volume) of your amplifier (or preamplifier, or perhaps Windows Soundcard setting) to achieve a 2.83Vrms sweep level. The easiest way to do that is with the PhotoLink device, though it can also be done with a 55Hz tone and an AC voltmeter or DVM. (If you are interested only in SPL and not how much voltage or power will compress your speaker, you can just adjust by ear to how loud you'd expect that voltage to play). Then click "Done" and start the sweep test in the Play Adjust form. Sweeps will be applied starting from a low level and increasing every few.

Should things get out of hand and you start hearing bad noises, you can click the **Halt** button!!

When the sweeps have completed (or you have Halted the process), a compression graph will be displayed.

You can re-display results of previous tests using the "Plot from files" button.

Reverb/ETC

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Reverberation is a measure of how quickly sound reflections die down in a room, and will depend on the frequency range of the sound, the size and shape of the room, what is in the room, and how the room surfaces are constructed. Determination of **RT60** can be made using this tool.

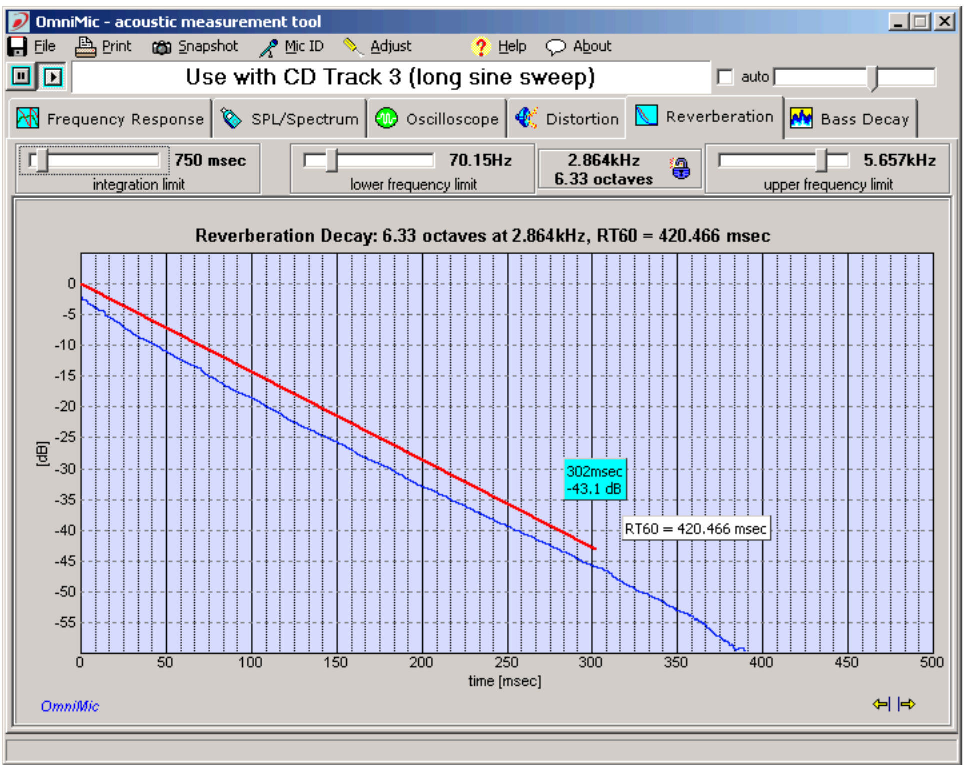
ETC stands for "Energy-Time Curve", which is a similar display, but which shows individual reflection spikes more clearly to help in locating where they are coming from. This display is also able to calculate **"Speech Transmission Index" (STI)** for auditorium and hall acoustics evaluations.

A third display option "**Log IR**" shows the dB level of the Impulse responses (like the one at the bottom of the Frequency Response tab page, but in a decibel format), a very similar result to the ETC in most cases.

These tests must be done while the sound system is playing **the "long sine sweep" test signals** (as downloaded from the Dayton Audio site, or off of the OmniMic Test Track CD or DVD, a generated WAV file, or played out from the soundcard of your computer). Use of other sound signals will ***not*** provide meaningful results!

Reverb (RT60)

OmniMic provides a graph of the reverberation decay curve. A properly done reverberation decay curve will drop 40 decibels (dB) or more, relatively smoothly, from left to right. Using the mouse cursor on the graph and the mouse button, you can obtain a measure of the **RT60** value -- the time needed for a reverberant field falling at that rate to die down by 60 decibels.



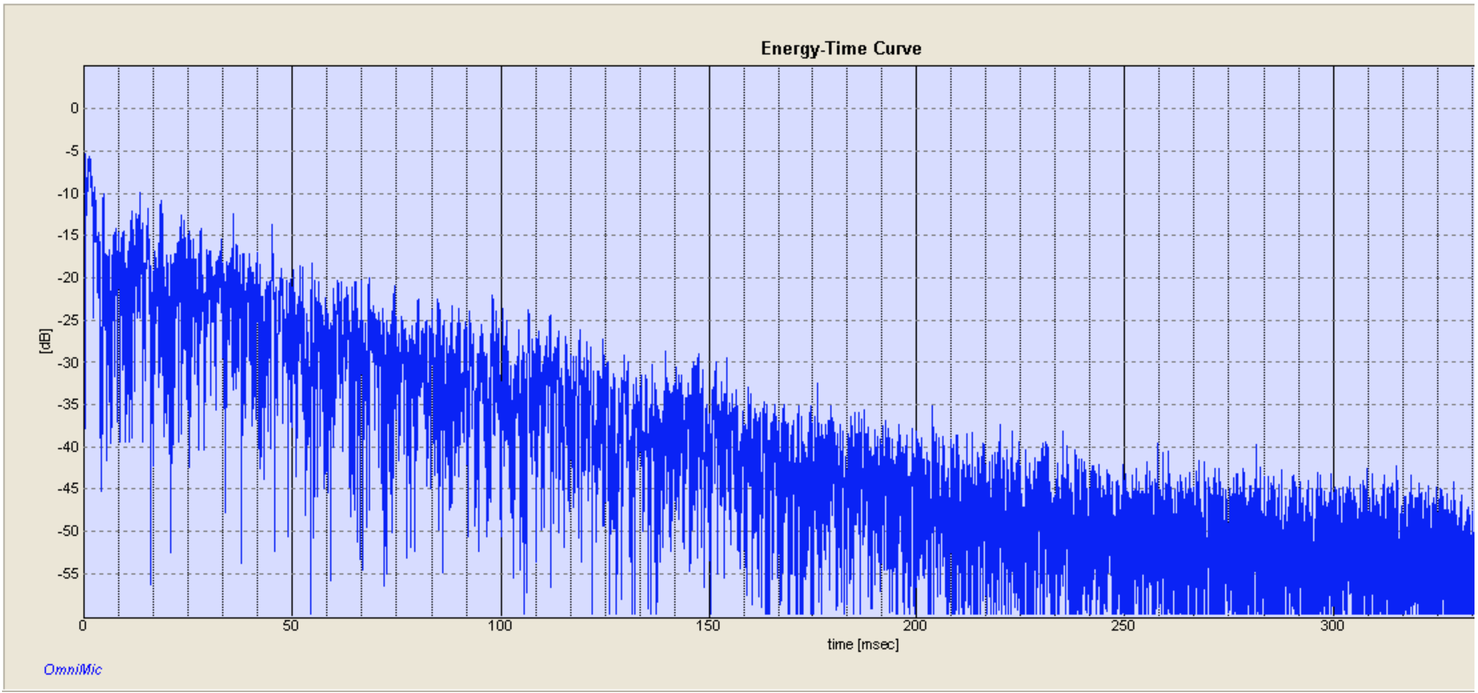
- Set the value of the "Integration time" to a value approximately equal to the expected RT60 of the room (around 500 milliseconds typically). The optimum value is the one that makes the decay line the straightest and longest.
- Set the lower and upper frequency limits to define each frequency range over which you wish to measure. The width of the range will be displayed in Hz and in Octaves. If you wish to force the controls to hold current number or octaves of range, click the small "lock" icon and the controls will track accordingly. Click again to unlock.
- Click on the graph to define a line **parallel** to the decay curve. The slope of this curve defines the value of RT60, which will be displayed on the screen until the next graph update. This may be easier to do if you first freeze the graph with the Pause button at top left of the OmniMic screen.

ETC (Energy-Time Curve)

The curve shown is the energy relative to the level of the energy in the first peak as an impulse comes from the speaker and reflects around the room. In other words, imagine the speaker sent out a sudden pulse rather than the sweep this test uses. (OmniMic calculates results equivalent to a pulse using a sweep as the sweep is better at rejecting noise and distortion). The first spike in the ETC is the original at 0dB and 0msec time, usually a direct signal from the speaker. The following ones are as reflected from the various surfaces around the room. You can often hold a piece of acoustic absorber (acoustic tile or even a pillow) near surfaces near the speaker or microphone to identify the sources of reflections. The controls at the top of the form allow you to filter the ETC to include only certain frequency ranges.

OmniMic can save and recall (as Added ETC curves) ETC measurement files, allowing multiple curves to be shown simultaneously. Curve files can also be smoothed before saving, to help in interpretation.

When in ETC mode and operating with full bandwidth, you can also have OmniMic calculate the **"Speech Transmission Index" (STI)** or the "Rapid Speech Transmission Index" (RASTI, a similar but less intensive version). To use this play the long sweep test signal from the system being tested (typically a public address or so the update rate will generally be slowed when calculating STI or RASTI. A result of "1" is perfect, "0" would be worst that can be expressed. Good signal to noise is desirable when making the measurement, so play the test signal loudly enough.



Bass Decay

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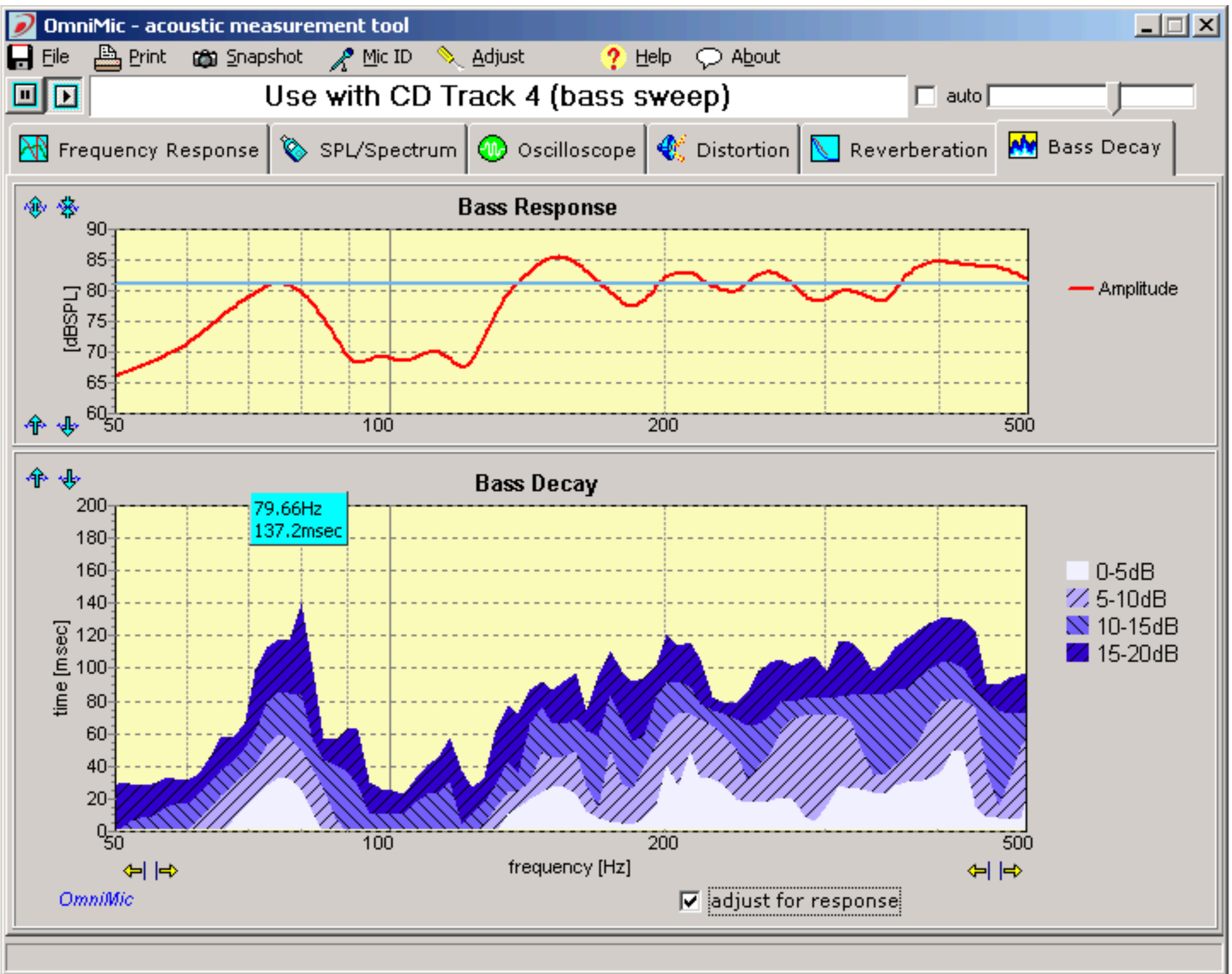
Use the Bass Decay analyzer for a non-standard measure of how bass energy decays in a room. Use only **OmniMic "bass sweep" tracks** to perform this test.

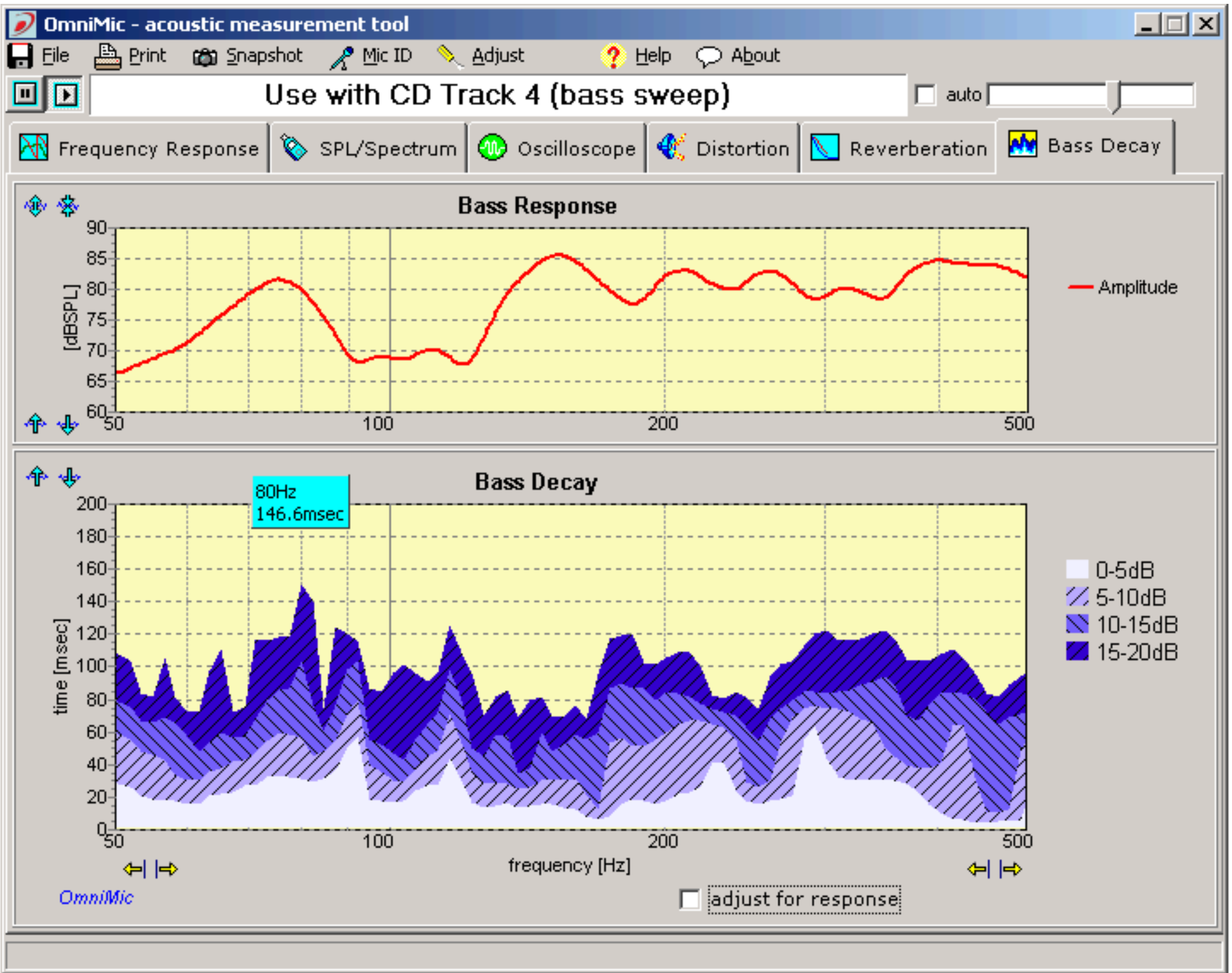
When a bass note is stopped within music, the sound in the room at some frequencies may still continue for some time. That is because the sound reflects back and forth between walls, resonating and forming "modes" before eventually dying down. This is not altogether bad and some reinforcement is normally desirable for natural sounding playback, but you would like to keep it under control and not have some notes sounding muddy or lingering much longer than others.

The top graph on the Bass Decay display shows the frequency response of the bass range, similar to that shown with the Frequency Response analyzer. The bottom graph shows how long it takes the sound to decay at each frequency. As shown on the legend to the right of the decay graph, the white area extends upward to indicate when the level drops no more than 5 decibels (dB). The light blue indicates when the level has dropped between 5dB and 10dB, etc.



At the bottom is a check box labeled **"adjust for response"**. This affects whether the variations in relative bass strength are included when calculating the decay graph.

- when the "adjust for response" box is unchecked, the bass decay shown does not take the frequency response into account. All decibel levels are with respect to whatever the response level is at each frequency. At frequencies where there is lower overall output (as shown on the upper graph), the decay may appear to be longer than actual because of background noise in the room that is inseparable from the low bass levels.
- when the box is checked, the decibel levels are with respect to the blue line shown on the frequency response graph. For instance, if the line is at 70dB SPL, then the white area of the Bass Decay graph shows how long it takes before the level dropped below 65dBSPL (that is, 70dBSPL minus 5dB), the light blue shows how long before the level dropped to 60dBSPL, etc.



Operating Notes:

- When working with the "adjust for response" box checked, you would normally set the blue line (**click inside the Bass Response graph**) to be in the upper, most flat, region of the Bass Response curve.
- The microphone should be placed out in the room, measuring at various listening positions. You can try relocating subwoofers or main speakers, or listening chairs to find optimum locations for these. For floor vibrations, spikes or pads below subwoofer boxes can also affect the bass decay (not always for the better).
- Remember that the response is strongly position dependent. Optimize the woofer placements and equalization for best overall results at all listening positions. This is generally easier to accomplished if multiple subwoofers are used.
- Slow decays should be less bothersome to the ears if response level is reduced at the problematic frequencies. If you are using an equalizer in the room, you can reduce your system's response to accomplish this. You can use the "adjust for response" checkbox to help find a trade off between tighter bass and flat response. But in general, decay speed normally follows the overall frequency response curve.

Reading Values on Graphs

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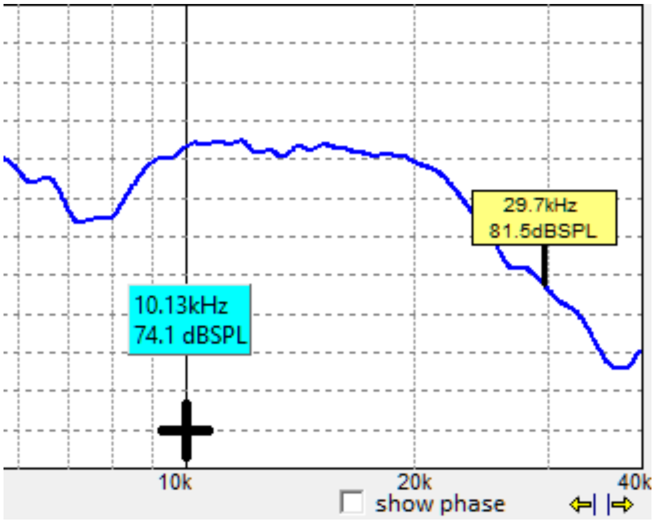

You can position the mouse cursor on most graphs to read data values corresponding to the vertical and horizontal location of the pointer. While the cursor is placed there, auto scaling will be temporarily disabled. Move the mouse cursor outside of all graph areas to allow graphs to autoscale when.

Also see "Tracking On-Trace Markers".

Scaling Graphs

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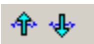
The various graphs and the SPL meter display can be scaled as suits the user. Or they can be set to automatically scale (vertically or meter range). Each graph has several arrow buttons that, when clicked, will adjust some of its coordinates.

 These compress or expand the curves vertically

 These move the curves up or down vertically. You should turn off automatic scaling if you want to adjust these manually

 These adjust the horizontal value of the left or right borders of the graph

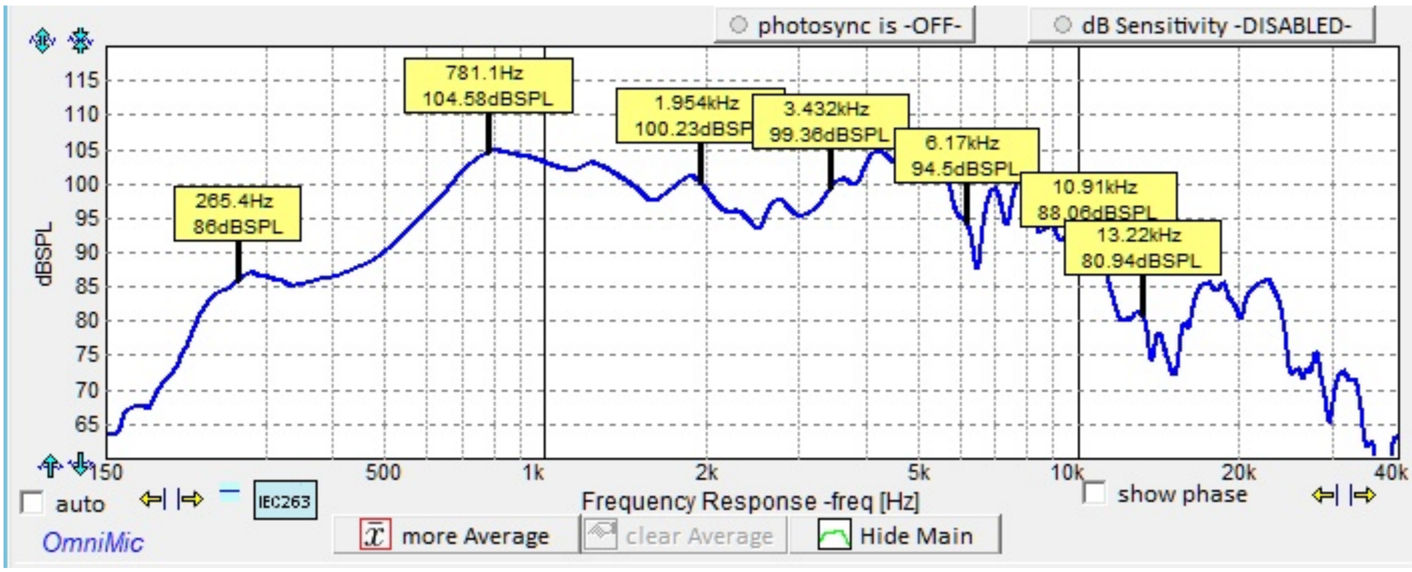
Automatic scaling will be temporarily disabled whenever the mouse cursor is over a graph, to keep it stable while reading data point values.

In addition, the entire OmniMic panel can be enlarged or reduced.

OmniMic backgrounds can be configured so that the plot backrounds are white or in some default colors. This setting will be remembered from session to session. The white background cannot be applied to Oscilloscope plots, however, these will remain green. The selection for this is under the "Config" menu.

Tracking On-Trace Markers

Up to 9 persistent **"floating" markers** can be placed on a live trace of Frequency Response, FFT or RTA Spectrum, or Harmonic Distortion SPL and THD traces.
Position the mouse cursor over the graph at the frequency you want the marker placed at, and tap one of the keys for "1" through "9".
Tapping "0" will allow you to clear all floating markers.
Right-clicking on a floating marker will allow you to delete it individually.



Printing Graphs

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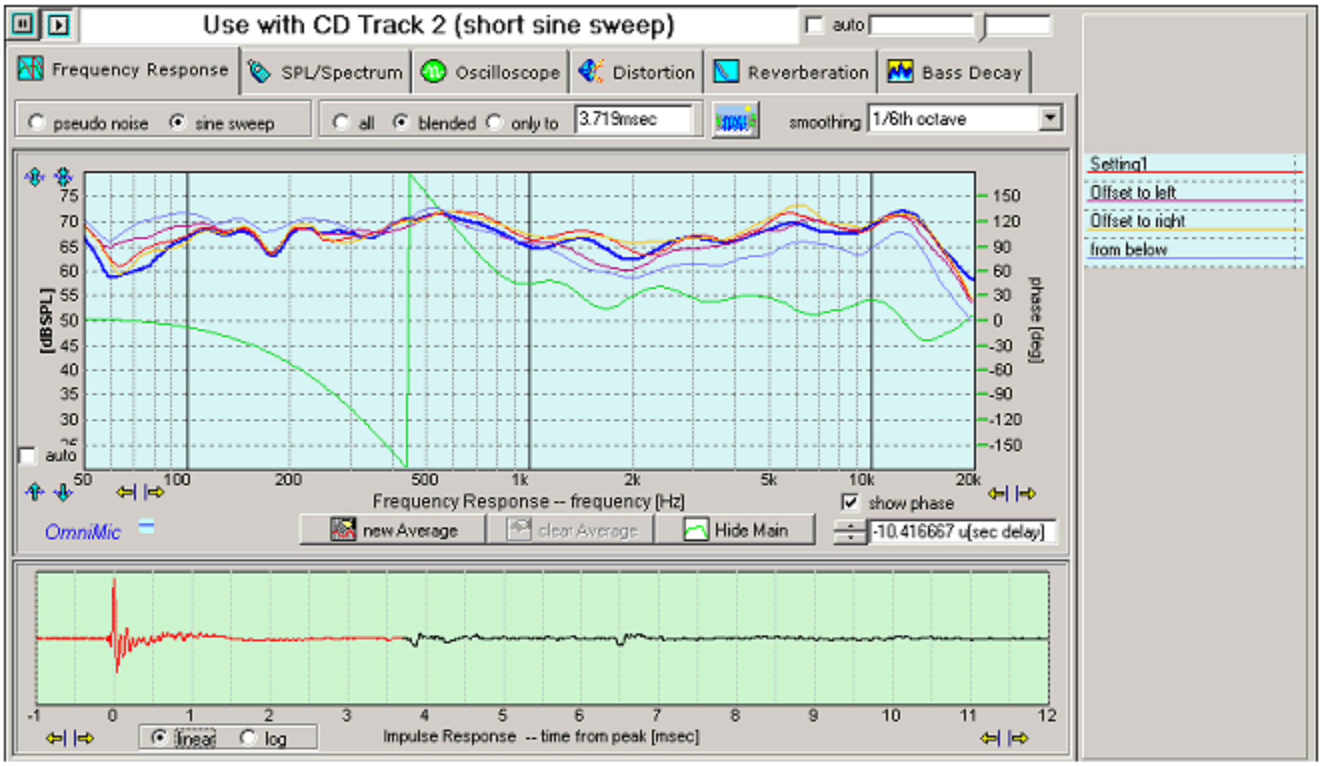
Print the current OmniMic screen image by clicking on the Print menu. You can select the printer to use with the "Printer Setup" menu on the main OmniMic screen. On most plots (but not Waterfalls or Polar displays) you can also print from a pop-up menu that appears when you right-click on the graph.

When a Frequency Response graph is shown with multiple "Added" curves (from files), the image that will be printed includes a list of the files used for each curve. Because of the limited space for showing the file names, short file names are recommended.



If you want to print only an individual graph (without the buttons or controls of the rest of the image), you can get a clean copy of it by using the "Snapshot" menu to save the image to disk in various formats, and using a graphics program (such as PaintBrush) to print the image.

Saving Graph Pictures to Disk ("Snapshot")

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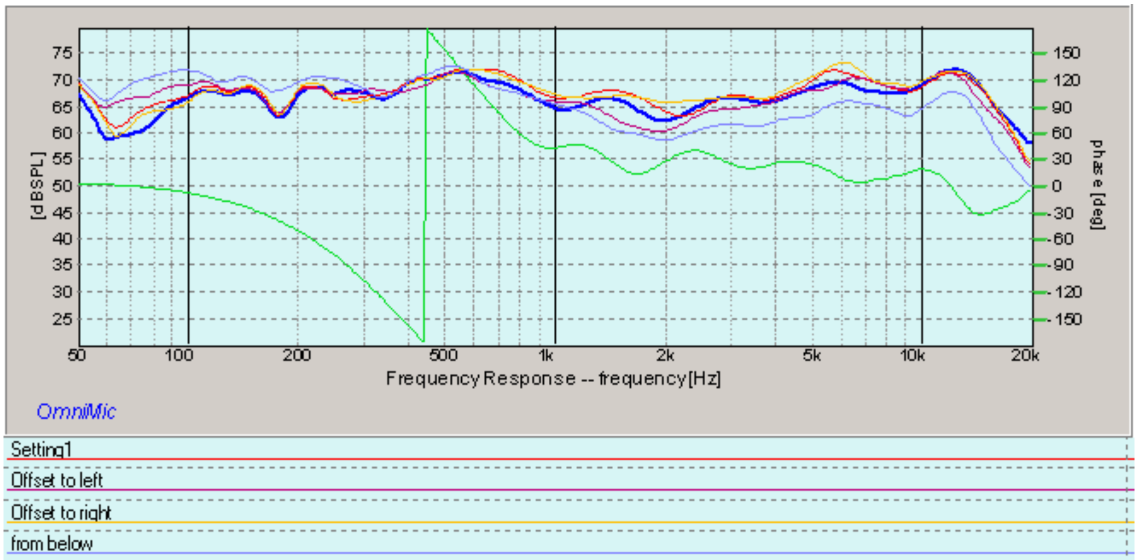
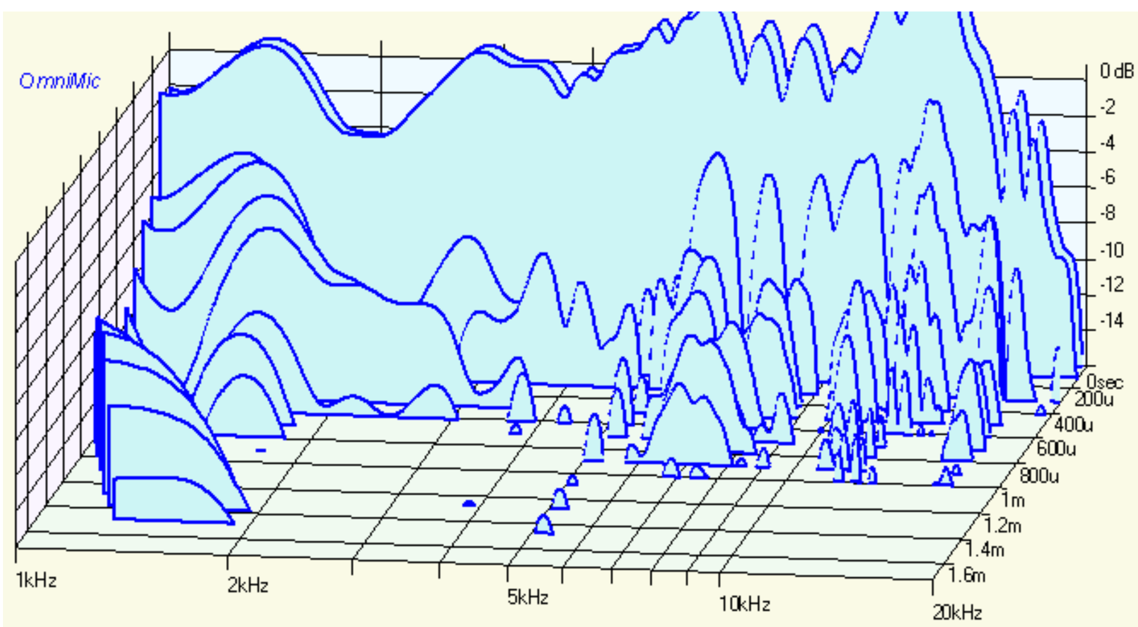
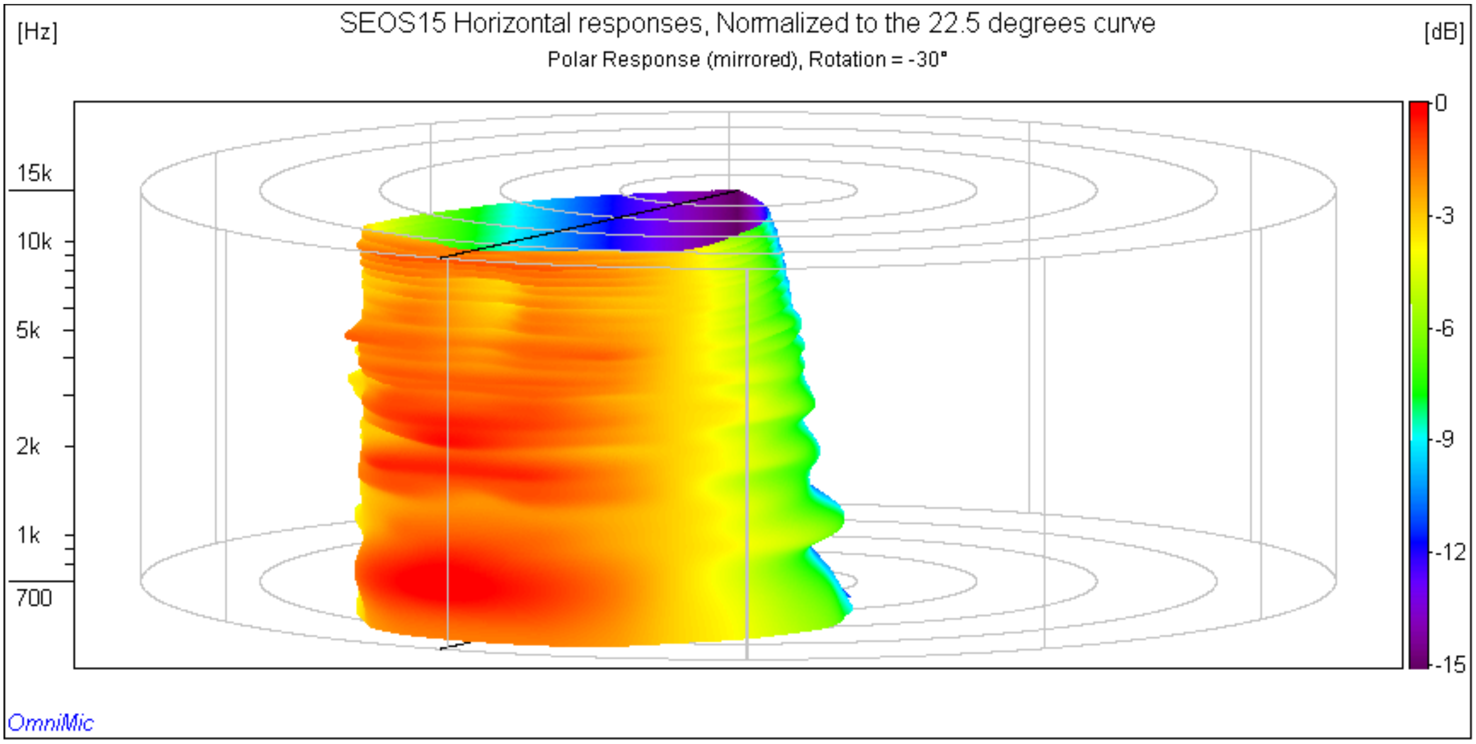
You can save a high quality picture file of the current graph by clicking on the Snapshot menu. These are useful for including into reports or published documentation, or for printing out separately using a graphics program. Buttons and controls shown on the graphs will not be saved to the file. However, if you have multiple curves (from data files) on a Frequency Response plot, a legend can be included in the graph identifying the curves with the files. You can also elect to add some notes to appear with the plot.



On most plots (not including Waterfalls or Polar displays) you can also command a snapshot from a pop-up menu that appears when you right-click on the graph.

You will be prompted for a file name. Files can be saved as bitmaps, PNG files, JPGs, PDFs, or as scalable WMF metafiles. Metafile or PDF displays can be scaled smaller or larger (when used with capable software like most word processors) without loss of resolution. Bitmaps are more universally supported, but may not expand or compress as well when inserted into documents. JPGs or PNGs are ideal for posting on internet web sites or discussion boards.





[Hz]
SEOS15 Horizontal responses, Normalized to the 22.5 degrees curve
Polar Response (mirrored), Rotation = -30°
[dB]
15k
10k
5k
2k
1k
700
0
-3
-6
-9
-12
-15
OmniMic

Freezing the Graphs

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By default, the OmniMic graphs and meters update, continually. It is always live.

At times you may wish to keep a graph steady for inspection or for reading off data points using the mouse cursor. This is easily accomplished by clicking the Pause button at the top left of the OmniMic screen.

To set it running again, click the Play button.

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Saving Data Files to Disk

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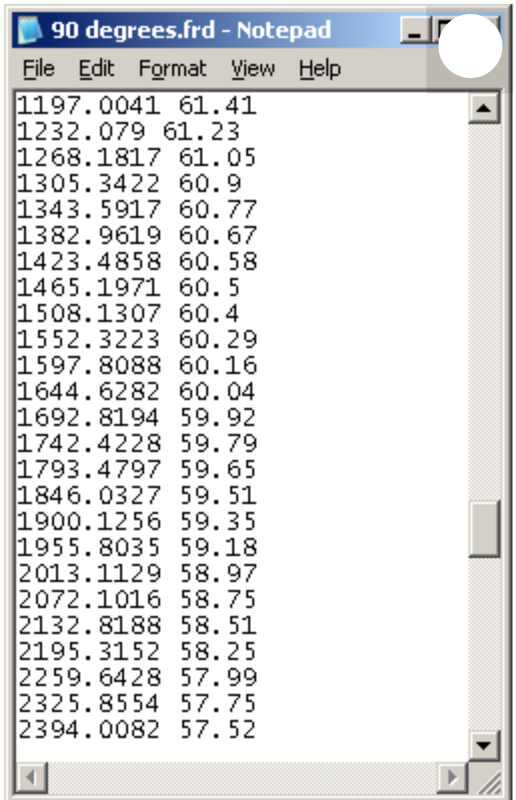
Data files can be saved to disk by clicking on the File menu. It will be saved in an ASCII text format importable into spreadsheet programs.




```
1197.0041 61.41
1232.079 61.23
1268.1817 61.05
1305.3422 60.9
1343.5917 60.77
1382.9619 60.67
1423.4858 60.58
1465.1971 60.5
1508.1307 60.4
1552.3223 60.29
1597.8088 60.16
1644.6282 60.04
1692.8194 59.92
1742.4228 59.79
1793.4797 59.65
1846.0327 59.51
1900.1256 59.35
1955.8035 59.18
2013.1129 58.97
2072.1016 58.75
2132.8188 58.51
2195.3152 58.25
2259.6428 57.99
2325.8554 57.75
2394.0082 57.52
```

Frequency response data (".frd") files saved this way can be reloaded back into the OmniMic program display using the AddedCurves menu. AddedCurves may be loaded individually or a list of curve files can be loaded or saved for fast recall of a group of curves.

You can also select a data file save operation for most plots from a pop-up menu that appears when you right-click on the graph. For Frequency Response plots, there are also options for easy auto-incrementing file name generation for making a series of related measurements, particularly helpful when you are collecting curves for a Polar Display.

To save in impulse response as a WAV file, right-click in the Impulse Response (lower) plot in the Frequency Response page. The time length of the saved file will be about 620msec, at the measurement sample rate (48ksps, or 96ksps for OmniMic40k).

The data curve reloading feature works only for Frequency Response and Energy-Time Curve (ETC) files. Other type files should be saved to disk as graphic "snapshots" if the plots are required later.

Test Signals and Files

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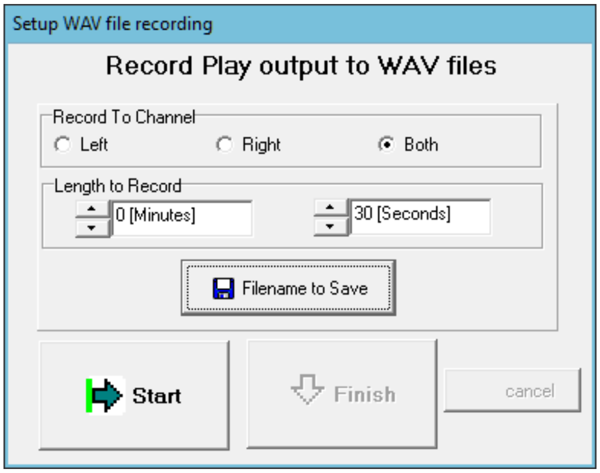
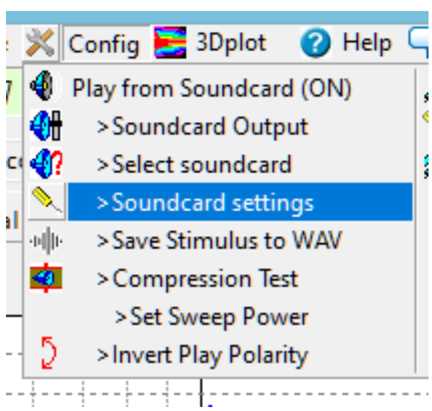
OmniMic test signal files can be downloaded from the Dayton Audio site for playback from external players.

You can also save OmniMic internally generated test signals to files yourself from the OmniMic software using the "Save Stimulus as WAV" submenu under the "Files" menu. Note that the files generated will NOT be affected by the Run/Pause button.



You can also drive the amplifier/speaker system from the computer soundcard using the Config menus. If you have more than one soundcard or DAC device, you can also select it from the same menus. When running from the soundcard, audio output will stop when OmniMic is paused or when you are saving a data file, and will mute if you hit the Esc key (F12 will turn it back on). When playing a signal for the 96kbps OmniMic40k, make sure that the soundcard is configured in the Windows settings to play at least at a 96kbps sample rate! This setting can be found through the Windows' Sound Controls use the "sound card settings" menu item under OmniMic's "Config" menu and follow the directions.



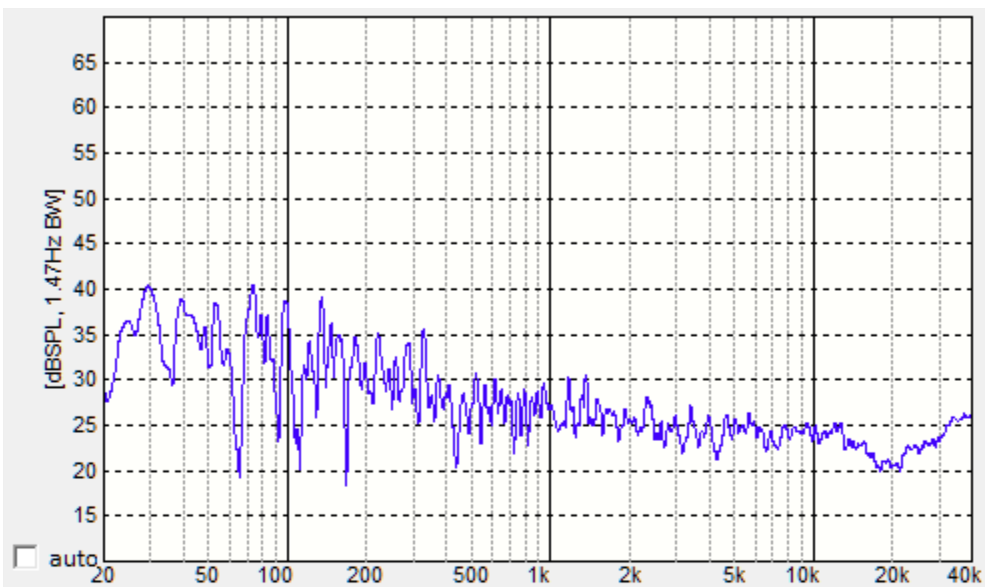
In most measurements, tracks in WAV, FLAC or MP3 format can be used for playback from computer systems or MP3 players. If using MP3, use one of the higher quality type MP3 compression formats (128k or better) for best results. Some ripping programs may do better than others in providing usable MP3 test tracks. For Frequency Response measurement using MP3, avoid using the Pseudo Noise stimulus signal.

Please note that an MP3 file or a normal CD player playing a Redbook CD (such as the original OmniMic test signal CD or DVD) does not have sufficient bandwidth for use with OmniMic40k microphones beyond 20kHz. Use a FLAC or WAV file as described above to play the proper 96k type signal through a capable audio player device. You can use the Omnimic CD or DVD with OmniMic40k, but usable results will result only up to 20kHz. Note that most TOSLINK optical interfaces also support only up to 22kHz output.

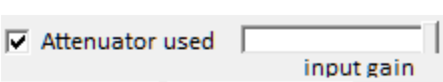
Dynamic Range Considerations

Measurement capability is limited by the dynamic range of both the equipment and the environment. Dynamic range is the difference in decibels between the noise floor and the highest levels that can be measured with acceptable distortion. OmniMic and OmniMic40k are designed as measurement microphones intended primarily for use in determining frequency responses and related characteristics.of audio systems. The small sizes of the capsules used, while good for minimizing directionality, do result in more background noise than would occur with, for instance, 1 inch capsules in typical music recording microphones*. Typically acoustic background noise in rooms where response measurements are done will be higher than the inherent noise of the microphones.

The OmniMic40k capsules are particularly small, and the equalization used to flatten them to 40kHz increases effective noise seen in the highest octave. This will be most evident in Spectrum Analyzer displays above 20kHz, where the electronic noise floor is seen to rise. Additional noise above 30kHz also is evident because of noise-shaping used in the converter chip inside the microphone.



To maximize the dynamic range use the highest gain setting of the slider at upper right that does not overdrive the converter circuitry. When that upper limit is approached, a warning will appear on OmniMic's screen telling you to turn it down.



With the OmniMic40k, the upper SPL limit can be increased by sliding the -24dB attenuator switch on the microphone body toward the back of the microphone body. When switched to this position, be sure to mark the "attenuator used" checkbox at the upper right of the form so the software knows to numerically account for the attenuation. In this position the electronic noise floor will be increased, so use the -24dB attenuator only when required to handle very high SPL levels. When the attenuator is used, the gain slider should be as far to the right as conditions will allow.
Of course, if you are high enough in level to need the attenuator switch, you should probably also be using **hearing protection** if you are in the room with it!!

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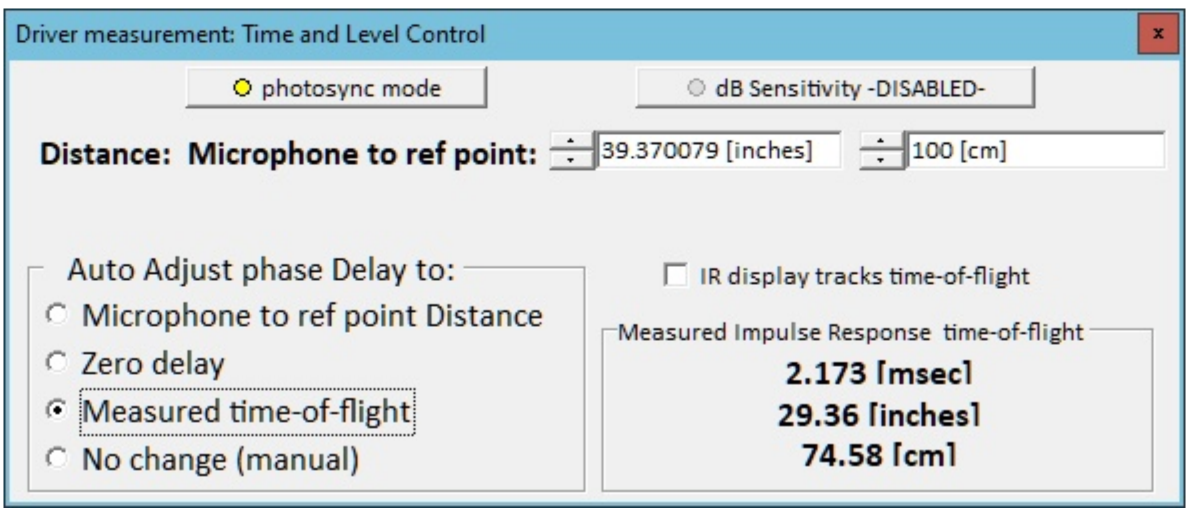
*While the higher noise floor in most cases would preclude using OmniMic40k as a recording microphone, it might be used for pickup in very high SPL and wide bandwidth situations needing flat response, such as with percussion instruments at close distances.

Making frequency response files for crossover network design

Designing loudspeakers is much easier than it was only about 10 years ago. Simulators such as Xsim or VituixCAD can help designers as the behavior of drivers and networks as well as driver interractions can be extremely complex. Effective use of simulators requires knowledge of

- Driver frequency response shape, both magnitude and phase
- Drivers' "SPL sensitivity", how well each converts drive voltage to sound level, typically included in the frequency response data. This depends on RMS voltage drive level as well as distance of the intended listeners from the loudspeaker.
- Relative distances of drivers' effective positions (distances) from the listener or microphone
- Driver impedance response (including again both magnitude and phase). The Dayton Audio DATS system is very effective for obtaining this data.
- Not essential, but: polar response data of each driver can also help handle how a loudspeaker system will behave at different listening angles, considered in some designs.

The first three and the last can be found with OmniMic. A standardized condition of measurement is with a drive level of 2.83Vrms and done at a distance of 1m. If these can't be the actual conditions of measurement, then the data can be compensated for the actual values.



Even with all these, final design must take into account baffle effects on radiation. There are also simulation tools to account for this. If it is an option, the chosen drivers can best have their data take while mounted on the intended baffle. It is also common to design at first without consideration of the baffle, and then readjust to shape the response accounting for baffle effects, as there will usually be adjustments after first listening tests anyway.

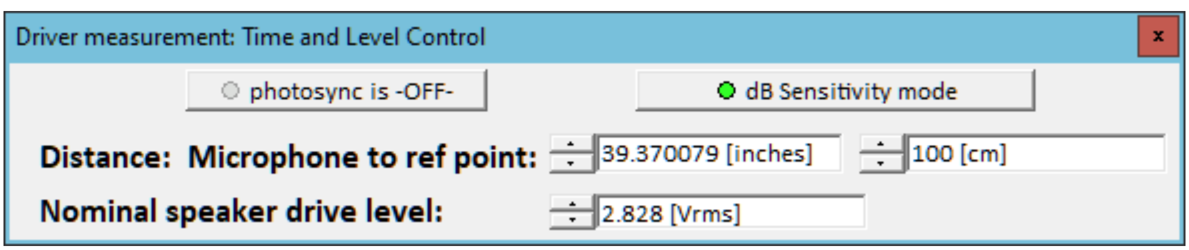
Lower frequency response (lower midrange and bass) does depend heavily on both baffle and cabinet parameters (volume, porting, open or closed surfaces), so these should always be investigated and anticipated first.

Getting the frequency response and sensitivity data

In multi-way loudspeakers, the way that various drivers interract is affected by their relative distances from the listener/microphone. The "starting points" of driver radiation, relative to the listener or to the baffle surface, varies with their construction. When making FRD frequency response data files with optical syncronization for crossover design, the included delay should be compensated based the measured distance of the OmniMic tip to the mounting baffle surface the driver is on. This can be done automatically with the PhotoLink (after you manually do the distance measurement by tape measure, mete stick, or laser) and should set all drivers on an even basis of delay. Without PhotoLink (such as with an original OmniMic), there is a relatively simple technique, described by the late Jeff Bagy, for dealing with delay effects based on multiple individual and combined driver measurements.

To account for SPL sensitivity, the measurements should be made in "dB sensitivity mode" to account for driver sensitivities. This will keep driver measurements compatible with others done the same way to allow you to investigate different combinations with a crossover simulator. If you don't have a PhotoLink box, you can also use generator tools in OmniMic to set the drive level to the standard 2.83Vrms.

The sound pressure that the microphone will pick up also depends on how far it is from the driver during measurement. Ideally, this should be done at the standard 1 meter (39.37 inches) distance. But if that is impractical because of small measuring areas, you can enter the actual distance into the dB sensitivity mode's settings box and the software will approximately compensate for the difference.



The compensation assumes an ideal 6.02dB attenuation per doubling of distance, the accuracy of which will depend on the driver technology (and sometimes, size) involved. Ribbons and electrostatic panels will drop off more slowly.

Index

How to... (some commonly asked questions)

Overlay frequency response curves with minimal user interaction

When you have a curve you want to keep on the screen during future measurements, and don't want to go through the steps of saving the file and then reloading it as an Added Curve, you can just go to the AddedCurves menu and click "Add Live Curve". (When you first do this you will be prompted for a directory in which such file curves will be saved, after that the directory will just be automatically used and the curve will simply be added to the list of Added Curves). Up to 40 such overlayed curves can be added. The colors of each curve will be changed in a succession, but you can change the color by clicking on the curve name in the AddedCurves menu.

Splice two or more frequency response curves together in a graph

This can be easily done using Added Curves. Just save the responses to FRD files, then use the "Added Curves">"Add" menu to bring in both curves (ONLY TWO CURVES! -- with no other AddedCurvess loaded).

When you load an Added Curve, or use the Added Curves menu to select an already-loaded curve, an editor/properties box will open for that curve. Use the controls in that box Uncheck the box marked "All Frequencies" to allow frequency range selection and use the given controls to configure the curves' frequency ranges that you want to use from each. For example, you might select 20 to 1000Hz for one curve and 800Hz to 20,000Hz for the other. The frequencies need not overlap at any frequencies, though it is a good idea to have some shared valid frequency range between them if you can. Also select "Show Phase curve" checkbox for both curves to make that visible.

You can adjust the gain (dB offset) and/or the delay (to connect phase responses, be sure to show the phases of both curves to do that) of each using the editor/properties form for Added Curves. Then in the editor for either curve use the "save spliced to another" button, and select a location on your computer (a folder or the desktop) where the new file is to be created. The resulting FRD file will be the spliced curve result (you can then load that one as another Added Curve if you wish to see it). If the original curves overlay each other, then they will be combined smoothly over the shared range. If there is a gap between the frequency ranges of the original curves, a smooth arc will be drawn to connect them.

Play test signals out of the computer's sound hardware (or sound card)

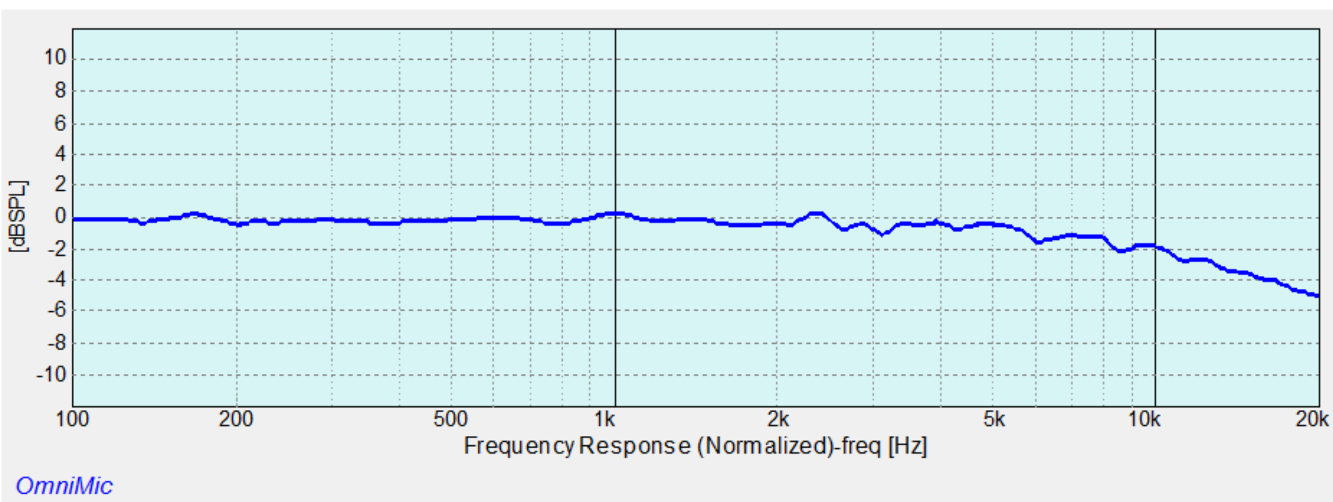
Look in the Config menu, there you can enable or select the sound, adjust the sound level or modify the output bandwidth (for Frequency Response or Distortion sweeps), or gain acces to the Windows sound controls.

Is there a fast way to disable sound output when I'm using the computer to generate test signals directly from OmniMic?

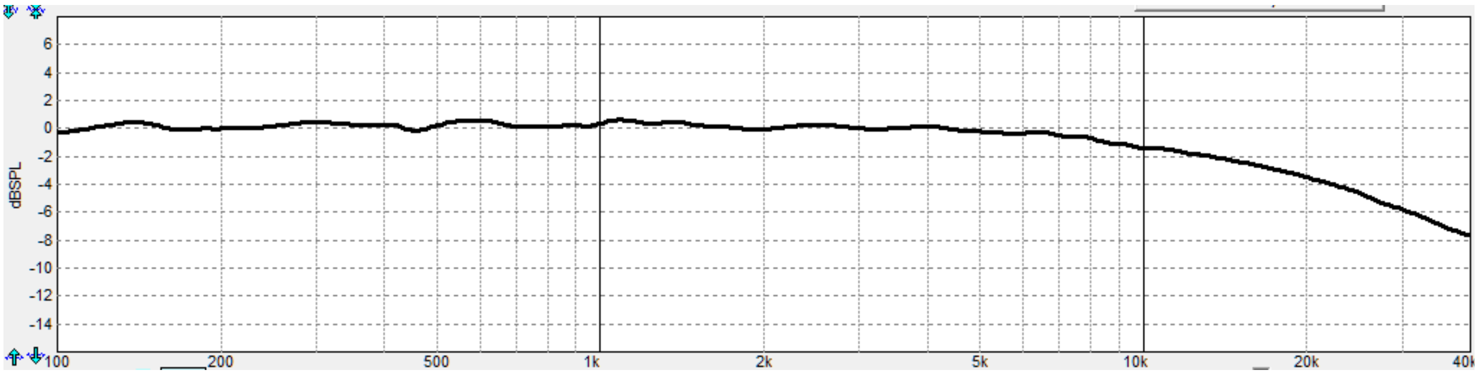
Yes, simply press on the [Esc] key on your keyboard. Press [F2] to start it again.
Or, you can **press the [Space Bar] to pause or un-pause the system**.

Point the microphone

OmniMic is a small capsule, narrow tip measurement microphone and its response is essentially omnidirectional at most frequencies. At very high frequencies, there is about 3dB reduction in perpendicular response at 13kHz (18kHz for OmniMic40k). This can be easily determined by making a set of comparative measurements with the mic pointing at and then perpendicular to a high frequency speaker, and then normalizing one by the other. Here is a result of such a measurement and process (showing the effect of perpendicular to directly aimed response for orignal OmniMic and OmniMic40k):



Mic facing upward compared to mic facing forward



Using more than one OmniMic microphone

The OmniMic software is able to work with several OmniMic microphones, though only with one at a time. And only one instance of the OmniMic software can be run on a computer at a time. But an OmniMic microphone cable can be removed from the USB port of the computer while the software is running, and a different OmniMic can then be plugged in to go on working.

But different OmniMics use different calibration files. So there is now an option in the Config menu for a "Microphone selector". This is a form that contains buttons and title blocks for up to 5 OmniMics. To set up for each, click on its title block (the round button at its left will then turn red) and then click on the "configure" menu in the selector form. You'll be prompted for a name or description of the mic, which might be something like "On Axis", "30 degrees", "nearfield" or the like. Then you will be given a file box for finding the calibration file for this microphone (which needs to be already on your computer).

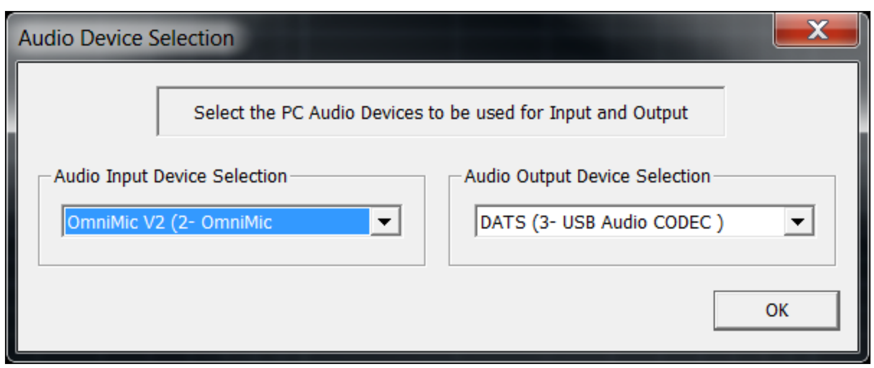
After this has been setup, you can quickly change between mics. Unplug the mic previously being used and plug in the new one. Then click on the name or button for that mic on the selector form and its calibration will be quickly loaded for new measurements.

Use a DATS device to provide OmniMic test signals

DATS can provide a consistent high-quality signal source for testing up to 20kHz. DATs should in general be set (in the Windows Control panel for **playback** devices) for "DVD quality" (48kHz, 2 channel mode). Its output could then be used to drive the power amplifier that drives the test speakers. For simpler connection or testing with OmniMic40k beyond 20kHz, a **Dayton Audio DAC01** is a good choice.

Sometimes use of computer based signal sources with AC powered audio equipment can result in 'ground-loops' or other noise from using common AC power sources. If the computer is a laptop, operating the laptop off of battery power will usually solve this issue. Sometimes changing power outlets for the computer or amplifier system can minimize the issue as well. There are also inexpensive USB ground isolator devices which can be found online and which can greatly reduce this kind of noise.

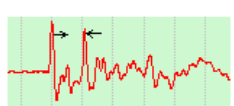
To use DATS, this procedure is best:
1) Connect the OmniMic and then the DATS unit to the computer.
2) Open both the OM and DATS software, go to the DATS Edit menu => Audio Device Selection and set playback (Audio output device) to DATS, recording (Audio input device) to OmniMic:



3)Click ok => Alt-Tab to view OmniMic main page and continue normally (i.e. click Config => Play from Soundcard, check SPL, mic ID etc.). The "Play from Soundcard" option can be found in the OmniMic "Config" menu.

Align two speakers in "time of arrival" at a listening position

Arrange your system so that the Test Track sound plays out of both speakers you are working with, but with any other speakers silent. Choose the Frequency Response page and play the indicated track with the microphone set at the desired position. Set the control that is near the top to "blended" and observe the Impulse Response display. If there is a time of arrival difference between the two speakers, you should be able to see two impulse peaks. The one to the left (near the 0 msec point) is from the speaker with the earlier delay time, the other is for the later one. If you have an electronic delay unit, adjust the delay applied for the earlier speaker so the peaks become superimposed. If you have more than two speakers you wish to align together, choose the latest (the one making the right-most peak) to be the reference and delay the others to align to its peak



Optimally position speakers within a room

Speaker and listener positioning within a room is probably the most significant factor determining how a system will sound. This can usually be best adjusted using the Frequency Response display and either the Pseudo-Noise or the Sine Sweep signal. Set the control near the top to "All" for a display in which all echoes, reflections, and room effects are included and apply 1/3rd octave smoothing. Start with one speaker playing at a time. The response will normally be irregular, but try moving the speaker position or listening position for the most uniform response, particularly at lower frequencies below 200Hz. For stereo, you will need to find two locations which can give a good stereo image together while optimizing the low frequency response. Very often such factors are not mutually attainable. Low frequency positions may also be optimized to minimize standing waves by use of the Bass Decay analyzer.

A strong advantage of using separate subwoofers (rather than only full range speakers) is that they allow you to optimize the mostly non-directional subwoofers for smoothest bass response and the upper frequency speakers for best stereo imaging. While bass tends to be non directional, the in-room response of the bass speakers is extremely sensitive to placement. You will almost always be able to get the smoothest overall low frequency response by using two or three subwoofers optimally placed in different parts of the room, rather than a single subwoofer - even if the single subwoofer were of much higher quality. The room position and diversity of placements will have significantly more effect on bass frequency response than could subwoofer response flatness.

Adjust an Equalizer (EQ)

OmniMic has a very sophisticated feature for manually or automatically configuring an equalizer, and downloading settings to MiniDSP type equalizers through "Biquad Parameters".
When adjusting equalizers:

- adjustment at lower frequencies should use a measurement setup similar to the one for positioning speakers (but the placement should be determined **first**). Strong boosts with the equalizer should be avoided at any frequency, but response peaks can be effectively removed ("notched") to any degree necessary.
- equalization at higher frequencies are usually best done only in moderation, and generally adjusted with echoes removed by setting the Frequency Response analyzer to "blended" and clicking the mouse within the Impulse Response display on a point just before the first strong reflection. Avoid sharp narrow-band adjustments on the equalizer at higher frequencies because they can cause significantly worse sound at slightly different listening positions.

Adjusting EQ for more than one listening position

For theaters or other situations with multiple listeners, compromise settings must be made to provide the best sound overall for all listeners. Begin by optimally positioning the low frequency speakers (or main, if full range) to avoid major problems at various seats, particularly those furthest from the center. Again, use of several well-placed low frequency speakers will be most effective for giving good results for all listeners.

Then, with the Frequency Response analyzer set to "All" and with 1/6th octave smoothing use the Averaging features. Make a series of measurements at all (or at least a good representative number) seats, starting with "Clear Averages" then "New Average" for the first, and clicking "More Averages" (or pressing the spacebar) once with the OmniMic placed at each position. This will result in an overall average room frequency response, which you should save to disk or print for future reference.

An alternate, and more powerful method, is to save FRD files of curves measured at each position and then bring up the "make Weighted Average" menu (under "Files"). There you can load and select the saved files and assign a relative weighting to each of them as desired. This will result in an Average curve which can be used for determining best overal equalization.

You would normally then adjust your equalizer to provide approximately the opposite of the resulting response; the new Equalizer Configuration form makes this easy. That is, if there were an 8dB peak (relative to response at most frequencies) in the 4kHz range, then if a flat response is desired, then the equalizer should be set for -8dB at 4kHz. Avoid applying large increases (more than about 7dB) with the equalizer and use only moderate and broad (not sharp in frequency) settings at frequencies above about 500Hz.

Arrange and test whether a frequency response falls within a specified range (QC loudspeaker testing)

The "Evaluate within" function (in the "Math" section of the Frequency Response menu will show a small rectangle at the top left of the frequency response graph that indicates whether measured responses fall within ("good") or outside ("out") of the first two curves specified in the Curves menu. The evaluation is performed only over the frequency range currently being displayed. To use this to determine whether new measurements fall within (for example) 3dB of a reference measurement, first make the reference measurement and save it to disk using the "File->Save Curve" menu. Then, in the "Curves" menu, choose "Add" and then select the file you just saved and then select an offset of 3dB on the form that appears. Use the "Curves->Add" menu again to choose the same curve, this time with an offset of -3dB. The two curves should show on the graph. Next, go to "Advanced Mode" (if you aren't already there) using the checkbox provided and in the "Math" menu, select "Evaluate within". You can select whether to allow the program to try to fit the curve by shifting it up or down by up to some maximum dB value. Finally, set the frequency range of the graph (using the yellow arrow buttons) to display the frequencies over which you wish the evaluation to be performed.

You can alternately define the two curve files to be used for limits by generating them using a text editor (simply load an FRD file into your text editor to see the simple format. All frequencies need not be present, but they do need to be in increasing order). Users familiar with spreadsheets such as Microsoft Excel can use these to edit FRD data for special custom limit curves, after resaving the resulting data with an FRD extension (or saving with a .txt or .prn extension and then changing the extension to .frd).

Measure frequency responses of Tweeters

OmniMic normally tests loudspeakers with a full range signal that briefly passes from below 2Hz to above 20kHz. Some speaker drivers, particularly tweeters, compression drivers, ribbons, or electrostatics cannot be safely driven at low frequencies. When testing devices such as these, you should always drive them through series capacitors (at least) to limit low frequency drive. You can calculate the approximate value to use as C=1,000,000/(6.2*f*Z) where f is the desired cutoff frequency, Z is the driver impedance, and C is the value in uF. Smaller values of capacitor are safer
.
Using such a capacitor causes a problem, though -- it also affects the measured response. If you test with a cutoff capacitor that acts far below where you tend to use the speaker then those effects might be ignored, but in that case you would be less protected.

Another option is to test with a pre-filtered (but still synchronous with OmniMIc) signal that has a known highpass response. The Config->Soundcard Output Level menu will provide a tool to allow you to taper at either frequency extreme of frequency sweep signals such as for Frequency Response, Harmonic Distortion or Compression.
IMPORTANT NOTE: You should **still use a series capacitor** (perhaps something large such as 100uF) to protect a sensitive driver from transients, hum, low sweep energy or other signals from the setup which might cause damage. **ALWAYS** use a blocking cap if you are testing a ribbon type tweeter without its crossover!

When you use those test signals, the measurements will be much more susceptible to noise below the rolloff frequencies, so you should adjust the displayed frequency range to avoid the lower frequencies.

Measure SPL sensitivity of loudspeakers ("2.83V 1w/1m")

Loudspeaker sensitivity is normally expressed as the SPL level that is sensed at 1 meter from the loudspeaker when it is driven from a voltage of 2.83Vrms. The ideal way to do this is with a PhotoLink device, which also has an indicator which shows when an OmniMic sweep signal is at this level. See documentation for the PhotoLink.

If you **don't** have a Photolink, the nominal level of the swept sine signals can also be determined (or set) by use of the 55Hz reference tones that are at available, as they are at the same peak levels as are the sweep tone test signals.

To use these, you may want to disconnect the speaker of the channel being tested (the output voltage of quality amplifier should not be affected by whether a speaker is being driven). Then attach a Digital Voltmeter (also known as a "DVM") to read AC Voltage across the amplifier terminals. Be sure to configure the voltmeter for AC Volts and that the probes connect to the meter's Voltage (and not its Current!) reading terminals.

Turn any equalizer or tone controls off. You can then play the 55Hz reference tone from the Soundcard Play Adjust Tool (or 50Hz test recording tracks) and measure or set the voltage level from the amplifier. Next, reconnect the loudspeaker, set OmniMic at the 1 meter distance from the loudspeaker and play the desired compatible sweep signal. Use the OmniMic software in its Frequency Response measurement mode for Sine Sweep, and the plot should then be a reading of SPL sensitivity (at the measurement distance used).

Some loudspeakers reference to a 1 watt level rather than to a voltage level. 2.83Vrms is equivalent to 1 Watt into an 8 ohm (resistive) loudspeaker. For a 4 ohm loudspeaker at 1 Watt, the voltage to use would be 2.0Vrms, for a 16 ohm speaker use 4.0Vrms. The most useful files, however, will always be as referenced to 2.83Vrms voltage level, as this defines the sensitivity of the driver which is needed for crossover designs.

FAQ

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Some answers to questions that may arise...

There is a message with a yellow background telling me "Check for Correct Signal Track!".

Most tests with OmniMic (other than "SPL/Spectrum" and "Oscilloscope") require particular test signal tracks to be played on the system that you are measuring. The tracks contain precise test signals necessary for the OmniMic software to do its analysis. **Do not substitute other test signals**, as any results would have no meaning! For example, although the "Pseudo Noise" (CD Track 1) signal sounds like pink noise, it is actually very different and use of pink noise in its place will not produce a useful result. The proper signal type names to play for any OmniMic test is displayed at the top of the OmniMic screen.

Played test signals seem to have gone crazy!

In rare cases, this can happen when the computer's recording and playback processes become very corrupted due to unusual processing interference or hardware problems (it's best to not have the computer doing unrelated complex operations while trying to make measurements!) or intermittent connection of a USB connector. In the **Config menu**, there is a menu item for **Reset System.** If you use this, there will be a several seconds delay while the system attempts to get itself aligned again, then the problem should be fixed. If that still doesn't fix it, try shutting down the OmniMic software, unplug the OmniMic, then restart the software and plug OmniMic in again.

Sometimes, the sweep tones may seem to "hiccup" or restart part way through a sweep. This is the software trying to position the output timing with respect to the OmniMic's acquisition process, and will usually self correct after a few sweeps. If it occurs often, there may be a difficulty in the setup, such as excessive noise levels or a bad connection.

What's the switch on the side of the Omnimic40k do?

The Omnimic40k is capable of handling signal levels beyond 140dBSPL (which is very loud) with low distortion. When you are working with levels above about 125dBSPL, you should put the switch toward the back (wider end) of the microphone. This inserts an internal 24dB attenuator into the analog signal path so that the A/D converter inside the Omnimic is not overdriven by the high signal levels. When this switch is in, be sure to put a check mark in the box, at the top right of the main form, that is labeled "Attenuator used" so that the software knows to correct the displayed levels to match the inserted attenuation.

What's the silver and black bump on the side of the Omnimic40k?

That's the sensor that picks up the timing signal from the PhotoLink box when you need to include true time of flight in measurements. See Using the OmniMic Photolink device

I get messages saying that my "Control Panel" settings are not correct for OmniMic (original version).

This can happen in modern Windows operating systems. For proper operation of OmniMic at higher sound levels, you must make sure that the system recognizes the microphone as a 2 channel microphone (the second channel is used for handling higher signal levels). The message gives specific, easy, directions for making the proper settings. The exact procedure may change in later Windows versions, so you may have to manually open the Windows Control Panel and locate the pages for making the needed settings. The settings may need to be done for each of the computer's USB ports that might be used with OmniMic.

The top of the OmniMic window shows the message "OmniMic not found".

This indicates that the presence of the OmniMic is not detected. OmniMic hardware must be attached to a working USB port on the computer for the program to be able to make measurements. Try using a different USB port on the computer and make certain that the blue indicator light (at top or back of the OmniMic) is illuminated. Unplug any other USB-based audio hardware from the computer, and if necessary try rebooting the computer. ***Please note that OmniMic cannot typically be used reliably with Windows versions earlier than Windows 7, with Macintosh OSes or under Linux or Linux/Wine.*** Windows 10 or 11 is advised.

The graphs of frequency response (or other measurements) is varying at lower frequencies.

This is probably because the test signal is not being played loud enough to overcome the background noise. Try playing the test signal louder or moving the microphone closer to the speaker(s). For many systems, the response at lower frequencies may be insufficient to overcome the noise.

Impulse response and phase traces seem to jitter left and right during measurements.

This is normal to a small degree. It should not happen when using OmniMic40k with its PlotoLink deveice. But because OmniMic without a Photolink can't know when the test signal was emitted at a loudspeaker, it must refer phase and time displays to the largest peak of the recovered impulse response shape, which can vary in level and polarity with responses that have multiple peaks of similar level. If you know the first peak is positive (or negative), then you can select the synch polarity with the [+] (or [-]) buttons at bottom left of the Impulse Response display.

Harmonic Distortion plots don't seem to make sense at frequencies where response dB values are low.

This again results from insufficient test signal level arriving at the OmniMic relative to noise. High levels of background noise or weak signal levels will show as distortion. You can find the distortion "threshold" of a measurement due to noise by simply silencing the test signal. Only distortion levels that appear significantly higher (at frequencies of interest) than this threshold when the test signal is playing should be assumed to be valid.

I can't measure harmonic distortion at higher frequencies -- the curve drops off the plot

Original OmniMic can "see" sound only in the range of human hearing (20kHz and below), so it cannot see harmonic energy that falls at frequencies higher than this. So 2nd harmonic cannot be measured above 10kHz; 3rd harmonic cannot be measured above about 6.7kHz, etc. (Double the frequency limits mentioned here, when using an OmniMic40k and an appropriate 96ksps test signal).

Why do square waves played through my speakers look weird (using the Oscilloscope)?

Most loudspeakers cannot reproduce square waves well. This may be due to imperfections in frequency response or phase response (the Frequency Response must be both flat and with phase near 0 degrees -- "linear phase"). The reproduction of a square wave that looks like the ideal typically requires the speaker (and OmniMic) to have flat, linear phase response from about 1/10th to 10 times the frequency of the square wave y ou are looking at. It will not be possible to show a square-looking waveform outside of the range of about 5 0Hz to 2kHz because the audio range of the microphone is 5Hz to 20kHz. Distorted looking waveforms could also be because of sound reflections off of surfaces within the room. Try moving the microphone closer to the speakers to see if the shape is better.

The frequency response at very high frequencies seems to drift around over time.

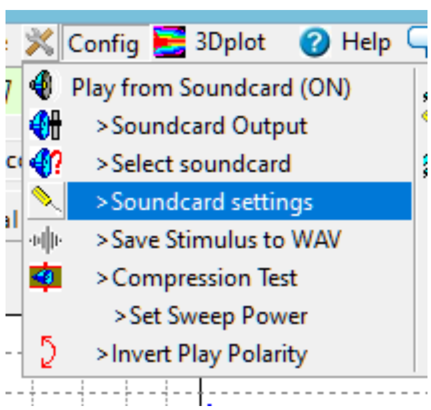
This is likely to be a side effect of slightly different sample clock rates used in OmniMic and your digital signal source. To minimize this effect, choose the "sine sweep" signal rather than the "pseudo noise" signal for measuring frequency response for your more critical measurements. Room positioning of loudspeakers and setting of equalizers at lower frequencies can generally be done using the more pleasant pseudo noise signal, if desired.

The impulse response of my speaker shows ringing *before* the impulse happens! That's impossible.

This happens when your speaker has strong response out to and beyond 20kHz (or 40kHz for OmniMic40k). OmniMic can only "see" out to about 23(46)kHz (displays to 20(40)kHz) because of its 48(96)kHz sample rate. It accurately calculates both magnitude and phase response right up to the highest audio frequencies, but as far as it can tell, the response drops completely beyond 23(46)kHz, and data above that is unknown. The correct impulse response that would be equivalent to a response that is strong with smooth phase to some frequency, and then its magnitude drops away above that frequency, will have "pre-ring" in the shown response. This doesn't mean that the speaker has output before it sees input -- it only means that an actual full band time response corresponding to a seemingly abruptly ending frequency response would have to have such a characteristic. If higher frequencies were known for the device being measured, and could be included, they would cancel the pre-ring (notice that the pre-ringing wave appears to have a period of approximately 1/23(1/46)kHz, well beyond audibility, where the data stops). Or, if the response rolled off naturally before 20(40)kHz (like human ears or most speakers do), that would result in a phase response, delay, and HF attenuation which would prevent the apparent pre-ring. But a perfect impulse response measurement of a broadband response that is strong up to a frequency where it suddenly disappears ***should*** have the pre-ring, it is **not** an error but is how the system must be represented in a limited bandwidth system.

I get almost no response above 20kHz:

With the original OmniMic hardware, response data is limited to 20kHz.
With OmniMic40k hardware, response up to 40kHz can be measured, BUT the test signal source and any following amplification, must be configured for 96kHz sampling rate or performan to at least 40kHz. Note that many class D amplifiers do not reproduce signals above 20kHz well or at all, so they are probably not idea for this purpose. If you are getting the test signal from your computer in Windows, make sure that its playback sample rate is set for 96ksps or higher.



My measurements show a rolled-off high end. I know my speakers are flat and my system is all very High-End.

First, make sure all system tone controls or equalizers are disabled (just in case). But check your signal source (file player, CD or DVD player, or even some computer sound cards) for "creative" signal handling -- some very expensive CD players are doctored to "fix" a misunderstood "pre-ring" time-domain problem and in doing so mess up the frequency response of their playback. Usually, inexpensive players don't have this problem. You may have to spend *less* money to get a *more accurate* test signal source!

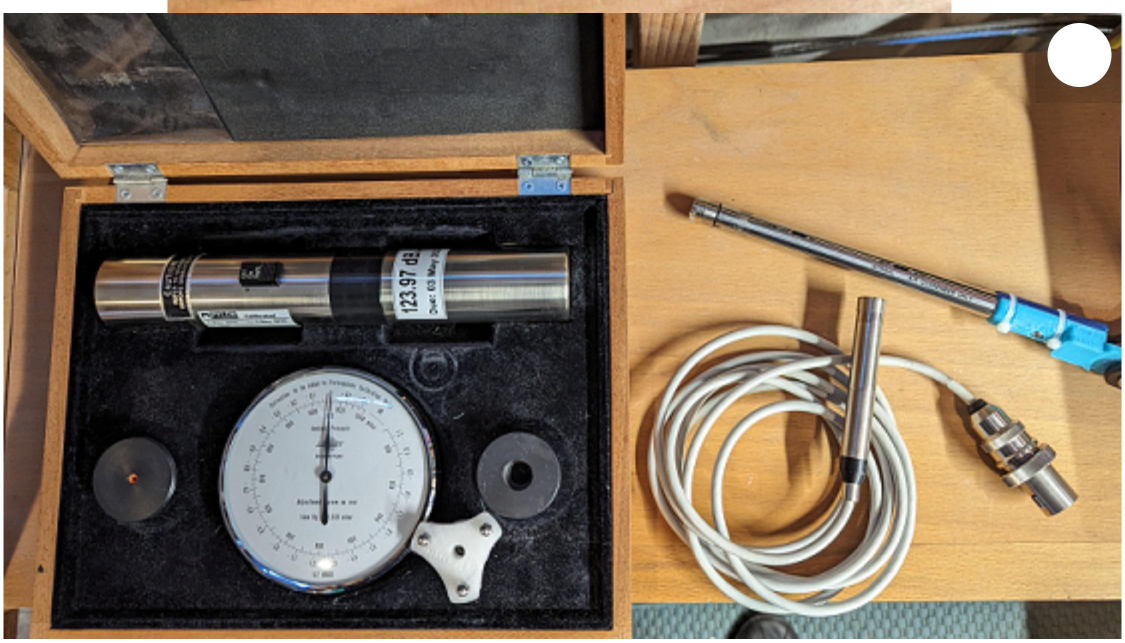
For PC-based **playback** systems used to provide the test signals, make certain that your computer is adjusted to play those in **DVD quality** (for origina**l** OmniMic Hardware) or **"Studio Quality" for use with OmniMic40k hardware**. That is, 48ksps rate for original OmniMic and 96ksps rate with OmniMIc40k hardware. PCs can and often do use different quality settings for different audio output devices (DACs, etc.) so it is best to check each one in the Windows Control Panel. You can use the "Config->Soundcard Settings" to get to the controls for this.

(Under recording interfaces, however, the orignal OmniMic hardware should always be set for 2-channel DVD quality)!!

If using a receiver or integrated amplifier for driving test speakers (when testing speakers, rather than the speaker and amplifier system), be certain that any tone controls for it are set to flat, and any EQ or loudness controls are turned off. Many class-D or class-T amplifiers (but not Hypex UcD or nCore, or IcePower) are not optimal for loudspeaker testing, as their frequency responses at high frequencies can be affected somewhat by loudspeaker impedances. A class-AB type amplifier or a class-D type mentioned above is usually preferable for most accurate testing purposes.

Calibration References of OmniMIc40k Hardware

Calibration of the OmniMic40k units is derived from the electrostatic deflection method and a MEMS DC responding accelerometer reference for frequency response shape. Accuracy will normally be within 2dB from 5Hz to 40kHz, and much closer over most of the audio bandwidth. Absolute SPL level (sensitivity) is calibrated based on a Bruenl and Kjaer pistonphone.



A master reference microphone consisting of a Bruel and Kjaer 4133 capsule with high voltage preamp/body was calibrated for frequency response shape (audio frequencies to 50kHz) using Bruel and Kjaer electrostatic actuator UA-033. The frequency response pressure curve was then corrected for free-field use per B&K published curves. The calibration lab was not able to calibrate low frequency response below 20Hz. Instead, a driven diaphragm with a DC responding accelerometer attached was measured by both accelerometer output and by the master reference microphone in the very-nearfield to generate a response curve from 5Hz to 200Hz. The upper frequency and low frequency curves were then combined to form the overall response shape curve of the master reference microphone.

The master reference microphone was then used in a free-field quasi-anechoic setup with a broadband loudspeaker to cross-calibrate reference microphonese of the same type 4133 capsules and preamps for production use. The process was performed driving OmniMic40k circuit boards fed from the microphone preamps' outputs. This provides reliable response data from several hundred Hz to 40kHz. At lower frequencies the production references were calibrated in the very-near field using a sealed 10" broadband woofer, comparing curves from the production reference and the master reference microphone to generate the low frequency response shape.

The frequency response curve shape of each OmniMic40k is derived by calibrating against the production reference microphones by the same technique used to calibrate the production references against the master reference microphone (combining separate high and low frequency curves determined in the far field and the very-near field respectively). The raw response shapes above 20kHz tend to be wavy within several decibles due to internal hardware equalization of the microphone response used to compensate approximately for general resonance effects. The individual response shape and SPL reference value are stored to an ".omm" file identified by serial number for use by OmniMic software to software correct measurements made by the microphones.

Calibration of absolute level uses a set of Metrosonic calibrator devices which were themselves calibrated using very stable Bruel and Kjaer 4220 pistonphones. The calibrator devices were used with a special adaptor to fit the narrow diameter of the OmniMic40k'ss barrel for individual absolute SPL calibration of the OmniMic4k units. The SPL calibration is performed with the OmniMic's "-24dB" attenuator turned off and with the sensitivity slider of the software set to the highest position.

Test CD Track Listing

The test CD used with the original OmniMic units cannot be used with OmniMic40k for measurement beyond 20kHz, as CDs can only reproduce up to 22kHz. It will operate correctly to 20kHz, but data at higher frequencies will not be meaningfully graphed. A full band (96ksps or higher) signal source such as the computer's sound output jack, or a WAV or FLAC file played through a high definition media device, will be required to measure audio content above 20kHz.

1 Track 1: Monophonic Pseudo-noise sequence

2 Track 2: Monophonic Short Sine Sweep

3 Track 3: Monophonic Long Sine Sweep

4 Track 4: Monophonic Bass Sweep

5 Track 5: Left Channel, Pseudo-noise sequence

6 Track 6: Left Channel, Short Sine Sweep

7 Track 7: Left Channel, Short Sine Sweep, bass removed

8 Track 8: Left Channel, Short Sine Sweep, bass and midrange removed

9 Track 9: Left Channel, Long Sine Sweep

10 Track 10: Left Channel, Bass Sweep

11 Track 11: Right Channel, Pseudo-noise sequence

12 Track 12: Right Channel, Short Sine Sweep

13 Track 13: Right Channel, Short Sine Sweep, bass removed

14 Track 14: Right Channel, Short Sine Sweep, bass and midrange removed

15 Track 15: Right Channel, Long Sine Sweep

16 Track 16: Right Channel, Bass Sweep

17 Track 17: Warning

18 Track 18: Left Channel, 50 Hertz Tone for voltage measurement

19 Track 19: Right Channel, 50 Hertz Tone for voltage measurement

5-Year Limited Warranty

See daytonaudio.com for details

All information presented is accurate at the time of publication.
Dayton Audio reserves the right to make improvements without notice.

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