SpectraFoo

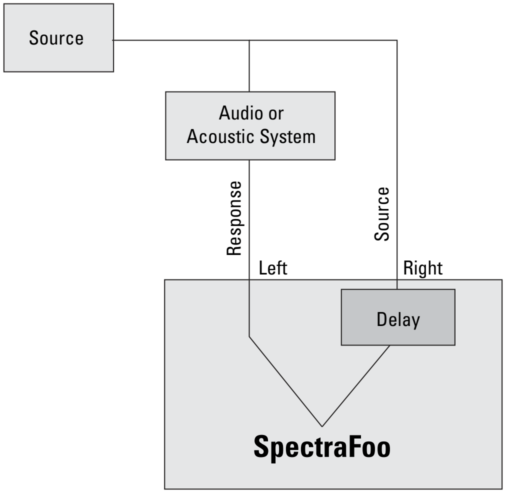
v4.0

User Manual

METRIC HALO

The Transfer Function

For live sound, equipment testing, or any application where you need to analyze the characteristics of acoustical and electrical audio systems, SpectraFoo provides Music Based Measurement (MBM™). SpectraFoo uses a transfer function display to show you the relative power and phase response between the left and right channels. The transfer function display assumes that the signal connected to the computer's right input is a source signal and the signal connected to the left input is a response signal.



SpectraFoo Signal Arrangement for MBM Transfer Function

When the source signal is the input to some audio processing arrangement and the response signal is the measured output of the system, you can use music as your "test tone". SpectraFoo uses the source signal as a point of reference and the transfer function display shows the differences in amplitude and phase between the source and response as function of frequency. This allows you to measure the properties of audio processing systems, including systems that contain acoustic elements. You can determine the amplitude and phase response of an equalizer in the presence of musical signal as easily as measuring the sound coloration of an acoustic space.

The transfer function really only makes sense when the source signal is the input to some audio processing arrangement and the response signal is the measured output of the system. It does not generally make sense if the the source and response are truly unrelated, as in the case of the two channels of a stereo mix (in this case meaning a multi–track mix, with panned elements, as opposed to a true stereo program, such as an X–Y recording).

Even in the case of a stereo mix, the transfer function may provide some information about the spectral balance of the recording, but, in general, the spectral balance is not constant in time.

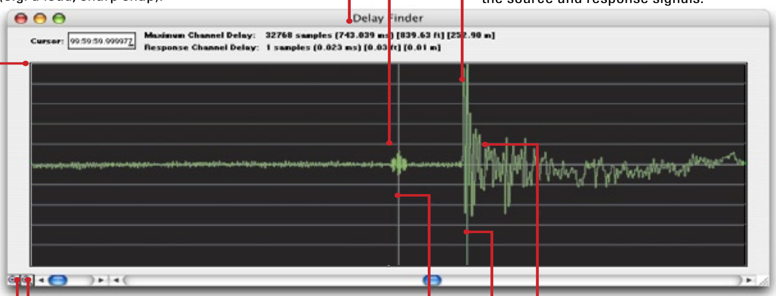
Even when the source and response signals are related, they usually will not be time-aligned. For example, if you are measuring the room response of an auditorium, there will be the speed–of–sound acoustic delay between the source signal and the response signal received at the measurement microphone. In order to properly measure the power and phase of the response signal relative to the source signal, the source signal must be delayed to time-align it with the response signal. SpectraFoo provides a delay detection feature that is accessed via the "**Compute Delay**..." button present in the Transfer Function window. The delay finder records a segment of both signals and computes the correlation of the signals with each other. By moving the cursor to the largest correlation of the two signals, you set the system delay and compensate for the measurement delay, time–aligning the two signals. Now the transfer function measurements will be accurate.

Transfer Function of a Room Response with a time delay

The delay finder uses the Fourier deconvolution theorem to compute the impulse response of the signal processing elements that the response signal has passed through. The deconvolution theorem allows SpectraFoo to, in essence, "divide" the stimulation signal out of the response signal and create the impulse response of the signal processing system being measured. The impulse response is the signal that we would measure if we stimulated the system with an impulse (e.g. a loud, sharp snap).

The impulse response is essentially noise for all times before the impulse has propagated through the system being tested.

This means that the time between Time "0" and the first spike in the impulse response is the delay between the source and the response signals. If you move the cursor of the delay finder to the first spike you will delay the source signal by the appropriate amount to time align the source and response signals.



Click here to Zoom in.

Click here to Zoom out.

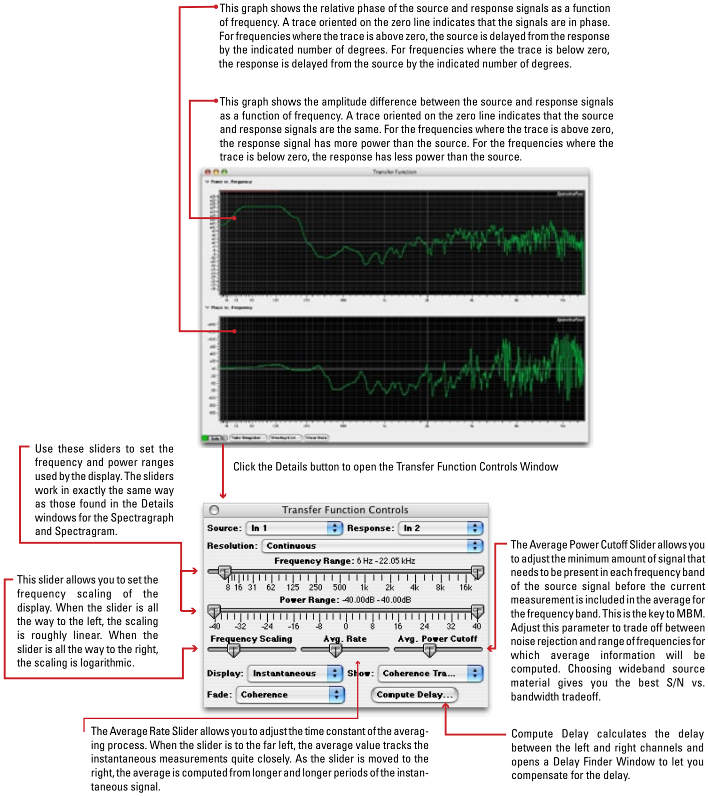
Time "0"

Cursor

The delay readouts provide precise information about the relative delay between the left and right channels for the current cursor position. This information is provided in samples, milliseconds, and in feet (assuming standard temperature, humidity and pressure). The maximum inter-channel delay supported by SpectraFoo is about 750 milliseconds.

All of the other spikes present in the impulse response are the time delays of the various "reflections" in the system. These can be real reflections from walls, tables, etc. or they could be virtual reflections from the digital delays in a reverb algorithm or electrical reflections due to improperly terminated studio wiring.

Time Aligning the measurement with the Delay Finder Window



Transfer Function after the measurement has been time-aligned

The parameter controls for the Transfer Function are accessible from within the Details window of the Transfer Function. This is consistent with the operation of all other SpectraFoo instruments. The Transfer Function details window is accessed by clicking on the show details button in the Transfer Function window or in the Master Controls window. In addition, the Transfer Function now has a solo button and an on/off button.

Coherence

The Transfer Function window has a Coherence trace which is displayed as a red trace in the power vs. frequency display. Coherence has a value of 0 when the trace is at the bottom of the display and has a value of 1 when the trace is at the top of the display and varies linearly in between.

Coherence is a measure of how well the response signal correlates with the source signal. When Coherence is at its maximum value of 1 for a given frequency band, the source and response are perfectly correlated and the Transfer Function is completely uncontaminated by noise. When Coherence is at its minimum value of 0 for a given frequency band, there is no correlation between the source and response and the measurement in this frequency band is invalid. Coherence can be used as a guide to determine which frequency bands are equalizable. Frequency bands for which Coherence is low cannot be corrected by equalization. Frequency bands for which Coherence is high are equalizable.

The Transfer Function Window

The Transfer Function system provides all the tools necessary to make high quality measurements of simple systems and will allow you to easily measure and correct small-scale sound reproduction systems like control-room monitors, home theaters, and small club sound reinforcement systems.

The transfer function environment is contained in the ⇢ **Transfer Function** window. There is only one Transfer Function window in the system. It can be shown, like any other instrument in the system, using the **Master Controls** window. The control item for the **Transfer Function** appears as the second item in the list in the **Master Controls** window.

The Transfer Function has a small number of controls in the main window:

1. ⇢ ▽ Power vs. Frequency ⇢**Power vs. Frequency** disclosure button – When this control is set to “disclosed” (arrow pointing down) the Power vs. Frequency data panel is visible. To hide the Power vs. Frequency data panel, set this control to “undisclosed” (arrow pointing right).
2. ⇢ ▽ Phase vs. Frequency ⇢**Phase vs. Frequency** disclosure button – When this control is set to “disclosed” the Phase vs. Frequency

data panel is visible. To hide the Phase vs. Frequency data panel, set this control to "undisclosed."

3. ⇢⇢**Instrument Enable** button – This is the button with the IEC power symbol. When this button is "On" (filled with green), the Transfer Function is running.

4. ⇢Solo⇢**Solo** button – When this button is "On" (filled with red) the Transfer Function is soloed.

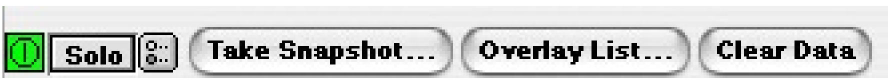
5. ⇢⇢**Show Controls** button – This is the button between "Solo" and "Take Snapshot…". Clicking on this button opens the Transfer Function Control window. The Transfer Function Control window provides all of the detailed controls used to adjust the display of the transfer function.

- <control>-clicking this button pops-up a menu that allows you to select the **Source** channel for the Transfer Function.
- <option>-clicking this button pops-up a menu that allows you to select the **Response** channel for the Transfer Function.
- <command>-clicking or clicking and holding this button pops-up the parameter library menu for the Transfer Function.

6. ⇢Take Snapshot...⇢**Take Snapshot**... button – Clicking on this button takes a snapshot of the current Transfer Function data and adds it to the **Transfer Function Snapshots** window. You can use Transfer Function snapshots for a number of measurement and comparison tasks. These are described in greater detail later.

7. ⇢Overlay List...⇢**Overlay List**... button – Clicking on this button shows the **Transfer Function Snapshots** window.

8. ⇢Clear Data⇢**Clear Data** button – Clicking on this button clears the current transfer function data and resets the transfer function to flat. This is useful when you start measuring a new system or device or move a test microphone and you want to start the measurement from scratch.

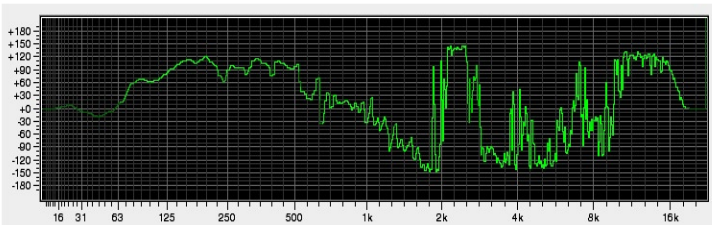


The Transfer Function data is displayed in two different panels:

Power Panel

The power panel displays the relative power between the response signal and the source signal as a function of frequency. The relative power curve is drawn in green. The vertical calibration is in units of dBr (decibels relative). The horizontal calibration is in units of frequency (Hz). The limits of both the vertical and horizontal scales are controlled with the Transfer Function control window.

The power panel also displays the coherence of the measurement as a function of frequency. The coherence curve is displayed in red. The vertical scale of the coherence is linearly scaled. The vertical limits of the coherence (which are not displayed) are "0" at the bottom of the graph and "1" at the top of the graph. These limits do not change when you change the power limits.



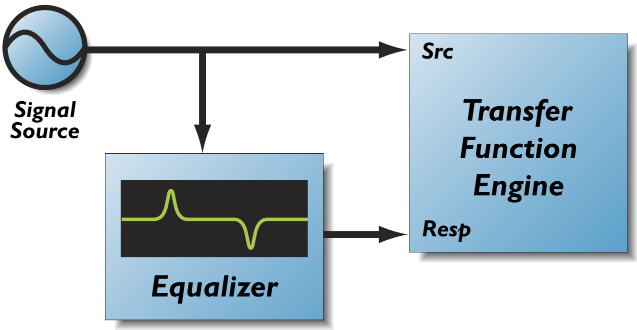
Phase Panel

The phase panel shows the relative phase between the response and source signals. The vertical scale of the phase panel is linear and is in units of degrees. Since phase is cyclic (that is, if the phase of a signal is x then x+360° is the same phase), the phase curve can wrap around from 180° to -180°. You can see this in the graph above at f = 2kHz. The phase does not really have a discontinuity at 2kHz, it has just wrapped around to +180°. The horizontal calibration is in units of frequency (Hz). The limits of the horizontal scale are controlled with the transfer function control window.

Understanding the Transfer Function

The purpose of the transfer function measurement is to allow you to find out what a particular electro-acoustical system is doing to the signal it is processing. We call the system being measured the System Under Test (SUT – also called DUT in other literature). The SUT could be as simple as an equalizer or as complex as a multi-zone sound reproduction system coupled with a measurement microphone (or many measurement microphones, for that matter). The key thing to understand is that ***every*** system ***changes*** the signal that runs through it.

An example of a simple measurement is finding the transfer function of a parametric equalizer. To make this measurement we would hook the equalizer up like this:



Simple Measurement setup

The signal source could be any source as long as it provides energy in the frequency bands that we are interested in. If we wanted to measure the response of the equalizer from 20Hz to 20kHz, a simple 1kHz sine wave would not be appropriate. Good source signals are:

1. Broadband musical sources. These sources have the advantage that they exercise the SUT in a way that is consistent with the way that we expect to use the SUT.

2. Pink Noise. Pink noise has the benefit that, on average, it has a power spectrum that is consistent with most musical program material. It also ensures that the entire audible bandwidth will be exercised. Pink noise can be annoying when played over a Sound Reinforcement system, but should not damage any system components (e.g. high-frequency drivers).

3. White Noise. White noise does not really match the characteristics of music but it does have a uniform density of energy in the entire audio band. It can be used for testing electronic components but is very annoying when used in acoustic testing. You also run the risk of damaging high-frequency drivers if you play white noise through a sound reinforcement system at high SPL.

4. Swept sines. A sine sweep that covers the audio band to be tested will allow you to make an extremely precise, highly noise-rejecting measurement. Swept sines do not bear any resemblance to actual musical signals, can be really annoying when used for acoustic testing, and do not allow you to compare what the transfer function is telling you to what you are hearing, but they do allow you to make very accurate measurements in noisy environments. You can use this type of source signal when you have to test a system while other aspects of a venue are still being set-up.

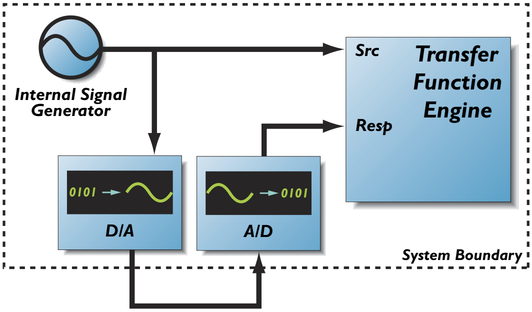
SpectraFoo's Transfer Function Measurement system needs to see both the source signal and the response signal to build the transfer function. The measurement created is a differential measurement and removes all of the common elements of the measurement path.

You can use an external signal (like music from a CD or DAT player), a SpectraFoo capture, or SpectraFoo's built-in signal generator as the signal source. SpectraFoo provides an internal loopback path for capture playback and the internal signal generator, so you can route the internal signal source back to the transfer function without using one of your A/D channels.

If you do use the loopback path, your system's digital

to analog and analog to digital converters (D/A/D path) are not common to both the source and the response signal.
If the frequency response of the D/A/D path is not perfectly flat (in both phase and power) its response will color the measured response.

You can easily measure your D/A/D chain by looping back your system's analog output directly into it's analog input. Make a measurement using the internally looped-back signal generator as the source and the system's analog input as the response. The measured transfer function is the response of your D/A/D chain.

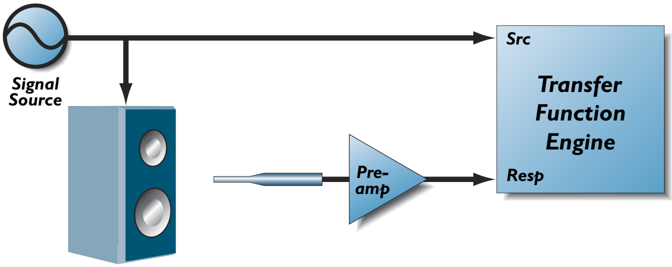


Setup for checking your D/A/D chain

If the response is sufficiently flat, you can use the internal loopback method. If it is not, then you must use an external loopback of the source as the source for the transfer function (since this will remove the D/A/D chain from the measurement).

One of the strengths of SpectraFoo's Transfer Function Measurement system is that it allows you to do Music-Based Measurement (MBM). This means that the system can automatically apply a threshold to the source signal to make sure that it is only measuring the response when the source is actually stimulating the SUT. MBM allows you to make accurate transfer function measurements even when the source signal is not stationary (like music) and when the signal "comes and goes" in various frequency bands (like music). MBM builds the transfer function up over time.

In order to be able to create an accurate measurement of the system response when the test signal is non-stationary, the response signal and the source signal have to be synchronized in time. If the signals are not synchronized the relative measurement of response will be wrong. For stationary signals (like pink and white noise), even if the source and response are somewhat unsynchronized the power measurement will be pretty good, but the phase measurement is useless.

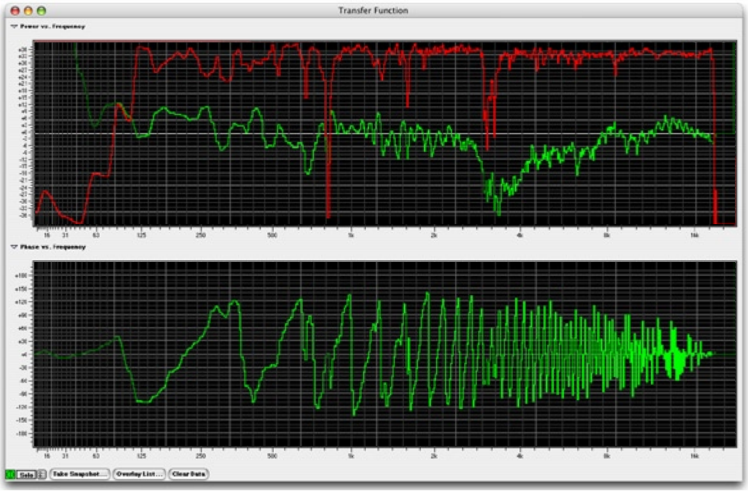


Acoustic test setup

Under normal circumstances the response signal will always be delayed from the source signal. Sometimes this delay can be very short (for example, if the signal source is external and the SUT is analog and contains no acoustic elements, the delay can be less than a sample). If the SUT contains acoustic elements (like a speaker), there is always the acoustic propagation delay. In any case, the delay must be compensated to achieve an accurate measurement.

When we make a typical acoustic measurement we are measuring all of the elements that are not common between the source path and the response path. In the case illustrated above, the speaker, speaker amplifier, acoustic space, measurement microphone, and preamp are not common. The amplifier, speaker and acoustic space are the elements that we want to measure. The microphone and preamp are not. In order to eliminate the pre-amp from the measurement we can pass the source signal through the same type of preamp or we can use a preamp that is flat (the LAB preamps from Earthworks are good for this purpose). We cannot eliminate the effect of the microphone, but we can reduce it by using a quality measurement microphone. Once again, Earthworks comes to the rescue with relatively inexpensive, high quality (flat from 9Hz to 30kHz) measurement mics.

When we start the transfer function using the setup shown in the figure above, we get a measurement that looks like this:



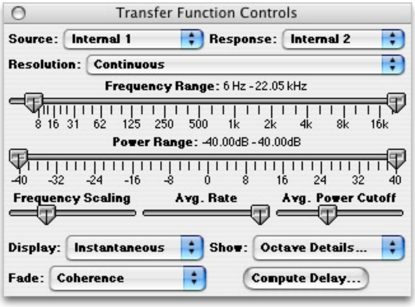
Transfer function without delay compensation

Since this measurement was made with the microphone close to the speaker, the acoustic delay is short. As a result the power measurement is reasonably accurate and the coherence is near “1” for most of the audio bandwidth. The phase, however looks kind of like a sawtooth. This indicates that there is a time delay between the source and the response. If the time delay was larger, the phase would be a sort of random line with values near zero and the coherence would be “0” or close to zero for the entire audio bandwidth. Basically, we need to time align the signals.

SpectraFoo provides internal delay compensation to allow you to time align the source and response signals. To accomplish this, you use the **Delay Finder**. The **Delay Finder** is accessed from the Transfer Function Control window.

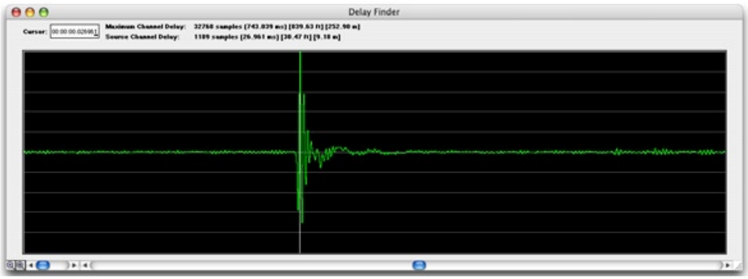
To time align the source and response signals:

1. If the **Transfer Function Control** window is not visible, open it by clicking on the **Show Controls** button in the **Transfer Function** window. The **Transfer Function Control** window will appear:



Transfer Function Controls Window

2.Click the **Compute Delay**... button. The **Delay Finder** window will appear:



Delay Finder window

3. The Delay Finder will automatically find the compensation delay and internally compensate the source signal.

The Delay Finder records 1.5 seconds of the source and response signal and computes the impulse response of the SUT.

The impulse response of a system is the signal that would come out of the SUT if you fed in an impulse. An impulse is sort of like a clap or a snap, but with an infinitely short duration of time. The impulse response of a perfect wire is a spike (the impulse) offset by the propagation delay through the wire.

For a real acoustic SUT, the impulse response will not be as simple as a spike. The time delay (lag) of the maximum value of the impulse response is, in general, the delay time of the SUT. If the SUT is acoustic, we expect to see a number of smaller copies of the impulse response at different lags. These copies correspond to acoustic reflections from walls, desks, etc.

If the SUT is time aligned, the impulse response will be relatively compact.

The impulse shown in the Delay Finder Window figure above is of a small professional PA speaker. The small dip in the impulse response that occurs before the main impulse corresponds to the start of the impulse response of the low-frequency driver. The larger spike is the start of the impulse response of the high-frequency driver. The speaker is not fully time-aligned. As a result, we will not be able to fully time-align the measurement.

SpectraFoo's internal compensation can delay the source signal by up to 1.5 seconds. This corresponds to an acoustic delay of almost 6 football fields (at standard temperature and pressure), and should be sufficient for most work. If you need to compensate a larger delay, you will have to use an external delay line.

The **Delay Finder** automatically finds the lag at the maximum value of the impulse response and uses that time delay to set the internal time delay compensation on the source channel. You can tweak the delay by moving the cursor in the **Delay Finder** window.

After we have compensated for the response delay the transfer function measurement becomes:

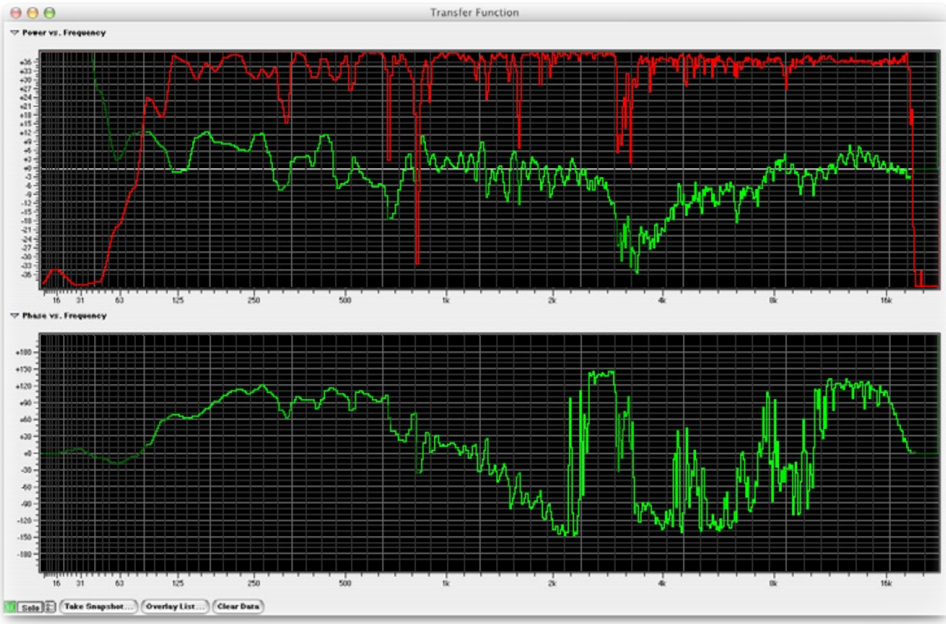


Figure 85: Time-aligned transfer function

Notice that overall the coherence of the measurement has increased and that the phase no longer has the "sawtooth" character. As explained earlier, the phase is not discontinuous at 2 kHz – it is just wrapping around due to the cyclic nature of phase. The reason we cannot get the phase measurement to appear continuous for this system is related to the fact that the system that we are measuring is not time aligned. When the measurement of the low end is time-delay compensated, the high end is not and vice-versa.

Interpreting the Data

So now we have a measurement of the transfer function of the system. This leads us to a few natural questions:

1. What does the data mean?
2. How do we interpret the data?
3. How do we use the data?

The transfer function measurement system calculates three different functions from the source and response signals. Each of these functions adds a piece of the puzzle and will be instrumental in interpreting and using the transfer function data.

The first function is the power response of the SUT. This is the effective equalization that the SUT applies to signals that run through it.

The second function is the phase response of the SUT. This is the phase shift/distortion applied to the signal by the SUT.

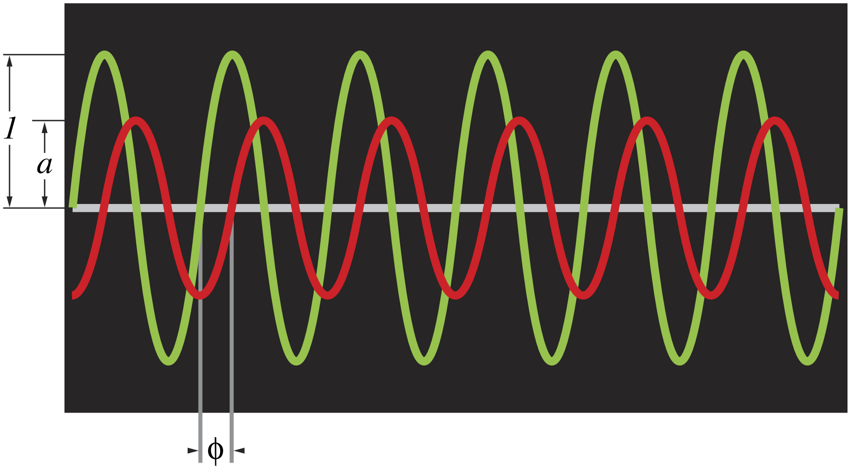
The final function computed by the transfer function measurement system is the coherence. The coherence is a measure of how much, on average, the response signal “lines up” with the source signal on a frequency by frequency basis. This is one of the most important pieces of the puzzle because it allows us to evaluate the quality and usefulness of the data. Coherence is a little hard to understand, so we’ll examine it in a little more detail.

About Coherence

Imagine the source signal is a simple sine wave. The response signal will, in general, be another sine wave with the same frequency but a different amplitude and phase. Mathematically:

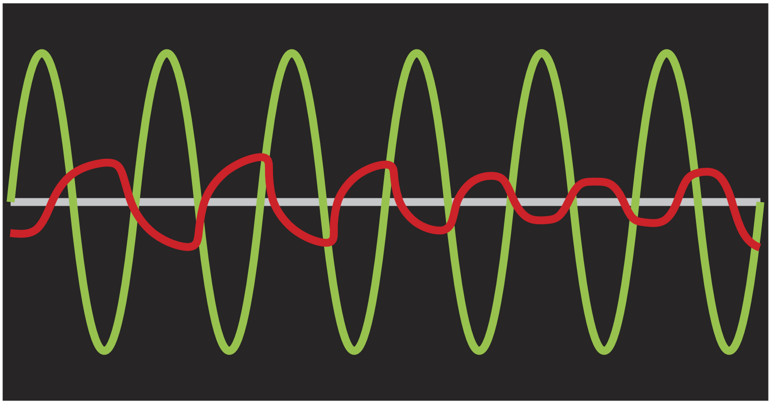
$$s(t)=\sin(2\pi ft)$$
$$r(t)=a\sin(2\pi ft+\phi)$$

The source signal $s(t)$ is a sine wave with unit amplitude and frequency f. The response signal $r(t)$ is a phase-shifted sine wave with amplitude a and frequency f. These two signals are shown below (the source signal is green and the response signal is red):



Simple source and response signals

Even though there is a phase shift between the source and response, the phase shift is constant in time. On average the signals line up the same way all the time. On the other hand, if the signals are not related to each other:



Simple source – Complex response signal

In this case, the phase shift between the source and response signals changes over time. This means that on average the signals do not line up with each other. The coherence will be close to zero.

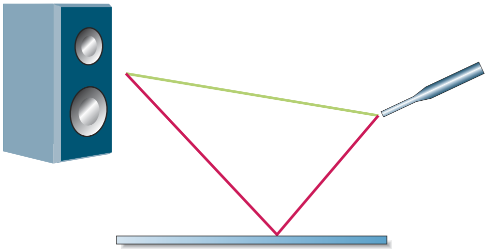
If the SUT is stationary (e.g. it is linear, does not change over time, and passes at least a portion of the source signal) the coherence will be “1” for all frequencies. This is what we expect when we make a time aligned measurement on an electronic system (like an EQ).

If the SUT is non-stationary or if the SUT does not pass any of the source signal in a particular frequency band, then the coherence in that band will be “0”. Unless we are having big problems, the frequency bands for which the coherence is small or “0” should be limited, even with acoustic tests. If the coherence is “0” or near “0” for most of the audio bandwidth, this indicates that the SUT is either not properly connected, the measurement has not been time-delay compensated with the delay finder, or that the SUT is sufficiently nonlinear that it is not possible to analyze it with the transfer function (a distortion generator is an example of such a device).

Under normal acoustic test conditions the response signal of the system will be the actual system response plus noise. The noise (which can be electronic or acoustic), is, by definition, uncorrelated with the source signal. The coherence of noise with any source signal (except an exact, time-aligned copy of the noise) is zero. As a result, the noise that corrupts the response signal will tend to decrease the coherence of the measurement.

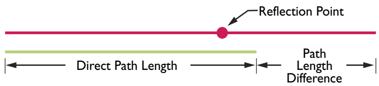
The fact that there is noise in the test environment is not normally a problem – when doing the test we arrange for the test signal to be louder than the surrounding noise. Even if the signal is not louder than the environmental noise at all times, the MBM thresholding allows the Transfer Function to only make measurements when the source signal is louder than the environmental noise.

There is one situation that is problematic when making acoustic measurements. When the system has acoustic reflections, the response signal may have hard or partial nulls at certain frequencies. These nulls depend on the specific geometry of the acoustic environment (especially the location of the test microphone) and don’t really represent the response of the system. A null is created at a specific frequency when multiple signal paths cause the test signal to cancel out at the test mic position:



Acoustic test with reflected paths

The reflected path introduces a copy of the signal with a time delay. The reflected signal takes longer to reach the microphone than the direct signal because of the path length difference.



Path length difference

If we call the path length Δl and the speed of sound v, the response at the microphone will have a null for frequencies where $f = n\frac{v}{\Delta l}$, and n is any integer. This is just a simple comb.

In any real acoustic environment there will be many reflective surfaces with many different reflectivities and phase shifts, so we will not see a simple comb filter but a complex set of partial nulls. At the partial nulls in the response at microphone position, little of the source signal will be detected; only the environmental noise will be measured. This means that at the nulled frequencies, the coherence will be very low.

This provides us with the key to understanding how to use the coherence. The coherence tells us where the measurement that we have made is valid. It is basically a measure of the quality of the measurement at the corresponding frequency.

In terms of correcting the response of acoustic systems, the coherence tells us which notches or dips in the response curve are *Equalizable* and which are due to nulls in the measurement. If the coherence is low at a dip in the response curve, we will not be able to use an equalizer to fix it.

If the null is due to the specific position of the measurement microphone, we can generate a better averaged response by making measurements at a variety of microphone positions and then averaging the measurements. This technique is described in more detail later.

On the other hand, if the coherence remains low for a variety of microphone positions, the cancellation of the source signal is not due to a specific reflection, but is actually part of the system response. In order to correct these types of problems, either the configuration of the system must be changed (e.g. moving the position of the speakers or adjusting the relative time delay of different clusters in the system) or the configuration of the acoustic space will have to be changed (removing reflections, adding diffusion, etc.).

About the Response Curves

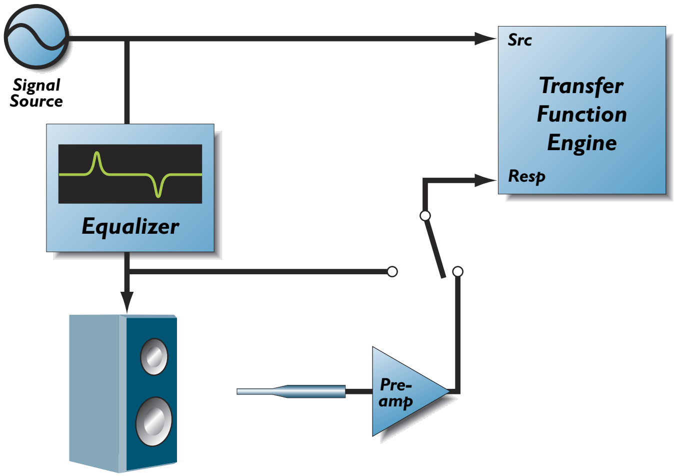
If the SUT is a simple processor (like an equalizer) the power response tells us the effect of the processor on the signal. Hopefully the response matches the expected response. In any case, the measured response is the actual response of the processor.

If the SUT is a sound reinforcement system, the expected (or more to the point, desired) response of the SUT is that of a wire. The system should, ideally, reproduce the signal exactly without any coloration. No real system will do this, but often the problems in a sound system are correctable. The data provided by the transfer function allows us to determine:

1. What parts of the system are correctable with equalization.
2. What equalization is required to correct the system.

As described above, the coherence allows us to determine what is correctable with equalization. The power response tells us what equalization is required to correct the system. The phase response can help determine what type of equalization will best solve the problem and also help determine if other phase changing components (like the crossovers) of a system are causing problems. It will also help you determine if the drivers in a speaker (or a cluster of speakers) are properly time-aligned (as shown before).

The basic system configuration that we will use to measure and correct an acoustic reproduction systems is:



Measurement and Correction setup

The equalizer in this setup is used to correct the response of the system. The switch above the pre-amp is used to select between the output of the equalizer and the output of the test mic.

To measure the system response:

1. Open the **Transfer Function** window if it is not already open.
2. Open the **Transfer Function Controls** window.
3. Flatten the system equalizer or bypass it. We want to measure the un-equalized response of the system.
4. Set the response input channel to the output of the test mic. You

can do this by selecting the proper input channel in the **Transfer Function Controls** window (if you are using multi– channel input hardware) or by routing your test mic to the computer input using an external routing system (e.g. a patch bay or a mixer).

5. Set the transfer function source channel to your signal source.

6. Turn on your signal source. Make sure that the signal source is a broadband source like pink noise or pop music. Classical music does not work as well. You must also ensure that the source is louder, on average than the environmental noise.

7. Make sure that you are getting both the source and response signal into Foo. You can use a level meter instrument to make sure that you are getting signal.

8. Check to see that the input levels of the two signals are reasonably equal in level. You can eyeball this with the level meter or you can use a power-balance meter to balance the input levels.

9. Adjust the **Avg. Power Cutoff** slider (in the **Transfer Function Controls** window) so that the pointer is in the middle. This will allow you to get an accurate time delay calculation.

10. Click the **Compute Delay**... button. The **Delay Finder** will appear and automatically compensate for the acoustic delay.

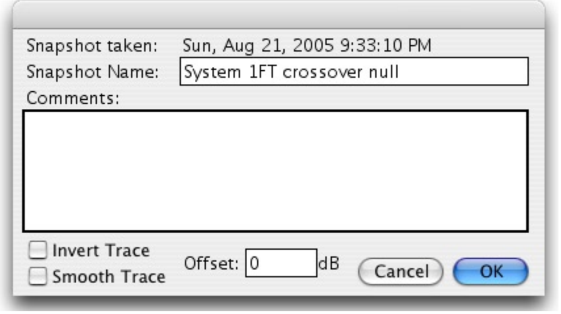
11. Adjust the **Avg. Power Cutoff** slider (in the **Transfer Function Controls** window) so that the pointer is near the left hand end of the slider. This will help the MBM thresholding system reject environmental noise. If the traces are not moving at all, select **Age** from the **Fade** pop-up menu. Ensure that the traces are flashing bright green all over the entire audio bandwidth. If they are not, decrease the **Avg. Power Cutoff** slider until the traces start flashing. (You do not want the trace to be completely bright green – just flashing).

12. Adjust the **Avg. Rate** slider so the pointer is near the middle of the slider. This will cause the Transfer Function to average over a

reasonable period of time.

13. After a short period of time the measurement should settle down. If all is well, the coherence should be near the top of the graph for most of the audio band. If you are using music, you may not see much activity in the very high end (above 16kHz).

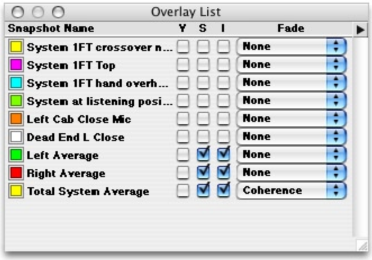
14. Click the **Take Snapshot**... button in the **Transfer Function** window. The Transfer Function Snapshot dialog will appear:



Transfer Function snapshot dialog

15. Enter a descriptive name for the snapshot (e.g. “Control Room Response”). The time and date of the Snapshot is automatically logged. You can enter any comments pertinent to the snapshot in the comments box. You can also tell the system to smooth the snapshot data (when it is displayed) and to invert the snapshot data (when it is displayed).

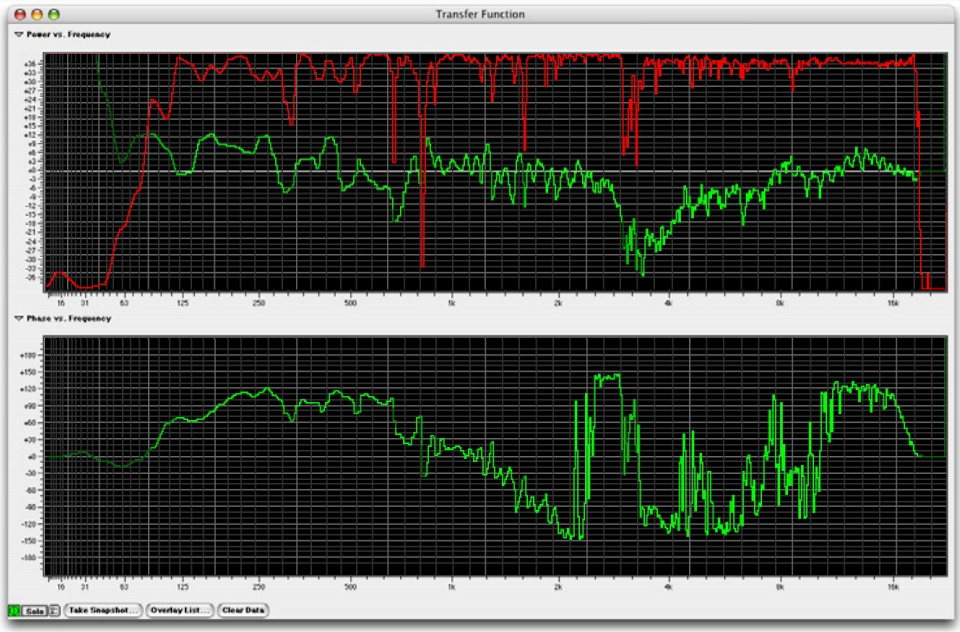
16. Click the **Invert Trace** checkbox and then the **OK** button. The **Overlay List** window will appear (the list in this figure has a number of overlays; yours will have just the new one):



Transfer function Overlay List window

You now have a complete measurement of the transfer function of your system. You may want to make a number of measurements for further averaging, but what we have now is good enough to illustrate the correction process.

Here is an example of a measurement that we have made using the procedure described above:



Sample transfer function

We see that the coherence is reasonably high (with a few narrow exceptions) from about 80 Hz to 18 kHz. Below 80 Hz, the sound system is not really reproducing the test signal and the environmental noise is pretty high. In addition, the test signal does not have a tremendous amount of energy below 60Hz. The test signal did not have much energy above 18 kHz, so we have not really measured the system response from that point on.

There are a few significant dips in the system response. Some of these are quite narrow and correspond to significant dips in the coherence curve. These dips are most likely caused by measurement nulls due to acoustic reflections and must be ignored since the coherence is so low.

Some of the dips in the response correspond to places where the coherence is high. These dips are real and can be corrected with EQ.

Notice that there are a series of closely spaced notches in the high end of the system response (8 kHz and up). These correspond to reflection nulls that will move around when the microphone position is changed, but will not disappear. They will make the system sound ragged on the high end and will also make the system sound phase-y when the listener moves around. These are caused by acoustic reflections and are best treated with diffusion or absorptive material.

We can now try to correct the system with equalization. What we want to do is adjust the system EQ to match the inverse of the system response. When we do this, we will pre-compensate the signal sent to the sound reinforcement system so that when it applies its effective equalization to the signal, the result balances out and leaves us with a (relatively) flat system response.

To correct the system:

1. Switch the response signal source to be the output of the system equalizer.

2. In the **Overlays** window, click the checkbox in the “V” column of the transfer function overlay that corresponds to the system response that you want to correct. This makes the snapshot vis-

ible in the **Transfer Function** window. (Make sure that the "I" checkbox is also checked. This Inverts the snapshot – which makes it look like the EQ curve that we need to correct the system.)The overlay is displayed in the color of the tile next to the snapshot name. If you want to change this color, click on the tile and choose the new color.

3. Select **Coherence** from the **Fade** popup menu of the snapshot in the **Overlays** window. This will fade the trace based on the coherence of the measurement. It allows you to see what portions of the response are equalizable.

4. Start your signal source. Pink noise tends to work best when you are measuring your EQ, because you don't have to listen to it. You can still use music if you want to be able to listen to the effects of the system EQ as you are changing the system EQ.

5. Since there is very little environmental noise in this measurement, you can decrease the setting of the **Avg. Power Cutoff** slider.

6. Again since there is very little environmental noise, and since we want to be able to see the changes in the equalizer response as we change the equalizer settings, you can decrease the setting of the **Avg. Rate** slider.

7. Click the **Compute Delay**... button to recompute the delay for the equalizer. The system will compute and compensate for the delay through the equalizer.

8. After a few seconds, the coherence curve should be near the top of the panel for the entire audio range and the power curve should be flat (this assumes that the EQ has been zeroed out).

9. Move the inverted system response curve so that the highest points on the curve are level with the live trace. You do this by clicking on the response curve and dragging it vertically until it is positioned properly. You want to make this adjustment so that you can correct the system using an all-cut EQ curve.

10. Adjust the settings on your system EQ to match the general trends of the inverted system response curve. Avoid trying to match steep peaks in the inverted system response… especially when the coherence is very low.

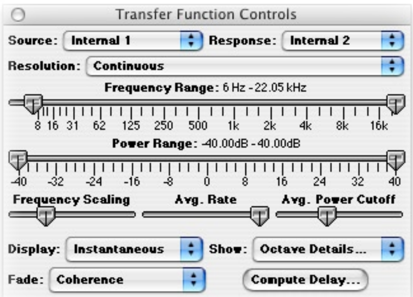
11. Assuming everything went well, your system has been corrected. It should now sound significantly better than when you started.

You can re-measure the system response (now with the system EQ switched in), to verify that the system has been substantially improved; and of course your ears will also let you know.

There are many other types of measurements you can make with the transfer function measurement system.

Transfer Function Controls Reference

You control the display and behavior of the transfer function window with the **Transfer Function Controls** window:



Transfer Function Controls window

Source pop-up menu – You use this menu to select the source channel.

Response pop-up menu – You use this menu to select the response channel.

Resolution pop-up menu – You use this menu to select the octave resolution of the analysis. You can select from:

1. 1 Octave – power is displayed as bars; each band is one octave

wide.

2. 1/2 Octave – power is displayed as bars; each band is one half of an octave wide.

3. 1/3 Octave – power is displayed as bars; each band is one third of an octave wide.

4. 1/6 Octave – power is displayed as bars; each band is one sixth of an octave wide.

5. 1/12 Octave – power is displayed as bars; each band is one twelfth of an octave wide.

6. 1/24 Octave – power is displayed as bars; each band is one twenty-fourth of an octave wide.

7. Continuous. – power is displayed as a continuous curve.

Frequency Range Slider – You use this range slider to control the visible bandwidth of the transfer function.

Power Range Slider – You use this range slider to control the visible power range of the transfer function. The power scale is automatically scaled to keep the display logarithmic.

Frequency Scaling Slider – You use this slider to control the scaling of the frequency axis of the transfer function.

Avg. Rate Slider – You use this slider to control the averaging rate of the transfer function analysis. The far left end of the slider is “No averaging” and the far right end of the slider corresponds to averaging over 2 minutes (approximately). The averaging done by the transfer function is a decaying exponential average, so it is difficult to assign a numerical value to the averaging rate. In general, acoustic systems require a larger averaging rate than electronic systems. If you use music for the test signal you need a larger averaging rate than if you use noise.

Avg. Power Cutoff Slider – You use this slider to control threshold level applied to the spectrum of the source signal. When the source signal level is higher than the threshold (this is computed for every spectral line in the spectrum) the transfer function adds the measurement to the average. If the level is below the threshold, the data is ignored. You can see which bands are above the threshold level by setting the **Fade** pop-up menu to **Age**. For every band that is above threshold, the response curves will be drawn in bright green. For normal acoustic testing, you want the response curves to be flashing on and off – not a constant bright green and not a constant dark green. This allows the MBM thresholding to reject environmental noise the best.

Display pop-up menu – This pop-up menu allows you to choose what data is displayed in the transfer function window:

1. **Instantaneous** – This mode displays each instantaneous measurement of the transfer function.

2. **Average** – This mode displays the thresholded average of the transfer function data.

Fade pop-up menu – This pop-up menu allows you to choose how the response traces are faded:

1. **None** – Don't fade the response traces.

2. **Coherence** – Fade the color of the trace based on the coherence of the measurement.

3. **Age** – Fade the color of the trace based on how long it has been since the source signal exceeded the threshold level.

Show pop-up menu – This pop-up menu select from a number options. Each option is independent of the others:

1. **Coherence Trace Is Visible** – Shows and hides the red coherence trace.

2. **Show Details** – Shows and hides the numerical readouts of the center frequencies and levels of each of the bands when the analyzer is in octave analysis mode.

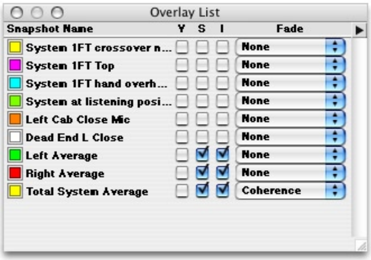
3. **Smooth Traces** – Makes the Transfer Function perform averaging in frequency space to display smooth response traces. This generally makes the response traces easier to interpret.

Compute Delay button – creates a new **Delay Finder** window and computes the impulse response of the system that is currently being measured. The **Delay Finder** will automatically compensate for the delay in the SUT.

Transfer Function Snapshot Overlays Reference

Use the **Take Snapshot**... button in the **Transfer Function** window to make a new snapshot from the current transfer function data.

All other snapshot overlay management is done from the **Overlay List** window:



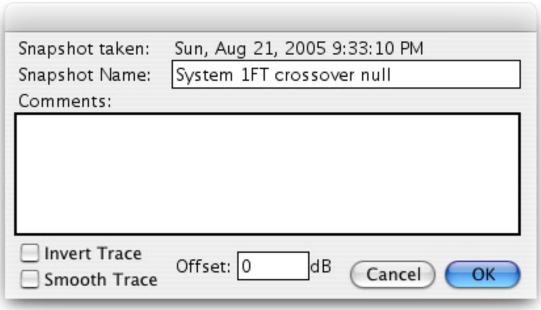
Transfer Function Snapshot Overlays window

This window lists all of the overlays that are loaded into the system. Each item in the list contains the following items:

1. A color well – This is the base color of the response traces in the snapshot. Clicking on the chip will bring up the color picker window and allow you to change the trace color.

2. The snapshot name – This is the name associated with the snapshot data. You can click on this to select the snapshot in the list. Double-click to bring up the transfer function snapshot edit dialog. You can change the name of the snapshot, its associated comments and the offset. The offset can also be controlled by

clicking and dragging the trace in the Transfer Function window. You can use the offset field to easily reset the trace position



Transfer function snapshot edit dialog

3. The 'V' checkbox – controls the visibility of the associated snapshot in the transfer function window. <option>-clicking this check box will set the state of all of the snapshots in the list.

4. The 'S' checkbox – controls the smoothing of the response traces of the associated snapshot in the transfer function window. <option>-clicking this check box will set the state of all of the snapshots in the list.

5. The 'I' checkbox – allows you to select if the response traces of the associated snapshot are inverted in the transfer function window. <option>-clicking this check box will set the state of all of the snapshots in the list.

6. The Fade pop-up menu – controls how the associated traces are faded in the transfer function window. <option>-selecting an item from this menu sets the state of all of the snapshots in the list.

The **Overlay List** window also has a mini-menu window command pop-down menu in the upper right-hand corner above the scrollbar (). This menu provides the following commands:

Save Selected Snapshots… – Select this command to save each of the selected snapshots to disk.

Load Snapshot... – Select this command to load a snapshot file from disk.

Remove Selected Snapshots... – Select this command to remove the selected snapshots from the list.

Average Selected Snapshots... – Select this command to create a new snapshot from the average of all of the selected snapshots. You can choose to weight the average with the coherence of the measurements. If you make a number of measurements with different mic positions and create a weighted average, you can remove the effects of local nulls and generate a more accurate measurement of the response of the system.

Smooth All Snapshots... – Select this command to smooth all the snapshots in the list.

Unsmooth All Snapshots... – Select this command to unsmooth all the snapshots in the list.

Invert All Snapshots... – Select this command to invert all the snapshots in the list.

Uninvert All Snapshots... – Select this command to uninvert all the snapshots in the list.

Show All Snapshots... – Select this command to make all the snapshots in the list visible.

Hide All Snapshots... – Select this command to make all the snapshots in the list invisible.

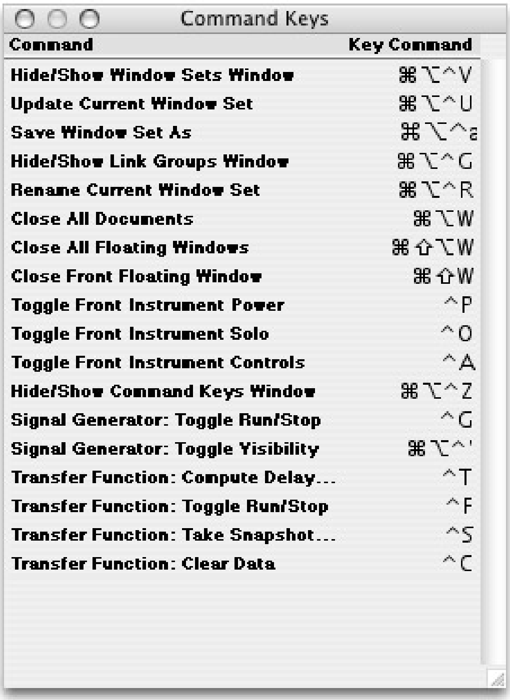
Ask About Unsaved Snapshots – When this item is checked, the system will ask you if you want to save an unsaved snapshot before it is removed from memory.

Cursors in the Transfer Function window

You can put a readout cursor in the Transfer Function window by moving the mouse cursor over one of the transfer function panels and holding down the <command> key. If the display is one of the octave analysis modes, the cursor will snap to the center of each band. The cursors in both the power and phase panel will move simultaneously. To move these two cursors separately, hold down both the <command> and <option> keys. To remove the cursor from the display, move the cursor to the vertical calibration of one of the panels.

Quick Menu Reference

Edit Menu This menu contains the standard editing commands. Currently **Select All** and **Show Clipboard** are not implemented. The **Edit Key Commands** menu item allows you to view or change the key commands for many standard menu commands:



The **Preferences** command allows you to bring up system-wide preferences for SpectraFoo. Currently, these preferences allow you to control the visual appearance of the components of SpectraFoo. In order to change the panel color you must have the “ColorPickerLib” installed in the extensions folder.This library comes standard with the Macintosh System Software. If this file is not installed,