

TRANSFER FUNCTION MEASUREMENTS WITH SMAART v8: QUICK GUIDE

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Why measurements are a quicker and safer way to optimize a sound system than doing it by ear

This article is intended for those who have not yet gathered the courage to start using measurement software as a support tool for sound system tuning, be it for lack of knowledge or insufficient encouragement. Here we'll cover the basics to guide you through the first steps such that the first contact comes out successful and encourages you to keep going. I recommend doing a quick first reading of the text and then going back to the beginning and reading it more thoroughly.

Making absolutely sure that there is nothing out of place within a system; that everything is in phase; that all compression drivers not only work, but perform consistently; that there is nothing disconnected, inverted, or with EQ or delay settings carried over from the last event; that our mixing board is doing what it is supposed to be doing... will avoid wasting your time trying to troubleshoot, by using trial and error, why one side sounds different to the other, or why your system does not sound the way it should.

While it can be hard at first, once you have practised the measurement techniques everything goes a lot faster than some of you might imagine. And, once the measurements are done, there is no room for doubts.

It goes without saying that the best time for practising is not while working in an event. This has to be learned when one is not under stress, at a time when nobody is bothering you and you can experiment. I'm sure you will find answers to questions that you always had, and that new questions will arise to which you'll find answers someday. People that do what we do are curious by nature.

In professional sound there exist two types of work that are widely different from each other, even though the same person may have to take care of both in small or medium sized rental companies. One is mixing, where it is the ear that counts, as do musical culture and good taste. Here the person behind the console is one of the musicians, even if not always recognized as such.

The other type of work, the one for which this article is written, is system tuning, which is diametrically opposed to mixing a band, and a type of work where our main tools are measurements, and only at the end we can rely on our ears to perform minor adjustments only if the previous work, which is purely technical, was correctly carried out.

If you do sound system tuning by ear, you risk noticing a problem mid-way through an event or, what's worse, once it is over. Or what would be even worse, never realizing it. We should never mix by measuring, nor tune a sound system relying entirely on our ears.

Nobody can reasonably justify that acquiring knowledge or using today's tools is unnecessary. Nobody can convincingly reason that phase aligning subwoofers, or a delay tower, can be done better and faster by ear than using acoustic analysis. So, you will not find a major tour without a systems engineer with measurement software and measurement microphones. And no, mayor tours will employ nobody to tune the equipment by ear. System adjustment is carried out by personnel who knows the job and the tools that are at our reach today, as well as being affordable by any professional.

We need our ears to mix, and add final touches to adapt the system to our taste or our mixing technique, with minor adjustments, once the purely technical issues have been dealt with, after being identified by measuring. As much as we like and intensely live professional audio, our subjective appreciations in a given time and context with respect to the tuning of a sound system may not always match those of the audience, musicians, co-workers, managers, producers ... or may not be sustainable over time. Not even the time that it takes for the concert to end.

Our measurement software does not lie, nor everything it shows is what it appears to be. The latter does not make it less valuable: we must know how it works and how to interpret it, as we'll go into detail below.

How an FFT-based measurement system works

A microphone connected to an electro-acoustical analysis (measurement) system will output voltage variations over time which are directly proportional to pressure variations, as seen in the following graph:

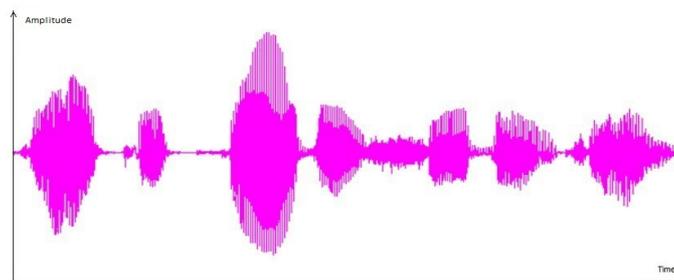


Figure 1: Signal picked up by a microphone.

Looking at these waveforms we can grossly estimate the signal dynamics, but little else. It does not provide sufficiently detailed information to be of any use with regards to the frequency response so that we can adequately evaluate the system we are measuring. Dual-channel FFT based systems process that signal as seen in the figure below:

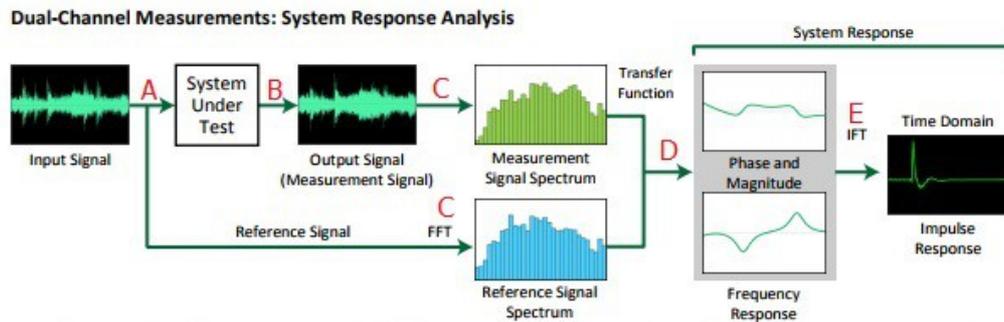


Figure 2: Block diagram of a two-channel FFT based measurement system.

The measurement system generates a signal that splits in two at point 'A': on one side it is fed to the system that we want to measure (say a loudspeaker, though it could also be a mixing console or an equaliser) and on the other side it goes back into the measurement system.

At point 'B', a microphone is picking up pressure variations produced by the loudspeaker, which receives the signal generated by our measurement system.

So far, we only have voltage variations as a function of time. On one side we have what we'll call the *Reference Signal*, which is the original signal generated by our system, and on the other side we have the *Measurement Signal*, which is the loudspeaker output picked up by the microphone.

At both 'C' points a Fast Fourier Transform (FFT) is applied to both signals, the *Reference Signal* and the *Measurement Signal*. FFT is a mathematical operation that allows for the spectral content of a signal to be obtained from its temporal variation.

At point 'D' we make a comparison between the frequency response of the two signals and from that comparison we obtain a magnitude and a phase response. It is what we call Transfer Function. At point 'E', running the inverse Fourier transform (IFT) of the comparison of the signals, we obtain the impulse response that characterizes, in this case, the measured response of the loudspeaker.

Looking at the diagram, we can conclude one important benefit from the use of an FFT-based 2-channel measurement system that we do not have with single channel systems. Since we are comparing the *Reference Signal* and the *Measurement Signal*, the type of signal we use does not matter, so long as it contains the range of frequencies we intend to measure. That is, we can use pink noise, a sweep, or music, for instance. One side benefit of the signal comparison is that it eliminates any undesirable effects introduced by the sound card.

Connections

The image below shows the connections required to measure a loudspeaker with a 2-channel FFT-based system:

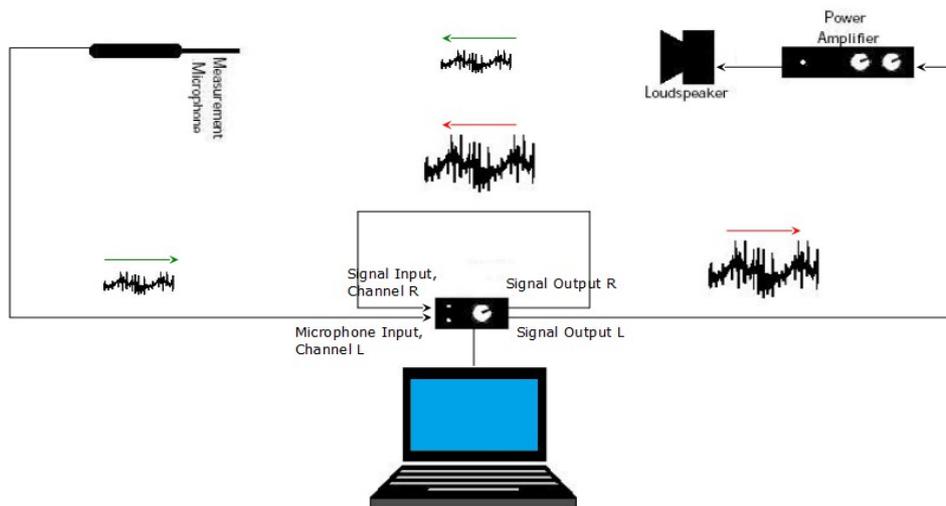


Figure 3: Connections for a two channel electro-acoustical measurement.

For clarity, on the diagram we placed the outputs on the right-hand side of the sound card and the inputs on the left. The connections are as follows:

Left Output: Connected to the amplifier that will, in turn, be connected to the loudspeaker.

Right Output: Looped back into the *Right Input*. It will be the *Reference Signal*.

Warning! The signal of the two outputs must be the same. We could not use, say, the two channels of a stereo wav file. Later in this article we'll explain how to configure Smart v8 to have the same signal on both channels.

Left Input: Connected to the measurement microphone. That is the *Measurement Signal*.

The connection diagram shows that, as explained earlier, the signal generated by our measurement system goes to the loudspeaker on one side and on the other one goes back to the right channel of the sound card. This is the signal we called *Reference Signal*, and denoted in the diagram by a red arrow.

The signal that gets picked up by the microphone, which we called *Measurement Signal*, is denoted by a green arrowhead and goes into the left channel of the sound card.

Warning! Many sound cards have a control knob, normally labelled "Mixer", with two positions, minimum and maximum, often labelled "Input" and "Playback" respectively. To be able to perform the measurements described in this article this control should be set at the "Playback" position, which will typically be the maximum clockwise position. Otherwise feedback is introduced from the inputs to the outputs, resulting in an unreal measurement of the system.

Propagation delay

As stated before, 2-channel measurement FFT-based systems compare the *Reference Signal* to the *Measurement Signal* from the microphone. This is what is referred to as finding the Transfer Function.

The *Reference Signal*, that comes out of the sound card and is looped back into it by means of a cable, gets to the input immediately, practically at the speed of light. However, sound that comes out of the loudspeaker travels very slowly, at the speed of sound, some nine hundred thousand times slower, before getting picked up by the microphone.

To be able to obtain a readable phase curve from the Transfer Function we'll need to introduce a delay on the input channel for the *Reference Signal* that is equal to the time that the wave takes to travel between the output and the input of the left channel. i.e., basically the acoustical wave propagation time.

This delay time will be calculated by the measurement system itself, only when we tell it to do so, and is a result of the comparison between the measurement signal and the reference signal.

In Smart, this propagation delay will be calculated when we press "Find"

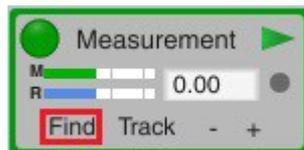


Figure 4: Smart will calculate the Propagation Delay when we click on "Find".

Coherence Graph

Quoting the Smart v8 manual, chapter 6, in the "Coherence" paragraph says : «Coherence is a statistical estimation of the causality or linearity between the reference and measurement signals in a transfer function measurement. Coherence does a good job of detecting contamination of the measurement signal by unrelated signals such as background noise and reverberation... The coherence calculation essentially asks the question, "How confident can we be that what we are seeing in the measurement signal at this frequency was caused by the reference signal?"»

As per Smaart v8's manual, coherence is mainly affected by these three factors:

- a) "A problem with the measurement system". For instance, we may not have synchronised the reference signal with the measurement signal by clicking on "Find". It could also be the presence of electronic noise, distortion, compression, limiting or crosstalk, though none of those are common according to the Smaart v8 User Guide.
- b) "Environmental noise causing measurement signal contamination".
- c) "Reverberation".

As stated earlier, quoting Smaart v8's manual, coherence detects if the measurement signal is contaminated by environmental noise or reverberation, i.e. it detects a low signal-to-noise ratio.

There is a case of low signal-to-noise ratio that is specially interesting, and that is when there are reflections or the signals from two systems arrive at different times. In those cases, since a cancellation is produced in one or more bands, the signal-to-noise ratio in those bands becomes very low, and, therefore, so does coherence. This is the case that we'll encounter most often and the one we'll discuss below.

Coherence is represented as a graph in the upper part of the magnitude response, and goes from 0% to 100%. There also exists the possibility of adjusting a coherence threshold such that the screen will not show those bands for which coherence is lower than the threshold we have set.

With electro-acoustic measurements it is very common that coherence is above 80% in the most part of the spectrum and that some bands appear for which coherence is low, which will typically indicate that there are two waves arriving at different times, normally due to the presence of a reflection.

A common error is to equalise a trough in the magnitude response for a band that has very low coherence. With all probability, that loss of level, combined with a low coherence level, is indicating that there is a reflection.

The next image shows the measurement of a system where a strong reflection can be appreciated in the impulse response, in the top window. Cancellations can be seen in the magnitude response around 400Hz, 1100Hz and 1700Hz, and low coherence at those frequencies as well as nearby ones. Also, at higher frequencies "rippling" can be observed which is due to cancellations that are very close to each other, and hence relatively low coherence can be appreciated at the frequencies at which those cancellations happen.

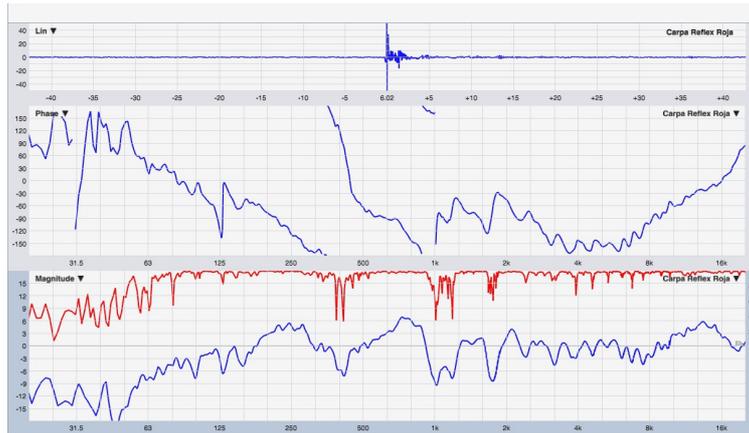


Figure 5: Effect of a reflection in an electro-acoustic measurement.

For comparison with the image above, for which we artificially produced a reflection for demonstration purposes, Fig 6 shows the same measurement without the reflection.

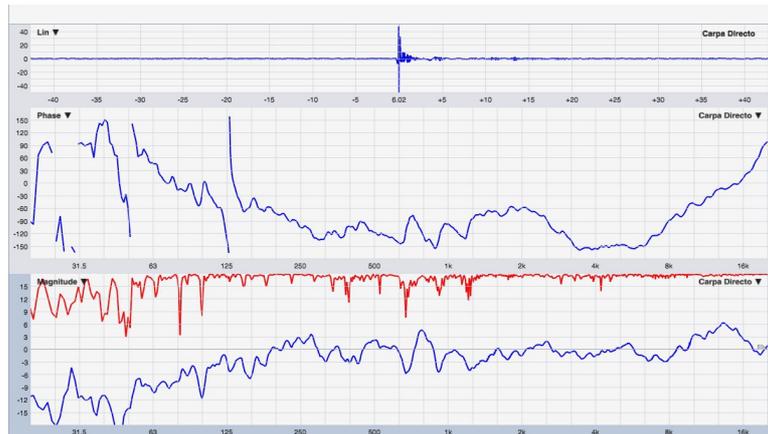


Figure 6: Same configuration as for Fig 5 but without the obstacle that caused the strongest reflections.

Fig 6 shows a single impulse response (though followed by other smaller ones further in time). The strong cancellations we saw before are no longer present, nor such low coherence can be observed, and no “rippling” appears at high frequencies. Furthermore, a more regular phase curve can be seen.

If we made the mistake of EQing the cancellation produced by a reflection we would see that the magnitude curve would get “fixed”, but if we looked at the coherence graph we would see that it would not change, it would continue being very low. And if we looked at the phase curve we would see a deformation, resulting from the different arrival times of the two signals as shown in figure 5.

These reflections happen every time we place a measurement microphone on a mic stand. The microphone picks up the direct sound from our system, but also the sound that bounces off the floor, as seen in Fig 7.

Other times that reflection can be produced by some element in the room or some obstacle that could be found in open air events. We need to learn to identify these reflections so as not to “fix” them with EQ.

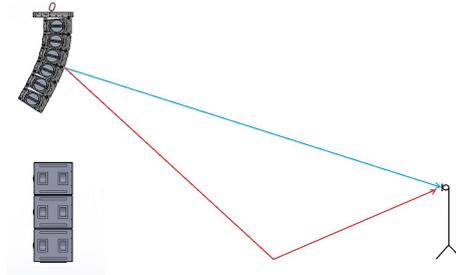


Figure 7: Reflected (red) and direct sound (blue) when a measurement microphone is used on a mic stand.

It is important to note that the cancellation shown by the measurement software in these situations only happens at the specific position where the microphone is located. At other positions, where the difference in distance between direct and reflected sound is not the same, the same cancellation will not be seen, and another one or more will appear at different frequencies.

Hence, when applying EQ to “fix” the magnitude response at that specific position, we would be colouring it at the rest of positions (where the microphone is not located), unnecessarily forcing the transducer, with the consequent risk of damaging it, as well as degrading sound quality by introducing equalisation that impairs the original signal.

If you've ever done this, now you'll be wondering: Why is it that, whenever there is a reflection and we apply EQ, the measurement software responds to that EQ and we achieve the same magnitude response that we would get without the reflection? To understand it, the following argument must be carefully followed. We'll make a distinction between the frequency at which total cancellation occurs and the nearby frequencies.

When two acoustical waves reach the same point with a time difference, there will be a total cancellation at the frequency or frequencies for which that time difference equals half a period or odd multiples of a half period, i.e., a difference of 180° or odd multiples of 180° in phase. At these frequencies, when the difference in phase is 180° or odd multiples of 180° , the resulting level is zero (i.e., no sound at all).

For frequencies above and below the first cancellation the phase difference will be lower than 180° , and hence we'll say that the cancellation (or the sum) is partial or, more correctly, that the sum between them will take place with a phase difference that is progressively lower as we move away from the cancellation frequency, going progressively from a total cancellation to a +6dB sum at the more distant frequencies. The same will happen for the next cancellation frequencies. The following table shows the resulting sum of two waves with the same level but different phase relationships. We can see that, naturally, the more the phase difference, the lower the result of the sum.

Example 1	Example 2	Example 3	Example 4
Magnitude1 = 100dB	Magnitude1 = 100dB	Magnitude1 = 100dB	Magnitude1 = 100dB
Magnitude2 = 100dB	Magnitude2 = 100dB	Magnitude2 = 100dB	Magnitude2 = 100dB
Phase difference = 0°	Phase difference = 45°	Phase difference = 90°	Phase difference = 120°
Sum = 106dB	Sum = 105.3dB	Sum = 103dB	Sum = 100dB

The sum being partial due to phase differences lower than 180° , when raising the level 10 or 15dB using EQ in the cancellation band, the measurement will show the magnitude response recovering. The fact that we see that level recovery for the frequencies with partial cancellation makes sense, specially with low frequency resolution and significant equalisation, such as the 10 or 15dB mentioned earlier. When summing two waves with an effective phase difference lower than 180° , a loss of level is produced which can be recovered by applying equalisation, but at the expense of forcing the transducers.

But why does the measured response seem to show that the level at the frequency of total cancellation is also recovered? The answer is simple: due to the environmental noise in the rooms where we are measuring. When there is a total cancellation the resulting pressure level is zero at the microphone position (i.e., no sound at all). However, the sound emitted by our loudspeaker at that frequency and reflected in the walls that comes back as background noise, results in the software showing that there is some level, which is what gets amplified when applying EQ. There is also the environmental noise in the room, that of the measurement and/or playback devices as well as the smoothing used in the measurement.

In other words, the result of the sum, at the microphone position, of the direct sound and the first reflection is a cancellation. However, the noise level at that same frequency is high due to the fact that equalisation has increased the level of the secondary late reflections, of lower level than the first reflection and which phase depends on the distance difference. Under normal circumstances it is mainly that background noise or reverberation that the measurement software is showing at the cancellation, and the reason for the low coherence level being shown is the low signal-to-noise ratio.

By way of summary, and to avoid wrong conclusions, I'll emphasize these two points:

1.- If a cancellation shows up on a measurement due to a reflection, we need to be aware of the fact that the cancellation at that frequency only occurs at the specific point where the microphone is at, and therefore it should not be equalised because we'll be unnecessarily colouring the response at the rest of positions.

2.- If, despite that, we applied equalisation, even though the software shows recovery of the magnitude response, we are being fooled by the background noise or reverberation, which is what we are really measuring at the cancellation frequency.

If we EQed the cancellation produced by a reflection we would "fix" with electronics a "problem" that occurs only at that specific position and which is not coming from the system but from an incorrect interpretation of the graph and an incorrect evaluation of the environment where the measurement is taking place, creating a problem where there wasn't one. As stated before, the software does not lie, but we have to learn how to interpret it.

Microphone and sound card

A range of affordable measurement microphones is available that can be used for this type of work.

It is advisable that, either the microphone has perfectly flat magnitude response, or else comes with a file for magnitude and phase response compensation. These compensation files, which are normally text files that are created specifically for each individual microphone, can be loaded onto the measurement software so as to counteract the small irregularities in the response, normally a high frequency roll-off.

Since we'll be taking equalisation decisions based on it, it will be good if you can make an effort and not buy a microphone that's too inexpensive. I would also not recommend purchasing a very delicate and/or expensive microphone. And I would recommend that, the moment you take it out of its box, when it is still new (and has not yet taken a fall to the ground), you take a reference measurement with a loudspeaker that you have around and do not use much, in repeatable measurement conditions, and compare the response from time to time so as to detect any deterioration in the response of the microphone.

If you were only to measure phase, i.e., make adjustments to phase, any microphone you may have will suffice. From your typical vocal electrodynamic microphone to a wireless microphone. Perhaps it would be good to start practising with a conventional microphone that you may already have and, over time, when you gain fluency and feel more confident measuring, invest in a measurement microphone with its corresponding compensation file.



As to the sound card, in theory we can use any. For taking these kind of measurements we do not need a card of exceptional quality. We're not recording audio, just comparing the signal that goes into a channel with the signal that goes into the other.

What we will definitely need is one with at least two channels, XLR microphone inputs, phantom power and level controls for input and output. It would also be advisable that it had 1/4" phone line level inputs for electrical measurements. Most current cards come with XLR-1/4" combo connectors.

Configuring inputs and outputs in Smart v8

We'll assume that both the sound card and Smart v8 (in that order) have already been installed on the computer. The first time we run Smart we'll be presented with a menu that takes us to the configuration for inputs and outputs, i.e., the screen you can see in figure 9. If, for whatever reason this does not happen, or we wanted to change the sound card, we would open the drop-down Config menu from the top bar as seen in figure 8 and take it from there.

Config--> I-O Config



Figure 8: Configuration menu for inputs and outputs.

On the I-O Config tab --> Input Devices,

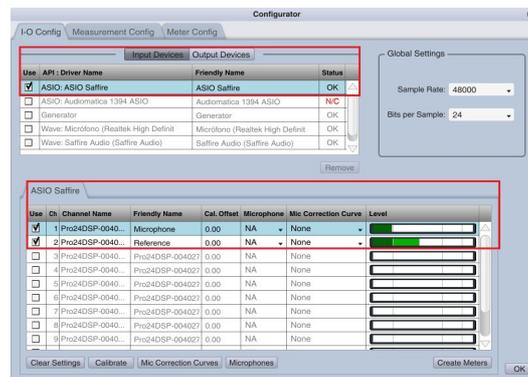


Figure 9: Input selection.

At the top of the page, select the sound card.

At the bottom of the page, select the two input channels and, clicking on "Friendly Name", assign the names "Microphone" and "Reference" to channels 1 and 2 respectively and unselect the rest.

Next, click on Output Devices,

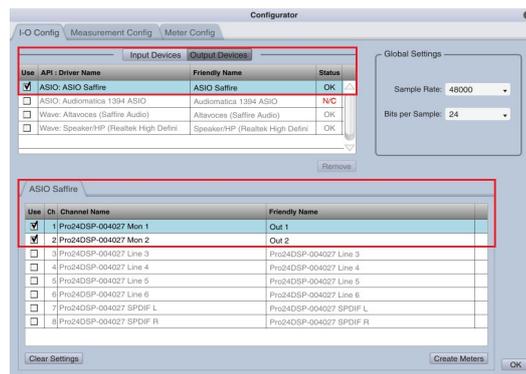


Figure 10: Selecting the outputs.

At the bottom of the page, select the two output channels and, clicking on “Friendly Name”, assign the names “Out1” and “Out2” to channels 1 and 2 respectively and unselect the rest. Now click on the OK button on the bottom right.

For the Signal Generator to come out on both the output channels, we'll need to go to:

Options--> Signal Generator

On “Device”, select the sound card if it is not selected already.

On “Main”, select the channel we named “Out 1”, and, select the channel we named “Out 2” for “Aux”. This way, the generator signal, which will be the *Reference Signal*, will come out of the two channels. Click on OK and continue.

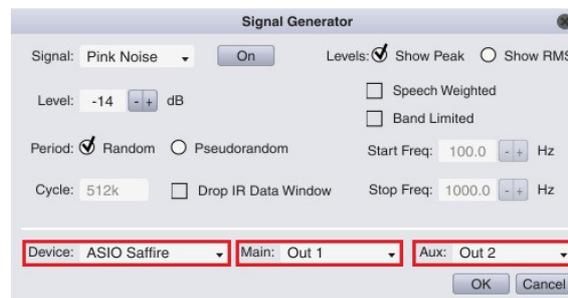


Figure 11: Selection of the card and the two channels through which the generator signal will be output.

First electrical measurement

The connection scheme would be like the one we saw earlier, but replacing the amplifier, speaker and microphone with the electronic device we want to measure. In this example it is a graphic equalizer. We can use the line input if the XLR input gets overloaded.

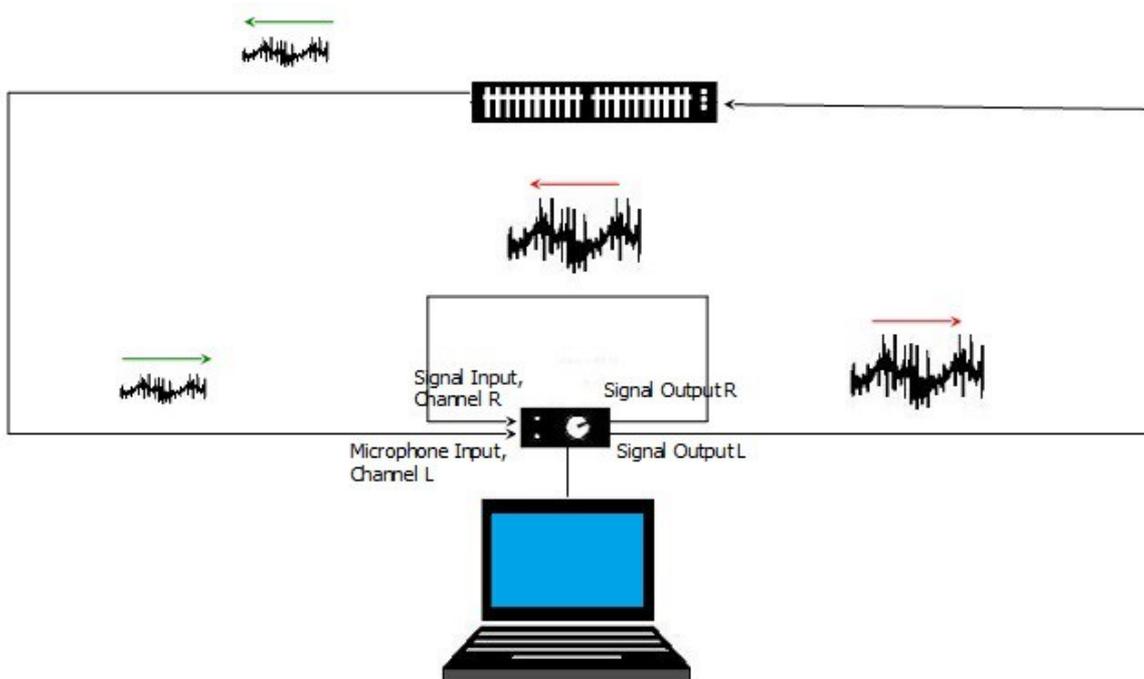
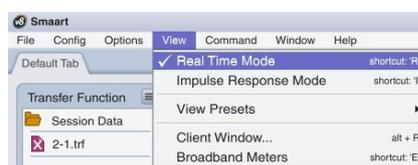


Figure 12: Connection diagram for an electrical measurement.

To perform a Transfer Function measurement, go to:

View--> Real Time Mode. On the bottom right click on "Transfer". If we wanted to see the impulse response on the top part, we can also click on "Live IR".



Figures 13 and 14: Selecting the "Transfer Function" mode.

Next we'll do as follows:

1.- Click on "New TF", on the lower bar, give it a name, click on OK and activate audio by clicking on the arrowhead

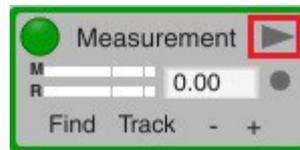


Figure 15: Active audio by clicking on the arrowhead.

2.- Adjust Averaging, Phase Smooth and Mag Smooth as seen in the image:

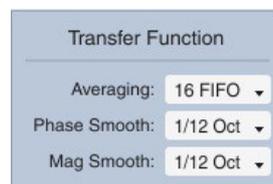


Figure 16: Initial setup of Averaging, Phase Smooth and Mag Smooth.

3.- If it isn't already, select "Pink Noise" and click on "On". By default the initial level in the generator is -48dB. Typically we'll need to increase that so as to have enough signal.



Figure 17: Activating the pink noise generator.

4.- Adjust the levels for the two inputs on the sound card so that they have the same unclipped level. If the microphone input is overloaded, we can use the line level input. The input VUmeters are the ones labelled M and R (which stand for "Measurement" and "Reference") and can be seen in green and blue in figure 18. Yellow input VUmeters will still produce a good measurement. The *Measurement Signal* goes into the left channel (labelled "M") and the *Reference Signal* goes into the right channel (labelled "R").

The signal level should not overload the input of the device being measured. It can be adjusted via the card's output potentiometer and the output level in the Signal Generator.

5.- Click on "Find", wait and click on "Insert".



Figures 18 and 19: Finding the Propagation Delay of the signal.

Caution! Make sure that the grey circle to the right of the delay field is not active (yellow). Otherwise Smart will be continuously adjusting the "Propagation Delay" and we will lose the phase relationship between two measurements. The grey circle turns yellow when delay tracking is active.

If we matched the levels of the *Measurement Signal* and the *Reference Signal* (M and R) well, we'll see a magnitude curve practically flat around 0dB at the bottom and its corresponding phase response around 0° at the top. And, if the propagation delay was right (under normal conditions, it will be) the coherence trace, which appears on the upper half of the magnitude graph, will be 100% across the spectrum.

Now we can, in this case, move the sliders in the graphic equaliser to see its real effect on our signal. We can do the same with mixing consoles, signal processors and so on. I encourage you to measure the electronic devices you work with.

Now that we know how to measure an electronic device, it would be advisable to repeat the measurement, this time connecting the card's output to its inputs, so as to check our measurement system. If both channels of the card have the same performance, we should see two practically flat lines on both the magnitude and phase graphs.

First electro-acoustical measurement

The connection diagram would be like the one we saw earlier in "Connections".

We'll assume that Smart is already configured as described above and that we are in "Transfer Function" mode. We can use any loudspeaker, even a hi-fi speaker that we may have at home.

Next we'll do as follows:

1.- For this first measurement let's place the microphone some 2 metres from the loudspeaker, at a point as distant from reflections as possible.

2.- Once in "Transfer Function" mode, we'll click on the arrowhead to activate the audio, as in the previous case.

3- Adjust Averaging, Phase Smooth and Mag Smooth also as in the previous case.

4.- Select "Pink Noise" and click on the "On" button to start the measurement.

If we want to perform the measurement using music instead, we'll do as follows:

Options--> Signal Generator

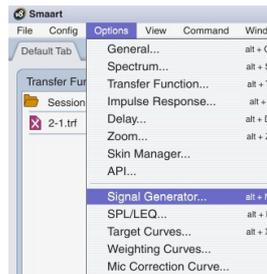


Figure 20: Changing the Signal Generator.

Under "Signal", select "File" from the drop-down menu.

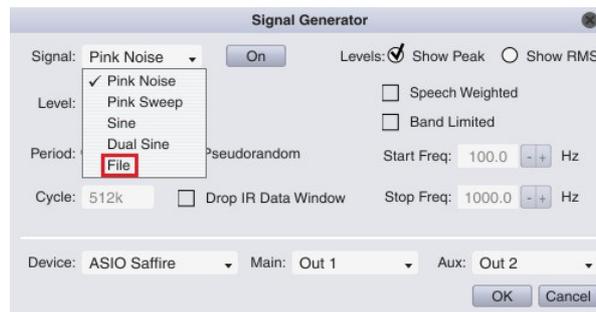


Figure 21: Selecting "File" from the drop-down menu.

Then click on "Browse" and select a WAV format file from the computer. It is not recommended to use music to measure subwoofers.



Figure 22: Selecting the track we want to play from our hard disk.

5.- Adjust the levels for the two input channels on the sound card so that they have the same unclipped level. The input VUmeters are the ones labelled M and R (which stand for "Measurement" and "Reference") and can be seen in green and blue in figure 23. Yellow input VUmeters will still produce a good measurement. The *Measurement Signal* goes into the left channel (labelled "M"), and the *Reference Signal* goes into the right channel (labelled "R").

The output signal levels need to be such that the sound from the loudspeakers is above the background noise in the venue where we are measuring.

6.- Click on "Find", wait and accept if a reasonable delay time is presented for the distance of two meters between the loudspeaker and the microphone. Typically the Propagation Delay proposed will be correct when measuring a mid-high loudspeaker, specially with pink noise.

If we matched the levels of the *Measurement Signal* and the *Reference Signal* (M and R) well, we'll see a magnitude curve centered around the 0dB line at the bottom and its corresponding phase response at the top. If we see the phase response 'disappearing' at the bottom and 'appearing' at the top (or the other way around), making reading of the phase trace more difficult, we can manually fine tune the Propagation Delay here:

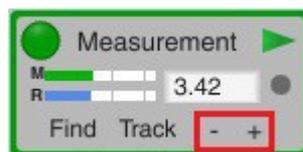


Figure 23: Adjusting the Propagation Delay to avoid truncation on the phase trace.

This way we add or take away delay until an easily interpretable un-truncated phase trace is obtained.

Acoustical measurements, unless performed in an anechoic chamber, will always show bands where coherence is not 100%. As explained earlier, one should not trust the readings at those frequency bands. Once we have finished doing these practice measurements and start doing real life measurements, it's advisable that we measure at several positions so as to give us a better idea of the average system response across the audience area.

If, when running the measurement, the coherence trace disappears at frequencies where coherence is low, this is due to Smaart's default Blanking Threshold for coherence, which hides coherence at frequencies where it falls below 20%. This threshold can be adjusted by dragging the arrowhead on the left; we'll move the arrowhead down if we want to see the trace for those frequencies where coherence is very low:



Figure 24: Modifying the coherence blanking threshold.

Frequency Smoothing

Loudspeaker specification sheets often use 1/3rd-octave smoothing for displaying frequency response curves. With such wide frequency smoothing, important details can be lost. Measurement software such as Smaart allow for a more detailed frequency display. Smoothing can be selected down to 1/48th-octave, or even no smoothing at all. I would start with a smoothing of 1/12th-octave and, depending on what we see or what we intend to measure, increase or decrease that.

This 1/12th-octave smoothing will typically be enough to appreciate reflections or temporal mismatches, but, if we detect a problem, to check how serious it is even less smoothing may be necessary.

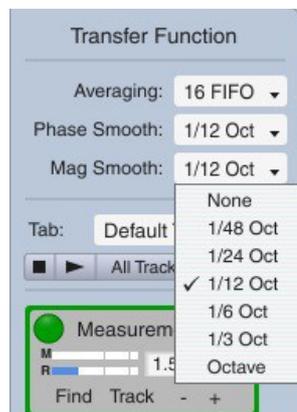


Figure 25: Frequency Smoothing drop-down menu.

Temporal Averaging

If we are looking for a stable graph, especially when using music as a *Reference Signal* or when in a noisy environment, it may be appropriate to use high temporal averaging. In these cases, you may want to start with averaging of 1 to 3 seconds and then decide whether you need more or less.

However, when adjusting EQ or phase we'll need for the graph to respond immediately to the changes being introduced to the system. In this case we'll need shorter temporal averaging. We can start with the 16 FIFO option, which shows the last 16 "frames".

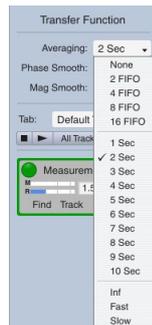


Figure 26: Temporal Averaging drop-down menu.

Feedback Frequency Identification

So as to provide an instant practical value to your measurement systems, even without the need for a measurement microphone, let's go through how to use Smart v8 in a live context to detect feedback frequencies.

If we want to identify feedback frequencies, let's connect the output of the mixing console that we want to monitor to a line level input on our sound card. In our case we'll use input 2 of the card. The connection diagram can be seen in the figure below:

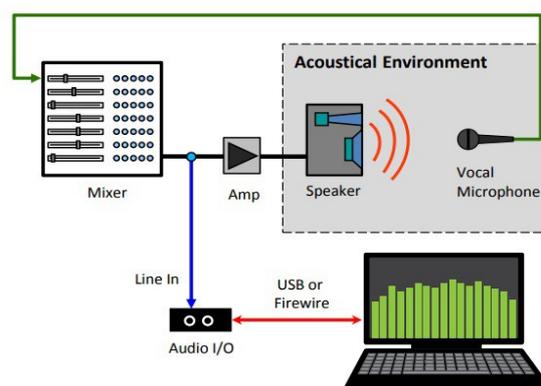


Figure 27: Electrical connection diagram to identify feedback frequencies with Smart v8.

Next, click on "Spectrum" and then on the split screen icon at the bottom centre.

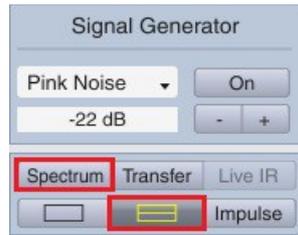


Figure 28: Screen configuration for feedback frequency identification with Smaart v8.

Since what we are analysing is the mixing console's output signal we'll activate input 2 ("Reference") on the card by clicking on the arrowhead.



Figure 29: Activating the signal input channel.

The image below shows the screen where we'll see the effect of an instance of feedback. On the upper right side of both windows "Reference" should be shown. Else, we'll need to click on the arrow to select it. On the upper left side of the top window "RTA" should be shown, while the upper left side of the bottom window should show "Spectrogram". Else, click on the corresponding arrowhead and select the inputs and the type of analysis as I indicated earlier.

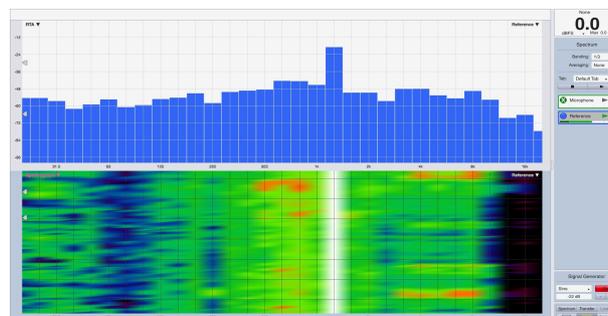


Figure 30: Screen for feedback frequency identification. RTA on top and Spectrogram below.

In our example we've used 1/3rd-octave smoothing and no Temporal Averaging, as you can see on the upper right side. The top part of the graph area shows an RTA, with frequency on the horizontal axis and amplitude on the vertical one. The RTA could be enough if we were watching the screen all the time. Since this can't always be the case we'll use the Spectrogram to help us.

The bottom part of the graph area shows the Spectrogram, with frequency on the horizontal axis and time on the vertical one. This way, even if we're not constantly watching the screen, when feedback takes place we'll be able to see the frequency that lasts over time, making it easier for us to identify the feedback frequency.

In our example you can see the effect of feedback at 1200Hz. On the RTA (upper window) we can see that the 1/3rd-octave band that corresponds to the feedback frequency is protruding. The Spectrogram (lower window) shows a continuous straight line on the vertical axis in the 1250Hz band. Since feedback often lasts over a certain amount of time (which sometimes feels like an eternity), the straight line will indicate the frequency at which feedback is taking place.

In our case, the straight line shows up in white on the spectrogram, but it could show in other colours depending on the feedback level and the dynamic range, which can be selected by the two arrowheads to the left of the spectrogram window. The Min and Max levels are initially set at -66dB and -30dB. If we do not see the feedback frequency on the spectrogram, we'll need to adjust the aforementioned arrowheads as appropriate. Each situation may call for different values. If correctly configured, when feedback occurs we'll be able to identify a straight line that lasts over time in the spectrogram; and in the RTA we should be able to see how one of the bars protrudes from the rest.

I recommend using the "Cue" outputs on mixing consoles to connect to the sound card's input. If it is more practical, there is also the possibility of monitoring the input of our measurement microphone instead of the mixing console outputs. In that case we would need to activate input 1, "Microphone".

Final considerations

These measurement systems offer many possibilities that I have not talked about. This is only the beginning. Smaart v8 has a detailed manual where you will be able to dig deeper and learn many more things.

Considering all of what you need to do measurements (computer, sound card, microphone, measurement software), the company that develops the software is the smallest of them all, and probably the one where you would find the most people who share your interests and the people you would be most comfortable engaging in conversation in a bar. They deserve making a living from their work like everybody else, just like yourself. So, I would like to ask you to use legally acquired software so that they can continue to help us achieve better sound.

Joan La Roda

**Senior Loudspeaker Designer
Work Pro**

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All screen captures and Smaart v8 Guide images in this article courtesy of Rational Acoustics (© Rational Acoustics LLC, 2016)