Single Channel MLS Measurement System

By Bohdan Raczynski, August 2015

Single channel measurement systems have been around for some time now, and are becoming more popular amongst DIY community. Cost and convenience are the most cited factors in opting for such a system as a measurement workhorse for loudspeaker drivers and systems.

However, there is also a technical aspect associated with the operation of single-channel measurement systems. This short paper attempts to examine the performance trade-offs between single-channels and dual-channel measurement systems.

A well respected, PCI sound card – Delta1010LT has been chosen as an example device, employing MLS measurement scheme. This sound card has been around for over 10 years and has become a very popular choice for DIY community. From the card's technical manual we have:

Frequency Response:	22Hz - 22kHz, -0.2, -0.4dB @ 48kHz; 22Hz - 40kHz, -0.2, -0.7dB @ 96kHz

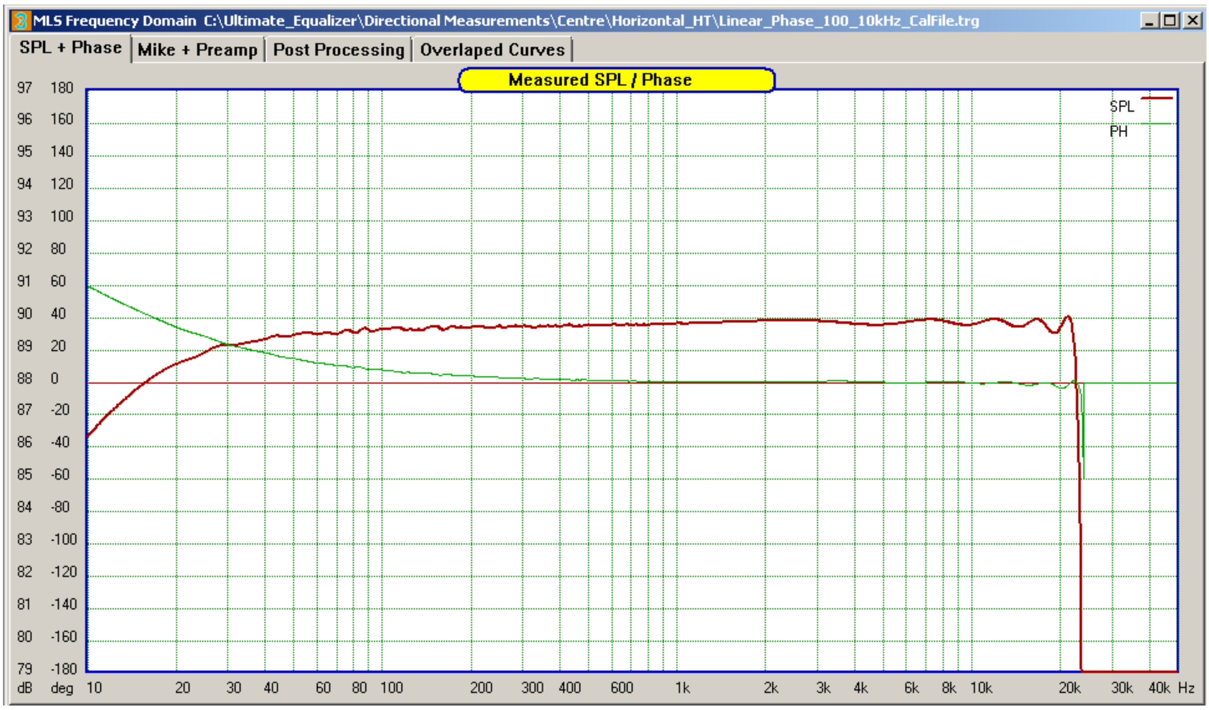
The above frequency response figures are quite respectable for a playback device. However, we must remember, that loudspeaker measurements engage the sound cards as a signal generator and also as a recording device at the same time. Therefore the already non-flat frequency response of the complete measurement system will be worse than the figures quoted above. By how much?.

Fortunately, we can quickly measure the combined frequency response of the output DACs + filter and input ADCs + filter. This is performed by looping the MLS output signal directly back into the soundcard input.

Before the test results are examined, we need to consider the level of performance we expect from the measurement system. Because it is a measurement system – nothing less. Typical loudspeaker design duties would require being able to resolve design issues manifesting themselves in fractions of decibels or 5-10degrees in phase response. Often, designers employ software optimizers, which are driven by accurate algorithms, and this type of activity heavily relies on accurate measurements. Moreover, contemporary subwoofer design often requires decibel accuracy below 20Hz.

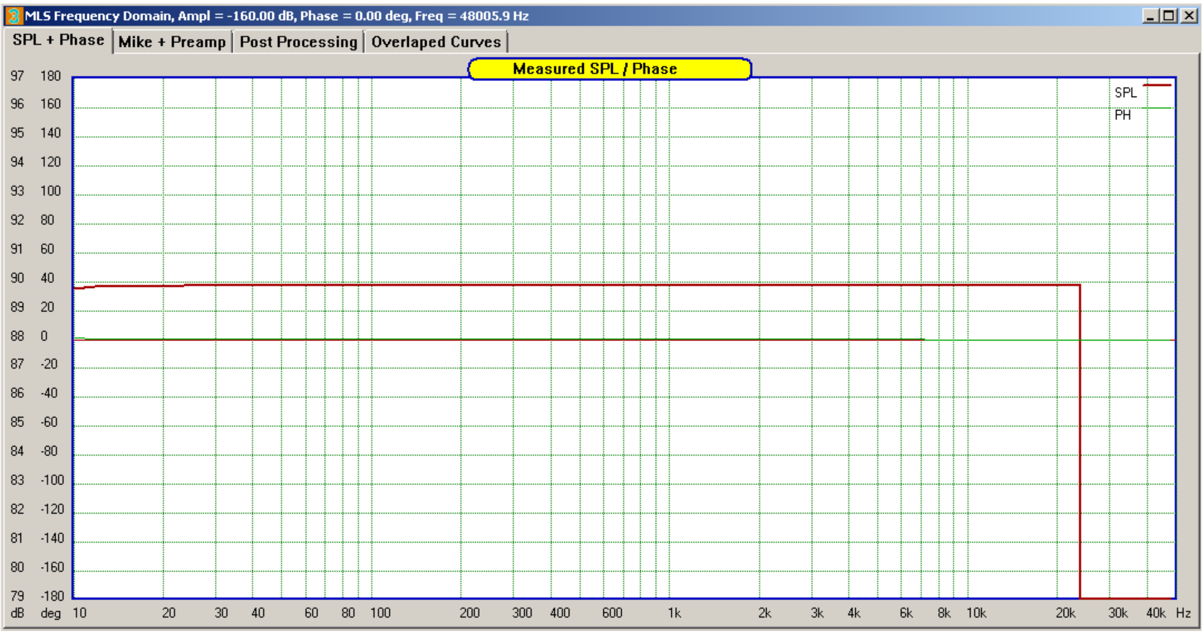
The point here is this: the measurement system must be as accurate as today's technology allows.

Now, we can examine the test results. The output frequency response is presented in vertical scale of 1dB/division. The FFT used to recover the SPL and Phase from impulse response has frequency resolution of 48000/262144 = 0.183Hz. We start with 48kHz sampling frequency data.



Single-channel frequency response measurement with output looped back to the input.

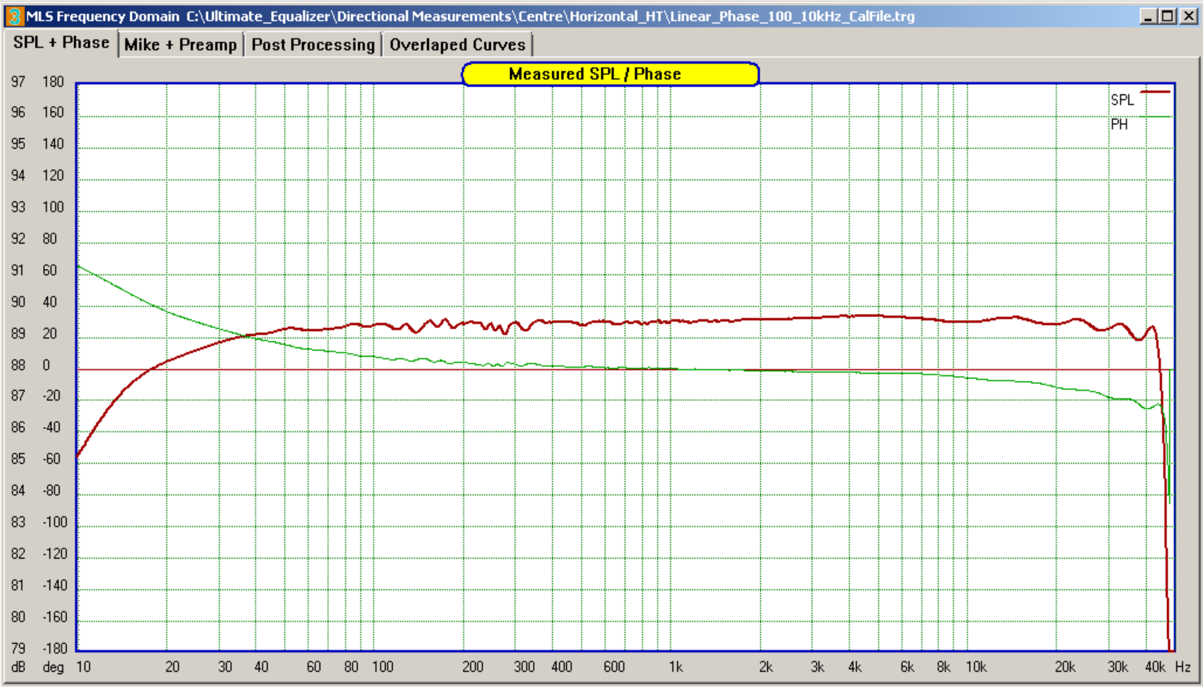
It is immediately observable, that frequency response is a non-flat one. Some of the small wiggles in the vicinity of 100Hz can be attributed to digital artefacts associated with MLS signal generation. However, the combined low-frequency drop is almost -3dB at 10Hz. This has a serious impact at the reliability of subwoofer design. Phase response is 60deg off target at 10Hz as well. Secondly, the wiggles from 3kHz-21kHz are possibly due to filtering in the sound card and can not be eliminated.



Dual-channel frequency response measurement with output looped back to the input.

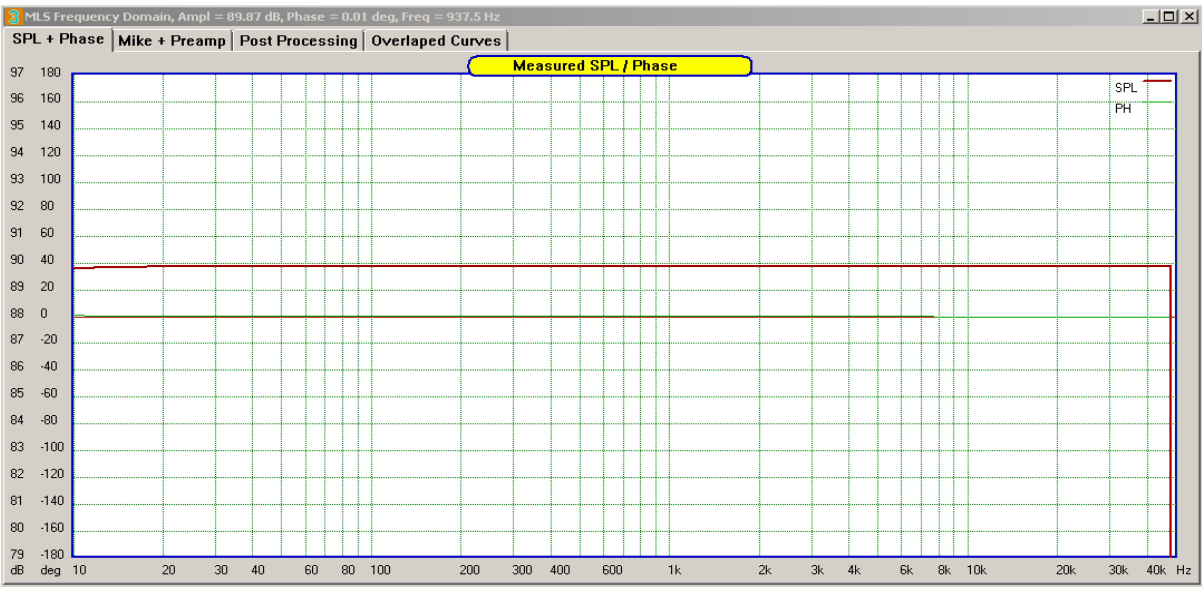
For dual-channel measurements we have essentially flat lines for SPL and phase. And this is pretty much what we expect from the measurement system – to be invisible.

For 96kHz sampling frequency The test returned the following SPL/Phase:



Single-channel frequency response measurement with output looped back to the input.

The results are similar to the 48kHz measurements, but marginally worse. For instance, the SPL level is -4dB down at 10Hz.



Dual-channel frequency response measurement with output looped back to the input.

Again, for dual-channel measurements we have essentially flat lines for SPL and phase. As before, this is pretty much what we expect from the measurement system – to be invisible.

The second aspect of single-channel measurement systems is lack of time reference for phase measurements. This deficiency is best explained by measuring phase response of a loudspeaker driver.

Dome tweeter example

We will now examine minimum-phase response of a popular Hi-Fi dome tweeter driver.

The dual-channel MLS system is actually designed to provide minimum-phase response of the measured driver, within the error of **+/- one sample time**. Here is how it works.

When you perform loop test, you will notice, that you will get flat phase response of the signal channel when you place the start of the FFT window at 10 samples before the peak of the impulse response – why?. This is because the reference channel is also automatically windowed with the fixed start of the FFT window also at 10 samples in front of the IR start. The loop test simply measured the true "minimum-phase" phase response of the sound card. However, each PC MLS system must be examined individually for the Reference Impulse response first.

Reference Impulse Response:

MLS Impulse Response Ref=41463.95, In=-95.36, Bin=60, Scroll[0 - 2000] Peak of 41463 at bin=60
MLS Impulse Response Ref=8584.96, In=1.29, Bin=59, Scroll[0 - 2000] Peak -1 of 8584 at bin 59.
MLS Impulse Response Ref=-3762.23, In=-1.57, Bin=58, Scroll[0 - 2000] Peak - 2 of -3762 at bin 58, the IR has gone large negative now. **For this MLS system, bin 59 is the start of the impulse response.**

Therefore, the start of the FFT window for the Reference impulse response is 10 sample times from the peak, or **9 sample times from the start of the impulse response.**

Now, we can apply the same technique to the loudspeaker measurement, and place the start of the FFT window 9 samples ahead of the start of the IR – and we'll obtain minimum-phase phase response of the loudspeaker straight away, with +/- one sample time error. To eliminate this small uncertainty error, we have to add/subtract small delay (or manipulate HBT slopes) to get the measured and HBT calculated phase into alignment.

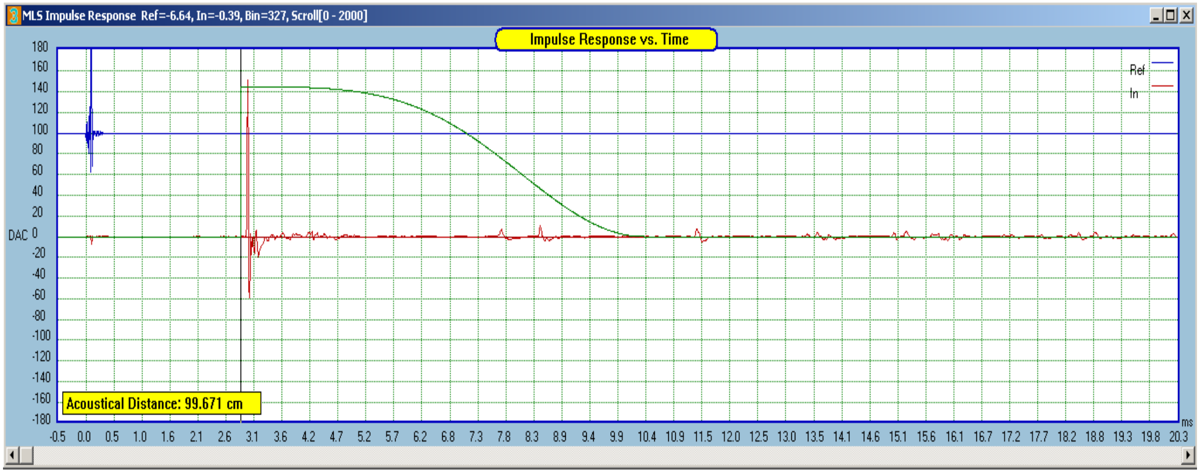
It is important to determine the **start of the impulse response** (not the peak), as various drivers have different rising slopes of the impulse response, therefore the peak will be located at various distances from the start of the impulse. So, first we need to find the peak of the impulse response:

MLS Impulse Response Ref=-7.84, In=1886.90, Bin=339, Scroll[0 - 2000] the peak is **In = 1886.90, located at Bin=339.** Next we move the cursor to the left of the impulse response, one sample time, and each time we monitor the "In" vale.

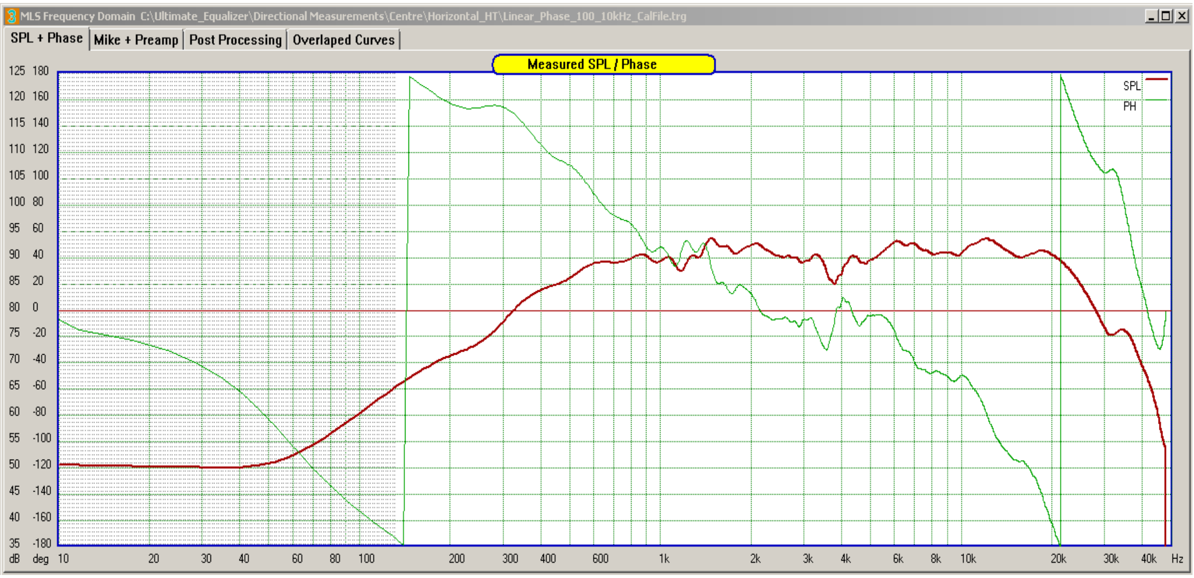
MLS Impulse Response Ref=-7.04, In=987.70, Bin=338, Scroll[0 - 2000]
MLS Impulse Response Ref=-14.45, In=147.21, Bin=337, Scroll[0 - 2000]
MLS Impulse Response Ref=-9.20, In=4.94, Bin=336, Scroll[0 - 2000] the In=4.94 and is very close to zero, therefore, we determine that the start of the impulse response is Bin=336.

Finally, we need to move the start of the FFT window 9 sample times (for this system it is 9 sample times) to the left from the start of the impulse response, 336-9 = 327.

Now, the start of our FFT window is located at Bin = 327. See figure below.



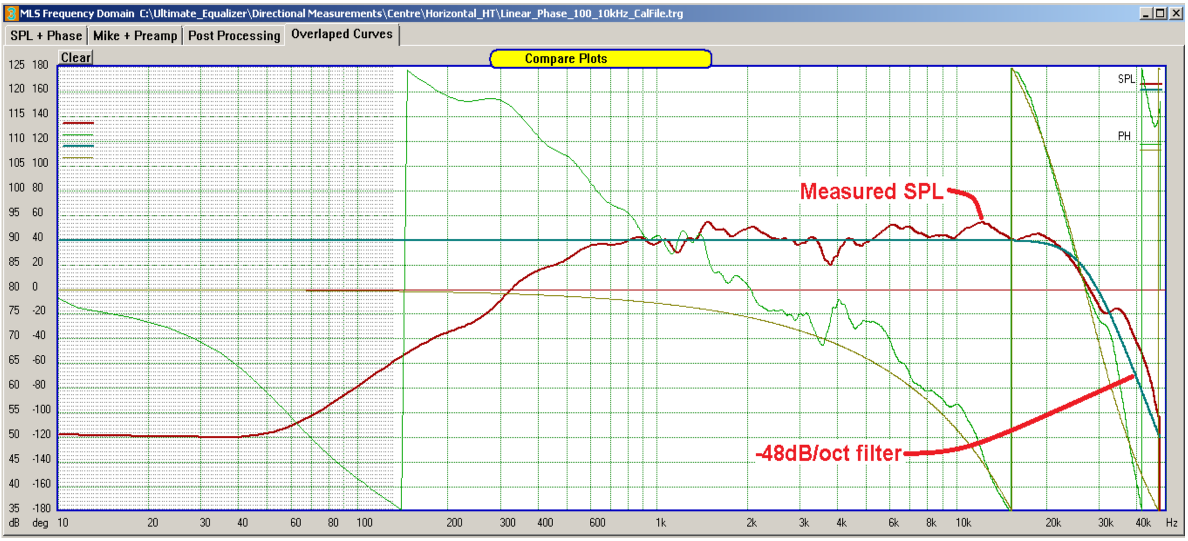
Next, we need to obtain the SPL and phase of the driver using FFT.
And here is the result.



To increase the level of confidence on the phase response, it is also suggested to use a band-pass filter, comparable with the loudspeaker amplitude response. This will give us filter's phase response, which we would use as an additional guidance for the locations of the 360deg transitions of the filter and the measured phase – they should be very close.

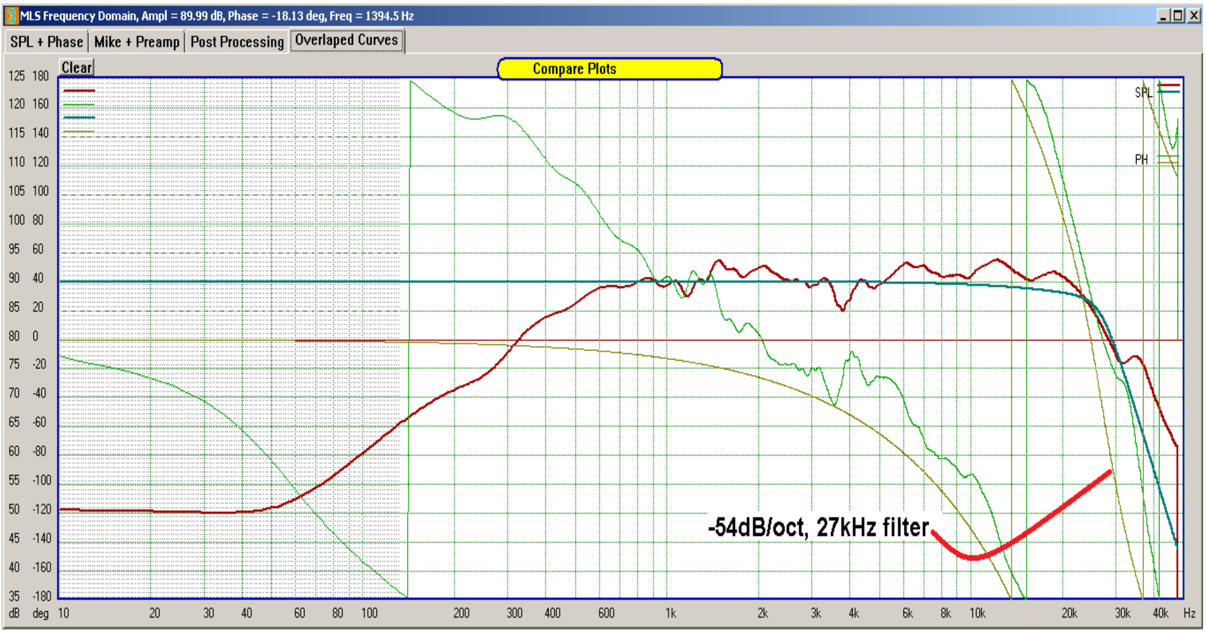
Since this is a tweeter example, and we are only interested in finding the high-frequency tail, I use simple low-pass filter located at 27kHz with -48dB/oct slope, as there needs to be a close match with the measured SPL. It is observable, that the slope of the filter is slightly slower than the measured response, so we assume **-51dB/oct as the asymptotic slope** of the measured driver SPL. Also, the 360deg transitions of both: filter and the measured SPL are very close.

In order the get the measured phase response phase response 360deg transition to overlap filter's phase transition, a small, **12usec delay was added to the measured SPL curve**.



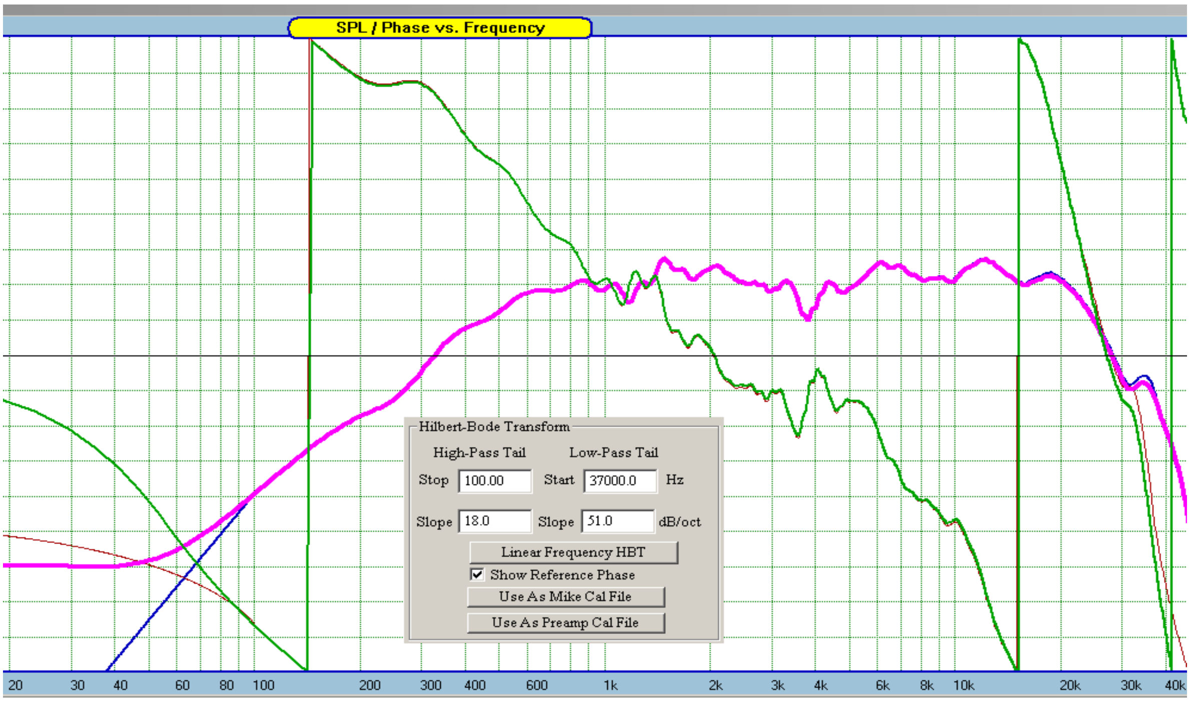
We can now run HBT with the high side asymptotic slope of 51dB/oct, to see how the whole picture works out.

Just for the record, using -54dB/oct (48dB + 6dB) filter also located at 27kHz appears to roll-off little too fast and produces transitions too early – see below.



I do not have a -3dB/oct filter in the available selection, therefore, the choice was to use asymptotic slopes of -51dB/oct.

Phase response of the -54dB/oct filter is -65deg at 5kHz, and phase response of the -48dB/oct filter is -55deg, therefore, the phase response of the -51dB/oct filter is located right in the middle of the two and is equal to -60deg at 5kHz. We have now achieved very good phase accuracy at 5kHz. The maximum error is 5deg/180deg = 2.7%.



We observe a perfect alignment of measured and HBT-derived phase responses assuming -51dB/oct asymptotic slope of the “guiding filter”.

HBT SPL – blue curve
HBT Phase – red curve

There is one other bonus of the MLS measured phase – it’s quite accurate at low frequencies. Measurements indicate, that dual-channel MLS system will give you minimum-phase (+/-1.5deg error) below 200Hz straight away by placing the FFT window as described above. Let’s assume, that we have a typical 3-way system with crossover frequencies at 500Hz and 5kHz. Uncorrected phase error will be increasing with frequency, so how much phase error is equated to +/- one sample time at 5kHz?.

48kHz sampling: +/-38deg, too high, needs guiding filter and HBT correction described in this paper.
96kHz sampling : +/-18deg, slightly too high, needs guiding filter and HBT correction described in this paper.
192kHz sampling: +/-9deg, good enough for first-cut design
384kHzsampling: +/-4.5deg, good enough for first-cut design

Conclusion

Two aspects of single-channel measurement systems were examined.

1. Combined frequency response of the measurement channel.

It is highly recommended for the single-channel measurement systems to have self-calibrating function. The non-flat frequency response of DAC/ADC + filters should be accounted for and provision needs to be made for calibrating these irregularities out, as the uncorrected responses are too deficient. Microphone calibration file (often provided by the manufacturers) should not be confused with calibrating the electronics involved in the measurement process. Such a file does not contain any calibration data for the electronics involved in playback side of the measurement. It is also doubtful, if the microphone calibration file contains precise data to calibrate out the electronics of the recording channel – this is because this calibration data is dependant on the measurement signal employed.

Dual-channel measurement systems are inherently self-calibrating, and calibrate out the irregularities in amplitude and phase for playback and recording automatically – as shown on the measurement results.

Single-channel measurement results presented above were obtained with an MLS measurement techniques. There are other technologies available, but in each case, it is essential to know (rather than speculate upon) the deficiencies in the measurement channel frequency response.

2. Time reference for phase measurements.

Here, it's difficult to offer recommendations for single-channels measurement systems. They just do not measure the absolute (minimum-phase) phase response.

On the other hand, dual-channel measurement systems are designed to measure phase response within +/- one sample time. The accuracy can be further improved by using the methods described above.