A Meaningful Loudspeaker Phase Response

Understanding Different Views and Thoughts of Phase Response Curves

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Introduction

In the current digital world, audio analyzers with FFT or TDS functions are easily able to show the magnitude and phase response of a loudspeaker. These software analyzers are not expensive and are widely used in live sound, installation and loudspeaker development. Several popular software analyzers are ARTA, Smaart, Systune and EASERA. And there are many others.

Phase response is often questioned. Many practitioners use the software functions such as: delay finder, auto delay finder or phase compensator to check phase response. However, most do not understand the basic principle behind the function. This article will discuss how to set a correct time reference to view phase response and assumes readers have experience in loudspeaker measurement.

Impulse Response and the Digital World

First, let's discuss an impulse response graph.

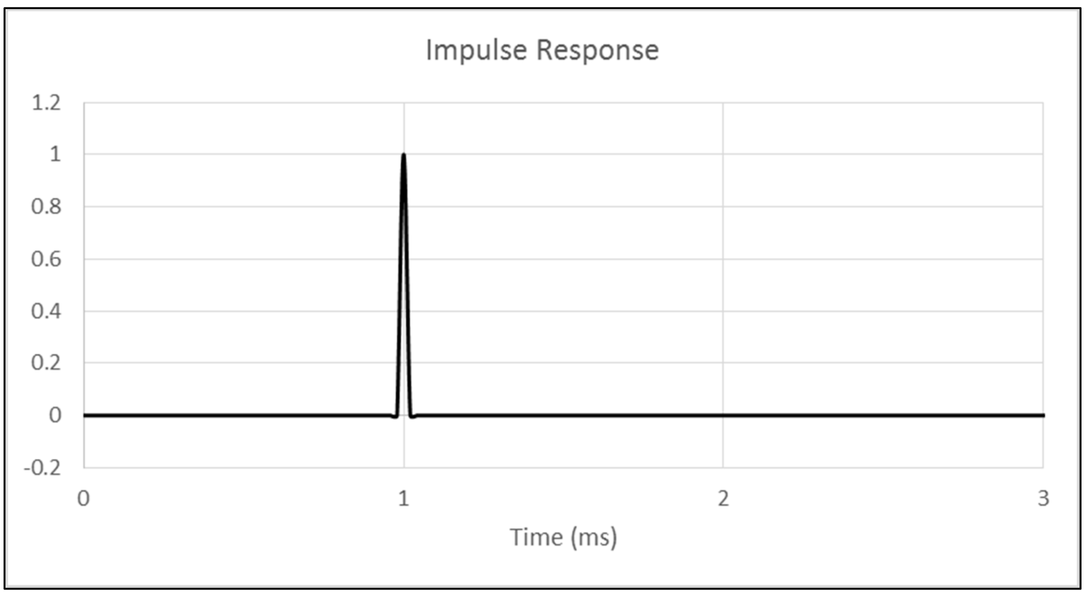

Figure 1

Figure 1 shows a perfect impulse response (Dirac pulse) with the peak at 1ms. This is a time domain graph showing that:

- ✓ An impulse of energy is detected at time = 1ms.
- ✓ The propagation delay of the signal is 1ms.
- ✓ The Y-axis shows a value of one, which shows the signal amplitude is +1.

If we zoom in a little bit, we can see in figure 2 that the impulse response starts before 1ms. The question is does this mean the exact arrival is just slightly before 1ms?

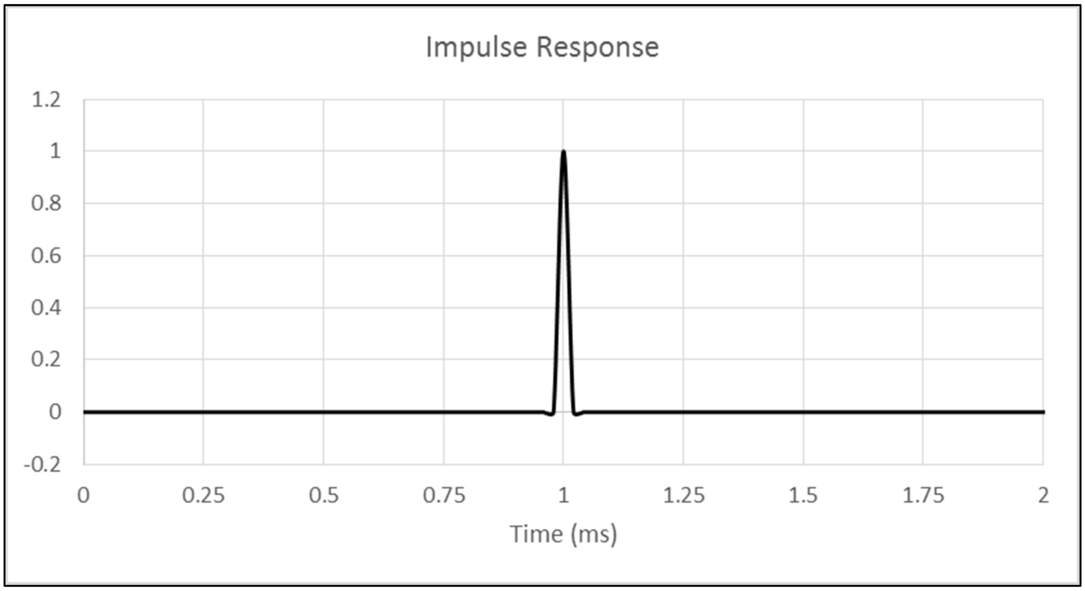

Figure 2

In visualizing a graph, a line is graphed to connect the dots. This helps us to see what is being graphed more easily. Let us see how figure 2 looks like when we only show the X and Y values with a dotted graph.

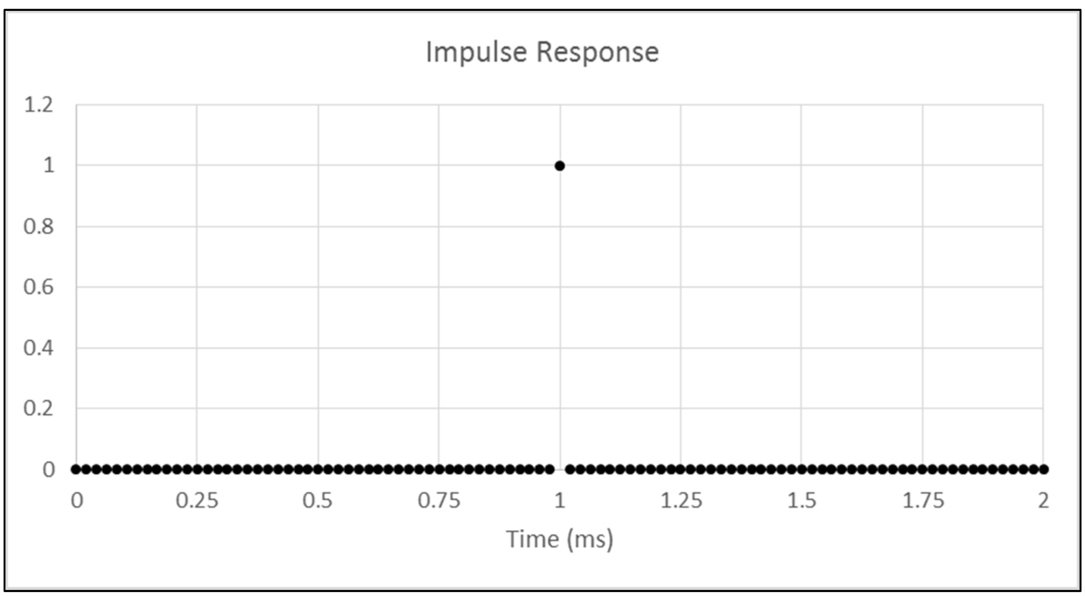

Figure 3

In figure 3, it is clear that the impulse only contains one energy spike that arrives exactly at 1ms. Understanding where an impulse response starts is the first step in reading the phase response. Not all analyzers can show a dotted graph, but it is important that we understand where the line/curve comes from in digital equipment.

Note: in digital, all values are discrete (not continuous) and the resolution of the time plot is determined by the sample rate. A Dirac pulse contains only one sample.

Propagation Delay and Phase Response

Why do we need to worry about the starting point of the impulse? Let us examine the effect of propagation delay on a phase response.

Propagation delay can result from different things such as:

1. Time of flight
   This is the time required for the direct sound to arrive to the microphone. Sound travels in the air at approx. 344 m/s (@20°C). It is important to know that sound requires time to travel in the air.
2. Processor's latency
   Digital processing or digital to analog or analog to digital conversions often requires processing time. A typical latency for a digital loudspeaker management system from A/D to D/A conversion is approx. 2ms. With heavy processing, a digital processor can require longer processing time, resulting in additional propagation delay.
3. FIR (finite impulse response) filters
   Using FIR linear phase filters may result in additional propagation delay. The filter's processing delay can take as small as 1ms up to >500ms depending on the applications.

In figure 4 below, let's assume a 'perfect' loudspeaker is measured using a dual-FFT and the blue colored impulse is obtained. A 0.5ms propagation delay is observed (the 'perfect' microphone is assumed at approx. 17cm away). The black colored impulse shows that propagation delay is removed correctly so the peak of the impulse is at 0ms. This is done by cyclic shifting of the impulse response (this will be explained below). The red colored impulse shows the location if the propagation delay is removed 0.5ms too much. The red colored impulse is an acausal response where the output is not caused by the input. An impulse before 0ms can also be thought that the output happens before the input is fed to the device.

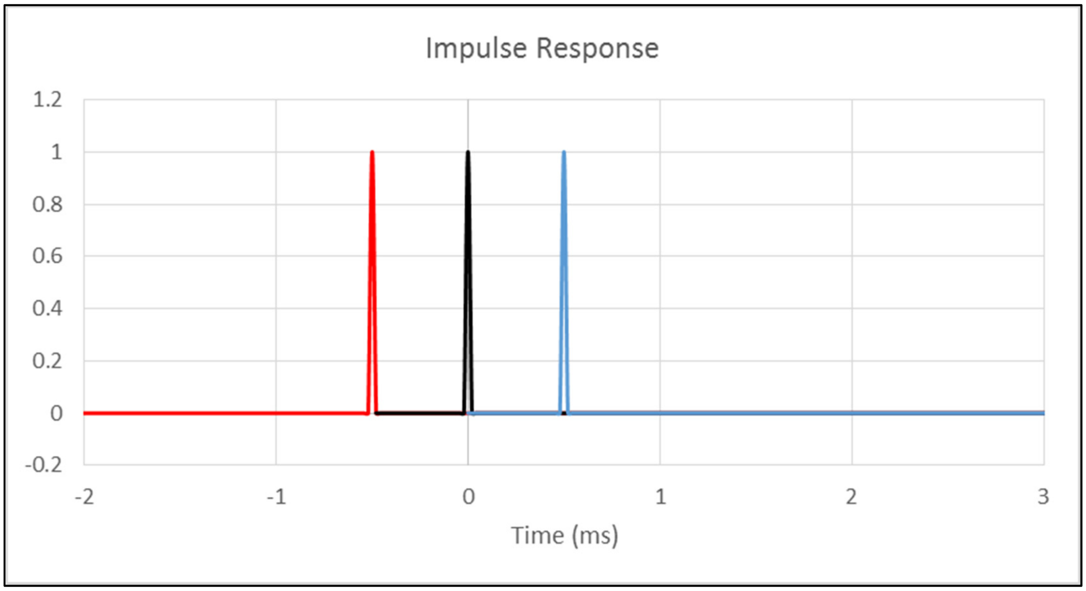

Figure 4

Figure 5 shows the phase response when the propagation delay is removed correctly, where the peak of the impulse is at 0ms. Please observe that the phase response is flat at 0deg and group delay is also at 0deg.

Group delay is the negative rate of change of phase with respect to frequency. The group delay equation can be seen below:

$\tau_g(\omega) = -\frac{d\varphi}{d\omega}$; Where:

ω (*omega*) is angular frequency or 2π times frequency.
φ (*phi*) is phase.
τ (*tau*) is group delay.

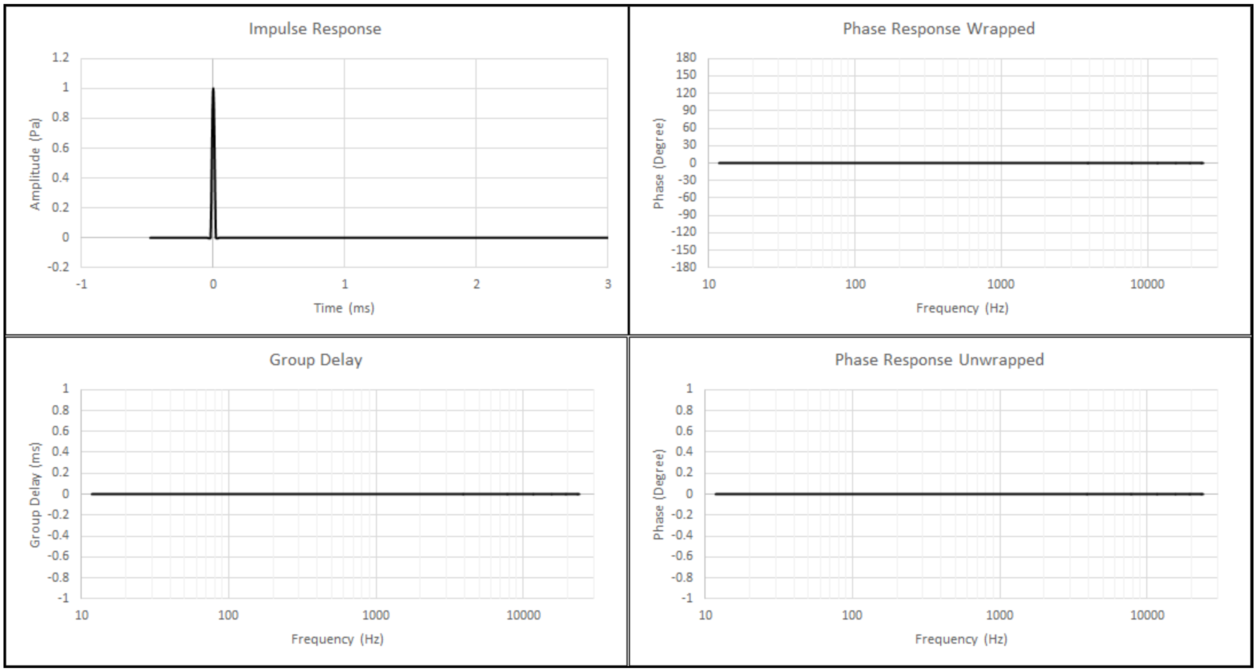

Figure 5

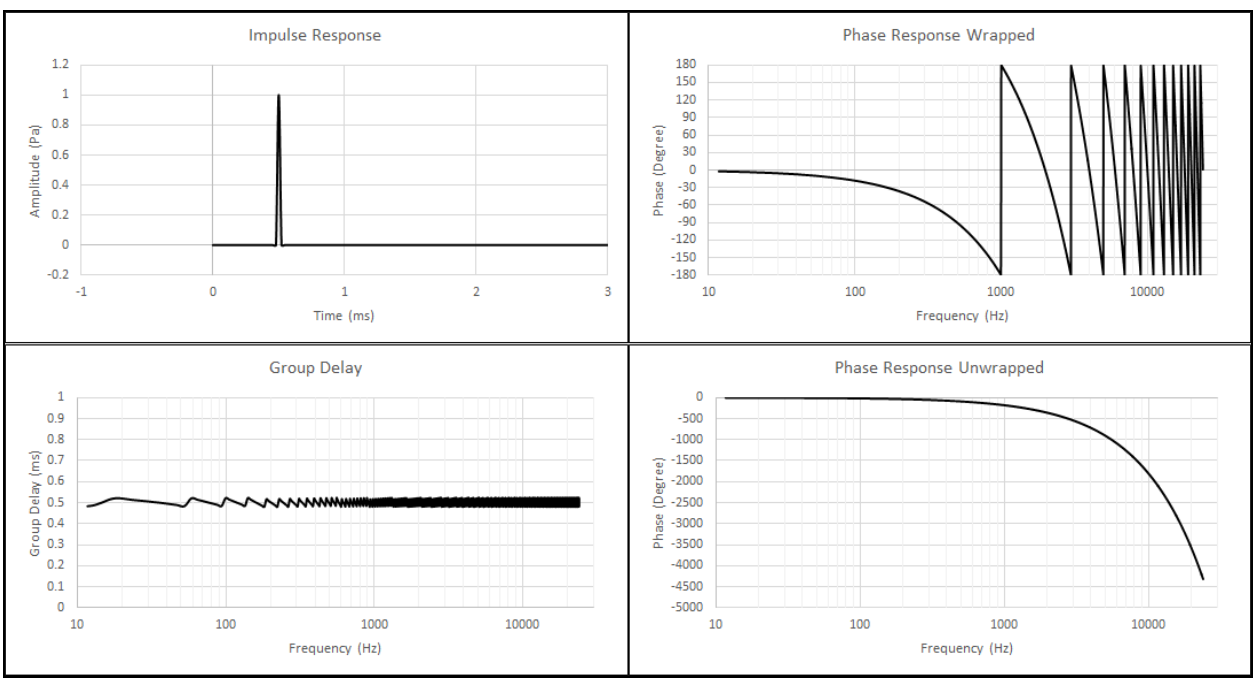

Figure 6

Moving on to figure 6, please observe the phase response when the 0.5ms propagation delay is not removed. To see the phase response clearly, figure 7 shows the zoomed version of the unwrapped phase response.

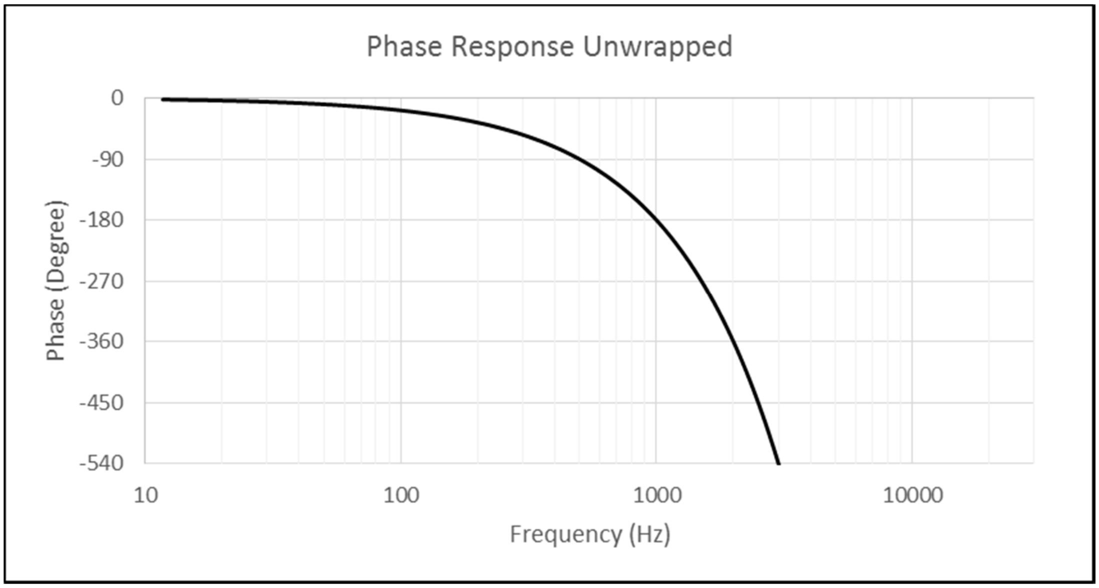

Figure 7

Using the following equation:

$1 = T \times f$; Where:

T is period (second).
f is frequency (Hz).

We can calculate that 0.5ms is a period of 2000Hz. The period of a sound wave is the time it takes to complete one cycle (360°). Therefore, we can see that the phase is -360° at 2000Hz. For a frequency of 1000Hz, it requires 1ms to complete a cycle, thus the phase is shown at -180°. In other words, 1000Hz lags by a half cycle and 2000Hz lags by a cycle in the presence of 0.5ms propagation delay. Without removing the propagation delay, all frequency components of the input signal are shifted in time by the same constant amount. This is not a phase distortion.

If we change the X-axis (frequency axis) of figure 7 to a linear scale, we can see a straight line. Please observe figure 8.

Note: The negative value in phase shows a condition where the phase lags at that frequency.

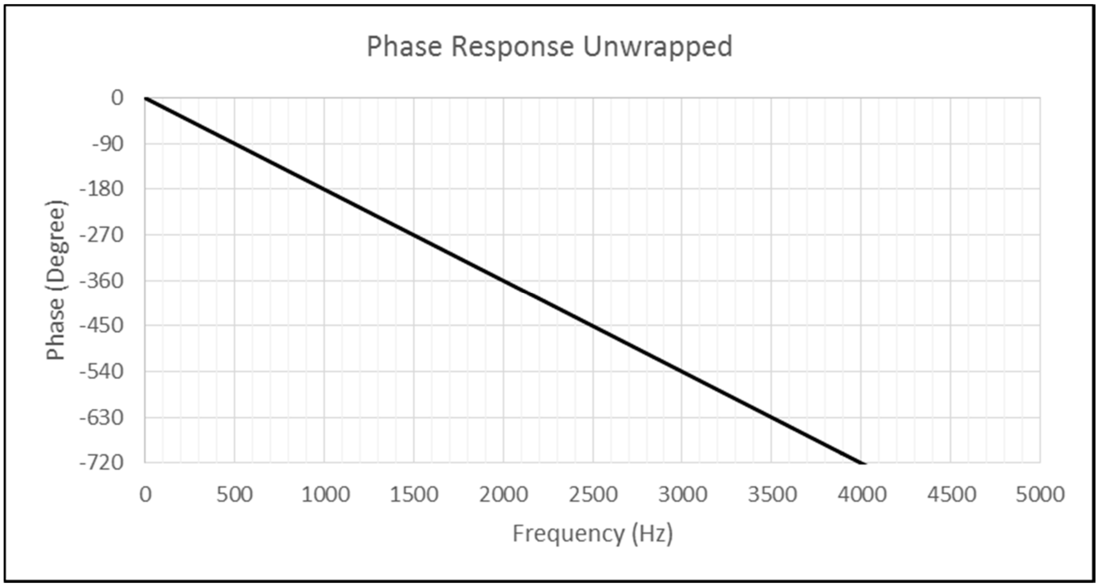

Figure 8

Another observation from figure 6 is the group delay. The group delay curve is at 0.5ms, corresponding to the 0.5ms propagation delay.

Let us observe what happens when the impulse is shifted 0.5ms too much. Please compare figure 6 and figure 9. Figure 10 is an overlay of phase responses from figure 6 (with 0.5ms propagation delay – blue curve) and figure 9 (removing 0.5ms too much of the propagation delay – red curve). The reader can observe the different direction of the phase response. Removing too much of the propagation delay will result in positive phase and negative group delay values.

It is important to understand that a small amount of propagation delay can drastically affect the phase response curve at high frequencies.

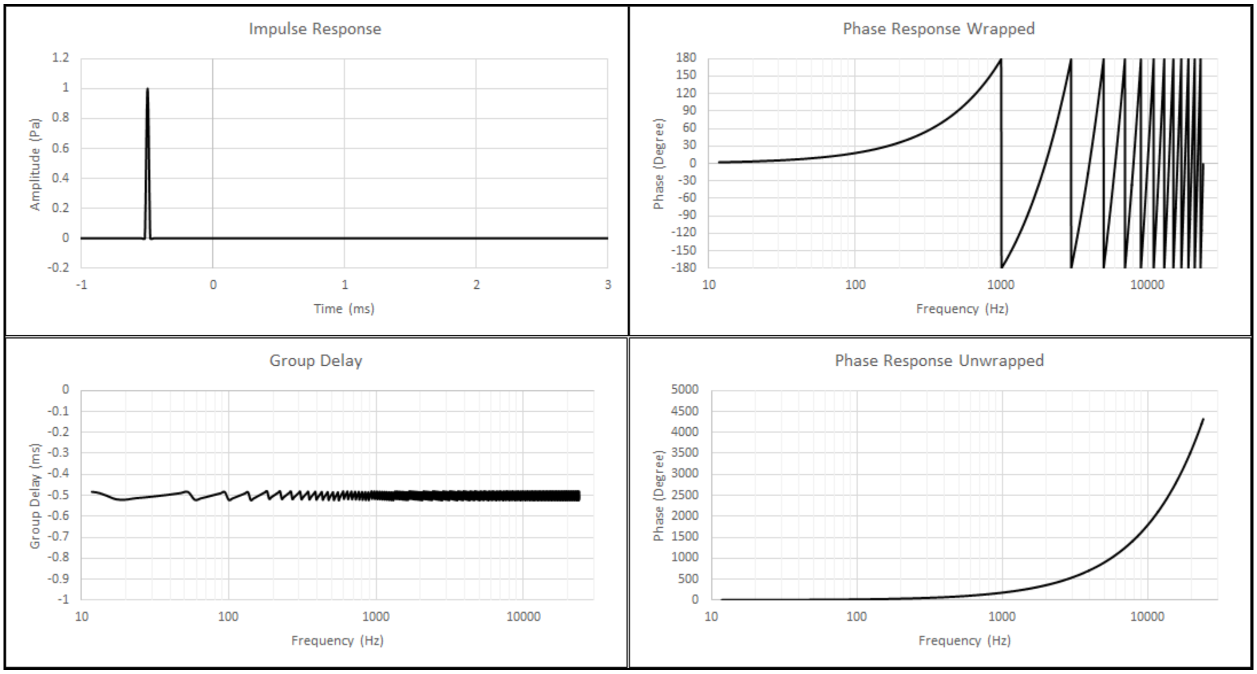

Figure 9

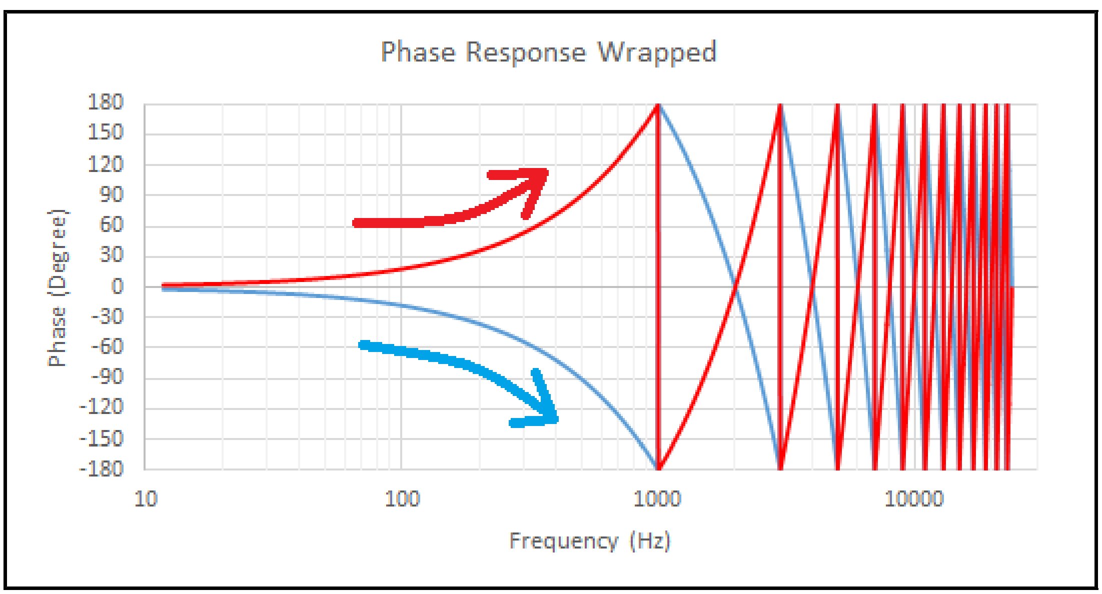

Figure 10

Removing Propagation Delay

Since propagation delay results in a linear phase shift, this can be easily predicted or calculated. A typical way of removing the propagation delay is cyclic shifting so the peak of the impulse is aligned at 0ms. It is commonly found in measurement software as auto delay finder, auto peak finder, normalize max to zero or others. This can be misleading for phase response reading if the frequency content above 1000Hz is lacking in intensity such as if influenced by a low pass filter.

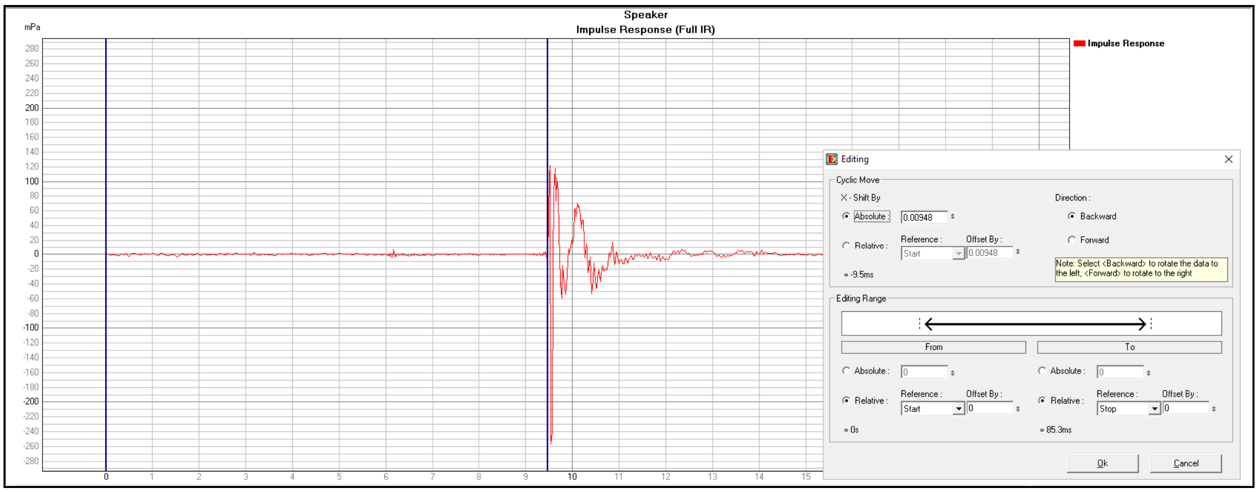

Figure 11

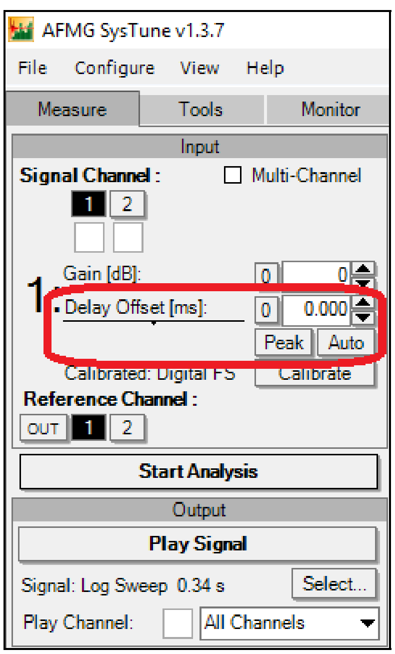

Figure 12

Figure 11 is a screenshot from EASERA where the user can remove the propagation delay manually by typing the delay amount under edit functions under cyclic move sub-menu.

Figure 12 is a screenshot from Systune where the user can remove the propagation delay manually by inputting the delay offset amount or using the peak button to cyclic move the impulse peak to 0ms.

Figure 13 is a screenshot from ARTA where the user can define the starting point of the impulse by placing the yellow cursor, thus removing the propagation delay.

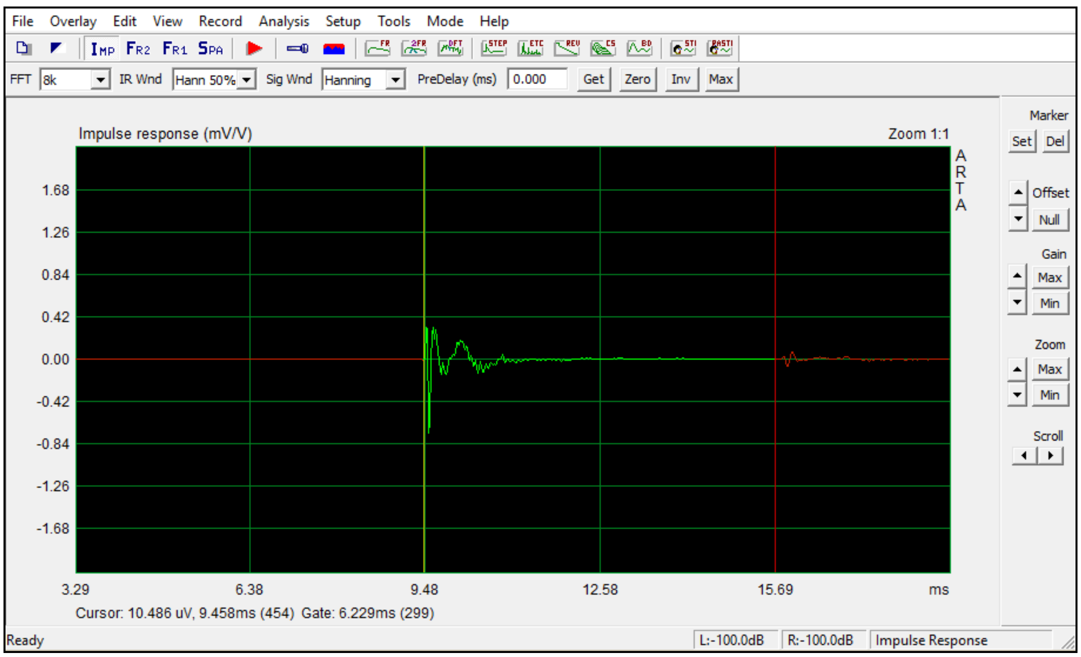

Figure 13

Different programs implement propagation delay removal differently. Several have a function to directly align the impulse' peak to 0ms, but others may require a manual placement of the cursor/marker. The placement of the cursor/marker can indicate a left rectangular window or a cut off.

One thing to remember is that the impulse response of the loudspeaker starts when the direct sound arrives to the microphone. It is important to understand that the peak of the impulse response does not always correspond to the initial energy arrival. Using the peak of the impulse response as reference point may result in a misleading or not meaningful phase response calculation.

Minimum Phase at Glance

Minimum phase is a condition where magnitude and phase responses have a predictable relationship. A change in magnitude response is followed by a change in phase response, corresponding to the Hilbert transform calculation. Most loudspeaker systems are not minimum phase system, but most transducers are. Another example of a minimum phase system are the infinite impulse response (IIR) filters with the exception of the all-pass filters.

The visual minimum phase relationship between magnitude and phase responses can be seen in figure 14.

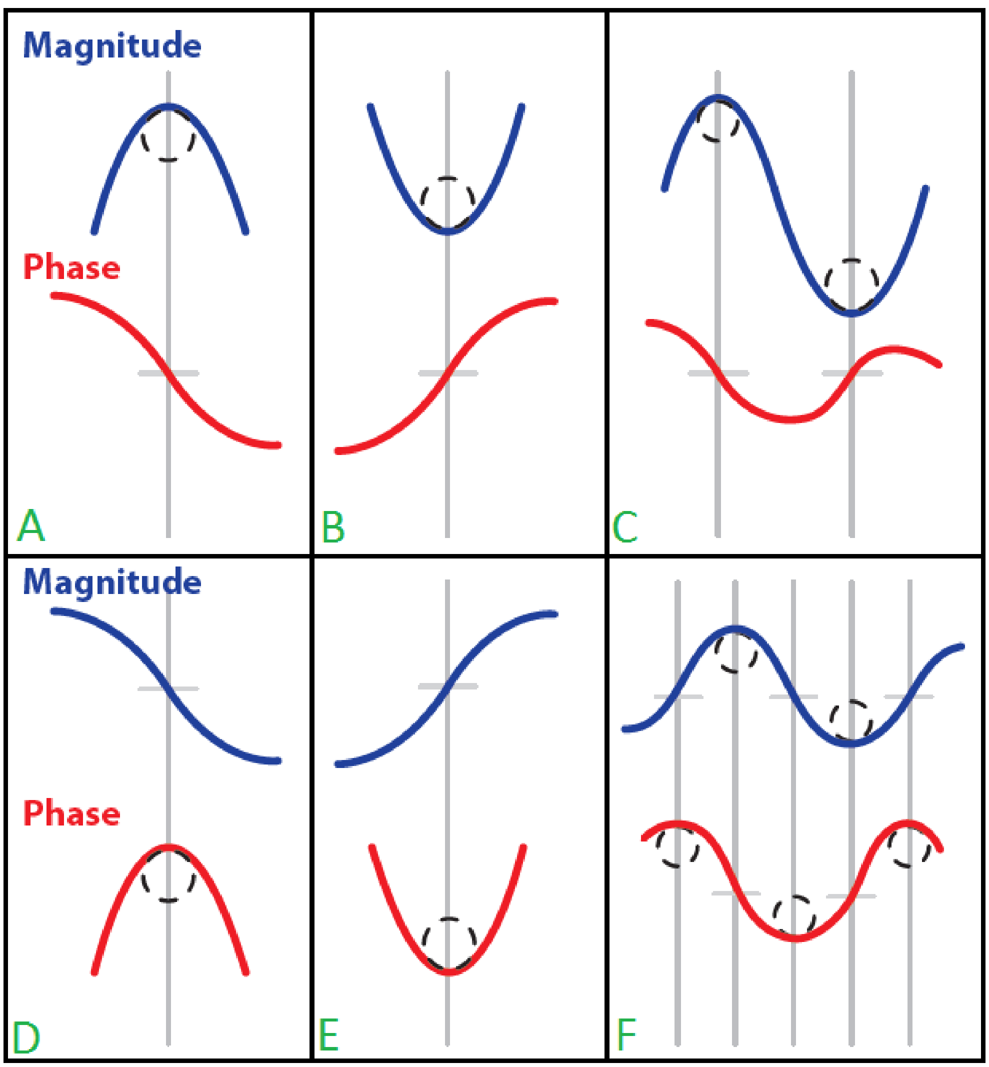

Figure 14

A positive value in phase shows a condition where the phase leads at that frequency. In a minimum phase system, phase can have positive values if the frequency response is rising up (from low to high frequency) such as a high pass condition, which usually occurs at the lower frequency response of a transducer. Another important point is a low pass condition where the frequency response is rising down. In a minimum phase system, this will result in minus phase values (lagging).

Let us observe a modeled one-way loudspeaker response with IIR filters, a high pass at 60Hz and a low pass at 12500Hz in figure 15 below.

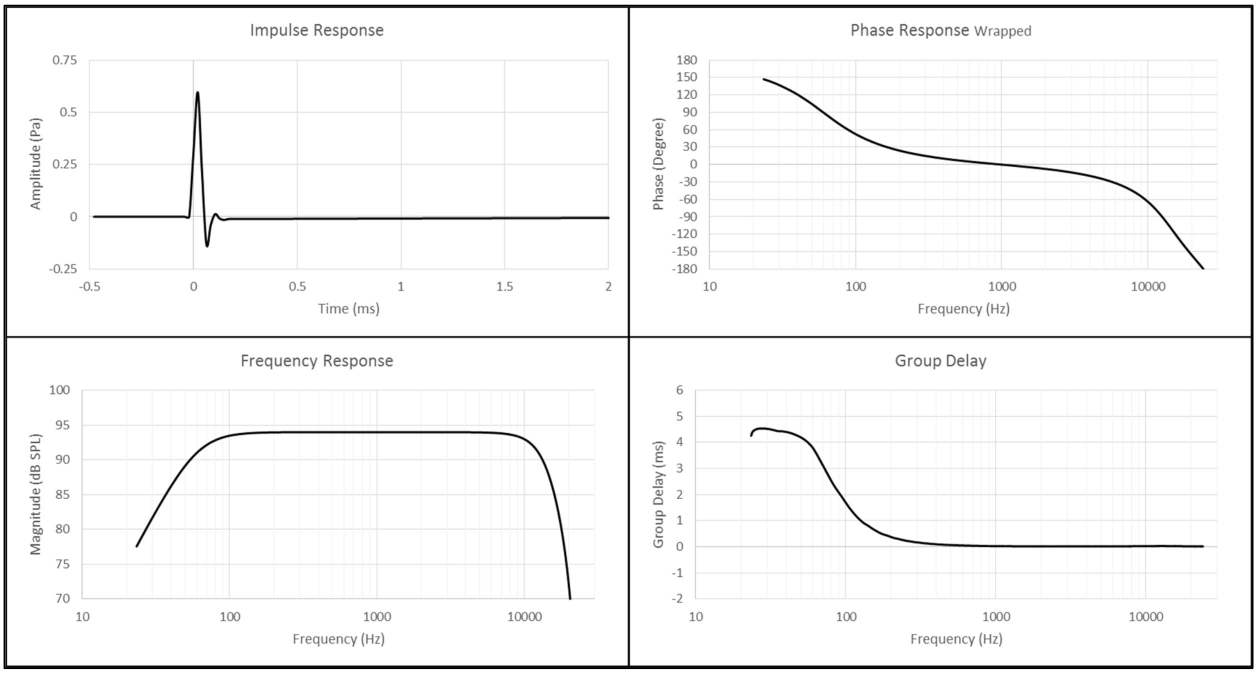

Figure 15

Several observations:

- ✓ At low frequency (below 200Hz), the phase value is positive due to the high pass condition (frequency response is rising up).
- ✓ At high frequency (above 8000Hz), the phase value is negative due to the low pass condition (frequency response is rising down).
- ✓ At frequencies around 1000Hz, the phase value is close to zero. This condition is expected since there are no changes in the frequency response at mid frequencies.

One little observation is the impulse peak that is shifted right a little bit. Let us observe figure 16 where the impulse response graph is changed to a dotted graph.

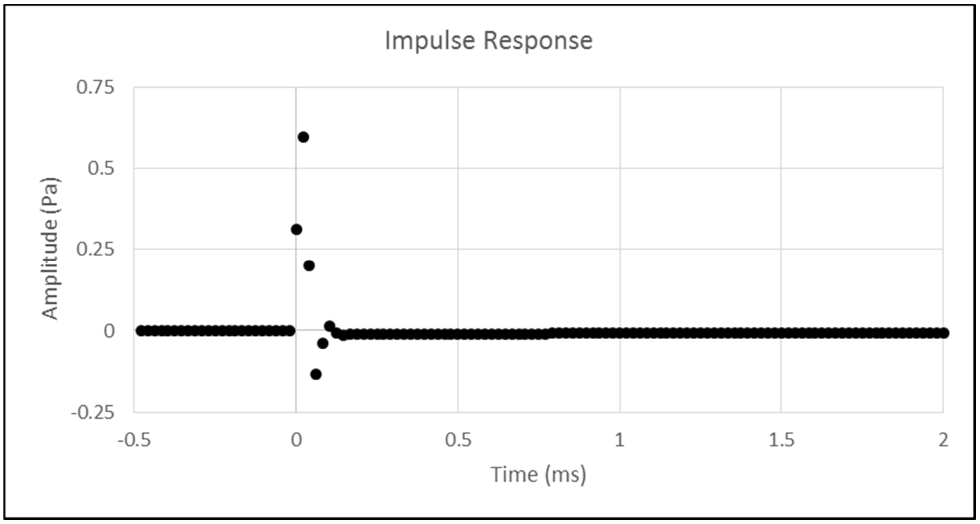

Figure 16

From figure 16, it is clear that the initial arrival of the impulse is correctly placed at 0ms. Most analyzers do not have the capability to change the graph from line to dotted graph, therefore it is important to always observe the transfer function (magnitude and phase responses) and impulse response at the same time.

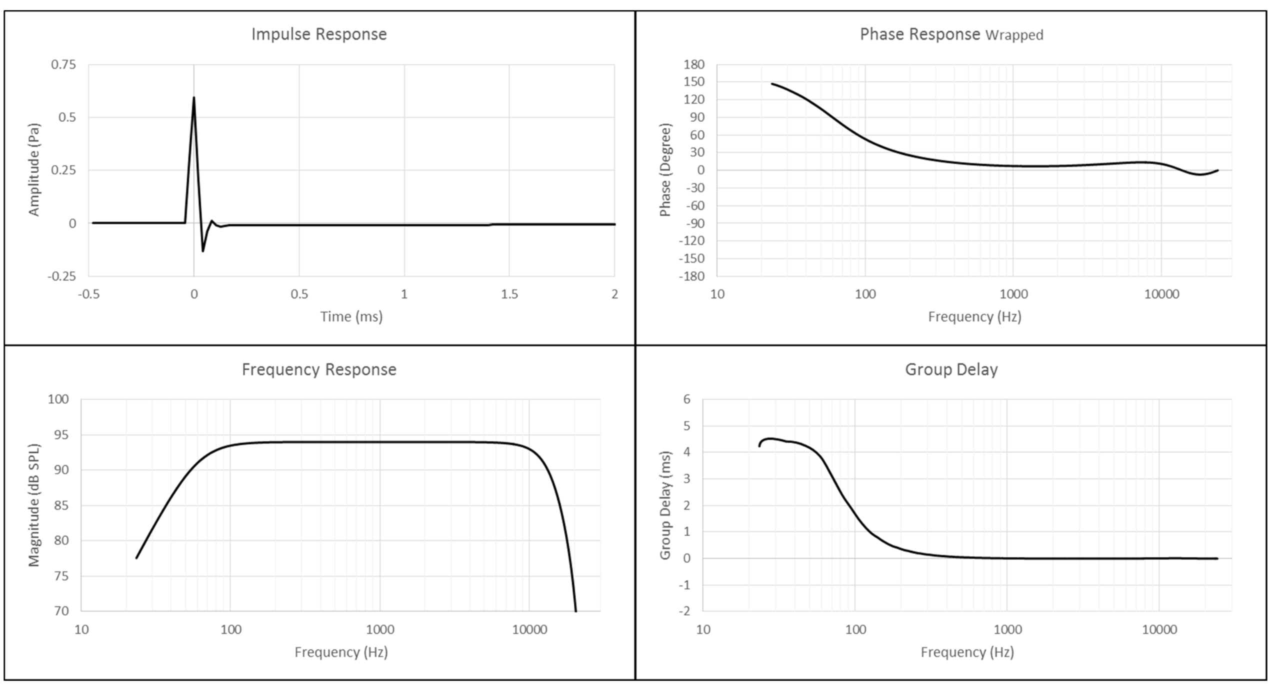

Figure 17

What happens if an auto delay finder is used? Let us observe figure 17.

- ✓ Frequency response does not change when the impulse response is cyclically shifted.
- ✓ The impulse response peak is placed at 0ms.
- ✓ Phase response becomes flatter, however please notice the values are positive/above 0deg up to 10000Hz. Positive values in phase can be an indicator that the impulse response is shifted a little bit too much, especially knowing that the frequency response is flat in the mid frequencies and rising down at high frequencies (above 8000Hz).

A phase response that has positive values can be observed in the next example, figure 18. A 2in woofer in a small closed box is measured at 1m away. The measurement is done without electronic filter and is a minimum phase system.

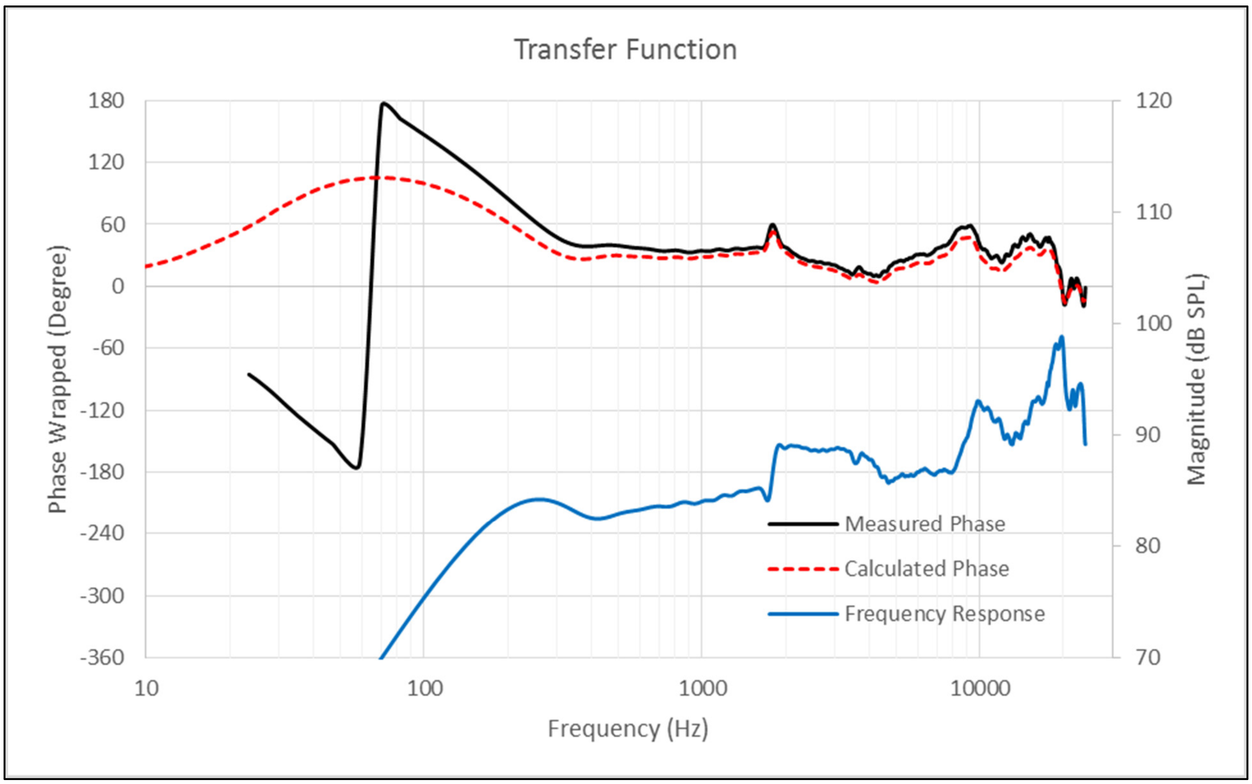

Figure 18

Several observations from figure 18:

- ✓ The calculated phase is the minimum phase response calculated from the magnitude response in ARTA. The calculated and measured phase responses are laying on the top of each other, showing that the propagation delay is correctly removed from the measured data.
- ✓ The phase values are positive all the way to 20000Hz. This is caused by the magnitude response that is rising up.
- ✓ Figure 19 points out the minimum phase condition by comparing figure 18 and 14. Please note the two green rectangular areas. Below 200Hz, the frequency response is rising; therefore, the phase response starts positive and goes down as shown in the figure 14 inset, graph A. At high frequencies, e.g. around 10000Hz, the measured frequency response follows graph C in the figure 14 inset. The reader can observe how the phase responses match between measured, calculated and graph C in the figure 14 inset.

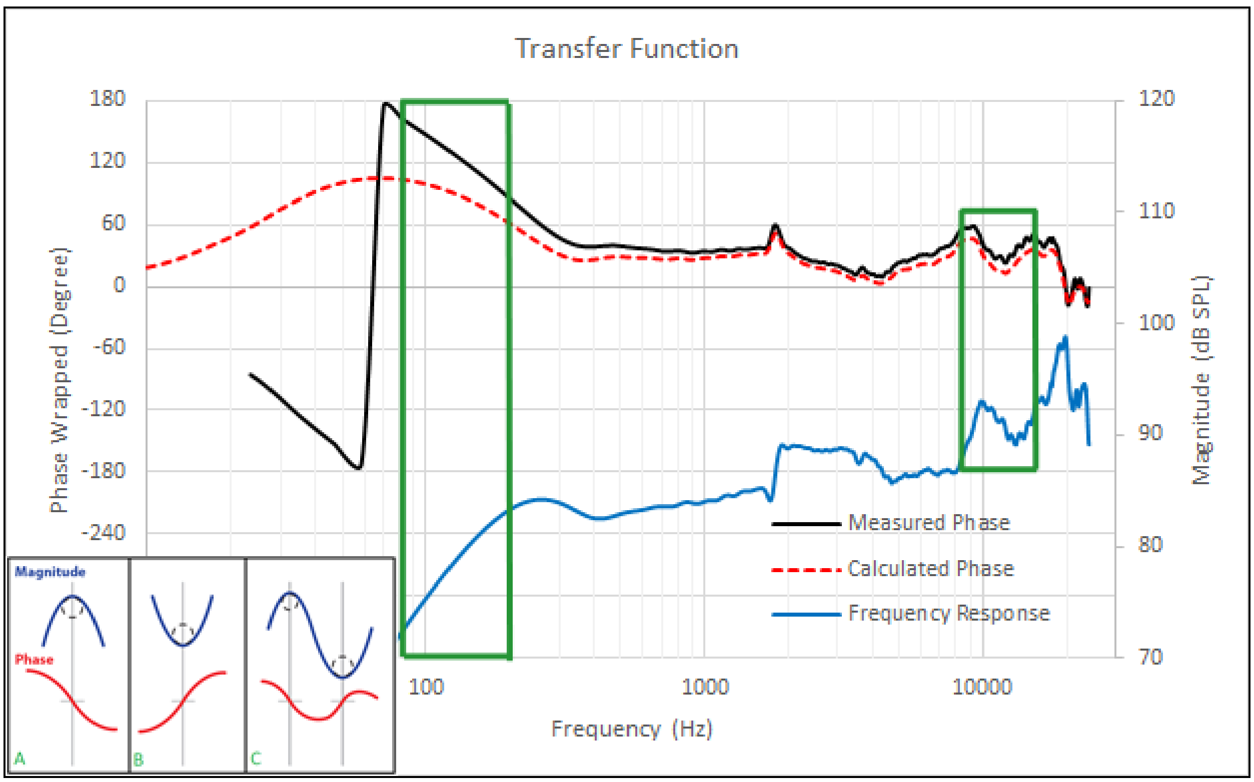

Figure 19

It is important to understand that phase response can be easily changed by cyclically shifting the impulse response and find a position where the phase response is the flattest. However the practitioner must also consider one question: is the phase response meaningful?

Meaningful Phase Responses

Meaningful in this context is useful for engineering use. Please see two phase responses in figure 20 below.

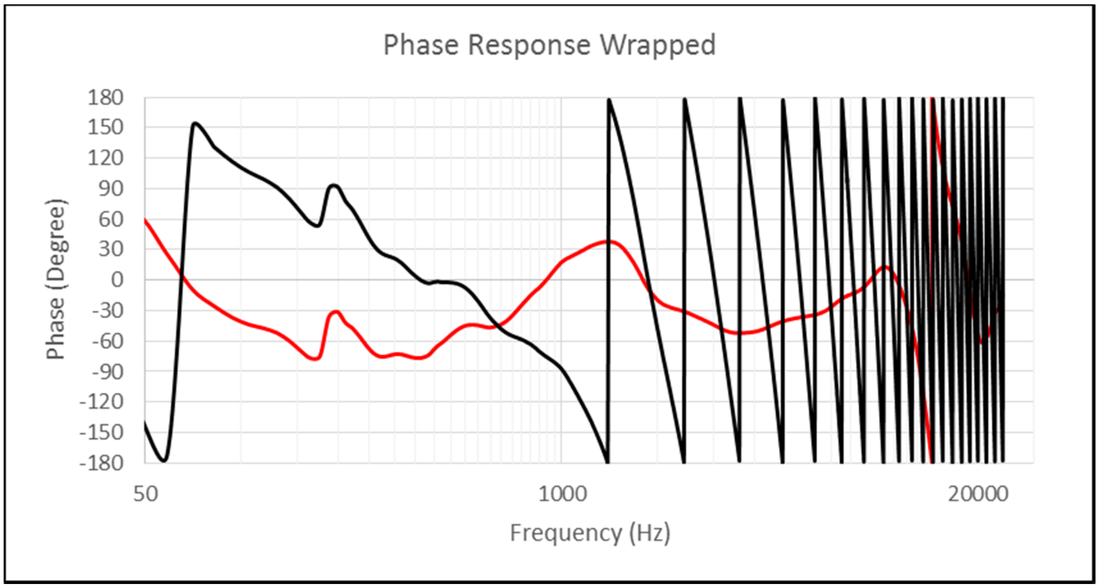

Figure 20

Let us do some visual observations of figure 20:

- ✓ Red curve is a better looking (flatter) phase response curve.
- ✓ Too many phase wraps on the black curve especially above 1000Hz.

Both curves in figure 20 are referring to the <u>same</u> measurement file. The black curve has the correct propagation delay removal, the red curve on the other hand is placing the peak of the impulse at 0ms (such as using auto delay finder function) with the input's polarity reversed.

There are always ways to make the phase response look flatter, but it does not mean it is useful for engineering use. We will discuss more about the black curve below. Let us discuss phase response with a comparison example.

A 12in 2-way horn loaded loudspeaker is measured passively without DSP and bi-amped with DSP. Please observe the on-axis measurement results in figure 21. Red curve is bi-amped with DSP and black curve is passive without additional DSP.

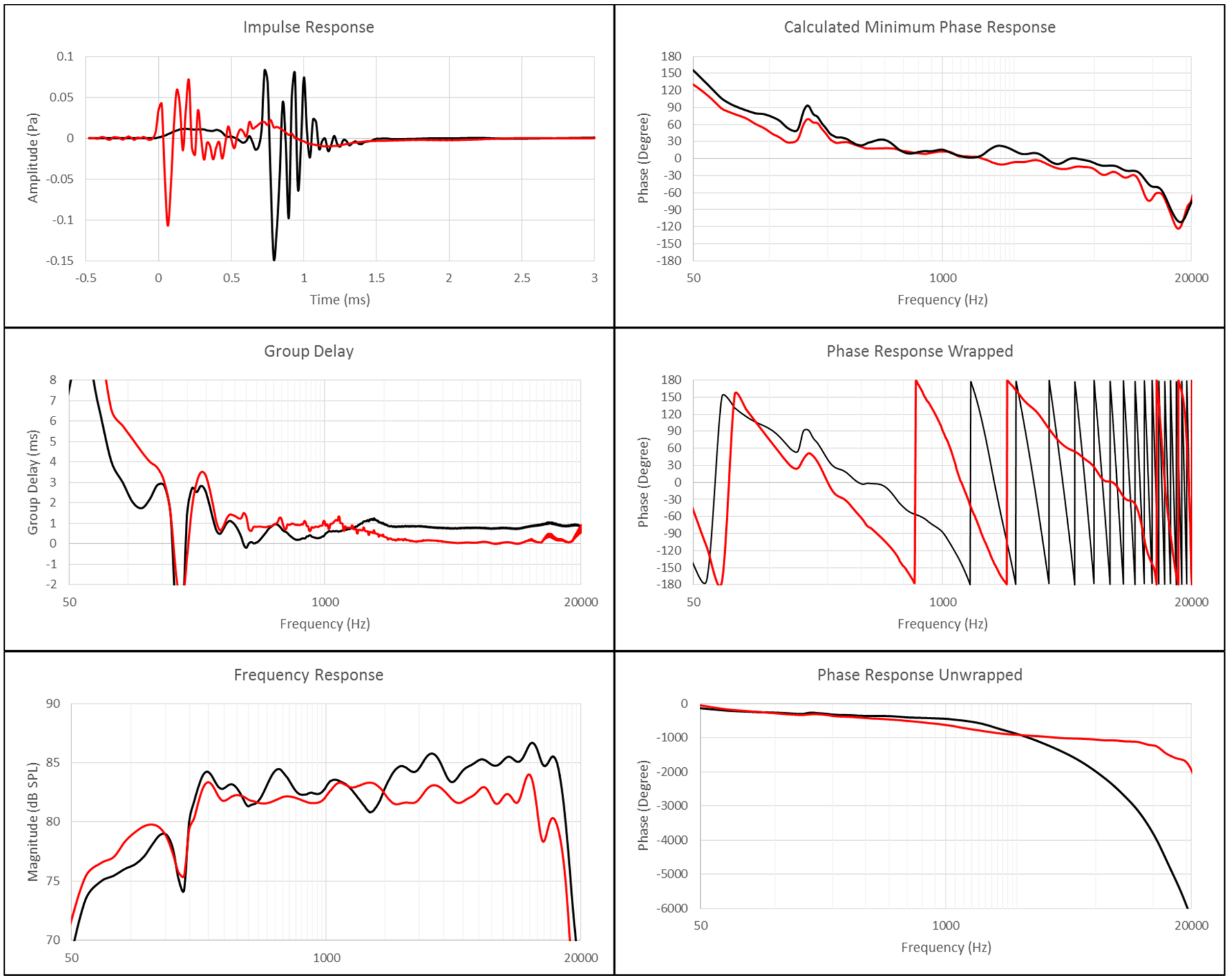

Figure 21

When an audio practitioner correctly removes the propagation delay in the impulse response, meaningful observations can be recognized. The discussions and observations below are focused to anything related to phase only.

- ✓ Calculated minimum phase response is different with the measured phase response.
  This condition explains that the loudspeaker under test is not behaving as minimum phase system in both passive and biamp setup. Their transducers are likely minimum phase, but as a loudspeaker system, the introduction of crossovers either passively or actively invalidates the assumption of a calculated phase from the frequency response as would be with minimum phase systems.

- ✓ Drastic phase drops/wraps on black curve at high frequency (above 1000Hz).
  The black curve is the loudspeaker run passively. If we notice the impulse response, the reader can see a small bump prior to the impulse peak. That small bump is likely the woofer impulse. It is smaller than the tall spike because the impulse response graph is 'dominated' by high frequency. When a low pass filter (at a frequency point below 2000Hz) is implemented to a woofer, the impulse height will be greatly reduced. The tall impulse peak is likely the tweeter, which comes later in time (approx. 0.7ms later). Since the loudspeaker runs with a passive crossover, this shows a condition where the tweeter's voice coil is located behind the woofer's voice coil, typically caused by the depth of the HF horn. The woofer's direct sound arrives first in the microphone and is followed by the tweeter 0.7ms later. As previously discussed in figure 6, the additional propagation delay (relative to the woofer that arrives first) creates the phase drops/wraps and creates higher group delay value at high frequency as discussed in the next point.

- ✓ Higher group delay value on black curve at high frequency (above 1000Hz).
  Following the previous discussion, due to the arrival difference between the tweeter and the woofer, the reader can observe the positive value of the group delay near 0.7ms above 1000Hz. This simply means that the tweeter is arriving 0.7ms behind relative to the woofer.
  The red curve group delay shows a curve at 0ms above 1000Hz. Comparing the red and black curves in the group delay and impulse response graphs, shows that a time delay is electronically used on the woofer to adjust the direct sound arrival difference.

- ✓ Higher group delay value on red curve at low frequency (below 100Hz).
  If we notice the frequency response, the magnitude of the red curve is higher than the black curve. This explains that a boost is performed in the woofer's low frequency region. This boost can be followed by high pass filter at lower frequency as processing are digitally implemented. Additional boost using parametric EQ and high pass filter will increase the group delay.

Let us go back quickly to figure 20, the red curve. The passive loudspeaker phase response can be shown flatter by placing the peak of the impulse response at 0ms and reversing the loudspeaker input's polarity during measurement. While this can be used for marketing/showing-off purpose, the phase response is misleading and is not meaningful for engineering work.

Conclusion

By removing propagation delay correctly, audio practitioners can derive valuable engineering information from a phase response.

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