A TRACK NAMED IMPULSE RESPONSE

Gaining an intuitive understanding of IR. *by Merlijn van Veen*

T he impulse response, or simply IR, remains an abstract concept to many. But if we think of the impulse response literally as an audio track in a digital audio workstation (DAW) like Pro Tools, Logic or Reaper, it might bring welcome clarity.

Let's start with Bob McCarthy's excellent explanation. The impulse response describes the answer to a hypothetical question: "What would this Device Under Test (DUT) look like on an oscilloscope when driven with an (im)pulse?" (**Figure 1**)

An oscilloscope displays audible sound as alternating current (AC) with positive and negative polarity where the ideal pulse (known as a Dirac pulse) would look like an instantaneous rise followed immediately by a fall without overshoot. It contains every audible frequency, lasting for exactly one cycle, starting at the same time and level.

Dual-channel fast Fourier trans-
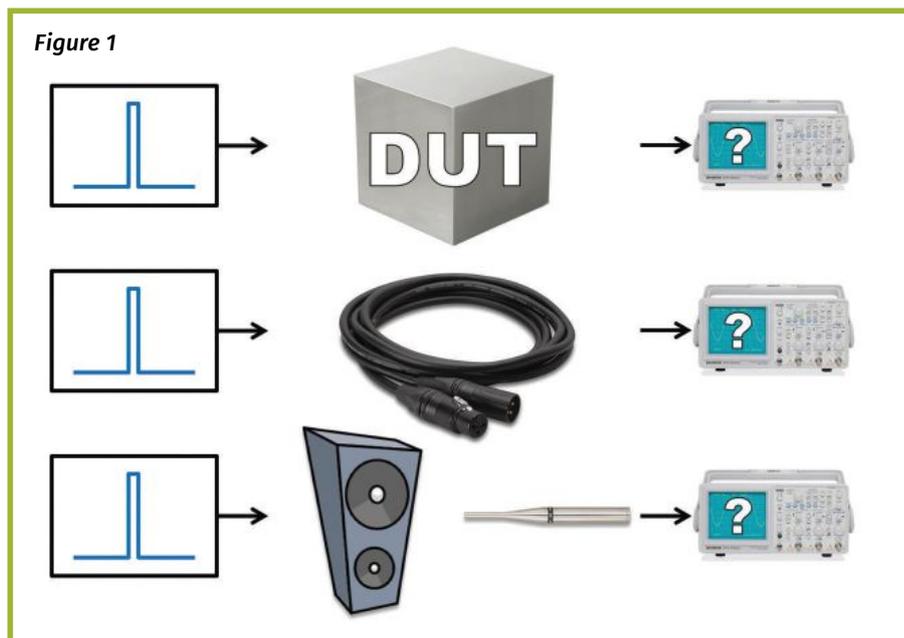
Figure 1

form (FFT) analyzers can obtain an IR either directly or compute it as a second-generation transfer function from the frequency domain by means of the inverse Fourier transform (**Figure 2**). This allows us to calculate the IR mathematically rather than having to fire an actual starting pistol, popping a balloon or clapping our hands. Time and frequency domain are reciprocal. They're joined at the hip like two sides of the same coin. It's exactly the same story told from different points of view.

If we introduce such a transient signal, e.g., a snare drum hit, into a microphone cable (a near-perfect conduit for our application), we expect it to come out at the other end instantaneously without any meaningful changes in time and/or level (Figure 1, Example 2). But what happens if we introduce a pulse into a DUT exhibiting a gratuitous amount of phase shift like a loudspeaker or, in this instance, a subwoofer (Figure 1, Example 3)?

Figure 3 shows a DAW (Reaper) with three tracks. Track 1 contains the IR of a microphone cable and Track 2 the IR of a typical vented subwoofer with

Figure 2

Inspired by "Getting Started with Smaart v7" by Rational Acoustics
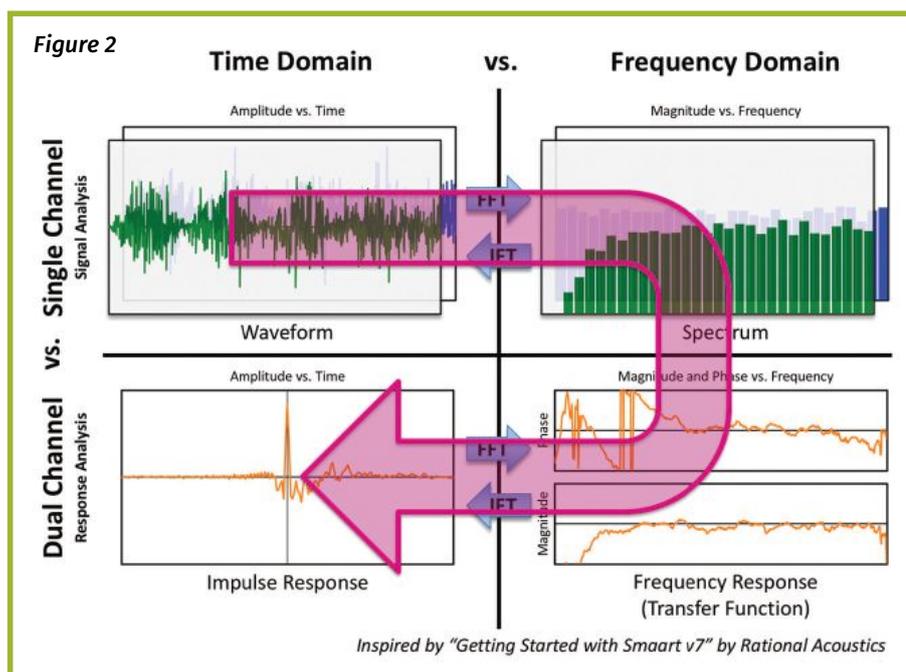
direct radiator(s) whose level has been normalized solely for visual purposes. Band-limiting a loudspeaker, like a subwoofer, reduces the amplitude of its IR which makes the waveform hard to see, particularly on the linear amplitude scale of a DAW or oscilloscope.

I've deliberately delayed both responses by 10 milliseconds (ms) so that their waveforms don't appear directly at the beginning of their respective tracks. Track 3 shows an arbitrary fragment of "The Sound of Silence" by Simon and Garfunkel (pun intended) to emphasize the waveform concept.

I've superimposed these very same IRs, as observed in an analyzer, on top of Tracks 1 and 2, respectively. When we conduct transfer function analysis, we use internal delay to synchronize the two channels, thereby compensating for propagation delay (latency and/or time of flight). Notice here that this practice acts like the actual playback position in a DAW.

If we scrub through the pulse in our microphone cable (Track 1) all audible frequencies will be heard instantaneously at the peak, i.e., a single-sample event. By contrast, the inherent phase shift of our subwoofer (Track 2) has stretched the pulse out over time and scrubbing through its track will typically let us hear only a single subwoofer frequency at a time. It's like sliding down the neck of a fretless bass guitar from high to low.

Let's pause briefly here and address why loudspeakers exhibit phase shift to begin with. By means of mechanical-electrical analogies, properties like friction, mass and stiffness (compliance) can be modeled in RLC circuits using resistors ("R"), inductors ("L") and capacitors ("C"), of which transfer functions can be calculated and measured.

In the case of vented enclosures (**Figure 4**, Left), the enclosure design alone (wood, screws and glue) will typically cause fourth-order high-pass filter behavior, including the inherent 360 degrees
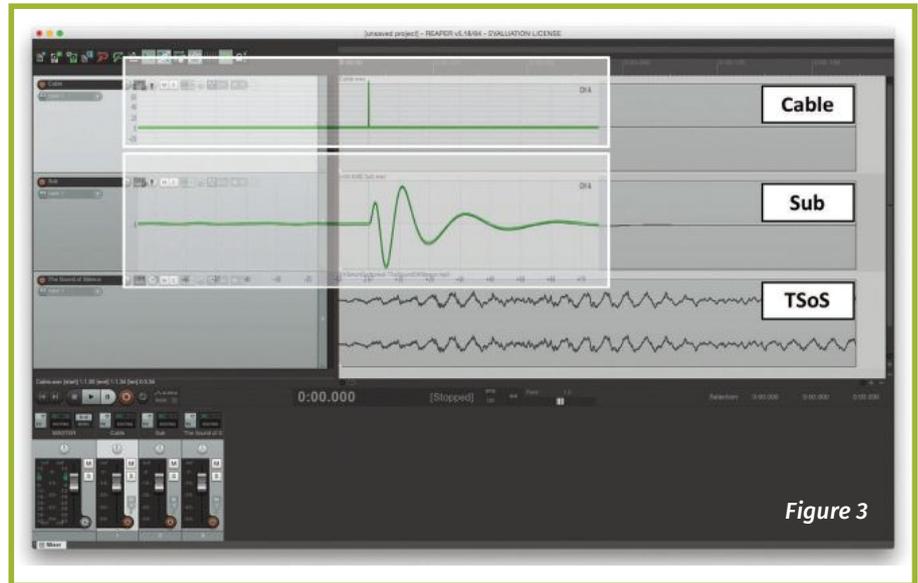

Figure 3

of phase shift, at the tuning frequency where port and direct radiator resonate. No electronics involved!

The vent (or port), however, is a double-edged sword. It extends low frequencies but at the same time enables the driver to "breathe" through the port, which until somebody bored a hole in the baffle was more challenging (trapped air - bicycle pump). Therefore, a subsonic electronic high-pass filter to prevent over-excursion (**Figure 5**, Row 2) or an alternative means of loudspeaker protec-

tion becomes mandatory.

To conclude this example, the subwoofer's bandwidth needs to be reduced to the intended range of operation. For vented enclosures (unlike band-pass designs), this is typically achieved by means of an electronic low-pass filter (**Figure 5**, Row 3). Upon completion, the compound phase shift typically equals three cycles (1,080 degrees) or more, throughout a mere 2-octave wide interval (**Figure 5**, Row 4)!
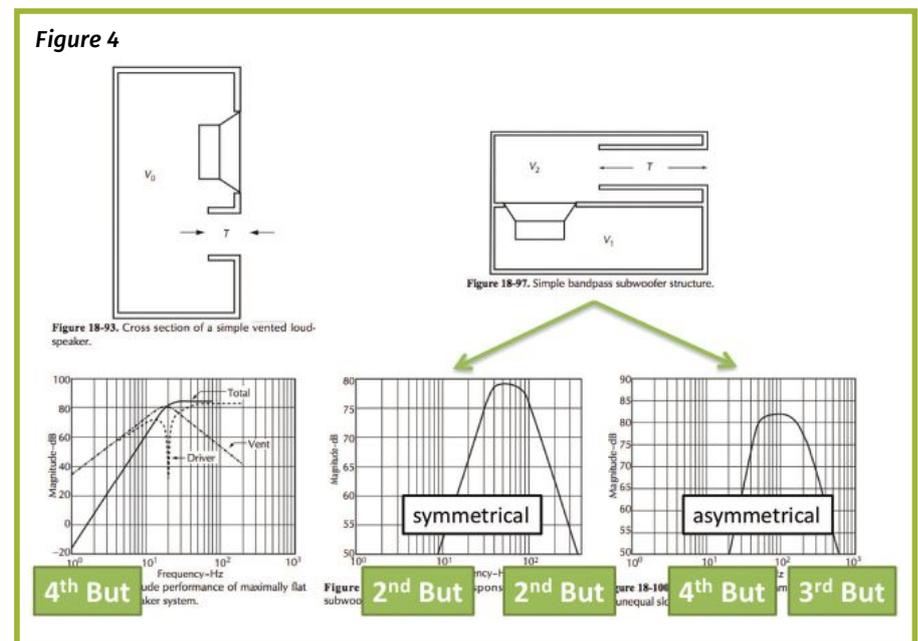
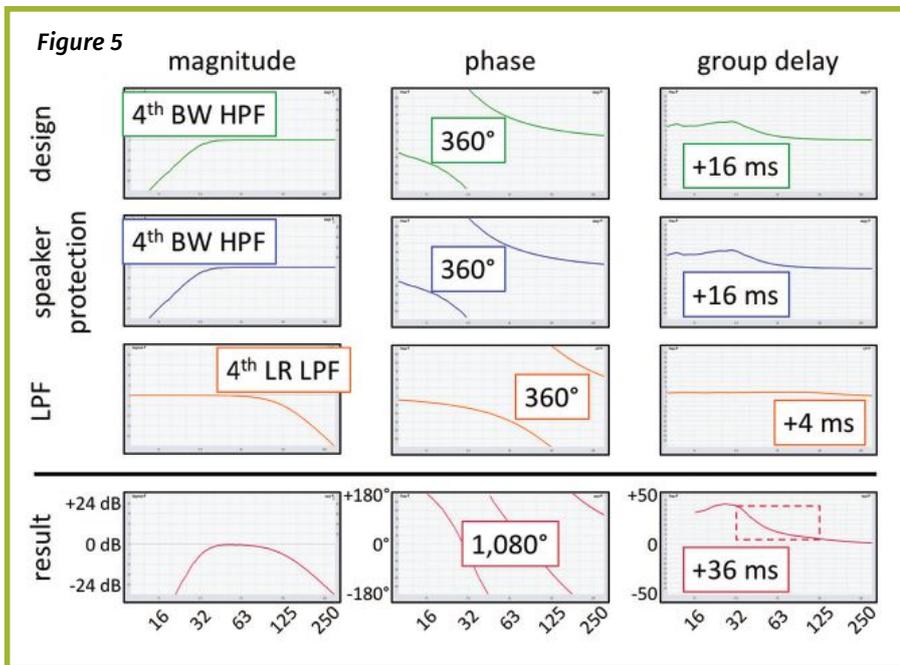This accumulated phase shift pro-

Figure 4

Figure 5

Figure 7

hibits loudspeakers from acting like microphone cables, which are free from any phase shift. And everybody wonders why a subwoofer is typically incapable of delivering more than a single frequency at a time?

TRACKING THE TRENDS

With that in mind, let's continue the original story. Loudspeakers are very likely to exhibit some degree of phase shift in their frequency response. Negative slopes in a phase trace indicate frequencies arriving late with respect to the reference signal, positive slopes indicate frequencies arriving early and no slope (horizontal tangent) indicates frequencies arriving in time with the reference signal (**Figure 6**).

These trends are not limited to any particular part of the frequency spectrum. More slope (regardless whether it's negative or positive) for the same frequency range means more time offset, less slope means less time offset.

In **Figure 7**, we "scrub" with the delay locator of an analyzer through the IR of a subwoofer. If you literally think of its IR as a track in a DAW (refer back to Figure 3), then depending on the current playback
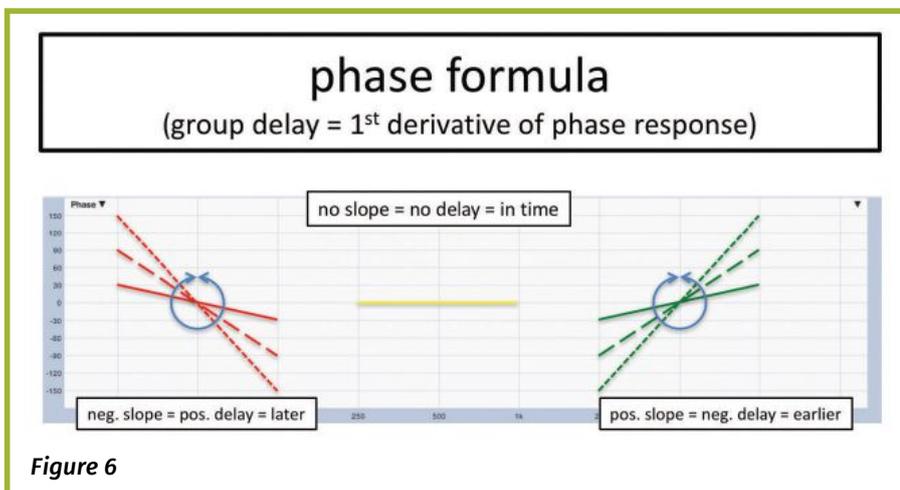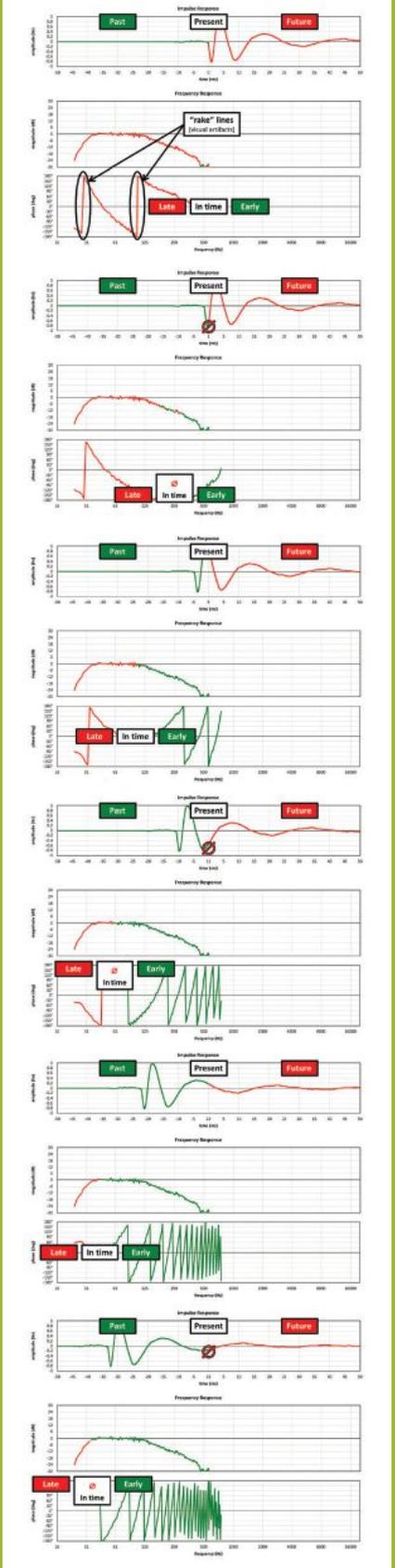

Figure 6

phase formula
(group delay = 1st derivative of phase response)

$$time = \frac{difference\ in\ time\ (cycles)}{difference\ in\ frequency\ (Herz)}$$

$$group\ delay = \frac{dt}{df} = \frac{-\left(\frac{[\phi_{high} - \phi_{low}]}{360°}\right)}{(f_{high} - f_{low})}$$

$$\frac{t}{f} = \frac{cycles}{\left(\frac{cycles}{seconds}\right)} = \frac{cycles}{1} \times \frac{seconds}{cycles} = seconds$$

Figure 8

position (present), some frequencies will already have been heard (to the left of our current playback position = past) whereas the remaining frequencies (to the right of our current playback position = future) have yet to be heard.

As you can see, this "song" takes time and the phase response that describes time over frequency confirms this – otherwise it would be ruler-flat like a microphone cable (without any phase shift). Regardless of where we are in the song, there's no visible range of phase trace without a change in slope. Only at a bending point, from negative to positive slope or vice versa, is the tangent horizontal. Here we observe a very narrow frequency span where measurement (output) and reference (input) are in sync.

In addition, the IR clearly shows that polarity is a moving target. The oscilloscope or DAW shows alternating current with alternating polarity. Some frequencies are in phase whereas others are out of phase. However, this does not affect important decisions regarding, e.g., crossover alignment. In such cases, the critical factor is matched phase responses in the overlapping region.

From a loudspeaker's phase response, we can calculate group delay. Group delay is complex math. It is the first derivative of the phase response (**Figure 8**). We can simplify it (but very badly, be warned) by visualizing it as frequency dependent delay which I don't define and is inaccurate anyway. Strictly speaking, group delay is the time delay of the amplitude envelopes of the various sinusoidal components.

The IR clearly shows that polarity is a moving target. The oscilloscope or DAW shows alternating current with alternating polarity.

Figure 9 shows common ways of observing time in an analyzer. For the subwoofer in this article, this particular arrangement enables us to estimate, at best, the way from IR (time domain) via group delay to phase response (frequency domain) and back. Our "song" begins at the very high frequencies with 0 ms and ends around 40 ms (45 feet/14 meters) later at 31 Hz!

Please remain calm! Just because I slapped a number on it is no reason to panic. Many readers have likely been aware of phase shift and group delay, but until now, very few likely considered them a problem. (Just kidding.) LSI

Merlijn van Veen (*www.merlijnvanveen.nl*) is a noted audio educator, and he also serves as senior technical support and education specialist for Meyer Sound.
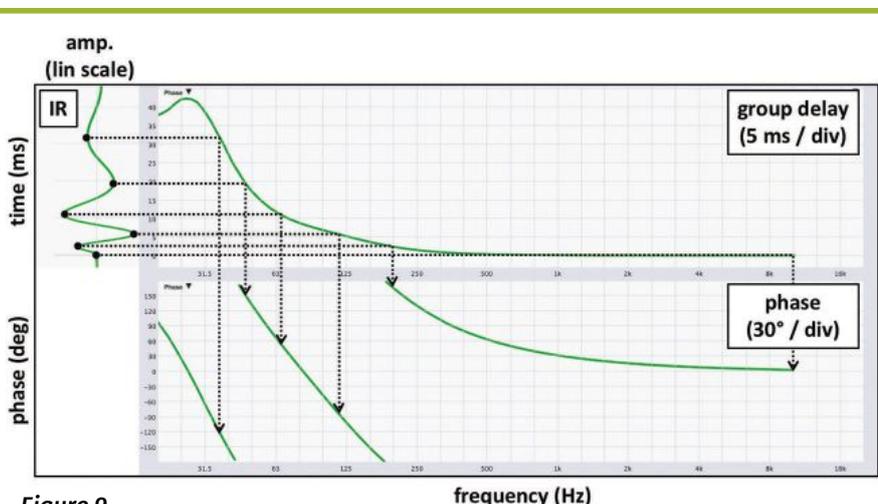

Figure 9