DON DAVIS

Measuring Acoustic Phase

One audio term often used and seldom properly understood is phase. Quite often when this word is used, the person actually means polarity. On other occasions, I have seen signal delay over the propagation path labelled as phase in a principal learned journal. In another similarly important learned journal, a tutorial article by a distinguished professor recommended teaching university-level engineering students about frequency response, directivity, and distortion measurements relative to the analysis of loudspeakers, but it failed to make any mention of phase. One cannot even find the word "phase" in the indexes of most basic texts on audio and acoustics. Yet phase can affect what we hear.

Indeed, many electronics-oriented engineering students have been taught that phase makes no difference, or is inaudible, at the frequencies commonly of interest to the human ear. It is true that most electronic circuits commonly encountered in hi-fi or sound systems have a minimum phase characteristic, but when one goes over to acoustic phase from electrical phase, the group of knowledgeable individuals familiar with acoustic phase measurements becomes quite small. (Those using Techron's TEF analysis or dual-channel FFTs to make loudspeaker and microphone measurements are an exception.)

Yet I sincerely feel that, had I been shown phase measurements first in my career, I doubt I ever would have bothered with acoustic amplitude measurements. Amplitude was simply what scientists first learned to measure, but it is not necessarily what was *important* to measure or necessarily most relevant to what we hear.

For those readers mathematically inclined, I recommend the referenced articles. They are correct, succinct, and thoughtfully thorough. It is really unfortunate that such a simple thing as phase had its mathematics labelled "complex numbers."

Painless Phase, Minus Math

The acoustic phase measurement can be understood as an observer merely having a different viewing point relative to a given signal. Figure 1 shows the two viewpoints of the analytic signal, called the real and the imaginary parts. As time progresses, the signal goes through 360° over and over again. The number of

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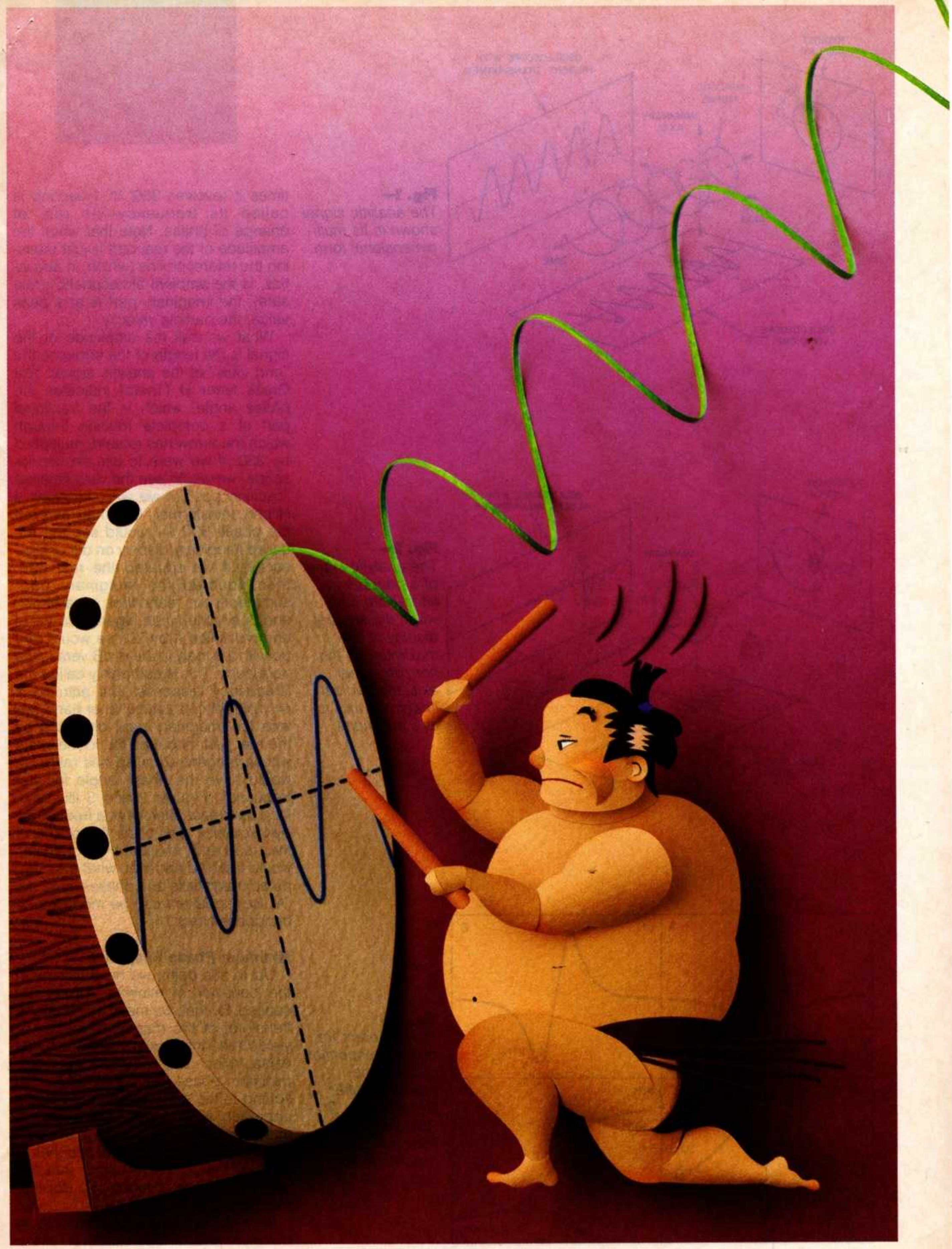

ILLUSTRATION: AJIN

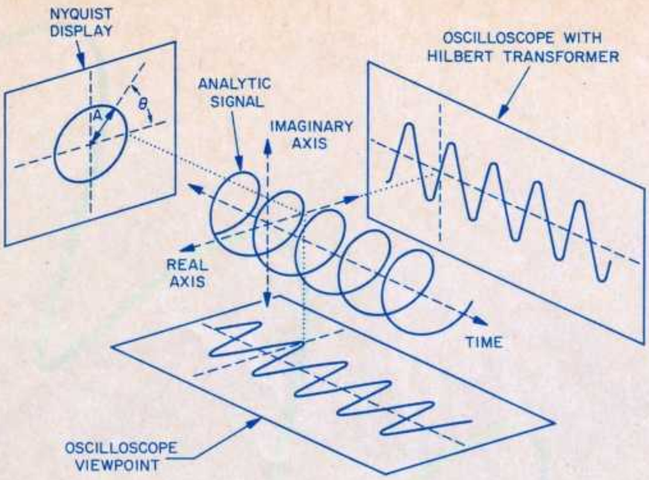

Fig. 1— *The analytic signal shown in its multi-dimensional form.*

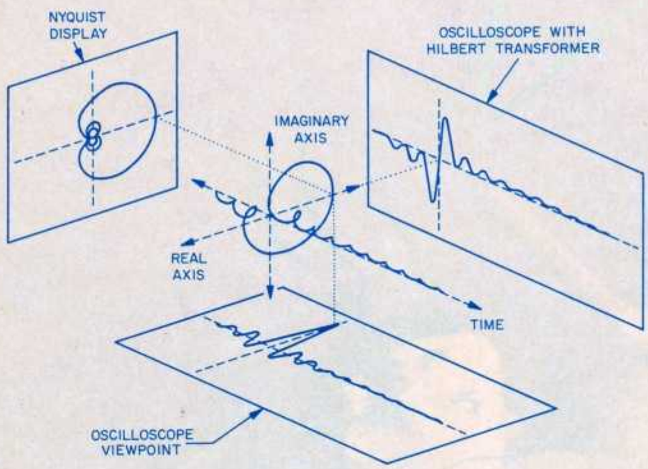

Fig. 2— *The analytic signal of a bandpass filter. Note that when the Nyquist display is at a maximum on the horizontal real axis, the vertical imaginary axis is at a minimum.*

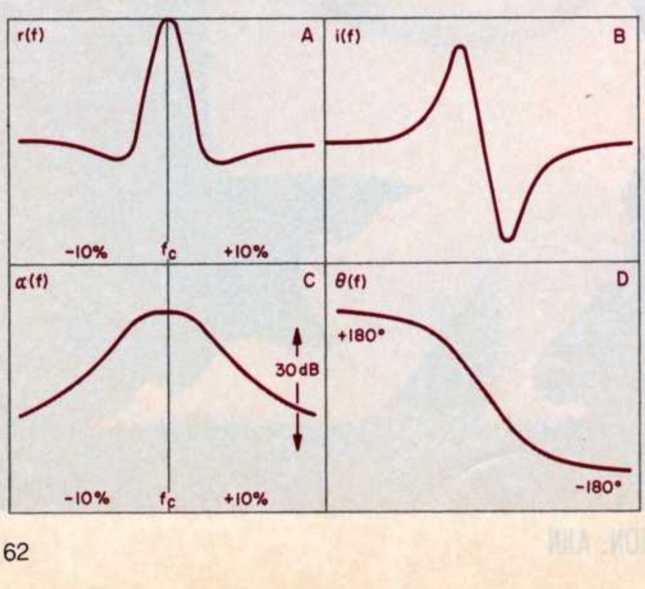

Fig. 3— *Four different frequency responses of the same bandpass filter shown in Fig. 2: Real part (A), imaginary part (B), magnitude (C), and phase (D).*

times it revolves 360° in 1 second is called its frequency—its rate of change of phase. Note that when the amplitude of the real part is just crossing the reference line (which, in acoustics, is the ambient atmospheric pressure), the imaginary part is at a peak value, the particle velocity.

What we call the amplitude of the signal is the length of the arrow on the "end view" of the analytic signal. The Greek letter Θ (Theta) indicates the phase angle, which is the fractional part of a complete rotation through which the arrow has rotated, multiplied by 360. If we were to use an oscilloscope, we would see the view labelled "oscilloscope"; if we were to install a Hilbert transformer, one that revolved the phase 90°, we would see the so-called imaginary display on our oscilloscope. If we squared the real part, then squared the imaginary part, summed them, found their square root, and finally took their logarithmic value and multiplied it by 20, we would end up with the magnitude in dB versus the frequency. This is commonly called the "frequency response" but actually is only part of it. If, on the other hand, we took the imaginary part, divided it by the real part, and then found the value whose tangent equalled this ratio, we would have the phase angle for that frequency. Figures 2 and 3 illustrate, for a bandpass filter, what a frequency-by-frequency plot of all the points of the real and imaginary frequencies would look like and then what the computed magnitude and phase would be. (All loudspeakers can be modelled as bandpass filters.)

Minimum Phase Response

Up to this point, we have illustrated the behavior of minimum phase response. By that, we mean that a Hilbert transform of the magnitude response yields the phase response, and vice versa. In fact, one of the quick tests for minimum phase response, without resorting to the more involved "S" plane approach, is to observe that a peak in the magnitude corresponds to the center of a slope in the phase. Further, a peak in the phase results in that frequency being in a center of a slope of the magnitude. (See Fig. 4.)

Why do we care if the response is minimum phase? One excellent reason

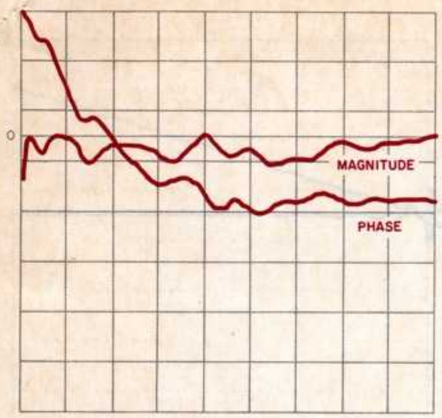

Fig. 4—*Phase and magnitude plots vs. frequency. The vertical scales are 6 dB/div. for magnitude and 45°/div. for phase.*

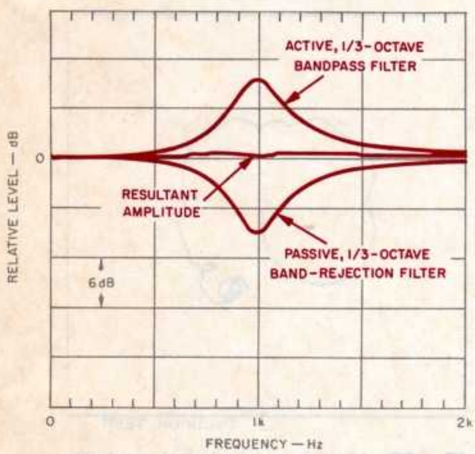

Fig. 5—*Magnitude plots for a bandpass filter, a band-rejection filter, and the resultant when both are in the circuit at the same time.*

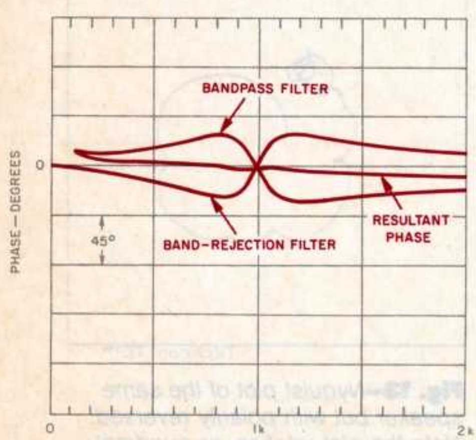

Fig. 6—*Phase plots of a bandpass filter, a band-rejection filter, and the resultant when both are in the circuit at the same time.*

is that minimum phase filters have uniform phase response for uniform amplitude response. (See Figs. 5 and 6.) Almost everyone has read at some time or another the statement, "I do not use equalizers because they cause phase changes." Of course equalizers do—they would not work if they did not—but the phase change the equalizers cause is a beneficial countering of a detrimental phase change caused by an unwanted bandpass effect in the system.

Signal Delay

Figures 7A and 7B show that the primary difference between an electrical and an acoustical phase measurement lies in the need to provide a signal delay in the measurement system. This delay is there mainly to compensate for the propagation path delay T(path) from the loudspeaker through the air to the microphone. A carefully calibrated signal delay allows for measurement of excess delay as well as for T(excess). Phase measurements use T(arrival) as their T(zero), and the phase measurement then becomes the difference between what the analyzer sent and what arrived at T(arrival). Perfection would be a flat response across the spectrum. In high-quality, real-life loudspeakers, we see the phase response first lead, then stay near zero, and finally lag, just as would be the case for a good-quality bandpass filter.

Let's have a look at this delay. Figure 8 shows the phase response of a small 4-inch loudspeaker in an infinite baffle (a totally enclosed box). The delay between the loudspeaker and microphone has been removed, so this is the true phase response. Figure 9 is exactly the same as Fig. 8, but the TEF analyzer's "quick difference" circuitry and software will show only the difference between what we had before and any changes we now introduce. If I move the loudspeaker back a quarter of an inch, Fig. 10 shows the delay as a straight line with a slope to the right. The signal delay is equal to the phase, in radians, divided by the frequency. Because it is a straight line, it is a constant delay at all frequencies. The steeper the slope, the greater the delay. If this measurement were to slope upward, that would indicate that the

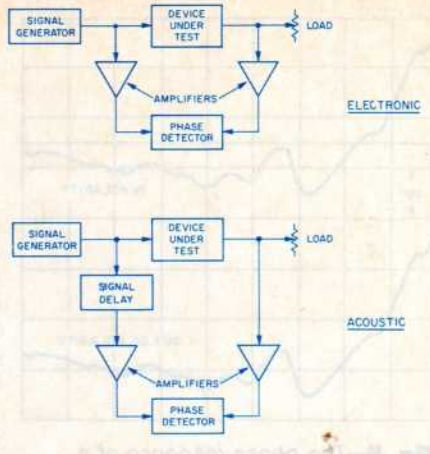

Fig. 7A—*Test setups for measuring electronic and acoustic phase. Note that acoustic phase measurements require adding a phase-calibrated delay device to the circuit.*

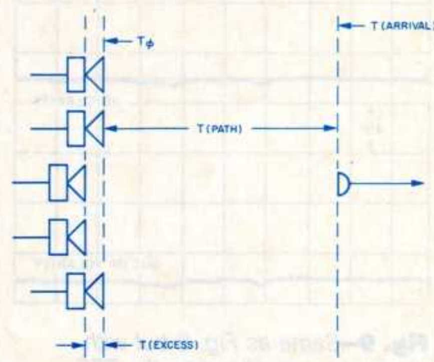

Fig. 7B—*The delay circuit shown in Fig. 7A compensates for the delay, T(path), shown here.*

loudspeaker was closer to the mike than its original position. As you can see in Fig. 11, signal delay is linear but phase is not. Another important distinction of phase is that it is frequency dependent.

Polarity

Often, we hear someone say, "I am going to phase my loudspeakers," when what he intends to do is make sure both diaphragms move in the same direction at the same time. The correct term for this is polarity. In my experience, Paul Klipsch was one of the very few who knew about this before phase measurements became common practice.

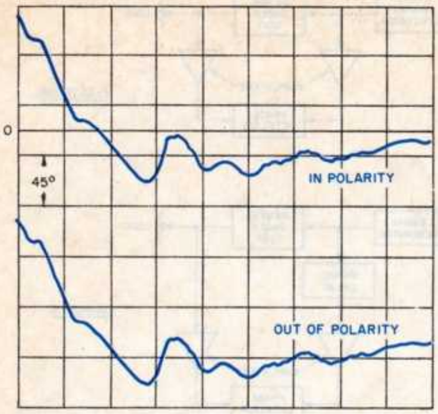

Fig. 8—*The phase response of a quality loudspeaker, both in and out of polarity. Note that polarity is* not *frequency dependent.*

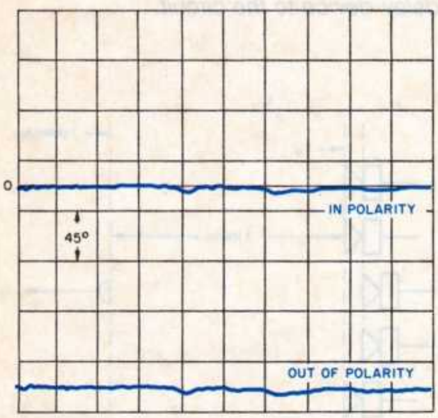

Fig. 9—*Same as Fig. 8 but with response normalized by the TEF analyzer's "quick difference" circuitry.*

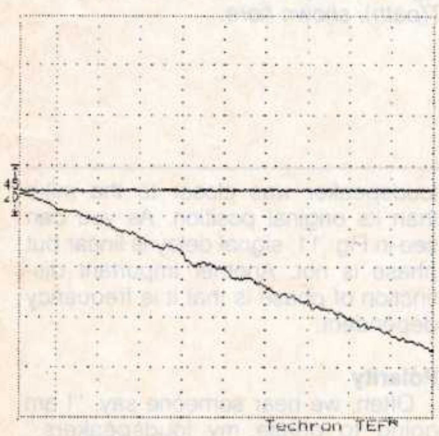

Fig. 10—*A signal-delay signature, sometimes mistakenly called phase delay. The slope's steepness shows the amount of delay; its flatness shows that delay is constant at all frequencies.*

Again, using the TEF's "differenced curve," let's reverse polarity (Figs. 8 and 9). Note that the phase response is the same curve but displaced 180°; it does not show up as delayed. Thus, a 180° displacement that is uniform indicates a polarity reversal. Looking at the Nyquist plot for both cases (Figs. 12 and 13) reveals quadrant shift. Note particularly the offset from the origin of the plot in Fig. 13. A distinguishing characteristic of polarity is that it is *not* frequency dependent—every frequency jumped 180°.

Those who would like to listen to phase interference in order to know what it sounds like should try the following simple experiment. Take two small single-cone speakers and place one on top of the other. Have a friend talk over both speakers through a single amplifier with a microphone. Now move one loudspeaker back approximately one foot and listen again. Have your friend move the loudspeakers back into synchronization (sometimes called alignment, though I feel synchronization is more correct) while he is talking through them. What you will hear with devastating clarity is phase interference.

Conclusion

Phase measurements can be much more complex than the concepts presented here. They can be used to locate poles and zeros in the complex plane. It is the highest resolution way to find circuit Q and the natural frequency. Phase measurements of loudspeakers are important, and variations in phase and polarity are audible. It is hoped that this simple discussion of what phase is and is not will prove useful when you next look at a simple amplitude response plot and realize how much information is missing. **A**

References

Preis, D., "Phase Distortion and Phase Equalization in Audio Signal Processing—A Tutorial Review," *Journal of the Audio Engineering Society*, Nov. 1982.

Preis, D., "Linear Distortion," *JAES*, June 1976. (This article is an absolute must for those who want to delve deeper into the subject.)

Luthropp, Dave, "Measure Phase Instead of Amplitude," Hewlett Packard.

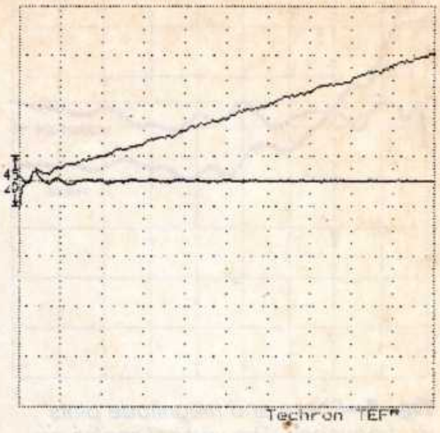

Fig. 11—*A signal-advance signature.*

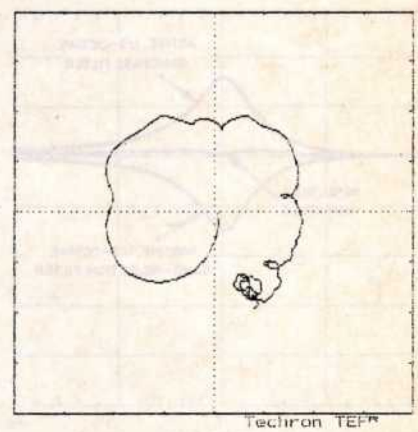

Fig. 12—*Nyquist plot of an in-polarity loudspeaker.*

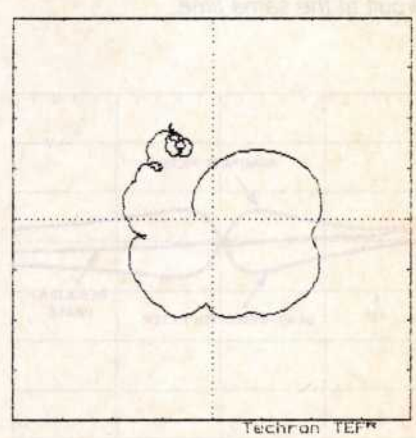

Fig. 13—*Nyquist plot of the same speaker but with polarity reversed. Note the plot rotation, or quadrant shift, and the offset of the origin relative to Fig. 12.*

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MEASURING ACOUSTIC PHASE

In illustrating my article, "Measuring Acoustic Phase" (February), I made two grievous errors. I failed to credit where two of the original illustrations came from (Figs. 2 and 3) and, in the case of one of them, picked the wrong one for the point I wished to make. The illustration I used as Fig. 3 was the time-domain analytic signal for a low-pass filter, whereas I had meant to choose the same for a bandpass filter.

When Andrew Duncan of Cerwin-Vega, the creator of the software that generated Figs. 2 and 3 in my article, read the piece, he contacted me to point out the omission of the credit and the misinterpretation. Duncan was then kind enough to generate the correct illustrations for the point I was trying to make and chided me on my failure to clarify in the text what was the time domain and what was the frequency domain when I discussed the manipulation of the signals for other displays. Because Duncan deserves credit for his work, and because I don't like erroneous material in print with my name attached, I offer, with his collaboration, the following clarification.

Figure 1 shows the relationships between measurements in the time domain (left side of chart) and the frequency domain (right side). To go from the left side to the right normally requires a Fourier transform, and to come back to the left side from the right normally requires an inverse Fourier transform.

To go from the real to the imaginary requires a Hilbert operator. The real part, in the time domain, has been given the name impulse response; the imaginary part is called the doublet response (Fig. 2). If the signal acquisition is done in the frequency domain, the real part is called the coincident response and the imaginary part is the quadrature response (Fig. 3).

In my article, the illustrations of the frequency-domain real and imaginary parts and of the phase and magnitude responses are correct, but the wording in the text implies that taking the impulse response's real and imaginary parts allows these calculations without first taking their Fourier transform. That implication is, of course, incorrect.

One can plot the envelope and the phase of the impulse response as well,

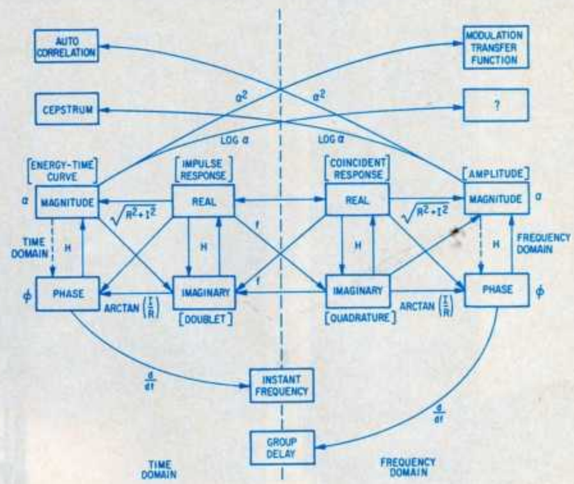

Fig. 1— A basic "road map" of acoustic measurements, in both the time and frequency domains. The box labelled "?" (upper right) is the forward Fourier transform of the complex log of the energy-time function. (After B & K.)

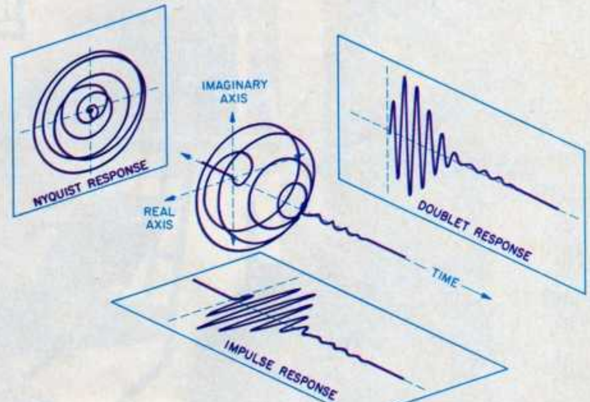

Fig. 2— A time-domain depiction of the analytic impulse response of a four-pole Butterworth ⅓-octave bandpass filter. Note the impulse response (real) and the doublet response (imaginary). All loudspeakers can be modelled as bandpass filters. (This and the following figures from Andrew Duncan.)

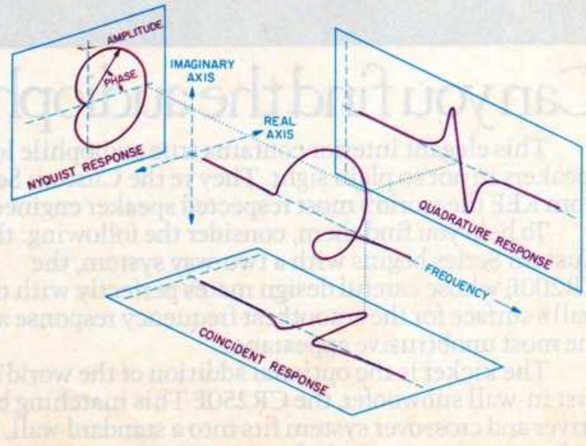

Fig. 3— A frequency-domain depiction of the filter in Fig. 2. Here, the real part is called the coincident response, and the imaginary part is called the quadrature response. The Fourier transform of the time-domain analytic signal uses this display.

[redacted]

in which case the envelope is called the energy-time curve (ETC) and the phase and magnitude curves are taken in the frequency domain (see Fig. 4).

Duncan further pointed out that my choice of words regarding the signal-delay display (Figs. 10 and 11) could and did cause confusion. The phase curve shown was linear, and the signal delay shown was constant.

There are few higher signs of respect than to have readers read with care and then share their thinking with you. My sincerest appreciation goes to Andrew Duncan for his constructive, helpful corrections and for the aid of his superb computer software in the depiction of these fundamental relationships.

References

Duncan, Andrew, "The Analytic Impulse," *Journal of the Audio Engineering Society*, May 1988 (Vol. 36, No. 5).

Schillinger, Joseph, *The Mathematical Basis of the Arts* (chapter on quadrant rotation, pg. 233), Philosophical Library, New York, N.Y., 1948.

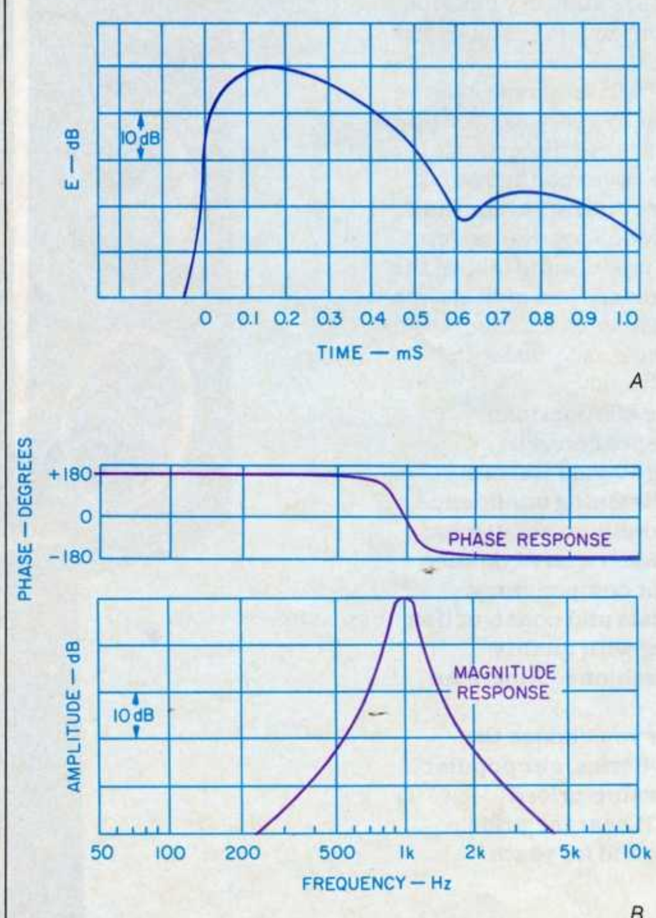

Fig. 4—In this illustration, the time-domain view shows the energy-time curve (ETC), which is the energy envelope over time (A). The frequency-domain transfer function (B) consists of the phase and the magnitude response of the signal.