Why we should care about phase

The sonic anomalies and performance of audio systems can be, and are, attributed to a variety of different processes yet the matter of phase is frequently overlooked. TONY WALDRON of CADAC Electronics says that getting phase correct is an essential for any piece of equipment that alludes to true quality.

WHAT A FINE INDUSTRY to work in! Just when we had developed a good understanding of how to design equipment using thermionic tubes, some bright spark decided to throw it all away and change to transistors. As soon as we had found out how to make transistors work to our advantage, the semiconductor industry developed cost-effective methods to package hundreds of transistors into a single chip, so we had to start over again. Now we are told to forget all about analogue electronics because it is more convenient to do our signal processing in the digital domain. This means that since I started designing systems with valve electronics, I have been involved with re-inventing the wheel four times now. But I'm not really complaining, because going back over the basic principles for the fourth time reminds you that electronics must obey the laws of physics.

As one would expect, there have been a large number of changes in the pro audio industry during the last 40 years. Back in the 1960s almost all professional audio equipment relied on circuits that used thermionic tubes. One of the less obvious differences from those days was that much of the equipment used by members of the audio profession was also designed and built by them. In the UK, that meant that if you decided to set up a recording studio, you needed to be able to design and build your own mixer. The well known recording companies built their own: EMI, Decca, Pye, etc., as did the BBC. Note that the live sound and commercial radio parts of the industry did not yet exist. And there was another important difference from today — designers of pro equipment spared no expense in the search for the highest possible audio quality. No audio professional went looking for low cost bits of kit, and there was a general expectation that high performance electronics would cost more than the equipment you used at home.

By 1963, the expansion of popular music culture was driving dramatic changes across the industry. Artists demanded to use the newly available multitrack tape recorders, which led to the need for increased recording session time, which in turn created the requirement for extra studio space. Very soon, independent recording studios started to appear: Lansdowne; Advision; De Lane Lee; Olympic. At about the same time Pirate Radio stations began broadcasting from boats moored just outside the 3-mile limit, and the first Rock Musicals appeared on Broadway and the West End. Recording engineers invented and developed brand new techniques: multitrack and stereo recording; chorus, phasing, and flanging.

To cope with all this, a whole new range of audio engineering manufacturers popped into existence: Sound Techniques; Neve; CADAC; Helios (in the UK); Spectrasonics; Quad 8; MCI; Harrison (in the US). Almost without exception, the newly-formed independent audio manufacturing industry embraced the relatively new transistor as the basis for their circuit designs. Physically smaller, more convenient and much easier to use, the transistor proved to be difficult to integrate into the high performance designs that were equivalent to the best tube equipment of the day. The more obvious reasons being very much reduced headroom, limited bandwidth and poor transient response. Nevertheless, by the early 1970s semiconductor electronics had been tamed and the search for high performance audio was well and truly in top-gear once more (overdrive still being sought today).

Into this heady mixture of science and art came improvements in loudspeaker design, and a better understanding of room acoustics. This led the artistic branch (balance engineers and producers) to demand higher definition electronics in the mixer/processing departments. Luckily, most of us involved in designing audio equipment in those far off days had learned much of our trade on-the-job, so we were pretty much used to juggling the various parameters that affected the performance qualities of our circuits to obtain the engineering compromises necessary to solve the problems. We found that higher definition at low frequencies was possible by improving the transient response and phase response of the circuits. Better high frequency definition was obtained by increasing the bandwidth of the input amplifier, which was itself involved with the phase response characteristics as we will see.

It turned out that understanding the vagaries of the phase response of a circuit or system was, and still is, essential for evaluating how good the design really is. Unfortunately, a graphical plot of a system's phase response does not look very flattering, as can be seen from Figure 1, so marketing executives have long since banned such things from publication (along with polar plots and power response diagrams for loudspeakers).

The phase response of a system is important, because the human auditory system is extremely sensitive to changes in phase. Detecting and processing the results of the phase relationship of a sound source allows us to locate its direction and position with extraordinary precision — even if the sound comes from behind us. Think of the situation where someone calls your name — in a room or in the street. The sound reaches first one ear and then the other ear. The brain calculates the angle represented by the difference in the arrival times of the sound at each ear, and we know precisely in which direction to turn to find the sound source. In fact, most of us are able to work out how far away to look for the sound source based on our experience of the natural loudness of the various sounds that we know about or regularly work with.

Now, the ability to respond to changes in phase evolved mainly for the preservation of life (in the days when humans roamed the earth as hunters and gatherers). Since then, many of us have adapted the same mechanism to help us appreciate the complexities of music and drama. A small number of audio professionals have become even more specialised in ear resolution, by training their auditory system to recognise extremely small phase changes at either end of the audio spectrum. For their efforts, they are often awarded the accolade of The Golden Ear Brigade. While this is often used as an insult, it is usually found that the person voicing the phrase in this manner has the hearing acuity of a dead slug.

As can be seen from Figure 1, the rate-of-change of phase is greatest at low and high frequencies. Listening tests reveal that humans are particularly sensitive to changes in phase when the rate-of-change of phase is excessive. For instance, we are easily able to detect changes in the timbre of low frequency sounds if the rate-of-change of phase in the reproduction system is more than about +5 degrees per octave. So, what is it that controls the low frequency phase response in electronics?

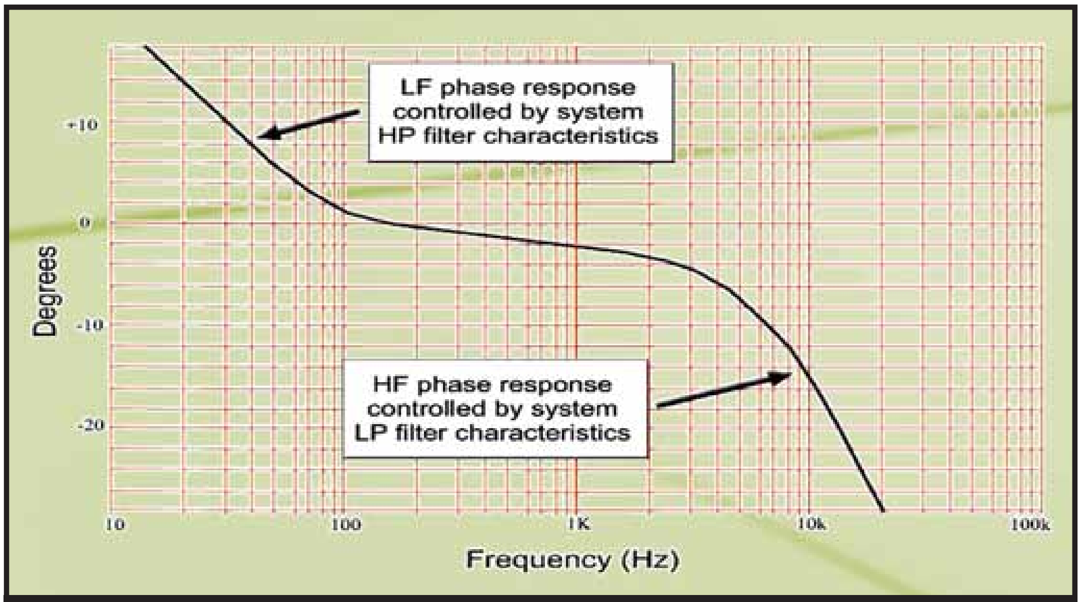

Figure 1. A typical phase response plot.

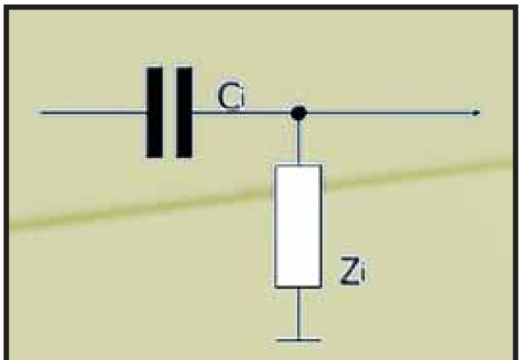

Figure 2. A single-pole high-pass filter

When we consider the frequency domain, it is immediately obvious that the input circuit for any electronic device is essentially a high-pass filter (Figure 2). The low frequency response is therefore determined by the value of Ci, for a given value of input impedance Zi. For a modern transformerless microphone amplifier, it is generally agreed that a reasonable value for Zi is around 1kohms. This is based on engineering experience that tells us that the input impedance (or load) should be at least 5 times more than the source impedance. So, an input impedance of 1k is based on the understanding that the source impedance of the microphone is about 200ohms. A simple calculation then tells us that the value for Ci only needs to be about 0.5µF, in order to get a '-3dB down' point at 20Hz.

It is not until we consider the phase response at 20Hz that we realise that such a low value for Ci will be a sad mistake. If the frequency response is -3dBu at 20Hz, then the phase response at the same frequency will be more than +30 degrees. The rate-of-change of phase is excessive and a listener who is used to working with sound sources that develop complex harmonics, will complain that the reproduced sound is not accurate. A quick look at most allegedly high performance microphone or line amplifier circuits will reveal values for Ci of at least 100µF.

At high frequencies, the phase response of a circuit or system turns negative, because the combined circuit elements behave as a series of low-pass filters. The rate-of-change of phase begins to increase rather rapidly above a frequency that is equivalent to 1/10th of the turnover frequency of the combined filter network. Thus, the behaviour of the high frequency phase response changes with the overall bandwidth of the circuit or system. Our natural ability to detect changes in phase allows us to resolve quite subtle changes in the overall bandwidth of a system.

In the early 1970s, I was involved in a series of tests to try and determine the minimum bandwidth required for an input amplifier. Using a calibrated microphone and a spectrum analyser we recorded the overall harmonic response of a number of different musical instruments. It was immediately clear that many acoustic instruments produced enormous amounts of harmonic energy well beyond the accepted audio bandwidth of 20Hz to 20kHz. Figure 3 shows an old Polaroid photograph of the Spectrum Analyser screen while a percussionist gently rattled a tambourine.

Six special microphone amplifiers were constructed to allow the input bandwidth to be switched from wide-open (100kHz) to 40kHz in 20kHz steps. A recording session was organised to record a simple jazz trio — piano, bass and drums. The listening tests

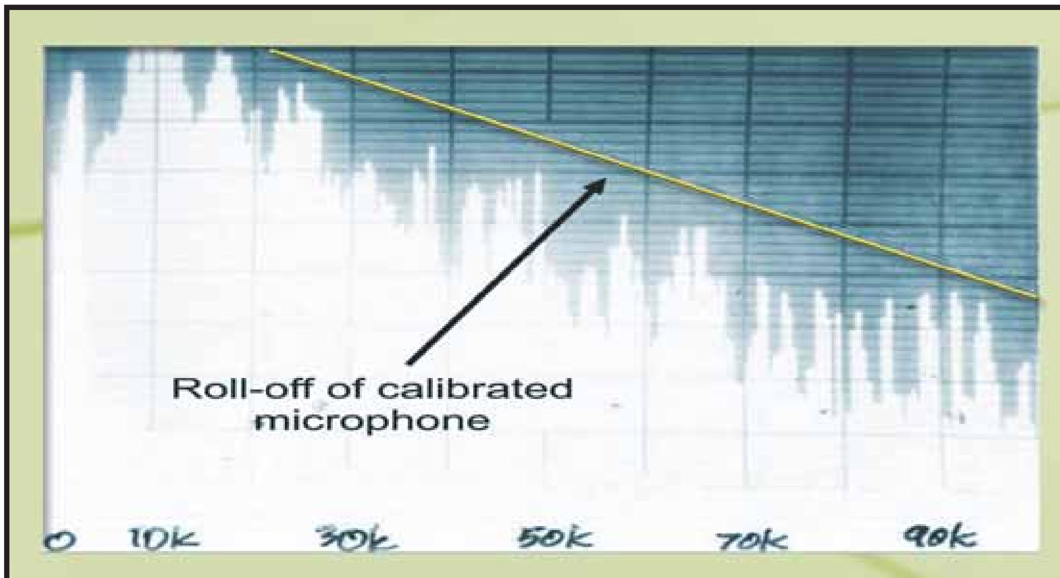

Figure 3. The amplitude/frequency response of a tambourine.

revealed that everyone in the control room could hear when the bandwidth of the amplifiers on the percussion microphones was reduced from 100kHz to 80kHz. Reducing the bandwidth to 60kHz further modified the percussion sounds, but also changed the timbre of the piano. When the bandwidth of all the microphone amplifiers was limited to 40kHz, the reproduced sound on all three instruments was heavily modified. Of course we had the advantage of being able to instantly compare the original sound sources with the electronic reproduction, by walking between the control room and studio.

Back in the lab, looking at the frequency domain on its own did not indicate any sort of electronic anomaly. But, when we checked the phase response, it was clear what was happening. As the high frequency turnover point was reduced, the rate-of-change in the phase response increased, and the frequency at which the phase response turned negative (1/10th of the upper band-pass frequency) fell from 10kHz to 4kHz (Figure 4). The increase in the rate-of-change of phase was always in the part of the audio spectrum that a human with normal hearing can easily resolve.

When we are designing or specifying high performance audio equipment, we need to be aware that a number of different parameters gang together to effect the overall outcome of the system response. Focussing on a single parameter will not be enough to explain what is actually going on, because all of the different parameters interact with each other, due to the fact that they are inextricably linked by basic physics. Note that so far, we have only discussed two!

High performance audio engineering requires that the frequency response of our input stages must have a bandwidth wide enough to minimise the rate-of-change of phase at the extreme ends of the audio spectrum. Unfortunately, checking the phase response of digital-audio systems is difficult and time consuming, but that is what we need to do if we are interested in accurate sound reproduction. Don't be surprised if you can hear the difference between 48kHz and 96kHz sampling convertors, but be very worried if you cannot. ■

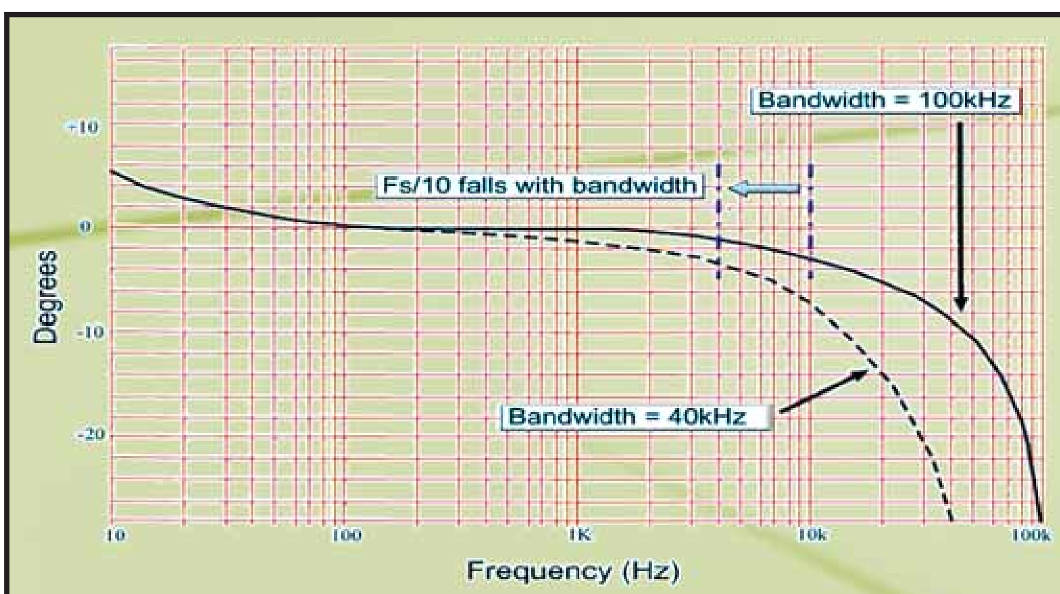

Figure 4. Comparing phase response plots as bandwidth is reduced.