equipment for these groups and concerts were basically offering beefed-up versions of the A-7. The woofer was often replaced with a 100-watt speaker such as the Altec 421. The single high-frequency horn was discarded in favor of a larger multicellular horn designed for wider dispersion, and which was driven by one or two high-powered transducers, such as the 100-watt Altec 290. These high-frequency horns were often packaged separately in their own road cases to facilitate shipping and set-up; again a potential source of driver misalignment. Often, due to lack of time, physical space restrictions, or just plain ignorance, the components of these two- or three-way systems were not set up with their drivers in the proper co-planar relationship. The result was as expected: a loud but not very intelligible concert sound system that was subsequently overdriven in an often futile attempt at clarity.

Recognizing this problem, several sound companies phased out the stackable array in favor of an integrated self-contained system, which again assured a co-planar relationship between components.

One of the first and foremost companies to go in this direction was McCune Sound in San Francisco. Working in 1972 with loudspeaker designer John Meyer, McCune developed the JM-3: a three-way tri-amplified system mounted in a single cabinet containing two 15-woofers, a custom designed mid-range horn and two high-powered tweeters. The mid-range driver was positioned in the same vertical plane as the woofers, and the high-frequency tweeters placed as far back in the cabinet as possible, without interferring with their dispersion pattern. In order to eliminate phase reversal problems in the crossovers, McCune went to a unity-sum type of crossover, which basically reduced the phase shift through the crossover point. Used in conjunction with separate subwoofer cabinets, these JM-3 systems could be arrayed in an arc to provide an evenly dispersed high-quality sound that was controllable and consistent throughout the room.

After a brief hiatus in Switzerland, where he designed his own phase coherent and time-corrected studio monitor speaker, Meyer returned to McCune Sound in 1976, and assisted in the development of a time-corrected, powerful loudspeaker of considerably smaller dimensions than the JM-3. Known as the SM-3, this speaker filled the need for a two-way co-planar loudspeaker that could be used as either a high-powered stage monitor, or as a compact general purpose sound-reinforcement speaker. McCune's chief engineer Bob Cavin found that while it was now possible to align the drivers with a very high degree of accuracy, most people who heard the speaker preferred to have the highs slightly leading the lows, and so it was aligned accordingly. The SM-3 speaker also utilized a summing crossover, and incorporated a driver diaphragm design patented by John Meyer (U.S. Pat. #4152552). The SM-3 is well protected by frequency selective fast-acting peak limiters, which make the unit virtually blow-out proof.

Early 1977 saw development of the SM-4B: a time- and phase-compensated version of the Altec 604. This speaker incorporated active all-pass delay and equalization networks, as well as Meyer's patented diaphragm designs. McCune engineer Steve Kadar engineered the replacement of the standard woofer cone with one capable of handling 150 watts. The SM-4B proved to be very successful in stage monitor use, and also for those difficult PA applications such as center-hung, "theater-in-the-round" systems.

The Altec 604 Family

The original Altec 604, introduced in 1944, soon became a favorite of recording studios that wanted the benefits of a two-way system, but didn't need the projection provided by a horn-loaded cabinet — and they certainly didn't need the size! (Figure 2.) Housed in a ported cabinet, this "co-axial" loudspeaker was aligned along the horizontal rather than vertical axis by positioning the high-frequency driver in the center of the woofer instead of above it. The result was three-fold: reduction in overall cabinet size; maintenance of cohesive imaging of a single-point source; and increased overall power and quality of sound compared to most earlier full-range speakers. Although other speaker manufacturers introduced similar co-axials at about the same time, the Altec 604 was pretty much a recording industry standard here in the States for many years.

As big an improvement as it was, time alignment of these speakers was actually off by between 0.5 and 1 msec., since the high-frequency driver, while in line with the woofer, was actually located behind it. McCune Sound therefore delayed the SM-4B's woofer by 0.57 msec. at 1.75 kHz, using an active all-pass network to bring it into line with the tweeter. Modifications were also done to the tweeter and woofer's support structures, to further reduce inherent phase problems and time abberations.

Figure 2
Altec 604E

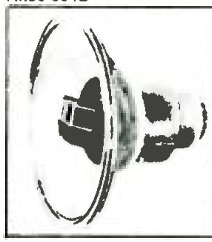


Figure 3
UREI 813

In 1977, UREI introduced a studio monitor speaker that was also built around the Altec 604, but which utilized a crossover with a small amount of built-in passive delay to move the apparent position of the woofer back to that of the tweeter. Known as the UREI 813 Time-Aligned™ Monitor, this speaker was based on an idea proposed and developed by Ed Long, who presented his work on the subject to the AES in 1976. (*A Time-Align Technique for Loudspeaker System Design*, Journal of the Audio Engineering Society, May 1976.) The new speaker, and the recording industry's reaction to it, is largely responsible for the attention being

Problems With Time

At the heart of the problem of time abberations and phase anomalies is time itself. Within the confines and definition of our physical universe, time is a basic operating principle; eliminate time and you can indeed do wonderous things — although probably nobody will notice! (That, however, is the subject of another article entirely.) Suffice it to say that without time there is no concept of distance; without distance there is no concept of movement; and without movement, a speaker cone has no purpose in life.

Fortunately for us, speaker cones do move, albeit not very willingly and usually not very linearly. Take a single full range speaker, for instance. Now if you feed this speaker with a source of full-frequency noise and measure the resultant waveform, you will find that all the individual sinewaves that made up the noise (and that fall within the upper and lower limits of the speaker) will be present in varying degrees of amplitude, depending on the quality and linearity of the speaker. One of the factors affecting response of this speaker has to do with the phase shifts and time anomalies caused by the physical properties of the unit itself. This problem is most easily understood when one looks at the impulse characteristics of the speaker in question.

Back in 1822, J. B. J. Fourier postulated a theory that has proven to be workable up to this point, and which deals quite well with the problem at hand. Germane to this discussion is his finding that an ideal impulse is actually made up of an infinite number of sinewaves, whose positive-going peaks all occur at exactly the same place and point in time. Therefore, a single signal impulse can occur whenever all the frequencies are triggered at exactly the same time. In the next moment some of the sinewaves start going negative (i.e., those at, say, 20 kHz) before others do (at 1 kHz, for example). A complete cancellation occurs at this point due to the vector addition of all these positive- and negative-going sinewaves.

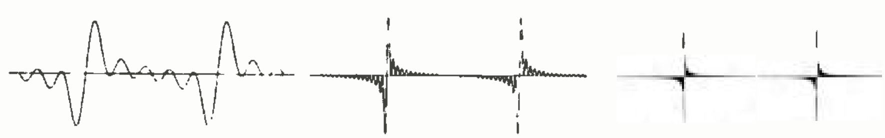
The impulse then stops, only to return again at an interval dependent on the period of the lowest frequency present (so long as all the other frequencies are again going positive at the same point in time).

The figure below is a computer-generated visualization of this phenomena, showing the interaction of up to 100 separate frequencies. It can be seen that the more frequencies involved, the more the resultant waveform approximates a perfect impulse.

Simultaneously triggering an infinite number of sinewave generators is not the only way a pulse can be generated — nor is it the easiest! But it should serve to show that, in theory at least, a perfect pulse contains the entire frequency range, and is the ideal signal to use to test the response of a loudspeaker. Basically, if you know what is going into a speaker you have only to measure what comes out to see where it is deficient.

Now due to physical properties of mass, inertia, and a speaker cone's interaction with the air itself, a loudspeaker will react to high frequencies faster than it will to low ones. Therefore, if all audible frequencies are present in a single impulse to a loudspeaker, the highs will lead the lows once the speaker starts to react to the signal. Hence the need to delay the highs in relation to the lows, in order to accurately reproduce the impulse as it was before.

By splitting a full-range speaker up into a multiway system, and then delaying the various components accordingly, a time corrected system is developed. The only other way to keep the syncronization is to speed up the lows so that the speaker reacts *before* a signal is present — a task that should be only attempted by those of you who are already proficient at eliminating time from your universe.



Computer-generated graphs illustrating the way in which an ideal impulse waveform can be considered as comprising an infinite number of sinewaves whose positive-going peaks all occur at the exactly same point in time. The left-hand trace represents two cycles of five integer frequencies, starting at 1 Hz. As more sinewaves are added — the center trace contains 30 integer frequencies, and the right trace 100 integer frequencies — the resultant waveform assumes the shape of a sharp impulse, whose interval is dependent on the lowest frequency present — in this case 1 Hz.
(Computer-generated graphs courtesy of Alexander Youill-Thornton — "Thorny".)