Loudspeakers

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Outside of recording conditions themselves, there is more variability in loudspeakers than in any other element or process in the sound recording–reproduction chain. No other element has received such wide attention in the pages of the *Journal of the Audio Engineering Society*, and we will assume that the reader is familiar with the basic methods of transduction employed in loudspeakers. Principles have not changed over the years, but attributes and specific goals have. We will look at the major developments of the last fifty years and attempt to point directions for the future.

INTRODUCTION: The centenary of the phonograph record can be divided just about equally into fifty years of mechanoacoustical art and fifty years of electroacoustical art. The introduction of electrical recording in the mid-twenties intersected the electroacoustical transducer development curve at a fairly high point, and there was no need for phonograph reproduction to be concerned with the crude devices that had evolved out of early radio and telephony. Rather, the electrical phonograph could benefit at its outset from the moving-coil direct-radiator loudspeaker developed by Rice and Kellogg in 1925. Curiously, the advent of electrical disc recording did not imply the immediate acceptance of electrical reproduction in the home. Maxfield and Harrison, inventors of the Western Electric disc recording system, said in 1926:

> "Where the power which can be supplied by the record is sufficient to produce the necessary sound intensity, as in the case of home use, it is in general simpler to design one single mechanical transmission system, than it is to add the unnecessary complications of amplifiers, power supply and associated circuits."[1]

This statement, at least on the surface, seems quite naive, but history was on the side of Maxfield and Harrison. The vast majority of home playback sets continued to be acoustical throughout the thirties, and it was only through attrition that they were supplanted by electrical playback systems during the years just prior to World War II. Actually, it was the LP record, introduced in 1948, that spelled the doom of the 78-r/min acoustical phonograph in many parts of the world.

Because playback of records in the home did not necessarily depend upon electronics, the first great emphasis on loudspeaker development did not come from the demands of that art, but rather from the needs of the sound motion picture industry as it developed in the late twenties. The development by Western Electric of high-power loudspeakers during the early years of sound motion pictures, continuing on into the mid-thirties, has left an important legacy scarcely changed up to the present day. In a later section of this paper we will examine their work in some detail.

THE CHALLENGE TO THE DESIGNER

Even experienced transducer engineers and systems designers cannot agree on which attributes and tradeoffs are most desirable in loudspeaker systems. Unlike any other transducer used in the sound recording and reproducing chain, a loudspeaker is called upon to convert substantial amounts of power, and in order to do this it must be fairly large in relation to the shortest wavelengths it must

[1] From Read and Welch, *From Tin Foil to Stereo*, 2nd ed., 1976.

reproduce. Typically, a wide-range loudspeaker system consists of at least three radiating units, each covering a specific portion of a frequency spectrum that may be nearly three decades wide. By necessity, the elements must be displaced spatially, so coherent radiation can be expected only over a fairly small solid angle. This is perhaps the most fundamental design compromise, especially if the designer is thinking in terms of the entire recording–playback chain and the goal of establishing a sound field in the playback environment relating conformally to the sound field present at the time of the original recording. In fact, too much loudspeaker design has taken place in isolation, treating the loudspeaker only as an entity in itself rather than as a subsystem in the recording–reproducing chain.

While it would be fairly easy to specify the attributes of an ideal loudspeaker in an ideal playback environment, it is no simple task to arrive at a consensus among loudspeaker engineers of what a good practical design for home use should be. If engineers have difficulty here, it is not surprising that most amateur, intuitive tinkerers run an even wider gamut in their efforts to come up with new loudspeaker designs. A survey of acoustical patent reviews in the United States will show that the attributes of unique and novel are not necessarily consonant with the laws of physics. It is a testament to the forgiving nature of the ear that many of these designs sound as good as they do.

Perhaps the best way to survey the broad aspects of loudspeaker design and practice is to do it historically. The ordering of attributes has changed many times over the years, and there have been distinct epochs that we might study in our assessment of the present, and future, states of the art.

A SHORT HISTORICAL REVIEW

As we said earlier, the demands of the talking motion picture provided the impetus for the first rapid development of quality loudspeaker systems. The efforts culminated in the monumental Bell Laboratories demonstration in 1933 of three-channel stereophonic long-line transmission between Philadelphia and Washington, D.C. The loudspeakers used for this study were developed by Wente and Thuras. Both high- and low-frequency units were horn loaded and were designed to yield maximum sensitivity over a controlled solid radiation angle with limited signal power input. In the mid-thirties, 10 watts of audio power was a practical upper limit, and these systems could deliver goodly acoustical levels with only that small amount of power. Typically, these systems produced sound-pressure levels (re 1 watt, 1 meter) of about 105–107 dB. Over the midband, these systems exhibited a directivity index of about 7–8, and the corresponding conversion efficiency was in the neighborhood of 15–25%. Loudspeakers such as these were large; the low-frequency assemblies were about 3.5 meters high, 2.5 meters wide, and perhaps 1 meter deep. Today, essentially the same devices are used in motion picture theaters, for general large-scale sound reinforcement, and, of course, at rock concerts. Physical parameters have not changed, but better magnet structures and higher temperature cements and materials have permitted these devices to handle much more power in today's applications with much more acoustical output.

The Western Electric legacy, as it may be accurately termed, carries over into one very important area of present high-fidelity practice, recording-studio monitoring. Gone are the large low-frequency horn assemblies; they have been replaced by single or dual 38-cm low-frequency units, usually in vented enclosures, along with horn-loaded medium-frequency units crossing over in the 500–1000-Hz range, and horn-loaded high-frequency units carrying the spectrum on from 2 to 7 kHz upward. Current practive has purposely cut the low-frequency unit sensitivity back by adding mass to the moving system in order to extend the low-frequency limit down to 30–35 Hz. (The earlier theater systems did not extend much below 50 Hz.) The routine availability of high audio power these days has made this a reasonable tradeoff, and current monitor systems typically exhibit sensitivities of 92–93 dB sound-pressure level (re 1 watt, 1 meter). Because of increased power handling due to the higher temperature stability of modern construction, the new monitor systems can easily reproduce levels in the normal control room exceeding 115 dB sound-pressure level and in substantially smaller enclosures than those of their precursors. This degree of high-level performance may be essential for modern pop/rock recording, but it is not necessary for reproducing music at realistic levels in the home.

A small revolution occurred in the mid-fifties, and it centered about the so-called acoustic suspension low-frequency loudspeaker system. Essentially, this is a low-frequency transducer mounted in a sealed enclosure small enough so that the stiffness of the moving system is due more to the enclosed air than to the characteristics of the loudspeaker spider or surround, which are usually made very compliant. These systems can provide excellent bass response in small enclosures because system resonances can be held to the 40-Hz range. For the first time, the bookshelf loudspeaker became respectable.

These systems were relatively inefficient, but their introduction coincided with a trend toward high ratings in power amplifiers. Typically, the sensitivity of an acoustic suspension system is in the range of 85–87 dB sound-pressure level (re 1 watt, 1 meter).

The fifties also saw the commercial introduction of practical electrostatic loudspeaker designs. These consisted of large rectangular panels, perhaps 2 meters high by 0.75 meters wide, and are invariably of "push–pull" design. The electrostatic loudspeaker is a dipole radiator; its polar pattern is "figure-eight," with the back radiation characteristically out of phase with the front. Further, because it is such a large radiator, the listener by necessity is located in the near field. The better electrostatic devices are characterized by a certain effortlessness of sound, and seem to convey a sense of spaciousness. They are inherently of low sensitivity and cannot be used successfully for high-level reproduction in the home. They have their counterparts in planar radiating systems of recent design using a printed "voice coil" on a large plastic membrane operating in the proximity of small high-energy magnets.

The sixties saw more emphasis on rational design techniques. The advancing state of the computer art made it possible to model many loudspeaker and dividing network components and their associated enclosures, observing the effects of changes in individual parameters in the design. Work in this area has continued up to the present, with ever increasing dividends for the manufacturer and relief from so many of the "cut and try" design practices of the past.

RELEVANT MEASUREMENTS AND DIRECTIONS FOR THE FUTURE

As we move well into the seventies—and into the second hundred years of the phonograph—we see significant improvements in loudspeaker design as audio engineers begin to pay closer attention to time-domain relationships between transducers. The problem is not new; in the early sound motion picture art there could be significant time delays between low-frequency enclosures and their associated high-frequency horn assemblies. But once these were adjusted to within a handful of milliseconds, the problem was believed solved.

In recent years, as recordings have improved so dramatically and as electronics design engineers have begun to isolate and remove subtle forms of distortion, the niceties of small errors in time-domain accuracy in loudspeakers become more important.

Measurement of the phase response of loudspeakers completes the overall transfer characteristic, which was viewed for so many years as only the amplitude portion of a complex quantity. Actually, the important aspect is not the phase response but rather its derivative with respect to frequency, the group-delay characteristic. With conventional instrumentation, measurements of complex transfer characteristics are difficult, and in recent years discrete Fourier transform methods have simplified the basis for analysis and given the engineer far more to work with. Using these techniques, a loudspeaker is energized with a series of pulses of known frequency distribution. These are observed and transformed into continuous plots of amplitude and phase, or group delay. Further transformations can isolate the minimum-phase part of the response from the total phase response, thus giving the designer a good idea of where his remaining design problems are.

The design of time-aligned systems demands better components than were customary in past years. Rapidly disappearing in serious design work are the highly colored "peaky" devices that were once the staple of the industry. A typical cone transducer has a useful frequency range over which, with a little attention, it will be well behaved. The low-frequency limit is usually established by its main resonance, while the upper limit is established by the tendency of the device to go into complex vibration or to narrow its radiation angle.

This latter qualification is important if the designer wishes to maintain fairly constant dispersion from the system, and proper concern for this often leads to the design of four-way systems. Certainly no less than three components can yield reasonably constant dispersion across the frequency band. Practical considerations usually come into play about 10 kHz, and designers are usually willing to let the response narrow somewhat as the high-frequency radiating surface becomes large with respect to the radiated wavelength.

Another area of loudspeaker performance is finally receiving the attention it deserves, its polar, or directional, response. Many loudspeaker systems are made up of an array of components, and the directional characteristics, the sum of all the radiators in a given direction, can be quite complex. Conventional measurement methods using band-limited noise average out many of the details in this "fine structure" of the systems and often give the impression that all is well, or at least sufficiently smooth. But it may not be so; the ear is sensitive to unevenness in polar response even when the energy, integrated over fairly narrow bands of noise, is reasonably constant. The effect on the experienced listener is one of decreased depth perception in the stereo array and a lesser sense of realism when compared with systems smoother in this respect.

In general there is a tendency, at least in the audiophile sphere, to go to lossier devices in an attempt to get smoother response. Many of the techniques favored for efficiency's sake can lead to resonant and peaky response, and as we have stated before, the tradeoff relates naturally to the application.

Distortion due to system nonlinearities is an important consideration in loudspeaker design. Surprisingly, large-signal nonlinearities, those due to exceeding fundamental linear limits, may not be as aggravating to the listener as various buzzes and noises due to break-up phenomena, audible often at low and moderate levels. Much development work therefore is being done in the materials area in an effort to isolate more stable and better cone and diaphragm materials from those tending to rattle or go into complex modes of vibration. Again, we are concerned with the fine structure of loudspeakers and are faced with the observation that two devices, measuring nearly identically by conventional techniques and differing only in cone material, may sound quite different, even to naive and inexperienced listeners. A typical measurement of fundamental, second, and third-harmonic response versus frequency for a loudspeaker will usually exhibit very choppy and erratic distortion curves, even when the fundamental is fairly smooth. Much of what is being plotted is simply spurious buzzes and noises filtered by the wave analyzer portion of the instrumentation.

THE ELECTRICAL-MECHANICAL INTERFACE

Most loudspeaker systems are designed to operate with a wide variety of amplifiers and with flat electrical input. It is further usually assumed that the amplifier has a negligible internal impedance and that it is effectively a constant-voltage device. Over the years a number of complex systems have been designed that view the amplifier–loudspeaker combination as a single system. Once this association is made, there are some new benefits. Biamplification (or triamplification) can be used to power each transducer with its own amplifier, selected for the optimum power level, and distortion at high signal levels is reduced in the process. Motional feedback may be employed for greater large-signal linearity in low-frequency devices. The design of dividing networks becomes easier to implement due to the high impedance levels at the amplifier inputs, and minimum-phase designs can be realized accurately and conveniently. Finally, moderate degrees of amplitude equalization may be incorporated into the overall design to facilitate loudspeaker–listening room integration.

OUTLOOK FOR THE FUTURE

There are not many major breakthroughs left in loudspeaker design; most of these were uncovered in the early days of the art. What will continue of course is the reordering of attributes and priorities as loudspeaker design keeps pace with development in the associated areas of the recording and reproducing chain. As each segment of the art improves, fine differences begin to be discernible between like components where none could be heard previously. And any difference at all implies that at least one of the components is less than ideal. Much work remains in developing objective judgments. The ear of a trained listener can detect subtle differences between amplifiers, for example, which cannot be measured by current testing methods, but fortunately the gap appears to be narrowing between cause and effect in this regard.

In the future more attention will be given to the listening environment itself, optimizing it for the job at hand. During the last six or seven years, the industry has seen the first step taken toward proper and consistent interfacing of monitor systems into recording control rooms, and this is a direction which serious home listening will take.