Loudspeakers exposed

With all the remaining components of an audio chain being increasingly refined, John Watkinson argues that the loudspeaker has become the limiting factor in audio quality. Here, he looks at the problems and presents solutions.

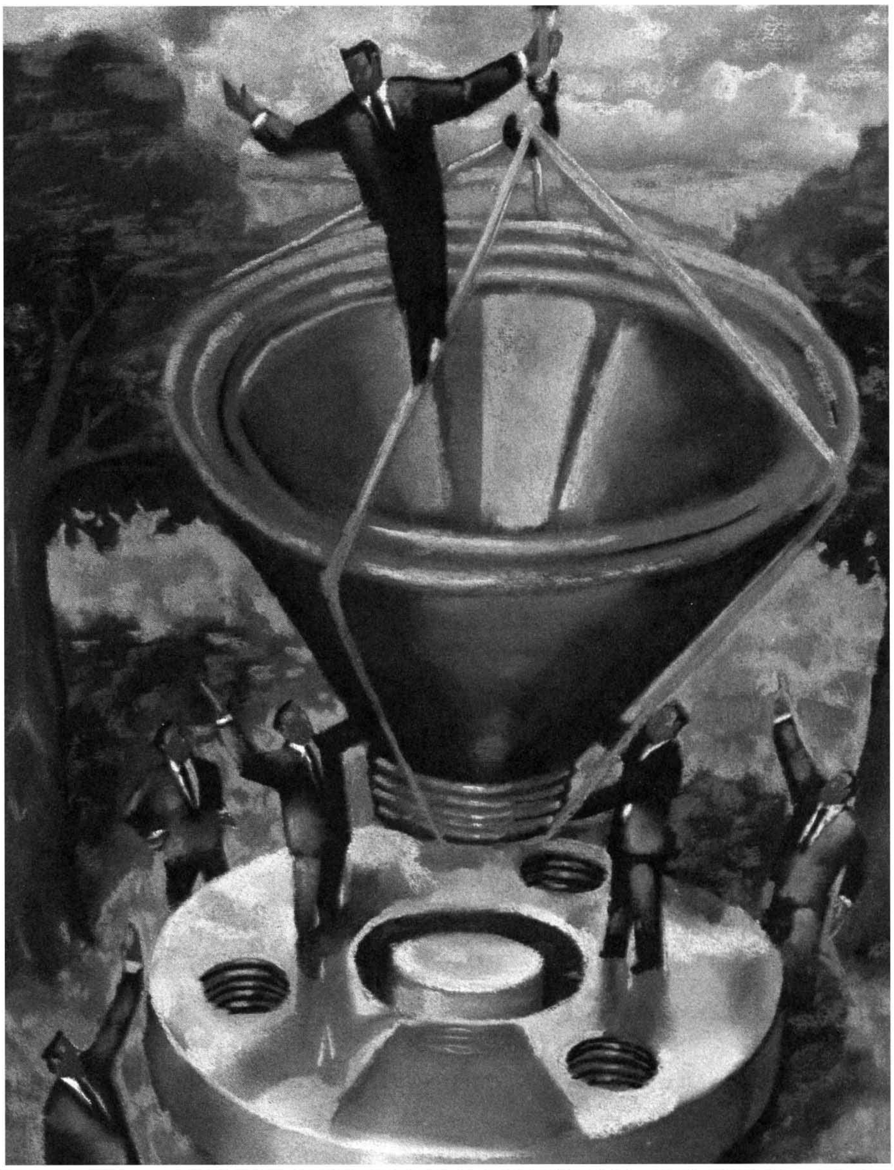

ILLUSTRATION JAMEL AKIB

Until recently, virtually every component in the audio chain was capable of causing audible impairment. If high quality was the goal a degree of determination and plenty of time was needed to adjust equipment to a finely balanced point faster than it drifted. Because things were never good enough there was a consistent research effort and this has given tangible results.

In a typical modern audio system, a microphone feeds an a-to-d converter, connected to a digital recorder, driving a d-to-a converter, a power amplifier and a loudspeaker. At some point a mixing console may be found. The weakest link determines the overall quality.

Modern microphones have an extremely flat frequency response, and adequate dynamic range and linearity. Modern converters using noise shaping and oversampling with 18 and 20-bit resolution are outperforming our ears, – provided some attention is given to clock jitter.

If a digital audio recorder uses digital i/o, then provided it doesn't use compression, it doesn't have a sound quality. Numbers coming in are the same as numbers going out. High quality modern mixing consoles have reached a stage where they are virtually transparent. Power amplifiers have reached a state where further developments will be in the area of efficiency and the friendliness of the load presented to the ac supply.

Most of the quality loss in a modern sound reproduction system is due to the loudspeakers, which for some reason have not seen the development of other components. In my opinion loudspeakers are now causing a quality bottleneck. Such an area is ripe for research because for a given effort the rewards will be more significant in comparison with more mature technologies where the returns diminish as the ideal is approached.

I should stress that I am interested in precise sound reproduction rather than in hi-fi. There was a time when the two were synonymous, but nowadays in many respects hi-fi has become a religion in which beliefs are more important than truths and enthusiasm replaces knowledge. The temples of hi-fi are the phenomenally expensive hardware installations and the high priests are journalists who find pseudo-scientific reasons to make the believers feel comfortable with the vast sums they have spent.

The laws of physics involved in audio reproduction are established beyond any shadow of a doubt yet they are regularly called into question by hi-fi journalists whose ejaculations usually serve only to raise the noise floor for the genuine researcher.

It is impossible to make other than accidental or empirical progress without a clear picture of the processes involved and an understanding of the key criteria. To determine what part of one's knowledge base can be trusted it is necessary to remove from it all of the myths and pseudo science and to establish what is and is not the case. It is surprising how long this takes if one is to be impartial and scientific about every spurious theory.

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Without a knowledge of psycho-acoustics it is impossible to assess the relative merits of differing approaches. The human hearing system is complex and highly sensitive in some areas, yet surprisingly casual in other areas. If this is understood, precision will be placed in areas of sensitivity, whilst shortcomings can be mitigated by placing them in other areas.

As audio systems are designed for human listeners, the criteria for audio quality can only be subjective. Audio systems form a window between the listener and the original sound. All that is necessary is to make that window larger than the sound passing through it in all respects. If human listeners are unable to detect an impairment, then the quality is sufficient and the window is big enough. Making it even bigger simply drives up the cost.

Listening tests are vital once all objective tests have been passed, but in order to be significant, such tests have to be properly conducted to avoid bias. I can listen to a loudspeaker as well as anyone, but unlike many, I do not consider myself competent to do so alone. This is simply because the spread of human hearing performance is so great that I cannot be truly representative. I will naturally listen to my own designs more favourably than those of others.

In a significant listening test, neither the operator nor the subjects must be aware of the reason for the tests, and the design of the tests must be approved by a statistician who can determine how likely it is that identical results could have been obtained by chance. I can only listen to a loudspeaker of my own design to ensure that it has no obvious warts, but to compare it meaningfully with another speaker of similar performance is beyond any individual.

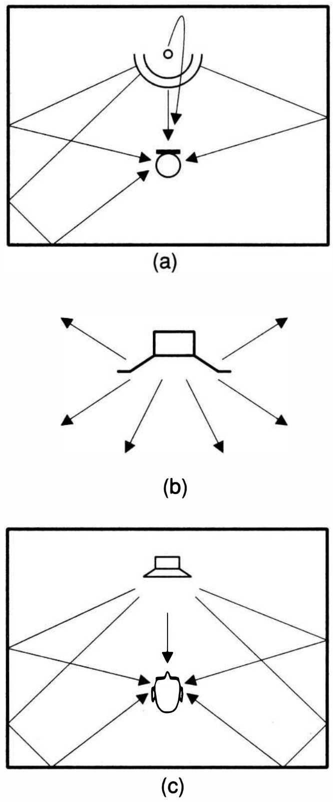

Fig. 1a) Sound approaches microphone from many directions due to ambience and reverberation. b) In anechoic conditions a single loudspeaker produces exactly the opposite of a). c) Loudspeaker in reverberant conditions simulates situation of a) at listener's ears.

The ideal

An ideal speaker might be one which was a sphere whose volume changed according to the input waveform. Such a device would behave as an ideal point source, having frequency independent dispersion and a frequency response like a ruler. What is more it would be perfectly linear and would not exhibit energy storage, which would also make it perfectly phase linear.

Some of these consequences bear explanation. A pulsating sphere acts as a point source because wherever one stands, the part of the surface nearest is moving directly towards and away from one. All points on the surface move in the same phase, therefore there can be no vibrations propagating across the surface of the sphere. Consequently there is no requirement to suppress such vibrations. Radiation cannot occur after the input ceases. If the output stops when the input stops, the system is phase linear.

A good microphone produces an accurate version of sounds approaching it from many directions. Even if a loudspeaker reproduced the microphone waveform exactly, the resulting sound is leaving in many directions. Spatially, a single loudspeaker is producing sound travelling in exactly the opposite direction to the original. Consequently reproduction of the original sound field is simply not possible.

Figure 1 shows the problem. Sound approaching a microphone at **a)** does so from a multiplicity of sources whereas sound leaving a single loudspeaker superimposes all of these sources into one. Consequently a monophonic or single loudspeaker is doomed to condense every sound source and its reverberation to a single point.

When listening in anechoic conditions **b)** this is exactly what happens. While the waveform might be reproduced with great precision, the spatial characteristics of such a sound are quite wrong.

However, when listening in a room having a degree of reverberation, a better result is achieved irrespective of the reverberation content of the signal. The reverberation in the mono signal has only time delay and no spatial characteristics whatsoever whereas the reverberation in the listening room has true spatial characteristics. The human listener is accustomed to ambient sound approaching from all directions in real life and when this does not happen in a reproduction system the result is unsatisfactory.

Thus in all real listening environments a considerable amount of reverberant sound is required in addition to the direct sound from the loudspeakers. Figure **1c)** shows that the reverberation of the listening room results in sound approaching the listener from all sides giving a closer approximation to the situation in a). Clearly better reverberation will be obtained when the loudspeaker is out in clear space in the room. So-called bookcase loud-

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speakers mounted on walls or shelves can never give good results.

Better spatial accuracy requires more channels and more loudspeakers. While the ideal requires an infinite number of loudspeakers, with care, as few as two speakers can give a convincing spatial illusion. The improvement in spatial performance using two speakers is enormous. Tests have shown that most people prefer stereo with poor bandwidth and significant distortion to pristine mono.

Two speakers can only give spatial accuracy for sound sources located between them. Reverberation in the listening room then provides ambient sound from all remaining directions. Clearly the resultant reverberant sound field can never be a replica of that at the microphone, but a plausible substitute is essential for realism and its absence results in an unsatisfactory result. This renders the traditional use of heavily damped rooms for monitoring suspect.

If realism is to be achieved, the polar diagram of the loudspeaker and its stability with frequency are extremely important. A common shortcoming with most drive units is that output becomes more directional with increasing frequency. **Fig. 2a)** shows that although the frequency response on-axis may be ruler flat giving a good quality direct sound, the frequency response off-axis may be quite badly impaired as at **b)**. In the case of a multiple drive unit speaker, if the crossover frequency is too high, the low-frequency unit will have started beaming before it crosses over to the tweeter which widens the directivity again.

The figure shows that the off-axis response is then highly irregular. As the off-axis output excites the essential reverberant field the tonal balance of the reverberation will not match that of the direct sound. The skilled listener can determine the crossover frequency, which by definition ought not to be possible in a good loudspeaker.

The resultant conflict between on- and off-axis tonality may only be perceived subconsciously and cause 'listening fatigue', where the initial impression of the loudspeaker is quite good but after a while one starts looking for excuses to stop listening.

The hallmark of a good loudspeaker installation is that one can listen to it indefinitely. Unfortunately such instances are rare. More often loudspeakers are used having such poor off-axis frequency response that the only remedy is to make the room highly absorbent so that the off-axis sound never reaches the listener. This has led to the well-established myth that reflections are bad and that extensive treatment to make a room dead is necessary for good monitoring. This approach has no psychoacoustic basis and has simply evolved as a practical way of using loudspeakers having poor directivity.

The problem is compounded by the fact that an absorbent room requires more sound power to obtain a given sound-pressure level. Consequently heavily treated rooms require high-power loudspeakers which have high distortion and often further sacrifice polar response in order to achieve that high power.

A conventional box shaped loudspeaker with drive units in the front will suffer extensive shading of the radiation to the rear and thus will create a coloured reverberant field. Clearly a much more effective way of exciting reverberation with an accurate tonal balance is for the loudspeaker to emit sound to the rear as well as to the front. This is the advantage of the dipole loudspeaker which has a figure-of-eight polar diagram.

Loudspeakers have also been seen with additional drive units facing upwards in order to improve the balance between direct and reverberant sound. These techniques work well but obviously in a dead room are a waste of time as the additional radiation will never reach the listener. The fault is in the room, not the speaker.

Air is not very dense. As a result it is not possible to influence very much mass at once. Thus it is difficult to radiate energy into air with a mechanical device because the mass of the moving part of that device will eclipse the mass of air influenced. In engineering terms a diaphragm has a high mechanical impedance but the air has a low impedance, resulting in a mismatch, meaning that loudspeakers will always be inefficient. With the almost limitless power from modern amplifiers this is a minor problem.

As an alternative the horn loudspeaker is a kind of acoustic transformer which raises the impedance of the air adjacent to the diaphragm in order to improve the power transfer. Unfortunately, acoustic transformers are difficult to make linear and the resulting distortion is difficult to eliminate.

A great problem with loudspeaker design is the span of wavelengths involved. These range

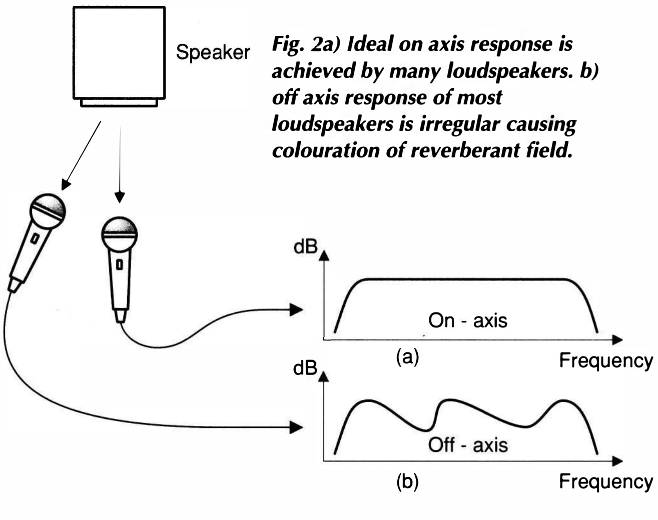

Fig. 2a) Ideal on axis response is achieved by many loudspeakers. b) off axis response of most loudspeakers is irregular causing colouration of reverberant field.

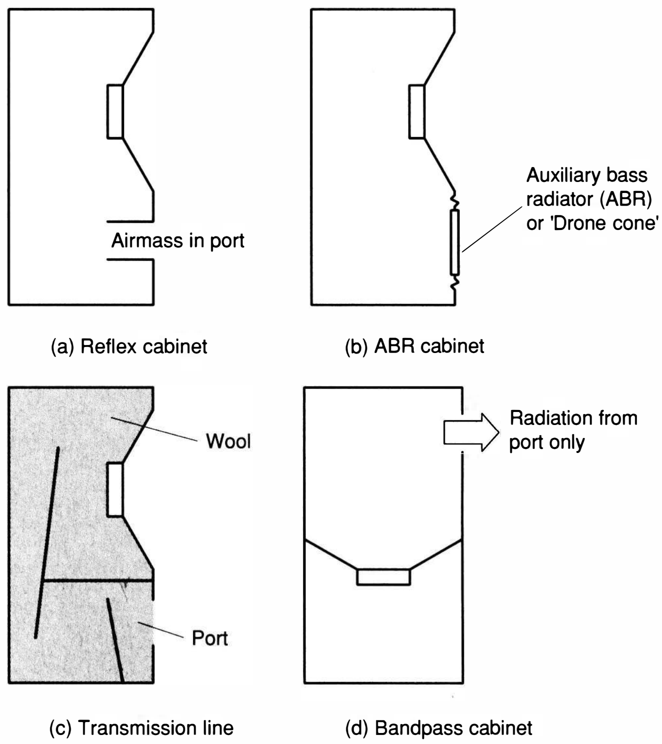

Fig. 3. Various attempts to reproduce low frequencies. a) mass of air in reflex duct resonates with air spring in box. b) air mass replaced by undriven diaphragm or auxiliary bass radiator. c) rear wave is phase shifted 180° in transmission line to augment front radiation. d) bandpass enclosure puts drive unit between two resonating chambers. None of these techniques can properly reproduce transients and active techniques have rendered them obsolete.

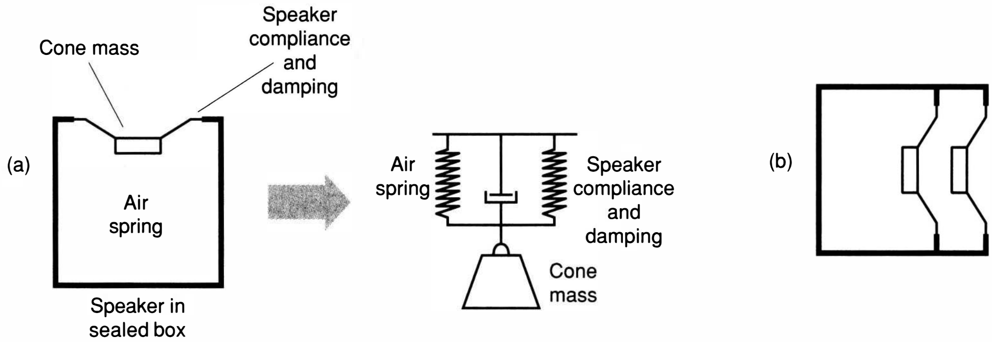

Fig. 4a) Sealed enclosure forms a non-linear air spring in parallel with driver compliance. This stiffens the compliance and raises the fundamental resonance. b) isobaric or compound woofer has tandem diaphragms.

from a few millimetres at the highest audible frequency to several metres at the lowest. There cannot be many disciplines in which mechanical motion is required over such an octave range.

Wave theory is dominated by the relative sizes of the source and the wavelength. Thus in a loudspeaker at the highest frequencies the transducer is much larger than the wavelength, whereas at the lowest frequencies it is much smaller. As a practical matter it is necessary to use more than one drive unit with a crossover network.

Reproducing low frequencies

In order to allow a diaphragm to generate low frequencies, it must be provided with an enclosure which prevents an acoustic short circuit. Provided the wavelength is larger than the enclosure, the resulting radiation will be omnidirectional and the result will be exactly the same as if a pulsating sphere had been used.

The lowest frequency to be reproduced is debatable and depends upon the material to be reproduced. If we want to be able to reproduce all musical instruments, we have to include the organ. Organ pedal notes don't start to be realistic unless a response is maintained to around 20Hz. At this frequency you do some of your listening with your chest – even at moderate sound-pressure levels. Low-frequency roll-off is unavoidable, but it must be monotonic and preferably have a slope of no more than 12dB/octave.

Most loudspeakers cannot faithfully reproduce the input waveform at low frequencies, but unless this is done, a loudspeaker is simply not accurate enough. An obvious example is the transient when an organ pipe begins to speak or stops speaking. The sound is distinctive and a good loudspeaker should reproduce it – but most don't. Further examples include marimbas and other bass percussion instruments like hollow logs.

Many loudspeakers employ resonances to obtain an extended frequency response in the mistaken belief that only steady state frequency response is important, **Fig. 3**. By definition, resonance works by storing energy. This energy is taken from the leading edge of a bass transient and added to the trailing edge. Again by definition a tuned loudspeaker cannot be phase linear. Consequently transient edges are blurred and unrealistic and arrive out of time with the treble energy. The correct term is linear distortion.

Therefore reflex loading, the auxiliary bass radiator and its more recent relative the bandpass enclosure, are unacceptable on fidelity grounds. These all achieve a lower frequency steady state response by destroying the waveform of bass transients. They have a steeper roll-off below resonance which is unnatural. The transmission line loudspeaker fails because there is an assumption that a phase shift in the line is as good as an inversion. Again this is unfortunately only true on continuous sinewave.

Reflex, auxiliary bass radiator, transmission line and bandpass enclosures are all traditional approaches which were the best that could be done with the simple electronics of the day. The active loudspeaker, which can easily be made phase linear, renders all of these approaches obsolete except for economy or to get high sound-pressure with old fashioned magnet technology. The only published techniques which do not violate the ideal are the sealed enclosure and its relative the isobaric. Untuned loudspeakers which do not store energy are essential for high fidelity because they can be made phase linear.

With a traditional approach to the sealed enclosure, the optimal reproduction of low frequencies requires a physically large loudspeaker. The mass of the diaphragm and the stiffness of the air in the enclosure behind it form a resonant system, as **Fig. 4** shows. Below resonance there is little output and so the lower the resonant frequency the better.

The smaller the cabinet, the higher the stiffness of the air within, and the higher the fundamental resonance. Also the internal pressures generated rise with small cabinets, resulting in a large force on the diaphragm and an increased likelihood of breakup. This is where the isobaric configuration scores by isolating the outer driver from the enclosure pressure.

> ***"While it is well known in engineering that pressure containment vessels should be cylindrical or spherical, loudspeaker designers cling to the rectangular box."***

The resonant frequency can be lowered by raising the diaphragm mass, but that reduces the efficiency too, causing a coil dissipation problem. The force on the diaphragm can be reduced by using a smaller diameter, but then the throw has to be increased, increasing distortion. Thus if a good low-frequency response and low distortion is required at reasonable sound-pressure levels, the traditional loudspeaker has to be large.

In strictly theoretical terms, a low-frequency loudspeaker only needs to be able to displace a sufficient volume of air to achieve the required sound-pressure level, and this has nothing to do with its enclosure volume. Thus in principle at least, a small low-frequency loudspeaker is possible, but this will not be based on the conventional approach and it will not be passive. With active techniques the motion of the diaphragm and its apparent resonant frequency are under the control of the amplifier designer.

Clearly a loudspeaker cabinet must be totally inert. As the interior of the cabinet is driven by a secondhand signal from the back of the drive units, there is no way that this can be allowed to radiate. As the area of the enclosure walls eclipses the area of the diaphragm, even small enclosure vibrations can have a serious effect on clarity.

While it is well known in engineering that pressure containment vessels should be cylindrical or spherical, loudspeaker designers cling to the rectangular box. The flat panels of a box are easy for carpenters to assemble, and eliminate the need for spending money on industrial design. But from any acoustic standpoint, they are inadequate. Has anyone ever seen a square submarine or a rectangular aerosol can?

Reproduction at higher frequencies

When a plane diaphragm transducer is much larger than the wavelength, it tends to produce plane waves which are directional. In the case of an unenclosed diaphragm, a bipolar response is achieved in which the front and rear radiations are identical but anti-phase.

Directionality rises with frequency and the result is that the highest frequencies can only be discerned directly on axis. As has been seen, this result is unacceptable and in a well engineered tweeter steps must be taken to avoid it.

At high frequencies, the cone acts as a mechanical transmission line for vibrations which start at the coil former and work out-

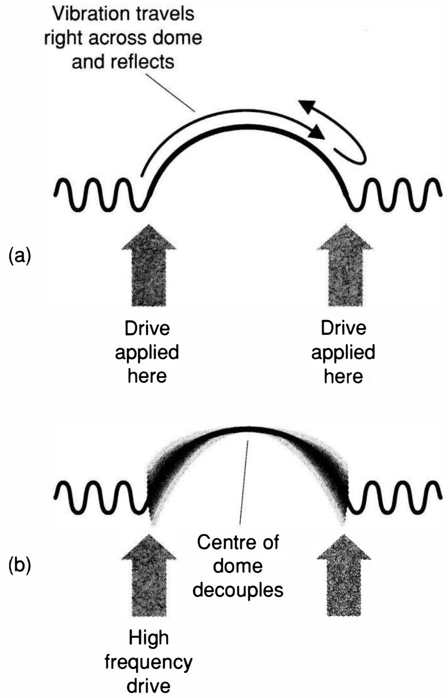

Fig. 5a) In a rigid dome there is nothing to stop vibrations travelling right across the apex and being reflected. b) At high frequencies, the centre of the dome decouples giving exactly the wrong characteristic for good directivity.

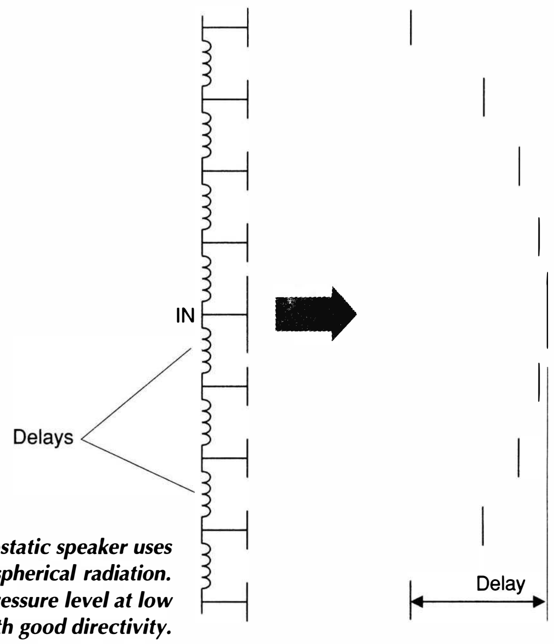

Fig. 6. Phased array electrostatic speaker uses delay lines to simulate spherical radiation. Can achieve high sound-pressure level at low distortion with good directivity.

wards. It is possible to introduce frequency dependent loss into the transmission line so that the higher the frequency the smaller is the area of the cone which radiates. Done correctly this yields a constant dispersion drive unit which simulates a sector of our ideal pulsating sphere.

The main concern is that there are vibrations travelling out across the surface and there must be a cone surround which acts as a matched terminator so that there can be no reflections.

If you consider the popular dome driver, to the casual observer it looks like a section of a sphere and should therefore be close to the ideal. Unfortunately, as has been pointed out many times in the literature, this is a myth. The dome moves on a single axis, and this is not the same thing at all as a pulsating sphere. Domes cannot be rigid, and so the vibrations from the coil must propagate inwards from the circumference to the apex. This causes two problems as shown in **Fig. 5**.

> ***"The passive loudspeaker has so many flaws that it is difficult to know where to begin."***

First, when the vibrations arrive at the apex, there is nothing to terminate them, so they must continue on until they arrive back at the coil. Consequently rigid domes must suffer from energy storage and hangover.

The alternative is to use a 'soft dome' which is lossy. In this approach, losses in the dome mean that the amplitude of vibration falls towards the centre. This is the exact opposite of what is wanted for good dispersion. Consequently domes can only work over a narrow frequency range and need to cross over to smaller units at frequencies where a transparent crossover cannot be achieved. As I showed earlier, this causes the directivity index to resemble a dog's hind leg. While the on-axis response may be flat at the sweet spot, the reverberant field will be extremely non-uniform.

From the theoretical standpoint, the dome has no acoustic merit. The practical advantage of the dome is that it can be fitted with an immense coil which can dissipate a lot of power without cremating itself.

In the electrostatic loudspeaker, the diaphragm does not need to be rigid because it is driven uniformly. As a result it can be lighter with corresponding benefits in efficiency, phase linearity, transient response and freedom from intermodulation distortion.

The electrostatic diaphragm is supported between two driving plates and the spacing is a compromise between the amplitude of motion possible and the drive voltage needed. They are invariably used in bipolar mode without a cabinet. While this is advantageous for exciting the reverberant field, it means that they suffer an acoustic low-frequency roll-off and are best used in conjunction with a linear phase woofer.

A large, flat, uniformly moving diaphragm beams dreadfully at high frequencies. The elegant solution of the *Quad 63* was to make the mechanically flat diaphragm behave like a sphere by splitting the electrode structure into concentric rings fed by lossy delay lines, as shown in **Fig. 6**. The outward propagation of vibrations across the diaphragm again allow a close simulation of a sector of the ideal pulsating sphere.

Again matched termination at the perimeter prevents reflections. Unfortunately when the Quad was designed, it was simply not possible to produce a woofer of matching quality and a full range electrostatic design having restricted sound-pressure levels was inevitable. With modern active woofer technology these restrictions no longer apply.

With a phase array electrostatic transducer used from the low midrange upwards it is possible to get staggering sound-pressure levels because of the sheer volume velocity available, but without sacrificing the low distortion and near ideal dispersion. Moving coil designs simply cannot reach these low distortion figures.

Loudspeaker electronics

One approach to improving loudspeakers is to treat the amplification, crossover and transducer stages as part of a single system having an overall transfer function. When this is done, a great many new avenues open. The tradition of building general purpose amplifiers which are remotely sited from passive loudspeakers built by someone else has nothing to recommend it.

The passive loudspeaker has so many flaws that it is difficult to know where to begin. The low-frequency response of a passive speaker is determined by the mechanical parameters and not by the control system and will be inferior for a given enclosure size.

It is intuitively obvious that the two outputs from a crossover network should sum to produce the original signal. Unfortunately in a passive crossover this requirement simply cannot be met. Having heavy woofer currents and their distortion products flowing in the same wiring as the tweeter drive, as a passive speaker does, is asking for trouble. One engineering tenet which is seldom broken with impunity is to put the power source near the load.

The only accurate solution is to use one power amplifier per transducer with the crossover function performed at signal level prior to the amplifiers. Power amplifiers are so cheap today that there is little excuse for any other approach.

Another advantage of integrating the amplifiers into the loudspeaker is that the endless and boring mythology of loudspeaker cable audibility is neatly sidestepped.

The future?

The traditional loudspeaker is so flawed that for high quality applications, the end of the road has been reached. Although countless learned papers have appeared pointing out the flaws of traditional speaker design, which are encapsulated in **Fig. 7**, there has been little reaction from traditional manufacturers who either lack the vision to see the future or who lack the wide range of skills needed to put the vision into practice.

In the future, the highground of precision sound reproduction will be captured by active loudspeakers whose design is based on a deep understanding of engineering, acoustics and psychoacoustics.

At a technological disjuncture of this kind, where an old technology is being replaced by new, the opportunity arises for entirely new companies to emerge and capture the market while the traditional suppliers do ostrich impersonations.