EAR TO THE GROUND

A NEW QUAD ELECTROSTATIC? ASKS BASIL LANE

IN JUNE 1979, at a meeting of the Audio Engineering Society, British Section, Peter Walker of Acoustical Manufacturing (QUAD) presented details of a new form of prototype electrostatic loudspeaker. That his work was considered to be of great importance was demonstrated by the unusually large audience.

The design is not yet ready for commercial release, nevertheless the lecture proved to be so interesting that a full report is given here.

As far as possible an attempt has been made to quote his speech *ad verbatim* since his explanation could not be further simplified without loss of detail.

Famous face, famous pesonality, Peter Walker of QUAD.

EVERY loudspeaker designer has looked at electrostatics' what a nice way of doing things, if only it would work! When a moving-coil speaker is made, you start with a lot of force on a speech coil and to match that to the load presented by the air, you need a large horn — and then it can be very good indeed. But an electrostatic speaker, which is a very thin sheet of comparatively large area with a small force on it, is already matched to the air load as if it were at the open end of the horn. In effect, there is no matching problem. Also, cones which are rather difficult to keep working over a wide bandwidth, are unnecessary. Electrostatic principles therefore, have their appeal, but also their problems.

If an electrostatic is made large enough to be loud at low frequencies, an octave higher in the audio band it begins to become too directional. Efficiency problems arise which have been solved by the use of a separate bass unit and mid/high frequency unit. Other devices have been thought of, including acoustic lenses, curved surfaces or even long strips, but all of these take something away from the basic simplicity of the device. After all, a simple electrostatic element is a very predictable thing and we know exactly what it can or cannot do.

When complications are added, such as curved surfaces, lenses and so on, it also happens that the acoustic results are also more difficult to predict. Tonight I hope to show a better way of producing an electrostatic speaker in which very nearly the whole performance is encoded in its electrical drive currents.

The point of this is that if the design can be accomplished using electrical terms to predict acoustic performance, then those same electrical parameters can be manoeuvred, to give a boost at high frequencies, or to provide delays, or to alter directivity. All this can be done with the sure knowledge that the acoustic performance will follow exactly the modifications to the current.

Let us start with the very simple and popular concept of a sound source as a pulsating sphere. Perhaps it can be imagined as a large beachball, the surface of which pulsates in and out over its whole surface. Such a sound source could be regarded as the equivalent of an omni-directional microphone.

Alternatively, the sphere could be made to vibrate to and fro along a single axis, the whole thing moving together. The result of this would be a directivity pattern in the form of a doublet, or figure of eight. It is the speaker equivalent of a ribbon microphone. Following that, one could imagine a sphere that is both pulsating and also vibrating along one axis. If the proportions of these two motions were adjusted correctly, the result would be a speaker having a cardioid directivity pattern. Of course, it is impossible to make any of these things for the obvious reasons that controlling the whole surface of a sphere accurately is beyond our engineering capability. Even if you were able to drive the whole surface correctly, the internal interference patterns would upset the results.

This is why, to control the directivity of a speaker over a wide band. Moving-coil systems use three units, one large in a large box, a smaller one in a smaller enclosure and a very small one on top. Of course, the small unit can come very close to perfectly reproduce the model I have described, but it is not powerful enough to reproduce low frequencies well.

It is my opinion that we pay a price for this complexity, it is not easy to add multiple units together to obtain a homogeneous sound. As an alternative, what we can do is to *imagine* the pulsating sphere. Then it should be possible to take a plane in space, 30cm or so in front of the pulsating sphere and plot all the air pressures and velocities through that plane.

If then we could make the plane surface then move in the same fashion to mimic the measured results, you would 'see', or rather hear, a perfect picture of this sphere. You cannot make the sphere, but it is possible to reproduce an acoustic picture of it. This is part of what we are trying to do.

A simple example will clarify this point. If I take a circular frame of about 90cm (3-feet) diameter and over it stretch a film of Melinex, 3.5 micrometers thick, it can be thought of as a plane in the air. If I hold it in front of my face and continue to speak, you will continue to hear me. The important question was did you hear a degraded sound quality?

Of course, the answer is yes, there were losses of about one decibel at 14kHz, and to cure this we could go on to experiment with other plastics to determine which would produce the best results. But this is not really necessary because the effects are quite predictable. The effect of that membrane between you and me could have been predicted over 100 years ago by Lord Rayleigh and all he had as test instruments were candle flames and elastic bands!

What is the point of this? Obviously, when I first spoke without the membrane, you heard me — my actual voice. When I moved the plastic sheet over my face, you were listening to a *reproduction* of my voice! I say it was a reproduction because the air pressure variations at your ears were due to the vibrations of the membrane.

Of course, I know the force causing the membrane to move, came from the air pressure variations produced by my voice. But you were not to know that — what you actually heard were the results of the membrane vibrating and sounding rather like Peter Walker.

So you now see, that if we could reproduce those forces on that membrane by an electrical method and they exactly mimic the effects of my voice on the membrane, then I would switch on the electrical circuit, go away elsewhere and you would hear Peter Walker.

How did the force distribute itself on the membrane? Obviously, the pressure reached the centre first and then reaches larger diameters on the plane progressively, until the outer edge is reached. It does this fairly quickly. At low frequencies, virtually the whole membrane moves together, but at higher frequencies we would see forced wave motions spreading from the centre, with an increasing number of expanding circles as the frequency increased.

In fact, in addition to the forced vibrations, there are other mechanical vibrations set up due to the stiffness and mass of the membrane. These travel to the outer edge at a much slower rate and have to be suppressed since they could colour the sound reproduction.

We have reached the first and important stage and that is to demonstrate that a membrane of this type has quite small and predictable degradation to a sound field. Thus, rather like an amplifier, all that is necessary is to make it good enough for its purpose.

In an electrostatic speaker it is possible to arrange ring-shaped electrodes on each side of this membrane, at a spacing of about 2.5mm, make them conductive and perforated to allow the air to move freely and finally to fix them so they remain rigid. If each of the concentric rings are then connected to each other by inductors and resistors, so that a signal applied to the centre circle passes on to each successive ring one at a time, we may be able to obtain the sort of force pattern we want.

This idea was in fact suggested over 50 years ago, so why was it not used and where does it go wrong? If this was all we did, the final reproduction would not be quite correct. First, with a membrane of about 90cm (3-feet) diameter, the output would fall by about 3dB/Octave below 300Hz. This is because the size of the membrane is smaller than the total wave front created by our imaginary pulsating sound source and so not so much air is being vibrated. Above about 300Hz, undulations of about 6dB to 8dB would appear in the frequency response because of the discontinuities

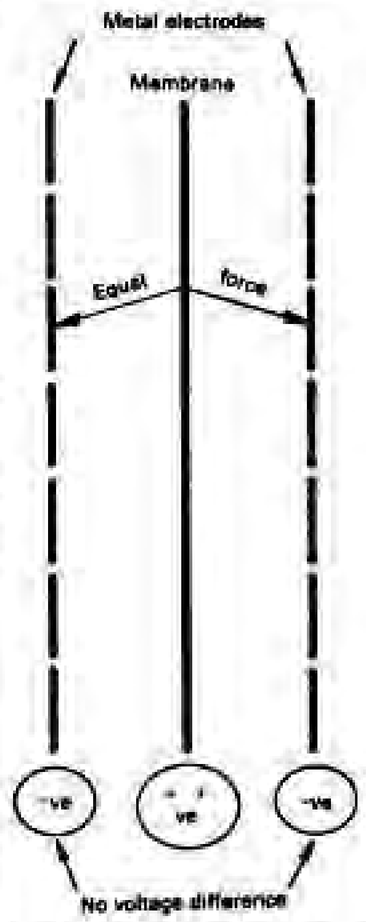
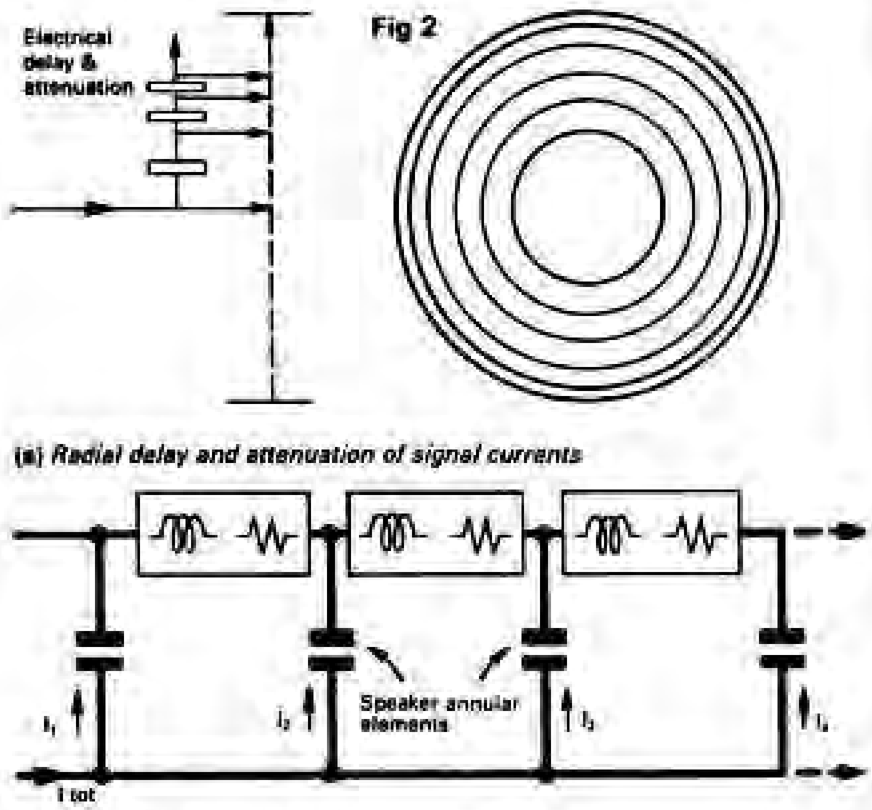


Fig 1 *Plates charge and open circuit show zero voltage difference when membrane is centreal*



(a) *Radial delay and attenuation of signal currents*

(b) *Simplified electrical circuit*

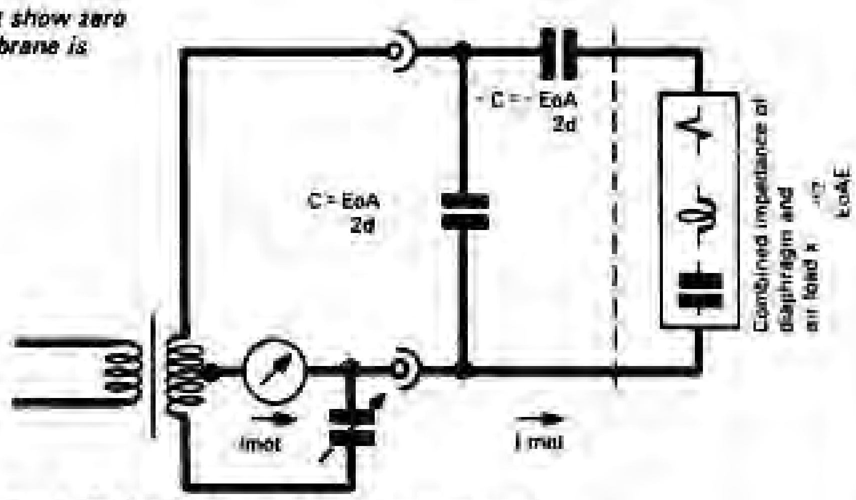


Fig 3 *Bridge for direct electrical measurement of membrane velocity.*

Typical motional current curve for constant applied voltage

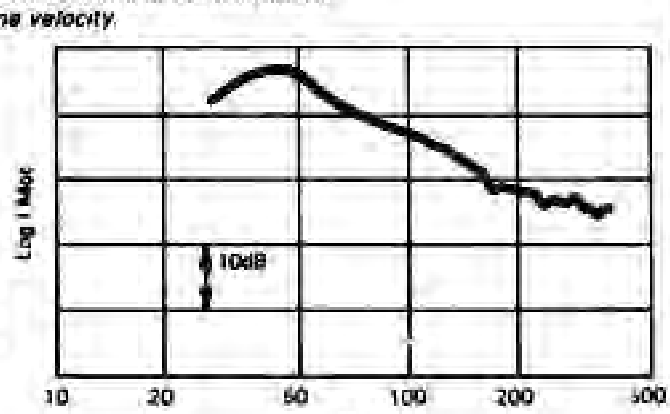


Fig 4

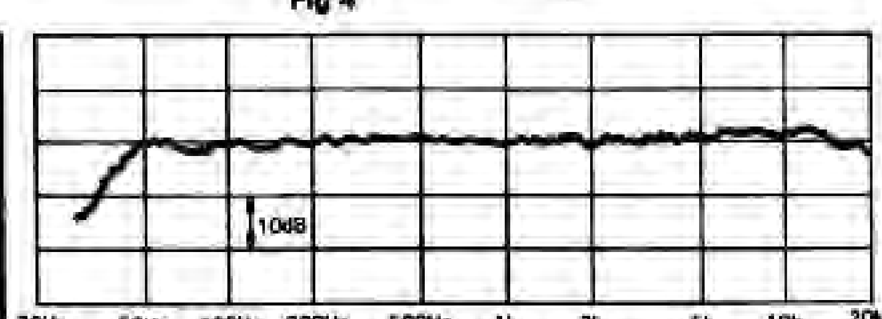
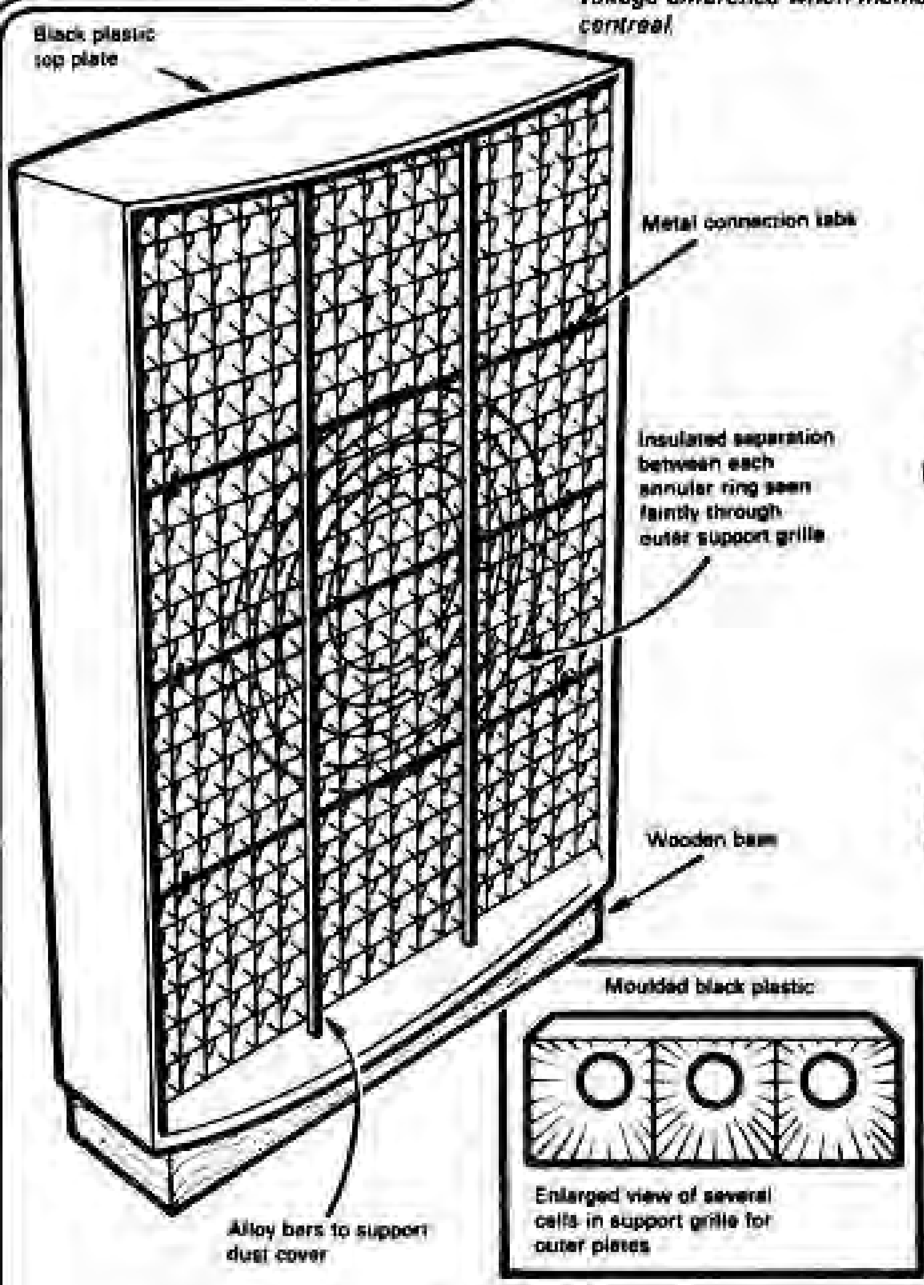


Fig 5 Outdoor axis response at 2 metres



Prototype QUAD ELS without covers overall size 90 x 45cm approx.

caused by the finite size of the membrane. The edge of the membrane reflects energy back in to the centre to produce interference patterns.

So how can we eliminate these problems — are they an obstruction to making a good loudspeaker? To understand that we need to look at the basic transduction method of the speaker, see how it operates as a microphone and because all these things are **reversible, calculate** its reciprocal behaviour as a speaker to see if a solution can be found.

Looking at the *Fig. 1*, the outer dashed lines represent the perforated metal electrodes, the solid line represents a membrane that is so light and so unrestrained that if we put the device in any sound field, then the membrane will move with the movement of the air particles of the sound field. There are no baffles or boxes surrounding it and at the moment, we do not define the total area.

The forces on the membrane will be equal and opposite because we have applied a charge as shown. Because the outer electrodes are equally charged with the same polarity, there is no voltage difference between them. But what happens if the membrane is now physically moved to the left? There is no electrical connection to the elements and therefore charge cannot migrate. So from school physics lessons, we remember that the voltage on the left electrode reduces in proportion to the distance moved by the membrane and conversely, the voltage on the right electrode rises in a similar proportion.

The force on the membrane has not changed because the gradient of the electrostatic field remains the same. However, a voltage difference has now appeared between the outer electrodes and we know that it follows a law of proportionality with the distance moved by the membrane.

We now only need one other ingredient to see how the device will work as a microphone since we already have a device that produces a voltage proportional to air-particle displacement. That is a point source of sound having a constant volume velocity will produce in the far field, an air displacement which is independent of frequency.

Putting these two facts together, we can now state that a source producing a constant velocity of air motion will produce a constant voltage on the electrostatic element. By reciprocity, we can also say that if current is pushed into the outer electrodes of the electrostatic element, pressure will be produced in the far field, that will be independent of frequency.

This can be calculated easily by using the very simple formula

$$\text{Sound pressure} = \frac{E}{d} \cdot \frac{I}{r} \cdot \frac{1}{2\pi c}$$

Nm^{-2} (Newtons per square metre)

This says that if you put a polarising charge (E) across two plates separated by a distance equal to 2d (say 2000V/mm), and you feed into the electrode a current of about 5mA (I), then at a distance along the axis (r) of say two metres and we accept that c is a constant (the speed of sound in air), then you will obtain a resulting air pressure of a little over two Newtons per square metre or just over 100dB of sound pressure.

You may look at this formula and ask what happened to the frequency dependency? In fact it is irrelevant, you will obtain 100dB of sound pressure at any frequency and also regardless of the diaphragm size, providing the current fed to the electrodes is 5mA!

Of course, there are limitations, because if you make the element too small, a 5mA current would break the device, so it has to be made large enough to accept the current and we will also have to accept that the lower the frequency, the greater will be the voltage that appears for the same 5mA.

This can be simply summed up in two basic statements:

1. If a linear electrostatic speaker is constructed as an acoustically transparent plane of uniform transduction with electrode currents separately accessible over the area of the plane, then the far field axis response is simply and directly related to the vector sum of the electrode currents.
2. The pressure at any angle off the axis can be derived by the summation of the currents for each elemental area, with due regard to its doublet directivity function and its spatial relationship.

Now perhaps, you can see what we are getting at. With this device, if the current in each of the ring electrodes is measured and then added together, it is possible to calculate exactly, the sound pressure at any point. Of course, some extra calculations are necessary to take into account the directivity function, but this is not all that difficult.

Taking a closer look at this (*Fig. 2a*), the speaker comprises a unit in which the outer electrodes are divided into six annular rings of equal area and the electrical signal is fed to the centre first and then on to the others *via* an inductance and resistance.

The circuit is actually a little more complex, but essentially, the purpose is to delay the outward spread of the driving signal in very much the same fashion as the sound pressure from my voice spread across the membrane used in the earlier experiment.

In practice it is important to gently attenuate the current and voltage down the delay line to prevent reflections. Remember that earlier, I had mentioned that the acoustic output of an ordinary membrane would show 6dB-8dB undulations in response. These are an exact analogue of the delay-line parameters, when the delay line is not terminated. So by electrical measurement and calculation, the delay line reflections can be eliminated and immediately, the acoustic output variations will disappear.

Obviously, the success of all this depends upon the unit being acoustically transparent and this needs a little qualification. First, the membrane is situated between the electrodes experiencing an equal force of attraction to either electrode. In practice this is not exactly stable and the membrane will always move to one or the other electrode. To eliminate this, tension has to be applied to the membrane, but this naturally affects the performance. The method used to determine the changes involve looking at the motional current of the speaker.

The next drawing (*Fig. 3*) shows a simple bridge circuit. If the polarising supply (E) is switched off, the radiation impedance becomes infinity leaving just the static capacitance of the speaker in the circuit. Now the bridge can be adjusted to give a null reading in the meter, using the balancing capacitor. If the polarising voltage is now restored, the bridge becomes unbalanced due to the motional impedance of the membrane and the meter shows the resultant current.

A graph produced by this method and showing the performance at lower fequencies is given in *Fig. 4*. The small variations at high frequencies are due to the reflection of the speaker output, reverberating around the room and being picked up by the speaker again.

The low frequency resonance is an instance of where the speaker is not completely acoustically transparent and the stiffness of the membrane coupled to the mass of the air load has some effect. At high frequencies, the speaker again departs from the ideal model because the mass of the diaphragm and that of the air around the electrode perforations produces a small loss above 10kHz.

By modifying the current drive to the speaker in an opposite direction, all these faults can be cured. The results are shown in *Fig. 5* and is a curve taken at two metres on axis. The acoustic power output is smooth up to about 1kHz and then reduces down by about 6dB at 8kHz. This indicates the effects of increasing directivity at high frequencies.

So finally, what is new about all this? Really, it is a lot of old ideas fitted together. Kellog in 1929 proposed the connection of a series of electrostatic elements by inductors as a delay line. His idea was to improve efficiency and reduce the power requirement from amplifiers. Shorter of the BBC took out a patent in 1941 describing the connection of a series of annular rings using resistors and Jansen, in 1953, suggested variations on the same theme.

In effect therefore, all I have done is to collect all these ideas and add a little work which says that if you can make the device acoustically transparent, then the performance can be predicted. We think that this is very important since it enables correction to the performance to be made very easily and after simple laboratory measurements.

Peter Walker

Note from Quad

In the July issue of PRACTICAL HI-FI our Chief Engineer is quoted as saying *'that the IMF loudspeakers are damned insensitive'*. We wish it to be clearly understood that this remark was based on experience with a particular loudspeaker some years ago and should not be construed as applying to current IMF production models which are of comparable efficiency to other high quality loudspeakers and of greater efficiency than our own electrostatic loudspeaker.

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