A Hexagonal Double-bass

E. J. Jordan presents the theory behind an unusual loudspeaker design employing four treble modules and two front-to-back bass units— all in a hexagonal cabinet suitable for home construction

WE start with the basic facts of sound radiation. If we consider a single cone loudspeaker unit mounted in a hole in the wall, the expression for radiated sound power is:

$$P_A = \frac{F^2}{Z_M^2} \times R_A \qquad 1$$

Where

P_A = Radiated sound power.

F = RMS force applied to the voice-coil.

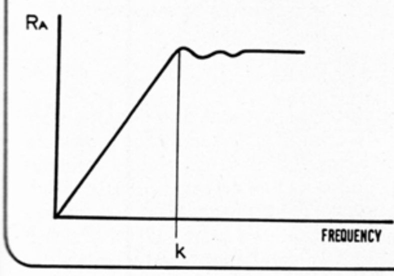
Z_M = Total mechanical impedance of moving parts.

R_A = Radiation resistance.

(For the sake of clarity, all constants are being omitted.)

The radiation resistance has the form shown in fig. 1, where it is seen that R_A has

FIG. 1



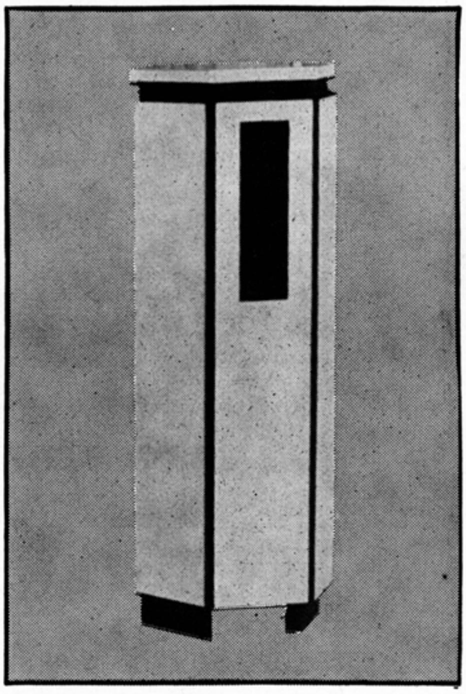
two distinctly separate parts above and below the 'knee' at frequency *k*. This frequency is an inverse function of cone diameter. For a cone of 250 mm diameter *k* will be approximately 800Hz, and for a 50 mm cone it will be about 4000 Hz.

The two parts of the radiation resistance curve are served by separate equations, and by substituting in equation 1 we have:

$$\text{Below } k \quad P_A = \frac{F^2}{Z_M{}^2} \times f^2 \times A^2 \qquad 2$$

$$\text{Above } k \quad P_A = \frac{F^2}{Z_M{}^2} \times A \qquad 3$$

We will assume for the next part that the force F applied by the voice-coil to the cone has a linear amplitude characteristic and is independent of frequency, which now leaves the entire performance characteristics of the loudspeaker defined as a function of Z_M. An electrical voltage waveform applied across the voice-coil will cause a corresponding oscillatory force to be applied to the cone, which we would then 'like' to respond with a corresponding oscillatory motion, but it does not. All would be well if Z_M were non-existent or even well behaved, but in due course we shall see that when Z_M is dissected into its constituent parts all sorts of non-linear and frequency-selective nasties come



crawling out. But we shall then see what can be done to tame them.

The three major components of Z_M are: the total moving mass M (which can vary with frequency), the stiffness 'S' of the suspension (which is usually non-linear), and the electro-magnetic damping R_M (which is usually anything but the correct value). The presence of mass and a stiffness component immediately indicates a resonant condition. These components in fact determine the fundamental resonant frequency of the cone f_0. Now, not having three hands, I am going to ask R_M to assume for itself a value such as to damp the resonance to a Q of unity. In this case, at frequencies below f_0 the motion of the cone is mainly controlled by the stiffness of the suspension, and putting this in equation 2 gives:

$$P_A = F^2 \times f^2 \times S^2 \times f^2 \times A.$$

$$\therefore \; P_A \propto f^4$$

This shows the radiated power to be falling at a hell of a rate as frequency falls (24dB/octave).

Above f_0, where the controlling component is the moving mass, equation 2 becomes:

$$P_A = \frac{F^2}{f^2 M^2} \times f^2 \times A^2$$

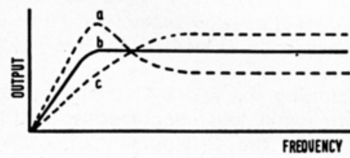
Here it is seen that the f^2 terms cancel and P_A is independent of frequency. How convenient! If, however, as is so often the case, the loudspeaker has been fitted with a magnet far too large for it, the value of Q will be much less than unity, in which case the damping component R_M will dominate, so again into equation 2 we have:

$$P_A = \frac{F^2}{R_M{}^2} \times f^2 \times A^2$$

$$\therefore \; P_A \propto f^2$$

This shows a condition of absolute bass cut at a rate of 12dB/octave. In practice it is not that bad, since R_M cannot exist on its own, but it does indicate the futility of excessive damping. No system can possibly incur a condition of oscillatory resonance if the Q is

FIG. 2



less than unity. Fig. 2 shows the effect of different magnetic flux densities on a given unit. It is seen that increasing the strength of the magnet does not so much increase the overall efficiency as tilt it about a certain frequency (which so happens to be *k*). In my view, there is absolutely no justification for the cost of the unnecessarily large magnets fitted to many loudspeaker units; but back to this anon.

Above the frequency *k*, if the moving mass continues to be the dominating component, equation 3 gives us:

$$P_A = \frac{F^2}{f^2 M^2} \times A$$

So now our *treble* power is vanishing at 12dB/octave. Is there no justice . . . . But for once, and just for once, our prayers are heard and the immutable power in charge of these things provides us with an artifice to overcome the problem. This is the fact that a force applied to the cone centre by the coil cannot be transmitted all over its surface instantaneously. There are therefore phase displacements between the centre and the edge of the cone from which it follows that parts of the cone surface are flexing.

One effect of this flexure is that the effective cone area is reduced, and by the same token so is the effective mass. Looking again at equation 3, it is apparent that if M is proportional to A:

$$P_A = \frac{F^2}{f^2 A^2} \times A$$

$$\therefore \; P_A \propto \frac{1}{f^2 A}$$

Now if, as we have said, A is decreasing as *f* increases, this will provide some compensation for the treble loss originally indicated.

At this point the directivity factor becomes relevant. At frequencies below *k*, whether we like it or not the loudspeaker is virtually omni-directional. Above *k* it becomes progressively more directional, and since the radiated power P_A is being concentrated over a progressively smaller area, although P_A may be falling the actual sound pressure level in the listening area could actually rise as frequency is increased. It must be noted that the directivity/frequency characteristic of a loudspeaker is determined purely by the dimensions of the effective radiating area which, as we have said, is a function of frequency. This leads to a very complex situation—but don't worry, I still know what I'm doing even if you don't.

By attention to the appropriate details of cone design, it is possible to evolve a flexure characteristic that will very closely compensate for the intrinsic reduction in treble power, so that a substantially frequency-independent sound pressure level is maintained over a wide listening area. To those who think that that is a tall order, I would point out that the moving-coil loudspeaker system in which this problem has been solved in any other way does not exist.

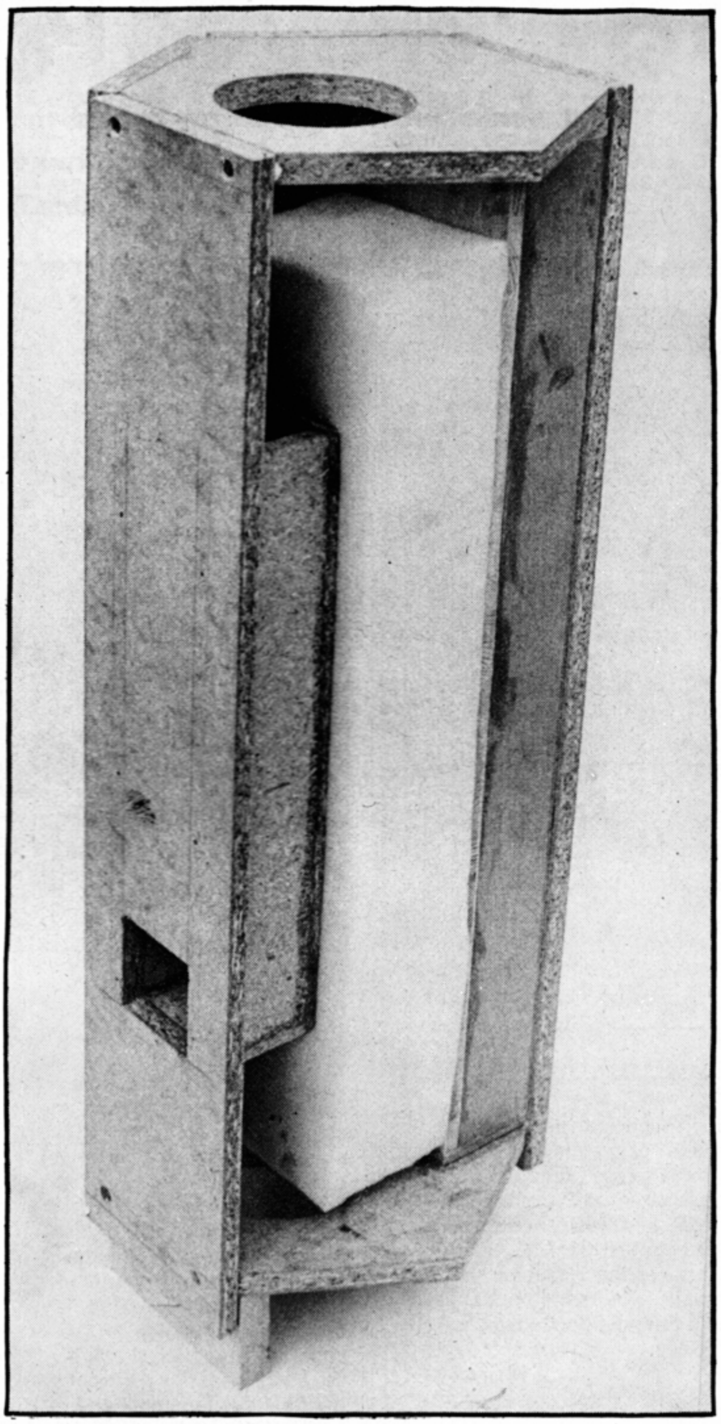
In developing the JORDAN 50 mm unit I decided to avoid 'over engineering' and to concentrate on the development of a basic loudspeaker module that could, on the one hand, be incorporated into 'money no object' systems which could be designed to meet practically any specification, and on the other hand be used to provide an outstanding performance in systems of relatively modest cost. In addition, there exists the means for providing performance features that I consider worth having, but lacking in other systems.

So, starting at the top end, if the upper frequency spectra of live speech and music are taken in conjunction with our ear charactersitics as shown by Fletcher & Munson, it is my opinion that 20 kHz is a very adequate top limit. This limit is set by the mass of the voice-coil. It was also decided to set the directivity characteristic to an included angle of 60° for reasons that will be seen later.

Considerable research was then undertaken to optimise the flexure characteristics of the cone, which, together with considerations of sound pressure levels and many other factors, led to the development of a cone of 50 mm diameter.

The selection of the cone material was to a large extent determined by a factor which, as far as we know, has not received any previous mention. If we accept that cone flexure is a fact of life, it is imperative that when at any instant a particular flexing force is removed, the cone restores very quickly to its original form. Many materials, especially plastics, do not fully restore for quite long periods. This delayed restoration rate inhibits the ability of the cone to follow rapid detail in the input waveform, and can be a source of considerable harmonic and intermodulation distortion, a fact sadly evident with many plastic domed tweeters.

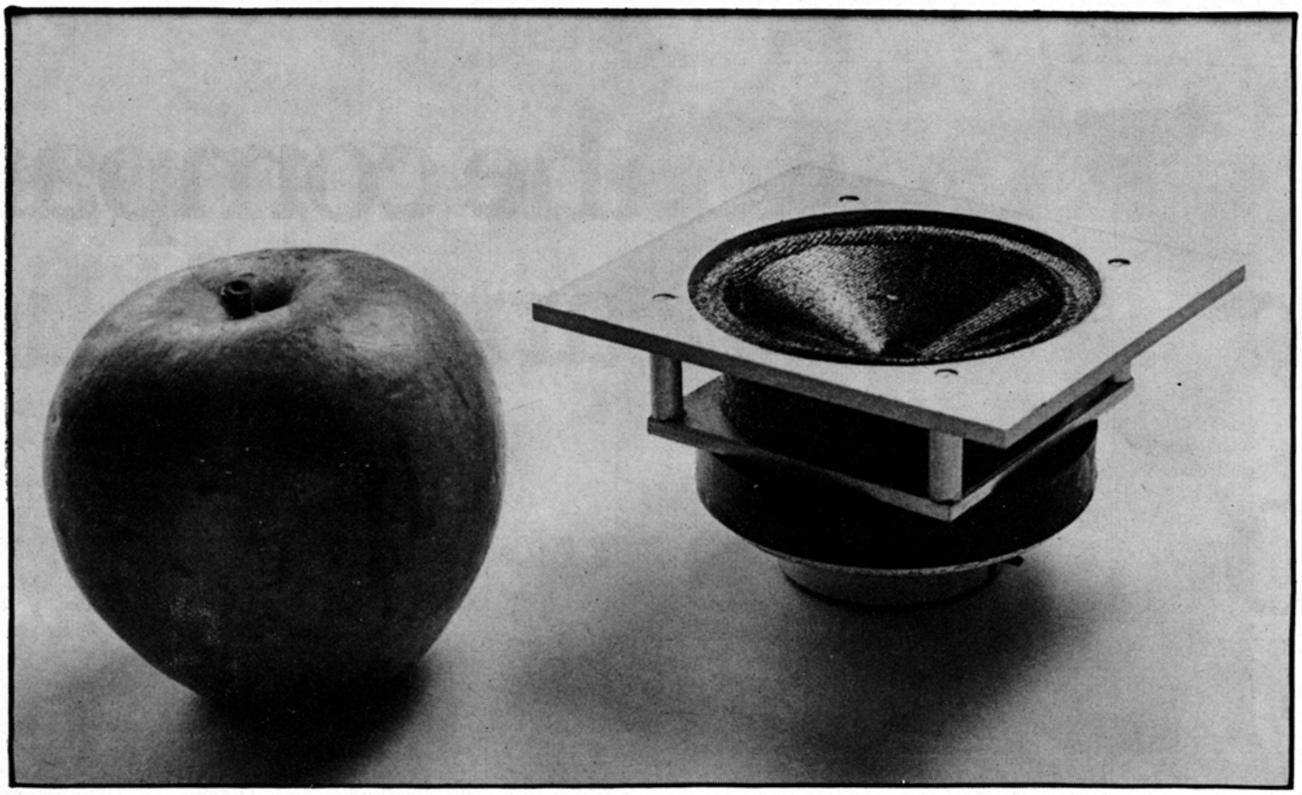
It was therefore decided to use an aluminium alloy having a very fast restoration rate. A further reduction in distortion was achieved by the use of a straight-sided cone, thus fully endorsing the findings described by Hugh Brittain in an article in the November 1952 issue of *Wireless World*. This article covered the development of the GEC 8 in. metal-cone unit, which became very well respected during the relatively short period of its existence.



Cut-away showing constructional details

A less desirable effect of cone flexure is the occurrence of resonant frequencies, and it might be supposed that the use of a metal cone would increase the problem. In fact, the reverse can be true. Try this simple test: Tap a wine glass and a plastic beaker. The wine glass will ring; the beaker won't, but it will produce a distinctly coloured sound. Now try again whilst gripping the edges of the wine glass and beaker with the fingers. The sound from the glass will be quite dead, but the beaker will still exhibit its coloration. The fact is that narrow-band high resonances found in low-loss materials are often much easier to 'kill' by damping than the wide-band low Q resonances found in 'lossy' materials.

The Q of a resonance is proportional to the moving mass, therefore by keeping this to a minimum we have a big starting advantage. Any residual resonances are almost eliminated by arranging for the cone surround to present the correct termination impedance. The moving-coil cone loudspeaker is particu-



The core of the matter—one of the treble drive units (on the right!)

larly amenable to this type of damping, since at high frequencies it is effectively a transmission line in which the force is applied at the centre and travels to the edge, where the energy can be absorbed. Dome units cannot operate in this way, hence the need to use intrinsically lossy plastics.

A 50 mm cone will have a *k* frequency of about 4 kHz, below which the radiated power remains constant provided the condition of mass control is maintained, as we have seen. The next thing to consider was how far down in frequency the unit would work within the bounds of a reasonable power handling capacity. At high frequencies this was limited to about 20 watts by the heating effects of the voice-coil. Using this as a criterion for the low frequencies, it was found that a frequency of 250 Hz could be achieved before exceeding reasonable limits of cone displacement. This meant that the unit could be safely used at this continuous power level with a 12dB/octave crossover operating at −6 dB at 150 Hz. This also gave a frequency of 100 Hz as a reasonable place for the bass resonant frequency of the unit. The *instantaneous* power capability is, of course, much higher: in excess of 50W.

A suspension system capable of allowing a total moving mass of only 1·3 grams to remain in control down to 100 Hz would obviously have to be much more sophisticated than the conventional arrangement. Accordingly, a roll surround having extremely low stiffness was developed, and an extremely linear restoring force provided by a unique suspension system housed *behind* the magnet and coupled to the cone via a very light, rigid microtube passing through the magnet system. Even the minute mass of this tube is decoupled from the cone at high frequencies. Particular attention was paid to the design of the voice-coil and the magnetic gap, to minimise inductive losses and to provide a drive system that was as linear as possible under peak drive conditions.

Having developed the 50 mm module, it was found that most of the major problems of total system design seemed to evaporate. The modules themselves require only to be mounted in an enclosed volume of not less than 1·5 litres per module when the most critical frequency range from 100Hz upwards is dealt with. For the bass end, there are many excellent drive units and enclosure designs that would effect a good match for a 50 mm system. In our own designs, however, we prefer to use the classical reflex principle according to A. N. Thiel.

Over a narrow band of frequencies the reflex enclosure operates as an acoustical transformer, inasmuch as it provides an improved match between the low impedance of the radiation resistance and the relatively high impedance of the cone. Also, like an electrical transformer, it provides phase inversion, so that over the operative band of bass frequencies the energy from the rear of the cone is radiated in phase with that from the front. The reflected acoustic load applied to the cone results in a considerable reduction in its amplitude of movement at these frequencies.

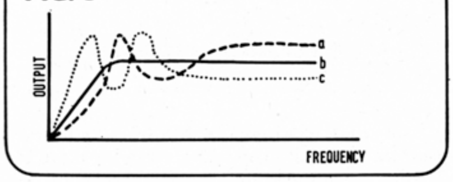
The advantages accruing from reflex loading are: high efficiency, increased power handling capacity, reduced distortion, and aesthetically acceptable enclosure dimensions. So why is reflex loading not far more generally used? There are two reasons. The principal one is that a reflex system has three critical frequencies where the reactive terms represent potential resonance. If the drive unit and enclosure are not correctly matched, these resonances will be manifest. However, with correct enclosure design and choice of loudspeaker parameters, the critical frequencies can be given a Q of unity, resulting in a non-resonant system which will operate with all the advantages claimed down to the fundamental resonant frequency of the loudspeaker drive unit.

The basic requirements for a correct match between drive unit and enclosure are few and simple: the reactive terms of the enclosure must be zero at the fundamental resonance of the drive unit; the stiffness reactance of the enclosed air volume must equal 0·62 that of the cone restoring force; and the free-air Q of the drive unit must equal 0·62. This brings us back to our earlier references to loudspeaker damping, as many of the drive units available to the designer are far too over-damped for good reflex loading. The effects of this are shown in fig. 3.

Another problem with reflex loading is one which the designer should not really be asked to accept. It is that at very low (rumble) frequencies there is very little loading on the cone, and excessive displacement can occur. At the risk of treading on toes, I would say that loudspeaker systems having performance standards of the level being discussed should not have to contend with equipment that rumbles. In any case, the correct place for a rumble filter is in the pre-amp, not in the loudspeaker system.

In all but the smallest of my Jordan system designs, a further feature is introduced in the form of twin push-pull bass drive units. This is a further distortion reducing technique designed to overcome the intrinsic non-linearity of conventional corregated cloth suspensions. It is characteristic of these devices that their stiffness tends to be higher in one direction than in another—and invariably in the same direction. Therefore, if two bass drivers are used with one fitted in

FIG. 3



reverse relative to the other, and the phase restored by reversing the connections, then the predominately even harmonic distortion will tend to be cancelled. It's a sort of 'double-bass' system.

The design of the crossover circuit is very simple and basic, since at the frequencies concerned the electrical impedance of treble and bass systems is substantially resistive. Also, the design does not incorporate the 'curve bending' components found in many systems which tend to flatten the frequency response at the expense of transient and phase distortion.

The circuit is shown in fig. 4, where it will be seen that it is a series rather than the more usual parallel arrangement. This reduces the power requirements for the inductors. The 2·2 µF non-polarised capacitor by-passes the 100 µF capacitors to compensate for the inductive losses of the latter at high frequencies. The overall approach described makes possible a very wide variety of total system designs, all of which are based upon the use of one crossover frequency only, *ie* 150 Hz.

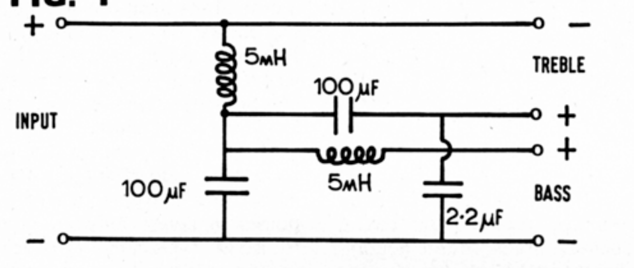
We are frequently asked if the addition of a 'super-tweeter' for the extreme high's would be desirable, and our answer is invariably *no*. We have yet to meet the tweeter capable of anything like the detail resolving power of the 50 mm module at high frequencies (with the possible exception of a ribbon), while the transient, phase and harmonic distortion incurred by most tweeters and their associated crossover circuits is a very high price to pay for a few kilohertz of ultrasonics, which very few of us can hear.

It will be apparent to the reader by now that our whole design philosophy is one of simplicity, which is soundly based upon long-established basic principals of electro-acoustics. We have shown how all the various mechanical and electrical components can add their own degradation to the sound quality, and how these have been reduced to a minimum. It must also be apparent that whilst our designs lack the complexity of many, no effort has been spared in attention to the details that matter.

So, having evolved techniques for the accurate translation of electrical waveforms into sound, are we home and dry? The answer is '**No**', not if we want to capture some of that elusive magic of which one can only be fully aware at a live performance. We are now onto the question of subjective qualities, which, since music is essentially an emotional experience, is in our view as important as the rest.

A live performance has width, depth and ambience which are often far from fully served by the conventional pair of stereo loudspeakers. It is quite a mistake to think that the provision of two separate sound sources handling left- and right-hand bits of the program will automatically combine to provide a homogeneous stereo format. The loudspeaker system has sometimes been referred to as a 'window on the orchestra'; if we stay with this concept for a moment, a stereo pair of loudspeakers could be analogous to two windows in the wall, each only a few square feet in area, with the orchestra on one side of the wall and the listener on the other. What we would like to do is take the wall away.

FIG. 4



I admit that this is something of a personal quirk, but I have found that I cannot enjoy music if I am conscious of it emanating from two relatively small areas, one each side of the supposed sound-stage. In fact, at its worst I find this more disturbing than mono, so over a period of many years I have indulged in considerable research in order to find a means of overcoming the problem.

An article of mine was published in *Wireless World* (February 1971) which fully outlined the problems and proposed a number of possible solutions, the ultimate one being the use of a long continuous line of drive units, extending horizontally right across the effective sound-stage. The units were interconnected with phase delay networks and the left- and right-hand inputs were connected at each end. This system worked extremely well, but had the disadvantage that it required the total devotion of at least ten feet of wall space.

To specify our aim, it is to produce in the listening room an effective sound-stage area consistent with the program. For a solo instrument or voice a mono system would be perfectly adequate, but in the case of an operatic production the ears expect a large sound area and may not be happy to accept two small ones. We would therefore like the loudspeaker systems (preferably in the form of left- and right-hand enclosures) to create a totally continuous sound-stage on which each voice and instrument is well defined and correctly positioned, irrespective of listening position, and leaving the listener with no awareness of the loudspeakers.

It was found that some speakers were better than others in this respect, and further investigation showed that this was very closely associated with the polar distribution pattern of the system. It appeared, in principal, that the broader the polar response, the worse was the performance in terms of the present discussion. About this time, various writers commented adversely upon the stereo performance of omni-directional loudspeakers. We saw earlier that polar distribution was solely a function of the physical dimensions of the sound-source; a wide distribution implied a small sound-source, and this seems to be exactly the way the ears hear it. Although a further complication is introduced by the room reverberation, the wider the polar distribution angle the greater will be the proportion of reflected to direct sound. This can give an artificial impression of a larger sound-source size, but will tend to confuse the actual stereo images even more.

It is very interesting to note that many loudspeakers using dome tweeters provide good stereo imagery, which is usually attributed to the wide dispersion characteristics of the tweeter. In fact it will be found that these systems also employ cone units for the mid-range, which becomes quite directional at the upper middle frequencies, and it is this and not tweeters that are responsible for the stereo performance. As a result of considerable work in this area, I am convinced that the best stereo image stability that can be obtained from a simple system is that provided by a cone loudspeaker handling all the middle and upper frequencies. By avoiding the use of high frequency crossovers and tweeters a very high degree of phase coherence is maintained, which makes a further very significant contribution to a well defined stereo performance.

Let us now examine the factors which determine how the location of a particular image is defined within the effective sound stage. Imagine two loudspeakers in the conventional stereo position, reproducing a central voice. If the listener is in the centre he will hear a central image of the voice, because the loudness from each loudspeaker will be equal, the time taken for the sound to travel to the listener will be equal, and the sounds will be in phase (fig. 5a).

If the listener now moves left of centre, the apparent sound level due to the left loudspeaker will be increased and that from the right speaker decreased. We will call this the *proximity effect*, and if the speakers have a wide polar distribution angle (which implies a

FIG. 5

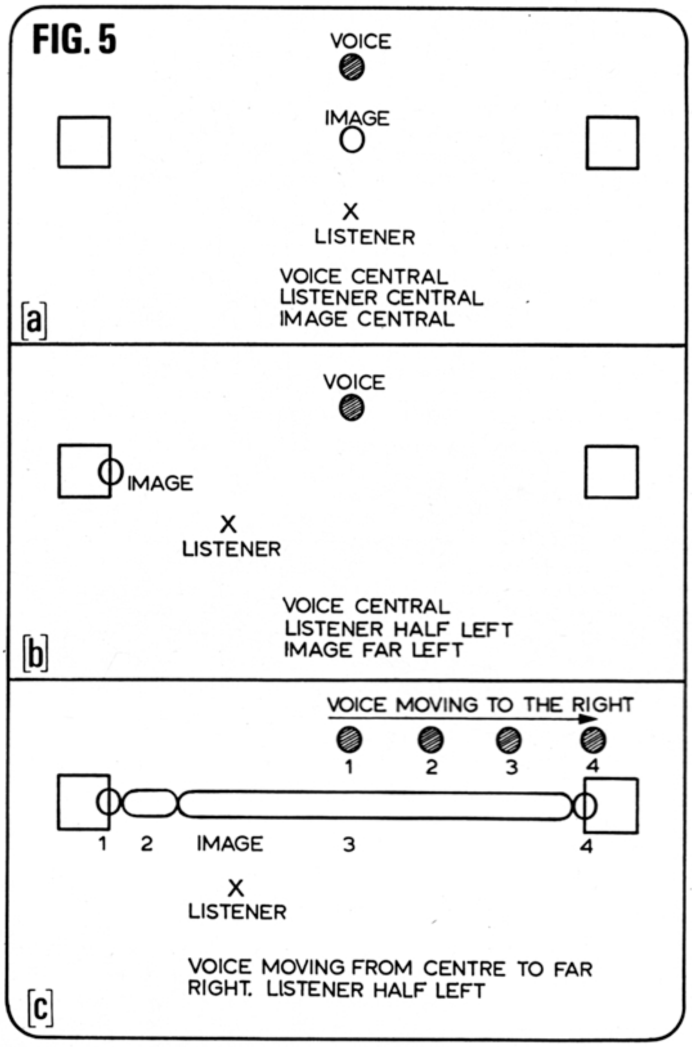
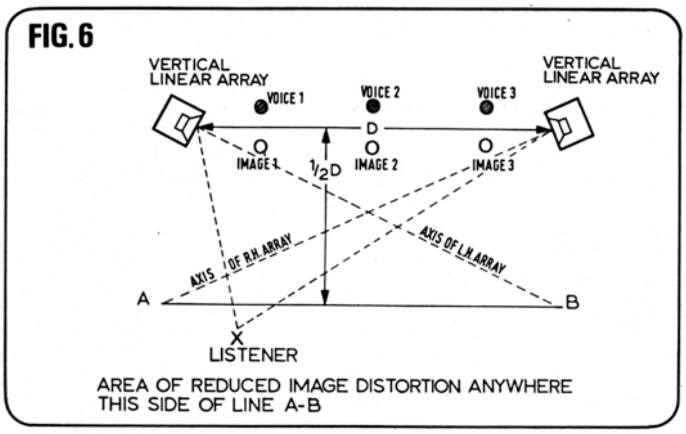


FIG. 6



spherical wavefront) the sound level at the listener due to either loudspeaker will be proportional to $1/d^2$. Also, the time taken for the sound to reach the listener will be decreased from the left speaker and increased from the right speaker. This is called the *precedence effect*, and the time taken will be proportional to distance.

So it is seen that if the listener moves left of centre, both proximity and precedence effects operate to shift the image to the left; and since with omni-directional systems the proximity effect obeys a square-law, even a small shift to the left on the part of the listener will tend to move the image over to the left loudspeaker—or it will become vaguely broad and have a markedly leftward bias (fig. 5b).

If now the actual voice were to move to the right, the sound level would increase from the right loudspeaker and decrease from the left; but the voice will have to move well over to the right before the loudness of the right loudspeaker is sufficient to overcome the proximity advantage of the left, which in any case will still retain its precedence advantage. At this point, our listener still patiently sitting left of centre will hear the voice now as an image widely stretched between the loudspeakers, with no positive location at all (fig. 5c).

When listening to music in this situation, most of the sounds will appear clustered around the nearest loudspeaker, with wide images of indefinable location stretching between the speakers. Background accompaniment will be heard from the further loudspeaker contributing extreme 'end of the line' instruments only. In practice the situation is not usually so bad as this, since most loudspeakers are not omni-directional and the proximity effect is reduced. Provided good phase coherence is maintained, an acceptable stereo performance can be achieved for reasonably centred listening positions—but the precedence effect remains.

The problems we have described can be almost completely resolved quite simply (if not very cheaply) by the use of specific polar distribution patterns. An approach which has proved extremely successful uses a left- and right-hand mirror-image pair of loudspeakers, each of which utilises four Jordan 50 mm modules fitted in a vertical linear array formation. The polar distribution pattern will be fairly restricted in the vertical plane down to about 300 Hz, but in the horizontal plane will have the same characteristics as a single unit. The overall distribution pattern will then be cylindrical up to the upper-middle frequencies, above which it will become progressively wedge shaped as frequency increases. The two arrays are inclined inwards so that their axes cross well in front of the listener (fig. 6).

Now what we have is, firstly, a cylindrical wavefront which provides a sound level at the listener proportional to $1/d$ so that the proximity effect is far less severe. It will also be noticed that as our listener moves to the left he will move progressively off the axis of the left speaker and more onto the axis of the right. This results in an effective phase delay in sound from the left speaker relative to that from the right, giving the latter an artificial advantage tending to cancel its loss of time precedence. By correct control of the polar response and correct positioning of the array, a wide effective sound-stage can be created, with the location of each voice or instrument well defined in its correct position irrespective of the listener. The effective stage width will automatically adjust to the requirements of the program, a central voice remaining central and the sound of a full orchestra filling the space between the loudspeakers. The situation will sometimes even 'hold' for listening positions beyond the width of the loudspeaker spacing.

The sense of realism that can be achieved from a full sound-stage, with the sounds of voices and instruments emanating from fixed points in space between the loudspeakers, is quite remarkable—especially when the latter appears to be producing no sound at all. This, I think, will be particularly appreciated by people who frequent live performances.

We now describe a complete loudspeaker system which we call *System Four*. It embraces all of the technological features discussed in the foregoing and it can be built without any great difficulty by the home constructor.

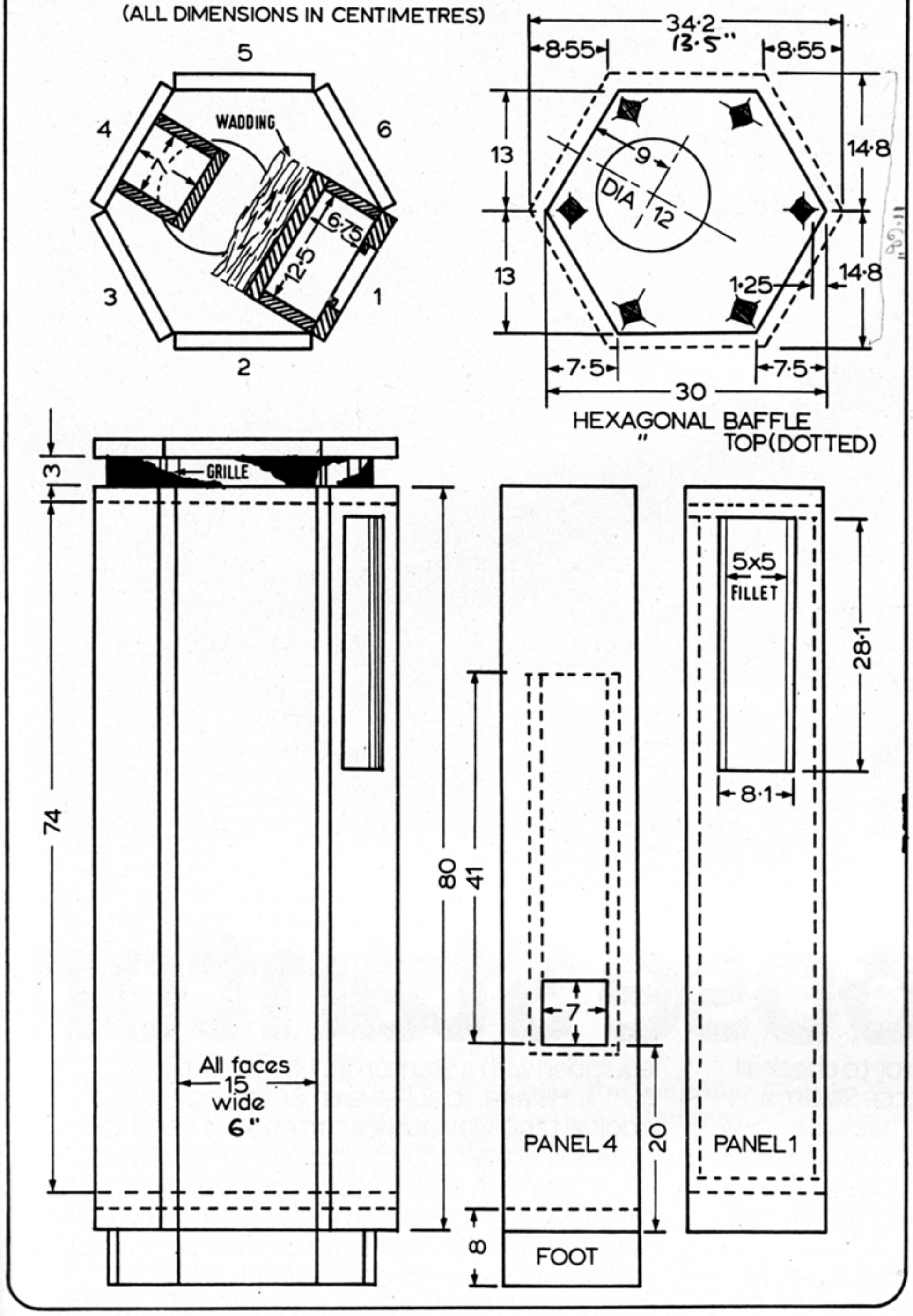
The complete system comprises two floor-standing enclosures, each incorporating a linear array, together with the twin-unit reflex arrangement for the bass. Correct angular positioning of the linear arrays is served by the hexagonal form of the enclosure, which also provides an unusual and attractive styling. The two bass units are fitted one at the bottom facing up into the enclosure, and the other at the top facing up out of the enclosure.

Construction of the enclosures is not difficult provided the hexagonal baffles are accurately cut. All joints must be substantially airtight. See fig. 7 for the general layout, which should be studied in conjunction with the Cutting List. Then proceed as follows.

(1) Assemble the three internal sides of the treble chamber, lay in connecting cables, lightly fill with fibre glass wadding (roof

FIG. 7 JORDAN SYSTEM FOUR

(ALL DIMENSIONS IN CENTIMETRES)



insulation grade) and fit to side 1.

(2) Assemble vent pipe onto side 4, and chamfer the edges to assist air flow.

(3) Fit 'feet' to sides 2, 4 and 6.

(4) Fit sides 1 and 4 to hexagonal baffles, making quite sure that they are square. Cables from the treble chamber should be brought out through the lower baffle.

(5) Fit sound absorbing material between vent and treble chamber. This may be a medium density foam block as shown in the photograph (p. 72), but a 'sausage' made from fibreglass wadding wrapped in polythene is preferred.

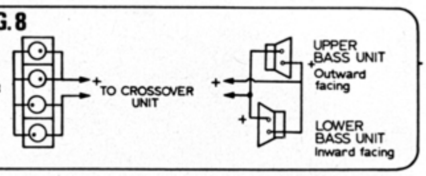
(6) Fit sides 2, 3, 5, and 6. Tape the insides of the joins and run a fillet of adhesive down the external 'V' grooves; alternatively, a filler may be used and run down with the finger to produce a convex curve in the grooves. The surfaces may now be finished to the constructor's taste.

(7) Connect the four Jordan 50 mm modules as shown in fig. 8 and fit, using an impact adhesive. Junctions between the modules may be taped, but this is not essential.

(8) Connect cables to upper bass unit and fit facing upwards on top of baffle. The cables should be brought out through a hole drilled in the lower baffle (see fig. 8).

(9) Fit lower bass unit onto the underside of the lower baffle so that it faces up into the enclosure. The Jordan crossover is also fitted onto the underside of the lower baffle, with suitable input connection facilities and all connections made. Make sure that the bass units are connected as in fig. 8.

FIG. 8



(10) Fit top hexagonal cover, using support blocks as shown or dowling. (It might be advisable to make this removable). Fit grille material wrapped round top supports and over 50 mm module array.

(11) Position both systems as shown in fig. 9, connect to amplifier, switch on, stand back and enjoy the music.

FIG. 9

For readers who would like to build a less expensive system now, and upgrade to a Linear Array later, the system may be built as above, but using only one 50 mm module. The latter is fitted at the top of each aperture, the remaining part of the aperture being blocked off with 1/8 in. hardboard. In due course this can be removed and the remaining three modules fitted to each baffle.

In addition to the above systems our literature contains four alternative designs, and if necessary we will undertake the custom design of systems to match any acoustic or aesthetic requirements. Jordan 50 mm Modules, crossover units and 100 mm bass drivers may be obtained directly from E. J. Jordan Ltd., 'Stoneyway' Bovingdon Green, Marlow, Bucks., SL7 2JH, or from Messrs. Wilmslow Audio, or Messrs. Badger Sound, who can also supply ancillary parts for the home constructor. ●

CUTTING LIST (for one enclosure)

18 mm	high density chipboard			
	Hexagonal top		1)	34·2×29·6 cm
	Hexagonal Baffles		2)	30×26 cm
	Sides		6)	80×15 cm
12 mm	high density chipboard			
Treble Chamber		Back.	1)	71·5×15 cm
		Sides.	2)	71·5×6·75 cm
		Top and Bottom.	2)	15×8 cm
Vent		Back.	1)	41×9·5 cm
		Sides.	2)	41×7 cm
		Bottom.	1)	9·5×8·2 cm
Timber		Feet.	3)	15×8×2·5 cm
		Top support blocks.	6)	3×1·8×1·8 cm

DRIVE UNITS (per speaker)
Four Jordan 50 mm modules (mid and treble)
Two Jordan 100 mm cone units (bass).