THE LINE-SOURCE LOUDSPEAKER AND ITS APPLICATIONS †

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A LINE-SOURCE LOUDSPEAKER is so called because its sound emitting element is long and narrow. It is frequently called a column speaker and there is no justification for reserving the word column for a particular kind of line-source. It has often been said that a line-source loudspeaker is much more sophisticated than a column, but this idea should not be encouraged. There are certainly many varieties of line-source and their differences will become apparent during the course of this article.

The fundamental characteristic of a line-source is that it propagates a fan shaped beam of sound. The plane of the fan is at right angles to the long dimension of the column. More precisely, the polar diagram is broad in the horizontal plane and shallow in the vertical plane when the column stands vertically.

The length of the column determines the angle within which the sound is confined in the vertical plane, the longer the column the smaller the angle. The angle also depends on the wavelength of the sound, the beam angle becoming smaller as the wavelength diminishes.

In addition to the main axial beam there are secondary beams above and below. Fig. 1 shows the vertical plane on the forward axis of a column. There is a set of secondary beams below the axis, but these are like those above it and have not been drawn in. The contours are equal-intensity contours. It is not a polar diagram as its co-ordinates are height and distance away from the column, the scale being arbitrary. The straight lines radiating from the column, each marked $-\infty$ indicate the directions of zero intensity or nulls in between the separate beams.

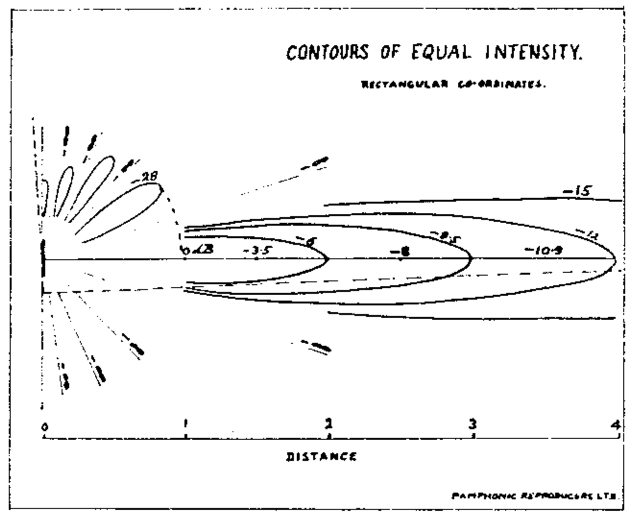

Fig. 1

The contours change as the frequency changes. At progressively lower frequency the upper secondary beams wheel anti-clockwise and the lower ones clockwise—each becoming extinguished in turn as it passes through the vertical line through the column. The main axial beam inflates as the first nulls off the axis wheel round until the forward propagation is virtually omni-directional at very low frequencies. If the frequency is progressively raised the main forward beam gets sharper, the nulls wheel in towards it and new secondary beams are born vertically above and below the column.

A more detailed study of radiation patterns will be given later; for the present it will be useful to discuss why and how this kind of speaker is used.

It is believed that the first line-source loudspeakers for commercial use were those installed at the White City Stadium in about 1933. (Fig. 2).

The problem here was to cover the whole stadium at minimum cost. The perimeter was over a mile round and a distributed system of

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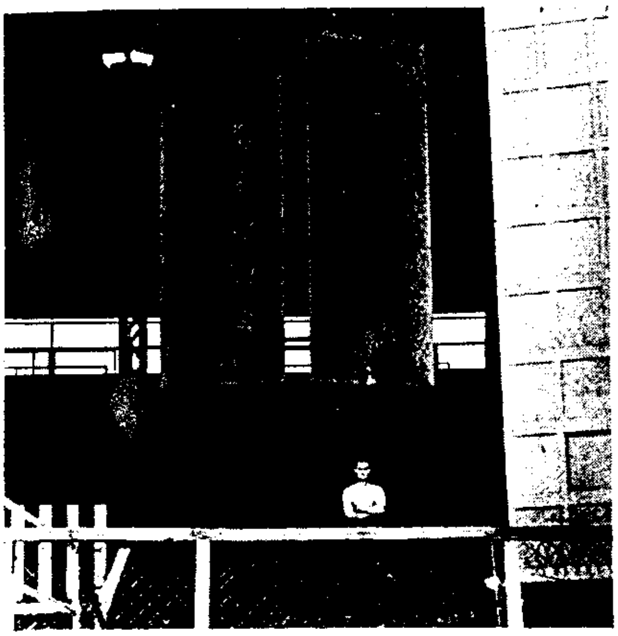

Fig. 2.

small loudspeakers would not only have been expensive as regards wiring, but the listeners would have suffered that aggravating effect of hearing all the loudspeakers in rapid succession. The columns in Fig. 2 were 16 feet long and 4 feet wide at the mouth of the flare. The power input was 125 watts each.

Very even coverage of speech and music was obtained as the flat broad beam penetrated right into the covered stands. Moreover, the whole central area was covered as well and this would not have been easy with a distributed system.

But it was not until after the Second World War that line-source loudspeakers came to be used indoors. During 1950 the acoustics of St. Paul's Cathedral were studied. At that time St. Paul's Cathedral had an induction P. A. System. Seven kilowatts of hi-fi power were pumped into loops of cable which encircled the crypt. Each row of seats on the Cathedral floor was fitted with a pair of loudspeakers and round each row went a copper band which was transformer coupled to the speakers. The crypt loops formed the primary and the banded seats the secondaries of what was a huge air cored—or cathedral cored—transformer. This arrangement allowed the rows of seats to be shifted about according to the layout required. There was no wiring in the ordinary sense. Unhappily the system created too much reverberation. Despite the fact that everyone in the seats was within 10 feet of a loudspeaker the direct to reverberant sound ratio was insufficient for really comfortable hearing.

The reverberation time of the Cathedral at mid-frequencies is 10 to 12 seconds and at that time nobody had been successful in putting in a sound reinforcement system which could cope with such a long R.T. Questions arose as to the possibility of including sufficient sound absorbing material to reduce the R.T. to a manageable figure, but this was difficult because aesthetic considerations did not allow architectural features to be interfered with. Eventually the Building Research Station of the Department of Scientific and Industrial Research was asked to advise the Dean and Chapter. Mr. P. H. Parkin, head of the acoustics section of that establishment then undertook a long series of measurements in the Cathedral. He received much publicity in the popular as well as the serious newspapers by being the first person to fire a revolver within its hallowed walls. In fact, he shocked one congregation by firing it during a service. The object of all this gunfire was, of course, frequency analysis of the reverberation.

His findings led him to recommend the fixing of as much damping material as could be accommodated in all the alcoves and recesses which were not easily visible from the Cathedral floor. This might reduce the R.T. by one or two seconds but would be insufficient by itself to make the acoustical conditions satisfactory for speech. The volume of the Cathedral is about five million cubic feet and for the most part its bounding surfaces are Portland stone and marble. With an R.T. of 10 seconds each spoken syllable goes on a two mile journey round the Cathedral, bouncing to and fro off the walls, floor and ceiling before it ultimately fades out.

So something in addition to damping material was needed and Parkin put forward the proposal that long line-source loudspeakers might be effective. His idea was to "shine" the fan-like beam down on to the congregation and if the vertical angle was small very little sound would reach the upper walls and ceiling.

Now the level of reverberant sound is directly proportioned to the power of the sound source and the level is the same throughout the building; it is entirely non-directional.

The direct sound from a non-directional loudspeaker falls in intensity inversely as distance is squared. So there is a critical distance at which the direct sound level falls to the level of the reverberant sound. At this distance the signal/noise ratio is unit. For a reverberation time of 10 seconds this distance —believe it or not—is about 23 feet.

But if a directional sound source is employed nearly all the radiated power is confined to a small solid angle so the energy density is greater at a given distance than from a spherical source of the same power. Simple geometry shows the increase to be proportional to $\frac{1}{\sin\theta}$, where θ is half the vertical angle. Fig. 3 illustrates the point. An all round looking line-source is at the centre of this toroidal wedge. The slice out of the near side has been removed merely to clarify the geometry of the figure. The main beam is

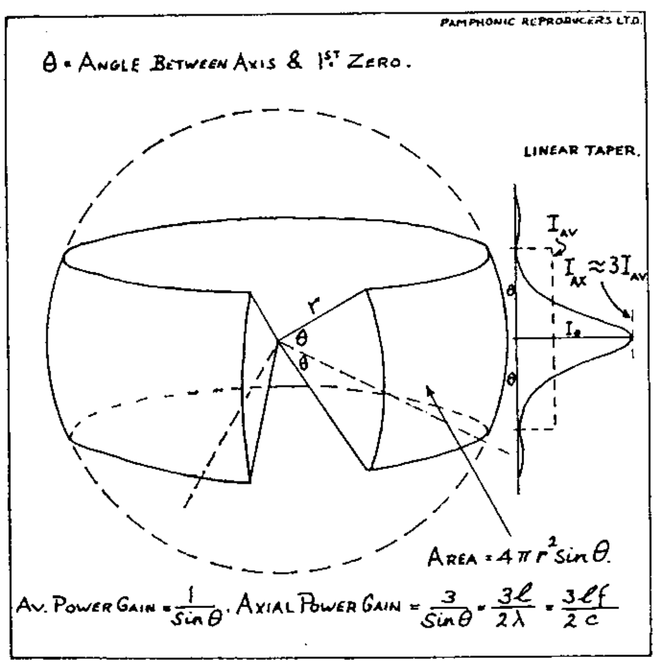

Fig. 3.

confined to the vertical angle 2θ so the radiated sound passes through the equatorial belt of area $4\pi r^2 \sin\theta$. The whole surface of the sphere is $4\pi r^2$ so the average power gain is $\frac{1}{\sin\theta}$. The intensity over the belt varies in the manner indicated on the right of the diagram, and for the particular case shown the level on the axis is three times the average, so the axial power gain over a spherical source is $\frac{3}{\sin\theta}$.

Parkin choose one eleven foot long column to cover the whole dome area. This had an axial gain of about sixteen times at 1,000 c/s, so it could throw the sound four times as far as a spherical source. This gives a range of 4×23ft, which is 92ft. A further advantage accrued from the fact that nearly all the sound fell on the congregation whose absorption coefficient was far greater than the fabric of the building. This resulted in less power getting free to reverberate and effectively increased the range to very considerably more than 92 feet.

The kind of line-source eventually used was quite novel and was the outcome of a combined effort on the part of Parkin's staff and the author's engineers. It is shown in Fig. 4, in front of the wrought iron screen under the chandelier. It was inclined at about 10 degrees from the vertical so that the centre of the beam cut the head level of the people about two thirds of the way across the dome seating area.

The transepts and nave were covered by additional six foot columns. The columns on the South side of the nave are shown in Fig. 5.

These additional columns were the cause of some serious difficulty which arose from the follow-on effect mentioned earlier in connection with the White City. The internal length of the Cathedral is over 450 feet, so sound takes nearly half a second to travel from end to end. The aural confusion caused by the several column loudspeakers could not be tolerated in the face of the already serious

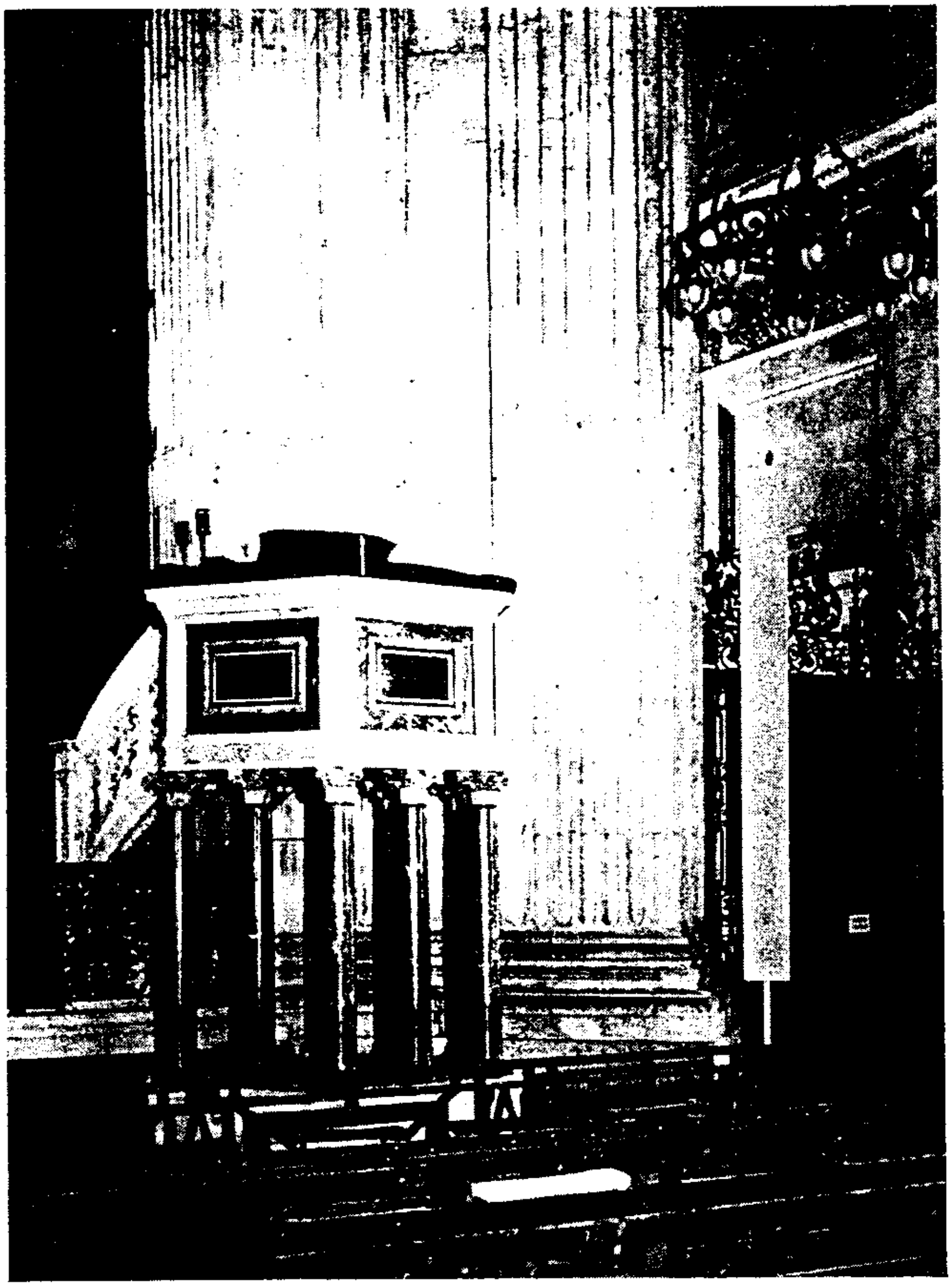

Fig. 4.

problem posed by the reverberation. So a delayed sound system was put in. This was essentially a tape recorder with several playback heads. Each playback head served such loudspeakers as required the same time delay as the transport time of the tape between the record-head and that particular playback head. Thus the loudspeakers spoke at the instant the original sound wave from the human speaker passed by them and this gave a very marked improvement in quality and naturalness. Fig. 6 shows the quantitive improvement obtained.

At the lower left hand corner of the diagram is the pulpit microphone and 11 foot column covering the dome area. Just over 100ft to the right is the first delayed nave column. Then there are two more with augmented delay to the right further down the nave. The dashed curve on the graph indicates the intelligibility without electrical time delays in operation. At the back of the nave it has

Fig. 5.

fallen to 50%, which is very poor. Switching-in the electrical delays resulted in the score indicated by the full line. Anything over 80% is good on a word basis as this results in a sentence intelligibility exceeding 95%.

The foregoing indicates the great advantage to be gained from using a highly directional sound source in reverberant buildings. There are very few in the world with acoustics as bad as St. Paul's, but many modern buildings are designed primarily for musical performance and the optimum R.T. for music is much too long for the clear audibility of speech.

The new Coventry Cathedral is an example. Here the R.T. is just about right for choral music, but line-source speakers were essential, as well as electrical time delays.

Fig. 7 shows one of the portable 8ft 6 in columns installed at Coventry. This one is a very modern presentation in keeping with the contemporary style of the building. The main columns for the lectern, pulpit and altar are 11ft long and in the same tubular style but suspended from the roof.

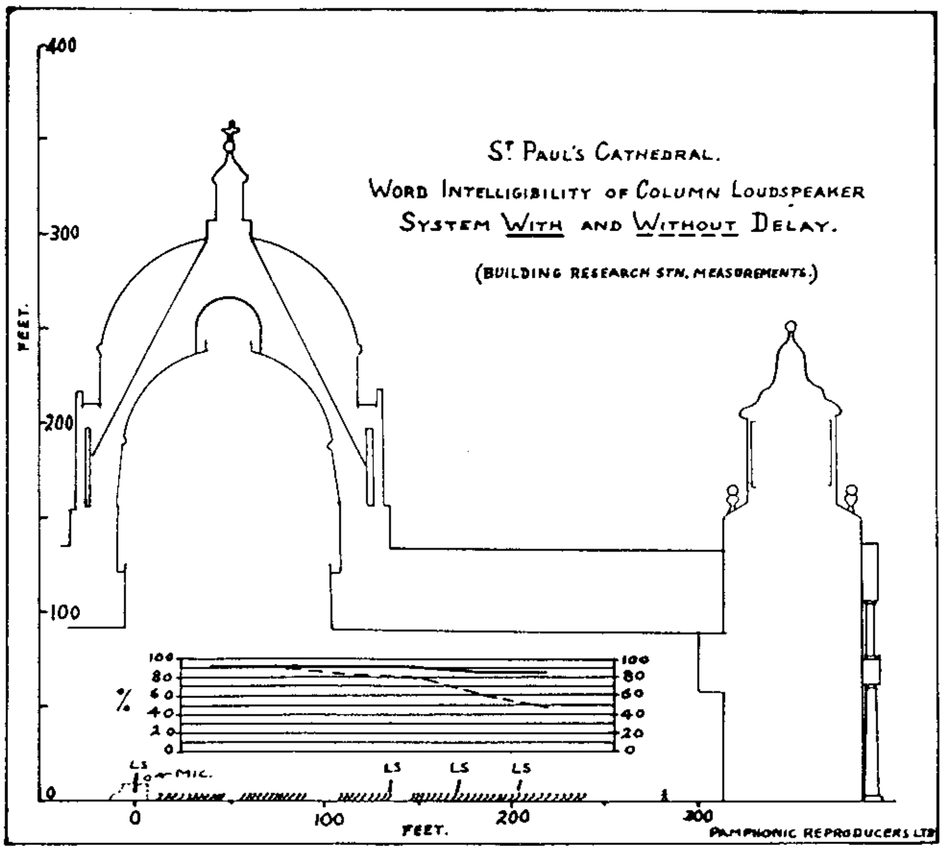

Fig. 6.

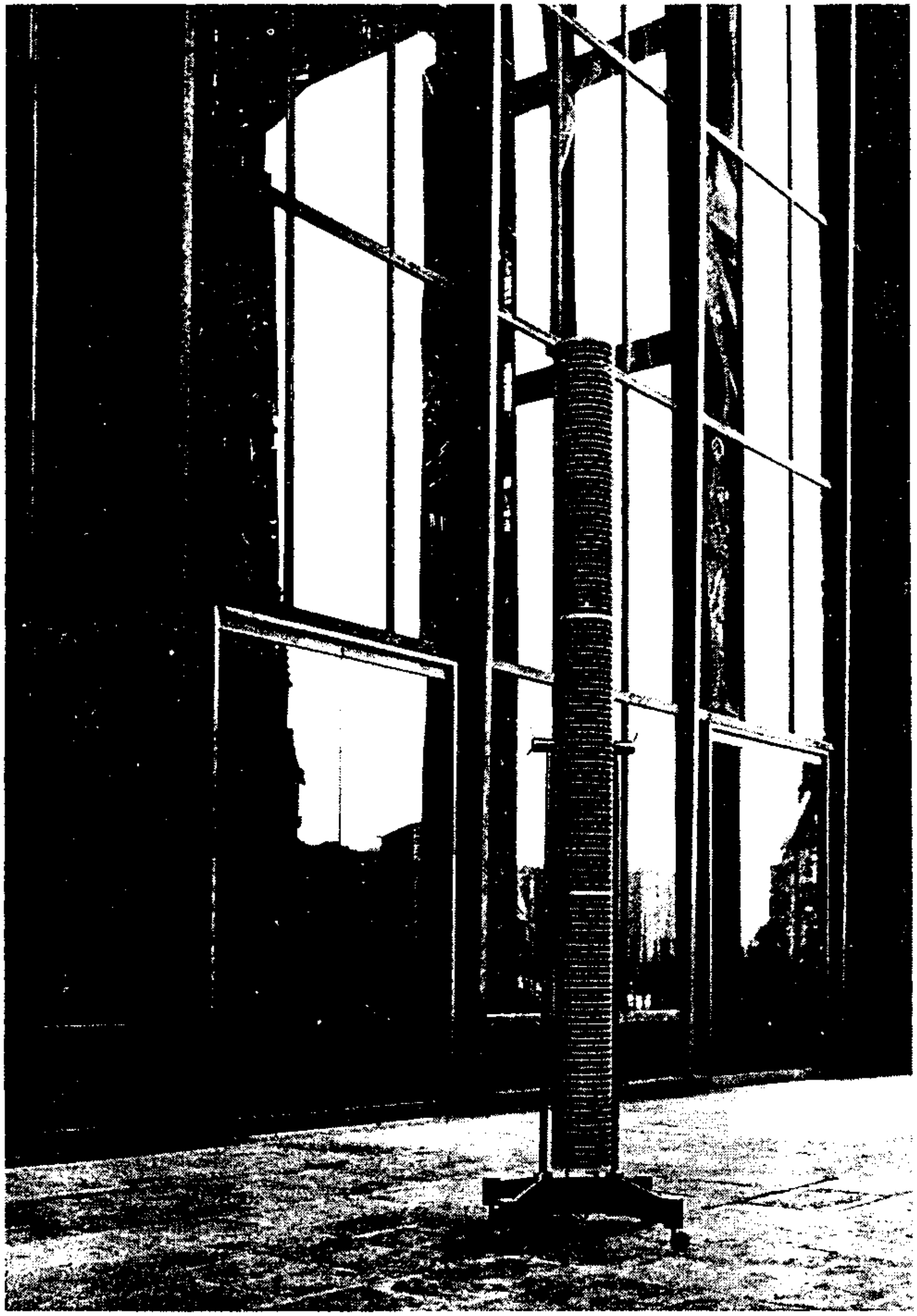

Fig. 7.

Fig. 8 shows the inside of the tube. The main column consists of a large number of 6 inch diameter loudspeaker units fixed in a vertical line. This arrangement is half way between a continuous line source in which the radiating surface would be of uniform width throughout its length and a line of true point sources apart the same distance as the loudspeaker units. There is a slight difference in the polar diagrams of these two extreme types, but in order to understand how a line source column works, it may be assumed that the individual units are true point sources.

In front of the middle of the main column

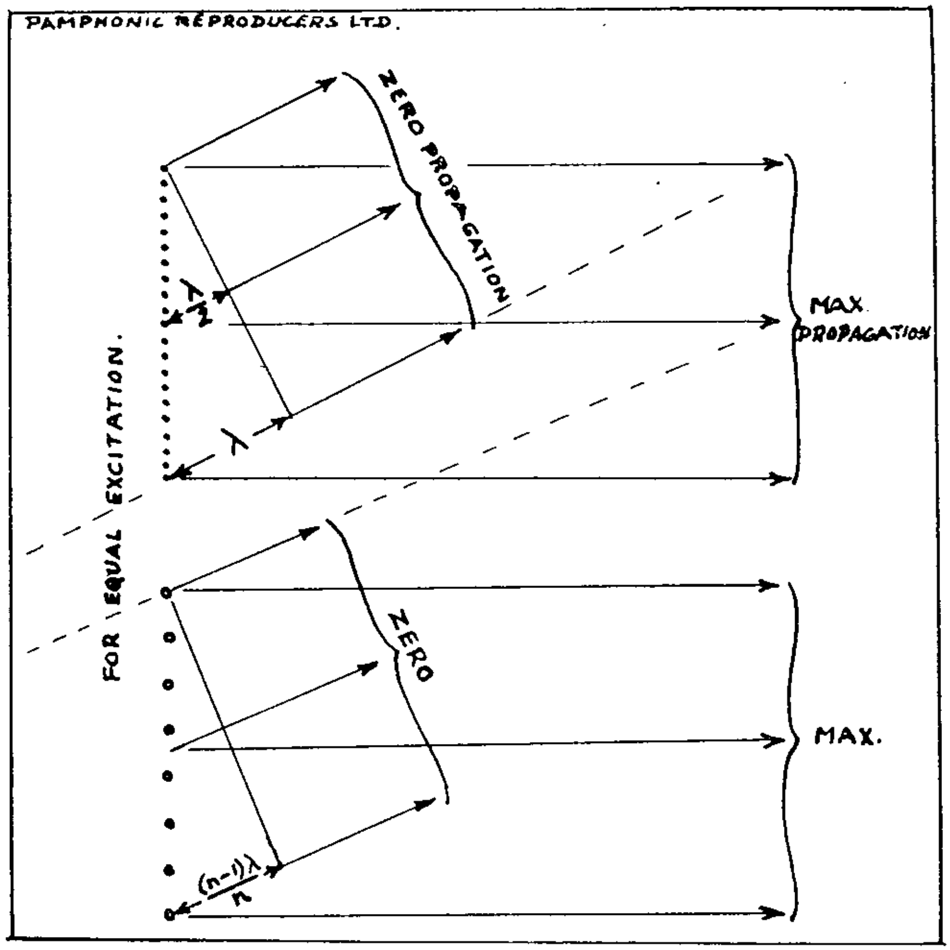

Fig. 9.

Fig. 8.

will be seen the tweeter column; the reason for its inclusion will become clear later on in this article.

Fig. 9 shows, at the top, a line-source composed of a large number of evenly spaced point sources, and below—one composed of only a few point sources. Consider first the source with a large number of closely spaced point sources.

At a great distance away immediately in front, the sound from all the separate sources arrives in phase. So all the sources reinforce one another in the direction exactly at right angles to the column. This will be so at any frequency.

But off the forward axis in the direction of the short arrows the total effect of all the sources will depend on the phase of the waves on arrival at the listener from each source. In the upper diagram we suppose that the bottom source is a wavelength further away from the

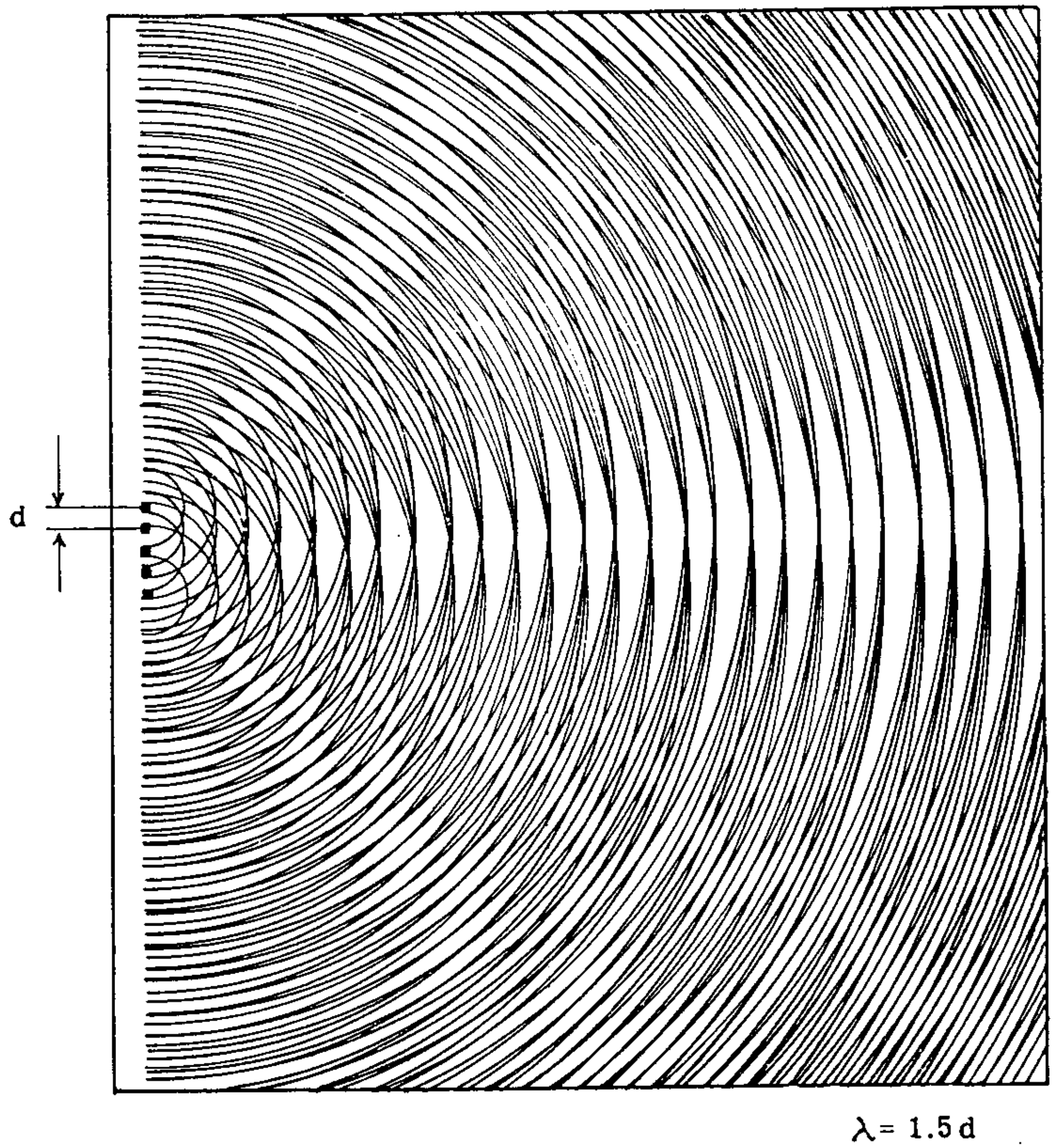

Fig. 10.

listener than the top source. The middle source is, therefore, a half wavelength further away.

For every source in the upper half of the column there is a source in a corresponding position in the lower half which is a half wavelength more distant from the listener. These pairs of sources therefore produce no net effect at the listener's position due to complete destructive interference. Since for every source in the upper half there is a corresponding source a half wavelength further off in the lower half, no sound at all is propagated in this direction.

This condition determines the first null off the axis. There will, of course, be a null at the same angle below the axial direction as the situation is symmetrical above and below the axis.

At this point it should be said that all the sources are operated in-phase with one another and they are all of equal strength.

This pairing of sources succeeds only when the number of sources is large. In the lower diagram (Fig. 9) there are only eight sources and the direction of zero propagation occurs when the bottom source is 7/8 of a wavelength further from the listener than the top one. This distance is always $\frac{n-1}{n}$ of a wavelength. So for four sources the distance would be $\frac{3}{4}$ of a wavelength.

As the angle off the axis increases we go through a series of secondary beams of lower intensity than the main beam. The number of secondary beams depends on the frequency and the length of the column.

Fig. 10 illustrates five sources on the left hand side. Each of these is the centre of a series of semicircles which represent sound waves being radiated. The wavelength is one and a half times the spacing in between each source. The main forward beam is very distinct. All the sources reinforce immediately forwards. The first secondary beams are just discernible radiating out from the centre of the column at about 45° above and below the forward axis. They are not so easy to see as they are a good deal less intense than the main beam. Another point to note is the straightness of the composite wavefront developing a little distance in front of the column. This shows very well the cylindrical form of the radiated wave.

Now, these secondary beams point in unwanted directions. The ones that go up into the air contribute nothing to the direct sound at any normal listening position; they only add to the reverberant energy in the building.

How they arise and how to suppress them will now be considered.

On the left of Fig. 11 there is a line of seven sources spaced distance "d" apart. To start with we consider the lower half of the column. This means chopping the centre source in half. At a remote point in the direction of the arrows each source on its own produces a sound pressure amplitude of A and the half centre source gives $\frac{A}{2}$. If the centre source is taken as phase reference, then at the remote point, the sound from the source below lags in phase due to the additional distance it has to travel. The extra distance, from the

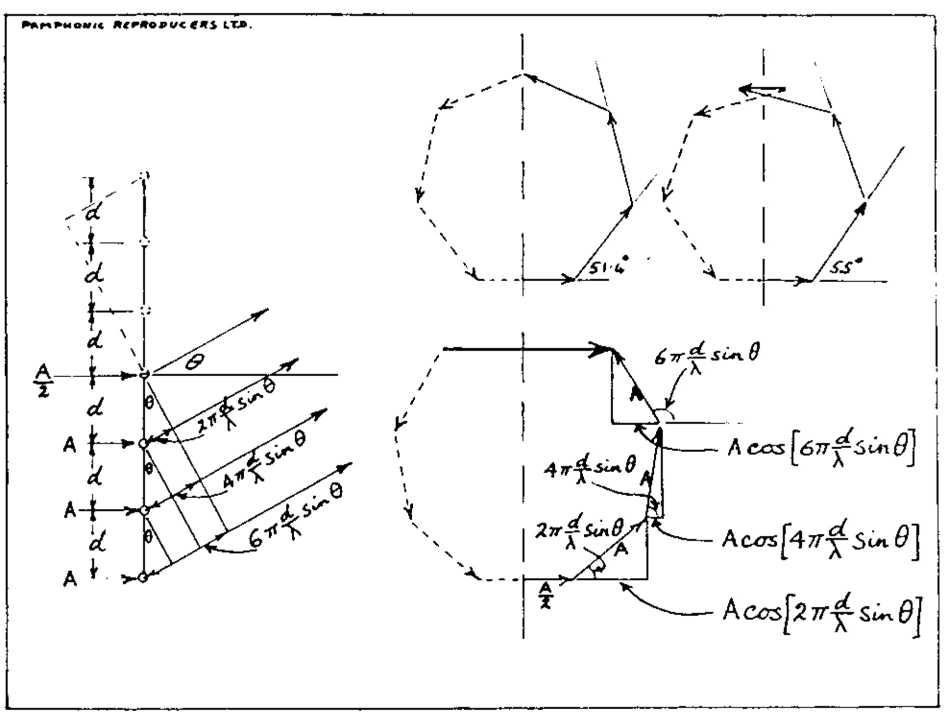

Fig. 11.

geometry, is $d \sin \theta$. To turn this into a phase lag it must be multiplied by $\frac{2\pi}{\lambda}$ where λ is the wavelength. So the phase lag of this source is $\frac{2\pi d}{\lambda} \sin \theta$ radians.

The phase lag of the next source down is twice this amount, as will be readily seen from the diagram. So this one lags on the centre source by $\frac{4\pi d}{\lambda} \sin \theta$ and by similar reasoning the bottom source lags by $\frac{6\pi d}{\lambda} \sin \theta$ radians.

The situation is shown in terms of vectors at the bottom right of the figure. The $\frac{A}{2}$ vector is the horizontal reference at the base of the polygon. Joined to its arrow head is the vector of length A representing the source next below the centre one. It is swung round at an angle $\frac{2\pi d}{\lambda} \sin \theta$ to the reference vector. To the arrow head of this second vector we attach the vector of the next source down and orient this one at an angle of $\frac{4\pi d}{\lambda} \sin \theta$ to the horizontal. The bottom source of the column is the next one at $\frac{6\pi d}{\lambda} \sin \theta$ to the horizontal.

The top half of the column can be dealt with in exactly the same way and the other half of the centre source vector together with the vectors for the other sources are shown dotted on the left hand side of the polygon. The large horizontal vector at the top joining our string of small vectors is the resultant at the distant listening point. It will be seen that it is *in phase* with the centre source. In fact, the resultant is always either *in phase*, 180° *out* of phase or *zero* with respect to the centre source.

If the listener moves round a bit and increases θ he will reach an angle where all the individual vectors form a closed polygon. The resultant is then zero and he is at the angle of the first null. The top left hand polygon shows this.

If he now moves round a bit further—that is, he increases his angle θ off the axis still more—we get the situation shown in the top right hand polygon. Here the end vectors overlap and there is a small resultant in opposite phase to the reference source vector. This is the first secondary beam coming up. It is now easy to visualize how this secondary beam will wax and wane as θ is increased still further. The vectors proceed to overlap still more and the maximum of the secondary beam occurs when the diagram becomes a pentagon. The resultant is then the vector joining the "eaves" of the pentagon as these are the locations of the head and tail of the end vectors. The length of this resultant is 1·62 times that of an individual source and since there are seven sources it is approximately 25% of the strength of the main beam on the axis. So the first secondary beam is 12dB down relative to the main beam. This is not always so. The strength of the secondary beams depends on the number of sources in the line and on whether they all receive uniform excitation or not. This last point is an important one that will be dealt with later on.

Attention must now be drawn to the cosine terms at the bottom right hand corner of the diagram. These give the horizontal components of the vectors and it is the algebraic sum of these which gives half the resultant vector. There is an identical set of cosines for the dotted vectors on the left hand side and these account for the other half of the resultant.

So the resultant can be expressed as the sum of the cosine series:

Resultant of all seven sources $= A_\theta$.

$$A_\theta = 2\left\{\frac{A}{2} + A\cos\left\{\frac{2\pi d}{\lambda}\sin\theta\right\} + A\cos \rightarrow\right.$$

$$\left\{\frac{4\pi d}{\lambda}\sin\theta\right\} + A\cos\left\{\frac{6\pi d}{\lambda}\sin\theta\right\} \quad \text{(i)}$$

If we substitute a for $\frac{2\pi d}{\lambda}\sin\theta$

we get $A_\theta = 2\left\{\frac{A}{2} + A\cos a + A\cos 2a + A\cos 3d\right\}$ (ii)

Quite obviously if the number of sources is increased a further cosine term is added for each pair of sources symmetrically disposed with respect to the centre of the column.

The sum of a series of this sort is:

$$A_\theta = A_0 \frac{\sin\left\{\frac{n\pi d}{\lambda}\sin\theta\right\}}{n\sin\left\{\frac{\pi d}{\lambda}\sin\theta\right\}} \quad \text{(iii)}$$

where A_θ is the effect at a distant point at an angle of $\theta°$ to the forward axis and A_0 is the effect at the same distance *on* the axis, n being the number of sources.

When the spacing (d) is very small compared with the wavelength, $\frac{\pi d}{\lambda}\sin\theta$ will also be small and since for small angles the sine of the angle is approximately equal to the angle itself, the denominator of (iii) can be written simply as $\frac{n\pi d}{\lambda}\sin\theta$. If there are a large number of sources as well, then nd = the column length (l) and this enables equation (iii) to be written as:

$$A_\theta = A_0 \frac{\sin\left\{\frac{\pi l}{\lambda}\sin\theta\right\}}{\frac{\pi l}{\lambda}\sin\theta} \quad \text{(iv)}$$

This would be the formula applicable to a continuous line source consisting of a long integral diaphragm. The individual sources would be infinitesimally spaced and infinite in number.

In order to determine the kind of radiation pattern produced by a particular number of sources spaced at a given distance apart it is much easier, and more instructive, to draw the cosine waves of equation (ii) on graph paper rather than the vector diagram. Cosine waves drawn freehand are fairly adequate and Fig. 12 illustrates a set.

This particular set shows the result of making the individual source strength diminish linearly from the centre of the column towards each end. That is, the coefficients A in the series progressively diminish as each term is added. At the top of Fig. 12 can be seen the set of coefficients. They indicate respectively the voltage applied to each of the seven sources. The centre source gets 4 volts, the ones next above and below it 3 volts, the next but one above and below 2 volts and the end sources 1 volt. *Now the extreme left hand ordinate is where $\theta = 0$ i.e., on the forward axis. The abscissa is in terms of $\frac{2\pi d}{\lambda}\sin\theta$, not θ itself. A θ scale could be applied to it and it would stop somewhere along the abscissa where $\sin\theta = 1$, i.e. where $\theta = 90°$. The set of curves is a universal set for any value of d, λ or θ.

Each of the cosine waves represents one of the cosine terms of the series shown in Eq. (1) with its own particular value of A. The sum of these cosine waves at any point along the abscissa gives the resultant effect at that particular value of $\frac{2\pi d}{\lambda}\sin\theta$. The dotted curve is, in fact, the sum of the cosines and gives the shape of the radiation pattern.

Only half of the main beam is shown, but the other half is to be imagined as an exact mirror image joined on to the left side of Fig. 12.

* *The sources are assumed to be moving coil loudspeakers. These are mass controlled devices and acceleration of diaphragm is proportional to coil current which is in turn proportional to applied voltage. Sound pressure at distant point is proportional to diaphragm acceleration and hence is proportional to applied voltage.*

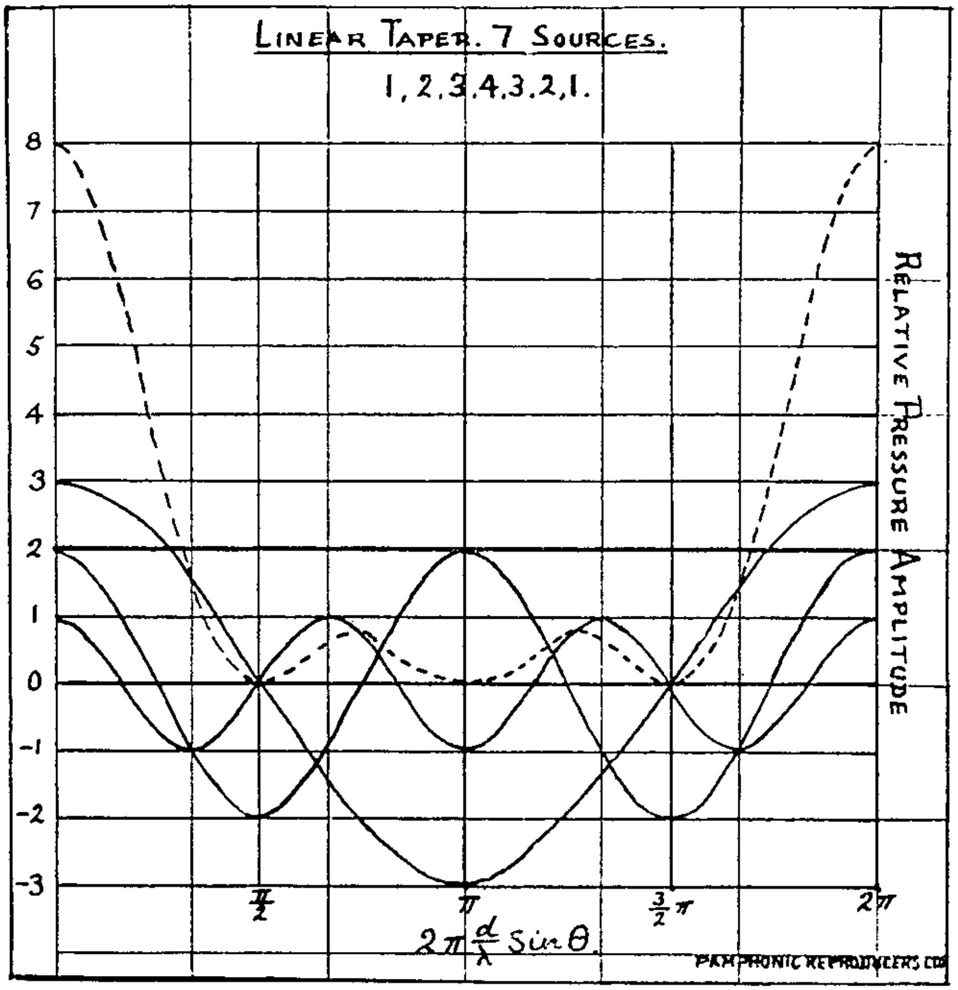

Fig. 12.

Since the *amplitude* of the cosine curves corresponds to the strength of the particular source it represents, the thick straight line running across from the figure 2 on the ordinate scale represents *half* the contribution of the centre source, its whole strength being 4. It will be remembered that the centre source does not change its phase as θ varies, and that is why its contribution is a straight line. The longest cosine wave represents the contribution of the source next to the centre one and its amplitude is 3 in accordance with the voltage feed chosen for it. The other two waves correspond to the remaining two sources and again have amplitudes proportional to their feed voltage.

As we move down the dotted line on the left, representing the main beam, we reach a zero or null point, and as we move further to the right it turns upward again and we go over a small hump which is the first secondary beam. Then we get another null and a second secondary beam and then another null. Then it starts to rise steeply into what looks like another main beam. It is, in fact, a main beam way off the axis and it occurs at an angle which makes the path difference to adjacent sources exactly one wavelength. Such a situation occurs, for instance, when the wavelength of the radiated sound exactly equals the spacing distance between the sources. Then you get two additional main beams, one vertically upwards and the other vertically downwards. This is a very undesirable state of affairs. Throwing all this power into the upper parts of the building is just what should be avoided. The only way to avoid this situation is to put in a low pass filter to chop-off at a frequency whose wavelength is still a good deal longer than the source spacing. If the sources are spaced nine inches apart then the filter must cut at say, 1,000 c/s, since the wavelength at 1,000 cycles is about a foot.

Above 1,000 cycles we must transfer to another set of more closely spaced units. This is where the tweeter column comes in, which was referred to in connection with the Coventry job. The tweeter column, as well as having more closely spaced units, is a quarter of the length of the long column. By shortening in this way the main beam is, prevented from becoming too sharp at high frequencies.

For speech reinforcement in reverber[a-] surroundings, the frequency band from [2-] cycles to 4,000 cycles is adequate for go[od] natural reproduction. In fact, any extensi[on] of this frequency range actually detracts fr[om] naturalness in that kind of invironment.

So, well designed columns for such app[li-]cations cover just four octaves. The lo[ng] column handles from 250 to 1,000 cycles a[nd] the tweeter column from 1,000 to 4,000 cyc[les] and the amplifying system should roll slowly above 4,000.

Having dealt with the big off-axis bear[ms] let us now consider the small secondaries.

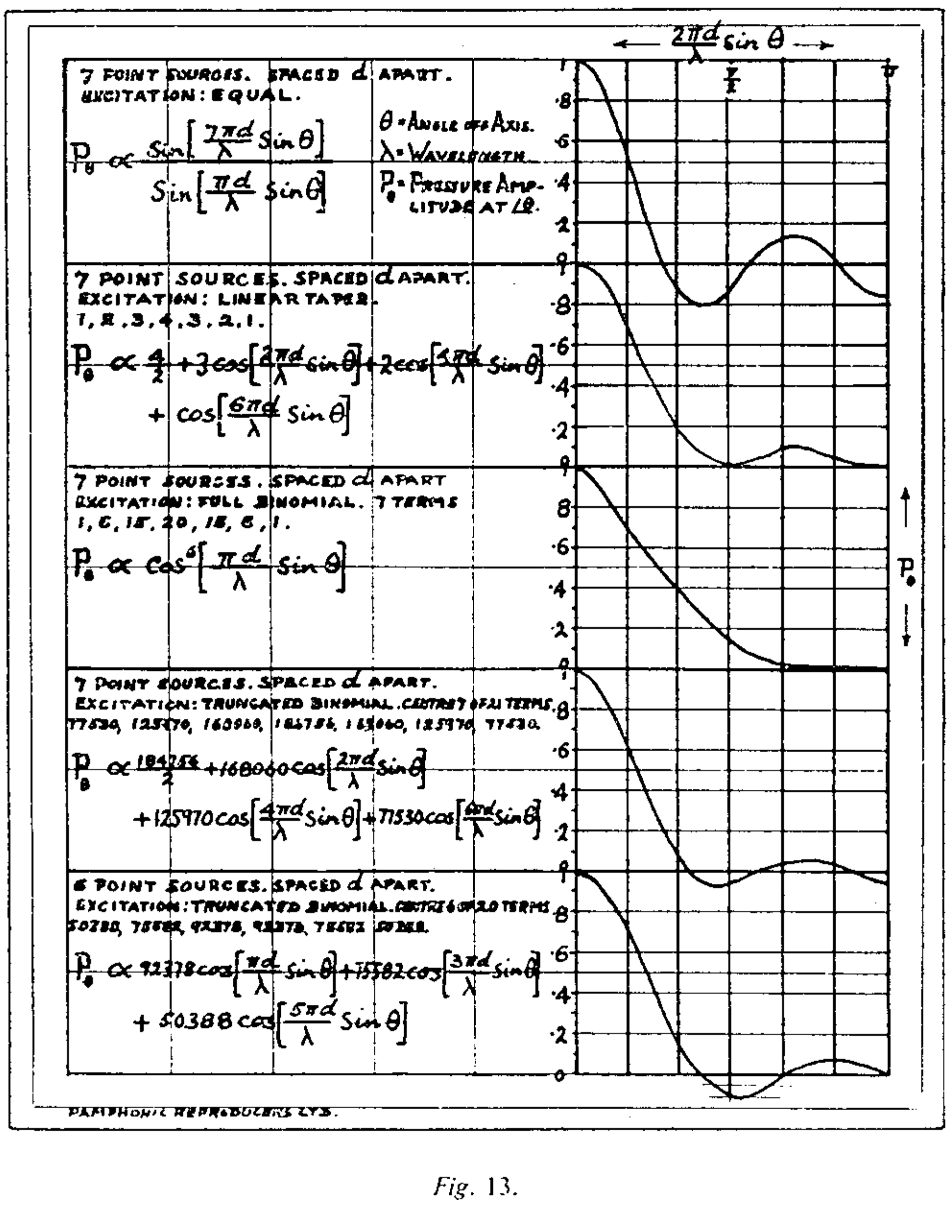

Fig. 13.

The linear taper feed does reduce the strength of these secondaries very considerably. The degree of suppression compared with a uniformly excited set of sources can be gauged from Fig. 13.

At top right is the pattern for seven sources *uniformly* excited. The first secondary beam is the trough below the zero line. It is opposite in phase to the main beam as shown in the vector diagram of Fig. 10. The next secondary is positive and almost as big. Then the beginning of the third secondary can be seen. Now these, as already mentioned, although much weaker than the main beam frequently cause discrete echoes off the ceiling of a building as well as adding to the general reverberant energy. It is very worth while taking all possible measures to suppress them.

A linear taper feed, which has been looked at in detail in Fig. 12, is the second curve from the top in Fig. 13. It will be seen how very much less vicious are the secondaries in this case than in the curve above. It can be seen that tapering has broadened the main beam, entirely suppressed the first secondary, and very much reduced the third. In fact, between 0 and π radians it has only one secondary whereas there are two and a half between 0 and π in the case of uniform excitation.

The next curve down—the middle one of the six—shows no secondary beams at all—this is obtained by feeding the sources in proportion to the coefficients of a binomial series of seven terms. To find the feed voltages for n sources the term $(1 + 1)^{n-1}$ is expanded and the resulting numbers are the voltages in the order they must be applied to the sources. The effect at any angle θ is given by the expression to the left of the curve.

However, it should be noted that the main beam angle is now very large and a much longer column would be needed to get the same beam angle as is obtained from a linearly tapered beam.

It will be seen that the end sources receive only 1 volt whereas the next ones to them receive 8 volts and the centre one 20 volts. The end ones hardly earn their keep—it is hardly worth connecting them up for the sake of such a tiny input. So a set of, say 21, binomial coefficients are taken and the seven small ones at each end are thrown away, leaving only the centre seven sizeable terms. The next curve down shows the result of such an operation. It will be seen that throwing away the end coefficients has raised some secondary beams. But they are weaker than those in the linear taper case, and the beam angle is actually narrower. This technique is called truncated binomial tapering and is patented for this application by Pamphonic. The lowest curve is for six sources instead of seven. The end seven terms of a 20 term series were thrown away. The reduction by one of the number of sources has made a marked difference to the first secondary beam. In this case the main beam is wider because the column length is shorter by one source.

Incidentally, these enormous coefficients do not mean similarly enormous voltages. They simply indicate the *relative* voltage levels of the sources. They are, in fact, the actual magnitudes of the coefficients for a binomial series with the number of terms indicated.*

To summarise: firstly, secondary beams should be suppressed as far as possible, as those which stick up in the air above the main forward beam give rise to echos, aggravate the reverberation and contribute nothing to the direct audible sound.

Secondly, all frequencies whose wavelength is equal to or shorter than the source spacing must be filtered out, otherwise there will exist, in addition to the secondary beams, high intensity main beams off the axis and these create a great deal more commotion than the secondaries which have been suppressed.

Thirdly, quite substantial suppression of the secondaries can be achieved by adopting a non-uniform feed to the several sources composing the column. Tapering off the feed in a linear manner from the centre to the ends of the column gives pretty good suppression. Tapering from the centre to each end in accordance with the binomial coefficients of

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* *The most efficient way of obtaining the particular set of voltages required is from a suitably tapped transformer.*

a series having the same number of terms as there are sources gives complete suppression but rather a fat main beam results. A truncated set of binomial coefficients gives a somewhat sharper main beam than a linear taper and the secondaries are less pronounced.

Fourthly, the frequency range should be divided between a long and a short column within the one loudspeaker to prevent the beam from getting too sharp at high frequencies.

It may be of interest to know that to maintain beam angles reasonably constant at about 30° over the frequency range 60 cycles to 16 Kc/s you would need four columns with appropriate filters. The longest would be 44ft and the shortest 8¼in long!

So far we have considered the angular distribution of the sound as it would appear to some mysterious observer at infinity, who enjoyed absolutely parallel rays of sound from all parts of the column.

In practice one is much more interested in the near field where the parallel ray theory is untenable. To what extent does this more practical approach upset what has been said so far?

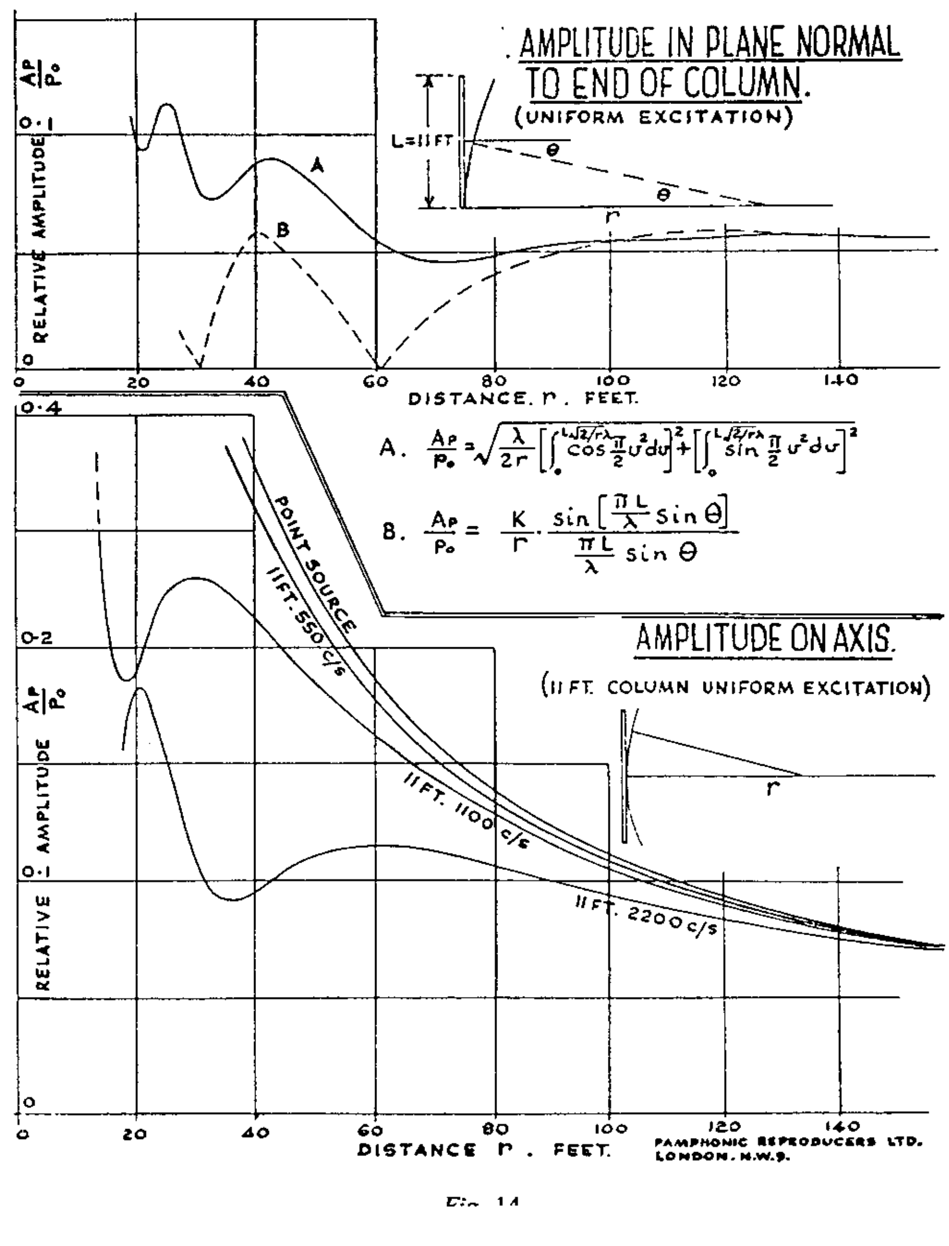

Fig. 14

At the top of Fig. 14 will be seen an eleven foot column with a line drawn from its base out towards the right. What will the sound pressure be at any point along this line? If we use the parallel ray formula the pressure should vary according to curve B. This curve is drawn for a frequency of 1·1 kc/s and the line-source is taken as "continuous" as explained in connection with Eq. (iv).

There is a null point at 60ft and another null at 30ft. These are not only within the range in which we are interested in practice, the position of this line—out from the foot of the column—just about coincides with the head level of the audience. So this is a most important case. To get the true sound pressure we must integrate over the surface of the column taking into account the difference in the angles from the top and bottom of the column to the listener.

Considered as a continuous line source, Equation A in Fig. 14 is the integral that has to be evaluated for the non-parallel ray situation. The result of this operation is curve A. It will be seen that the nulls have disappeared and the secondary beam maxima are now just ripples in a smooth curve. Only at distances exceeding 100 feet does the parallel ray theory start to be tenable at this particular frequency.

Another point worth noting is the constancy of sound pressure with distance. It alters by little more than 6dB beyond 20 feet. This is due to the inverse square law being compensated by the diminishing angle θ. As θ diminishes the listener gets nearer the main beam axis and the sound pressure increase due to this just compensates for the increased distance away.

This effect is one of the most valuable features of a line-source loud-speaker. Listeners close to it and listeners far away enjoy almost the same sound level.

The lower part of Fig. 14 gives the *axial* variation of sound pressure with distance. The top curve shows the inverse square law applicable to a point source of sound. The bottom curve is the result of integrating over the eleven foot column for a frequency of 2·2kc/s. The curve above is for 1·1 kc/s. The next one above that is for 550 c/s and it will be seen that this one is very nearly an inverse square law as close as 30 feet.

So it will be evident that some of the desirable characteristics of a line-source arise out of the breakdown of the parallel ray theory in the immediate neighbourhood of the column.

Before going on to consider open-air applications, mention should be made of the radiation from the backs of the loudspeaker units. This should be attenuated either by total enclosure, in which case adequate absorption material must be contained within the enclosure, or by allowing some back radiation to escape through an acoustically "lossy" backing to the cabinet. The latter method can be made to give a cardoid distribution in the horizontal plane by the interference pattern set up between front and back radiation.

The general situation out of doors is quite different from indoors and the important aspects of open air sound coverage will now be covered very briefly. Secondary beams get lost in the clouds and nobody cares. Even if they do get reflected down again they cause no trouble, in fact, the secondaries both above and below the main beam can be quite useful as will be seen later on.

One of the features of a uniformly excited column is its comparatively sharp main beam. This is very desirable when the sound has to be fired over a great distance. One can also pump much more power into a uniformly excited column than one which has a tapered feed. The centre unit on a tapered column can be driven at its maximum power handling capacity, but because of the taper feed all the remaining units are relatively under-powered. So the input power of a uniformly excited column can be much greater—2·54 times greater in the case of a seven unit column rising to three times greater for a column with an infinite number of units, i.e., a continuous line-source.

As was the case at the White City, a single source of sound, if it can be thrown far enough, generally gives a more economic system because of the saving in wiring cost. But the snag that usually limits the range is

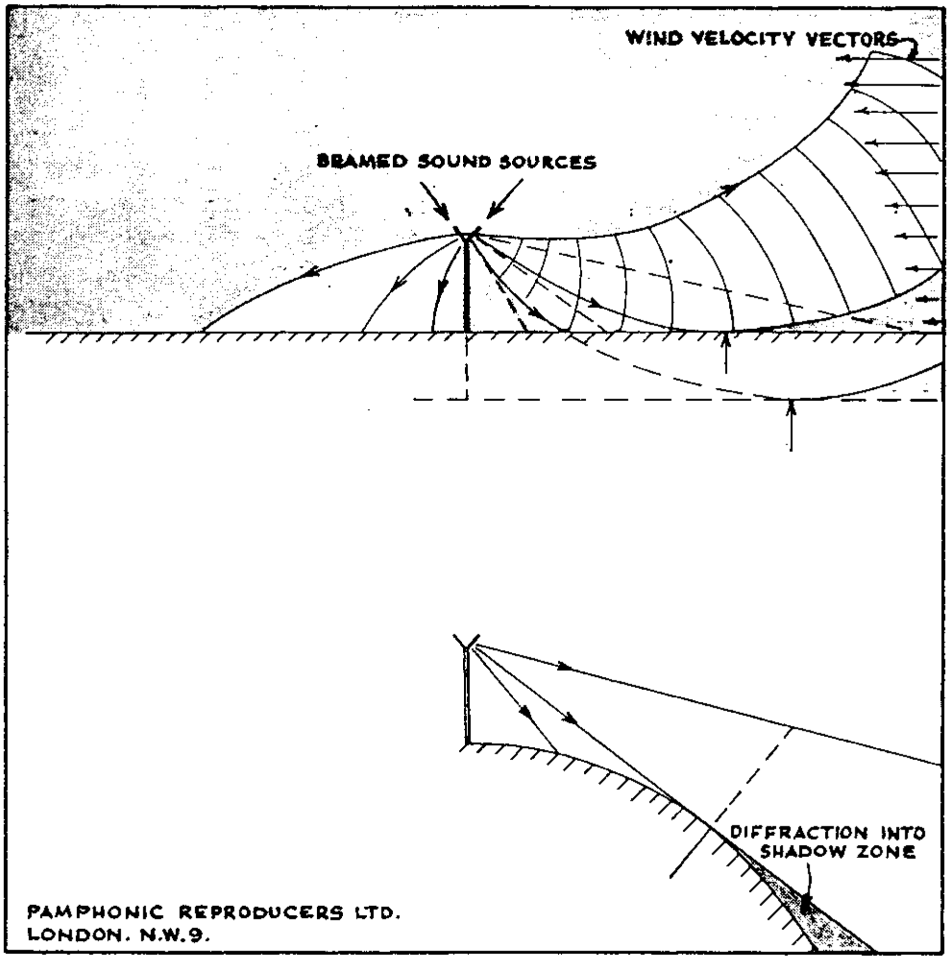

Fig. 15.

the wind. An adverse wind—that is one which blows against the direction of sound propagation—bends the sound up into air.

In Fig. 15 the top picture shows diagramatically two columns inclined downwards on the top of a pole—one facing each way. The three dashed lines radiating downwards from the right hand column delineate the main beam and its axis. The ground is "illuminated" with sound, from close to the pole out to the extreme right of the picture when no wind blows.

If a wind blows from right to left and its velocity increases with height in the way indicated by the wind vectors drawn on the right, sound near the ground will make more progress in a given time than sound at a greater altitude.

The wavefronts, therefore, begin to lean over backwards and since they progress in a direction always normal to the wavefront there comes a time when they leave the ground altogether. The mechanism of this can be seen from the diagram. There is in fact a limiting ray which grazes the ground at the arrow point against the hatched ground. Beyond this point there is a shadow zone in which the sound level will drop very severely.

On the other side of the pole—the downwind side—the rays are bent downwards and audibility is limited to a distance at which the top side of the beam hits the ground. Beyond this point secondary beams will fall and these will extend the range somewhat. A point source, i.e., an omnidirectional one, will reach no further to windward but sound cover down-wind will be extended because there is no "top of the beam".

Now, below the hatched ground level there is a dotted line which represents where the ground would be if the pole were higher. You will notice that the limiting ray grazes the ground at a point somewhat to the right of where it did before, so the shadow zone is further away. Indeed, it can be shown that the range at which the shadow zone starts is roughly proportional to the square root of the height of the loudspeaker, so it is essential with outdoor systems to get the loudspeaker as high as possible off the ground. However, there are some effects which relieve the situation in the shadow zone. At the bottom of Fig. 15 is a distorted representation of the upper diagram. Here, instead of curved sound rays and flat ground, we have curved ground and straight sound rays. This is quite legitimate and is frequently done in radio propagation diagrams in which ray bending has to be taken into account. Now in this lower picture it is clear that the curved ground forms a screen which casts the shadow. Whenever there is a screen in the way of a series of plane waves of this kind, the shadow is not absolutely sharp edged; there is diffraction down into the shadow and this diffraction is quite considerable when the wavelength is long. One is used to dealing with diffraction in terms of light, but with light, of course, the wavelength is millionths of an inch not feet and the amount of light which gets into the geometric shadow is minute. With sound the amount which gets into the shadow zone is very valuable in extending the range of a loudspeaker, but it does mean that the intensity of the source must be high. If you calculate how far away is the shadow zone from a loudspeaker placed at, say, a height of 20ft with a wind of 30 m.p.h. with the usual velocity gradient, you will be surprised to find that it is about 100 yards away up-wind. But we do know from experience that we can hear a good deal further away than this and the reason is that a lot of the sound is diffracted into the shadow. Moreover, the wind is never streamline, it is usually turbulent, and as such it reflects and refracts the sound and scatters it into the shadow zone.

A temperature gradient has a similar effect on the sound as a wind gradient. The normal daytime temperature falls with increasing height. Sound travels faster in warm air than it does in cold air so the sound near the ground where the air is warm travels faster than it does higher up and this bends the rays upwards, as does an adverse wind. Now when we have temperature inversion, which we frequently get on a Winter's day and also in the evening, especially in the Summer, the sound bends downwards in the same way as it does on the down-wind side of a sound source. This is very noticeable on a Summer evening, when sitting out in the garden cocks crowing many miles away may be heard. Similarly all the sounds in the countryside around are heard extremely clearly; this is caused by temperature inversion. All the sounds which are generated in the local farmyard go up into the air and are bent down again by the inverted temperature gradient into the garden. Sometimes the temperature and wind effects cancel one another out, for instance, an adverse wind and temperature inversion can leave the sound completely

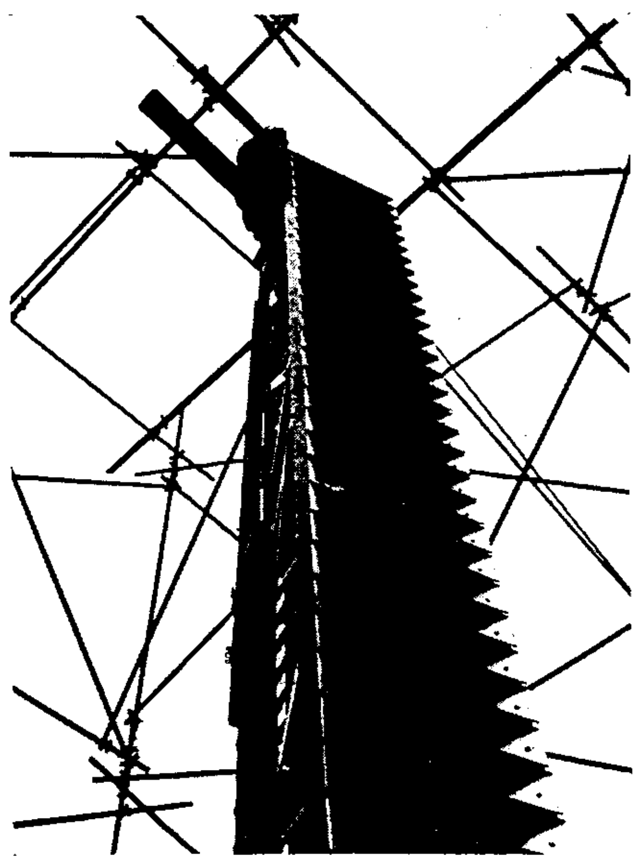

Fig. 16.

unbent and it will propagate in straight lines. On the other hand, the normal daytime temperature gradient, together with an adverse wind, can bring the shadow zone very close indeed to the source. Having regard to what has just been said about the need for a really intense sound source out of door, Fig. 16 shows the scale of a long-range outdoor column. It is 20ft long, takes a kilowatt of audio input and on speech it would give fair audibility up to two miles. This particular column has been heard quite clearly at three miles, but, of course, at that distance the weather conditions do affect it

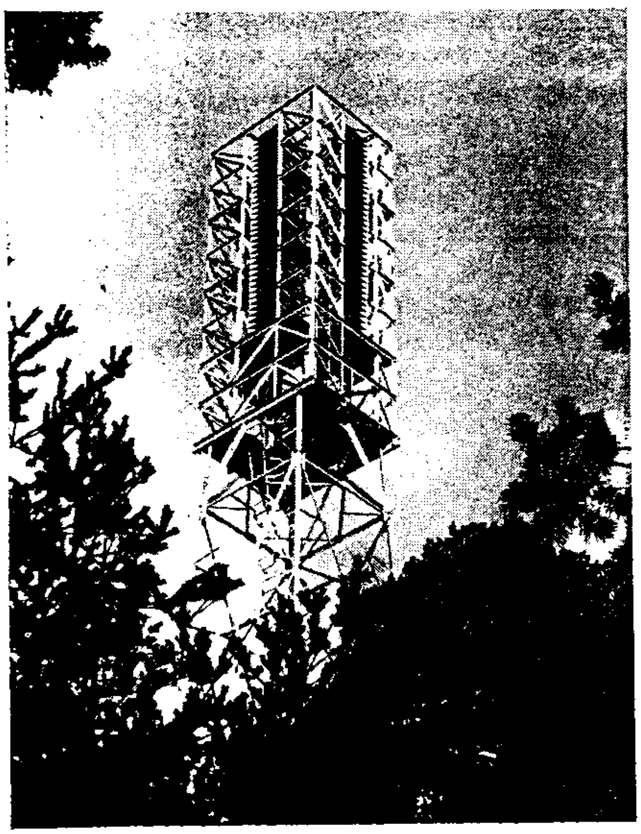

Fig. 17.

very considerably. It is being prepared for tests and is being rigged up in scaffolding so that it can be twisted round and its polar diagram checked in the horizontal plane. It weighs about a ton. Three columns of this kind are installed at the top of a tower, a little over 100ft high, at Broadmoor in Berkshire, Fig. 17. Broadmoor, being an institution where the criminal insane are kept, needs a device whereby the local population can be warned in the event of a break-out. The actual warning signal at Broadmoor is not speech, it is a tone, in fact, it is a mixed tone consisting of a frequency of about 450 c/s with another of about 470 c/s. The idea of using two frequencies is to prevent nulls forming in the polar diagram, particularly in the horizontal polar diagram. There are three of these columns at the top of the tower and because they are separated horizontally in

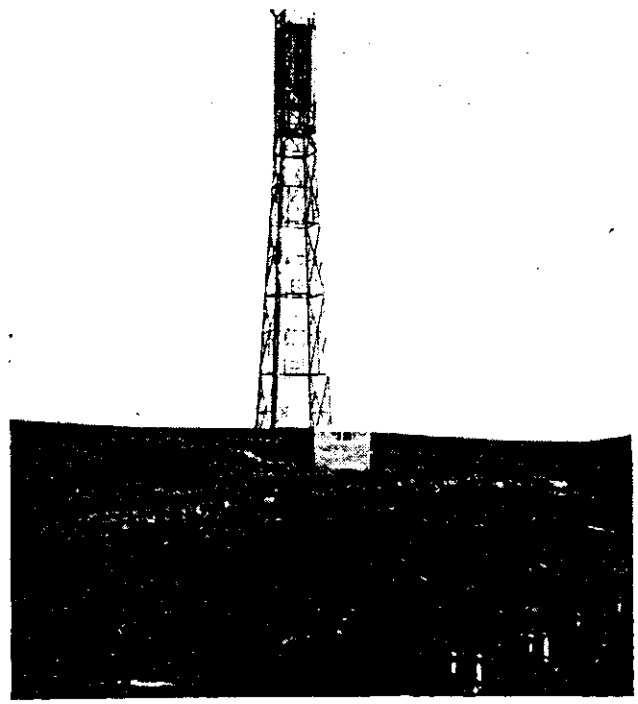

Fig. 18.

space, they would generate an interference pattern in the horizontal plane with several nulls if only one frequency were used.

With a tone of this kind the range is three miles at nearly all times. Some statistical work was done on the wind velocity at Broadmoor and it is considered that on only 16 days of the year would it be difficult to hear this warning signal at a distance of three miles.

Fig. 17 shows the tower and two of these line-source columns. There is a ladder up the tower and platforms at various levels so that the units can be reached for servicing purposes. Each column is composed of flat mono-planar horns with pressure units at the throat. It approximates very closely to a continuous line-source, as opposed to one made up of discrete sources. Fig. 18 shows another of these giant loudspeakers, again at the top of a 120ft tower. Each column is 20ft long and there are four of them directed roughly North, South, East and West. This particular installation is at Spadeadam in Cumberland, the Blue Streak Missile Test Site. These columns do handle speech and this is one of the essential requirements of this particular warning system.

The quality of speech is phenomenal. It can be heard a couple of miles away over the moore where the silence is normally quite impressive; and it is really most uncanny to hear good quality speech rolling across the moors apparently from nowhere.