ELECTRICAL ALIGNMENT OF ACOUSTIC SOURCES

(Passive Time Delay Compensation)

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DEFINITIONS:

Drive Unit

A moving coil/moving piston transducer designed to cover a specific portion of the audio frequency band with the minimum number of acoustic, mechanical and electrical compromises.

Loudspeaker

A collection of drive units arranged to optimally cover the whole audio frequency band. Each drive unit is fed from an electrical filter network designed to complement the acoustic response of the unit.

Crossover

A collection of electrical filter networks designed to split the whole audio band into frequency bands suitable for feeding to individual drive units.

Amplitude

The acoustic sound pressure generated by a drive unit or loudspeaker system at any particular frequency. Constant amplitude infers constant sound pressure. Amplitude response refers to a plot of amplitude with increasing frequency - the frequency usually shown on a logarithmic scale because of the nature of music and natural sounds and the response of the human ear.

Phase

A measure in periodic or oscillating waveforms of the movement of one particular portion of a waveform with respect to another portion or a fixed reference. One complete cycle of a wave corresponds to 360° phase shift with reference to the fixed starting point of the wave. (Note $360^\circ = 2\pi$ radians)

Frequency

The number of complete oscillations contained within a time of one second, for a periodic or oscillating waveform. Usually the units are Hz or cyles per second (CPS). Note that since one cycle is also 2 radians as above, frequency is also referred to as radians per second designated W (omega) in electrical formulae. $W = 2\ f$ where f is cycles per second or Hz.

The concept of phase is somewhat difficult to grasp - especially in the field of acoustics. However it is important to have some concept of phase however simplified to appreciate the concept of time delay since the two are related.

An illustration of phase

Consider a bicycle with large visible tyre valves. With the bicycle at rest the tyre valves are set so that both valves are pointing straight down at the ground. The bicycle is wheeled along for one revolution of the front wheel until the front valve is at the same position again. If the back wheel is the same diameter as the front then the back wheel valve will also map out one revolution and return to a position pointing downwards. It will be exactly in step with the front wheel. As an electrical engineering concept the back wheel valve is exactly in phase with the front wheel valve. i.e the phase difference is zero.

Supposing the rear wheel is slightly larger than the front wheel. As the front wheel maps out one complete revolution the rear wheel valve falls short of pointing straight downwards. It could be measured as say 10° short. In other words the position mapped out by the rear wheel valve is now 10° out of phase with the front wheel valve. As the bike is wheeled along so the error in back wheel valve position is cumulative. On the second revolution of the front wheel, the rear wheel is now 20° in error and so on. After 18 revolutions of the front wheel the rear wheel would be 180° out of phase $(18 \times 10)^{\circ}$ and whilst the front valve pointed straight down the rear valve would point straight up. They would be exactly out of phase with each other. A further 18 revolutions of the front wheel would bring the rear wheel apparently in phase with the front wheel again. But the rear wheel is actually $(36 \times 10^{\circ}) = 360^{\circ}$ out of phase. Although the valves are both pointing straight down again after 36 revolutions of the front wheel, the back wheel has only done 35 revolutions i.e. it has now lagged behind the front wheel by one complete revolution of 360 degrees.

To an observer who looks at the bicycle at the beginning of this experiment, goes away, and looks at it again after 36 revolutions of the front wheel, both valves appear to be in phase with one another. To the person pushing the bike and observing the position of the valves the rear wheel is 360° out of phase with the front wheel.

The moral of this story with regard to understanding phase and bicycles is:

1) The phase of anything must be quoted as relative to some fixed reference. The phase of the back wheel was examined with reference to the front wheel. The back wheel could have been taken as the reference in which case after 35 revolutions of the back wheel the front wheel would be 360° ahead (or leading) of the back wheel. Hence the terms phase lag or phase lead, designated as - or + phase change respectively.

2) Multiples of 360° (or 2π Radians) appear to give "in phase" conditions. Multiples of 180° (or π Radians) appear to give "out of phase" conditions.

3) To specify completely any phase response we need to know what happens during the history of the observation. The person observing, going away and re-observing has a different concept of what happened during the event to the person observing the event continuously. The observer during the event could have noted the number of revolutions of the front wheel and the transient observer should have asked the question, either "how many wheel revolutions have taken place since I last saw the bike"; or "what speed has the bike travelled over what distance? Answers to these questions would give the phase response of the back wheel reference the front in terms of

(i) phase vs revolutions
(ii) phase vs distance travelled
(iii) phase vs time

Combining (i) and (iii) would give us the phase with respect to number of revolutions in a certain time. This is the concept of phase vs frequency used in loudspeaker measurements.

4) The bearing on the back wheel will need less oiling than the front bearing since it goes round fewer times. Chopper owners take note.

Phase Response in Loudspeakers

The amplitude response plot for a drive unit is well known but just to recap this is what it means. A perfect drive unit should reproduce sound pressure at a constant level independent of the driving voltage frequency but at a level dependent on the on-axis conversion efficiency. A practical drive unit can do this very well through its "pass band" and the designer of the loudspeaker system arranges for the passbands of drive units to add together to cover the whole audio band. Because of the laws of physics the extreme low frequencies are governed by factors such as cabinet volume, lowest frequency reproduced Q factor and acoustic conversion efficiency. Drive units therefore behave as band pass devices operating over a specific portion of the audio band. This means that they will reproduce a specific range of frequencies with constant amplitude and outside this range they will attenuate or offer much reduced output. Badly designed drive units will emphasise certain frequencies outside the pass band sometimes to an intolerable level. The band pass characteristics of the drive units in a typical two way system are shown in figure 1.

In just the same way the phase response can be plotted. This is the phase of the acoustic radiation generated by the drive unit, on axis with respect to the oscillating input voltage. A constant level horizontal line shows that the acoustic radiation is in phase with the driving voltage and all is well. (The back wheel is in step with the front wheel).

Because of the nature of electrical filters, transducers and the associated laws of physics where there are changes in the amplitude response there are corresponding and quite predictable changes in the phase response. A typical phase response for the bass unit in figure 1 is shown in figure 2. Over the operating band the acoustic phase is following the input voltage phase.

The analogy with the bicycle can be extended here. Supposing the rear wheel diameter of the bicycle was variable. On first pushing the bike the rear wheel is very large and the rear valve position lags a long way behind the front valve position. As more distance is covered the rear wheel diameter is gradually reduced until when equal to the front wheel the diameter is held. The valves are now in step (but not necessarily pointing in the same direction) for some distance and then suddenly the rear wheel shrinks in diameter, the rear value goes round quicker with respect to the front valve and therefore starts to increase its lead in position..

The horizontal portion of the phase response will not coincide with 0^o in practice, but has been drawn at 0^o for clarity. Similarly when the rear valve of the bike is in phase with the front valve although they are rotating at the same speed the valves do not necessarily point in the same direction at any instant in each wheel revolution.

There is a mathematical relationship between amplitude response and phase response for electrical networks. This can be used to test the quality of a drive unit since the amplitude response can be simultated by an electrical filter. If the test is successful the drive unit is termed "minimum phase" The test comprises examing the amplitude response for small deviations i.e. maxima, minima and points of inflection (maximum rate of change of amplitude) and comparing these deviations with the phase response. Every amplitude maxima or minima should correspond to a point of inflexion in the phase response and vice versa.

Most well designed drive units exhibit minimum phase characteristics over their respective pass bands. An important property of a minimum phase drive unit is that if an electrical filter is placed in the signal path to equalise the amplitude response (measured anechoically)then corresponding equalisation and improvement to the phase response will ensue. Equalisation of course refers to any electrical network including graphic equalisers and parametric equalisers. Therefore aberrations in amplitude for minimum phase systems can be corrected by graphic equalisers provided the system response is separated from the room response.

So far we have examined drive units individually for phase and found them to be classical if well designed. But to make a loudspeaker system we need to combine the acoustic output from two or more drive units to cover the whole audio band. In addition we must ensure that to an observer or listener the acoustic radiation appears to come from one source without wildly characteristic identifications giving any clues that a multiple source is present.

In the analogy with the bicycle we find that we don't really like riding it too much as the saddle position seems to go up and down a lot - although there is a good bit on the journey where the wheel diameter stays constant. Notice that even in this increasingly tenuous analogy the amplitude of the rear wheel diameter and the phase of the rear tyre valve with respect to the front valve are inextricably linked. Amplitude and phase go together.

In order to combine drive units successfully the exact nature of sound radiation from a drive unit should be examined. For example : Where does the sound appear to be generated? What are the consequences of the speed of sound in air with respect to the phase of the components making up the overall sound image? Why do some two way speakers sound better when lying on the floor?

Acoustic Source of a Drive Unit

A well designed drive unit, operating over its intended band, correctly mounted in the right size of cabinet will radiate spherical sound waves as shown in Fig. 3. This applies to drivers with wide dispersion over their operating band. If we could measure the radius of the sound waves we could easily predict the origin or virtual source. Hence we could specify where the acoustic source was in relation to the hardware of the drive unit. However, life is rarely this simple. Figure 3 shows the concept and Figure 4 shows how we can measure the position of the source.

The Measuring Principle

Unfortunately we can't just put a microphone in front of a loudspeaker and measure the phase response. What we would measure is the phase response of the acoustic radiation present at the microphone after traversing a distance 'D' from the loudspeaker. Because sound travels at a constant speed in air the effect on the phase of the various components is drastic. Consider a low frequency signal traversing the space with a wavelength of say 500 mm. (approx. 700 Hz) compared with say a higher frequency signal with a wavelength of say 125 mm. (2800 Hz.) If the microphone were 1000mm from the drive unit then the phase of the low frequency signal (with respect to the input voltage) would have rotated through $(1000/500) \times 360^{\circ} = 720^{\circ}$. The higher frequency signal would have phase rotated by $(1000/125) \times 360^{\circ} = 2880$. Also by the time the microphone measured the radiation the drive unit, it would be doing something entirely different because of the delay in the radiation traversing the distance to the microphone at the speed of sound (speed of sound = 345 µ sec. therefore the microphone measures the drive unit 2.9 milliseconds after the event). The microphone would measure a phase difference between 700 Hz and 2800 Hz of $(2880 - 720)^{\circ}$ even though the radiation left the cone of the driver perfectly in phase.

In the bicycle analogy this is rather like trying to assess the phase of the valves on a bicycle whose frame length increases with the distance travelled.

So to compensate for this effect we introduce a similar delay of electronic origin into the reference input of the phase meter. The reference delay is adjusted to exactly compensate for the acoustic delay caused by the distance between driver and microphone and therefore cancel the effect out. As far as the phase meter is concerned it now thinks that the microphone is exactly at the acoustic centre of the sound source. By measuring the delay accurately (and knowing the air temperature) the distance from the microphone to the acoustic source can be found and referred to some point on the driver chassis or magnet assembly.)

For the Jupiter J40 bass unit the acoustic source is 62 mm from the front chassis surface which puts it on a plane coincident with the front surface of the magnet ceramic. For the treble unit the source is 20 mm behind the front mounting plate surface. Both sources are on the axis of symmetry of the units.

So by measuring techniques we can determine the acoustic source with a high degree of accuracy. In the bicycle analogy if we could compensate for the varying frame length, knowing the phase relationship between the valves and the phase change at a point in the journey together with the difference in diameter of the wheels we could calculate where the bike started off from.

We don't have to use the diameter of either wheel in the calculation, just the difference in circumference.

Phase and Time Delay

Back to the bicycle analogy. Supposing the bicycle is moving at a constant speed. The back wheel is slightly larger than the front one and we have managed to arrange that the frame and rear wheel do not change in size with distance covered. Two observers have to shout when the valves point vertically downwards, one observer for each wheel. At a time when both wheels appear to be in phase (could be any multiple of 360° phase change for the back wheel remember) both shout together. Then a curious thing happens. With each revolution of the front wheel the shout from the rear wheel observer gets later and later. A time delay appears between shouts of first observer and second observer which gets longer as the bike travels on. Eventually after 36 revolutions of the front wheel they both shout together again. But the rear observer would have shouted one less time than the front observer. If they were to count the revolutions and shout the number, the same numbers would get further and further apart in time. So a phase change is the same as a time delay in particular a constant phase change with frequency is equal to a fixed time delay. Note that the phase must be constantly and lineally increasing with respect to linear frequency.

So:

$$\text{Time} = \frac{\text{Rate of change of phase}}{\text{Rate of change of frequency}} = \frac{d\phi}{dw}$$

The relationship is really very simple. Figure 5 shows a graph of increasing phase change with frequency and the resulting associated time delay values.

The reverse is of course true and a time delay can be compensated for by introducing a network which shifts the phase a constant amount with frequency. If the amplitude response is unaffected then this circuit would be able to compensate for inherent time delay errors in loudspeakers, between drive units.

The Loudspeaker System

Consider the loudspeaker of figure 6 which is a conventional loudspeaker system with two drive units mounted on the front baffle. By measurement the acoustic sources of the HF and LF units are located at (1) and (4) respectively. The LF unit is fed through a low pass filter to control the acoustic response and reject high frequency signals which fall outside the designed pass band of the unit. It so happens that all low pass networks have inherent phase shifts which give rise to an apparent time delay. (high pass network do not have time delays). Indeed this explains why the LF unit source is behind the moving cone - because the LF unit on its own acts like a low pass filter. The lower the cut off point of the filter the greater is the associated time delay. So feeding the LF unit through a low pass electrical filter delays the signal more and the acoustic source is set back from position (4) in figure 6 to position (3). In the case of the Jupiter bass unit this represents a distance of 8 mm. So the point (3) is actually 70mm behind the front chassis surface when a crossover is applied.

The treble unit as discussed earlier has its acoustic centre at (1) which is 16 mm behind the front plate.

In a conventional crossover network the acoustic sources are assumed to be coincident in space so that the electrical signals are fed to each unit in a way which allows the energy to add at the crossover point thus ensuring a smooth amplitude response transition from LF to HF. Where the sources are not coincident measures have to be taken to take account of this. In the case of the dual concentric the problem is fairly easy and low roll off slopes can be used. With discrete systems where acoustic sources can be widely spaced they can be aligned to a plane. Conventional ways of doing this so far are:-

1. Filler Drivers

   A technique based on complex mathematical analysis, sounds terrible and is uneconomical in the use of hardware.

2. Sloping Baffle

   Tricky cabinet build techniques, aesthetically not accepted, problems with continuous off axis listening unless height is carefully set. No real solution.

3. Stepped Baffle

   Shelves in baffle create reflective and dispersive surfaces which interfere with amplitude response. Vertical polar response terrible.

All the above solutions attempt to generate acoustic sources on a plane which is vertical and parallel to a vertical plane through the listeners ears.

This is an acceptable condition since the path length difference from observer or listener to the LF and HF units is small i.e. the angle subtended by the two units at the listener's ear is very small. See figure 7. If the listener is midway between the two units the path length will be equal and all will be well. Apart from the disadvantage noted above all the above solutions are feasible.

Another solution which is incorporated in the Jupiter and Venus systems is to introduce a time delay into the treble unit feed so that in Figure 6 the acoustic source is pushed back a distance dl to a point (2) in the vertical plane containing the LF unit source. The disadvantages of sloping cabinet fronts, auditioning off axis, diffractions and reflections from stepped baffles and the raucous noises of filler drivers are all eliminated at a stroke.

The time delay is generated by a second order all pass network which gives negligible amplitude change but a linearly increasing phase lag with increasing frequency. The circuit has a phase change of 360° over a specified operating band depending on the component values chosen.

In the example quoted the Jupiter has a time delay difference of 70 - 20mm = 50. This represents a time delay of approximately 152 ms which can be achieved by the circuit in Figure 8. Referring to an earlier figure 5 a phase change rate that gives 180° (π) at approximately 2.6 kHz would suffice.

The operation of the circuit can be quite easily understood if one remembers that inductors pass lower frequencies and not higher ones and capacitors pass higher frequencies and not lower ones. Parallel combinations of L & C give open circuit at one particular frequency and series combinations of L & C give a short circuit at one particular frequency. i.e. the L's & C's are opposite in action and the parallel and series combinations are also opposite in action :

At low frequencies L1 and L1' pass the signal which passes from right to left across the output terminals. At middle frequencies L1 C1 and L1' C1' become open circuit and L2 C2 L2' C2' become short circuit, so the signal passes from left to right across the output terminals (180° phase change). At still higher frequencies C1 and C1' pass the signal and the signal direction reverts to right to left. However during the beginning the inductors L1, L1' give a 90° phase shift and at the end of the sequence C1 C1' give another 90° shift. The sum total amounts to 360° at the output terminals with respect to the input terminal voltage.

The amplitude can be seen to remain constant by looking at Figure 9. The bandpass and bandstop effects add up to zero attenuation, provided the Q Values of the parallel and series combinations are carefully controlled.

The Benefits of Source Alignment.

With conventional crossover networks (i.e. those not containing delay elements) the axis on which the energy adds at one crossover point is perpendicular to a line through the acoustic sources. Figure 10 shows this. A certain amount of improvement to this condition can be effected by putting the treble unit out of phase with respect to the mathematical phase or messing about with the crossover points of HF and LF units to "gap" the crossover point. Another solution is to design the crossover with very high cut off rates outside the pass bands. These solutions work tolerably well but are all something of a compromise.

Another problem which occurs with non aligned sources is in the off axis response. Consider figure 11 which is a top or plan view of the speaker in figure 10. Even assuming the designer has made a good compromise and the''on axis' response is good (Axis 'A' in fig 11) when the off axis positions are inspected then a varying path difference occurs between LF and HF source. Clearly the response in the crossover region is not going to be the same at 'B' or 'C' as it is at 'A'. This means that the total energy response of the loudspeaker in a live room will show problems at all angles except the on axis or zero angle. Whilst such problems exist in the amplitude response of the loudspeaker and we cope with them in a normal listening environment the stereo information and time delay effects confuse the ears and do not allow a precise stereo image to be constructed.

Figures 12 and 13 show similar diagrams for a speaker with sources aligned to a plane. The axis of best performance is now horizontal in fig 12 and even off axis in the horizontal plane the energy at the crossover point is not subject to path differences. The total energy into the room is therefore more complete in terms of stereo information.

The points made above answer the questions posed earlier. Some speakers sound better when lying on the floor because the best additional axis is pointing downwards.

The time delay solution overcomes most of the problems above although obviously the vertical polar response will not be as good as the horizontal. The solution to this problem is to use a dual concentric. However at reasonable listening distances these effects are minimised and are no greater than other two or three way systems.

Phase, Time Delay and Aural Perception

Olson, a very eminent acoustician states in various parts of his books that :

"The phase of a harmonic affects the threshhold of perceptible distortion as well as the complex sound. This statement contradicts the so called Ohm's Auditory Law that the ear tends to analyse the compounds of a complex sound regardless of the phase relations."

3 references are given of experimental evidence to support this statement. Also on the subject of Auditory Localisation:

"The human hearing mechanism can localise sounds with great accuracy. This property is due to two effects, namely : the difference in intensity and the difference in phase between the two ears. The difference in phase between the sounds at the two ears is due to the difference in time arrival at the two ears. The difference in intensity at the two ears is due to diffraction".

Clearly to reproduce a sound field capable of fooling the ears we must take account of maintaining the time delays between different parts of the complex sound waveforms and also recreate an amplitude field both on and off axis which is equal to that at the original recording microphones.

Acoustic source alignment does just this by introducing a constant phase shift to align sources with the attendant benefits in off axis response, lack of time smear distortion and the creation of very real stereo effects.

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FIG. 1.

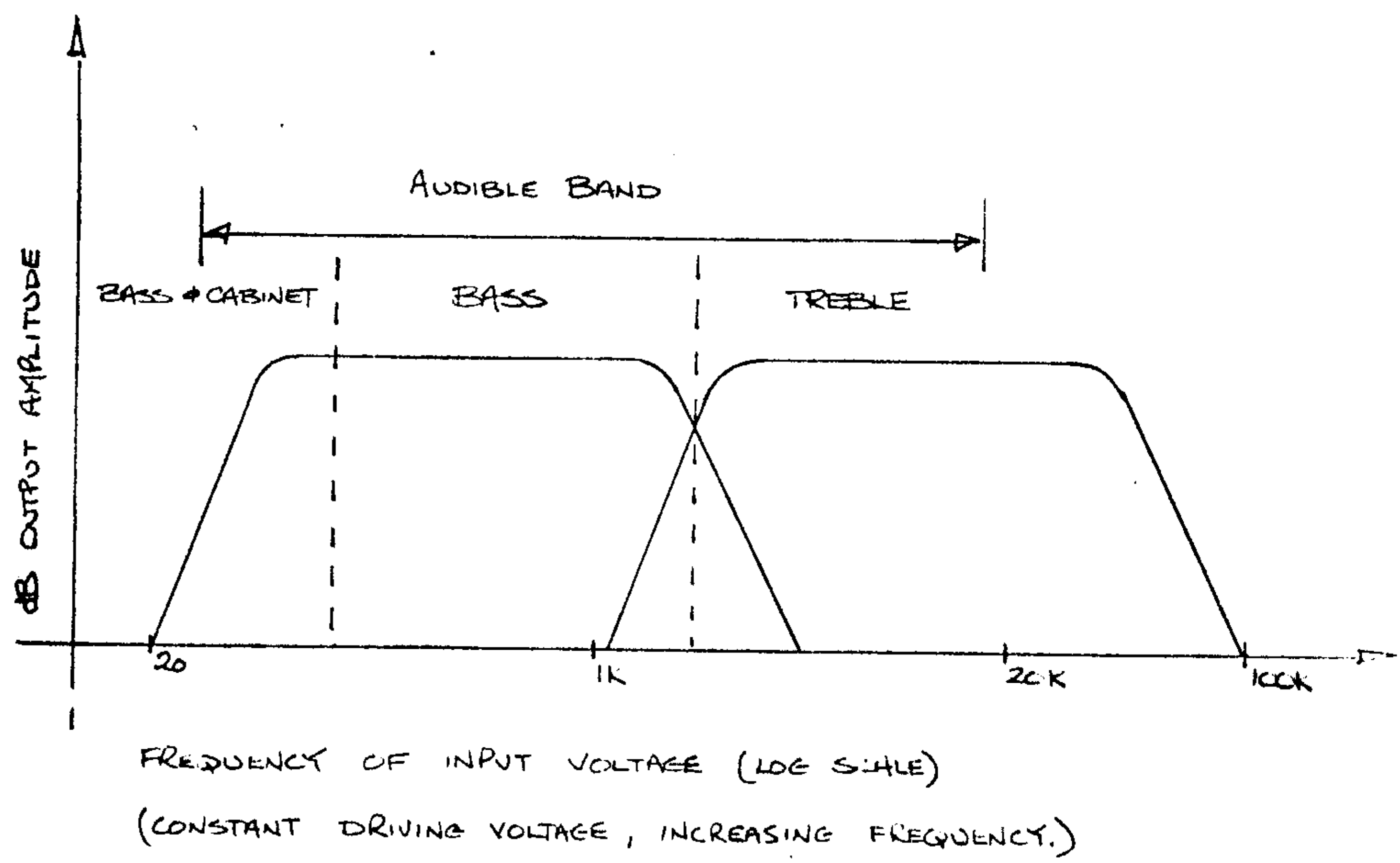

AUDIBLE BAND
BASS + CABINET
BASS
TREBLE
dB OUTPUT AMPLITUDE
20
1K
20K
100K
FREQUENCY OF INPUT VOLTAGE (LOG SCALE)
(CONSTANT DRIVING VOLTAGE, INCREASING FREQUENCY.)

FIG 2.

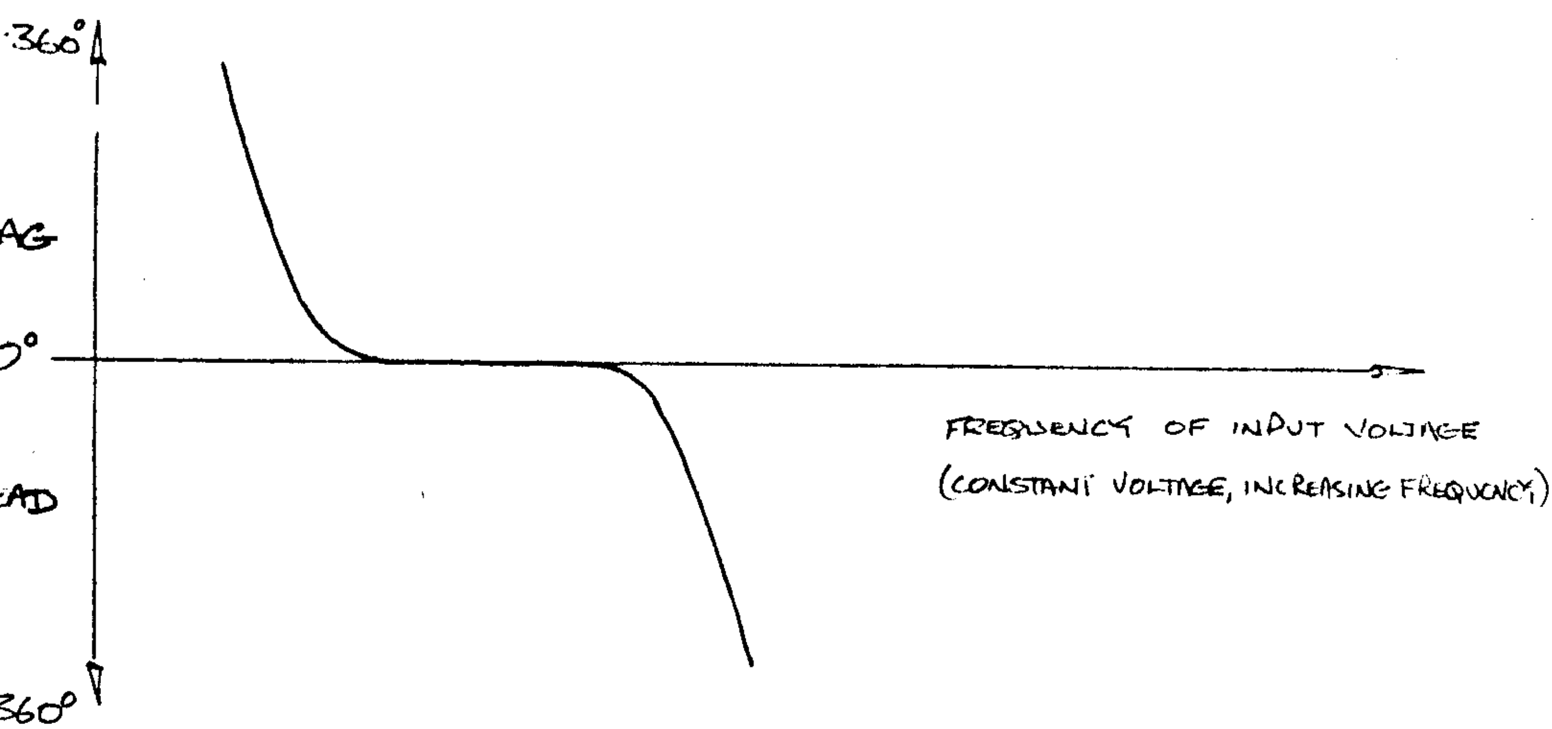

360°
AG
0°
AD
360°
FREQUENCY OF INPUT VOLTAGE
(CONSTANT VOLTAGE, INCREASING FREQUENCY)

FIG. 3

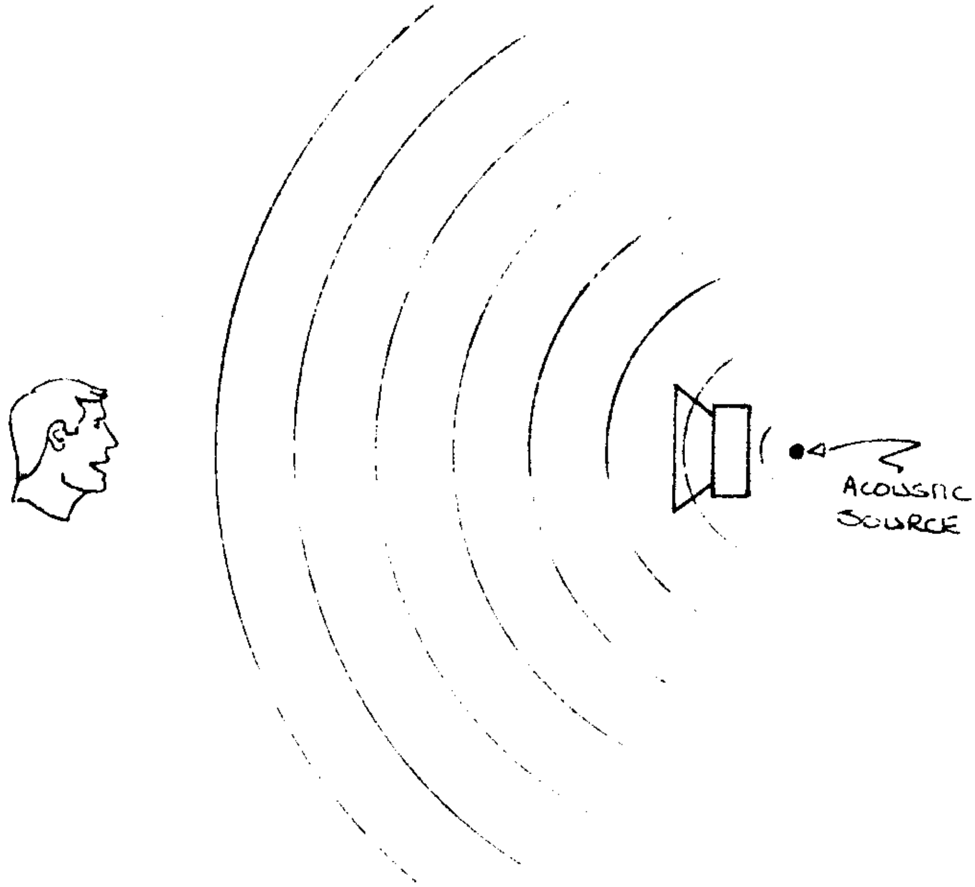
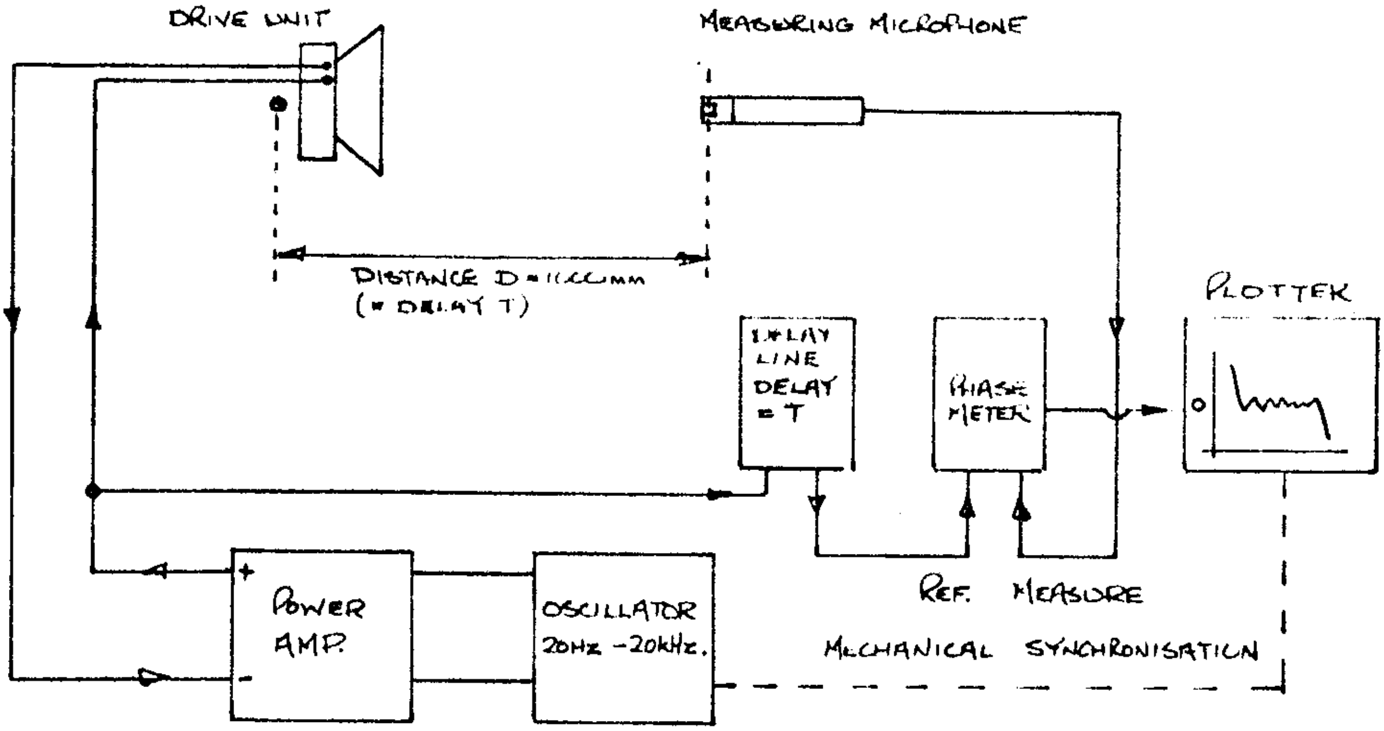
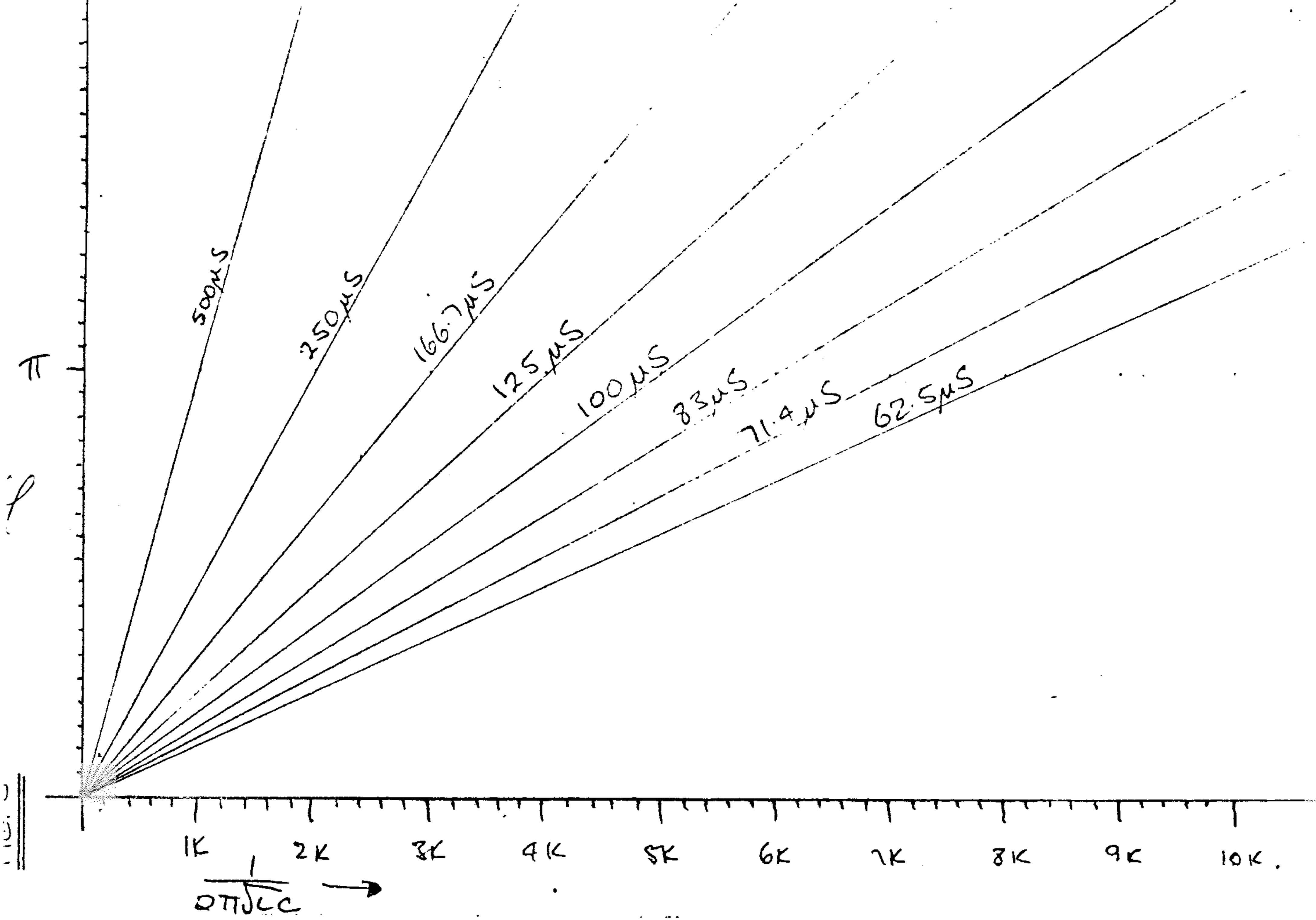


FIG. 4



NOTE: PHASE METER HAS TWO INPUTS - ONE FOR THE REFERENCE VOLTAGE.



π
500µS
250µS
166.7µS
125µS
100µS
83µS
71.4µS
62.5µS
1K
2K
3K
4K
5K
6K
7K
8K
9K
10K
$\frac{1}{2\pi\sqrt{LC}}$

FIG 6

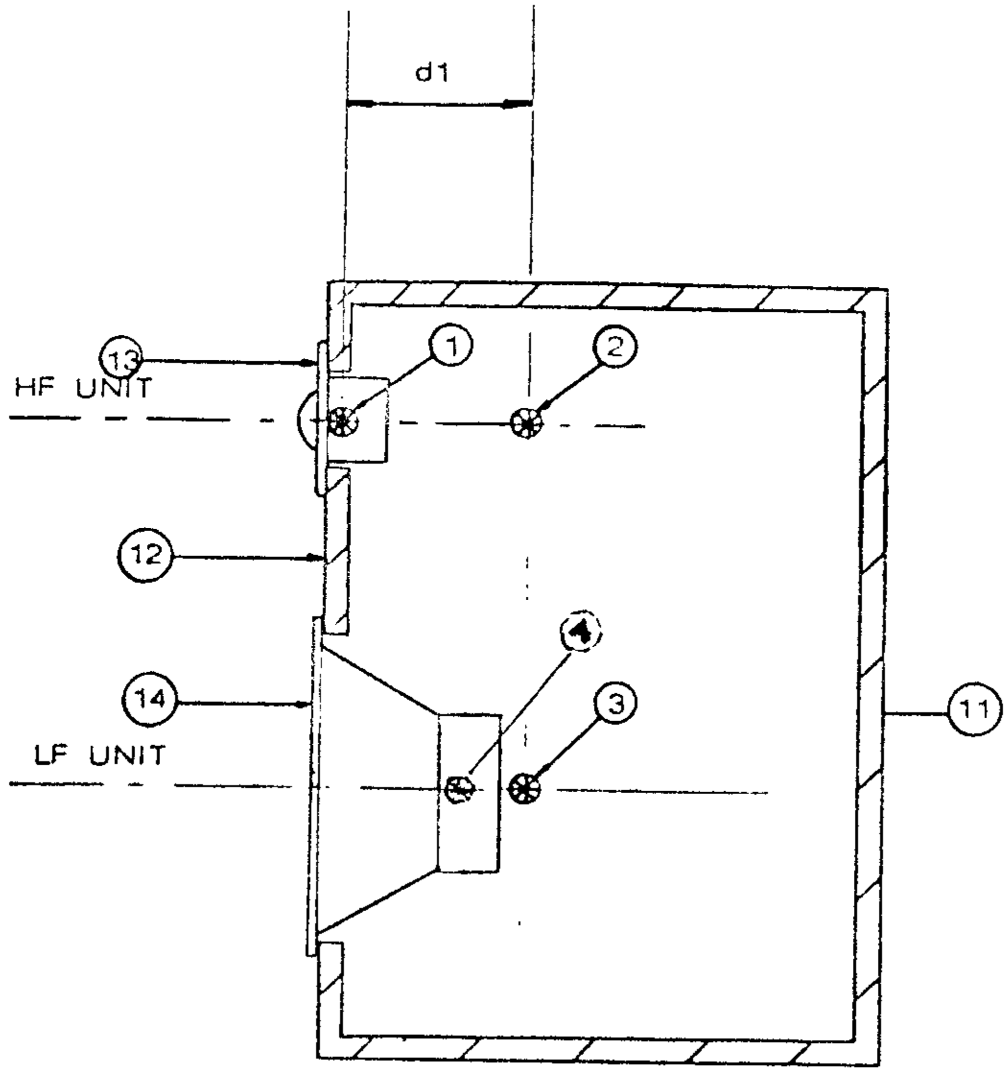

d1
13
1
2
HF UNIT
12
4
14
3
11
LF UNIT

FIG 7

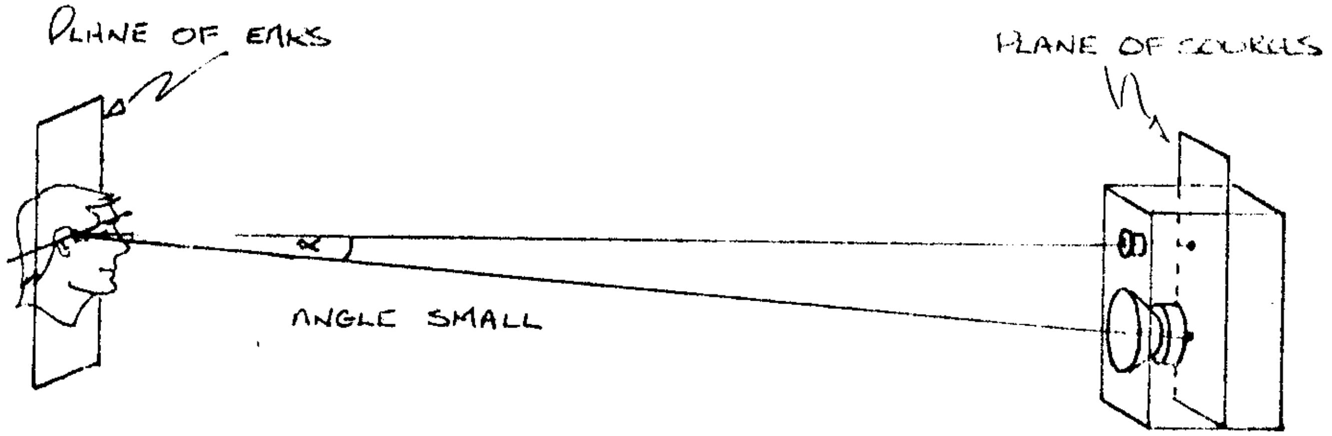
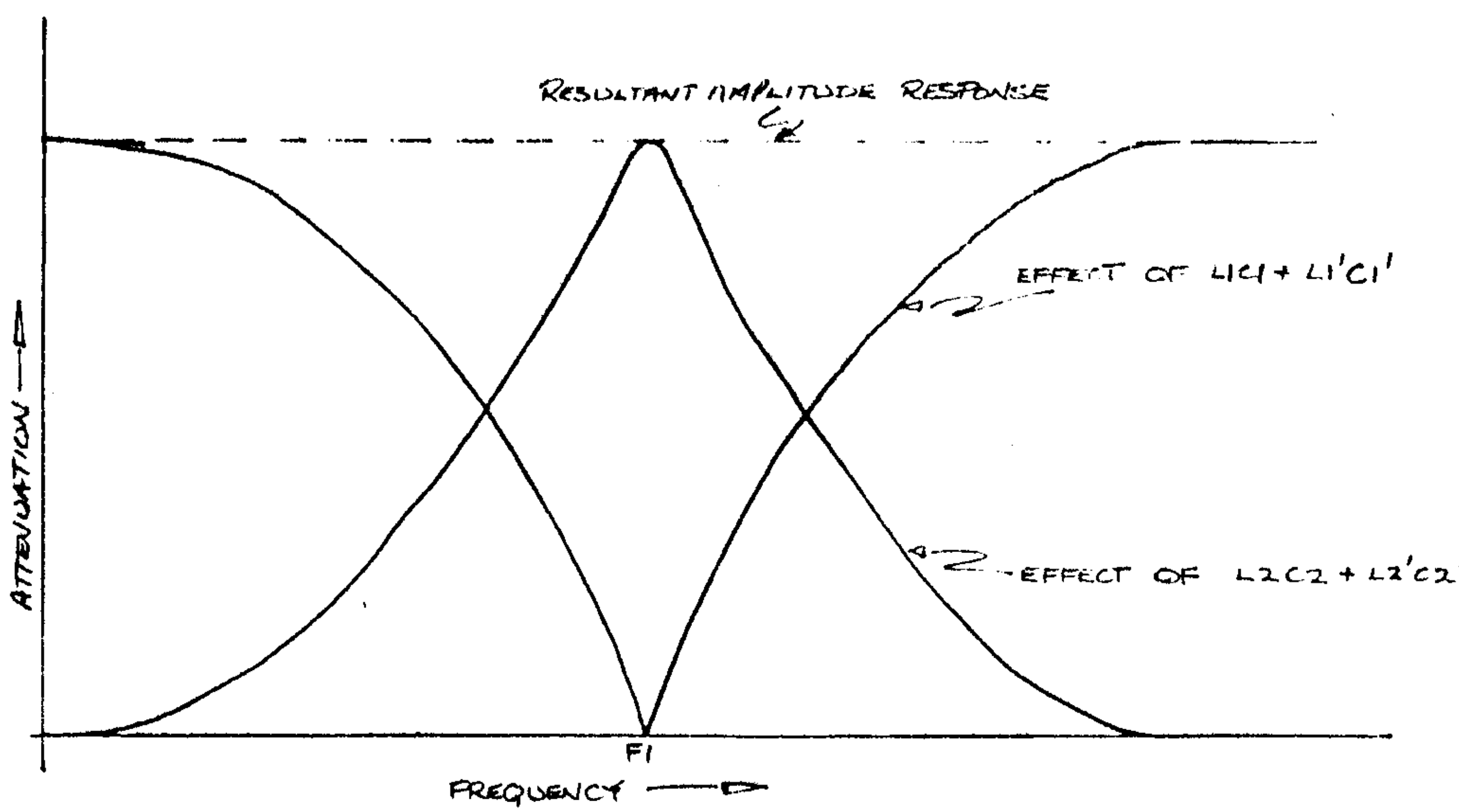

PLANE OF EARS
PLANE OF SOURCES
α
ANGLE SMALL

FIG. 9



$$F1 = \frac{1}{2\pi\sqrt{LC}}$$

FIG. 10

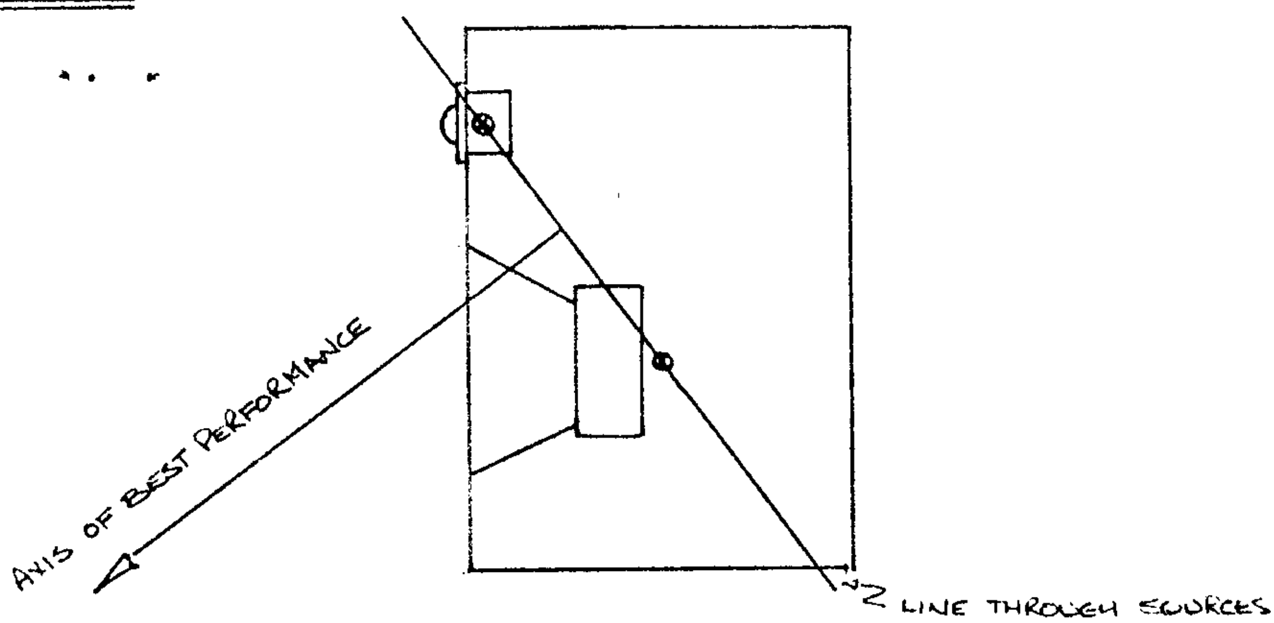


FIG. 11

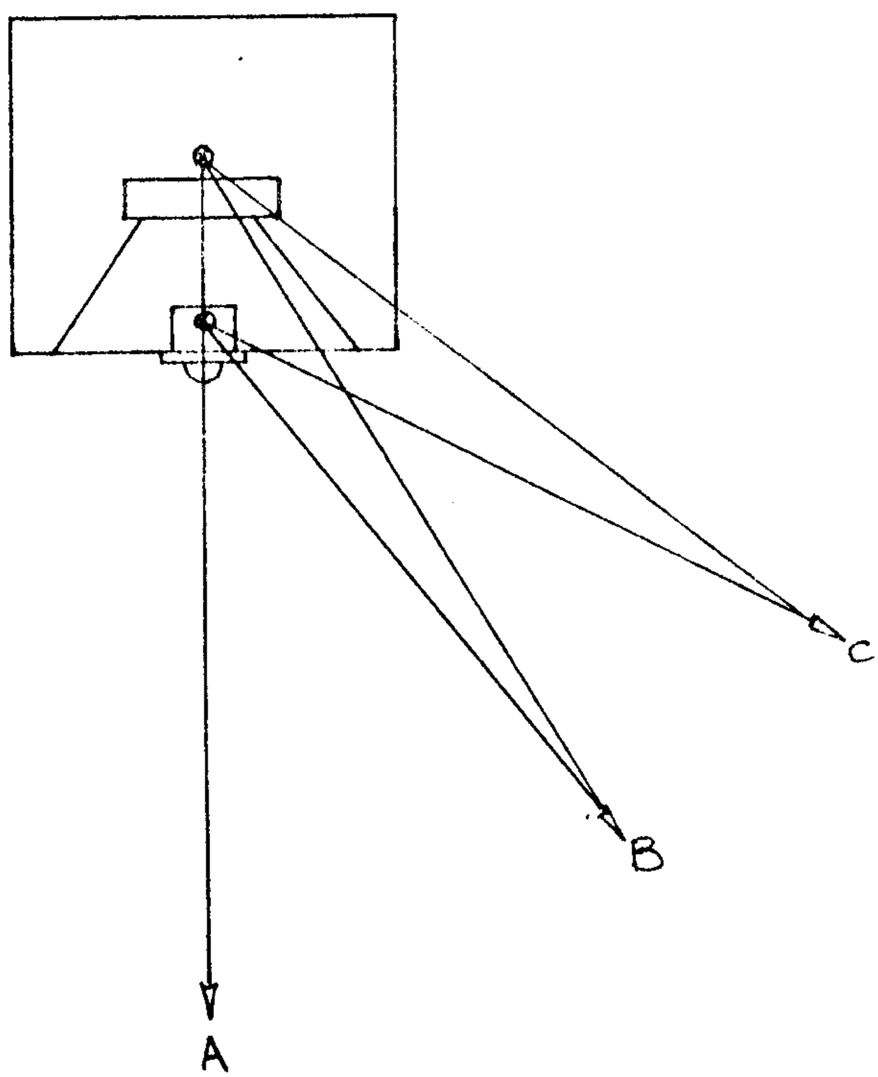


FIG. 12

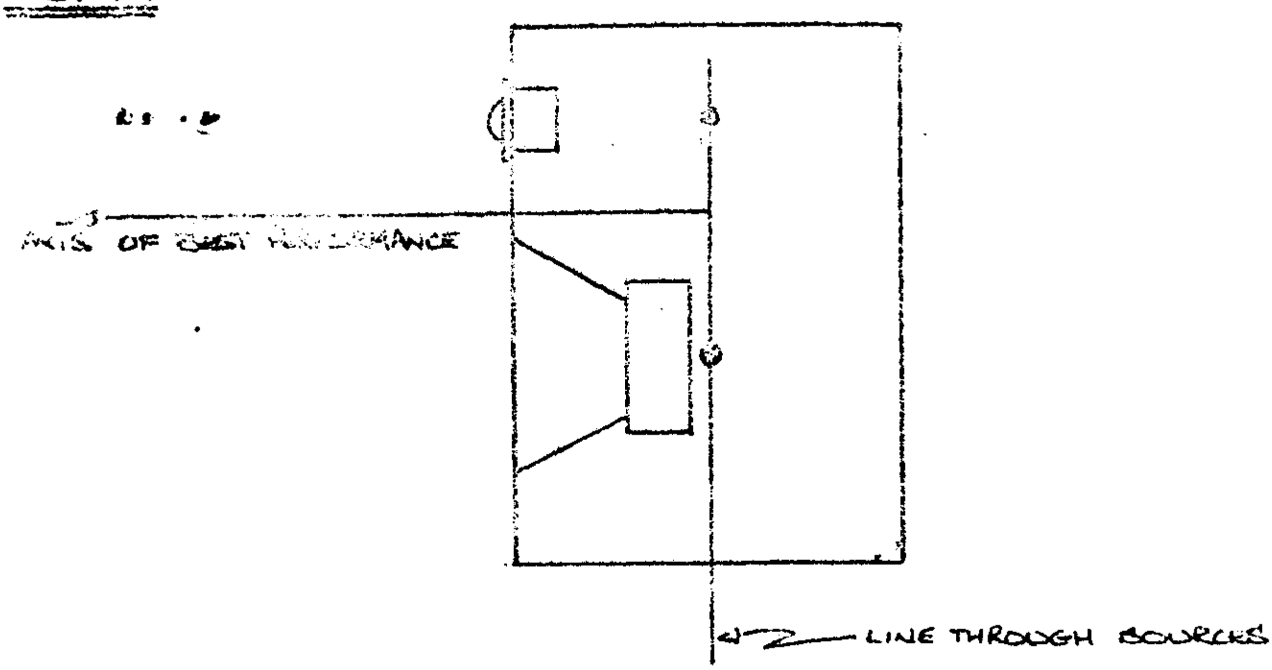


FIG. 13