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EXPERIMENTAL VERIFICATION OF A LOW FREQUENCY LOUDSPEAKER MODEL FOR AN ENGINEERING APPLICATION

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INTRODUCTION

In the control of low frequency noise active attenuators are set to play an increasingly important rôle. With very few exceptions the transduction of the electrical to the cancelling acoustic signal is accomplished by means of a loudspeaker. The requirement for highly accurate control of signal amplitude and phase places stringent demands on the elements of an active noise control system and whereas this is comparatively easy to achieve with most of the system elements, commercially available loudspeakers fall short of the performance needed unless modifications are made. In order to know what modifications must be carried out it is necessary to understand how the parameters governing the loudspeaker's response are connected and in particular their criticallity at different parts of the spectrum. Although equivalent circuits for loudspeakers are described in the literature, the model presented in this paper has been developed to express, in an analytical manner, the acoustic pressure developed as a function of the applied input voltage. The transfer function derived can be directly applied so enabling particular frequency characteristics to be engineered. The accuracy of the model has been checked by measurement and the experimental determination of the various loudspeaker parameters will be discussed.

MODEL DERIVATION

The model is based on the application of fundamental physical rules to the motion of the component parts of the loudspeaker (Figure 1) and an analogous electrical circuit of this (Figure 2) is well known from the literature [1,2]. Taking the elements from the block diagram of Figure 1, the equation of motion for the loudspeaker cone can be derived yielding:

$$\overline{F}_d = m \frac{\partial^2 \overline{\zeta}}{\partial t^2} + K\overline{\zeta} + \lambda \frac{\partial \overline{\zeta}}{\partial t} + \overline{Z}_{rad} \, \Delta S \frac{\partial \overline{\zeta}}{\partial t} \tag{1}$$

where $\overline{F}_d$ is the driving force generated by the electromagnetic induction in the voice coil. From this follows the relation between the acoustic pressure produced and the applied input voltage which for periodic signals gives the transfer function:

$$H(s) = \frac{s\tau_1}{s^3\tau_2^{\,3} + s^2\tau_3^{\,2} + s\tau_4 + 1} \tag{2}$$

with $\tau_1 = \frac{Z_{rad}\ Bl}{KR}$, $\tau_2^{\,3} = \frac{L}{R}\frac{m}{K}$

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$$\tau_3^2 = \frac{L}{R}\frac{\lambda}{K} + \frac{m}{K} + \frac{L}{R}\frac{Z_{rad}\,\Delta S}{K}$$

$$\tau_4 = \frac{L}{R} + \frac{\lambda}{K} + \frac{Z_{rad}\,\Delta S}{K} + \frac{B^2 l^2}{KR}$$

Equation (2) is the general form of the transfer function. However the radiation impedance, Z_{rad}, depends upon the mounting of the loudspeaker and thus the function must be adapted for a specific configuration. This has been done for the case of the loudspeaker mounted in an infinite baffle. A general expression for the radiation impedance of a baffled piston in three-dimensions is unnecessarily complicated for the purpose of the model and consequently simplification to one-dimension (i.e. axial) has been made. The acoustic impedance at an axial distance r from a baffled piston of radius a is given by [3]:

$$\overline{Z}(r,o,t) = \rho_o\, c \left[e^{-jkr} - e^{-jk\sqrt{r^2+a^2}}\right] \quad (3)$$

As $r \to 0$, $\overline{Z}(r,o,t) \to \overline{Z}_{rad}$

giving $\overline{Z}_{rad} = \rho_o\, c \left[2 \sin^2 \tfrac{1}{2}ka + j \sin ka\right]$ (4)

For a typical low frequency drive unit ka<<1 at low frequencies. Thus equation (4) simplifies to:

$$\overline{Z}_{rad} = \rho_o\, c \left[\frac{a^2}{2c^2} s^2 - \frac{a}{c} s\right] \quad (5)$$

substituting this into equation (2) gives:

$$H(s) = \frac{A\, s^2\,(a/2c\, s - 1)}{s^4\tau_5^4 - s^3\tau_6^3 - s^2\tau_7^2 - s\tau_8 - 1} \quad (6)$$

with

$$A = \frac{Bl}{KR}\,\rho_o\, c\, \frac{a}{c}$$

$$\tau_5^4 = \frac{L}{R}\frac{\Delta S}{K}\,\rho_o\, c\, \frac{a^2}{2c^2}$$

$$\tau_6^3 = \frac{L}{R}\left[\frac{m}{K} + \frac{\Delta S}{K}\,\rho_o\, c\, \frac{a}{c}\right] - \frac{\Delta S}{K}\,\rho_o\, c\, \frac{a^2}{2c^2}$$

$$\tau_7^2 = \frac{m}{K} + \frac{L}{R}\frac{\lambda}{K} + \frac{\Delta S}{K}\,\rho_o\, c\, \frac{a}{c}$$

$$\tau_8 = \frac{L}{R} + \frac{\lambda}{K} + \frac{B^2 l^2}{KR}$$

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Three of the parameters contained in the time constants τ_5 to τ_8 cannot be simply measured since they are compound variables from various elements of the loudspeaker. These are the mass m, the spring constant K and the damping λ.

The mass m is an effective mass:

$$m_{eff} = m_{cone} + m_{\text{air at back of piston}} + m_{\text{air at front of piston}} \tag{7}$$

With the loudspeaker mounted in an enclosure, Beranek [1] gives an empirical expression for the air loading on the piston which substituting into equation (7) gives:

$$m_{eff} = m_{cone} + (\Delta S)^2 \frac{B'\rho_o}{\pi a} + (\Delta S)^2 \frac{0.23}{a} \tag{8}$$

B' is a constant dependant upon the dimensions of the back enclosure.

The effective spring constant K is composed of the mechanical suspension coupled with the air compliance and is related to the effective mass by:

$$K_{eff} = \omega_o^2\, m_{eff} \tag{9}$$

where ω_o is the resonant frequency for the loudspeaker mounted in the enclosure.

The damping λ is an effective damping term:

$$\lambda_{eff} = \lambda_{suspension} + \lambda_{\text{air in back enclosure}} + \lambda_{\text{air in front of piston}} \tag{10}$$

Substituting the non-suspension terms shown by Beranek [1] gives:

$$\lambda_{eff} = \lambda_{suspension} + \Delta S^2 R_{\substack{\text{acoustic of}\\ \text{back enclosure}}} + \Delta S^2 R_{\substack{\text{radiation of}\\ \text{front of piston}}} \tag{11}$$

Equation (6) is the analytical function relating acoustic pressure at the loudspeaker piston surface in an infinite baffle to the input signal voltage. This can now be extended to give an expression for the pressure developed at a distance, r, on axis from the piston surface:

$$|\bar{p}\ (r,o,t)| = |\bar{H}(s)|.|\bar{V}_{in}|.\ \frac{\sin \frac{1}{2}kr\ (\sqrt{1+(a/r)^2}-1)}{\sin \frac{1}{2}ka} \tag{12}$$

This expression represents the model of the loudspeaker and enables the effect of changes in particular parameters to be assessed.

EXPERIMENTAL VERIFICATION

Two loudspeakers have been used for testing the model: a KEF B139 drive unit and a SEAS 33F KWA unit.

For both these units the required model parameters have been measured and the resultant pressure predicted by the model has been compared with the on-axis sound pressure amplitude response measured under anechoic conditions.

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The model has been derived for a loudspeaker mounted in a baffle of infinite extent thus predicting radiation into a half-space. For practical reasons the units have been built into enclosures with the consequence that the radiation is no longer restrained to the half space. Therefore for sensible comparison, the unit's directivity must be taken into account and appropriate correction applied. To calculate this correction the on-axis pressure response has been measured both in front and behind the loudspeaker. This results in an expression for the sound pressure level as if radiation were into a half-space:

$$L_{\text{into half-space}} = L_{\text{frontal}} + 20\log_{10}\left[\frac{L_{\text{frontal}} + L_{\text{back}}}{L_{\text{frontal}}}\right] \qquad (13)$$

The values obtained from this expression represent the corrected sound pressure from the loudspeaker and therefore can be directly compared with the levels predicted by the model.

As far as possible the parameter values for the model have been experimentally determined. Where this has not proved feasible, values have been taken from the manufacturer's data.

- The radiating surface area was measured as the piston surface area in combination with an equivalent radiating area corresponding to the outer piston suspension.
- The Bl-factor was statically determined by applying a known force to the piston and measuring the DC current required to return the piston to its equilibrium position.
- The electrical resistance was measured using a standard bridge, but direct measurement of the inductance was not possible due to the emf generated by the motion of the voice coil in the magnetic field thus resulting in an erroneous value for L. For this reason the inductance was assumed from the manufacturer's data.
- Since a direct measurement of the loudspeaker's moving mass was only possible by dis-assembly of the drive unit, the manufacturer's data was again used to provide a value for m_{cone}. With this, the effective mass, m_{eff}, was calculated from equation (8).
- The spring constant K was then determined from equation (9) using the calculated value for m_{eff} and the measured resonant frequency of the drive unit mounted in the enclosure.
- As the damping governs the broadness ($\Delta\omega$) of the resonance peak, its value can be determined from the electrical impedance curve of the loudspeaker in its enclosure by:

$$\lambda_{eff} = \frac{\omega_o}{Q} m_{eff} \qquad (14)$$

where $Q = \omega_o/\Delta\omega$.

The value for the damping parameter in the model is λ_{eff} less the last term of equation (11). However for the cases of the loudspeakers presented in this paper this term is negligible.

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By substituting the values of these parameters into the expressions for A, and τ_5 through τ_8 the relative significance of the terms in the transfer function (equation (6)) can be assessed for different frequency ranges. Ignoring the insignificant terms then gives:

$$\text{for } f < 500 \text{ Hz}: H(s) \approx \frac{\alpha s^2 (1 - \beta s)}{s^3\tau_6^{\ 3} + s^2\tau_7^{\ 2} + s\tau_8 + 1} \quad (15)$$

$$\text{for } f < 100 \text{ Hz}: H(s) \approx \frac{\alpha s^2}{s^2\tau_7^{\ 2} + s\tau_8 + 1} \quad (16)$$

with $\alpha = \rho_0 c \frac{a}{c} \frac{Bl}{RK_{eff}}$ and $\beta = \frac{a}{2c}$

The loudspeaker amplitude response derived from equation (15) was then compared with measured results for both the KEF and SEAS units.

COMPARISONS AND CONCLUSIONS

The measured free field on-axis response for the KEF B139 unit is shown in Figure 4 together with its directionality corrected curve and the model response. Up to approximately 350 Hz the agreement is good with a maximum deviation of 2 dB. (With the limitations imposed by the anechoic chamber, the accuracy of the free field response measurements cannot be guaranteed below 30 Hz). Above 350 Hz the measured response is seen to deviate significantly from the model and an explanation for this could be that the loudspeaker cone is no longer behaving as a rigid piston. In order to further check the validity of the model a second loudspeaker (SEAS 33F KWA) with substantially different parameter values was measured. The results are given in Figure 5 and again good agreement was obtained. To illustrate the value of the model, Figure 6 shows the effect of changing the cone mass and effective spring constant of a loudspeaker on its predicted amplitude response.

The model discussed in this paper has been found in general to be a good analogue of the low-frequency behaviour of a loudspeaker. The facility for being able to judge the effects of altering loudspeaker parameters constitutes a powerful tool in tailoring the acoustic response for particular applications.

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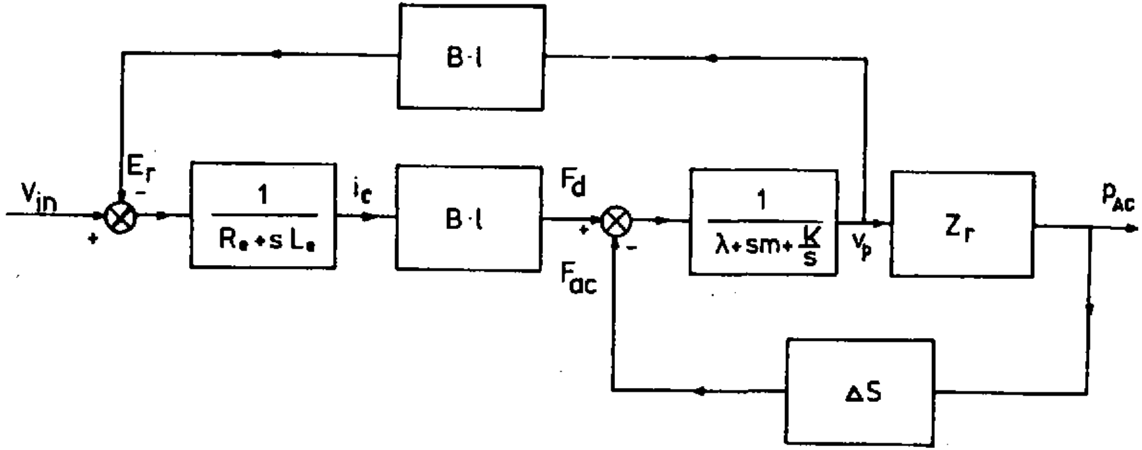
[3] Kinsler and Frey, 'Fundamentals of acoustics', J. Wiley and Sons, 1982



V_{in} = applied input voltage

E_r = EMF

R_e = electrical resistance of the voice coil

L_e = inductance of the voice coil

B = magnetic flux density

l = length of the voice coil wire

λ = mechanical damping

m = total moving mass

K = spring constant

v_p = piston velocity

Z_r = specific radiation impedance

ΔS = radiating surface of the piston

Figure 1 : Block diagram of a loudspeaker.

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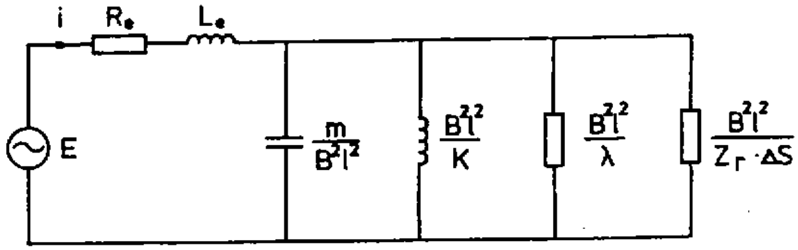

Figure 2 : Equivalent electrical circuit of a loudspeaker.

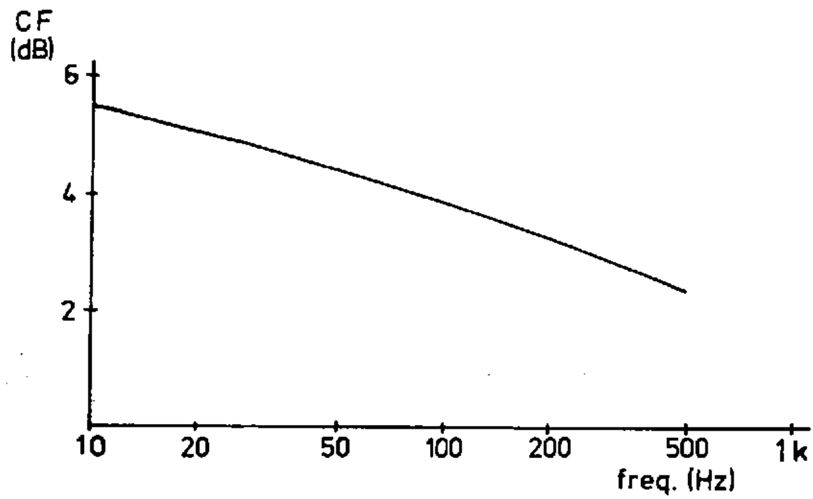

Figure 3 : Loudspeaker directionality correction factor (CF).

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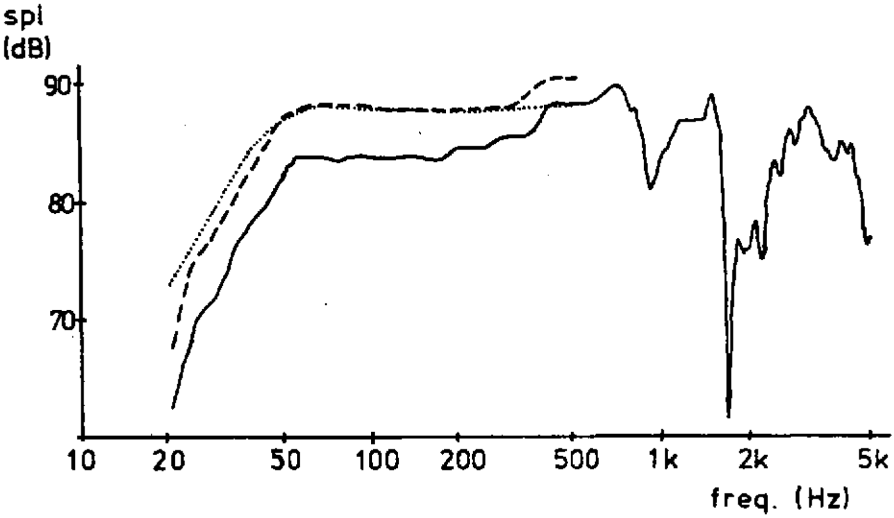

Figure 4 : Response of the KEF B139 (——); corrected for directionality (---) and response predicted from the model (·····).

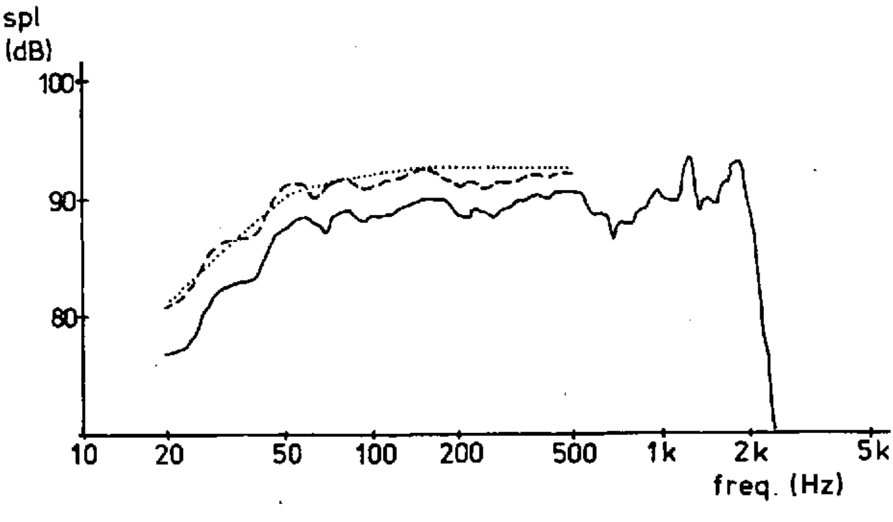

Figure 5 : Response of the SEAS 33F KWA (——); corrected for directionality (---) and response predicted from the model (·····).

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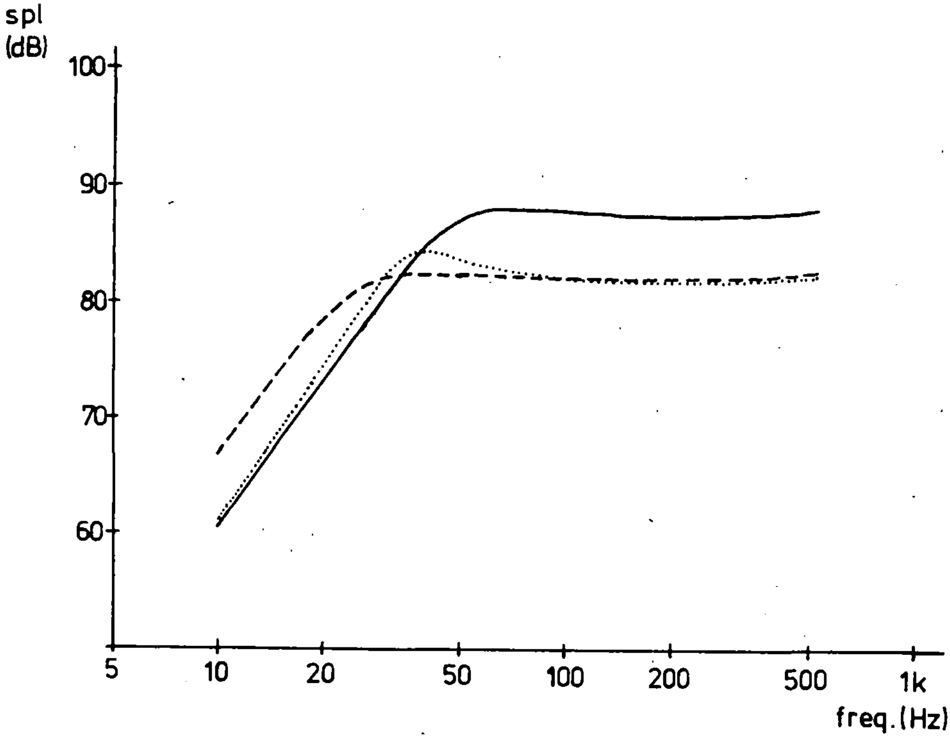

Figure 6 : Predicted response of KEF B139: unmodified (——), doubled cone mass (·····), doubled cone mass and halved effective spring constant (---).

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SETTING UP HIGH POWER SPEAKER ARRAYS FOR ROCK FESTIVALS

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OVERVIEW

In the context of home hi-fi, or even public address, "high power" can mean a few hundred watts. This paper discusses speaker systems of up to 50,000 watts nominal power. In this context, the power is the electrical V A fed out of the amplifier to the speakers and not the actual acoustic output. The usual use for such high power arrays is for open air rock concerts but the same concepts are also used for certain military tasks or where long distance sound transmission is required for simulation or modelling of proposed large industrial installations.

BACKGROUND

When electronics started to come into music the first introduction was in the dance bands of the late '40s with a 25 watt amplifier for the "electric guitar". At about the same time the very crude public address systems were being improved using line column speakers instead of horns to get some semblance of flatness. The next stage on the road was the 3-guitar groups of which the Beatles were the best known. The 3 guitar amplifiers were of the order of 50/100 watts each and contemporary measurements suggest levels of 95dBA were typical.

By the late '60s/early '70s "walls" of speakers had started to be built in the attempt by each "heavy" band to outgun each other. Now levels of 110dB were not uncommon. Indeed, in 1972 an Australian band was measured by one of the authors, playing at a fairly constant level of 122dBA. At the time a large disco company was seriously thinking of putting this band into a noise sensitive club. They tried it for two nights. The predicted disaster resulted and as a direct result the club was eventually sold.

Rental companies have always played a big part in the entertainment world, but it was around this time that they came into pre-eminent positions. The cost of the big systems was becoming too expensive to buy for any except the richest groups. Thus, rental became the norm for one day events and also for whole tours. Rental could be just the equipment, but more usually, for events of the magnitude we are discussing it included the crew to run the system.

From the early days rental companies faced a diacotomy of interests. On the one hand renting out inefficient walls was superficially rewarding in terms of immediate profit, but on the other hand a reputation for clean sound produced by a small efficient system would probably be rewarding long term, even though in the short term not so attractive.

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While each company took it's own commerical decisions, an external factor crept in around 1972. This was, of course, noise pollution. By 1973 this had built up to such an extent that citizens action groups were actively working against the major open air rock festivals. Indeed, they had good reason for this, as no real attention was paid to environmental problems by many event organisers. All efforts were directed towards the band's requirements. The argument was, and to some extent still is, that they are the stars and everything must be done to keep them happy. This is no different to the problem Hollywood has faced for half a century.

In 1975 Entec and Cirrus Research started a project to evolve a new method of setting up the speaker stacks with three objects in view. The work was undertaken for the National Jazz Federation who are the organisers at Reading Rock. The N.J.F. have always had an enlightened attitude to technical problems and for them, noise control was a technical problem which required both engineering and socio-political investigations. Thus three problems were defined on an engineering front, with the non-engineering problems left open ended.

Firstly, the sound quality had to be improved. This was defined for these purposes as improving the system transparency. A totally transparent system, i.e. one which simply makes the level higher, is not what rock groups want. However it was a necessary first task on the way to meeting their exact needs.

Secondly, the external sound field had to be reduced as much as possible. Ideally down to the levels of BS 4142 - a maximum of say 6dB above the background levels. At the time this seemed an impossible task, as levels 30dBA above the background were common and there really was no concensus that thought should be given to the exposed population. While the project was funded by the N.J.F. the pressure was on them from the local council, who were not only the statutory authority, but also the landlord of the festival site.

Finally, the system should be more controllable with more predictable performance. While this would seem a statement of the obvious, in truth it is not. There was a large body of opinion that wanted the mystique of a system that needed 'expert' handling. A controllable system would also be easier to use if the environmental lobby became even more powerful. At the very least we would have the knowledge of what was possible and be able to make rational judgements on any new proposed site. In 1985 this seems obvious, but in 1973 no-one had any real idea what could be achieved, least of all the project team. It was here that the social surveys and non engineering matters would come in.

At the outset it was clear that sudden revolutionary changes were not the way to proceed. Certainly the bands would not like to face a new system when they arrive to perform and indeed it was not reasonable to ask them to do so at that time. They have for sale an image and a sound. Should the sound system not be what they expect they naturally feel annoyed. Thus, the balance between good clean sound with good attenuation was and is very delicate. One must add to this the natural conservation of the group management and sound crew. This conservatism is natural when the ephemeral nature of the commodity is considered. Indeed, from the point of view of bands, it is the only possible response.

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PROGRESS

Between 1974 and 1977 the technique was to raise the height of the speaker systems with heavy angling. This was to essentially make a curved front system rather after the manner of a fresnel lens. There was much argument that this gave different results to 'normal', but objective tests showed that the sound was cleaner with much less interference between individual units and therefore less random sound lobes. Of the authors one has a classical musical bent, while the other is trained in mixing for rock groups. This liaison did give certain advantages in setting up systems and in resolving arguments when a bands manager claimed "The sound is wrong". It very rarely was. However we cannot pretend that the early days were easy. It took a long time for the two authors to be able to communicate at any sensible level, they had different perceptions of reality and the priorities involved and until these were merged, or at least totally understood, progress was slow.

SUDDEN CHANGE

In 1978, for reasons unconnected with this paper, the system was taken back to the 'wall'. Naturally the environmental gains of the previous two years were wiped out at a stroke. The N.J.F.and the council together decided that a totally radical system should be implemented forthwith and that in future no system could be used without the prior approval of the noise control team. This by now included the two authors and the Environmental Health Officer responsible for the festival. This joint programme produced rapid results.

CURRENT SYSTEM

In 1979 the new method of setting up was accordingly put into place. This system, the result of much effort, was simple in essence but complex in realisation. Each stage source, be it drums, voice, guitar speaker, organ etc., is fed into a separate microphone. Each microphone signal is then passed down on a cable to a mixing desk at the side of the stage and a second mixing desk some 50m out in the audience. Up to 32 channels are routinely employed. Each channel has totally conventional studio control of tone etc., In fact the mixing desks would be perfectly at home in a studio. The separate channels are then sub-grouped; again with tone control; and grouped again down to a single channel in either mono or stereo. This processed signal is then fed to digital delay units and passed to separate speaker stacks in two lines out from the stage.The basic block is shown in Fig.1

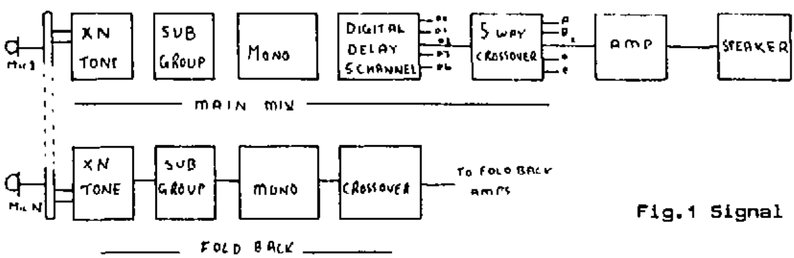

Fig.1 Signal path

SETTING UP HIGH POWER SPEAKER ARRAYS FOR ROCK FESTIVALS

Each speaker stack has a 5 channel crossover in approximately double octave bands. The output of each crossover is fed to high power - 500 watt linear amplifiers which in turn feed speakers.

Thus, the system has a series of individual sources each controllable in time, frequency response and spatial directionality. Each source is at a much lower level than would be the case of a single stack and because of heavy angling, the top of the radiation angle is just below the horizontal.

Each double band on each tower had two amplifiers to give a safety margin, thus the system had 9 stacks with at least 8 amplifiers per stack, making a total of 96 amplifiers (including foldback) each of about 500 watts, all running together. A total of 264 speaker cabinets were involved in the system.

This then is the main PA system which has to be measured, tested and set up as a working whole. Fig.2 shows the delay tower arrangement.

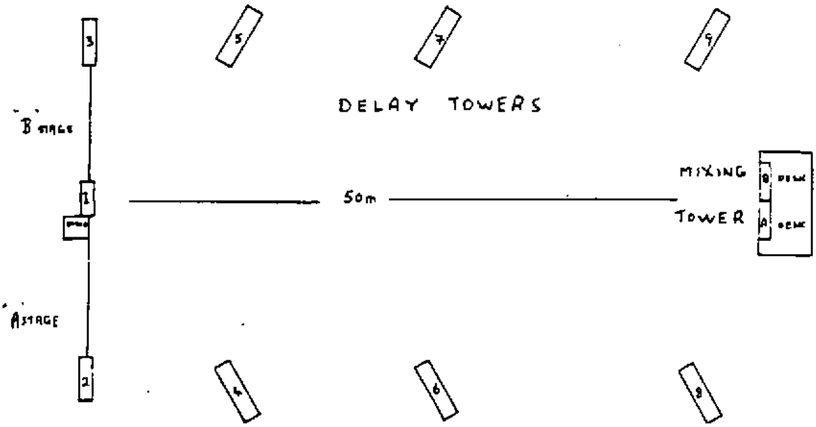

Fig.2 Delay tower arrangement

The second mixer, on stageside, is to give 'Foldback' to the artists on stage. Clearly at the levels in use and the propensity of the artist to move about carrying a microphone, a complex tone control mix is needed to prevent acoustic feedback. This is done with third octave filters in each foldback or monitor speaker and the curve adjusted by trial and error until no 'howl' is perceptable. The stageside speakers are complimented by 'wedges' at the stage front. These are the obvious signs on T.V. of the system. The sidefill speakers originally piled on top of each other in a haphazard manner, are now carefully arranged in line column with appropriate spacing to keep a wide flat beam across the stage with as little spill into the surrounding area as possible. The total power of the stage monitors is of the order of 12% of the whole.
However, in 1980 the stage monitors were identified as the main component in the escaping noise, hence the attention to careful stacking. Reading had an extra problem in that two stages are in use to give continuous music. Thus each

microphone and mixer was duplicated for each stage, finally coming together at the input to the delay units. The foldback system stayed independant on each stage. In use one stage is sound checked while a band is performing on the other. This requires a highly trained crew and really good noise cancelling microphone intercom units to talk to the stage being set up. Remember the normal ambient level is well over 100dBA.

STACK PERFORMANCE

When it was decided to have multiple delay towers as described, it was necessary to carefully design the stack so that the penumbra of the speaker was roughly parallel to the ground. In this way the rollover effect of the ground wave would ensure that the beam in fact never went above the horizontal. To do this meant the actual dispersion pattern of the speaker stacks be known. The various manufacturers of the horns, bins and drivers were consulted, but it soon became clear the data did not exist. Or if it did, no-one would admit to it. Accordingly a simple experiment was set up to measure the actual radiation angle. This is a perfectly simple, totally routine thing to do in the laboratory. However, when half a tonne of speakers with a throat area of 5 square metres is involved, the task is somewhat more difficult. The experiment was finally done using the biggest fork lift truck available to hold and elevate the stack, while the measurements were made in the bucket of a 'cherry picker' of the sort used to change street light bulbs. At times like this, a good case can be made for scale modelling.

Having obtained the data it was a simple matter to design the speaker stacks such that the area around each delay tower has a flat and even field, yet the cut off at the back of the area is very sharp.

SETTING UP THE STACKS

The first part of the setting up procedure is to get the delays correct to a first order. Then each individual stack is equalised. To do this a hand-held real time analyser is used with pink noise played through each section, i.e. each double octave band in turn. Pink noise is not the best for noise control purposes, this ideally needs a noise shaped to the spectrum of typical rock music. However to set up and equalise a system, the sound crew are used to pink noise and thus they continue using it.

When the head sound man is happy with each stack and it is correctly aimed for good attenuation, the delays are set accurately. This is done by pulsing the system with a very short pulse and either looking at the output of the audio channel on an oscilloscope or more usually adjusting until no double pulse can be heard. The trained ears of the crew can get it correct far quicker than an engineer with measuring equipment.

The field is then surveyed using simple sound level meters to ensure that we have a good even field varying by at most 5dB. Again ears are the best judge of a changing mix. That is when the overall level, the vector sum, is correct but each individual double band is not. Much time is taken up by this and the R.T.A. is very useful. In future we would be looking at a computer programme to automatically plot the field using one of the new generation hand held R.T.A.

connected on line to the computer. This has however not yet been tried in a real situation.

ENVIRONMENTAL TESTING

At this stage the team leaders change hats and start the formal testing for propagation in the far field. In this case the far field is not the classic definition, but the field outside shouting range. Here the radio links have transformed the task. What used to take all day, is now the work of an hour or so.

The stacks are individually fed with the simulated pink rock using the curve described by Griffiths or in some locations pink noise. The output is held at the same Leq for a minute or so and the several operators in the field take simultaneous measurements. These are reported back by radio and the results plotted.

If all is as predicted and for the last 4 or 5 years this has been the case, the whole system is then checked, all at once, on full power. This is the acid test. If the total attenuation is as designed, all that remains is to plot each single double band for reference purposes. If not, adjustments have to be made and the tests repeated. Officially at this time, the local environmental health officer is invited to repeat the test. However at Reading and in the G.L.C. area the officers have not only been with us while we carried out our tests, but usually have lent personnel and equipment to speed up the task. Thus they can be reasonably satisfied that all is as it seems.

Occasionally, local politicians have complained that this liaison between the poacher and gamekeeper takes away some right of the council to take action if things go wrong. The supposed argument is that the promoters will claim that the council gave advice and the error is a direct result. In fact nothing could be further from the truth. The sole person responsible should be the consultant, or if one is not employed, the head of the sound crew. No matter what help the council give, it must be the clear duty of the consultant to be responsible and nothing can take away the statutory rights and duties of the council.

RESULTS

The results of all this have indeed been impressive. The sound quality is as good as or better than the best of home hi-fi units. Naturally when delays are used, the only echo free area is in a straight line down the centre. In actual use, the echo does not pose a problem and the predictions of Britsh Rail sound have proved groundless.

On the environmental front, the results have been better than predicted. The levels out in the surrounding area are of a lower level than the unrestrained audience applause. Clearly, that is all that can be sensibly achieved unless steps are taken to silence the audience. As for being predictable, the third target, the noise team can modify the mix on-line during the performance in the certain knowledge that adjustments will give the required results out in the field.