THE SOUND DISTRIBUTION IN THE FAR FIELD OF IRREGULAR RECTANGULAR AND CIRCULAR ARRAYS OF ACOUSTICAL SOURCES WITH DIRECTIONAL PROPERTIES

by
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for
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Introduction

The study of sound fields from multiple acoustical sources dates back at least to work published by Wolff and Malter [1] in 1930 and Nyquist [2] in 1932. It is the acoustical equivalent of the work on optical interference started by Thomas Young in 1801. The analysis of interference phenomena is based on Young's priciple of superposition of displacements or an equivalent parameter.

Recent studies of sound fields from multiple acoustical sources include those of Hartman [3], Sinclair [4], [5], [6], Meyer [7], Russell [8], Ahnert [9] and others. This small sample alone considers linear and rectangular arrays [3], [4], [5], [6], [7], [9] and sources arranged in arcs or cylinders [5], [8]. Most of these studies considered effects in the far field only, while others [3], [7], [9] included near field effects. Some studies considered phase relationships between the individual sources. The effects of the directional properties of the individual sources was included in some of the work [4], [5], [7]. Two of the papers [5], [7] examined the sound field in directions other than those defined by the principal axes of the array, and one [3] considered the field in the Fresnel region or very near field along the principal axis perpendicular to the array. A few of the above papers [3], [4], [5] included theoretical derivations and experimental verification.

The present study provides the means of examining the far fields of regular and irregular arrays, in the horizontal and vertical planes for rectangular arrays and in the plane of the array for circular arc arrays. The effects of the directional properties of the sources are included in the analyses. A few examples are examined.

The Models

The model used here for the rectangular array case is any array of acoustical sources whose positions can be described in cartesian co-ordinates. There are no restrictions on source positions such as regular spacings in rows or columns. The sources need not be in the same plane. The only restrictions are

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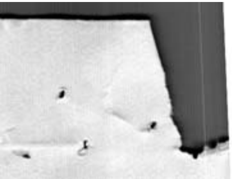


that the sources be identical, with equal acoustical outputs and that their axes be parallel.

The co-ordinates are x positive horizontally to the right looking at the array from the listening area, y positive upward and z positive toward the listening area. The co-ordinate axes are parallel to the source axes. The position of the origin can be selected arbitrarily, typically but not necessarily coinciding with one of the sources.

The model for the circular arc arrays is for sources arranged in a plane at angles to each other such that the longitudinal axes all intersect at a point. This point is used as the origin of a set of polar co-ordinates. The sources need not be equidistant from the origin and the angular separation need not be equal.

For arrays which are convex toward the listening area, the co-ordinates are R positive, with angles positive for rotations of the source longitudinal axis upward or to the right when viewed from the listening area and negative to the left or downward. For arrays which are concave toward the listening area, the signs on the angles are reversed.

For all sources the directional properties are approximated by an empirical formula.

General Theory

Following the method described earlier [4], the origin is taken as the reference point for phase and acoustical path length. For a point at a distance from the array very much larger than the array dimensions, let the sound pressure P be the real part of

$$P = A(\theta) e^{j(\omega t + \phi)} \tag{1}$$

where ω is the angular frequency and $A_i(\theta)$ is a function of the angle between the array longitudinal axis and the direction of the distant listening point. It gives the ratio of the sound pressure due to the source as a ratio of its on-axis value at the same distance. For the i^{th} source shown in figure 1, the pressure contribution is given by

$$P_i = A_i(\theta) e^{j(\omega t - kS_i + \phi_i)} \tag{2}$$

where

$$k = \frac{2\pi}{\lambda} = \frac{2\pi f}{c} \tag{3}$$

λ is the wavelength, f is the frequency and c is the speed of sound. ϕ_i is the phase of the i^{th} source and S_i is the distance by which the path length from the i^{th} source to the distant point exceeds the distance from the origin to that point. In this study, ϕ_i - 0 for all i.

For an array of n sources, the total pressure P is given by

$$P(\theta)=\sum_{i=1}^{n} A_i(\theta)e^{j(\omega t - kS_i)} \tag{4}$$

$$= e^{j\omega t}\sum_{i=1}^{n} A_i(\theta)e^{-jkS_i} \tag{5}$$

The square of the pressure amplitude is given by

$$P_o^2 = \left[\sum_i A_i(\theta)\cos(-kS_i)\right]^2 + \left[\sum_i A_i(\theta)\sin(-kS_i)\right]^2 \tag{6}$$

Alternatively, $P_o^2 = \left[\sum_i A_i(\theta)\cos(kS_i)\right]^2 + \left[\sum_i A_i(\theta)\sin(kS_i)\right]^2$ (6a)

For the rectangular case, all $A_i(\theta)$ are the same for any given θ, hence for this case only,

$$P_o^2 = A_i(\theta)\left\{\left[\sum_i \cos(-kS_i)\right]^2 + \left[\sum_i \sin(-kS_i)\right]^2\right\} \tag{7}$$

The sound pressure level SPL(θ) is given by

$$SPL(\theta) = 10 \log P_o^2 \text{ dB.} \tag{8}$$

SPL(0) is the level on the array longitudinal axis relative to that of a single source.

$$SPL_{rel}(\theta) = SPL(\theta) - SPL(0) \tag{9}$$

Additional Theory for Rectangular Arrays

For an irregular rectangular array, the additional path length of the i^{th} source for a direction θ_h from the axis in the horizontal plane as shown in figure 2 is given by

$$-S_i = \delta_{i1} + \delta_{i2} \tag{10}$$

$$\delta_{i1} = \frac{z_i}{\cos\theta_h} \tag{11}$$

and $\delta_{i2} = (x_i - z_i\tan\theta_h)\sin\theta_h$ (12)

so that $-S_i = \frac{z_i}{\cos\theta_h} + (x_i - z_i\tan\theta_h)\sin\theta_h$ (13)

Similarly, for the direction θ_v in the vertical plane,

$$-S_i = \frac{z_i}{\cos\theta_v} + (y_i - z_i\tan\theta_v)\sin\theta_v \tag{14}$$

For the special case of θ_h, θ_v = ± 90 degrees, (13), (14) are indeterminate in the form given and must be replaced by

$$-S_i = x_i \sin\theta_h \tag{15}$$

$$-S_i = y_i \sin\theta_v \tag{16}$$

The empirical function $A_i(\theta)$ describing the single source directional properties is defined in the following way. Let θ_c be the angle between the -6 dB angles in a given plane, then the sound pressure level SP(θ) at angle θ relative to the axial value

is approximated by

$$SP(\theta) = 30\,F^3(\theta)\{[3.3 - F(\theta)]F(\theta) - 3\} \qquad (17)$$

where

$$F(\theta) = \left|\frac{\theta}{\theta_c}\right| \qquad (18)$$

The form of (17) is shown in figure 3, with a polar plot for a 60 degree horn in figure 4. $A_i(\theta)$ is given by

$$A_i(\theta) = 10^{0.05\,SP(\theta)}, \text{all } i. \qquad (19)$$

The values of (17), (18), (19) will in general be different for the horizontal and vertical planes.

Additional Theory for Circular Arc Arrays

For a circular arc array, the additional path length S_i as shown in figure 5 for the i^{th} source at radius R_i and angle α_i is given by

$$-S_i = R_i \cos(\theta - \alpha_i) \qquad (20)$$

As (17) really gives the dependence of sound pressure levels relative to the source axis, then for arcs, (18) must be modified to

$$F_i(\theta) = \left|\frac{\theta - \alpha_i}{\theta_c}\right| \qquad (21)$$

Hence,

$$SP_i(\theta) = 30\,F_i^3(\theta)\{[3.3 - F_i(\theta)]F_i(\theta) - 3\} \qquad (22)$$

and

$$A_i(\theta) = 10^{0.05\,SP_i(\theta)} \qquad (23)$$

Equation (18) is a special case of (21) when $\alpha_i = 0$ for all i.

Computer Programs

A set of computer programs based on the above theory has been written to execute the calculations needed to predict sound pressure levels on horizontal and vertical planes in the far field of rectangular arrays and in the array plane for circular arc arrays. The programs are in GWBASIC for IBM compatible computers. Listings are given in the Appendix.

The programs can be utilizad by executing the following steps.
1) With the system in DOS, load GRAPHICS.
2) Now load GWBASIC.
3) Insert the disc with the programs into disc drive A and an initialized disc to accept data into disc drive B.
4) Load and run A:RHARRAY.

All further instructions are given as prompts on the screen as the programs run. Because calculated data is stored on the disc in drive B, polar curves can be reproduced at a later date

without having to re-enter the input data and wait for the calculations to be completed. It can take several minutes to complete the calculations.

The units of measure required for the input data are inches for linear dimensions and degrees for angles. The grid of the polar graph output includes concentric circles at 10 dB intervals with the outer circle representing 20 dB and hence the third circle in represents 0 dB. The central point is -40 dB. Any calculated values less than -40 dB are not shown as though these values disappear through a hole in the middle of the graph.

As long as a polar graph remains on the screen, any number of hard copies can be printed. This can be achieved by setting the printer to the top of the next page after a print-out is complete and then simultaneously pressing Shift and PrtSc. The procedure can be repeated until the required number of copies have been printed. This is a very slow process.

Examples

A pair of Renkus Heinz SR-1A units side by side at 30 degrees angular separation is the first example. The horizontal polar plot at 2 kHz. is shown in figure 6.

A regular 2x2 rectangular array of four Renkus Heinz SR-1A cabinets is the next example. Horizontal and vertical polars at 2 kHz. are shown in figures 7 and 8. One each of the top and bottom units was then inverted and the horizontal spacings displaced away from the regular positions in the plane of the array. Three of the units were then moved out of the original array plane. The resulting polars are shown in figures 9 and 10.

Further examples are shown in the following figures. The horizontal polar at 2 kHz. for a regular arc of four Renkus Heinz SR-1A cabinets is shown in figure 11 and that for an irregular arc is shown in figure 12.

Conclusions and Recomendations

By disturbing the positions of sources in regular rectangular arrays to render the spacing irregular, it is possible to achieve polar acoustical radiation patterns with fewer directions having severe phase cancellation than with regularly spaced arrays. This can be seen by comparing figures 7 and 8 with figures 9 and 10. By inverting two of the units, the vertical spacings have been changed from one of 30 in. to two of 18.75 in. and one of 11.25 in. This results in reducing the spacings and rendering them less regular. Similarly, the horizontal spacings have been reduced and rendered more random. Longitudinally displacing the sources by different distances further randomizes the acoustical path differences. A similar effect is observable with circular arc arrays as can be seen from figures 11 and 12.

It is recomended that arrays of loudspeakers should have irregular spacings and should lie neither exactly in a plane nor on a circular arc if possible, so as to reduce the number of directions for severe phase cancellation. It is also recomended that more test cases be investigated to further verify this conclusion.

References

[1] Wolff I. and L. Maiter, "Direct radiation of sound,"J. Acoust. Soc. Amer, Vol. 2, Feb. 1930.

[2] Nyquist H., Bell Systems Tech. J. II, 1932, pp 126 - 147.

[3] Hartman W. H., "Directional characteristics of phased audio reproducers," AES preprint no. 1026, May 1975.

[4] Sinclair R., "Stacked and splayed acoustical sources, Part I," AES preprint no. 1389, Nov. 1978.

[5] Sinclair R., "Stacked and splayed acoustical sources, Part II," AES preprint no. 1515, May 1979.

[6] Sinclair R., "Examination of the far field from phased arrays and the prediction of a conventional antenna far field from mouth measurements," report for XonTech Inc., Van Nuyes, CA., 1984.

[7] Meyer D. G., "Development of a model for loudspeaker dispersion simulation," AES preprint no. 1912, 1982.

[8] Russell D., "Electronic control of speaker directivity, a practical application," AES preprint no. 2389, Nov., 1986.

[9] Ahnert W., "Comb-filter distortions and their perception in sound reinforcement systems," AES preprint no. 2565, 1988.

APPENDIX

```
10 'PROGRAM RHARRAY
20 '
30 'To select options of calculating sound fields for new arrays, plotting
previous results or exiting.
40 '
50 CLS:PRINT:PRINT "The disc with this program should be in disc drive A and a d
isc for storing     results in drive B."
60 INPUT "Is this disc in drive A <Y or N>";AN$:IF AN$<>"Y" THEN 110
70 PRINT:PRINT "Do you wish to calculate field for new array <1>, plot results a
lready stored   <2> or exit <3>?"
80 INPUT "Select option <1, 2 or 3>:",OPT%:ON OPT% GOTO 90,100,120
90 RUN "A:ARRAYCAL"
100 RUN "A:ARRAYPLT"
110 PRINT:PRINT "Start over with this disc in drive A."
120 END
```

```
10 'PROGRAM RHAR1, shortened version of RHARRAY
20 '
30 'To select options of calculating sound fields for new arrays, plotting previ
ous results or exiting.
40 '
50 PRINT:PRINT "Do you wish to calculate field for new array <1>, plot results a
lready stored   <2> or exit <3>?"
60 INPUT "Select option <1, 2 or 3>:",OPT%:ON OPT% GOTO 70,80,90
70 RUN "A:ARRAYCAL"
80 RUN "A:ARRAYPLT"
90 END
```

```
10 'PROGRAM ARRAYCAL
20 '
30 'FAR FIELD SOUND DISTIBUTION FOR ARRAYS
40 '            by Rex Sinclair
50 '          for Renkus Heinz Inc.
60 '
70 'Input of data common to rectangular and circular arrays.
80 '
90 DIM X(3,30),SP(180,2),AR(30),ALPH(30),ALPHR(30)
100 CPI=4*ATN(1):DTR=CPI/180:KTF=2139:NTT=LOG(10)
110 PRINT:INPUT "Enter identification of sources: ",ID$:FL$="B:"+ID$
120 INPUT "No. of sources (30 max.)";N%
130 INPUT "Choose rectangular array (1) or circular arc (2): ",T%:IF T%=1 THEN 1
90
140 PRINT "Enter radial distance (inches) from origin and angle (deg.) from axis
 for each  source as prompted."
```

```
150 PRINT "Origin is centre of curvature of arc.  Positive angles are upward or
right for  arrays convex toward listening area, or downward or left for arrays c
oncave     toward listening area."
160 PRINT "Enter distance and angle for source no.:"
170 FOR SN%=1 TO N%:PRINT SN%;": ";:INPUT AR(SN%),ALPH(SN%):ALPHR(SN%)=ALPH(SN%)
*DTR:NEXT SN%
180 INPUT "Enter angle between -6dB points in plane of arc for a single source:
",THC:THCR=THC*DTR:GOTO 240
190 PRINT "Enter x, y, z co-ordinates (inches) of each source as prompted.  Co-o
rdinates   are from any reference point, typically a source."
200 PRINT "Positive x is to the right looking at array from listening area, posi
tive y is upward."
210 PRINT "Positive z is toward listening area.":PRINT "Enter x, y, z in inches
for source no.:"
220 FOR SN%=1 TO N%:PRINT SN%;": ";:INPUT X(1,SN%),X(2,SN%),X(3,SN%):NEXT SN%
230 INPUT "Enter angles between -6dB points in horizontal and vcertical planes:
",TH6(1),TH6(2)
240 INPUT "Frequency";F:K=F/KTF
250 '
260 '
270 'Transfer of input data to disc.
280 '
290 OPEN "O",1,FL$:PRINT #1,ID$:PRINT #1,N%:PRINT #1,F:PRINT #1,T%:IF T%=2 THEN
480
300 FOR J%=1 TO 2:PRINT #1,TH6(J%):NEXT J%
310 FOR SN%=1 TO N%:FOR K%=1 TO 3:PRINT #1,X(K%,SN%):NEXT K%:NEXT SN%
320 '
330 '
340 'Calculation for rectangular arrays and transfer of data to disc.
350 '
360 FOR R%=1 TO 2:FOR Q%=0 TO 180
370 FOR P%=1 TO 2: SUMC=0:SUMS=0:ANG=(-1)^P%*Q%*DTR
380 FOR J%=1 TO N%:IF Q%=90 THEN S=(-1)^P%*X(R%,J%) ELSE S=X(3,J%)/COS(ANG)+(X(R
%,J%)-X(3,J%)*TAN(ANG))*SIN(ANG)
390 SUMC=SUMC+COS(K*S):SUMS=SUMS+SIN(K*S):NEXT J%
400 P2=SUMC*SUMC+SUMS*SUMS:QN=Q%/TH6(R%):SP1=30*QN^3*((3.3-QN)*QN-3)
410 SP(Q%,P%)=10*LOG(P2)/NTT+SP1:IF Q%=0 THEN SP0=SP(Q%,P%)
420 SP(Q%,P%)=SP(Q%,P%)-SP0:NEXT P%
430 PRINT #1,SP(Q%,1):PRINT #1,SP(Q%,2):NEXT Q%:NEXT R%:GOTO 610
440 '
450 '
460 'Calculation for circular arc arrays and transfer of data to disc.
470 '
480 PRINT #1,THC:FOR SN%=1 TO N%:PRINT #1,AR(SN%):PRINT #1,ALPH(SN%):NEXT SN%
490 FOR Q%=0 TO 180:FOR P%=1 TO 2:SUMC=0:SUMS=0:QP=(-1)^P%*Q%:ANG=QP*DTR
500 FOR J%=1 TO N%:FQ=ABS(QP-ALPH(J%))/THC:SP1=30*FQ^3*((3.3-FQ)*FQ-3):IF SP1<-1
00 THEN SP1=-100
510 A=10^(.05*SP1):S=AR(J%)*COS(ANG-ALPHR(J%))
520 SUMC=SUMC+A*COS(K*S):SUMS=SUMS+A*SIN(K*S):NEXT J%
530 P2=SUMC*SUMC+SUMS*SUMS:IF P2<1E-10 THEN P2=1E-10
540 SP(Q%,P%)=10*LOG(P2)/NTT:IF Q%=0 THEN SP0=SP(Q%,P%)
550 SP(Q%,P%)=SP(Q%,P%)-SP0:NEXT P%
560 PRINT #1,SP(Q%,1):PRINT #1,SP(Q%,2):NEXT Q%:GOTO 610
570 '
580 '
590 'Transfer of common data to disc and return to master program.
600 '
610 PRINT #1,SP0:CLOSE #1:RUN "A:RHAR1":END
```

```
10 'PROGRAM ARRAYPLT
20 '
30 'To plot results calculated in ARRAYCAL.
40 '
50 DIM X(3,30),SP(180,2,2),AR(30),ALPH(30)
60 DTR=ATN(1)/45:PRINT:INPUT "Enter identification of array to be simulated: ",I
D$:FL$="B:"+ID$:OPEN "I",1,FL$
70 INPUT #1,ID$:INPUT #1,N%:INPUT #1,F:INPUT #1,T%:IF T%=2 THEN 130
80 FOR J%=1 TO 2:INPUT #1,TH6(J%):NEXT J%
90 FOR SN%=1 TO N%:FOR K%=1 TO 3:INPUT #1,X(K%,SN%):NEXT K%:NEXT SN%
100 DIR$(1)="Horizontal":DIR$(2)="Vertical"
110 FOR R%=1 TO 2:FOR Q%=0 TO 180:FOR P%=1 TO 2:INPUT #1,SP(Q%,P%,R%):IF SP(Q%,P
%,R%)<-40 THEN SP(Q%,P%,R%)=-40
120 NEXT P%:NEXT Q%:NEXT R%:GOTO 160
130 INPUT #1,TH6(1):FOR SN%=1 TO N%:INPUT #1,AR(SN%):INPUT #1,ALPH(SN%):NEXT SN%
140 FOR Q%=0 TO 180:FOR P%=1 TO 2:INPUT #1,SP(Q%,P%,1):IF SP(Q%,P%,1)<-40 THEN S
P(Q%,P%,1)=-40
150 NEXT P%:NEXT Q%
160 INPUT #1,SP0:IF T%=1 THEN 210
170 R%=1:DIR$(1)=" ":GOSUB 420
180 PRINT "Coverage angle = ";TH6(1);" deg.":PRINT "Source no."," R","Angle on
arc"
190 FOR SN%=1 TO N%:PRINT USING "   ##         ####.###         ####.##";SN%,AR(SN
%),ALPH(SN%):NEXT SN%
200 PRINT "On-axis SPL relative to":PRINT USING "one source = ###.## dB.";SP0:GO
TO 260
210 FOR R%=1 TO 2:GOSUB 420
220 PRINT "Coverage angles (horizontal, vertical) = ";TH6(1);",";TH6(2);" deg."
230 PRINT "Source no.   x         y         z"
240 FOR SN%=1 TO N%:PRINT USING "    ##    ####.###  ####.###  ####.###";SN%,X(1
,SN%),X(2,SN%),X(3,SN%):NEXT SN%
250 PRINT "On-axis SPL relative to":PRINT USING "one source = ####.## dB.";SP0
260 GOSUB 370
270 FOR P%=1 TO 2:DRAW "BM 556,130"
280 FOR Q%=1 TO 180:QR=Q%*DTR:X2=460+2.4*(SP(Q%,P%,R%)+40)*COS(QR)
290 Y2=130-(-1)^P%*(SP(Q%,P%,R%)+40)*SIN(QR)
300 LINE -(X2,Y2):NEXT Q%:NEXT P%
310 INPUT DUM$:CLS:IF T%=2 THEN 330
320 NEXT R%
330 CLOSE #1:SCREEN 0:RUN "A:RHAR1"
340 '
350 'Subroutine to plot polar grid
360 '
370 FOR RAD=24 TO 144 STEP 24:CIRCLE(460,130),RAD:NEXT RAD
380 FOR D=0 TO 330 STEP 30:DRAW "TA=D;NU 60":NEXT D:RETURN
390 '
400 'Subroutine for instructions and print-out of common data.
410 '
420 PRINT:PRINT "Set printer to top of page.":PRINT "For hard copy of polar, pre
ss Shift PrtSc   when prompt (?) appears.  When print-out  is complete, hit Ente
r to continue."
430 PRINT "If no print-out is required, simply hit Enter after prompt (?).":INPU
T "Now hit Enter to continue. ",DUM$
440 CLS:SCREEN 2:PRINT ID$;", No. in array = ";N%;", frequency = ";F;" Hz.
    ";DIR$(R%)
450 RETURN
460 END
```

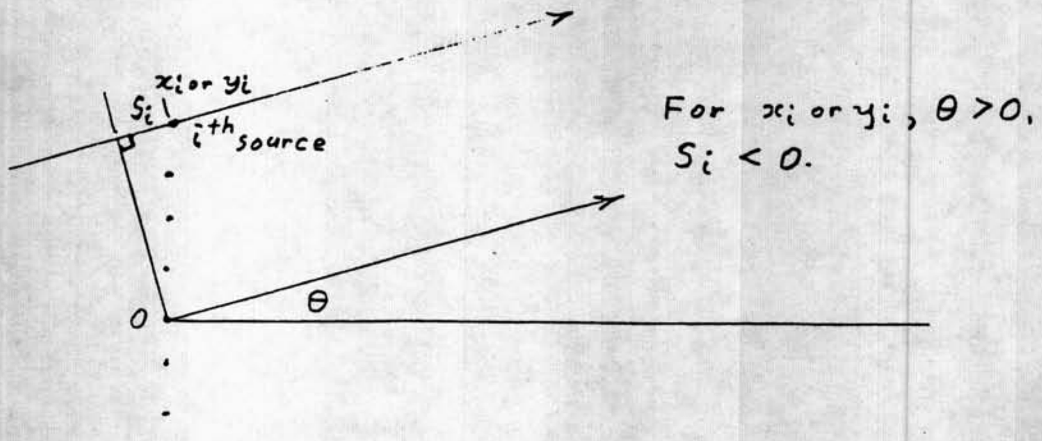


Fig. 1. Path length difference of i^{th} source in a regular rectangular array.

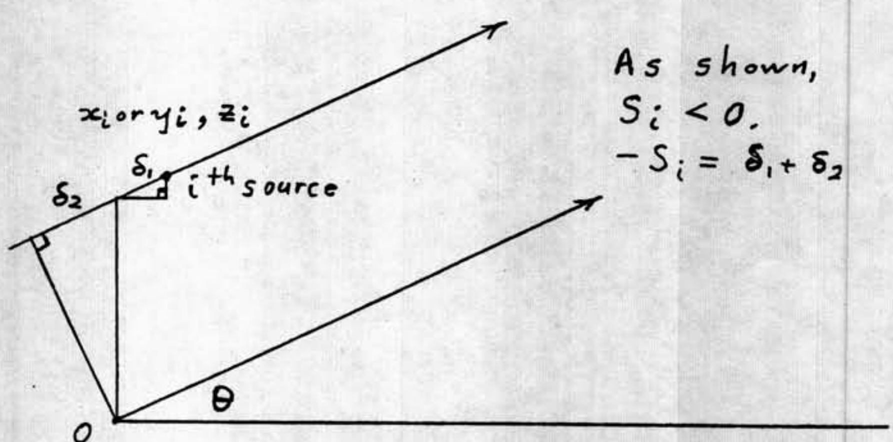


Fig. 2. Path length difference of i^{th} source in an irregular rectangular array.

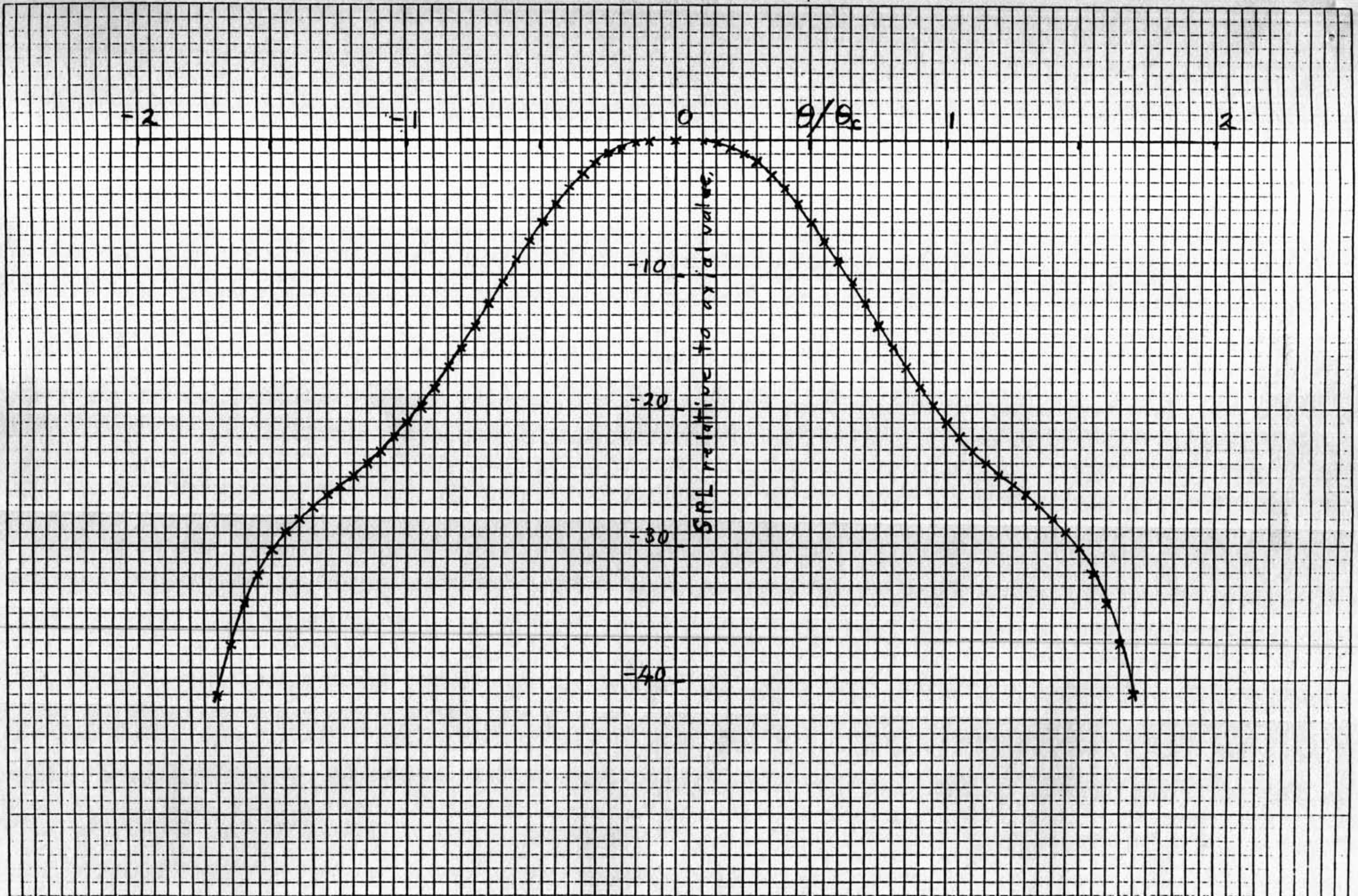


Fig. 3. Empirical SPL dependence on off-axis angle.

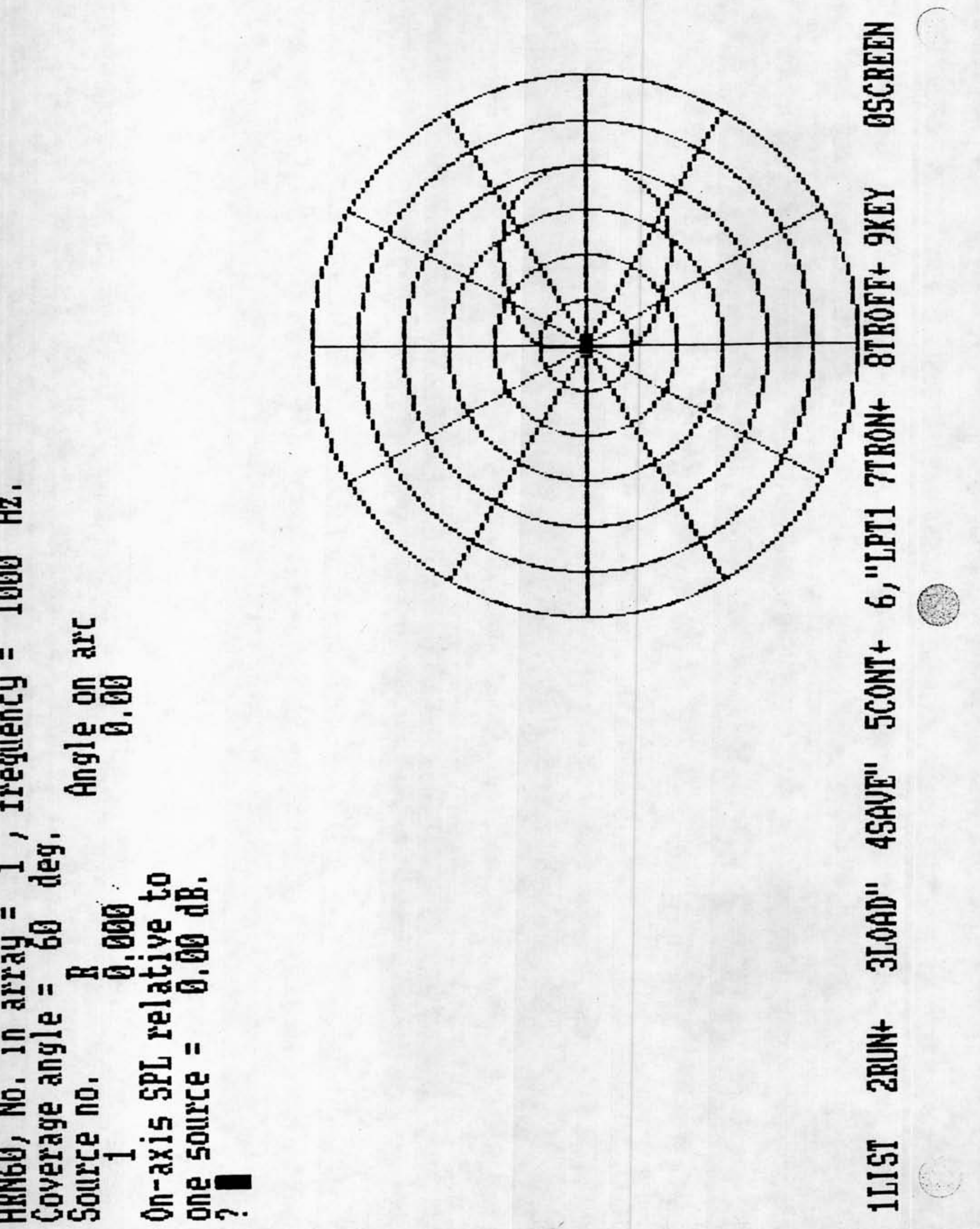
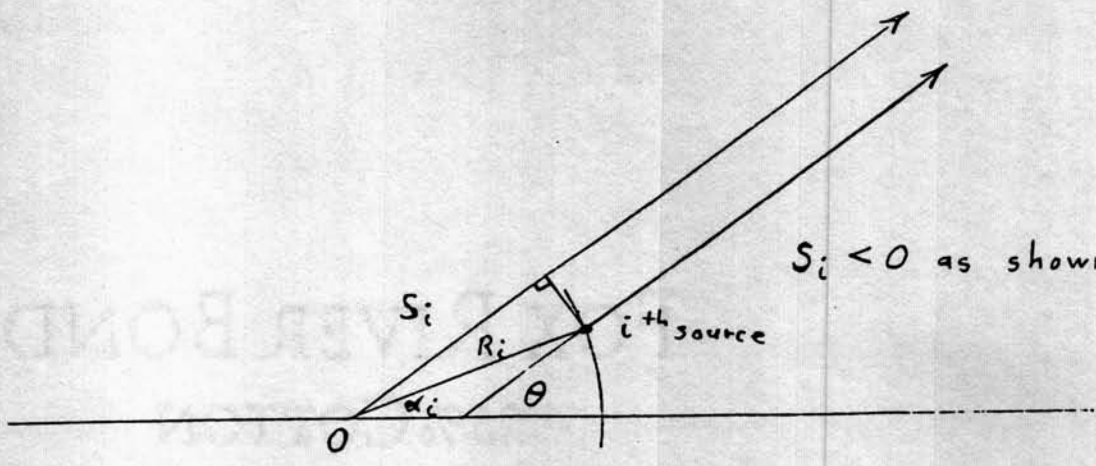
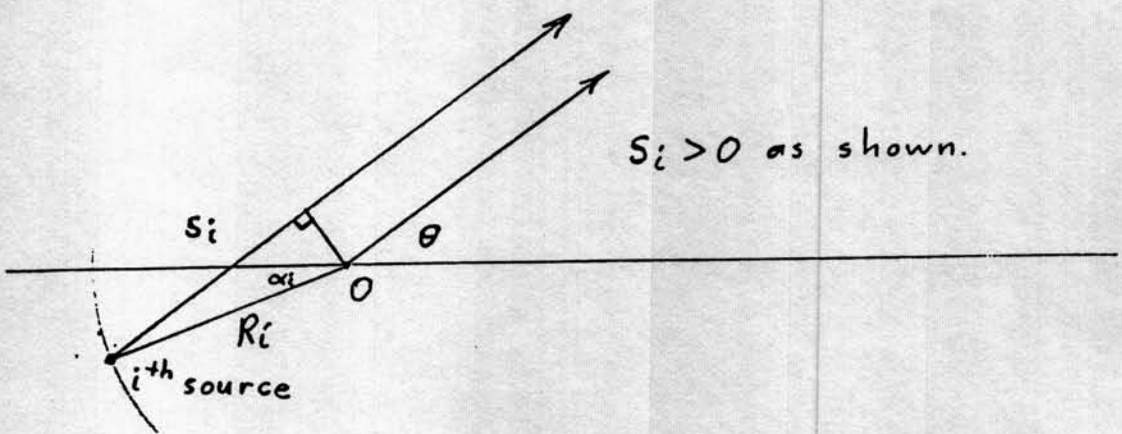


Fig. 4. Empirical SPL relationship applied to a 60 degree horn, $\theta_c = 60$.



a) Convex array



b) Concave array

Fig. 5. Path length difference of i^{th} source in convex and concave circular arrays.

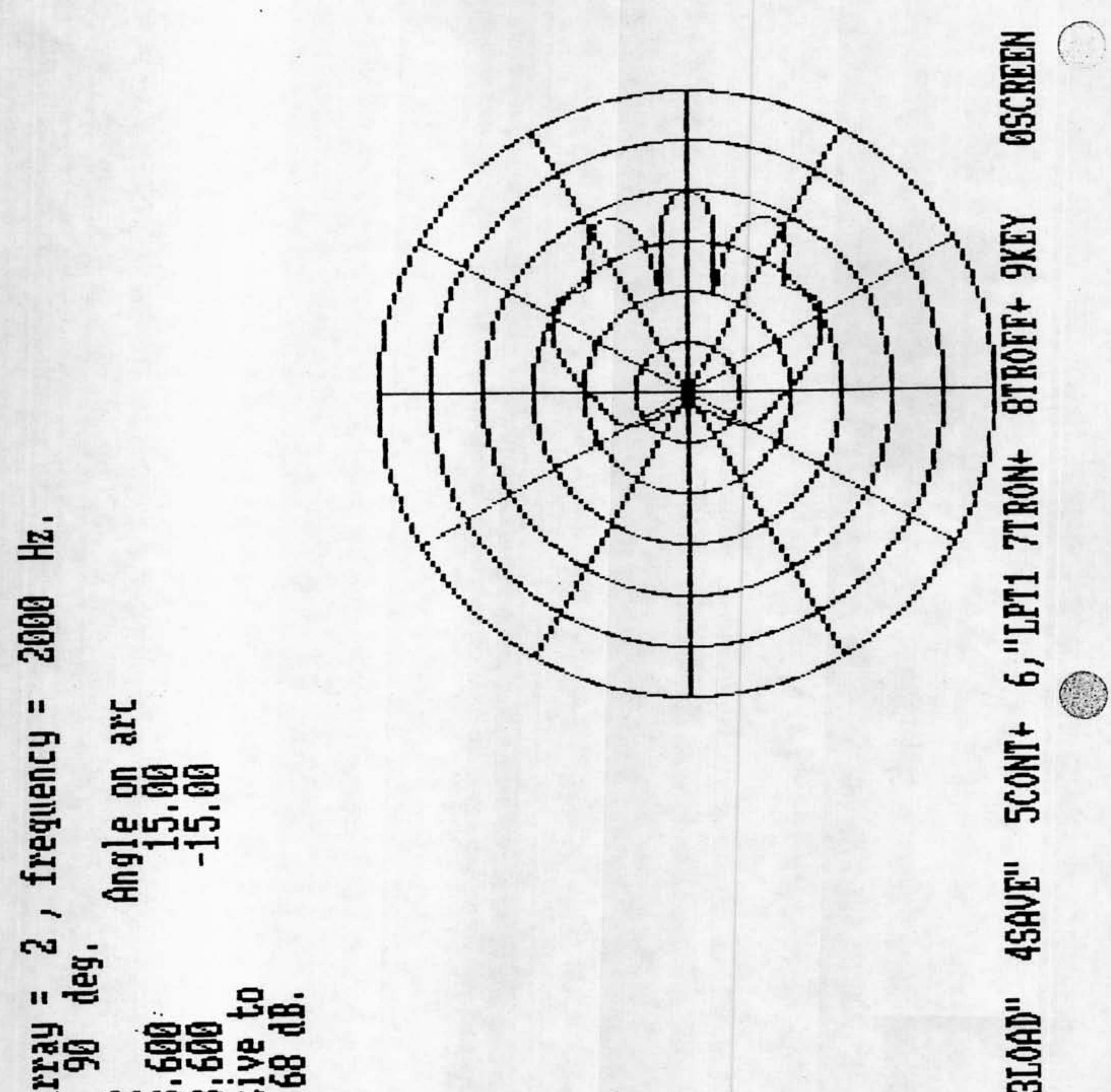
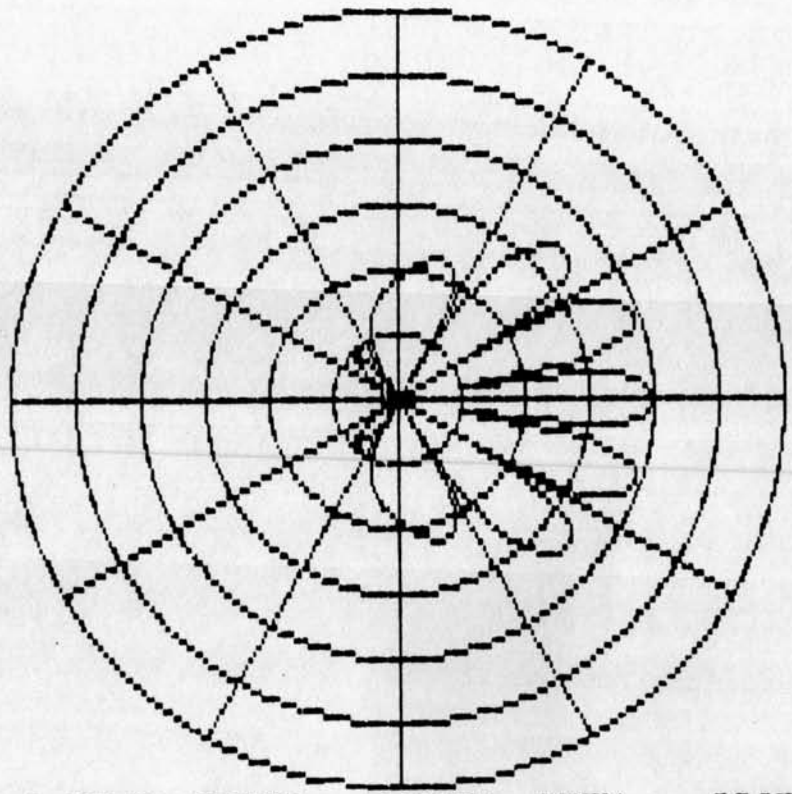


Fig. 6. Horizontal polar pattern for two Renkus Heinz SR-1A cabinets side by side with 30 degrees angular separation at 2 kHz.

REC4REG, No. in array = 4 , frequency = 2000 Hz. Horizontal
Coverage angles (horizontal, vertical) = 90 , 60 deg.

Source no.	x	y	z
1	0.000	0.000	0.000
2	19.750	0 000	0.000
3	0.000	30 000	0.000
4	19.750	30 000	0.000

On-axis SPL relative to
one source = 12.04 dB.
? █



1LIST 2RUN← 3LOAD" 4SAVE" 5CONT← 6,"LPT1 7TRON← 8TROFF← 9KEY 0SCREEN

Fig. 7. Horizontal polar pattern of 2x2 regular rectangular array of Renkus Heinz SR-1A cabinets at 2 kHz.

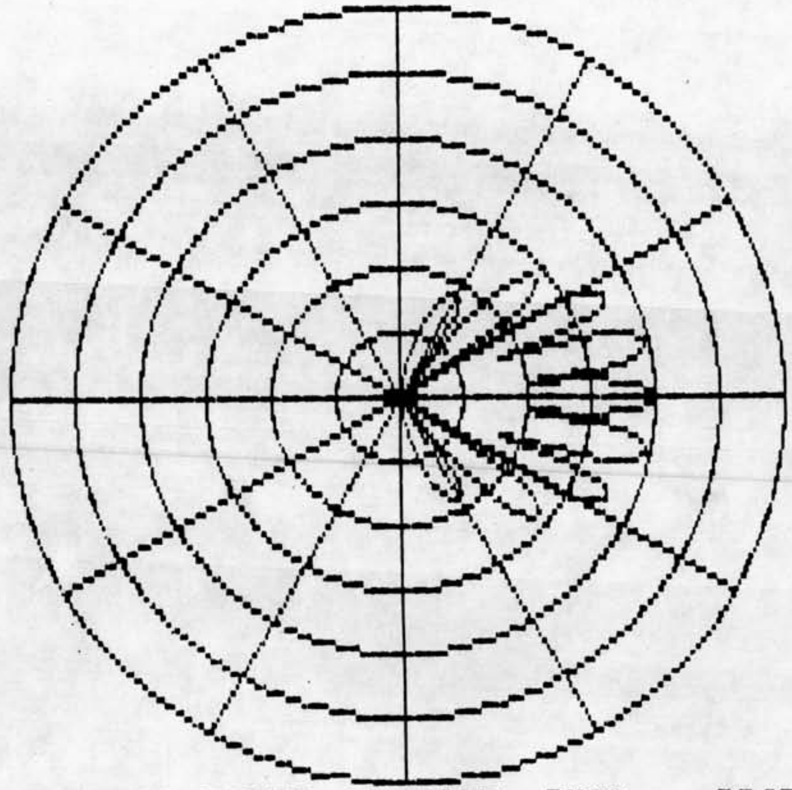
REC4REG, No. in array = 4 , frequency = 2000 Hz. Vertical

Coverage angles (horizontal, vertical) = 90 , 60 deg.

Source no.	x	y	z
1	0.000	0.000	0.000
2	19.750	0.000	0.000
3	0.000	30.000	0.000
4	19.750	30.000	0.000

On-axis SPL relative to
one source = 12.04 dB.

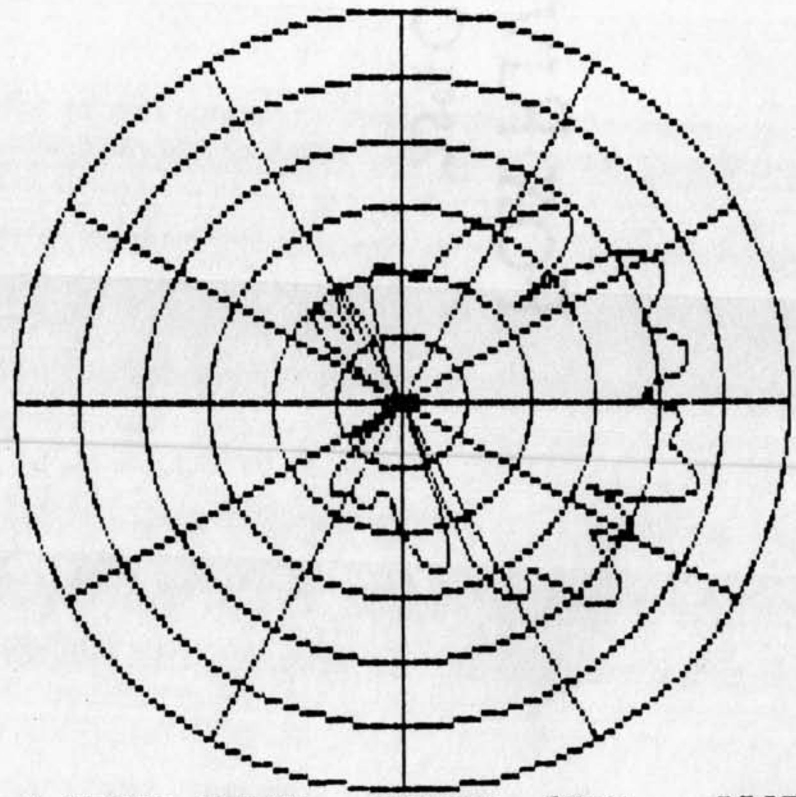
?



1LIST 2RUN← 3LOAD" 4SAVE" 5CONT← 6,"LPT1 7TRON← 8TROFF← 9KEY 0SCREEN

Fig. 8. Vertical polar pattern of 2x2 regular rectangular array of Renkus Heinz SR-1A cabinets at 2 kHz.

```
REC4IRR, No. in array =  4 , frequency =  2000  Hz.          Horizontal
Coverage angles (horizontal, vertical) =  90 , 60  deg.
Source no.     x          y          z
     1       0.000      0.000      0.000
     2      28.750    -18.750      9.000
     3       6.000     11.250     -3.000
     4      22.750     30.000     -6.000
On-axis SPL relative to
one source =     1.44 dB.
? █
```



```
1LIST   2RUN←   3LOAD"   4SAVE"   5CONT←   6,"LPT1  7TRON←   8TROFF← 9KEY    0SCREEN
```

Fig. 9. Horizontal polar pattern of 2x2 irregular rectangular array of Renkus Heinz SR-1A cabinets at 2 kHz.

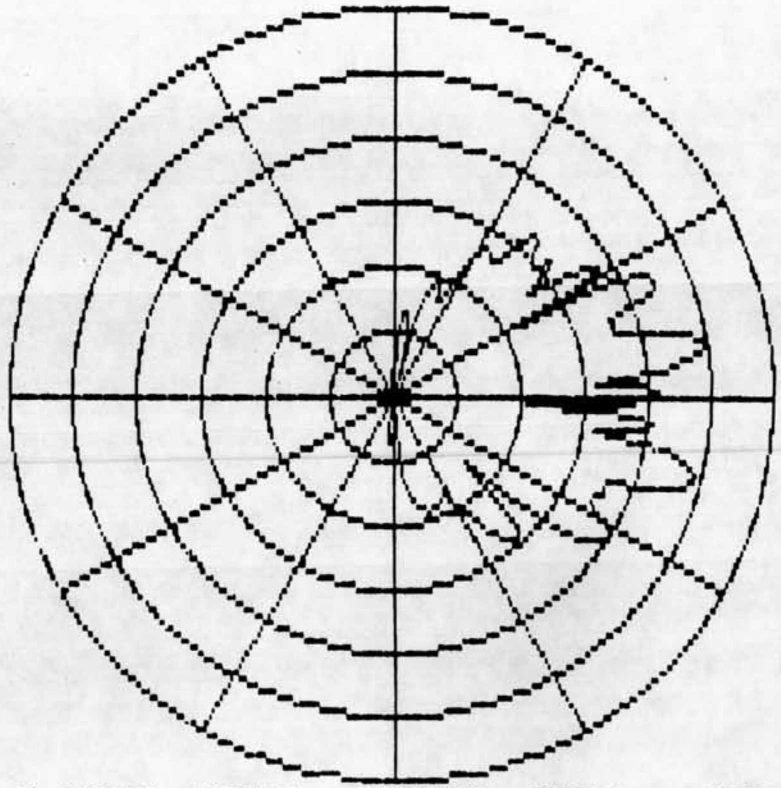
REC4IRR, No. in array = 4 , frequency = 2000 Hz. Vertical

Coverage angles (horizontal, vertical) = 90 , 60 deg.

Source no.	x	y	z
1	0.000	0.000	0.000
2	28.750	-18.750	9.000
3	6.000	11.250	-3.000
4	22.750	30.000	-6.000

On-axis SPL relative to
one source = 1.44 dB.

?



1LIST 2RUN← 3LOAD" 4SAVE" 5CONT← 6,"LPT1 7TRON← 8TROFF← 9KEY 0SCREEN

Fig. 10. Vertical polar pattern of 2x2 irregular rectangular array of Renkus Heinz SR-1A cabinets at 2 kHz.

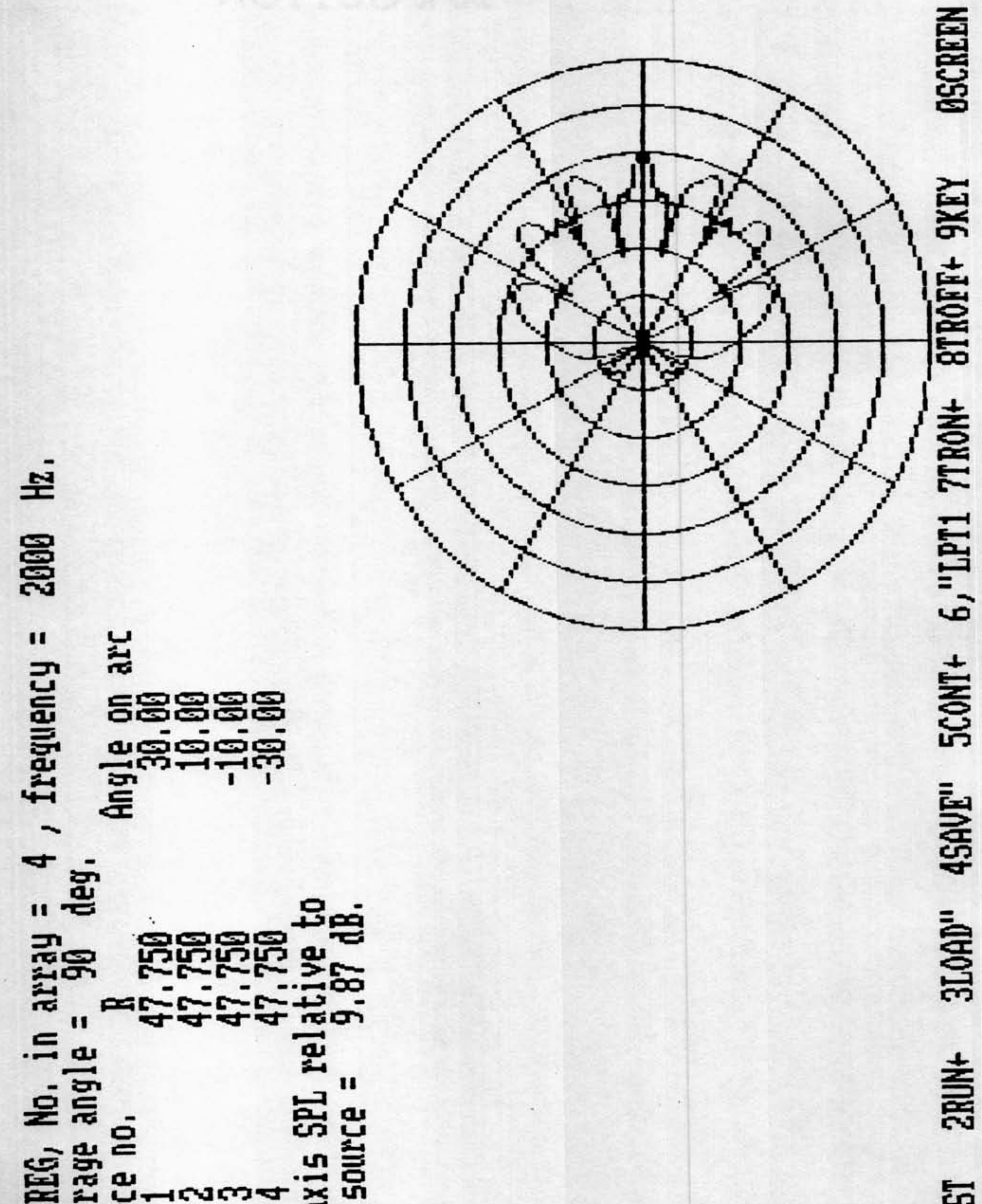
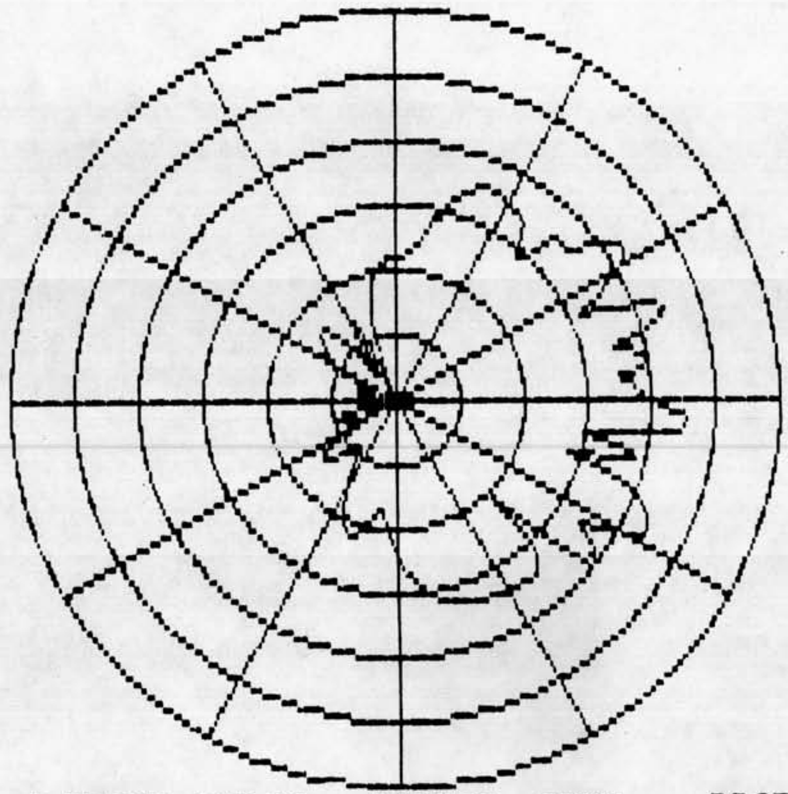


Fig. 11. Polar pattern for regular circular arc array of four Renkus Heinz SR-1A cabinets at 2 kHz.

ARC4IRR, No. in array = 4 , frequency = 2000 Hz.
Coverage angle = 90 deg.

Source no.	R	Angle on arc
1	49.750	30.00
2	47.750	9.50
3	40.750	-12.00
4	52.750	-30.00

On-axis SPL relative to
one source = 4.11 dB.
?



1LIST 2RUN← 3LOAD" 4SAVE" 5CONT← 6,"LPT1 7TRON← 8TROFF← 9KEY 0SCREEN

Fig. 12. Polar pattern for irregular circular arc array of four Renkus Heinz SR-1A cabinets at 2 kHz.

ADDENDUM

Extension to other planes

For rectangular arrays, polars can be found for planes other than the horizontal and vertical ones. New x and y axes, x' and y' can be defined by a rotation β of the co-ordinate system about the z axis. The appropriate transform equations are:

$$x'_i = x_i \cos\beta + y_i \sin\beta \tag{24}$$

and

$$y'_i = y_i \cos\beta - x_i \sin\beta. \tag{25}$$

For circular arrays, the co-ordinates α', R' in a plane rotated β relative to the plane of the arc are given by:

$$\alpha'_i = \tan^{-1}(\tan\alpha_i \cos\beta) \tag{26}$$

and

$$R'_i = \frac{R_i \cos\alpha_i}{\cos\alpha'_i}. \tag{27}$$

For both kinds of array, the appropriate values of θ_c can be found from published contours, interpolated from horizontal and vertical values or determined experimentally. Similarly, the position of the apparent apex in the appropriate planes can be found by interpolation or experimentally.

Program Changes for Alternative Computer Configuration

The computer programs listed in the Appendix are for IBM compatibles with two floppy disc drives labeled A and B. To run on machines having one floppy disc drive labeled A and a hard disc labeled C, the programs should be loaded into disc C. Drive A should be used to store calculated data onto a floppy disc. The following program changes are then needed.

RHARRAY
Line 50, last statement, PRINT "The disc with this program should be in disc drive A and a disc for storing results in disc drive B." becomes PRINT "Do you have a disc for data storage in disc drive A <Y or N>?"; .
Line 60 INPUT "Is this disc in drive A <Y or N>?";AN$:IF AN$<>"Y" THEN 110 becomes INPUT AN$:IF AN$<>"Y" THEN 110
Line 90 RUN "A:ARRAYCAL" becomes RUN "C:ARRAYCAL". Line 100 RUN "A:ARRAYPLT" becomes RUN "C:ARRAYPLT".
Line 110 PRINT:PRINT "start over with this disc in drive A." becomes PRINT:PRINT "Start over with formatted disc in drive A."

RHAR1
Line 70 RUN "A:ARRAYCAL" becomes RUN "C:ARRAYCAL".
Line 80 RUN "A:ARRAYPLT" becomes RUN "C:ARRAYPLT".

ARRAYCAL
Line 110, last statement, FL$="B:"+ID$ becomes FL$="A:"+ID$.
Line 610, last statement, RUN "A:RHAR1" becomes RUN "C:RHAR1".

ARRAYPLT
Line 60, second from last statement, FL$="B:"+ID$ becomes FL$="A:"+ID$.
Line 330 RUN "A:RHAR1" becomes RUN "C:RHAR1".

RH3-160, no. in array = 3 , frequency = 160 Hz. Vertical
One third octave pink noise.
On-axis SPL relative to source no. 1 alone = 20.41 dB.
?

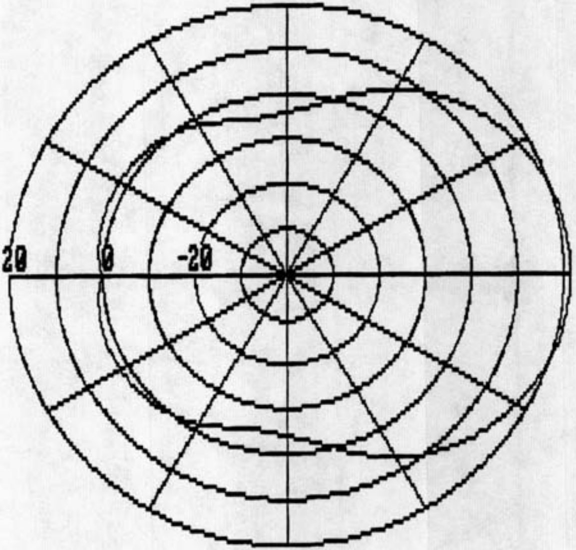


Figure 15f. Low frequency line array vertical polar prediction at 160 Hz.

RH3-200, no. in array = 3 , frequency = 200 Hz.   Vertical
One third octave pink noise.
On-axis SPL relative to source no. 1 alone = 11.57 dB.
?

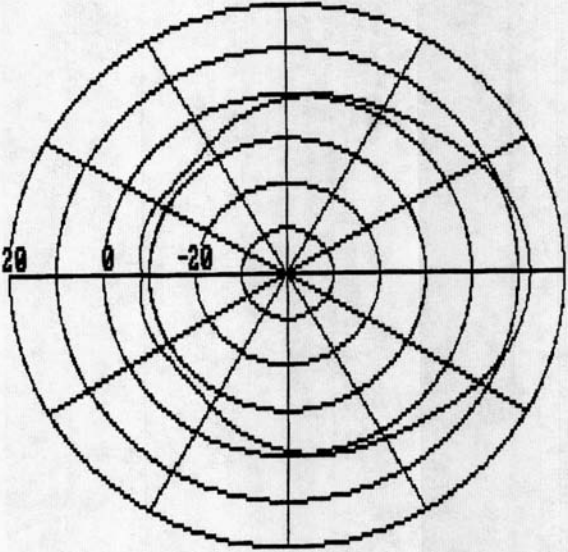


Figure 15g. Low frequency line array vertical polar prediction at 200 Hz.

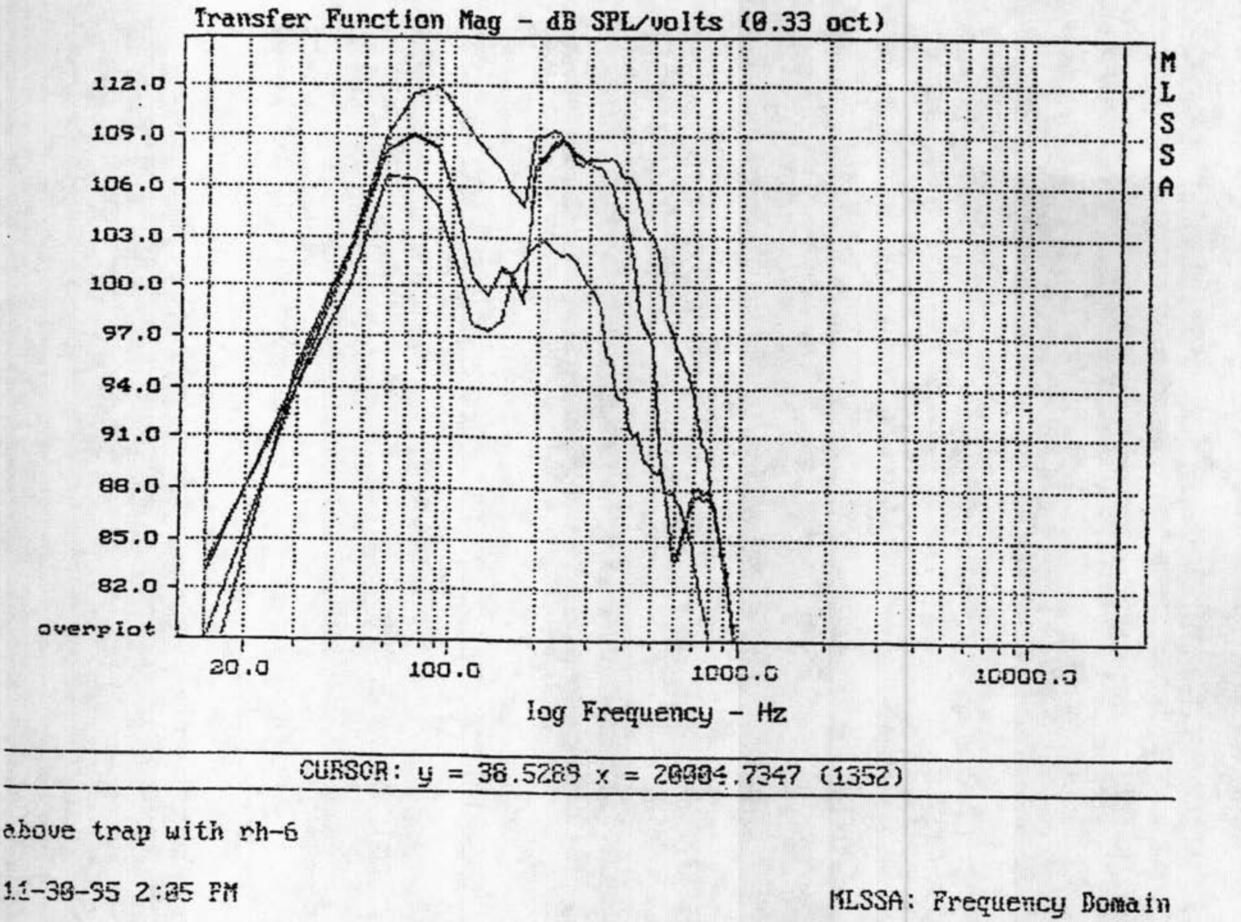


Figure 18. Low frequency line array measurements made on axis, 45 degrees, and 90 degrees in the vertical plane above the array. Notice the lobing at certain frequencies in the hemisphere above the array, generally in the direction of the ceiling of a room. This characteristic makes the array pretty much unusable in this direction.

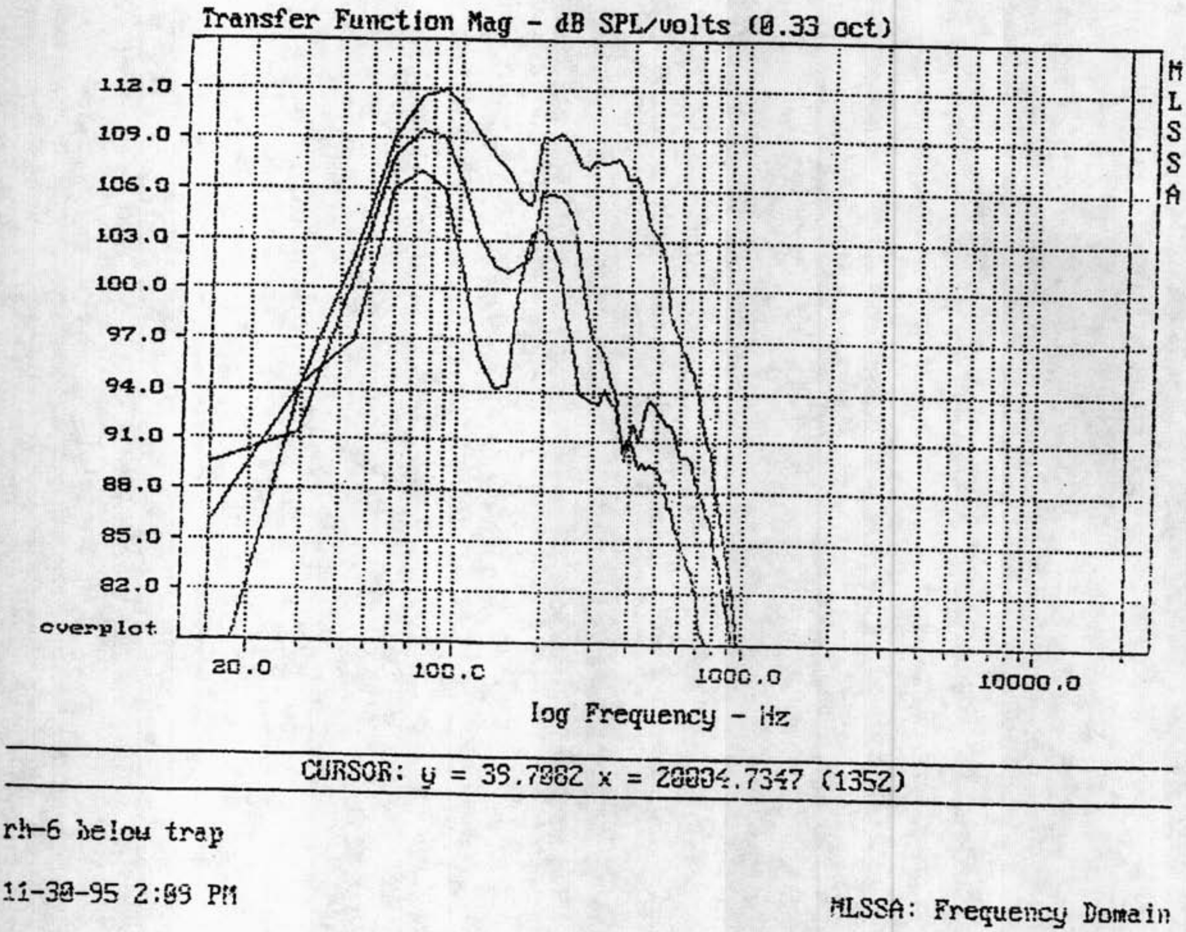


Figure 19. Low frequency line array free field measurements made on axis, 45 degrees, and 90 degrees in the vertical plane below the array (in the direction of what would be the seating area). Performance 90 degree off axis is at least -6 dB for all frequencies down to 90 Hz, -5 dB down to 70 Hz.

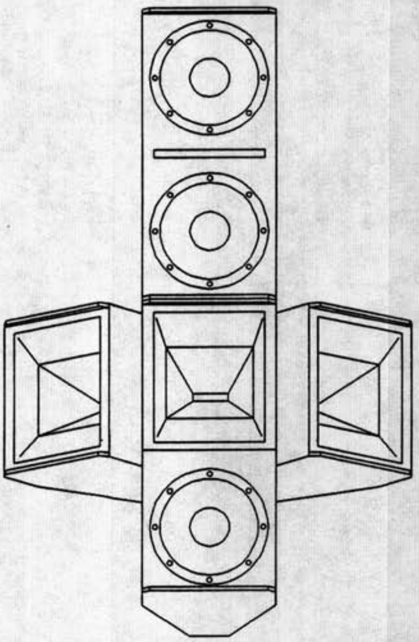


Figure 20. A 3x40 mid high array configuration with a single low frequency line array.

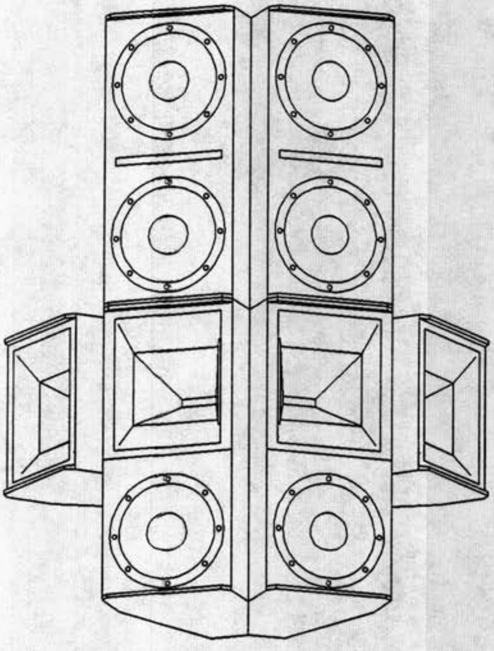


Figure 21. A 4x40 mid high array configuration with a double low frequency line array.

Minimizing Interference in Circular Arc Arrays

For a distance to the listening area very much larger than the array dimensions, let the sound pressure P be the real part of

$$P(\theta) = A(\theta)\varepsilon^{j(\omega\Xi\tau - kSi)}$$

where P is the sound pressure, ω is the angular frequency, $A_i(\theta)$ is a function of the angle between the array longitudinal axis and the direction of the distant listening point. It gives the ratio of the sound pressure due to the source as a ratio of its on axis value at the same distance. For the *i* th source shown in Fig. 1, assuming identical sources, the pressure contribution is given by

$$P_i = A_i(\theta)\varepsilon^{j(\omega\tau - kSi)}$$

where $k = 2\pi/\lambda = 2\pi f/c$, λ is the wavelength, f is the frequency and c is the speed of sound. S is the distance by which the path length from the I th source to the distant point exceeds the distance from the origin to that point.

For an array of n sources, the total pressure P is given by

$$P(\theta) = \sum_{i=1}^{n} A_i(\theta)\varepsilon^{j(\omega\Xi\tau - kSi)}$$

The square of the pressure amplitude is given by

$$P_0^2(\theta) = [\sum_{i=1}^{n} A_i(\theta)\cos(k\, S_i\,)]^2 + [\sum_{i=1}^{n} A_i(\theta)\sin(k\, S_i\,)]^2$$

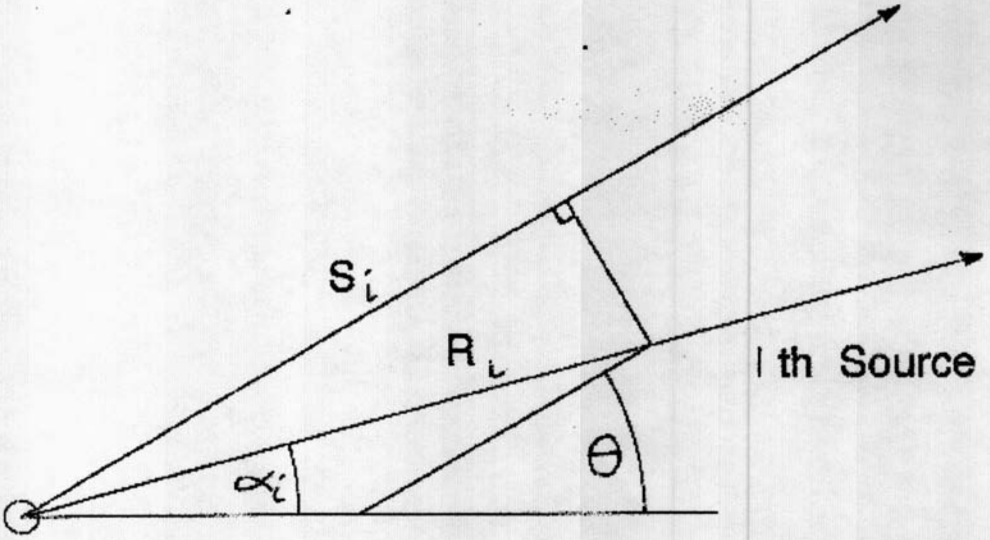
where $A_i(\theta) = A_i(\theta - \alpha_l)$

For a circular arc array, the additional path length Si as shown in Fig. For the ith source at radius R and angle is given by

$$-S_i\,(\theta) = R_i\cos(\theta - \alpha_l)$$

The smaller R is, the smaller are S_i differences. This means less interference. Ideally R = 0.

Fig. 1



Minimizing Interference in Circular Arc Arrays

Definitions:

1. P = sound pressure

Assumptions:

1. The sources are identical with equal acoustic output.
2. The sources are arranged in a plane at an angle to each other such that their axis intersect at a point. This point is used as the origin of a set of polar coordinates (0 in Fig.1).
3. The distance to the listening area is very much larger thasn the array dimensions.

General Theory

P = A

where P is the sound pressure, ω is the angular frequency, A(θ) is a function of the angle between the array longitudinal axis and the direction of the distant listening point. It gives the ratio of the sound pressure due to the source as a ratio of its on axis value at the same distance. For the I th source shown in Fig. 1, the pressure contribution is given by where $k = 2\pi/\lambda = 2\pi f/c$, λ is the wavelength, f is the frequency and c is the speed of sound. S is the distance by which the path length from the I th source to the distant point exceeds the distance from the origin to that point.

For an array of n sources, the total pressure P is given by

$P(\theta) = \Sigma A_i(\theta)\varepsilon^{j(\omega\Xi\tau-kSi)}$

The square of the pressure amplitude is given by

$P_0^2(\theta) = [\Sigma A_i(\theta)\cos(k S_i)]^2 + [\Sigma A_i(\theta)\sin(k S_i)]^2$

and

The empirical function A (θ) describing the single source directional properties is defined in the following way. Let be the angle between the -6 dB angles in a given plane, then the sound pressure level SP(θ) at angle θ relative to the axial value is approximated by

$SP(\theta) = 30F_i(\theta)$

where for arcs $F_i(\theta) = |(\theta - \alpha_i)/\theta_c|$

For a circular arc array, the additional path length Si as shown in Fig. For the ith source at radius R and angle is given by

Hence, $SP_i(\theta) = 30\ SP_i(\theta) = 30F_i^3(\theta)\{[3.3- F_i(\theta)]F_i(\theta)-3\}^3(\theta)\{[3.3- F_i(\theta)]F_i(\theta)-3\}$

and $A_i(\theta)= 10^{0.05SPi(\theta)}$

COPY

United States Patent [19]

Lemon

[11] Patent Number: 4,862,508

[45] Date of Patent: Aug. 29, 1989

[54] METHOD FOR LARGE-SCALE MULTIPLE SOURCE SOUND REINFORCEMENT

[75] Inventor: John Lemon, Shipbottom, N.J.

[73] Assignee: U.S. Sound, Inc., Philadelphia, Pa.

[21] Appl. No.: 61,099

[22] Filed: Jun. 10, 1987

[51] Int. Cl.[4] H04R 1/00

[52] U.S. Cl. 381/156; 381/182; 181/145; 181/152

[58] Field of Search 381/24, 88, 90, 182, 381/156, 205, 186, 89; 181/145, 152

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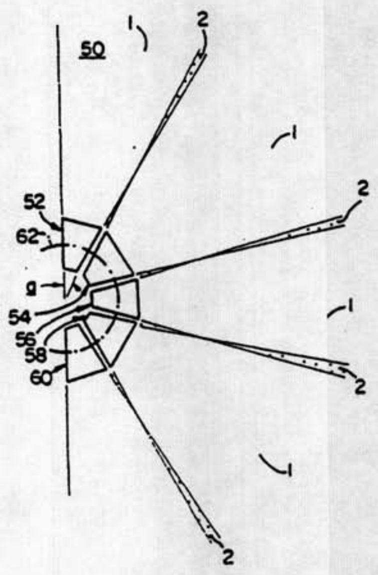
Primary Examiner—Forester W. Isen

Attorney, Agent, or Firm—Steele, Gould & Fried

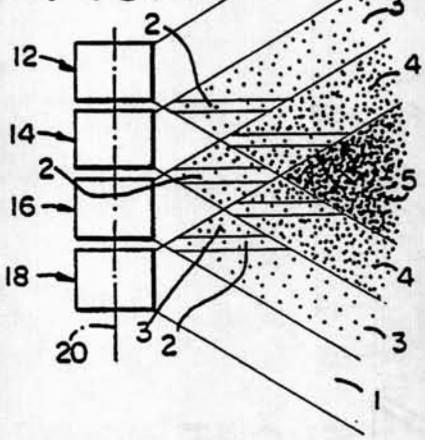
[57] ABSTRACT

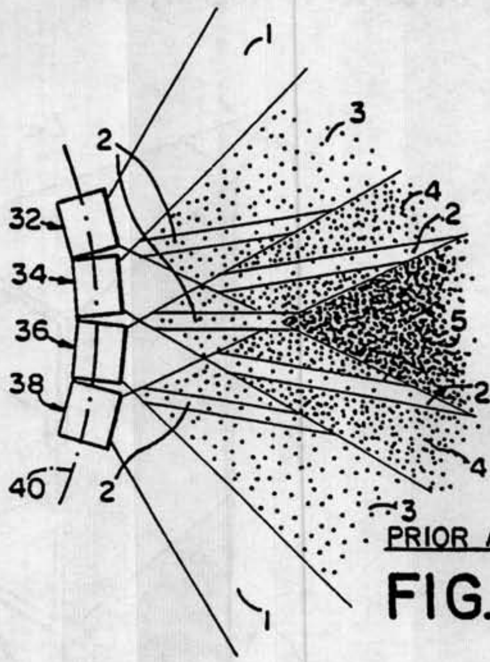
An improved method for transmitting sound at high power levels over a wide angle zone of dispersion without distortion, comprising the step of emitting sound waves from a plurality of individual sources, each characterized by a relatively narrow, wedge-shaped envelope of sound projection, such that adjacent edges of respective sound projection envelopes are in substantial alignment and do not overlap, whereby the absence of interference between sounds emitted from different sources precludes sound distortion and enables uniform sound dispersion and high sound quality throughout the zone. The sound waves are preferably emitted from electroacoustical loudspeakers having loudspeaker enclosures shaped to conform to the edges of their respective sound envelopes.

18 Claims, 2 Drawing Sheets



PRIOR ART
FIG. 1





PRIOR ART
FIG. 2

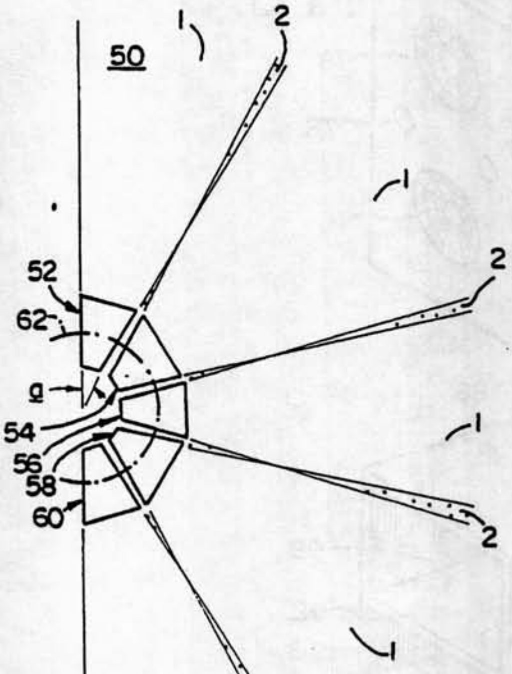


FIG. 3

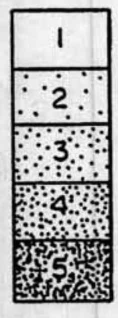


FIG. 4

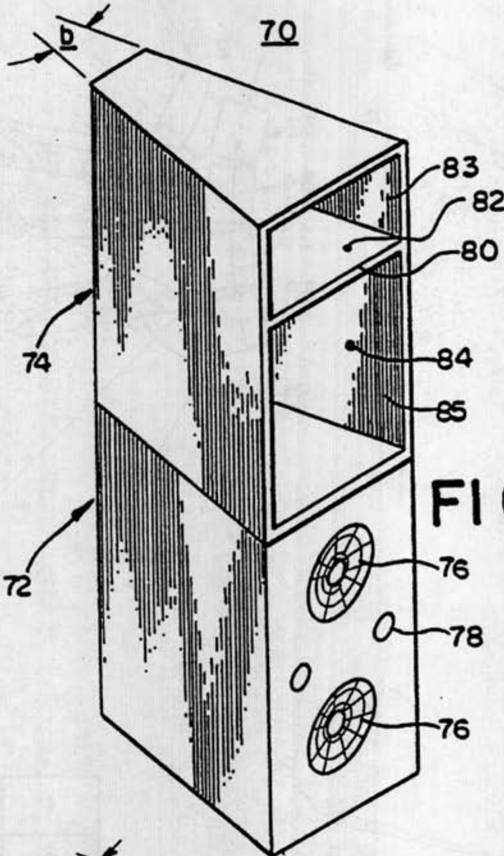


FIG. 5

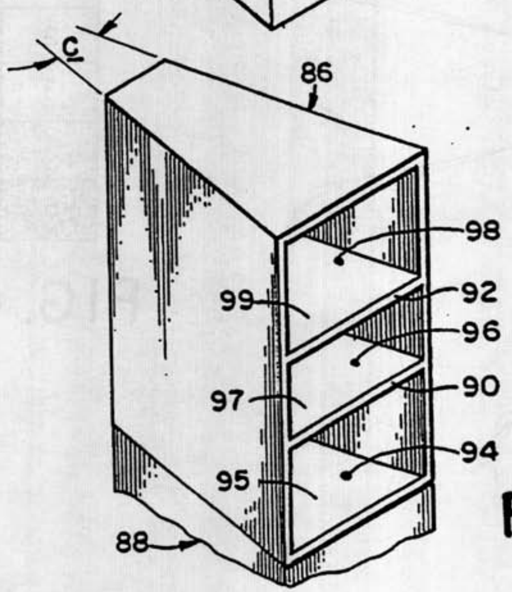


FIG. 6

METHOD FOR LARGE-SCALE MULTIPLE SOURCE SOUND REINFORCEMENT

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates to the field of multiple source high-power sound systems, and in particular, to high fidelity, high intelligibility sound transmission systems for concerts and the like.

2. Prior Art

Audio speaker systems of particularly high power are commonly used for concerts delivered in auditoriums, arenas and amphitheaters, both indoors and outdoors. Typically, individual speakers or "boxes" are stacked or "flown" in a large, closely spaced array in multiples. A typical "wall of sound" system as known in the prior art is shown in FIG. 1, and generally designated by reference numeral 10. A large array of speakers is often referred to as a "concert rig". The concert rig 10 comprises four speaker systems, designated 12, 14, 16 and 18, disposed on a straight line 20. Each speaker system comprises loudspeakers of different sizes and designs, which are appropriate for efficiencies in bass response, mid frequency response and high frequency response, respectively. The four speaker systems interact with one another, creating a plurality of different zones across the listening area or composite zone, wherein sound emitted by each of the speaker systems will remain pure and undistorted, or will be mixed with sounds emitted from one, two or three of the other speaker systems, creating sound confusion and loss of intelligibility. The principal cause of intelligibility loss in the arrival of sound from various sources at different times. Due to path length differences. The different zones are marked and shaded according to the chart shown in FIG. 4. The zones of purest sound are clear, without shading, and identified by the numeral 1. The least intelligible, most distorted sound is shaded the darkest, and identified by numeral 5. Pure sound, that is, sound which is emitted from only one source, and is not mixed with sound emitted from any other source, is transmitted along the outer edges of the zone and immediately in front of each of the speaker systems, as indicated by numeral 1. A number of short zones 2, known as "hi-fi alleys" are formed. The sound in each of the hi-fi alleys emanates from two sources, but inasmuch as the sound contribution from each of the two sources is substantially equal, the overall sound is of generally good quality. Zones designated by reference numeral 3 indicate sound confusion and loss of intelligibility due to unequal contribution from two sources. Zones marked by reference numeral 4 denote more confusion and loss of intelligibility, due to unequal contributions from three sources. The zone designated by reference numeral 5, which will be the single largest zone in the listening area, indicates maximum confusion and interference and maximum loss of intelligibility, due to unequal sound contributions from all four sources. It will be appreciated by those skilled in the art, and indeed by those who attend concerts where such concert rigs are utilized, that more than four speaker systems or sources are often used, for example six to eight sources for covering a wide angle zone of 100 degrees or more. Four sources are illustrated in FIG. 1 in order to reduce the difficulty of illustrating the problems of the prior art without unduly complicating the drawing.

A further difficulty stems from a demand perceived by the those presenting concerts to provide the maximum in sound level, which in turn requires the generation of high sound pressure levels and high dynamic range. Many concert boxes are literally filled with amplifiers and related devices in order to generate as much sound power as possible. Concert rigs such as that shown in FIG. 1 have, unfortunately for those presenting and attending concerts, become synonymous with load sound of inferior quality.

Some improvement has been achieved by a concert rig 30 as shown in FIG. 2, wherein each of the speaker systems or sources 32, 34, 36 and 38 are splayed outwardly from one another, being disposed upon a common arc or shallow curve 40. This arrangement has the effect of modestly increasing the size of the zones 1 of pure sound and the zones of hi-fi alleys 2 of equally mixed sounds. However, it is easily seen that the vast majority of the composite listening zone comprises sound zones designated 3, 4 and 5, which are of noticeably inferior quality. The angle between such splayed sources is typically between ten degrees and twenty degrees, and the system is intended to cover an overall zone of between sixty degrees and ninety degrees. The multiple overlap of sound generated by each of the typical concert rigs 10 and 30 shown in FIGS. 1 and 2 results in a highly unintelligible, acoustically blurred sound quality. Only those listeners who are very close to any one of the individual systems will receive relatively intelligible and undistorted sound. Such listeners are also likely to be deafened by the sound pressure levels.

This invention overcomes the difficulties of the prior art by providing a composite listening zone in which pure, unmixed, undistorted sound is delivered to virtually 100 percent of the listening zone, by emitting sound waves from a plurality of individual electroacoustical sources, each of a constant directivity type and characterized by a relatively narrow, wedge-shaped envelope of sound projection, such that adjacent edges of respective sound projection envelopes are in substantial alignement and do not overlap. The absence of interference between sounds emitted from different ones of the sources precludes sound distortion and enables uniform sound dispersion and high sound quality through the listening zone. Even the extent to which sound from adjacent sources may mix, it mixes equally in hi-fi alleys, a situation which provides at least good sound quality, if not the best sound quality. This can be appreciated by reference to the concert rig 50 shown in FIG. 3, which will be described in detail hereinafter.

SUMMARY OF THE INVENTION

It is an object of this invention to provide large-scale multiple source sound reinforcement delivering high fidelity sound throughout an entire listening zone.

It is another object of this invention to provide large-scale multiple source sound reinforcement delivering high intelligibility sound throughout the listening zone.

It is yet another object of the this invention to provide large-scale multiple source sound reinforcement without interference between any of the multiple sources.

It is yet another object of this invention to provide large-scale multiple source sound reinforcement employing zero-overlap sound projection from each of the sources.

These and other objects of the invention are accomplished by an improved method for transmitting sound at high power levels over a wide angle zone of dispersion without distortion, comprising the steps of: emitting sound waves from a plurality of individual sources, each of a constant directivity type and characterized by a relatively narrow, wedge-shaped envelope of sound projection; and, positioning the plurality of speakers in side by side relationship so that adjacent edges of respective sound projection envelopes are in substantial alignment and do not overlap, whereby the absence of interference between sounds emitted from any of the speakers precludes sound distortion and enables uniform sound dispersion and high sound quality throughout the zone. The method preferably comprises the further step of emitting sound waves from speakers having loudspeaker enclosures shaped to conform to the edges of their respective sound envelopes, such loudspeaker enclosures being thereby substantially trapezoidal in plan. In various preferred embodiments of the invention, the method further comprises the step of configuring each of the sound envelopes to define angles of sound dispersion which are less than or equal to approximately forty degrees, thirty degrees and even twenty degrees.

These and other objects of the invention are also accomplished by a speaker array for transmitting sound at high power levels over a wide angle zone dispersion without distortion, comprising a plurality of individual electroacoustical loudspeaker sources, each of a constant directivity type and characterized by a relatively narrow, laterally wedge-shaped envelope of sound projection, disposed in side by side relationship such that adjacent edges of respective sound projection envelopes are in substantial alignment and do not overlap, whereby the absence of interference between sounds emitted from different ones of the sources precludes sound distortion and enables uniform sound dispersion and high sound quality throughout the zone. In the presently preferred embodiment, the loudspeakers are disposed in a plurality of loudspeaker enclosures, each of the enclosures having a shape which conforms in plan to the edges of the envelope of sound projection generated by the loudspeakers disposed within the enclosure, such loudspeaker enclosures being thereby generally trapezoidal in plan.

BRIEF DESCRIPTION OF THE DRAWNGS

For the purpose of illustrating the invention, there are shown in the drawings forms which are presently preferred; it being understood, however, that the invention is not limited to the precise arrangements and instrumentalities shown.

FIG. 1 is a diagrammatic illustration of the sound interference patterns resulting from a linear wall of sound concert rig, according to the prior art;

FIG. 2 is a diagrammatic illustration of the sound interference patterns resulting from a splayed wall of sound concert rig, according to the prior art;

FIG. 3 is a diagrammatic illustration of the sound dispersion pattern of a concert rig in accordance with this invention;

FIG. 4 is a chart illustrating the scale of shading used for identifying the various degrees of sound purity and sound interference and distortion in the concert rigs of FIGS. 1, 2 and 3;

FIG. 5 is a perspective view of a speaker system source suitable for use in a concert rig as shown in FIG. 3; and,

FIG. 6 is a perspective view of an alternative speaker system source to that shown in FIG. 5.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The operation of concert rig 50 in accordance with the principles of this invention is diagrammatically illustrated in FIG. 3. Sound is transmitted into a composite listening zone by a plurality of sound sources 52, 54, 56, 58 and 60. The composite listening area requires sound dispersion throughout an angle of approximately 180 degrees. Each of the sources 52 through 60 is designed to emit sound waves in a relatively narrow, wedge-shaped envelope of sound projection. In order for the five sources to cover the composite listening zone of 180 degrees, each of the sound sources must define a wedge-shaped envelope of sound projection having a dispersion angle a of approximately 36 degrees. The sounds are emitted radially outwardly from position on a substantially circular arc 62, if the sources are considered point sources or vertical line sources for purposes of illustration. As "real" speakers, the sources are disposed in side by side relationship along the substantially circular arc 62 so that adjacent edges of respective sound projection envelopes are in substantial alignment and do not overlap. With regard to the sound interference level chart shown in FIG. 4, and utilized in the description of the prior art systems of FIG. 1 and FIG. 2, it can be seen that each of the sources 52 through 60 generates a large zone 1 of pure sound, a narrow hi-fi alley 2 being formed between each of the zones 1. Whereas most of the composite listening zones in the prior art systems of FIGS. 1 and 2 consists of zones 3, 4 and 5, which are indicative of substantial interference and poor sound quality, almost all of the composite listening zone of the system shown in FIG. 3 is substantially pure, undistorted sound, the only deviation being the good sound quality of the narrow hi-fi alleys. The absence of interference between sounds emitted from any of the sources precludes sound distortion and enables uniform sound dispersion and high quality throughout the composite listening zone.

It is a further step of this invention to emit sound waves from sources or speakers having loudspeaker enclosures which are shaped to conform to the edges of their respective sound envelopes, referred to as a "minimum envelope" enclosure. Accordingly, each of the sources 52 through 60 is substantially trapezoidal in plan, whereas the loudspeaker enclosures of the sources used in the systems shown in FIGS. 1 and 2 are substantially rectangular. The triangular gaps between the splayed enclosures used in FIG. 2 result in acoustic diffculties on their own account.

A suitable speaker system for forming each of the sound sources 52 through 60 is shown in FIG. 5, and generally designated 70. Each such sound system is divided into two parts, a low frequency or "bass" speaker system 72 covering the frequency range of sound from approximately 35 Hz to 125 Hz and a "mid-high" speaker system 74 covering the frequency range from approximately 125 Hz to 20 KHz. Sounds at a frequency of approximately 20 KHz are at the very upper range of human hearing intelligibility. Separate amplifiers are used for the low, mid and high frequency loudspeaker components and an active crossover/equalizer control network is used to assign the appropriate signals to the appropriate amplifiers. Although the use of separate loudspeakers and crossover/equalizer networks is known in the art, the use of a mid-high speaker system, which essentially covers the entire range of intelligible sound, is a key to the invention's unusual sonic excellence and has never been used in concert rigs before. The mid-high unit operates with unusually small coverage angles, from as much as approximately 40 degrees horizontal dispersion to as little as 20 degrees horizontal dispersion. The angle of vertical dispersion is approximately 20 degrees.

Loudspeakers for the mid-high range speaker **72** must be of a constant directivity type, for example model EV-HP 420, available from Electro-Voice, Inc. of Buchanan, Mich. Inasmuch as the loudspeaker itself down not form a part of the invention, the loudspeaker is not shown in detail. A partition **80** divides the speaker **74** into an upper compartment **82** and a lower compartment **84**. Upper compartment **82** is adapted to receive a high frequency loudspeaker **83**, emitting sound in the range of approximately 1,100 Hz to 20 KHz. Compartment **84** is adapted to receive a mid frequency loudspeaker **85** for emitting sound in the range of approximately 125 Hz to 1,100 Hz. Those skilled in the art will appreciate that the preferred crossover point from bass speaker to mid-high speaker will vary according to circumstances, falling into the range of at least as low as 125 Hz and at least as high as 150 Hz.

The mid-high box **74** is substantially trapezoidal in plan, and defines a dispersion angle b of approximately 30 degrees. The mid-high speaker **74** rests on the bass speaker **72**, which is a vented "bass box" having two bass loudspeakers **76** and two vents **78**, the vents **78** being used to tune the bass box as is known in the art. The bass box **72** is also provided with a minimum envelope enclosure which conforms in plan to the dimensions and shape of the mid-high speaker **74**. Even though the bass emissions are not subject to the same narrow envelopes of sound transmission, the minimum envelope shape contributes to the sound quality. Moreover, the narrow envelopes of sound provided by the special mid-high speakers control almost all of the spectrum of audible sound.

An alternative source is shown in FIG. **6**, wherein a mid-high speaker **86** defines an envelope of sound projection having a dispersion angle c of only 20 degrees. In this embodiment, one mid frequency loudspeaker and two smaller mid frequency loudspeakers are utilized. Partitions **90** and **92** divide the mid-high speaker into compartments **94**, **96** and **98**. Compartments **94** and **98** receive mid frequency loudspeakers **95** and **99** operating in the range of approximately 150 Hz to 1,100 Hz. Compartment **96** receives a high frequency loudspeaker **97**, operating in the range of approximately 1,100 Hz to 20 KHz. A vented bass box **88** is similar to bass box **74** shown in FIG. **5**, except for having a minimum envelope enclosure corresponding to the angle of dispersion c of mid high speaker **86** and having a maximum preferred frequency of 150 Hz.

Extreme wide band directivity control is achieved, and a "house" engineer can so arrange the array of speaker sources that individual areas in an audience are covered with minimum overlap or interference from adjacent sources. Even the transition zones, the hi-fi alleys between sources, are acoustically very well behaved. Total zone control within the composite listening zone is possible for the first time. Both sound level, particularly in the critical intelligibility band of 500 Hz to 2,000 Hz, and frequency response can be tailored for each zone. For example, listeners in distant seats will require high output from a source and more frequency "boost" to perceive the same sound quality and level as listeners closer to the source. Moreover, dramatic improvements in stereophonic "image" can be achieved by "skewing" the directivity patterns of "left" and "right" arrays, each array formed by plural sources as described herein. The stereo image can be made effective to a much larger part of the audience, and in fact, to most of the audience. Such stereophonic imaging is simply impossible to achieve when most of a composite listening zone is filled with overlapping and confused sound dispersion patterns from multiple sources.

The use of a mid-high speaker as described enables the highest possible sound pressure levels to be developed for a given input power. The use of smaller capacity loudspeakers, even in slightly larger numbers, reduces system cost, as less amplification is needed to achieve desired sound levels and quality. In most cases, the system can "coast", particularly in the "vocal" range. The system provides universally clear sound reinforcement with low distortion and wide dynamic range.

Overall, the reinforcement of sound from multiple sources according to this invention is particularly advantageous, in providing: non-overlap zonal coverage; lightweight design; convenient system packing for similarly shaped bass and mid-high speaker boxes; and, the use of vented enclosure technology for the bass boxes. The non-overlap zonal coverage enables listeners to hear sound from only one source, providing the best intelligibility. The high directivity results in minimum reverberation and the design of the horns in the loudspeakers provides maximum efficiency for the lowest distortion and minimum power requirements. Sound quality uniformity in various audience zones can be obtained and the stereo image can be vastly improved throughout a much larger portion of the composite listening zone.

The invention can be embodied in other specific forms without departing from the spirit or essential attributes thereof, and accordingly, reference should be made to the appended claims, rather than to the foregoing specification, as indicating the scope of the invention.

What is claimed is:

1. An improved method for transmitting sound at high power levels over a wide angle zone of dispersion without distortion, comprising the steps of:

emitting sound waves from a plurality of individual sources, each source having a constant acoustic power output at a predetermined angle of divergence and over a predetermined frequency range covering substantially a full audio range of frequencies above approximately 125 to 500 Hz, the sources being of a constant-directivity type and characterized by a relatively narrow, wedge-shaped envelope of sound projection with an angle of sound dispersion not more than approximately forty degrees; and,

positioning the plurality of said individual sources in side-by-side relationship so that adjacent edges of respective sound projection envelopes are in substantial alignment and do not overlap, whereby the absence of interference between sounds emitted from any of the sources preclude sound distortion and enables sound dispersion and high sound quality throughout the zone.

2. The method of claim 1, wherein each said source has sections for distinct frequency ranges, the sections likewise emitting sound at constant-directivity over said predetermined angle of divergence.

3. The method of claim 2, wherein sound is emitted from a mid frequency source and a high frequency source, and further comprising the step of dividing signals into said mid frequency source and high frequency source using a cross-over network.

4. The method of claim 1, wherein said individual sources are electroacoustical loudspeakers having loudspeaker enclosures shaped to conform to the edges of their respective sound envelopes.

5. The method of claim 4, comprising the further step of forming each loudspeaker enclosure to be substantially trapezoidal in plan.

6. The method of claim 1, further comprising the step of emitting the sounds radially outwardly from positions on a substantially circular arc.

7. The method of claim 1, comprising the step of configuring each of the sound envelopes to define an angle of sound dispersion which is not more than approximately thirty degrees.

8. The method of claim 3, comprising the step of configuring each of the sound envelopes to define an angle of sound dispersion which is not more than approximately twenty degrees.

9. The method of claim 1, further comprising the steps of emitting the sound waves from two arrays of the plurality of sources and skewing the sounds emitted from the arrays to achieve stereophonic imaging throughout a large portion of the dispersion zone.

10. An improved method for transmitting sound at high power levels over a wide angle zone of dispersion without distortion, comprising the steps of emitting sound waves from a plurality of individual constant-directivity sources, each source having a constant acoustic power output at a predetermined angle of divergence and over a predetermined frequency range covering substantially a full audio frequency range above approximately 125 to 500 Hz, the individual sources being characterized by a relatively narrow, wedge-shaped envelope of sound projection with an angle of sound dispersion not more than approximately forty degrees, such that adjacent edges of respective sound projection envelopes are in substantial alignment and do not overlap, whereby the absence of interference between sounds emitted from different sources preclude sound distortion and enables uniform sound dispersion and high sound quality throughout the zone.

11. The method of claim 10, comprising the step of configuring each of the sound envelopes to define an angle of sound dispersion which is not more than approximately thirty degrees.

12. The method of claim 11, comprising the step of configuring each of the sound envelopes to define an angle of sound dispersion which is not more than approximately twenty degrees.

13. The method of claim 10, wherein the sources are electroacoustical loudspeakers having loudspeaker enclosures shaped to conform to the edges of their respective sound envelopes.

14. The method of claim 13, comprising the further step of forming each loudspeaker enclosure to be substantially trapezoidal in plan.

15. The method of claim 10, further comprising the step of emitting the sounds radially outwardly from positions on a substantially circular arc.

16. The method of claim 10, further comprising the steps of emitting the sound waves from two arrays of the plurality of sources and skewing the sounds emitted from the arrays to achieve stereophonic imaging throughout a large portion of the dispersion zone.

17. The method of claim 10, wherein each said source has sections for distinct frequency ranges, the sections likewise emitting sound at constant-directivity over said predetermined angle of divergence.

18. The method of claim 17, wherein sound is emitted from a mid frequency source and a high frequency source, and further comprising the step of dividing signals into said mid frequency source and high frequency source using a cross-over network.

* * * * *

RH3-250, no. in array = 3 , frequency = 250 Hz. Vertical
One third octave pink noise.
On-axis SPL relative to source no. 1 alone = 8.49 dB.
?

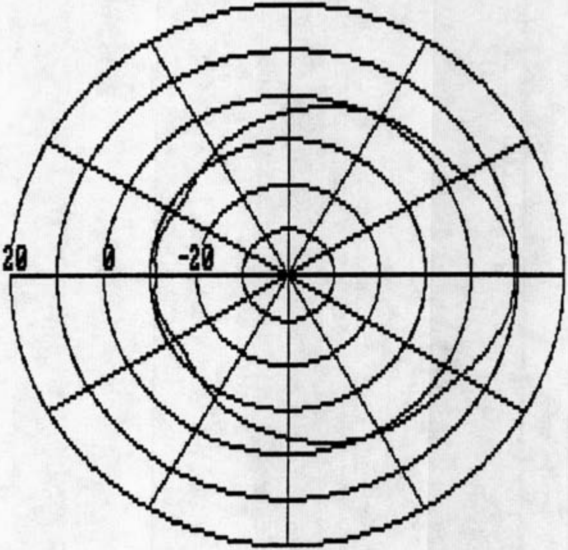


Figure 15h. Low frequency line array vertical polar prediction at 250 Hz.

RH3-315, no. in array = 3 , frequency = 315 Hz. Vertical
One third octave pink noise.
On-axis SPL relative to source no. 1 alone = 6.76 dB.
?

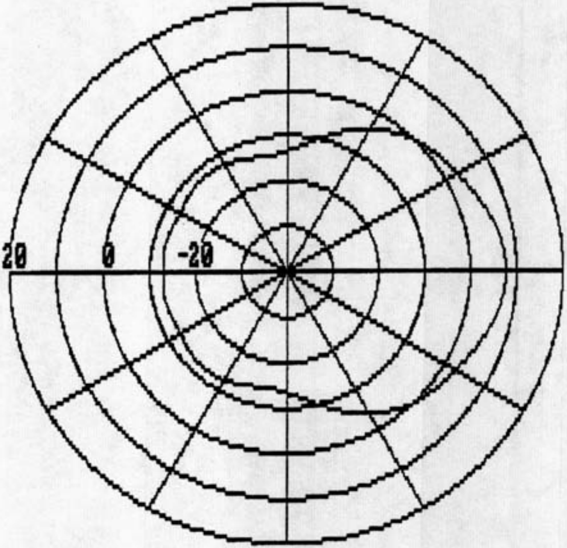


Figure 15i. Low frequency line array vertical polar prediction at 315 Hz.

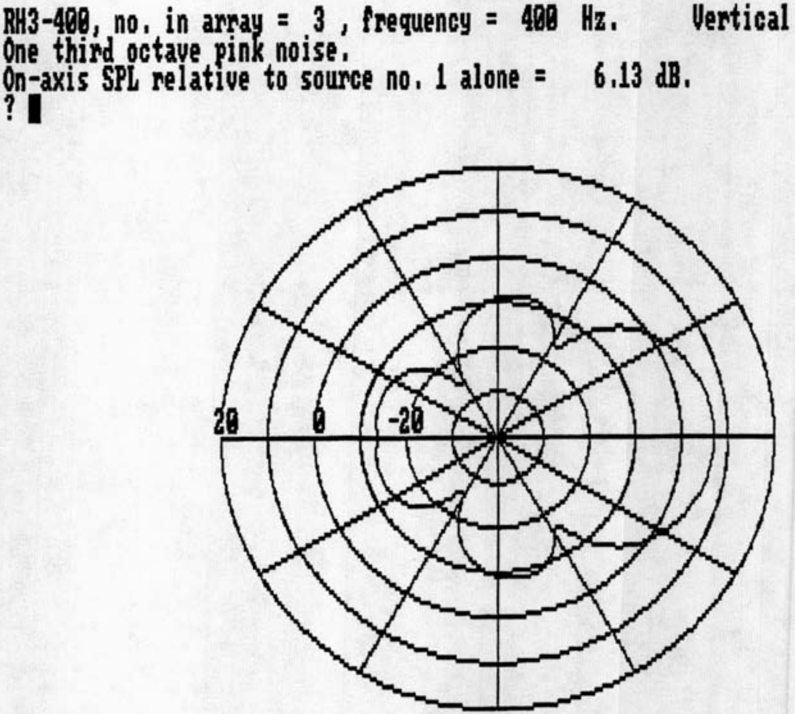


Figure 15j. Low frequency line array vertical polar prediction at 400 Hz.

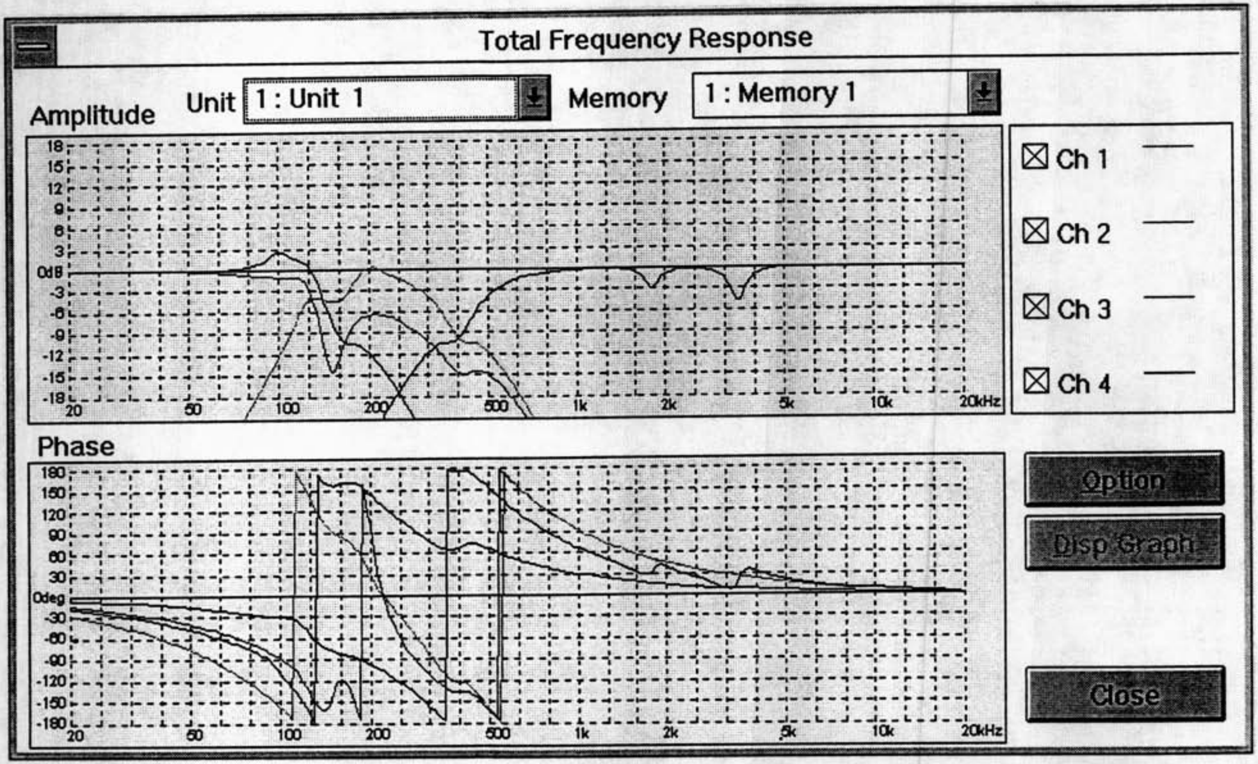


Figure 16. Low frequency line array and mid/high signal processing showing amplitude and phase for the research prototype. In order to maintain close phase response between low frequency drivers, the use of all pass filtering is required in addition to parametric and crossover filtering. To achieve proper power matching, level adjustments to the individual driver amplifiers is required.

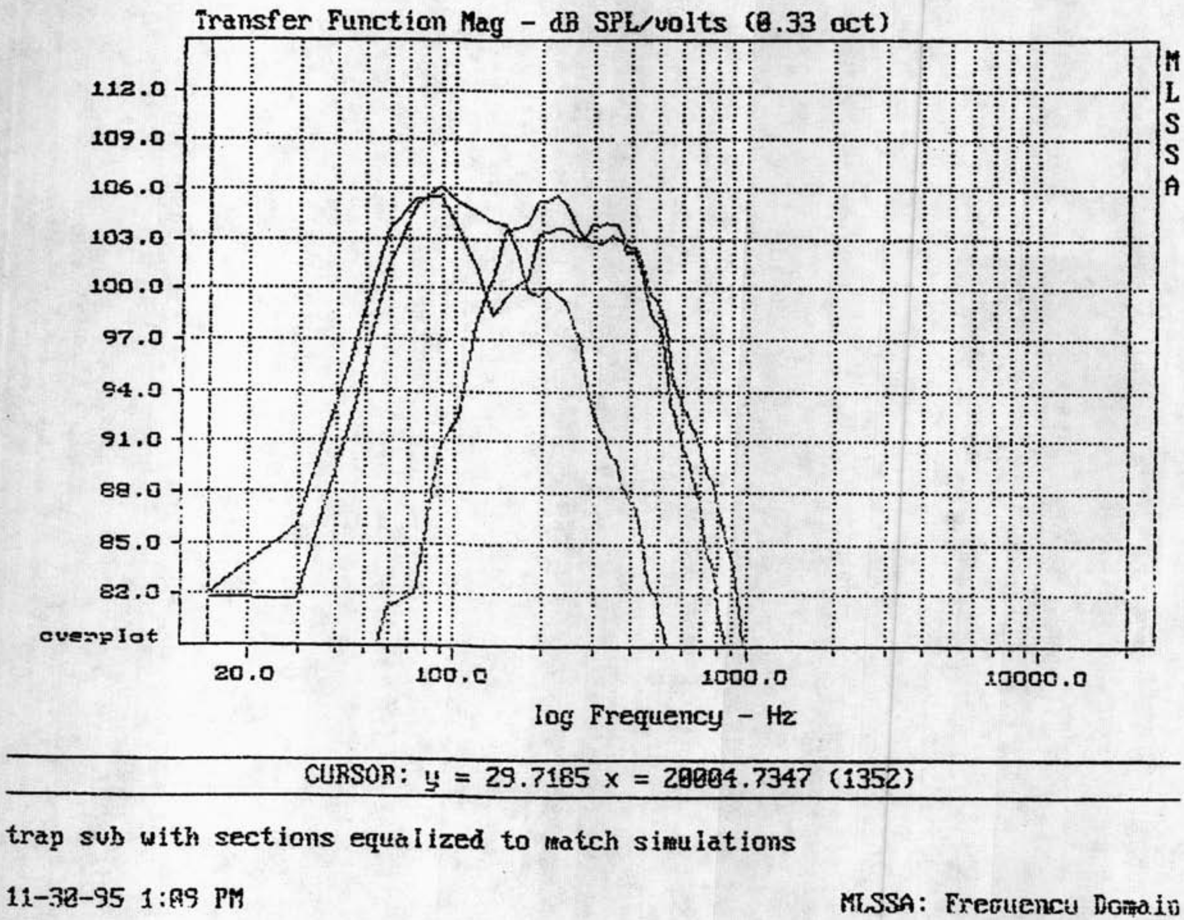


Figure 17. Measured individual low frequency line array drivers illustrating the frequency response for each device.