CALREC

Sound Field System

calrec by ams
P.O. Box 31864 • Seattle, WA 98103-1864
E-Mail: 42:IMC 889
(206) 633-1956 • Telex: 4900001180 (CALUI)

CALREC SOUNDFIELD MICROPHONE MARK 4.

CIRCUITS DESCRIPTION, TEST AND ALIGNMENT.

CONTENTS.

CIRCUIT DIAGRAMS:

SOUNDFIELD MK. 4 CONTROL UNIT	:	SF5020-3-7
SOUNDFIELD MK. 4 MATRIX MODULE	:	SF5020-3-5
SOUNDFIELD MK. 4 MICROPHONE	:	SK4050-56 (62, 63)

GENERAL ARRANGEMENT:

SOUNDFIELD MK. 4 MICROPHONE- GENERAL ARRANGEMENT	:	SF4050-64
SOUNDFIELD MICROPHONE CONTROL UNIT MK. 4 FRONT PANEL	:	SF5020-2-10
NORMAL CABLE CONNECTION CABLE CONNECTION VIA STUDIO LINES	 :	SK4050-59 SF4050-65

1. POWER AND INDICATORS:

The Soundfield Mk. 4 Control Unit may be operated from 200/250 or 100/120 volt A.C. Mains. There is a voltage range selector on the rear panel which should be set prior to connection. This unit has a mains filter incorporated and, therefore, must have a mains earth connected.

The system will operate down to 180 (90) volts A.C. (with slightly reduced microphone voltage) and the lower voltage figures below are taken at this input.

Transformer T1 produces 15/20-0-15/20 volts A.C. which is full-wave rectified to ± 18/24 volts across C65, C66. Two stabilisers (1 & 2) control the main power rails to the Control Unit, matrix card and metering card at ±15.6 volts (C67,68). (Test points 8, 9, 10 and 11).

A green L.E.D. (POWER, LD4) is illuminated from the + rail.

Reduced voltage rails of ± 7.5 volts for CMOS analogue switch circuits are provided by zener diodes D23, 24 (C82, 83). (Test points 12 and 13).

A voltage tripler circuit is connected in series with the + supply source (D18, C63 and D19, C64) to produce a voltage of +56/80 volts at C64 (+plate/ground). This is controlled and stabilised by TR1, TR2 and IC21 to +53 volts (C69) for the microphone amplifiers supply and capsule polarising. (Test points 2 and 1).

Two other L.E.D. indicators : SOUNDFIELD IN (yellow LD2) and SOUNDFIELD TO REC (red LD3) are switched from the +15.6 volts rail.

The MUTES and OSCILLATOR TEST L.E.D. indicator (red LD1) is switched on by any of 4 CAPSULE MUTE selections or the OSCILLATOR TEST button. Any MUTE selection causes the L.E.D. to flash; the OSCILLATOR TEST button switches the indicator on continuously.

The flashing sequence is provided by a bi-stable, IC1a, driving an invertor, IC1b. The L.E.D. is connected to IC1a via a diode (D5) which may be over-ridden by a Transistor TR3 connected directly to the OSC. button. The flashing sequence is initiated by any MUTE button or CMOS analogue switch IC19C opening a series circuit in the input of IC1a. IC19C (also 19B, 19A, 20A, 20B, 20C see below) is "opened" by another pole on the OSC button which releases the control circuits (to logic 1) on IC19 and IC20 and also mutes any capsule signals not already selected to mute. (Test points 4 and 5).

The bi-stable thus produces a square-wave output with a period of about ¾ second on MUTE and on OSC. TEST (even though the L.E.D. remains steady in the latter condition). This output is used to pulse the oscillator tone outputs in a special way to identify the 4 B-format components, fully described in Section 2.

IC19B releases a short-circuit on the oscillator IC, A2b allowing it to start.

After a short period to allow the oscillator to settle, decided by R192 and C86, IC 19A - 20C - 20B and 20A CMOS analogue switches change-over the inputs of the record amplifiers to the oscillator input IC, A2a.

The control signals to IC19A, 20B and 20A (X, Y and Z) are over-ridden in a specially coded manner in the oscillator pulse circuit described in Section 2. This coded switching returns the record amplifier inputs to the microphone circuits for a few milliseconds at a time but no signal is heard in the intervals because of the overall capsule mute switching by the OSC. TEST button.

2. ALIGNMENT OSCILLATOR.

The oscillator A26 starts when "released" by IC19B. The oscillator operates at a frequency of approximately 1KHz and produces an output of about + 7 dBu at A2b output. A2a provides a drive output suitable for the record amplifiers; since these amplify +15B and the tone output is required to be OdBu, A2a is set by RV9 to -15Bu by observing OdBu at B-FORMAT RECORD XLR5-M socket. TH2 provides temperature compensation to the output level by working in an opposite fashion to TH1 over a normal ambient range. (Test points 6 and 7)

It is fundamentally important that B-Format signals are properly identified to avoid connection errors. It is important that such identification can be quickly and simultaneously recognised aurally or visually on metering.

For this reason the W signal receives continuous tone; the X tone is interrupted by the negative-going output of IC22 for a period decided by C91 and R207 which switches IC19A away from the tone via D7. This period is chosen to allow a peak programme meter to fall at least 4dB before the tone returns. 4dB is sufficient to ensure that a bargraph with 3dB steps and a long decay time will extinguish at least one segment. (Test point 14).
Simultaneous with the negative-going pulse from IC22, a positive-going pulse from IC1 (R201, C89, D12) to the "clock" input of J-K Flip-Flop IC18a produces a change in levels at the outputs (Q, $\bar{Q}$). The next change is after 1 cycle from IC1 which represents a half-cycle at IC18a output. Thus a further integration of the negative pulses of this output (R204, C90, D9) to IC20b causes a similar interrupt period to the Y tone but at half the frequency of X. (Test point 15).

By repeating this process from IC18a ($\bar{Q}$) output to IC18b "clock" input (R199, C87, D10) and a third integration circuit (R203, C88, D11) a similar occult period is applied to the Z tone but at half the frequency of Y i.e. one quarter the frequency of X. (See Fig. 4). (Test point 16).

The interruption periods, when they occur together, are exactly in-phase. After only 1 or 2 seconds the signals can be visually recognised on the bargraph meters and tape machine meters, or aurally if heard individually.

*Moreover, some fascinating matrix tone patterns are produced at the Stereo outputs especially if the Soundfield controls are introduced. It is strongly recommended that the engineer/user firstly acquaints himself with the sound of the individual tone sequences and then listens to a stereo output whilst operating the Polar Pattern and Angle controls.

Remembering that Omni is wholly W, Figure-of-eight, Angle 0° wholly X and Figure-of-eight, Angle 180° wholly Y some very fast checks can be made.

The results become complex but recognisable at intermediate settings especially when Z is introduced by operating the Elevation control. With practice, the Azimuth control can be checked in all 4 sectors as can the Dominance control.

*If the Control Unit is being aligned for the first time or if the engineer/user is unfamiliar with the Soundfield microphone, it may prove better to return to these sophisticated experiments following alignment and/or familiarity.

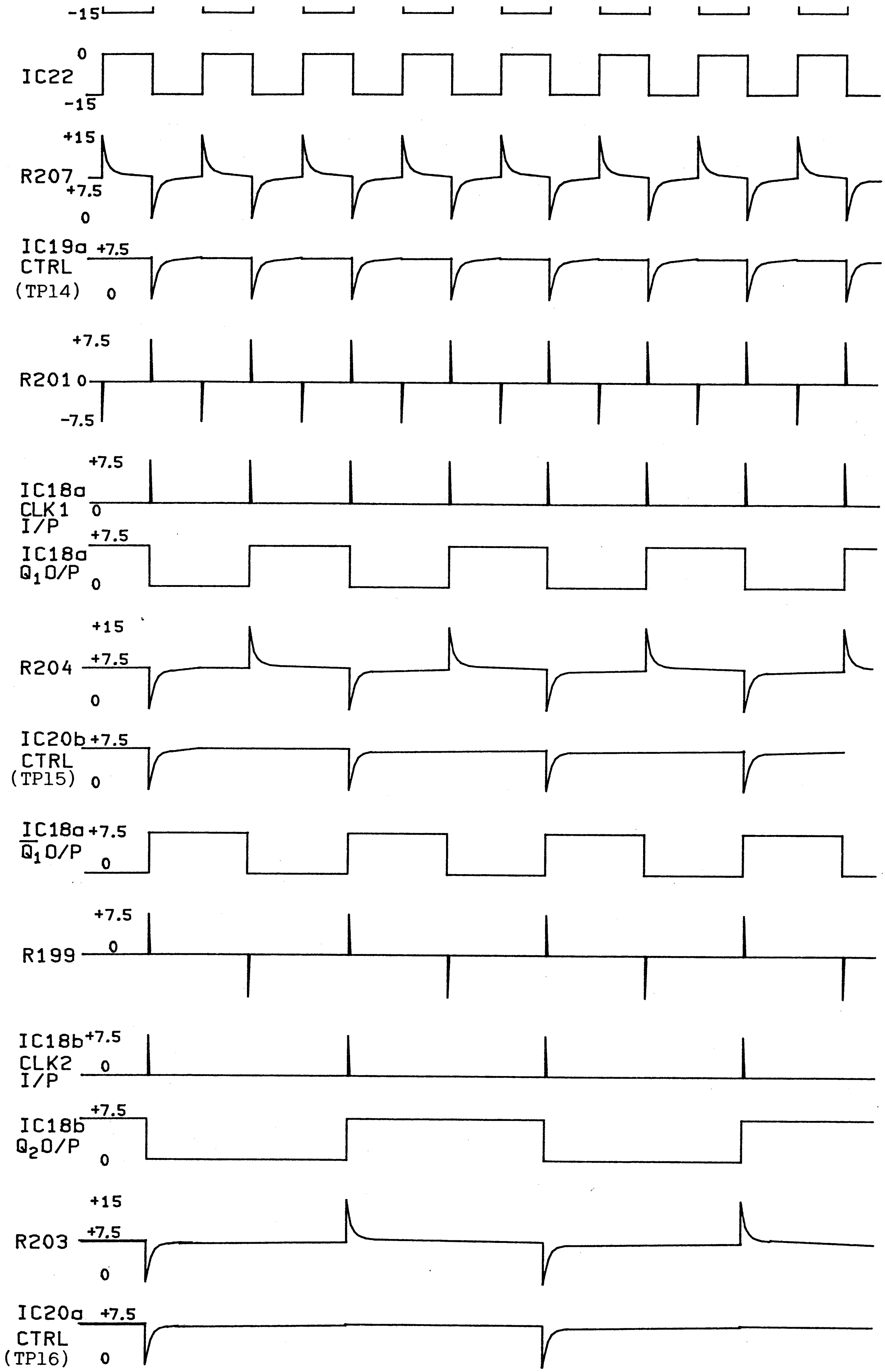


FIG 4: ALIGNMENT OSCILLATOR SEQUENCES

SF P2/2

3. METERING.

This consists of 4 coloured bar-graph type indicators.

They are connected to the B-Format REPLAY amplifiers so that Record Level is metered in the normal mode, post Sound-Field if so selected, or Replay Level when the button TAPE is pressed.

The bar-graphs are calibrated to illuminate 5 green segments at OdBu, line-up tone and the panel is marked at this position. The calibration is such that the alignment tone only just illuminates segment 5 so that if a tape machine is being aligned prior to recording, the recorder replay gain should be set so that segment 5 just turns on.

Segments 8, 9 and 10 are red so as to indicate overload: segment 8 will just illuminate at +8dBu representing peak programme.

The indicators have a very approximate "Peak programme meter" attack and decay times but are not intended as a serious replacement for superior programme metering.

Each of the 4 circuits consist of an amplifier (IC 23 & 24) and rectifier and law-shaping components. Pre-set level controls RV 10, 11, 12, 13 should be set at OdBu at the B-FORMAT RECORD output XLR5M socket so that segment 5 illuminates and extinguishes at -0.3dBu.
(Test points 18, 19, 20 and 21).

4. RECORD AMPLIFIERS.

These are A4a, A4b, A5a and A5b and each has a nominal gain of +15dB. They should be tested at full RECORD LEVEL gain. (OdB attenuation).

Signals should be injected at pins 2, 3, 4 and 5 at the Matrix Card DIN 32 connector and read at the B-FORMAT RECORD XLR5M socket with the OSCILLATOR off, and DUB, SOUNDFIELD REC not selected.

The signals are amplified 15dB, in-phase unless INVERT is selected whan Y and Z are inverted as required for a microphone hung upside down.

Note that when the 4 RECORD outputs are set for OdBu, the bar graph meters are illuminated up to and including sector 5. A reduction of -0.3dB should extinguish sector 5.

The RECORD LEVEL control should be set at the marked "O" level for -10dBu outputs (previously set at full level for OdBu) by adjusting RV1, RV2, RV3 and RV4 for W, X, Y and Z respectively.

The control should be checked for good tracking on all 4 signals over the marked "working sector": -15/+5dB relative to "O". Maximum error between any 2 signals should be 0.5dB.

It is necessary at this stage to check the first 3 positions of the GAIN switch: O, 5 and 10. At these positions the capsules polarising voltage is reduced to give capsule attenuation of -15, -10 and -5dB respectively.

Check that capsule voltage holds steady at +50V at all other positions and that it reduces as below:-

Pos. 10 : 32.5 V
Pos. 5 : 20.5 V
Pos 0 : 13 V

Check at Pin 19 Matrix Sub Card Skt.

5. REPLAY AMPLIFIERS.

These are A6a, A6b, A7a, A7b and each is a simple voltage follower. The amplifiers present a constant input impedance of 10K-ohms to the B-FORMAT PLAY signals from the tape recorder. They provide a low source resistance to several alternative paths for the B-Format signals so as to eliminate gain changes due to switching circuits in/out, adjusting controls etc.

They are quickly checked by connecting a OdBu, 1KHz signal to the B-FORMAT PLAY XLR5F socket to each signal pin in turn and observing a corresponding output at the B-FORMAT RECORD XLR5M socket with the DUB button depressed and the RECORD LEVEL control at mark 0 i.e. Line-up.

If the box is being aligned for the first time, set RV5, RV6, RV7 and RV8 for OdBu output at X, W, Y and Z pins respectively.

6, SOUNDFIELD CONTROLS.

Azimuth, Elevation and Dominance are effected by adding or subtracting appropriate other B-Format components to each basic B-Format signal either in-phase or inverted to achieve a desired result.

For example if it is required to effectively move the "front" of the microphone 45° to the left then the "new X" or X', must now consist of equal components of X and Y but so that the amplitude of X' is identical to that of X, the X and Y constituents must each be reduced by a factor of 2 or -3dB. (Fig 1).

Similarly 45° to the right requires the same component of X i.e. $\frac{X}{\sqrt{2}}$

but now the added Y component requires to be $\frac{-Y}{\sqrt{2}}$

Thus the Azimuth control in the Front sector merely reduces X by 3dB at each end relative to centre, the signals remaining in-phase, whereas the ADDED component, $\frac{Y}{\sqrt{2}}$ (-3dB) at left, disappears at centre and re-appears as $\frac{-Y}{\sqrt{2}}$ (-3dB) at right i.e. INVERTED.

This is the key to understanding and testing all Soundfield circuits: The BASIC signal component is varied in amplitude from its 0dB neutral position whilst remaining IN-PHASE (it actually INCREASES in the Dominance circuit) whilst the ADDED signal component appears in opposite phase on either hand from a zero value at neutral. The maximum value of the ADDED signal at the ends of the controls is always $\frac{1}{\sqrt{2}}$ or -3dB.

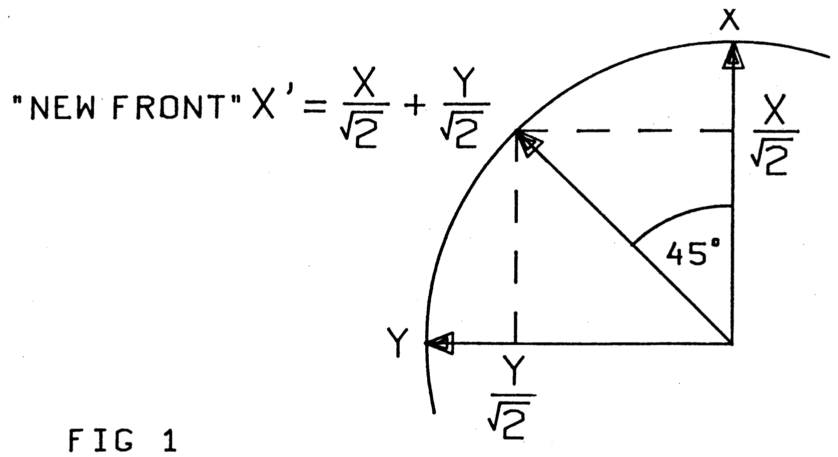


FIG 1

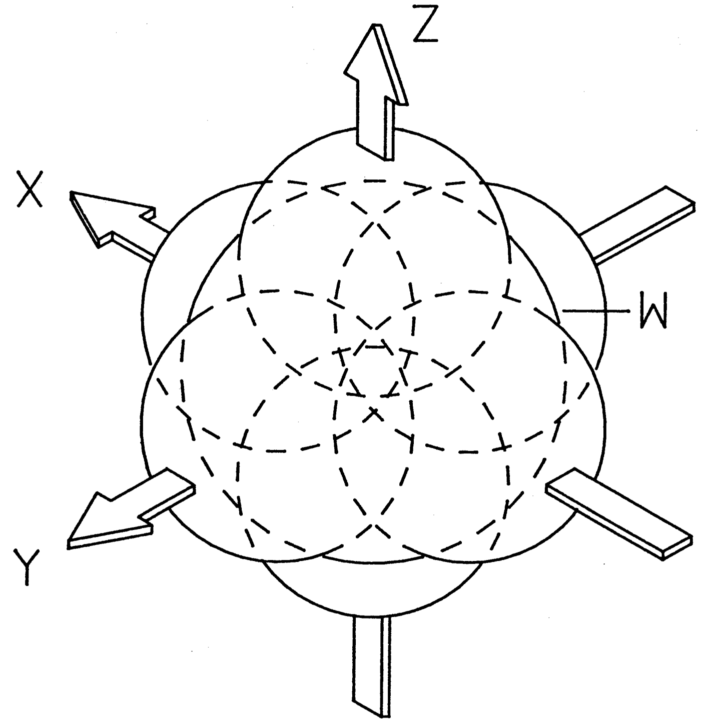


FIG 2

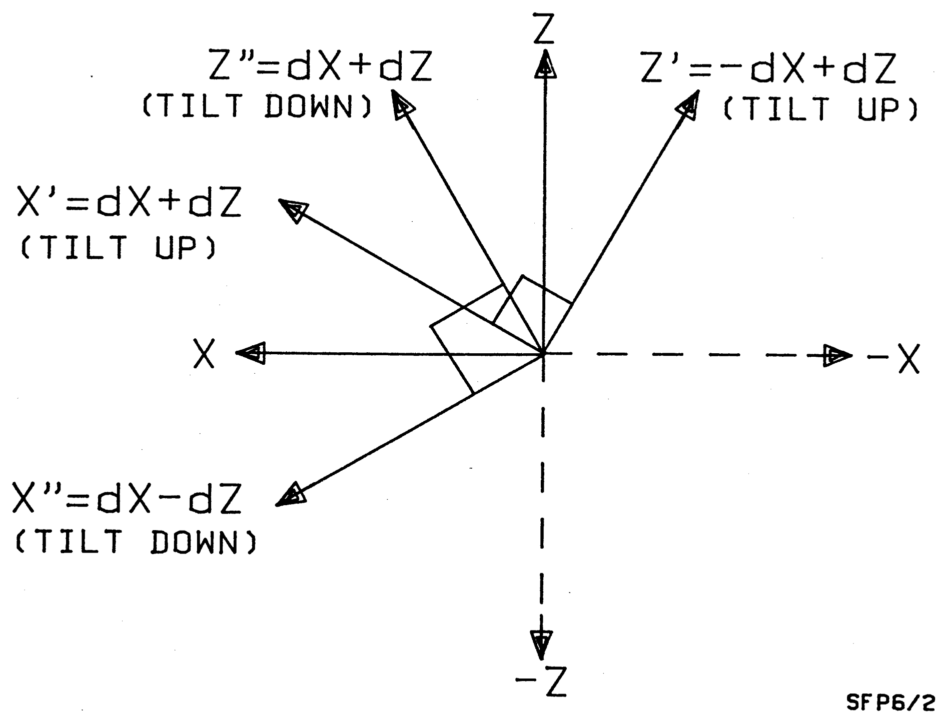


FIG 3

SFP6/2

Once this principle is understood, the required PHASE of the added signal to achieve a desired result can easily be determined by reference to the B-Format signal vectors (Fig. 2). For example if the microphone requires to be tilted up, (Elevation control) X' must now contain some +Z and Z' some -X, or tilted down, X' must contain some -Z and Z' some +X and so on (Fig. 3). Each of the 3 B-Format directional signals has a "positive hemi-sphere" in which its vector lies, and an opposite "negative hemi-sphere".

Dominance is slightly more complicated (but only slightly). This control breaks the basic symmetry of the soundfield sphere and causes it to "bulge" forwards or backwards. This is achieved by adding or subtracting some X to the W component to make it into a front or back facing SUB-CARDIOID* and at the same time adding or subtracting W to the X component to form a front or back facing HYPER-CARDIOID*.

*It must be remembered that equal quantities of W (pressure, omni-directional) and X (pressure-gradient, figure-of-eight) components form a FRONT FACING CARDIOID. If X is subtracted (inverted), this is a BACK FACING CARDIOID. If $W > X$ then the polar pattern is said to be a SUB-CARDIOID; if $X > W$, a HYPER-CARDIOID. There are a range of SUB and HYPER-CARDIOIDS but only one CARDIOID. A fuller explanation of this appears in Section 7.

An interesting feature of the Dominance circuit is that to achieve the required patterns described above, two circuits firstly produce two cardioids, one facing front $(W + \frac{X}{\sqrt{2}})$ and one facing back $(W - \frac{X}{\sqrt{2}})$, (remember X is normally $\sqrt{2}$ the amplitude of W). The Dominance control now simply increases one whilst reducing the other or vica versa and W' and X' are rematrixed from these by appropriate addition and subtraction.

Test and Alignment of Soundfield Circuits.

Note that normal B-Format signal levels from the microphone increment the pressure-gradient components X, Y and Z by +3dB relative to W. However, in alignment in this section all signals are relative to a OdBu level.

The Soundfield Section is best tested by injecting signals at the B-FORMAT PLAY XLR5F socket and taking readings from the B-FORMAT RECORD XLR5M socket. Press the DUB button; set a line-up level on the RECORD LEVEL control for all 4 signals. The metering will also register output levels if <u>not</u> selected to TAPE. Press the SOUNDFIELD IN and REC buttons.

If the Soundfield section is working correctly then reading as UNDER-LINED should result and the intermediate tests can be ignored.

Set Azimuth, Elevation and Dominance initially to neutral.

AZIMUTH.

Ensure there are <u>no</u> Z or W inputs.
A15b (C44) = $X_1 + Y_1$.
(Test point 26)

FRONT

$$X_1 + Y_1 = \frac{X}{2} + \frac{Y}{2}$$

LEFT

$$X_1 + Y_1 = -\frac{X}{2} + \frac{Y}{2}$$

RIGHT

$$X_1 + Y_1 = \frac{X}{2} - \frac{Y}{2}$$

BACK

$$X_1 + Y_1 = -\frac{X}{2} - \frac{Y}{2}$$

N.B. $\frac{X}{2}, \frac{Y}{2}$ = -6dB; MINUS SIGN = INVERTED SIGNAL.

A15a (C46) = X2 + Y2.

FRONT

$$X_2 + Y_2 = \frac{X}{2} - \frac{Y}{2}$$

LEFT

$$X_2 + Y_2 = \frac{X}{2} + \frac{Y}{2}$$

RIGHT

$$X_2 + Y_2 = -\frac{X}{2} - \frac{Y}{2}$$

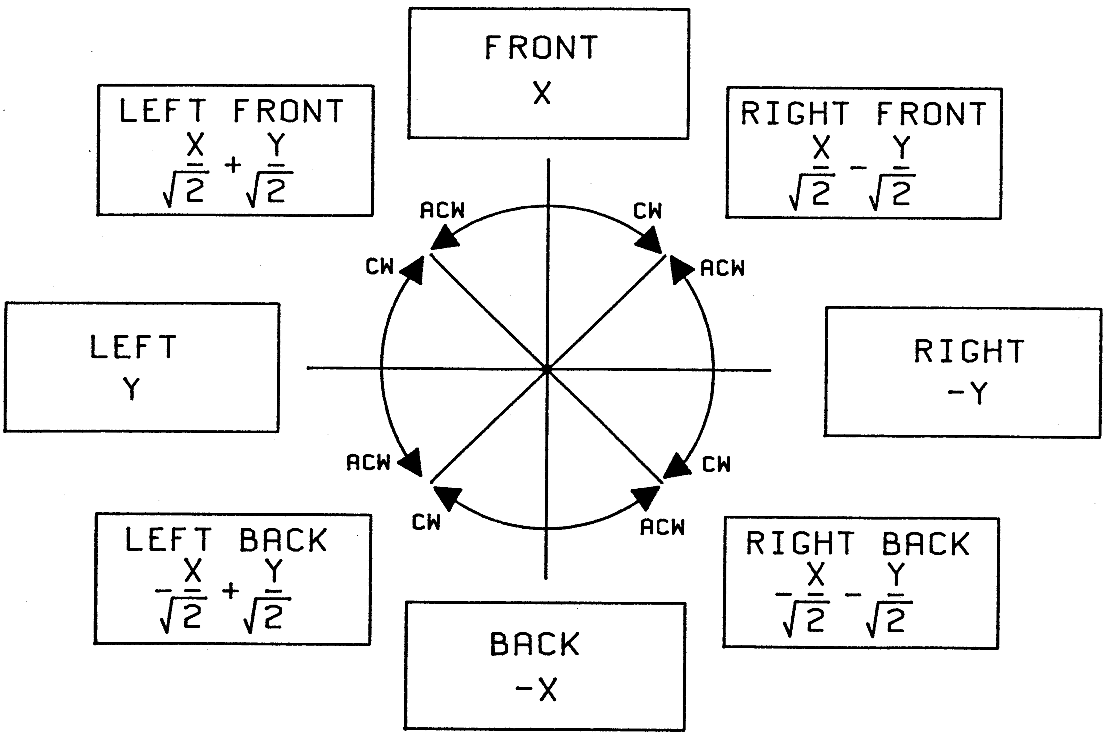
BACK

$$X_2 + Y_2 = -\frac{X}{2} + \frac{Y}{2}$$

A12b (C52) = X5 (X OUTPUT FROM AZIMUTH CIRCUIT) (Test point 30).
AT NEUTRAL AZIMUTH, X5 = (X_1 + Y_1) + (X2 + Y2).

AT ACW (Full anti-clockwise) X5 = 2 (X_1 + Y_1)
AT CW (Full clockwise) X5 = 2 (X2 + Y2)

N.B. This is only true for Elevation at neutral and no Z input assuming that all is well around A12b. If not, then the following readings must be taken as X3 at A17a (C49): THE RESULTS WILL ALL BE -3dB and INVERTED REL. TO A12b (BELOW).

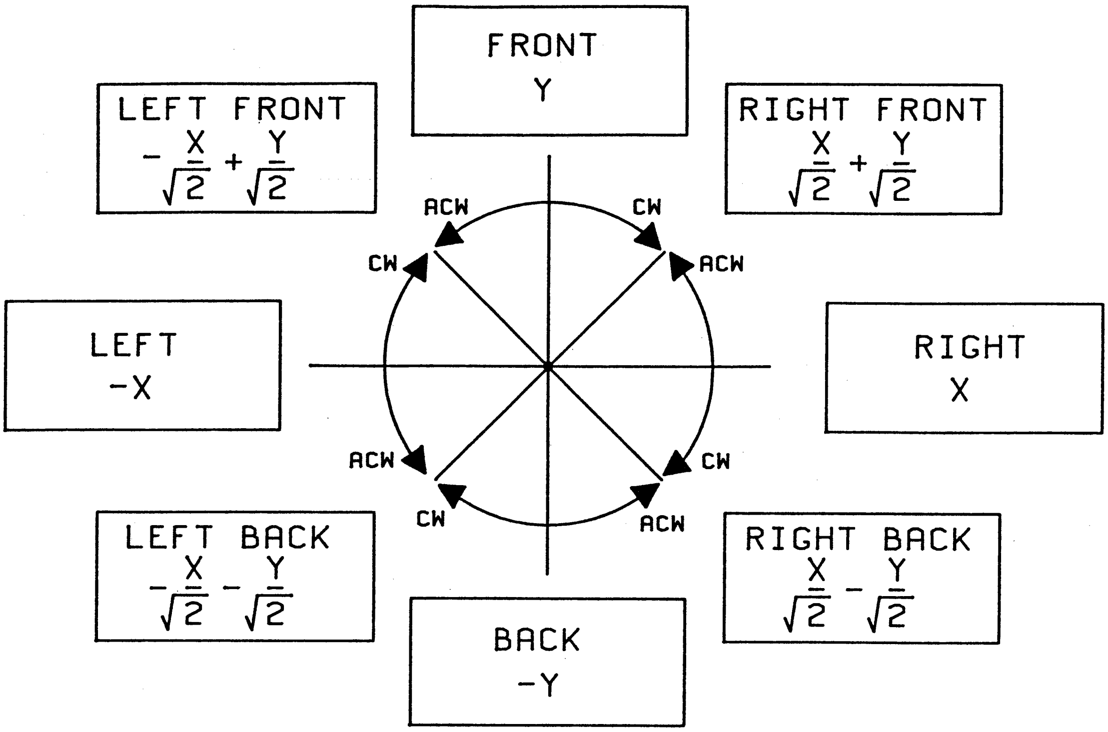


X OUTPUT FROM AZIMUTH.

N.B. $\frac{X}{\sqrt{2}}$, $\frac{Y}{\sqrt{2}}$ = -3dB; MINUS SIGN = INVERTED SIGNAL (AND SOUNDFIELD IF ELEVATION AND DOMINANCE ARE NEUTRAL & NO FAULTS.

A16a (C60) = Y3 (Y OUTPUT FROM AZIMUTH AND SOUNDFIELD CIRCUITS).

AT NEUTRAL AZIMUTH, $Y3 = (X_1 + Y_1) - (X2 + Y2)$
AT ACW $Y3 = -\sqrt{2}(X2 + Y2)$
AT CW $Y3 = \sqrt{2}\,(X_1 + Y_1)$



Y OUTPUT FROM AZIMUTH AND SOUNDFIELD.

N.B. Note how in each sector the principal signal remains in-phase and reduces -3dB on either hand whereas the ADDED signal, being zero at neutral, appears at -3dB in (appropriate) OPPOSITE phase at each end of the control.

ELEVATION.

Ensure that AZIMUTH is at neutral, FRONT.

A17a (C49) = $X3 + Z_1$ (Test point 28)

$X3 + Z_1 = -\frac{X}{\sqrt{2}} - \frac{Z}{\sqrt{2}}$

A17b (C70) = X4 + Z2 (Test point 29)

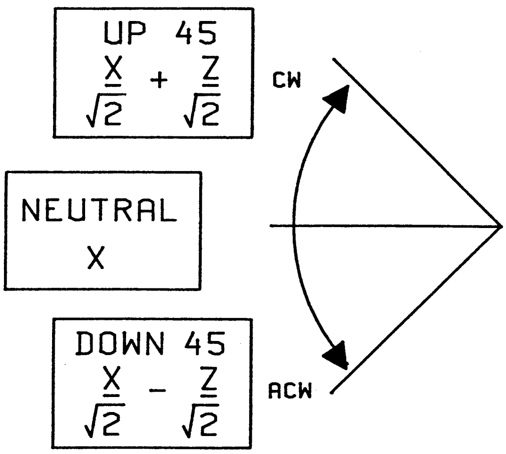
$X4 + Z2 = \frac{X}{\sqrt{2}} - \frac{Z}{\sqrt{2}}$

A12b (C52) = X5 (X OUTPUT FROM ELEVATION CIRCUIT) (Test point 30)

AT NEUTRAL, $X5 = \frac{-(X3 + Z_1) + (X4 + Z2)}{\sqrt{2}}$

AT ACW (DOWN 45), X5 = X4 + Z2

AT CW (UP 45), $X5 = -(X3 + Z_1)$



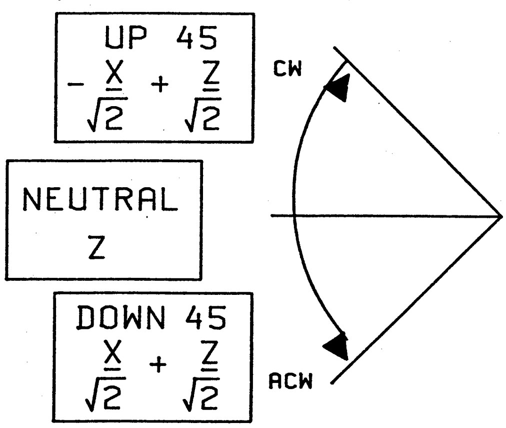
X OUTPUT FROM ELEVATION (& SOUNDFIELD IF DOMINANCE IS AT NEUTRAL AND NO FAULTS.

A16b (C62) = Z3 (Z OUTPUT FROM ELEVATION AND SOUNDFIELD CIRCUITS)

AT NEUTRAL, $Z3 = \frac{-(X3 + Z_1) - (X4 + Z2)}{\sqrt{2}}$

AT ACW (DOWN 45), $Z3 = -(X3 + Z_1)$

AT CW (UP 45), $Z3 = -(X4 + Z2)$



Z OUTPUT FROM ELEVATION AND SOUNDFIELD.

N.B. $\frac{X}{\sqrt{2}}, \frac{Z}{\sqrt{2}}$ = -3dB; MINUS SIGN = INVERTED SIGNAL.

DOMINANCE.

Ensure that AZIMUTH and ELEVATION are at neutral and FRONT ($X5 = X$).

$$A12a\ (54) = W_{,} + X6 \text{ (Test point 31)}$$
$$= (2 - \sqrt{2})W - (\sqrt{2} - 1)X$$

$\boxed{\begin{array}{l} W\ -4.65dB\ \text{IN-PHASE} \\ X\ -7.65dB\ \text{INVERTED} \end{array}}$

Since X is normally enhanced $\sqrt{2}$ (+3dB), this is a BACK FACING CARDIOID of Max. level (0°) $2(2 - \sqrt{2})$.

$$A13b\ O/P = W2 + X7 \text{ (Test point 32)}$$

AT NEUTRAL DOMINANCE, $W2 + X7 = -(W_{,} + X6)$
$$= -(2 - \sqrt{2})W + (\sqrt{2} - 1)X$$

$\boxed{\begin{array}{l} W\ -4.65dB\ \text{INVERTED} \\ X\ -7.65dB\ \text{IN-PHASE} \end{array}}$

AT ACW, BACK DOMINANCE, $W2 + X7 = -(1 + \sqrt{2})\ (W_{,} + X6)$
$$= -\sqrt{2}W + X$$

$\boxed{\begin{array}{l} W\ +3dB,\ \text{INVERTED} \\ X\ 0dB,\ \text{IN-PHASE} \end{array}}$

AT CW, FRONT DOMINANCE, $W2 + X7 = -\dfrac{1}{(1 + \sqrt{2})}\ (W_{,} + X6)$
$$= -\left(\frac{2 - \sqrt{2}}{1 + \sqrt{2}}\right)W + \left(\frac{\sqrt{2} - 1}{1 + \sqrt{2}}\right)X$$

$\boxed{\begin{array}{l} W\ -12.3dB,\ \text{INVERTED} \\ X\ -15.3dB,\ \text{IN-PHASE} \end{array}}$

i.e. The action of the Dominance pot. section VR7a INCREASES the level of the BACK FACING CARDIOID from A12a x $(1 + \sqrt{2})$ (+7.65dB) when turned ACW for BACK Dominance, and REDUCES it $\dfrac{1}{(1 + \sqrt{2})}$ (-7.65dB) when turned CW for FRONT Dominance. The signals are also INVERTED by A13b.

$$A13a\ O/P = W3 + X8 \text{ (Test point 33)}$$

AT NEUTRAL DOMINANCE, $W3 + X8 = -(2 - \sqrt{2})\ W - (\sqrt{2} - 1)X$

$\boxed{\begin{array}{l} W\ -4.65dB\ \text{INVERTED} \\ X\ -7.65dB\ \text{INVERTED} \end{array}}$

AT ACW, BACK DOMINANCE, $W3 + X8 = -\left(\dfrac{2 - \sqrt{2}}{1 + 2}\right)W - \left(\dfrac{\sqrt{2} - 1}{1 + 2}\right)X$

$\boxed{\begin{array}{l} W\ -12.3dB\ \text{INVERTED} \\ X\ -15.3dB\ \text{INVERTED} \end{array}}$

AT CW, FRONT DOMINANCE, $W3 + X8 = -\sqrt{2}W - X$

$\boxed{\begin{array}{l} W\ +3dB\ \text{INVERTED} \\ X\ \ 0dB\ \text{INVERTED} \end{array}}$

Note that at the Neutral position, since X is normally enhanced $\sqrt{2}$ (+3dB) this is a FRONT FACING CARDIOID of max. level (0°) $2\ (2 - \sqrt{2})$ and that the action of Dominance pot. section VR7b INCREASES its level from A13a x $(1 + \sqrt{2})$ (+7.65dB) when turned CW for FRONT Dominance and REDUCES it x $\dfrac{1}{(1 + \sqrt{2})}$ (-7.65dB) when turned ACW for BACK Dominance. The signals are also inverted by A13a.

$$\underline{A14a\ (C56)} = W4\ \underline{\text{(W OUTPUT FROM SOUNDFIELD CIRCUITS)}}$$
$$W4 = -\frac{1}{2(2 - \sqrt{2})}\left[(W2 + X7) + W3 + X8\right]$$

($\dfrac{1}{2(2 - \sqrt{2})}$ is the gain of A14a, and A14b below; minus sign indicates A14a inversion.)

AT NEUTRAL DOMINANCE $\boxed{W4 = W}$

AT ACW, BACK DOMINANCE, $W4 = \sqrt{2}W - \dfrac{X}{\sqrt{2}}$

$\boxed{\begin{array}{l} W\ +3dB,\ \text{IN-PHASE} \\ X\ -3dB,\ \text{INVERTED} \end{array}}$

i.e. W becomes a BACK FACING $\overline{\text{SUB-CARDIOID}}$.

AT CW, FRONT DOMINANCE, $W4 = \sqrt{2}W + \dfrac{X}{\sqrt{2}}$

$\boxed{\begin{array}{l} W\ +3dB,\ \text{IN-PHASE} \\ X\ -3dB,\ \text{IN-PHASE} \end{array}}$

i.e. W becomes a FRONT FACING $\overline{\text{SUB-CARDIOID}}$.

A14b (C59) = X9 <u>(X OUTPUT FROM SOUNDFIELD CIRCUITS)</u>

$$X9 = \frac{1}{2(\sqrt{2}-1)}\left[(W2 + X7) - (W3 + X8)\right]$$

AT NEUTRAL DOMINANCE, $\boxed{X9 = X}$

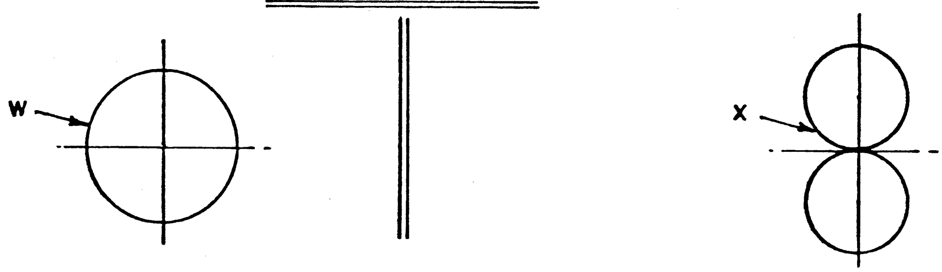
AT ACW, BACK DOMINANCE, $X9 = \frac{-W}{\sqrt{2}} + \sqrt{2}X$ $\boxed{\begin{array}{l}\text{W -3dB, INVERTED}\\ \text{X +3dB, IN-PHASE}\end{array}}$

i.e. X becomes a BACK FACING HYPER-CARDIOID.

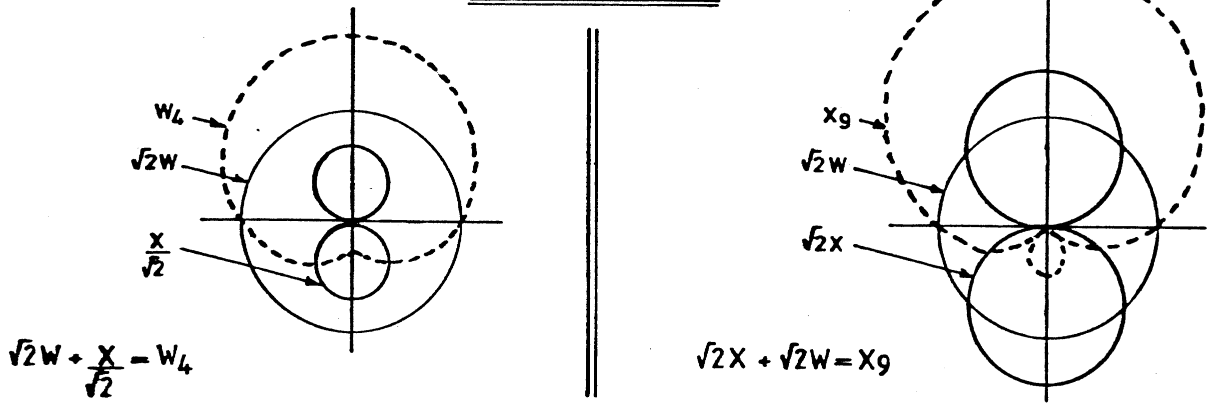
AT CW, FRONT DOMINANCE, $X9 = \frac{W}{\sqrt{2}} + \sqrt{2}X$ $\boxed{\begin{array}{l}\text{W -3dB, IN-PHASE}\\ \text{X +3dB, IN-PHASE}\end{array}}$

i.e. X becomes a FRONT FACING HYPER-CARDIOID.

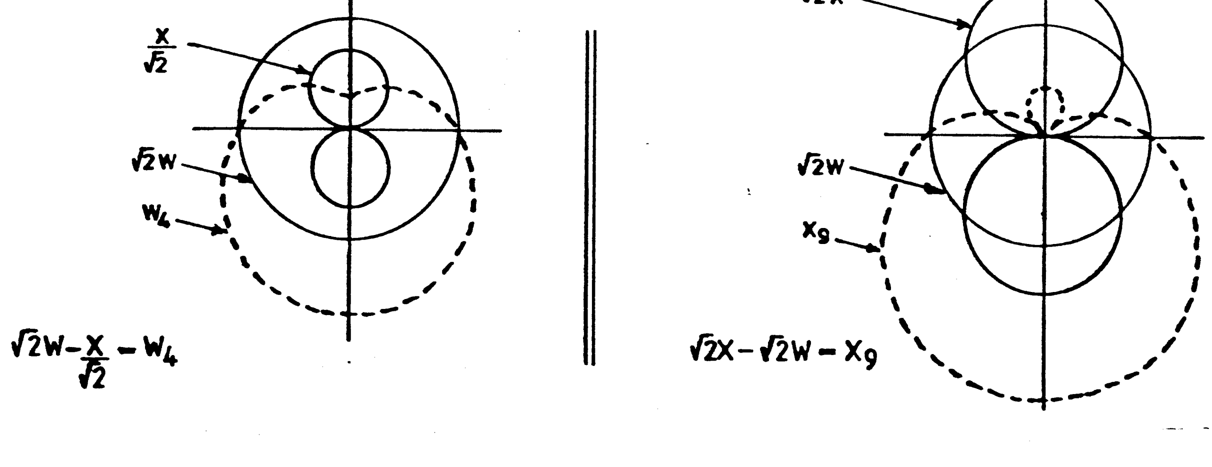
<u>Neutral Dominance</u>



<u>Front Dominance</u>



<u>Back Dominance</u>



7. STEREO OUTPUTS.

There are 3 stereo outputs from the Control Unit:
-the main STEREO OUTPUT available on 2 XLR3M sockets, LEFT and RIGHT on the rear panel: (these signals are NOT subject to the MONITOR gain control)
-two STEREO jack sockets, one on the front panel and one on the rear panel: both subject to the MONITOR gain control.

The stereo output circuits are fed with B-FORMAT signals X, W and Y from the RECORD amplifiers (and hence the microphone) in the normal mode (Z, the height component, is not used in stereo except in the Soundfield circuits to effect elevation when required). If TAPE is pressed the circuits are switched to the B-FORMAT signals from the tape recorder via the B-FORMAT PLAY socket. On DUB the RECORD LEVEL control may be introduced.

The main STEREO OUTPUT levels are approximately OdBu programme level for OdBu B-Format programme inputs. Level adjustment is available in the DUB mode. The MONITOR jack outputs may be adjusted from OFF to +10dB relative to the main levels.

There are two further controls for when the unit is being used as a STEREO MICROPHONE, or when dubbing from B-Format to Stereo, or (experimentally) when monitoring a B-Format recording in Stereo. These are POLAR PATTERNS and ANGLE, the latter allowing adjustment of the "angle-of-point" of the synthesised stereo pair of microphones.

The circuits around these controls : A8, A9a, A10a, A10b and MONITOR: A11a, A11b, may be checked precisely as follows:-

With the control unit on TAPE and OdBu, 1KHz signals at the appropriate pins of the B-FORMAT PLAY XLR5F socket should give output signals LEFT AND RIGHT XLR3M as follows:

<table>
<tr><th>INPUT</th><th>PATTERN</th><th>ANGLE</th><th>LEFT OUTPUT</th><th>RIGHT OUTPUT</th></tr>
<tr><td>W</td><td>Omni
Cardioid
Fig-of-8</td><td>Any</td><td colspan="2">OdB INVERTED
-6dB INVERTED
OFF</td></tr>
<tr><td>X</td><td>Fig-of-8</td><td>0°
90°
180°</td><td colspan="2">-3dB INVERTED
-6dB INVERTED
OFF</td></tr>
<tr><td>X</td><td>Cardioid</td><td>0°
90°
180°</td><td colspan="2">-9dB INVERTED
-12dB INVERTED
OFF</td></tr>
<tr><td>Y</td><td>Fig-of-8</td><td>180°
80°
0°</td><td>-3dB INV.
-6dB INV.
OFF</td><td>-3dB IN-PHASE
-6dB IN-PHASE
OFF</td></tr>
<tr><td>Y</td><td>Cardioid</td><td>180°
90°
0°</td><td>-9dB INV.
-12dB INV.
OFF</td><td>-9dB IN-PHASE
-12dB IN-PHASE
OFF</td></tr>
<tr><td>X & Y</td><td>Omni</td><td>Any</td><td colspan="2">OFF</td></tr>
</table>

NOTE 1: that the general inversion is one of circuit convenience only and that truly, only Y at the RIGHT output is inverted with respect to other signals. (Reference to Fig. 2 will remind the user that the RIGHT hemisphere is negative for Y).

NOTE 2. In standard B-Format the pressure-gradient signals X, Y (Z) are enhanced 3dB so if the table is now viewed with this in mind, it may be seen that with the STEREO MICROPHONE set for CARDIOID and 0° ANGLE, both outputs (LEFT and RIGHT) would be made up of W = -6dB and X = -6dB i.e. true CARDIOID patterns, facing FRONT.

When the angle is set, say, for 90°, the Y component appears at -9dB and X falls to -9dB (-12 on the test). It has previously been seen that -3dB $\left(\frac{1}{\sqrt{2}}\right)$ equates to 45° vector addition such that the TOTAL pressure-gradient component (X and Y combined) remains at -6dB level and the patterns remain CARDIOIDS one now pointing 45° LEFT at the LEFT output and, since Y is INVERTED for RIGHT, a second one pointing 45° RIGHT at the RIGHT output.

Examples of the inter-play of these controls and general POLAR PATTERN theory may be seen in Fig. 5.

Finally, check the output of the 2 MONITOR JACKS: The level should be variable to approx +10dB relative to the table by use of the MONITOR GAIN control.

Fig 5a. Mono polar patterns

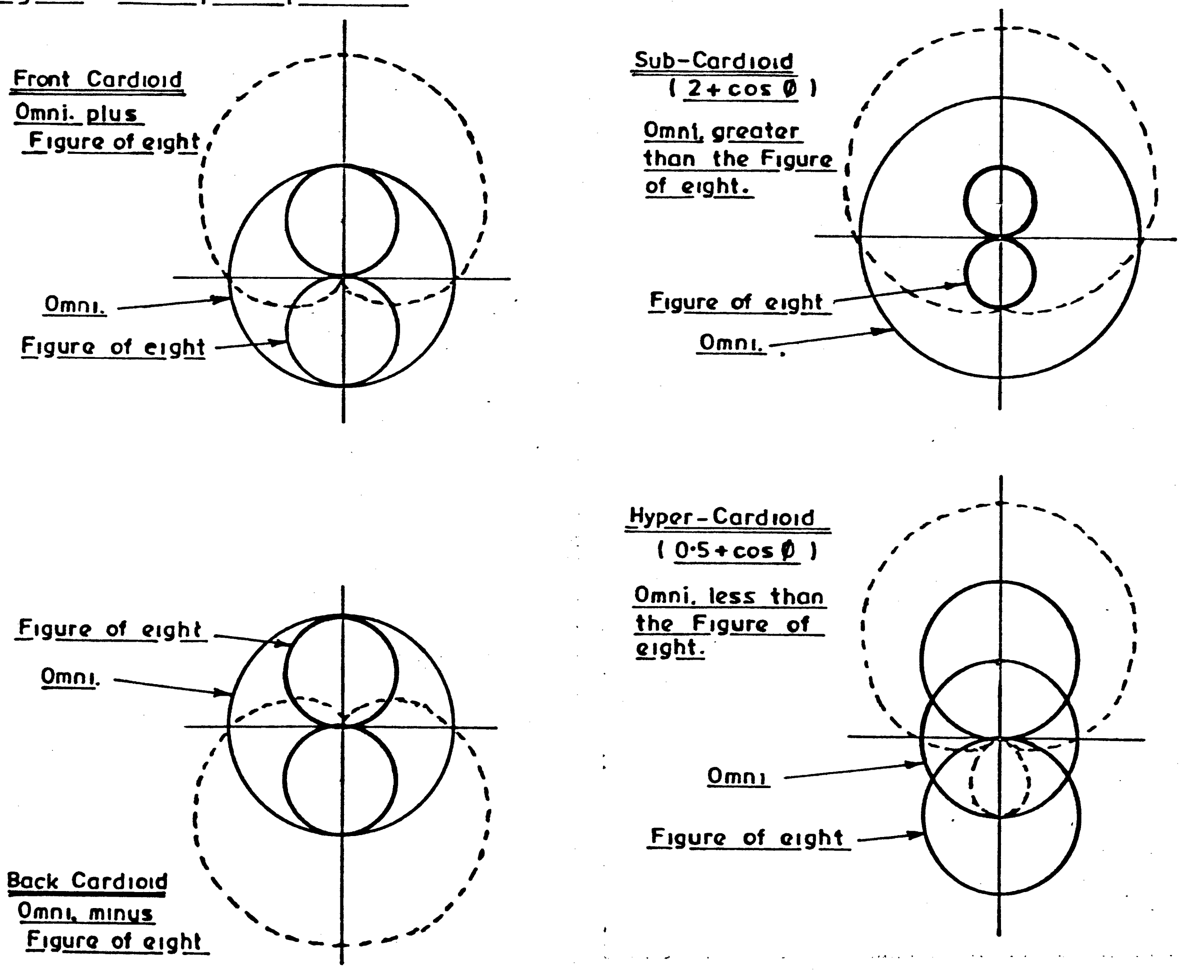
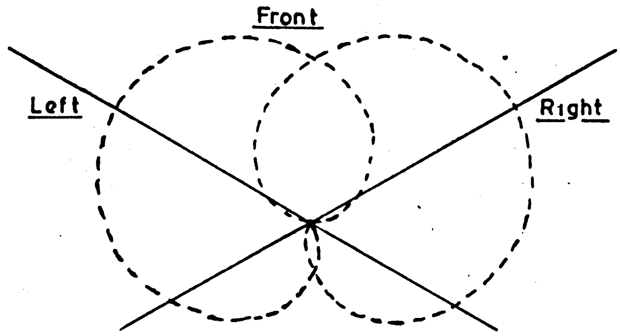
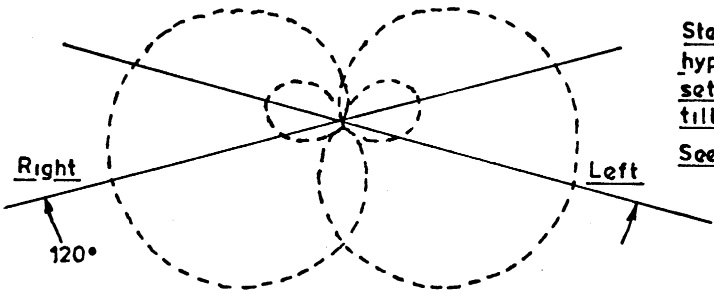


Fig. 5b. Two of many Stereo configurations from the Soundfield microphone.

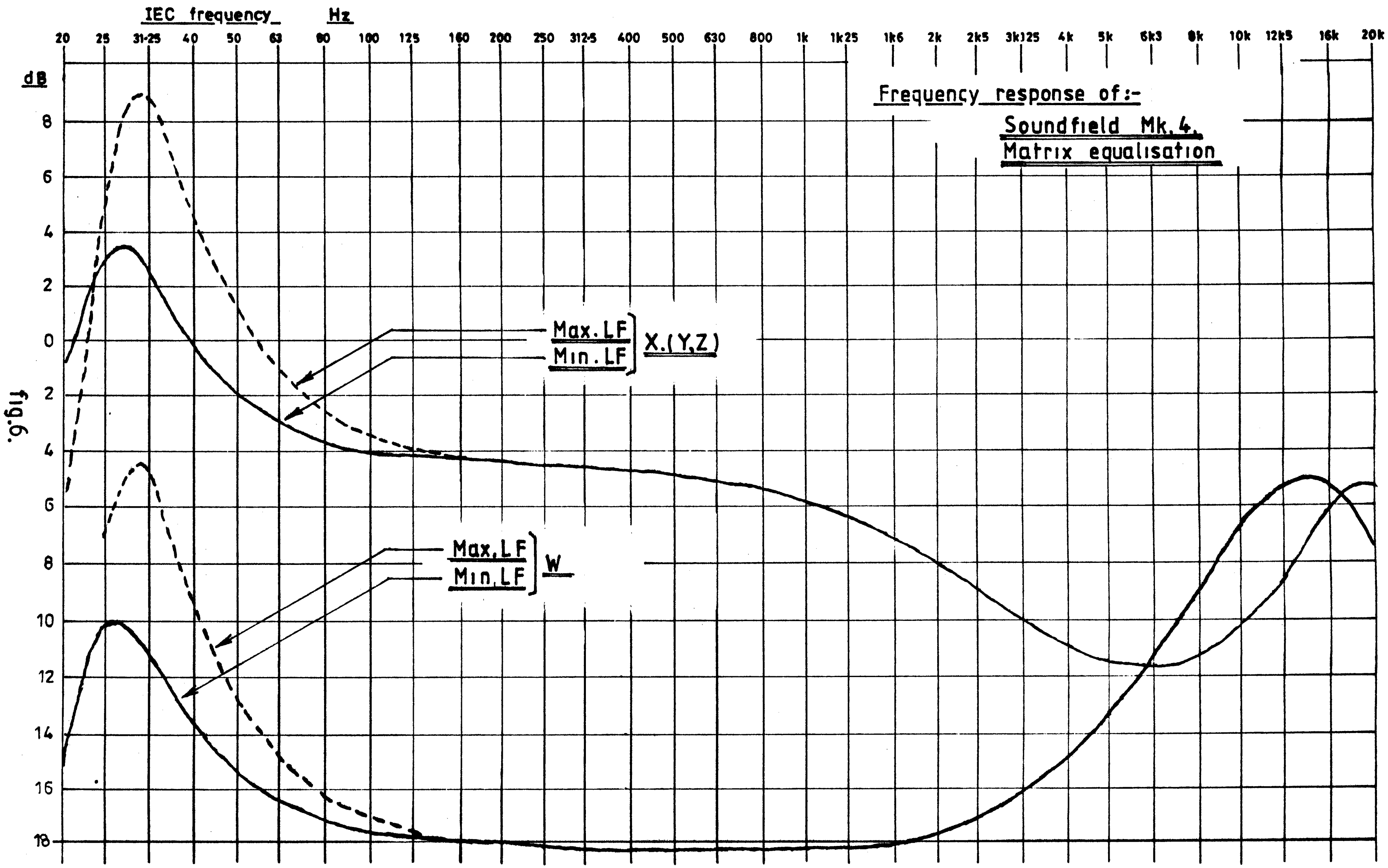


Stereo pair of synthesised cardioid microphones.



Stereo pair of synthesised hyper cardioid microphones, set at Angle 120°, and tilted downwards.

Seen from front.



IEC frequency Hz
20 25 31·25 40 50 63 80 100 125 160 200 250 312·5 400 500 630 800 1k 1k25 1k6 2k 2k5 3k125 4k 5k 6k3 8k 10k 12k5 16k 20k
dB
8 6 4 2 0 2 4 6 8 10 12 14 16 18
Frequency response of:-
Soundfield Mk.4.
Matrix equalisation
Max.LF
Min.LF
X.(Y,Z)
Max.LF
Min.LF
W

fig.6.

8. MATRIX MODULE.

This module is fitted separately to the Soundfield Mk 4 Control Unit and may be withdrawn at the rear following the removal of 4 screws. It is set for one particular microphone only.

It uses a total of 19 operational amplifiers in 10 packages. A1a, 1b, 2a and 2b are electronically balanced input amplifiers each with a gain of 6dB, for the 4 "A" signals from the microphone. Each can be adjusted for a good rejection to unwanted common mode signals. A3a, 3b, 4a and 4b provide L.F. boost and filtering, which, together with L.F. boost/filter circuits to the B-Format signals around A7b, 8b, 9b and 10b and the SET L.F. trim-pots allow a range of adjustment of about 2-10dB between 25 and 30Hz and a (combined) slope of 24dB/octave below 20Hz (see Fig. 6). Amplifiers A1a, 1b, 2a and 2b also couple with the GAIN switch in the control unit which sets the amplifiers gain between 0 and 25dB in 5dB steps. These gains are set on the GAIN switch as follows: Positions 0, 5, 10 and 15 correspond to 0dB, 20 = 5dB, 25 = 10dB, 30 = 15dB, 35 = 20dB and 40 = 25dB. (At positions 0, 5 and 10, the capsules polarising voltage is reduced to provide 15, 10 and 5dB attenuation respectively at the microphone capsules).

Following these stages are 4 CAPS SENS pre-set adjustments with further connections to the control unit circuits for MUTE. Capsule signals LB-, RF- and RB+ are also fed to 3 inverting amplifiers having unity gain, A5a, 5b and 6(a). An inverted version of LF+ is not required. The + and - signs attached to the capsule signal symbols indicate up and down respectively; they represent the tilt of the capsules in the tetrahedron and are dropped in the matrix formulae below.

The 7 signals are combined in matrices around A7a, 8a, 9a and 10a to produce the 4 B-Format signals as follows; note that a minus sign here indicates the inverted signal:-

$$X = -LB + LF + RF - RB$$

$$W = LB + LF + RF + RB$$

$$Y = LB + LF - RF - RB$$

$$Z = -LB + LF - RF + RB$$

Reference to Fig. 2 will remind the user/engineer of the required form and direction of the B-Format signals, the evolution of the matrix formulae can be clearly seen when this Fig. is considered in combination with the drawing of the microphone capsule configuration.

The combination of the capsule signals in this way give rise to gain and phase changes which need equalisation. For example, adding capsule signals (in-phase) will produce a larger signal at low frequencies than subtracting them (2 inverted). Thus X, Y and Z require considerably more gain at L.F. than W.

Moreover X, Y and Z each require +3dB boost relative to W in Standard B-Format.

The general addition of pressure-gradient signal components to produce X, Y and Z, causes a progressive rise in signal whereas the spacing of the capsule causes phase and gain loss at H.F. to all the signals. Finally the capsule types used in the Soundfield microphone (special sub-cardioids), have a slightly falling L.F. response below about 100Hz.

The required equalisation to compensate all these effects accurately to 10KHz (and fairly accurately to 15KHz) is built around A7a, 7b, 8a, 8b, 9a, 9b, 10a and 10b. The X, Y and Z circuits are identical. The L.F. boost circuits are identical for all 4 signals and may be recognised as C44, C45, R85 and R86 for W for example. This L.F. boost is additional to that around A3a, 3b, 4a and 4b where the SET L.F. trim-pots are located.

The gain/frequency equalisation curves for the amplifiers may be seen in Fig. 6. Note that the highest frequency where the curves are level is 500Hz : this is used for the test signal frequency from now on.

Note that the balanced line inputs are D.C. de-coupled from the input cable socket connections which are biased to +50V at high resistance so as to permit alternative microphone lines carrying +48V phantom power to be used for the microphone connection without problems. There are also protection diodes to prevent connection pulses from reaching the input amplifiers.

For initial alignment set ALL presets on the module to full ANTI-CLOCKWISE (min). Set the Control Unit GAIN switch to 0 and the RECORD LEVEL to 0 (Line-up). Ensure INV, OSC, TAPE, SOUNDFIELD IN etc are not depressed. Connect 0dBu, 500Hz to the LB input in correct phase and check for (approx.) -0.5dBu INVERTED at the W output (XLR5M).

Using 2 x 120R resistors matched better than 0.1%, set the CM REJECTION trim-pot RV1 at 500Hz and seal. Set CV1 at 10KHz and seal. Check the LB MUTE button. Check that CAPS SENS, RV9 increases the gain +6dB approx. and that the GAIN switch increases the gain in 5dB steps (±0.3dB) between settings 15 and 40. Reset these to minimum.

Check the W SENS, RV14 and the equalisation frequency response against Fig. 6. Look for a correct L.F. boost and filter response and adjustment range on SET LF, RV5. Reset the trim-pots to minimum (ACW).

Repeat the CM REJ, CAPS MUTE, CAPS SENS, GAIN check and SET LF for LF, RF and RB inputs. (No need to re-check W but use the W output).

TRIM-POTS						
CAPS SIGNAL	CM REJ		SET LF	CAPS SENS	B SIGNAL	B SENS
LB	RV1	CV1	RV5	RV 9	X	RV13
LF	RV2	CV2	RV6	RV10	W	RV14
RF	RV3	CV3	RV7	RV11	Y	RV15
RB	RV4	CV4	RV8	RV12	Z	RV16

Return to the LB input. Check for (approx.) +13.5 dBu IN-PHASE at the X output (XLR5M). Check the X SENS, RV13 and the equalisation frequency response against Fig. 6. Look for a correct (minimum) L.F. boost and filter response. Reset RV13 to min. Check the LF, RF and RB inputs at 500Hz only : LF and RF should be INVERTED, RB IN-PHASE.

Repeat the above checks at Y and Z outputs. For Y : LB and LF are INVERTED and RF, RB IN-PHASE. For Z : LF and RB are INVERTED and LB, RF IN-PHASE. Finally leave all 8 trim-pots at ELECTRICAL centre +3dB from minimum.

The gain in all stages of the recording chain including the microphone amplifier, may be seen in Fig 7. The main graphs assume the CAPS SENS, and X, W, Y, Z SENS pre-sets set at centre although limits are shown at the acoustic line-up level. Note that there is -3dB loss in the "transmission system" comprising the balancing resistors and RF rejection inductors at the output of the microphone amplifiers, the cable and the input RF rejection inductors at the control unit input to the matrix module.

It is interesting to note that if only one channel is checked for W, the resulting output signal is -9dBu at line-up whereas for X, Y and Z it is +5dBu, i.e. for X, Y and Z, +2dB more than the line-up for 4 capsules,which is +3dBu due to the X, Y, Z 3dB emphasis. This is because signals add in-phase for W but subtract for X, Y and Z.

Note also that the calculation for equivalent input noise relative to the W output benefits 3dB for 4 capsules relative to one channel since, as previously stated, signals add +9dB whereas noise adds only +6dB (2 "doublings" of 3dB each).
(Test points 22, 23, 24 and 25 may be used for trouble shooting).

9. MICROPHONE AMPLIFIERS.

These are located in the body of the microphone on one P.C. card and all 4 are identical.

Each comprises a low-noise field effect transistor with a very high input resistance of 2000 meg-ohms, current feeding an operational amplifier which returns series negative feedback to the FET source such that the combined gain is +20dB.

This circuit arrangment ensures that the operational amplifiers contribute negligible noise to that produced at the FET's input EVEN WHEN THE EXTENSIVE HF BOOST IS LATER APPLIED TO THE W SIGNAL. (This was a limitation of earlier designs using an FET in the familiar source follower mode). The resulting equivalent input noise of the 4 microphone channels to the W output is approximately 19 phons which is comparable with large single capsule microphones. (Early leaflets conservatively claimed only 20-22dBA).

The FET stages are biased by a single potential divider network in the gate circuit which gives a measure of D.C. negative feedback from the drain. This ensures repeatable D.C. conditions as shown on the circuit against FET tolerance spreads.

The operational amplifiers are A.C. coupled to the FET's and are thus biased independently to half the the supply voltage of 20V by one potential divider chain for all 4 stages (R54, R55, C26). Each has DC feedback via a 10M-ohm resistor.

A.C. feedback for +20dB is decided (for LB for example) by R9 and R6 in parallel with R7. C3 and C4 provide D.C. isolation so as not to upset the important separate stage and FET biasing. The choice of R7 at the low value of 100R ensures negligible noise contribution from this source. C3 and C4 at 48 and 100uF respectively are however necessary for adequate L.F. performance and even then, there is slight L.F. roll-off. This is made up together with any boost required for capsule LF roll-off in the SET LF circuits as part of the Acoustic Tests in the matrix module as previously described.

The output of the operational amplifiers feed BALANCED line outputs which are connected one leg to the amplifier and one to earth via 100R close-tolerance resistors. The lines are D.C. de-coupled from the amplifiers and biased to +50V to allow alternative conections to the control unit to be made by microphone studio lines already carrying +48V phantom power. Protection diodes prevent connection pulses from reaching the operational amplifers.

The D.C. supply to the microphone amplifiers arrives at +50 volts so that it may also use a microphone live having +48V phantom power on it if required. Following suitable RF filtering (which is also applied to all other connections) the voltage is dropped via a 1K-ohm, 1W resistor (R56) located in the stem of the capsules mount. This ensures the capsules are warmed very slightly to dispel condensation.

The connection continues via a L.E.D. (D11) which is mounted in alignment with a hole in the microphone case at CENTRE FRONT to assist in alignment when mounting at a recording session. The supply to the amplifiers is finally set at +20V by the zener diode D12 and filtered by C38 and C39.

The capsule polarising voltage is brought in separately since it is progressively reduced from the normal value of +50V at the lower gain settings as explained in the section describing the matrix module. The voltage reductions are achieved by zener diodes such that, when an alternative connection to the microphone is achieved via studio microphone lines carrying +48V phantom power, the phantom voltage is suitably over ridden (It is fed via a 6.8K-ohm resistor as part of a standard specification and is designed to be reduced to approx +12V by some microphones).

First * alignment of the microphone amplifiers should be carried out prior to connection of the capsules as follows: (These connections are easily removed if necessary) Test connections should be made via 47pF polystyrene capacitors.

General tests are best carried out at low gain settings i.e. GAIN OdB and RECORD LEVEL at O (line-up) with the CAPS SENS and X, Y, Z SENS AT CENTRE, (as left following the Matrix module tests), the gain of one channel from the capsule input to the W output is -31dBu input for -9dBu output or +22dB. One channel to the X, Y, or Z output is 14dB more i.e. 36dB at 500Hz.

To avoid interference from hum fields, use -20dB input at 500Hz and look for approx +2 and +16dBu outputs respectively, mute unused channels to avoid interference. The level at the microphone amplifiers outputs should be -0.5dBu and, as a check of the cable and connections, -16.5dBu should appear on the 100R of the "earthing leg" of each circuit. The level at the matrix module input amplifier outputs is +3dBu approx. (There is -3dB loss in the "transmission system" which is mostly the 100R resistors and +6dB gain in the matrix module input amplifiers).

The frequency response at the microphone output (AT THE SOCKET : BALANCED LINE) is detailed on the microphone amplifier circuit if required. Otherwise this may be checked at the W output and should follow graphs seen in Fig 6.

Next check the microphone amplifiers clipping level by increasing the input signal to approx -2dBu and observing the W output. X, Y and Z will clip first at +22dBu output when fed individually in this way, at approx -14dBu input.

Now set the GAIN to +35 and confirm each channel at -42.5dBu input at 500Hz for OdBu W output (±0.5dB). If this is satisfactory, note the precise gain and disconnect the input signal. Ground the free ends of the 4 x 47pF capacitors and place the microphone on an earthed metal plate so that the temporary capacitors are guarded and clear of mains interference.

Using the MUTE buttons, check each channel for a noise output of -70dBA. approx, (Equivalent to -112.5dB, "A" weighted at the microphone amplifier input). Look for consistent performance across all 4 amplifiers; ensure no hum is present. Change any suspect components.

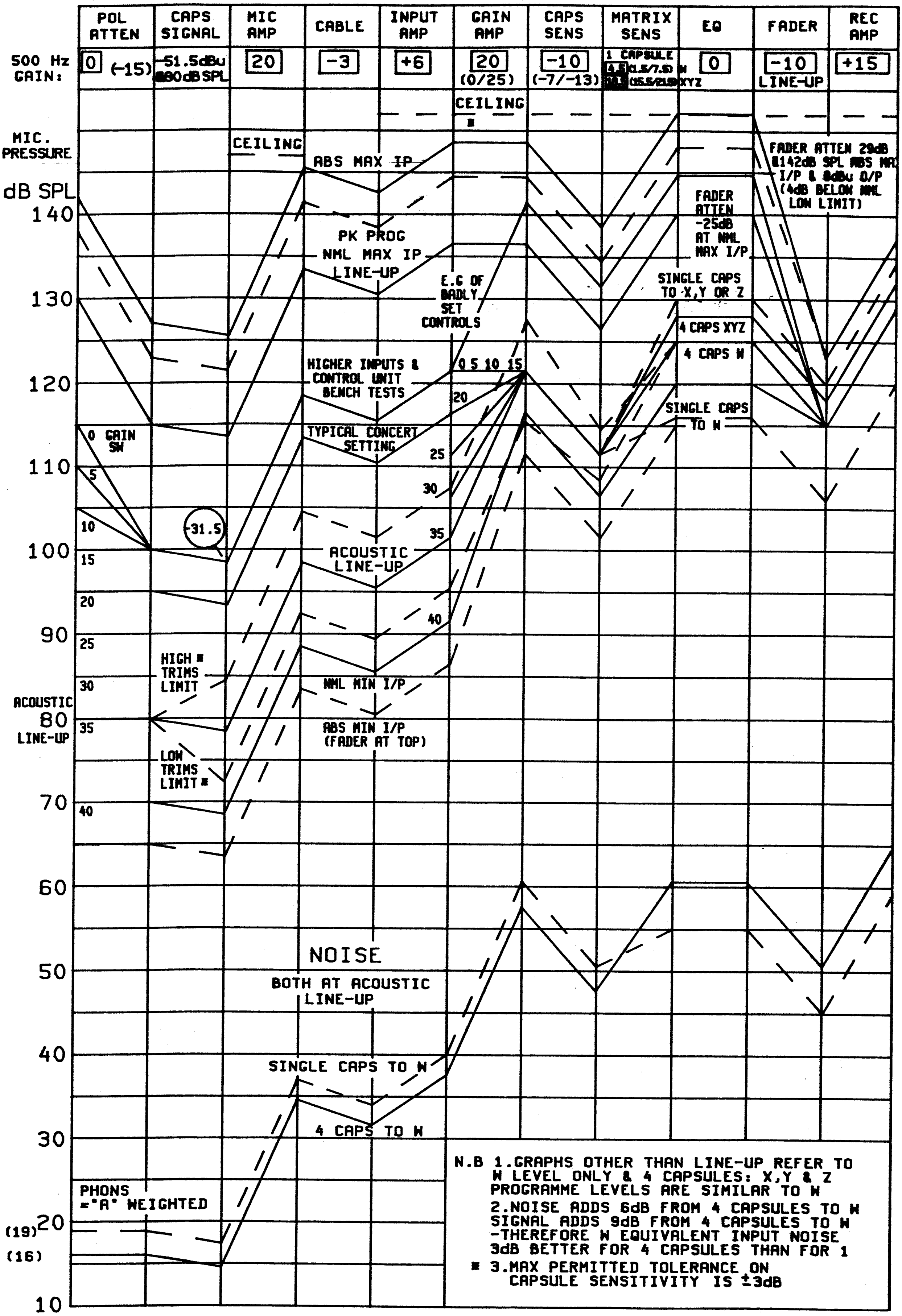
Finally check the noise of all 4 channels together to the W output at -64dBA approx.

Remove the temporary capacitors and connect the capsules. Replace the microphone cover. Confirm operation.

The equipment is now ready for the acoustic alignment.

*N.B. Alignment other than noise may be carried out without disconnecting the capsules by feeding the test signal (of -42.5dBu for 0dBu at the W output) to the polarising connection at the capsules end of the microphone via a 10uF, 63V polystyrene or polycarbonate capacitor. In this way all 4 channels are fed simultaneously through the capsules which, in combination with the MUTE buttons, allows some very rapid checks to be made.

SOUND-FIELD Mk4 SIGNAL & NOISE LEVELS
FIG.7



SF4LEVELS 0,0,0.9 KF 22 JUL 83

10. ACOUSTIC TESTS.

Place the microphone in a vertical position with its Front approximately 1 metre from a tone source (test L.S.) in an anechoic chamber. There should be a calibrated probe microphone near the S.F. microphone set to maintain a pressure of 80dB SPL (2 microbars) independent of frequency over the range 20Hz to 20KHz.

Set the Control Unit to Line-up as follows:-

GAIN	:	35
RECORD LEVEL	:	0 (Line-up)
UPRIGHT OR INVERTED	:	As appropriate (Note that if MIC is tested INVERTED, when it is rotated to the LEFT the RIGHT capsules turn towards the L.S. first and vice versa. i.e. LF becomes RF and vice versa LB becomes RB and vice versa).
MUTES	:	ALL MUTED
OSC TAPE DUB SF IN SF REC	:	ALL OUT
ANGLE	:	0°
PATTERN	:	FIGURE-OF-EIGHT
MONITOR	:	For a suitable phones level.

Point the LB of the mic. to the L.S.

Remove the LB MUTE and switch on at 500Hz.

X output should be +4dBu approx. (+2dBV)*

Use phones and/or a 'scope to ensure there is no interference.

Look at LF, RF and RB in turn.

Now adjust X SENS for +4dBu average (+2dBV)

Then run through again setting LF, RF and RB SENS for exactly +4dBu (+2dBV) at X output at each position.

Now check with LB and RF on together for a figure-of-eight along the LB-RF axis and peak levels of 0dBu (-2dBV).

Repeat for RB and LF.

Finally check with all capsules on together for a figure-of-eight along the Front/Back (X) axis and peak levels of +3dBu (+1dBV).

Trim CAPS SENS, X SENS as necessary.

Check the frequency response on the X axis and patterns at other frequencies.

The frequency response should rise slightly at 50Hz.

ENSURE THAT 500Hz IS REPRESENTATIVE OF GENERAL MID-BAND SENSITIVITY.

(The dBV meter will read dBu (approx.) by increasing the sensitivity to -2dBV).

Transfer to W output and set PATTERN to OMNI.

Point LB to L.S. and mute other capsules.

Check at 500Hz for -9dBu approx. (-11dBV).

Repeat for LF, RF and RB.

Set W SENS for -9dBu average (-11dBV).

Switch on all capsules and check at all positions for a steady OdBu. Trim W SENS as necessary.

Check the frequency response at FRONT (200Hz* to 20KHz) and patterns at other frequencies.

ENSURE THAT 500Hz IS REPRESENTATIVE OF GENERAL MID-BAND SENSITIVITY.

*Below 200Hz the response will fall off so proceed as follows:-

Reselect LB only and turn it towards the L.S.

Increase the measuring amp. gain temporarily to bring 500Hz to 0 on the meter.

At 50Hz trim LB LF boost for level frequency response.

Repeat for LF, RF and RB checking for 0 at 500Hz each time, adjusting the appropriate LF boost trim-pot.

Return to all four capsules and normal amplifier sensitivity.

Check for level response 500-50Hz (0, -1dB).

Transfer to Y output and set PATTERN to FIGURE-OF-EIGHT and ANGLE to 180°

Repeat as for X but only adjust Y SENS initially.

IF THE SENSITIVITY AND PATTERNS DO NOT CONFORM ON THE Y AXIS IT MAY BE NECESSARY TO TRIM LB, LF, RF AND RB SENS BUT IF SO, THE X AND W TESTS WILL HAVE TO BE REPEATED UNTIL THE BEST AVERAGE IS OBTAINED. (WITH A LITTLE PRACTICE THE CORRECT RESULTS WILL APPEAR).

The frequency response at LF will now rise more than X due to the LF adjustments in the W tests.

Transfer to the Z output. Phones/'scope monitor is now only possible by introducing the ELEVATION control to MAX UP with SF IN.

The individual capsule sensitivities of +4dBu (+2dBV) at each position may be checked and also LF-RF and LB-RB pairs for -2dBu (-4dBV) max at CL/CR and CF/CB respectively. Furthermore, a continuous null should be achieved with all capsules on.

However, it is better to re-site the microphone in a horizontal position, FRONT DOWNWARDS and pointing at the loudspeaker at normal, FRONT.

Check LF at TRUE LF position for +4dBu (+2dBV) at 500Hz.
Check RF at TRUE RF position.
Look at RF ~~and~~ RB in turn at TRUE RB AND RF positions respectively.

Now adjust Z SENS for a good average of +4dBu (+2dBV) at each position.

LF and RF should give a figure-of-eight on axis LF-RB with OdBu (-2dBV) max. levels.

Similarly LB and RB should give a figure-of-eight on axis LB-RF, with OdBu (-2dBV) max. levels.

Finally check for a figure-of-eight on the FRONT-BACK axis with +3dBu (+1dBV) max. levels and correct nulls at LEFT and RIGHT with all capsules on.

Trim Z SENS if necessary. DO NOT COMPROMISE X AND Y PATTERNS BY ADJUSTING CAPS SENS TRIM-POTS.

Check the frequency response on the Z axis and patterns at other frequencies. ENSURE THAT 500Hz IS REPRESENTATIVE OF GENERAL MID-BAND SENSITIVITY.

If all the above tests are satisfactory, take records of the following:-

FREQUENCY RESPONSE	:	W at 0°	
		X at 0°	
		X at 90° CENTRE LEFT (AND RIGHT)	
		MONO CARDIOID AT FRONT	ANGLE = 0°
		MONO CARDIOID AT BACK (180°)	
		STEREO CARDIOID AT 45° LEFT	ANGLE = 90°
		STEREO CARDIOID AT 45° RIGHT	

Now check correct gain and reduction in the W and X front levels at GAINS 40, 30, 25, 20 and 15 for 5dB steps. Check more closely GAINS 10, 5 and 0, particularly the latter, for 5dB (approx) sensitivity change at each position but more importantly, correct relationship between W and X and no loss of X patterns.

(X tends to increase slightly in sensitivity at low polarising voltage which occurs at the "0" GAIN position).

Finally check noise to the W and X outputs and for a cardioid at 0°. The results should be as follows:-

W	:	18 phons (dB, A weighted)
X	:	22 phons
Mono Cardioid	:	16 phons*
(Tolerance	:	±1dB)

Note the actual results in the frequency response graph margins.

*The Mono (or Stereo) cardioid gives optimum noise due to cancellation effects of coherent noise from individual capsules in the formation of cardioid(s) from X and W. Omni noise is about 2dB worse and figure-of-eight about 4dB worse for a similar front sensitivity.