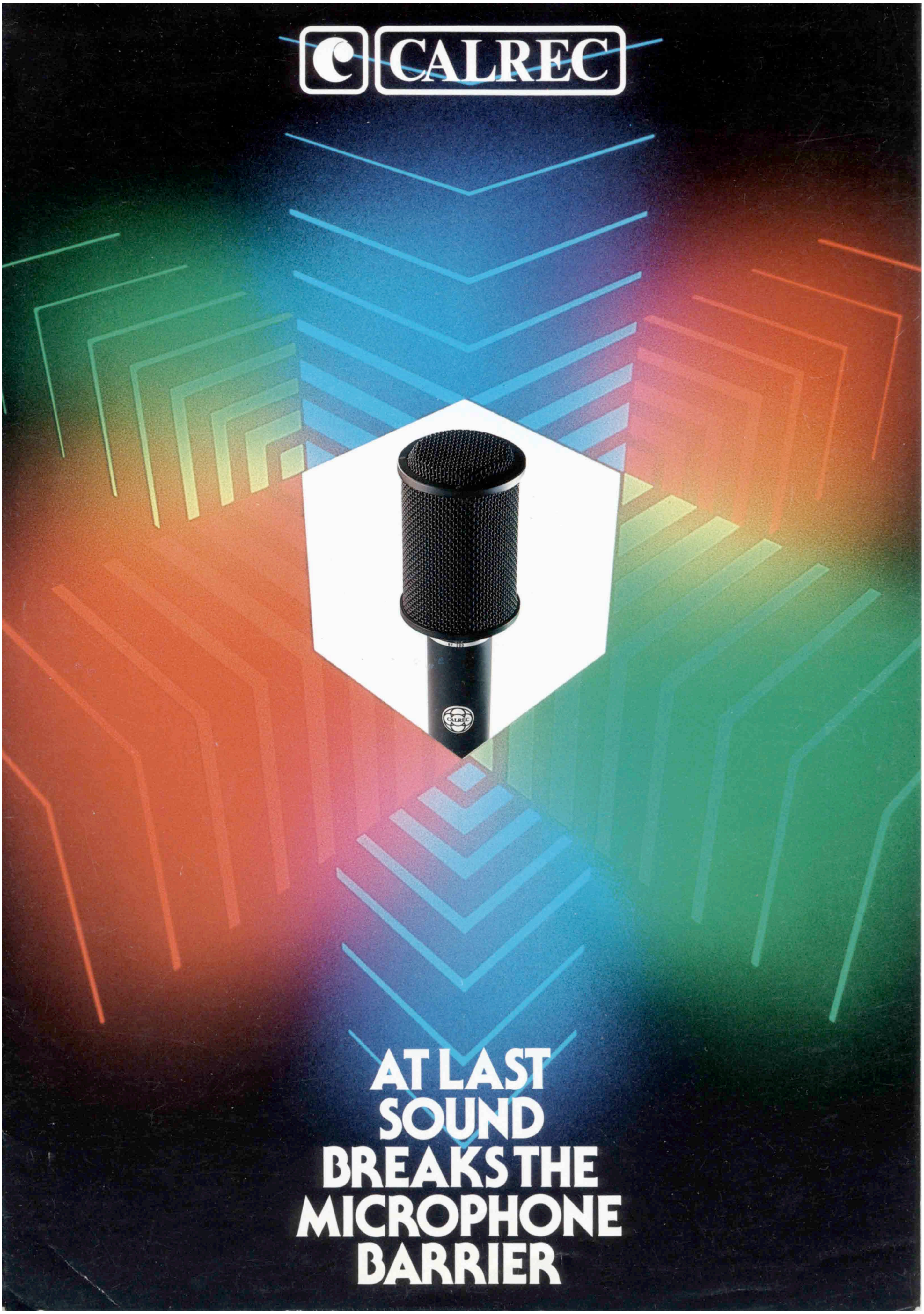

CALREC
CALREC
AT LAST
SOUND
BREAKS THE
MICROPHONE
BARRIER

CALREC

Producers of the first and only Soundfield Microphone System

THE FEATURES

* The only truly coincident stereo microphone in the world.
* The most accurate polar patterns in the world.
* Equally level frequency response to both on and off axis sounds.
* Very low noise performance for the digital era.
* Totally steerable both horizontally and vertically either live or in post production.
* Variable stereo capsule angle and polar pattern either live or in post production.
* Variable zoom, forwards or backwards either live or in post production.
* Perfect mono compatability from the stereo output.
* Very high maximum sound pressure level capability.
* Separate outputs for Ambisonic Surround Sound.

The Mk IV Soundfield Microphone is the result of over 10 years research by mathematicians, scientists and audio engineers. It stems from the most exhaustive study ever conducted into psychoacoustics, the hearing mechanisms of the human brain and their relationship to natural sound reproduction. Calrec Audio Ltd., are proud to have been associated with this work which has produced the first fundamental step forward in transducer design for many years.

The Soundfield Microphone forms part of the Ambisonic Surround Sound technology which was developed under the auspices of the United Kingdom National Research Development Corporation and is the subject of U.K. Patent No. 1494751 and U.S.A. Patent No's 3997725 and 4042779, together with all corresponding patents in other countries and all other patents pending.

THE PRINCIPLE

It has long been recognised that the main problem with stereo coincident microphones and discrete microphones used as coincident pairs is that they are not coincident, nor are they uniformly non-coincident. In almost all microphone arrays, phase errors due to capsule spacing become significant between 1 to 2KHz, and can be in excess of 90 degrees at frequencies as low as 1.5KHz, dependent on the angle of arrival of the sound at the microphone head.

The philosophy of the Soundfield design is to eliminate, as far as is practically possible, these random phase errors and produce a stereo output whose two signals appear to have originated at the same point in space.

The concept of Soundfield is based on the mathematical theory of sampling on the surface of a sphere and is best explained by considering the properties of a theoretical loudspeaker with an omni-directional pattern radiating spherical wave fronts.

If a spherical diaphragm were constructed, sufficiently light and unrestrained that it moved in exactly the same way as the air particles under the influence of the spherical sound waves produced by the point source speaker, then the diaphragm would be acoustically transparent (see fig. 1). If the point source speaker were removed and the diaphragm replaced with a dense array of speakers driven (with equal amplitude and phase) so as to have exactly the same displacement as the original diaphragm, then the spherical waves would continue to radiate exactly as before and an external observer would hear no difference (see fig. 2). If those speakers were replaced by microphone capsules then the exact opposite would apply, that is the combined output of all the capsules would exactly represent the pressure (or omni-directional) signal present at the CENTRE of the sphere. Such a device would, in principle, be an ideal omni-directional microphone.

A similar exercise could be conducted using a theoretical figure-of-eight loudspeaker. The pattern of diaphragm displacement would be more complex in that there would be two hemispheres moving in antiphase separated by a stationary equatorial area.

However, it is still possible to mimic this motion using a dense array of loudspeakers and by similar reciprocity show that, as a microphone, it would have a figure-of-eight response whose centre point was at the centre of the original sphere, not on the surface where the transducers were situated. Unique to the Soundfield system is the fact that these two functions may be simultaneously combined in one array and, by different combination of the capsule outputs, both the omni-directional (zero order) and the figure-of-eight (first order) response of the same point in space can be obtained. Indeed with the theoretical array of dense transducers it would be possible to extract microphone polars of higher orders, but in practice many factors, not the least of which is cost, limit the number of capsules used to characterise the surface of the sphere to four.

These are arranged in a regular tetrahedron and allow the generation of the single spherical harmonic of order zero which corresponds to sound pressure only and is, therefore, omni-directional, and the three spherical harmonics of order one which correspond to pressure gradient and are, therefore, figure-of-eight. These represent the three dimensions of the live sound and can be thought of as the component parts X=front-back gradient, W=pressure, Y=left-right gradient and Z=up-down gradient (see fig. 3). These components signals are called B-FORMAT and are available as nominal 0dBm outputs and inputs as well as the stereo outputs.

All normal microphones (i.e. other than specialised super directional types) belong to a single family responsive to a mixture of pressure and pressure gradient which is produced around either a single diaphragm, or a pair of diaphragms with a single electrical output. In any such systems the gradient component has a response that rises with frequency above approx. 1KHz. The pressure response falls with frequency above the same figure and a well designed capsule will utilize these properties to achieve a level frequency response to on axis and 180 degrees off axis sounds. At all other positions the frequency response performance is sub-optimal and must always be so because the pressure and gradient signals cannot be separated.

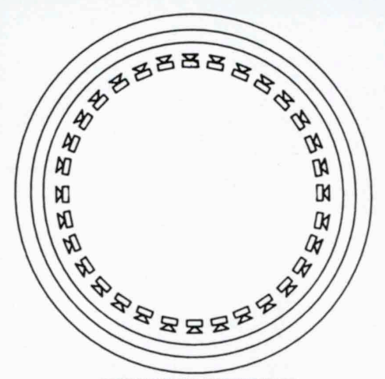


FIG 2 SOUND WAVES CONTINUE TO RADIATE EXACTLY AS BEFORE

The B-Format signals of the Soundfield System exactly represent the pressure and gradient components of a conventional microphone, but are entirely separate electrically. It is, therefore, possible for the first time ever, to correct the inherent defects of frequency response, and thus ensure substantially perfect patterns at all frequencies and on any axis.

The normalised B-Format signals may be stored on tape for post session processing or combined immediately into two apparent microphone signals of any characteristic between omni and figure-of-eight by simple vectorial addition. For example, combining the X signal (forward figure-of-eight) equally and in phase with the Y signal (left facing figure-of-eight) produces a 45 degree left pointing figure-of-eight. The same signals combined anti-phase produce a 45 degree right pointing figure-of-eight. This process creates a 90 degree crossed pair of microphones whose component parts are the same electrical signals, the frequency responses of which have been normalised prior to the sumation. This is one of the most important features of the system and its effect on image stability in the stereo picture has to be heard to be believed.

The example given is an over simplification to show the principle involved. The full facilities offered by the system allow the generation of two effectively coincident microphones, the angles of which may be continuously varied between 0 degrees (Mono) and 180 degrees, the polar patterns of the pair may be continuously varied between omni-directional and figure-of-eight, through all the cardioid positions. The direction of point may be steered through 360 degrees horizontally and plus or minus 45 degrees vertically and the complete array may be effectively zoomed closer to or farther away from the sound source. Moreover, this control over the microphone may be exercised either live at the time of recording 2 track stereo OR DURING POST SESSION PROCESSING OF THE B-FORMAT MASTER TAPE.

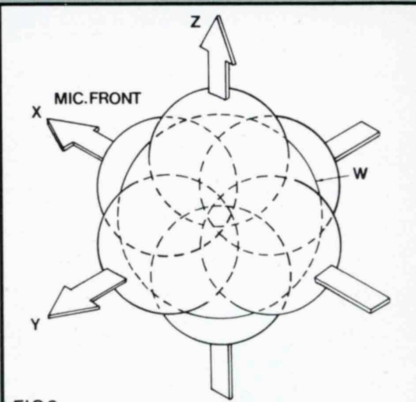


FIG3

THE FUTURE

The Soundfield is undoubtedly the most accurate and natural sounding microphone available today and certainly the most flexible, but its superiority does not stop there! The B-Format signals contain total first order information in a spherical form and as such are capable of supplying directional information to any present or future surround sound systems, particularly the Ambisonic system, for which it was originally conceived.

FIG1

The Mk IV Control Unit

TECHNICAL DESCRIPTION

The Units : Soundfield Mk 4 microphone in presentation box.
Mk 4 Control Unit.
100 metres connecting cable on a drum (SFC 1).
Mains cord.

Optional Accessories : Microphone Mounting Bar.
Anti-vibration mount.
10 metre Microphone Head lead/Extension cable (SFC 2).
Splitter cables type SFS 1/SFS 2 allow the microphone to control unit connection to be made via 5 studio XLR-3 balanced tie lines thus placing the control unit in the listening room (both cables are required).
5 pin XLR female to 4x XLR-3 male B-Format record output leads (SFS 3).
5 pin XLR male to 4x XLR-3 female B-format replay input leads (SFS 4).
(IT IS IMMATERIAL WHETHER OR NOT THE TIE LINES CARRY STUDIO 48 VOLT PHANTOM POWER).

The Microphone : This consists of four special capsules mounted in a regular tetrahedral array, followed by four extremely low-noise amplifiers driving four balanced outputs through a common 12-pole connector. The amplifiers receive power from the Control Unit through the same connector.
There is a LED which may only be seen from the front of the microphone to assist in correct placement. The amplifier powering is arranged to keep the capsules warm and dry thus avoiding condensation problems. The microphone can not be used without the control unit.

The Control Unit : In the rear of this unit there is a removable module which includes the microphone input connector and which contains the matrix amplifiers ADJUSTED TO THE PARTICULAR MICROPHONE. It, therefore, carries the microphone serial number.
The matrix is Factory set under anechoic conditions for precise matching of the four capsule signals and B-Format output signals to the remainder of the Unit. Should a user suspect or damage his microphone, he need therefore, only return the microphone and this module.

The Controls :

- **Capsule Mutes** : The four microphone signals may be muted as required to check the continuity of the connecting cable(s).
- **OSC Test** : Provides a 0dBu 1KHz signal at the B-Format record output continuous on W and pulsing at differing rates on X, Y & Z, to assist in track identification. Also gives a 0dBu stereo output when pattern control is set to omni.
- **LED (RED)** : Flashes warning when mutes are activated, steady warning when Oscillator is on.
- **Invert** : Corrects the matrix signals to allow for the microphone to be suspended in an inverted mode.
- **Gain Switch** : O to +40dB Microphone Input Gain in 5dB steps.
- **Gain Fader** : Fine gain adjustment +5, -15dB approx DENOTED BY ARC LINES. Below these lines for Fade up/down only.
- **Recording Level Indicators** : Respond to the above controls and monitor the Record Output B-Format signals.
- **Tape** : Allows the B-Format Tape Replay signal to be compared, monitored or replayed.
- **Soundfield In** : Connects the Soundfield controls into the B-Format signal path AFTER the TAPE control so that adjustments of these controls DO NOT affect the B-Format recording thus allowing experimentation during the session.
- **Soundfield Rec** : Connects the Soundfield controls into the B-Format RECORDING SIGNAL PATH to allow correction of a mis-aligned microphone. (The Soundfield controls may not be used in both paths simultaneously).
- ***Soundfield Azimuth** : Allows the microphone to be rotated in the horizontal plane ±45° from the four cardinal positions FRONT, LEFT, RIGHT, BACK:— NO BUTTONS = FRONT; BOTH BUTTONS = BACK.
- ***Soundfield Elevation** : Allows ±45° of vertical tilt from the horizontal.
- ***Dominance** : By an ingenious interplay between pressure and pressure-gradient characteristics, the direct to reverberant signal ratio and the microphone Front/Back sensitivity are adjusted in such a way that the microphone APPEARS to move fore and aft from its fixed position. Front Dominance narrows the aural scene slightly (where actual movement would widen it) but this does little to spoil the illusion.
- ***Stereo Microphone** : The B-Format signals are reduced to almost perfectly coincident stereo signals with two additional controls:
  - ***Polar Patterns** : Continuous adjustment from omni-directional through sub-cardioid, cardioid, hyper-cardioid to figure-of-eight.
  - ***Angle** : Continous symmetrical adjustment between LEFT and RIGHT from 0° to 180°.
- **Monitor** : Provides off/+10dB continuous adjustment to 2 stereo headphones sockets, 1 on front panel, 1 on rear panel.
- **Dub** : Allows a B-Format replay to INCLUDE the Fader Gain Control for dubbing purposes.
- **Power** : Indicates DC power on board from the Mains input.

NOTE THAT ALL THE "STEERING" CONTROLS (MARKED) MAY BE ADJUSTED POST-SESSION IF DUBBING FROM B-FORMAT TO STEREO.

Connectors : Microphone Input : DIN 12 pole
B-Format Record : XLR 5-M
B-Format Replay : XLR 5-F
PIN 1 : EARTH
PIN 2 : X : Front/Back Pressure-Gradient 1) (Recommended
PIN 3 : W : Omni, Pressure 2) Analogue
PIN 4 : Y : Left/Right Pressure-Gradient 3) Tape Track
PIN 5 : Z : Up/Down, Pressure-Gradient 4) Format).

Stereo Outputs : 2 x XLR 3-M
PIN 1 : EARTH
PIN 2 : SIGNAL
PIN 3 : EARTH

Headphones/Monitor : 2 x Stereo Jacks
TIP : LEFT
RING : RIGHT
SLEEVE : EARTH

Mains Input : IEC, with 2A fuse and adjustment for 100-120 volts or 200-240 volts.
AC, 50-60Hz.

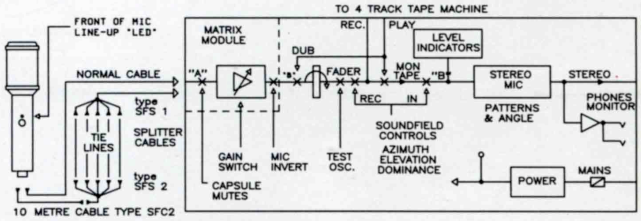
Specification :

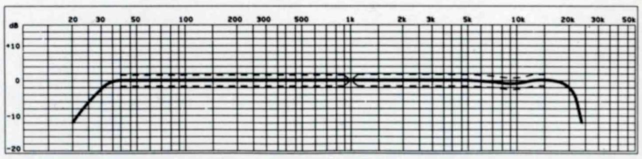
Microphone front sensitivity at 0dB gain, fader 0	115dB SPL
Microphone acoustic line up at 35dB gain, fader 0	80dB SPL
Maximum sensitivity at 40dB gain, fader +5	70dB SPL
Minimum sensitivity at 0dB gain, fader −15	130dB SPL
Maximum input for less than 0.5% THD	140dB SPL
Frequency range	20-20, 000Hz
Equivalent self-noise, IEC 179	16 to 17dB-A (Cardioid)
Microphone Outputs: A-Format, 48 volt phantom powered, balanced.	
Source impendance	100 ohms approx.
Sensitivity	approx 1mV/microbar
Control Unit Outputs at line-up	0dBu, unbalanced
Maximum output levels	22dBu.
Minimum load:	
B-Format & Stereo Outputs	600 ohms
Monitor/Headphone Outputs	400 ohms per side

Source impedance of all outputs : Approx 33 ohms.

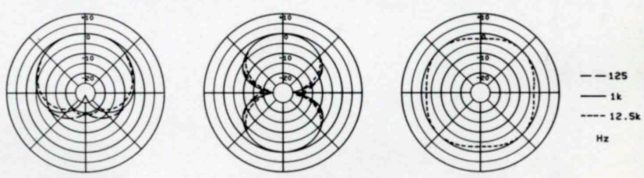
Weights : Microphone : 500 grammes (18oz.)
Control Unit : 3500 grammes (8lbs.)

Dimensions : Microphone : Length : 241mm (9½")
Body dia. : 38.1mm (1½")
Head dia. : 63.5mm (2½")
Control Unit : Length : 425mm (16¾")
Depth : 254 mm (10")
Height : 100mm (4")





FREQUENCY RESPONSE OF THE STEREO OUTPUT SET TO CARDIOID



SOUNDFIELD MICROPHONE STEREO OUTPUTS AT 3 POLAR SETTINGS

This British made precision product is guaranteed against faulty materials and workmanship for a period of 12 months.

CALREC AUDIO LTD

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DESIGNED & PRODUCED BY OEC DESIGN & DISPLAY · BRIGHOUSE · WEST YORKSHIRE

L'exploitation globale du champ acoustique par le "Soundfield Microphone" de Calrec

Encore une réalisation anglaise, loin d'être nouvelle, puisqu'elle en est à son quatrième modèle ("Mark IV"), très prisée Outre-Manche. Elle fut présentée aux services de notre ORTF, sans trop de succès, semble-t-il. La génèse de ce capteur nous ramène à l'époque, où la tétraphonie soulevait beaucoup de controverses ; surtout sous sa forme dite "matricielle", répartissant sur deux canaux (disque stéréophonique) un savant mélange de quatre messages microphoniques, comprenant nécessairement les deux normaux de la stéréophonie, complétés d'informations relatives à l'ambiance du local, où fut effectuée la prise de son. Lors de la restitution, un décodeur (réalisant des combinaisons linéaires des deux canaux disponibles, multipliés par des coefficients bien choisis), sans parvenir à séparer entièrement les composantes du mélange, les orientaient, en priorité, vers l'un des quatre (parfois trois) haut-parleurs obligés. Des sommes considérables furent englouties par quelques firmes, pour assurer la suprématie mondiale à leur méthode "matricielle" (la lutte de CBS et Sansui revêtit parfois un caractère épique). En Angleterre, deux chercheurs, le Professeur Felgett et Michael Gerzon, eu égard à l'impossibilité d'accéder à une vraie tétraphonie aux 4 canaux séparés ; jugèrent plus important d'en suggérer la plus satisfaisante illusion, étudièrent le problème sous l'angle psycho-acoustique pour définir un compromis optimal baptisé "Ambisonics". Les premières prises de son expérimentales, selon la méthode "Ambisonics", furent réalisées avec quatre microphones séparés, orientés convenablement, aux diaphragmes rapprochés. De là, naquit l'idée d'un microphone à quatre diaphragmes quasi-coïncidents, complété d'un module de traitement électrique des quatre messages, pour les combiner au mieux. "Calrec", dont les microphones électrostatiques jouissaient d'une excellente réputation, accepta de tenter l'épreuve, et c'est ainsi que vit le jour ce "Soundfield Microphone" explorateur du champ acoustique, aux quatre capsules électrostatiques identiques légèrement directionnelles (sans atteindre la cardioïde), disposées (fig. 5) selon quatre faces d'un tétraède, en deux groupes de deux rectangulaires (les réponses sont électriquement compensées pour annuler leurs séparations dans l'espace). On dispose ainsi d'un moyen d'échantillonner et caractériser des ondes sonores émanant de toutes les directions possibles.

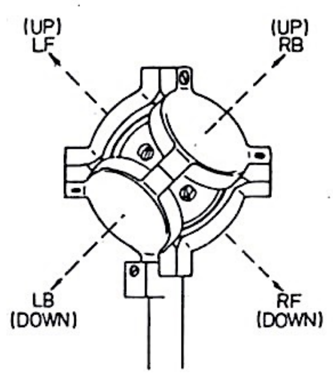

Fig. 5 : Idée de la disposition dans l'espace des quatre diaphragmes du microphone "Soundfield" de Calrec (il est assez difficile de se représenter leur situation ; un modèle en carton du tétraèdre support aide beaucoup). Les 4 diaphragmes sont repérés comme suit : RP (UP) soit secteur droit arrière, vers le haut ; RF (down) secteur droit antérieur, vers le bas ; LF (UP) secteur gauche antérieur, vers le haut ; LB (down) secteur gauche arrière, vers le bas. Ces quatre échantillons du champ acoustique, pratiquement prélevés en un même point (après correction), convenablement dosés et combinés par l'intermédiaire du module de liaison obligatoire, donnent énormément de liberté au preneur de sons stéréophoniques. Ne pas oublier que ce microphone Calrec fut initialement conçu pour le procédé "Ambisonics", de tri ou tétraphonie britannique, conçu par l'équipe Felgett-Gerzon. L'aspect extérieur est très classique, même un peu vieillot.

Pour diverses raisons, la tétraphonie échoua, et il en fut de même du procédé "Ambisonics" (il est toujours question de sa résurrection). Ce qui importe pour nous, c'est que le microphone conçu à son intention demeure et fut même perfectionné ; car s'il paraît improbable de l'utiliser, dans un proche avenir, pour une multiphonie, il peut assurer des prises de son stéréophoniques à deux canaux, avec des possibilités de traitement des signaux sonores (captés idéalement en un seul point) sans équivalent ; grâce au dosage électronique et à la répartition (par le module associé) des tensions des quatre capteurs. Un opérateur compétent peut, pratiquement, tout faire : obtenir l'équivalent de deux microphones orientables, aussi bien verticalement qu'horizontalement, et d'angle ajustable ; faire varier la distance apparente d'une source sonore (effet de zoom acoustique), etc...

Eu égard à ses très riches possibilités le constructeur présente ce microphone composite comme tout spécialement adapté à l'ère "numérique", (très faible bruit résiduel), il peut aussi encaisser 140 dB de pression acoustique, à moins de 0,5% de distorsion. Bien entendu, si le procédé "Ambisonics" revenait au premier plan, il serait tout prêt à le servir.

Rémy Lafaurie
1984 Sono

CALREC

Sound Field System

calrec by ams

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E-Mail: 42:IMC 889
(206) 633-1956 • Telex: 4900001180 (CALUI)

CALREC SOUNDFIELD MICROPHONE MARK 4.

CIRCUITS DESCRIPTION, TEST AND ALIGNMENT.

CONTENTS.

1. POWER AND INDICATORS.
2. ALIGNMENT OSCILLATOR.
3. METERING.
4. RECORD AMPLIFIERS.
5. REPLAY AMPLIFIERS.
6. SOUNDFIELD CONTROLS.
7. STEREO OUTPUTS.
8. MATRIX MODULE.
9. MICROPHONE AMPLIFIERS.
10. ACOUSTIC TESTS.

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 SK4050-59
CABLE CONNECTION VIA STUDIO LINES : 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.

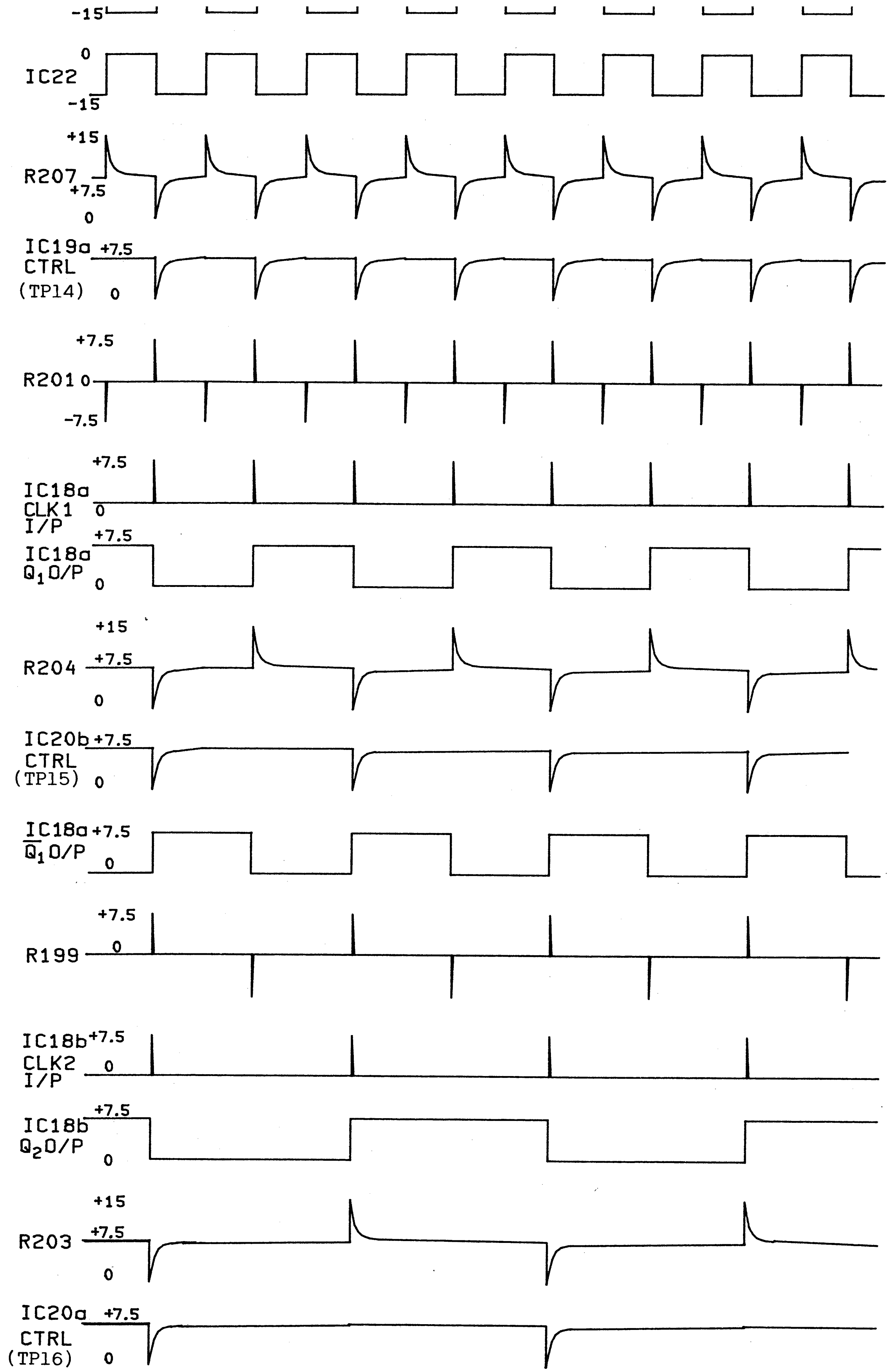


FIG 4: ALIGNMENT OSCILLATOR SEQUENCES

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: 0, 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 OdB 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.

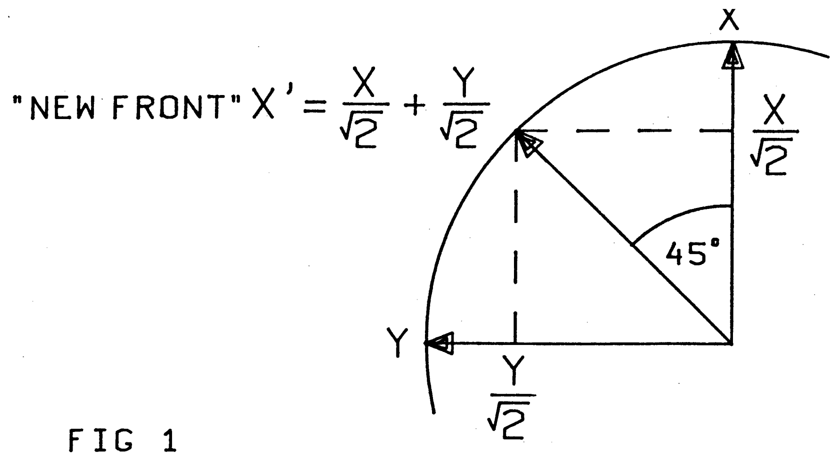


FIG 1

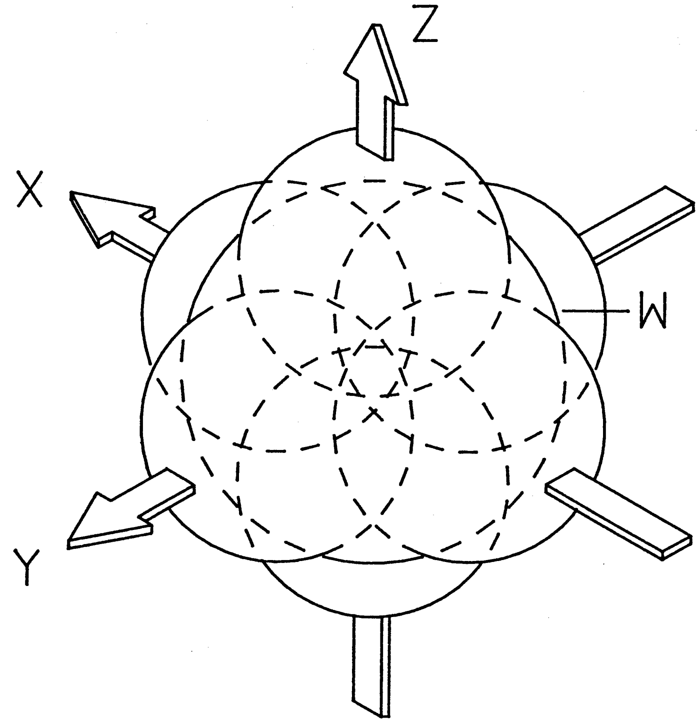


FIG 2

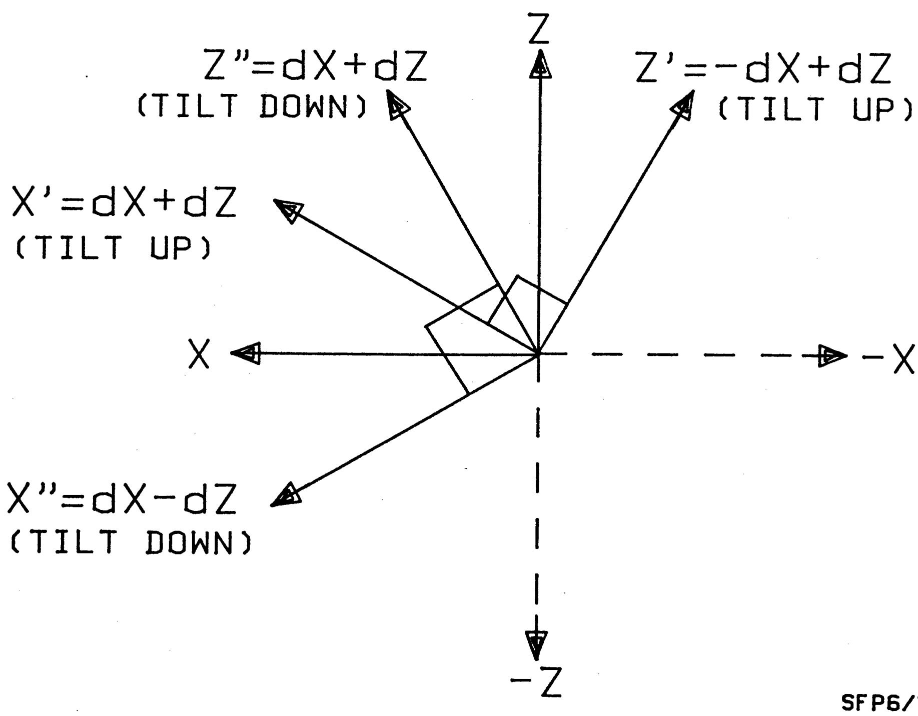


FIG 3

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 not selected to TAPE. Press the SOUNDFIELD IN and REC buttons.

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

Set Azimuth, Elevation and Dominance initially to neutral.

AZIMUTH.

Ensure there are no 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
$$X_2 + Y_2 = \frac{X}{2} + \frac{Y}{2}$$

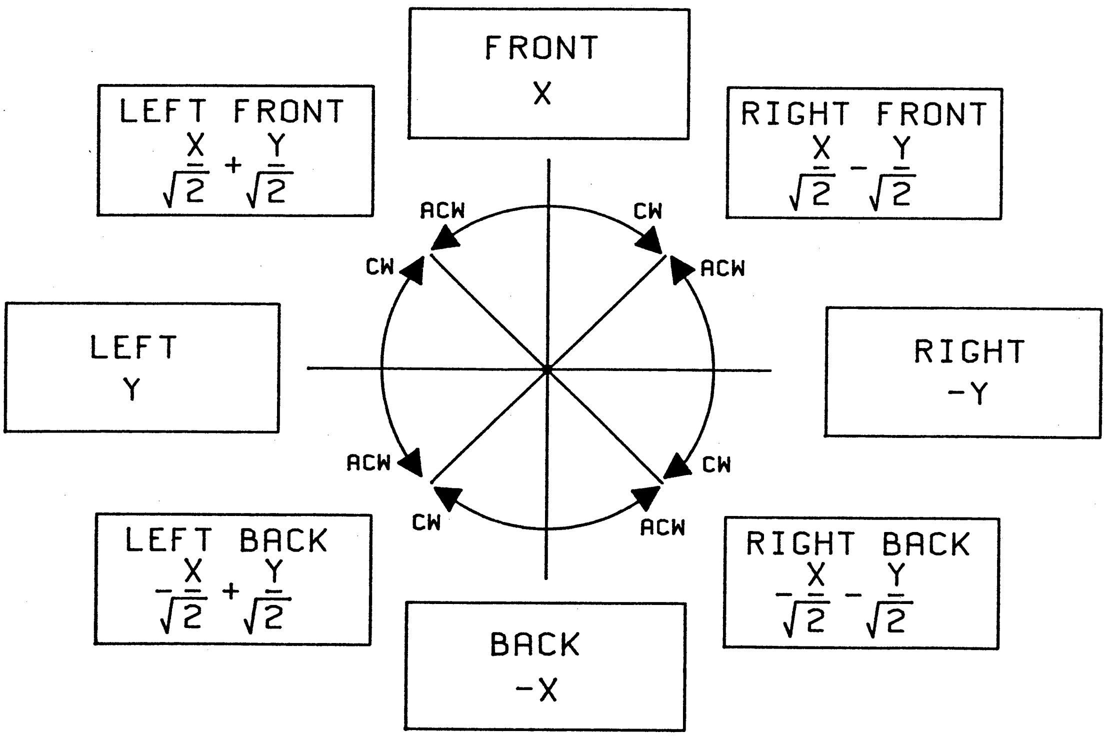
RIGHT
$$X_2 + Y_2 = -\frac{X}{2} - \frac{Y}{2}$$

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).

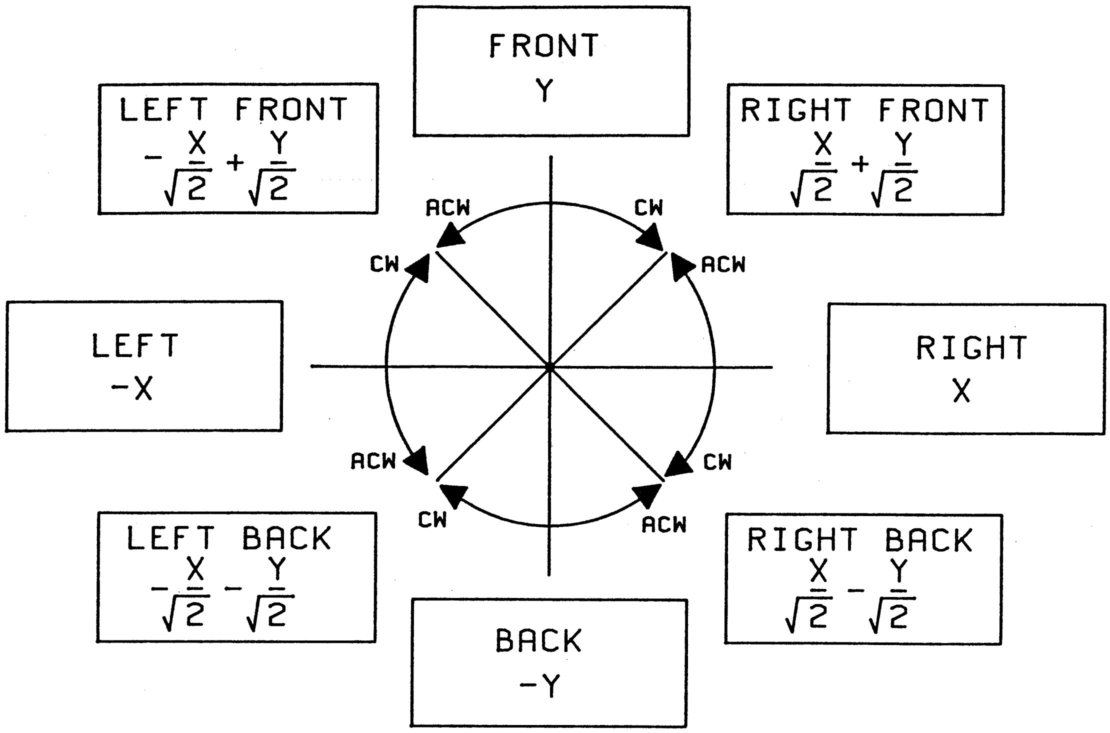


X OUTPUT FROM AZIMUTH.

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, (Test point 28)

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

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

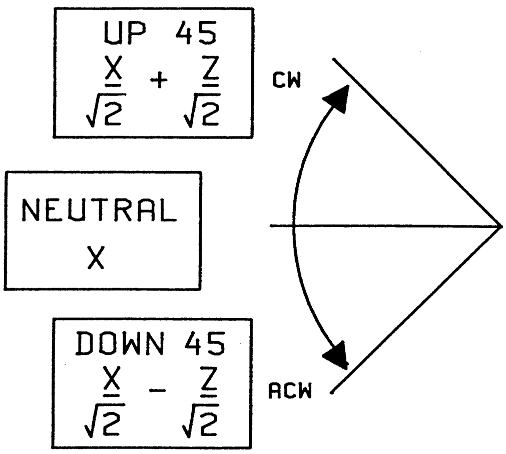
$$X4 + Z2 = \frac{X}{\sqrt{2}} - \frac{Z}{\sqrt{2}}$$

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

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

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

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



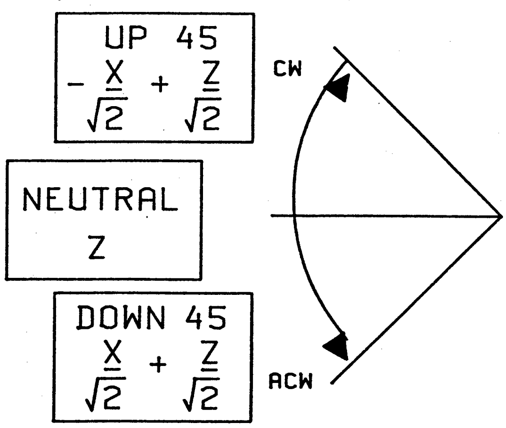
X OUTPUT FROM ELEVATION (& SOUNDFIELD IF DOMINANCE IS AT NEUTRAL AND NO FAULTS.

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

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

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

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).

$$A12a\ (54) = W_1 + X6\ \text{(Test point 31)}$$
$$= (2 - \sqrt{2})W - (\sqrt{2} - 1)X$$

W	−4.65dB IN-PHASE
X	−7.65dB INVERTED

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, $W2 + X7 = -(W_1 + X6)$
$$= -(2 - \sqrt{2})W + (\sqrt{2} - 1)X$$

W	−4.65dB INVERTED
X	−7.65dB IN-PHASE

AT ACW, BACK DOMINANCE, $W2 + X7 = -(1 + \sqrt{2})\ (W_1 + X6)$
$$= -\sqrt{2}W + X$$

W	+3dB, INVERTED
X	0dB, IN-PHASE

AT CW, FRONT DOMINANCE, $W2 + X7 = -\dfrac{1}{(1 + \sqrt{2})}\ (W_1 + X6)$
$$= -\left(\frac{2 - \sqrt{2}}{1 + \sqrt{2}}\right)W + \left(\frac{\sqrt{2} - 1}{1 + \sqrt{2}}\right)X$$

W	−12.3dB, INVERTED
X	−15.3dB, IN-PHASE

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

W	−4.65dB INVERTED
X	−7.65dB INVERTED

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

W	−12.3dB INVERTED
X	−15.3dB INVERTED

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

W	+3dB INVERTED
X	0dB INVERTED

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.

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

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

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

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

W	+3dB, IN-PHASE
X	−3dB, INVERTED

i.e. W becomes a BACK FACING SUB-CARDIOID.

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

W	+3dB, IN-PHASE
X	−3dB, IN-PHASE

i.e. W becomes a FRONT FACING 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, $X9 = \frac{-W}{\sqrt{2}} + \sqrt{2}X$

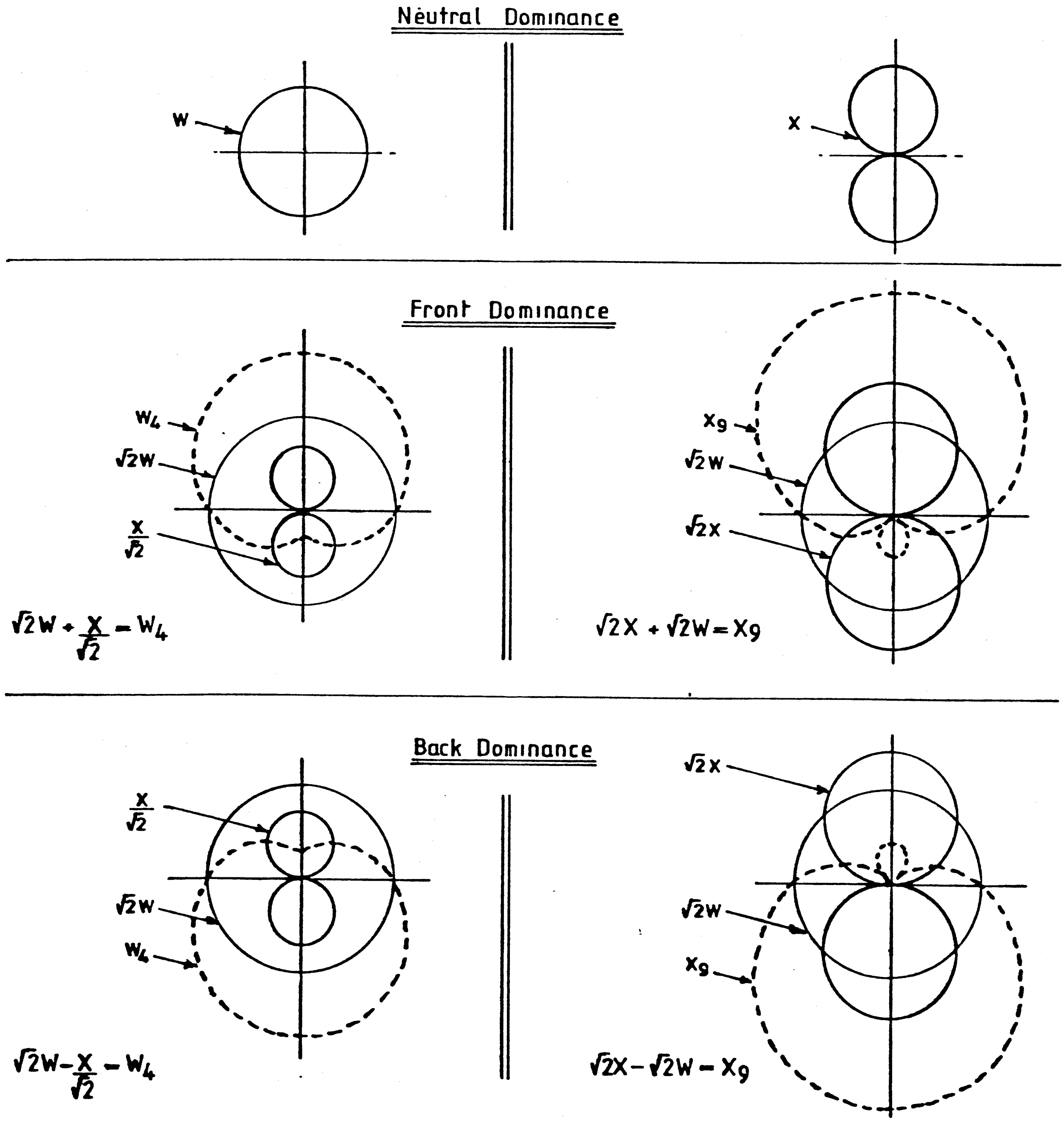
W	-3dB, INVERTED
X	+3dB, IN-PHASE

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

AT CW, FRONT DOMINANCE, $X9 = \frac{W}{\sqrt{2}} + \sqrt{2}X$

W	-3dB, IN-PHASE
X	+3dB, IN-PHASE

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



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 +1OdB 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, A1Oa, A1Ob 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

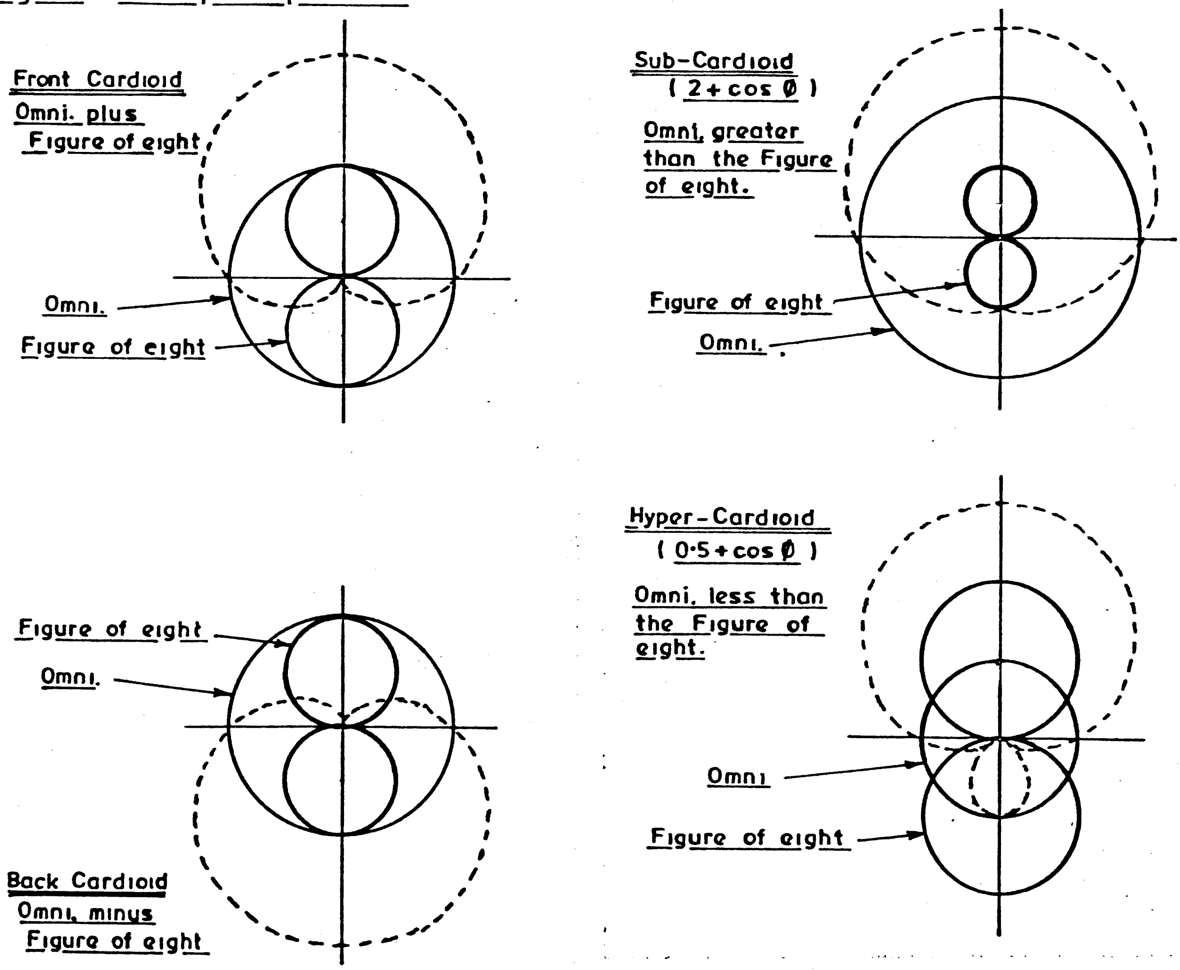
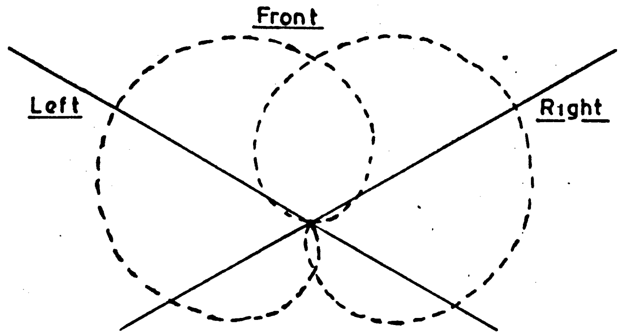
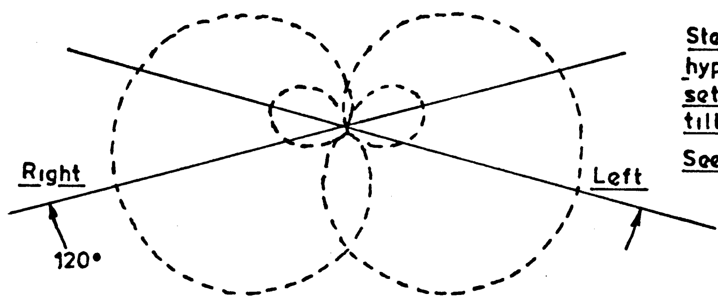


Fig. 5b. Two of many Stereo configurations from the Soundfield microphone.



Stereo pair of synthesised cardioid microphones.



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

Seen from front.

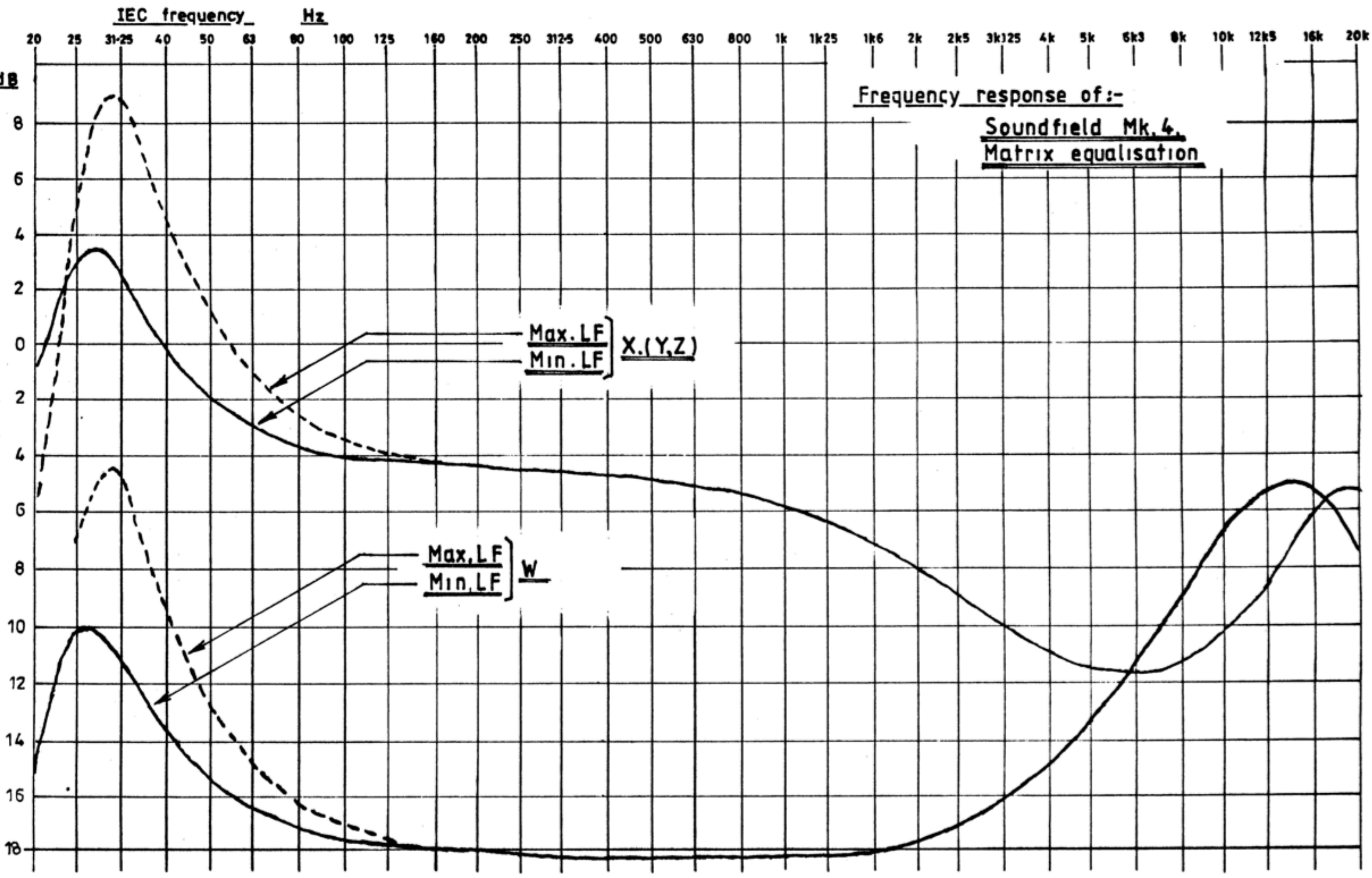

IEC frequency Hz
20 25 31·25 40 50 63 80 100 125 160 200 250 312·5 400 500 630 800 1k 1k25 1k6 2k 2k5 3k125 4k 5k 6k3 8k 10k 12k5 16k 20k
dB
8
6
4
2
0
2
4
6
8
10
12
14
16
18
Frequency response of:-
Soundfield Mk. 4.
Matrix equalisation
Max. LF
Min. LF
X. (Y, Z)
Max. LF
Min. LF
W

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 ($\pm$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 <u>all</u> 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.

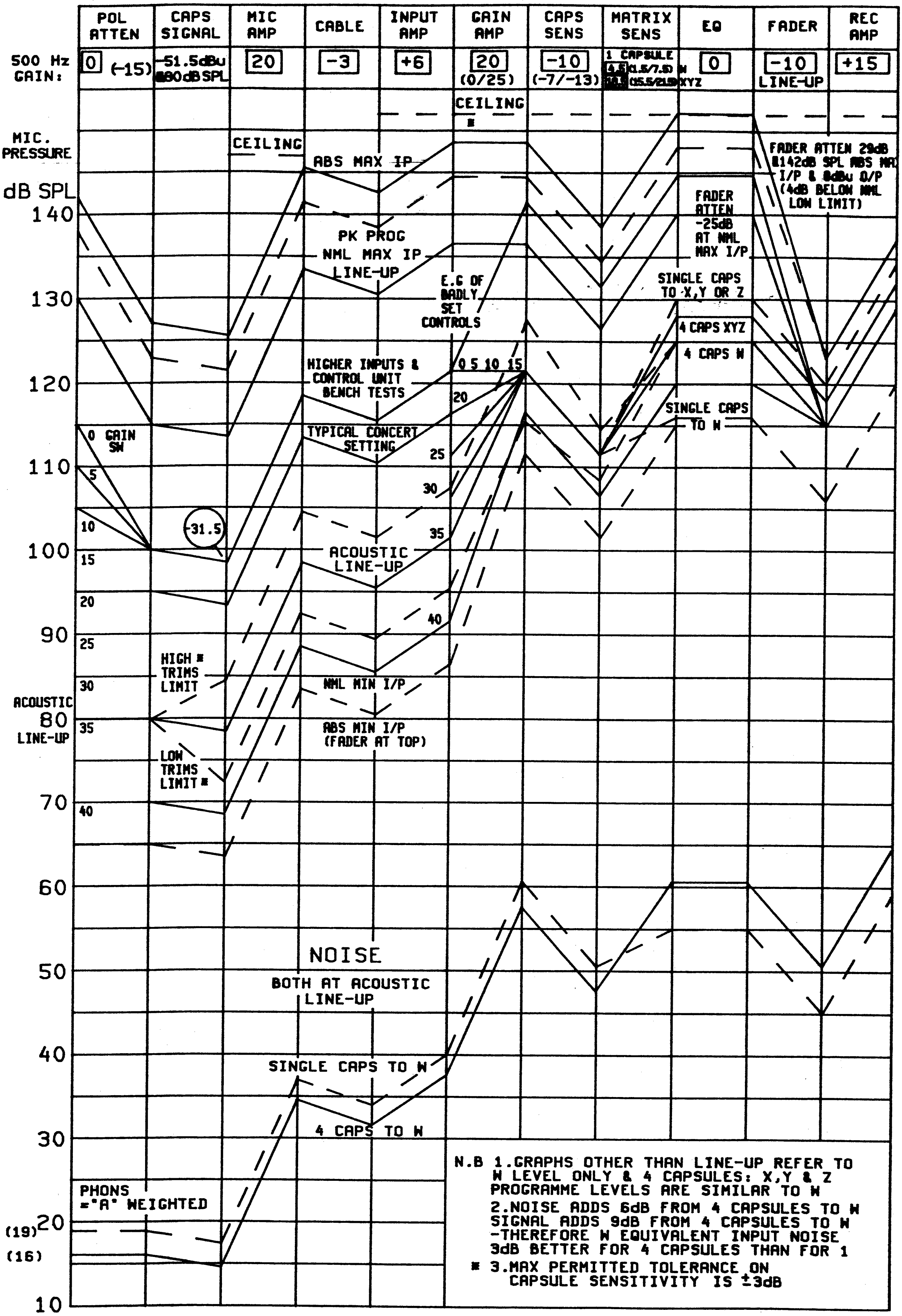
Finally check the noise of all 4 channels together to the W output at -64dBA approx.

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 0dBu. 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 0dBu (-2dBV) max. levels.

Similarly LB and RB should give a figure-of-eight on axis LB-RF, with 0dBu (-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:-

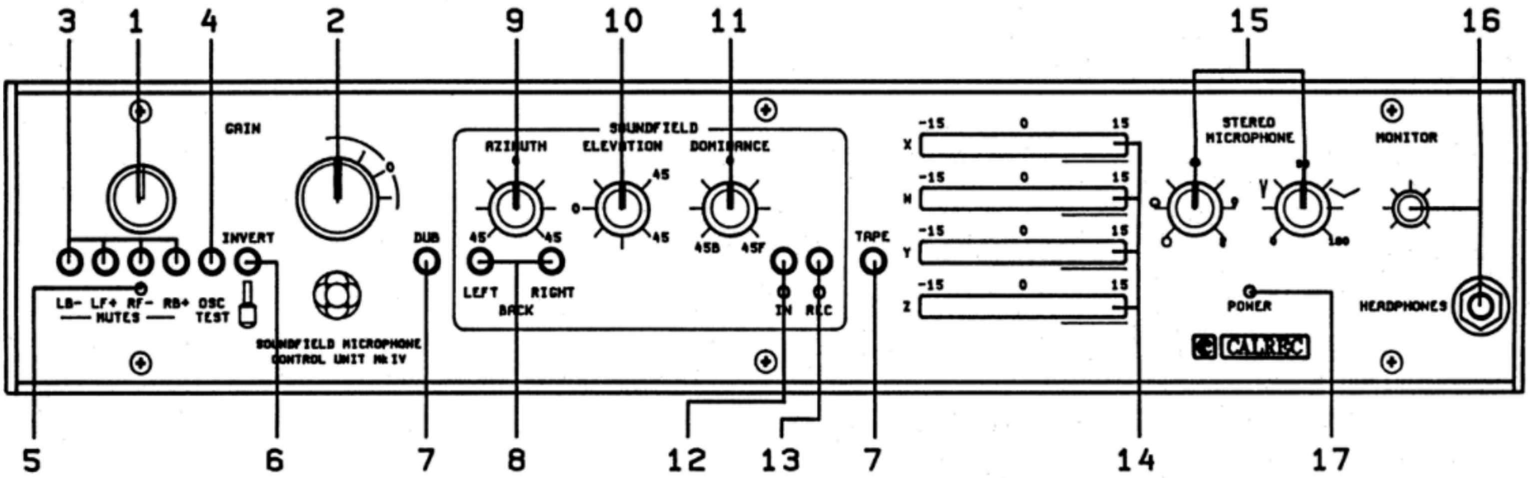
W	:	18 phons (dB, A weighted)
X	:	22 phons
Mono Cardioid	:	16 phons*
(Tolerance	:	$\pm$1dB)

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.

SOUNDFIELD MICROPHONE CONTROL UNIT MK4

B-Format Record : XLR 5-M
B-Format Replay : XLR 5-F
PIN 1 : EARTH
PIN 2 : X : Front/Back
PIN 3 : W : Omni
PIN 4 : Y : Left/Righ
PIN 5 : Z : Up/Down


3
1
4
2
9
10
11
15
16
GAIN
SOUNDFIELD
AZIMUTH
ELEVATION
DOMINANCE
STEREO MICROPHONE
MONITOR
INVERT
DUB
TAPE
LB- LF+ RF- RB+ OSC
MUTES
TEST
LEFT
RIGHT
BACK
IN
REC
X
W
Y
Z
-15
0
15
POWER
HEADPHONES
SOUNDFIELD MICROPHONE CONTROL UNIT Mk IV
CALREC
5
6
7
8
12
13
7
14
17

INTRODUCTION

The Calrec Soundfield Microphone System MK IV is a unique product offering a hitherto unobtainable degree of accuracy in the generation of coincident stereo and mono microphone patterns. The user is able to steer and move the generated microphones both in real and post production time. A fully three dimensional output signal suitable for encoding to any surround system and in particular to the UHJ family of Ambisonic surround sound systems is available.

The System comprises:-

1. The MK IV Soundfield Microphone in presentation box

2. The MK IV Soundfield Control Unit

3. 100 metres of connecting cable on a drum (SFC 1)

4. Mains cord

5. A flight case to house items 1, 2 & 4

OPTIONAL ACCESSORIES

Microphone Mounting Bar. AKG HD-30
Anti-vibration mount.
10 metre Microphone Head lead/Extension cable (SFC 2)
Splitter cables type SFS 1/SFS 2 allow the microphone to control unit connection to be made via 5 studio XLR-3 balanced tie lines thus placing the control unit in the listening room (both cables are required).
5 pin XLR female to 4x XLR-3 male B-Format record output leads (SFS 3).
5 pin XLR male to 4x XLR-3 female B-Format replay input leads (SFS 4).
(It is immaterial whether or not the tie lines carry studio 48 volt phantom power).

A foam windshield and a Rycote total Windgag are currently in development and will be available shortly.

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

THE CONTROLS

1. **Input gain** - controls the sensitivity of the microphone over a 40dB range in 5dB steps.

2. **Main Fader** - should normally be operated as near to the zero mark as possible but allows a +5dB to -15dB fine trim. Below -15dB is for fade only. The +10dB position may be used for maximum sensitivity in extreme cases.

These two controls should be used in exactly the same way as the microphone input stage and fader of a conventional audio mixing console.

3. **Mutes - (LB- LF+ RF- RB+)** Interrupt the head lead circuit from the microphone capsule/head amplifier to the control unit and should only be used for continuity checks. All four circuits should be muted and then each one released in turn to establish the presence of a signal. UNDER NO CIRCUMSTANCES WHATSOEVER SHOULD RECORDING TAKE PLACE WITH ANY MUTE BUTTON DEPRESSED.
   The main use of the mutes is for checking continuity when the head lead has been split-out into existing tie-lines using the SFS 1 and SFS 2 splitter leads. It is good Soundfield practice, though, to check before every recording, as head leads do get damaged, and, while under normal circumstances it would be obvious from the monitored output that all was not well, under difficult monitoring conditions on headphones a discontinuity could go un-noticed.

4. **Osc. test** - the test oscillator produces a 0dBm/1kHz tone at the 'B format' record outputs which should be used to align the multi-track tape machine record circuits. The tone is coded for later identification of the recorded tracks and the following track plan is recommended as a standard for professional use.

Tape Track	B-Format Signal	Oscillator Coding approx. Secs On	Off
1	X	1	1
2	W	Continuous	-
3	Y	3	1
4	Z	7	1
The bargraph meters are arranged in this order thus making direct/off tape comparison very easy.			

The Y tone sequence is twice as long as the X sequence and the Z is twice as long as the Y. IT IS IMPERATIVE THAT

TEST TONES BE RECORDED AT THE BEGINNING OF EACH SESSION and preferably on each reel of tape. If a noise reduction system is used, the Soundfield test tones should be after those of the noise reduction system. The recorded tones may then be used to align the tape replay section and to check for correct connections of tracks.
The absolute level of the replayed test tone is not important but it is vital to adjust each track to the same relative level.

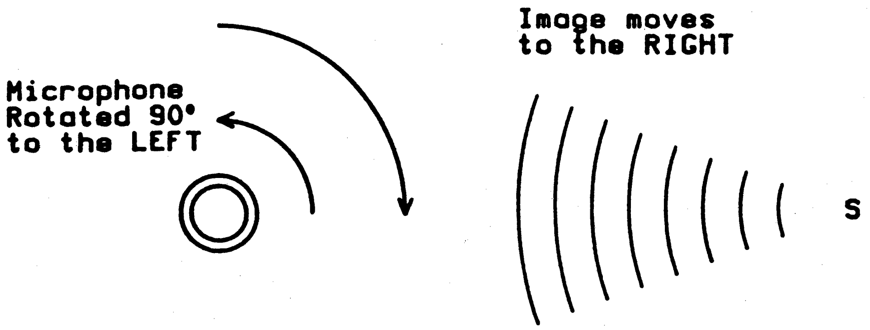
5. **Red LED** - Flashes when any of the mute buttons are depressed and continuous when the oscillator is in operation. UNDER NO CIRCUMSTANCES SHOULD RECORDING TAKE PLACE WITH THIS LED ILLUMINATED.

6. **Invert** - when the microphone is suspended upside-down the left-right Y signal and the up-down Z signal are reversed and must be corrected, otherwise the operation of azimuth and elevation would be reversed.

7. **Tape and Dub** - although physically separated, these two buttons are interactive in use. The normal condition is for the B-format output of the microphone input stage to be routed direct to the stereo output via the soundfield controls if selected. "Tape" substitutes the microphone signal with the off tape input to the B-format replay socket and would normally be used for 'tape check' during recording. 'Dub' disconnects the microphone completely and routes the B-format replay input to the main fader and thence to the B-format record output and the stereo output. This action reverses the roles of the B-format inputs and outputs AND SHOULD NEVER BE USED DURING RECORDING. The facility allows B-format to B-format and/or B-format to stereo dubbing with gain control and "Soundfield" control if required. The polar pattern and capsule angle controls are always available in the stereo output.

8. **Left/Right/Back buttons** - with neither buttons selected the normal front of the microphone is indicated by an LED within the body of the mic shining through a hole in the case just above the "Calrec" logo. This is the centre front of any generated stereo configurations and the centre front stage of the Ambisonic surround sound signal.

   Selection of either Left or Right moves the electrical front through 90° in the appropriate direction and selection of both (Back) moves the electrical front through 180°. It should be noted that this has exactly the same effect as physically moving the microphone and the sound images will, therefore, move in the opposite direction.



Note that when the Microphone is rotated to the LEFT, the Image previously at the FRONT moves to the RIGHT of the Stereo (or Surround) picture, and Visa-Versa.

When working in stereo, using directional patterns, the sound source would appear "off mic" but in surround sound the image would simply move to the new location.

9. **Azimuth** - allows up to plus or minus 45° of continuous variation on the direction of point selected on the Left/Right/Back buttons. The same rules of steering apply as in the previous section.

10. **Elevation** - allows plus or minus 45° continuous variation on the vertical alignment of the actual microphone.

11. **Dominance** - in effect, this is a form of zoom control allowing the generated microphone to be moved closer to or farther from the original sound source than the actual microphone. The effect in stereo is not exactly the same as moving the microphone because the stereo image does not widen as would be expected, in fact it narrows slightly, but this effect can be corrected with the 'Capsule angle' control. In surround sound the apparent movement is even more realistic.

12. **Soundfield "In" Button** - routes the B-format signal through the 'Soundfield' control section. If no soundfield correction is to be made the section should be switched out of circuit to avoid accidental adjustment.

13. **Soundfield 'Rec' Button** - In normal or "Dub" operation the "Soundfield" controls are inserted into the B-format signal after the B-format Record outputs and only affect the stereo output. "Rec" allows their insertion into the B-format record outputs, thereby enabling corrections to be made onto 4 track tape as well as the stereo output.

14. **Metering** - The four, bargraph LED meters show the signal levels of the four components of the B-format signal, X, W, Y and Z as they appear either at the B-format output or off tape at the B-format input. In either case they show the effect on signal level of any Soundfield adjustments AND DIRECTLY REPRESENT THE SIGNAL LEVEL AT THE B FORMAT OUTPUTS. When used directly as a stereo microphone or on subsequent mixdown of a B format signal to stereo they only show the energy being fed to the stereo output circuits and do not show the energy level of the stereo output. Nevertheless they should not be allowed to peak into the Red. The stereo output of the microphone should be adjusted, using the 'Dub' facility and the internal oscillator to give the desired line-up level AFTER the correct tonal balance has been achieved with the "Soundfield" and "Stereo Microphone" controls.

15. **'Stereo Microphone' controls** - the polar pattern control is graduated from omni-directional (Ø) at the anti-clockwise end through cardioid at "12 o'clock" to figure-of-eight at the clockwise end and smoothly adjusts the polar pattern of the generated microphone(s) through all the intermediate sub-cardioid and hyper-cardioid positions. The capsule angle control is graduated from Ø degrees to 18Ø degrees and smoothly adjusts the angle of the generated microphones between the two extremes. With the control set to zero the two outputs would be of microphones pointing in exactly the same direction from exactly the same point in space and would therefore be identical mono signals.

16. **Monitor** - controls the signal level of the stereo output to the two headphone sockets. (One on the front panel one on the rear panel).

17. **Power** - indicates that the control unit is connected to a mains power source and that the DC supply from the power unit is present on the circuit board.

CHECKS AND ALIGNMENTS

Because the Soundfield System produces near perfect figure-of-eight patterns it is possible to substantially check the microphone using speech tests only, preferably using headphones.

Set the microphone on a stand or boom at a comfortable height to walk around but low enough to get your head over the top. Set the controls as follows:

1. Mutes — normal (off)

2. Osc — off

3. Invert — to suit microphone position

4. Gain — to give a comfortable listening level

5. <u>Azimuth</u> "O"

6. <u>Elevation</u> "O"

7. <u>Dominance</u> "O"

8. <u>Left/Right/Back</u> de-selected (forwards)

9. <u>In</u> de-selected

10. <u>Rec</u> de-selected

11. <u>Tape</u> de-selected

12. <u>Stereo Mic Pattern</u> figure-of-eight

13. <u>Stereo Mic Angle</u> 0^{o}

First mute all the capsules and release each one in turn to check the continuity of the head lead. If a speech check is conducted at this point it should be possible to ascertain the direction of each individual capsule in the horizontal and, if the test room is not too reverberant, the vertical.

LB-	=	Left back down
LF+	=	Left front up
RF-	=	Right front down
RB+	=	Right back up

Left and right are defined as being the left and right hand side of the stereo picture ie. they will be reversed if you face the front of the mic.

This test is optional provided that a roughly equal signal is received from each capsule. (Bearing in mind that two face forwards and two face backwards and that two face upwards and two face downwards).

De-select all the mute buttons and select the Soundfield <u>IN</u> button. Reference to the control setting list will confirm that the unit is now set to produce a forwards facing mono figure-of-eight.

<u>TEST 1</u>

To speech check, start at the front of the microphone at about 150mm (6 inches) range and progress clockwise around horizontally, noting the signal peaks at 0^{o} and 180^{o} and the nulls at 90^{o} and 270^{o}. Similar nulls should also be found immediately over the top and underneath the microphone. (See fig 1)

TEST 2

Aim the microphone 90° to the right by selecting the "Right" button and repeat the tests. (See fig 2). The signal peaks and null will have moved round 90° relative to the LED and Badge.

TEST 3

Repeat the test with "Left" selected and results should be as TEST 2.

TEST 4

Repeat the test with "BACK" and results should be as TEST 1. (See fig 1).

TEST 5

Return the controls to forward facing, as in TEST 1 and select 45° clockwise on the "AZIMUTH" control. The peaks and nulls will now have moved 45° to the right of mic front (ie. halfway between figures 1 and 2).

TEST 6

Repeat test 5 with the "AZIMUTH" control selected to 45° anticlockwise. The peaks and nulls will now be 45° to the left of mic front. (Again, halfway between figures 1 and 2, but at 90° to test 5).

TEST 7

Repeat test 5 with the Left/Right/Back buttons selected to "LEFT". The peaks and nulls will be in the same position as test 6 (45° clockwise from the left hand side of the mic is the same position as 45° anticlockwise from the front).

TEST 8

Repeat test 6 with the Left/Right/Back buttons selected to "RIGHT". The peaks and nulls will be in the same position as test 5 (45° anticlockwise from the right hand side of the mic is equal and opposite to 45° clockwise from the front).

TEST 9

Return the AZIMUTH to 0° and turn ELEVATION to +45°. The peaks should now be at +45° relative to microphone FRONT and -45° relative to microphone BACK as in figure 3.

TEST 10

Repeat TEST 9 at -45° and note that the peaks move to this position relative to microphone FRONT and +45° relative to microphone BACK as in figure 4.

TEST 11

Return the ELEVATION control to Ø° and whilst speaking at microphone FRONT note that increasing DOMINANCE increases microphone sensitivity and reducing DOMINANCE reduces it.

TEST 12

Return the DOMINANCE control to Ø°. Exercise the POLAR PATTERN control and observe that the microphone works as a single cardioid with a null at 18Ø° (microphone BACK) in CARDIOID position and all around equally in OMNI position.

TEST 13

With the microphone set as a CARDIOID check that the DOMINANCE has a similar effect as TEST 11. Return the DOMINANCE control to Ø° and de-select "SF IN".

TEST 14

Set the STEREO controls to CARDIOID and 9Ø° and observe that the LEFT signal peaks when speaking at 45° MICROPHONE LEFT (to your right) and the RIGHT one at 45° MICROPHONE RIGHT (to your left)' with corresponding opposite nulls.

CONCLUSIONS

If all these tests have produced the results described, then the system can be assumed to be working correctly. There are no servicable parts within the soundfield System. Maintenance should always be carried out by a qualified service engineer in possession of a current service manual.

IMPORTANT

The microphone is factory adjusted and aligned to the control unit by the A to B Module which is housed in the rear of the control unit and contains the microphone input socket.

The microphone serial number is engraved on the panel of the A to B module to indicate that the microphone and A to B should always be kept together.

If the microphone is changed for any reason, this module should be withdrawn by removing the four fixing screws, inserting the mic cable plug and pulling gently. The A to B connects to the control unit by means of a gold plated edge connector.

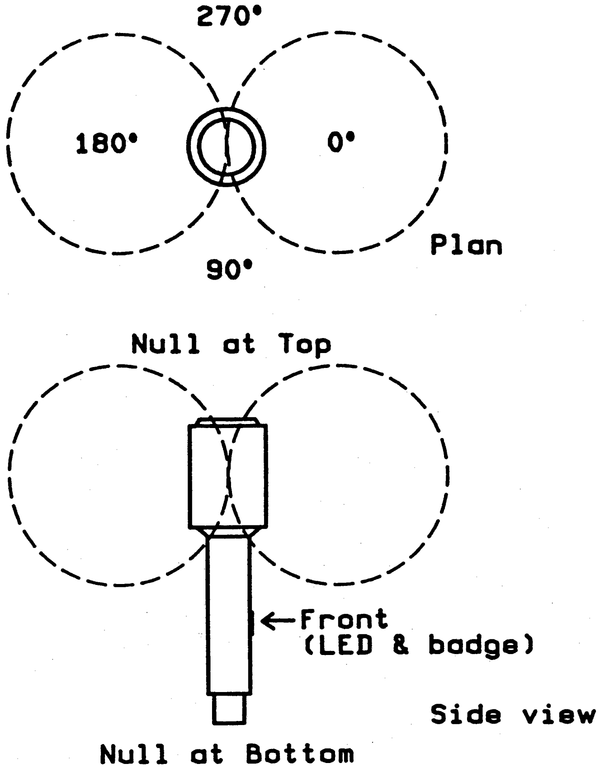


Fig. 1: Figure of Eight: Angle, Azimuth, Elevation: 0°,
Azimuth: FRONT.

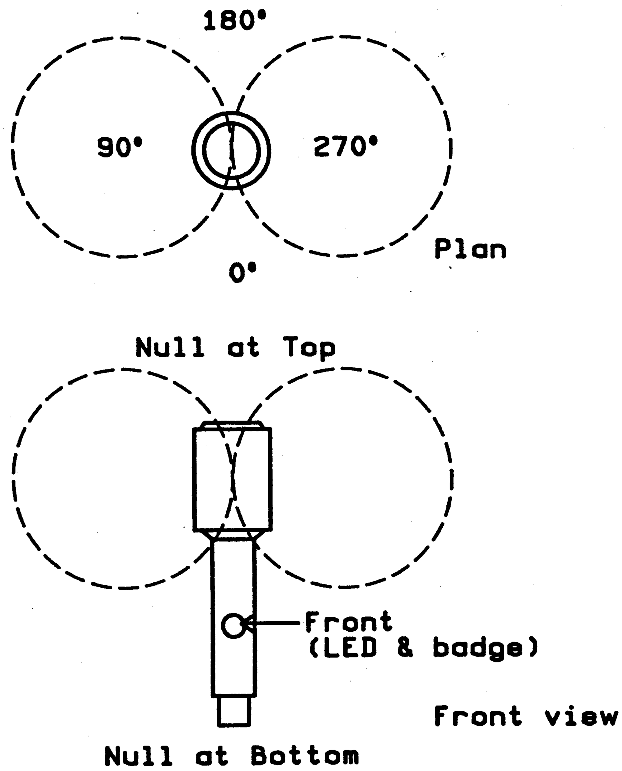


Fig. 2: Figure of Eight: Angle, Azimuth, Elevation: 0°,
Azimuth: LEFT or RIGHT.

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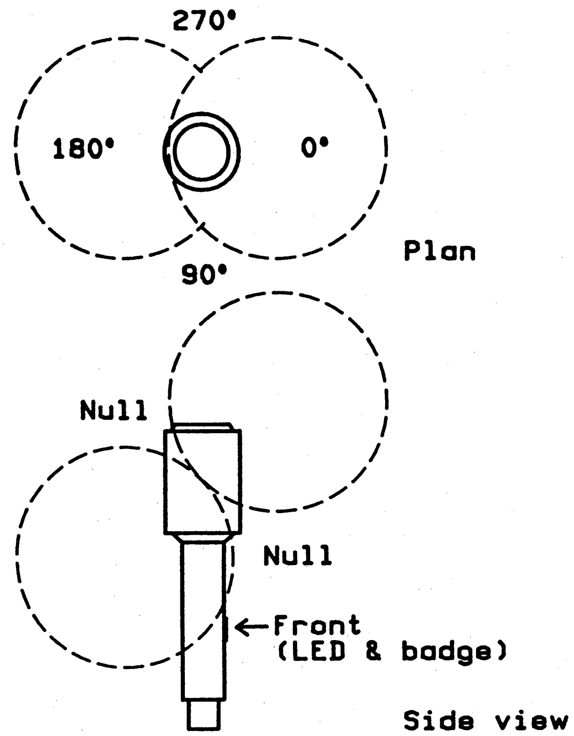

Fig. 3: Figure of Eight: Angle, Azimuth, Elevation: 0°,
Azimuth: FRONT,
Elevation: +45°.

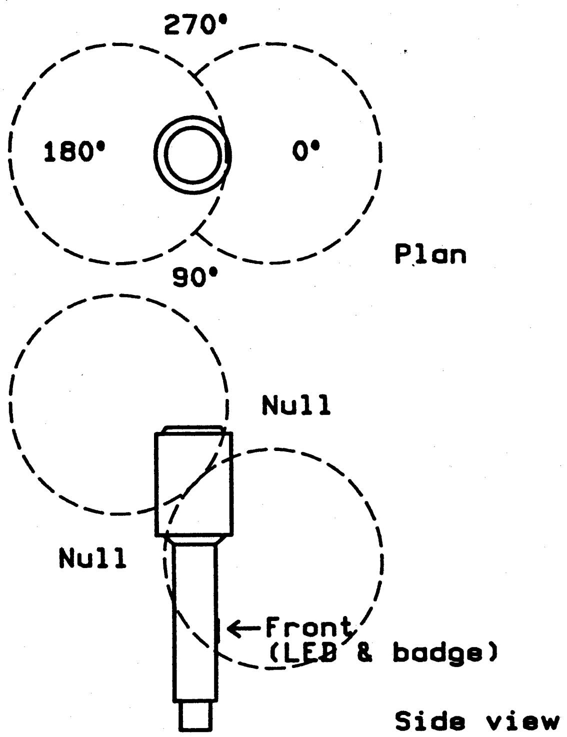

Fig. 4: Figure of Eight: Angle, Azimuth, Elevation: 0°,
Azimuth: FRONT,
Elevation: -45°.