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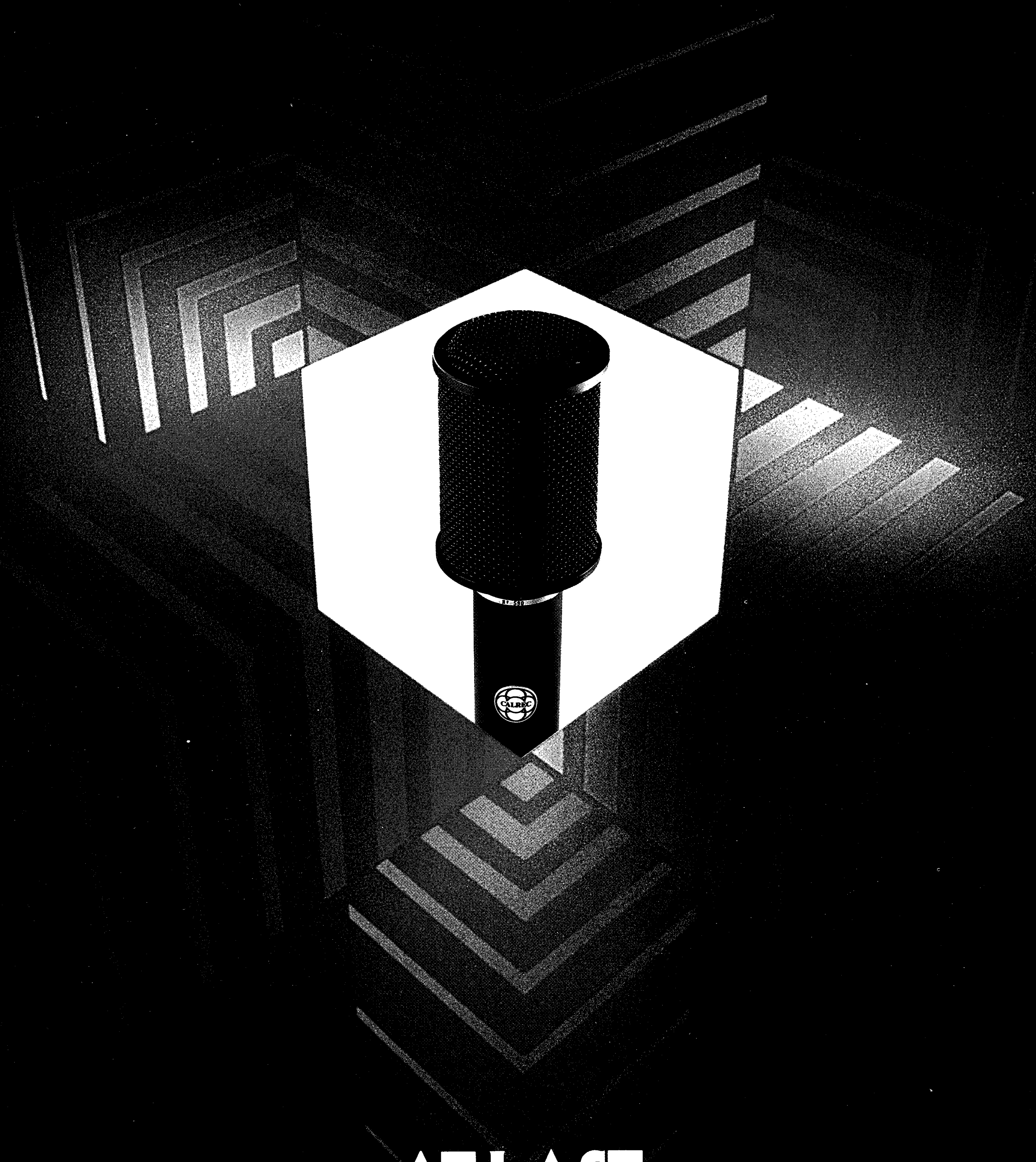


AT LAST SOUND BREAKS THE MICROPHONE BARRIER

CALREC

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 notice 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 pressue (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 transducers were situated.

One centrally important point not yet explained is that the two previous functions may be 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 (1) left-right, (2) front-back, (3) up- down. (see fig. 3).

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 a single diaphragm. In any such system 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 an axis and 180 degrees off axis sounds. At all other positions the frequency reponse performance is sub-optimal and must always be so.

In the B-format signals of the Soundfield Microphone the pressure and gradient components of an apparent single diaphragm are presented as separate electrical signals whose performance may be optimised independently.

The normalised B-format signals may be stored on tape or combined immediately into two apparent microphone signals of any characteristic between omni and figure-of-eight, the angle between the two may be varied from 0 degrees to 180 degrees and the pair may be pointed in any direction both horizontally and vertically and at all times the two signals will remain truely concident.

A significant advantage of handling signals in B Format is that simple circuits allow the microphone to be "steered"; that is rotated, tilted and (apparently) moved backwards and forwards. Moreover these Soundfield steering controls may be exercised together with the stereo controls DURING POST SESSION DUBBING.

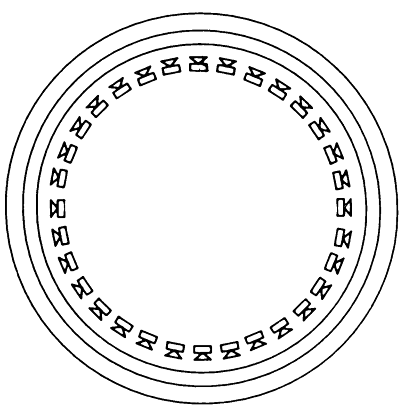


FIG 2 SOUND WAVES CONTINUE TO RADIATE EXACTLY AS BEFORE

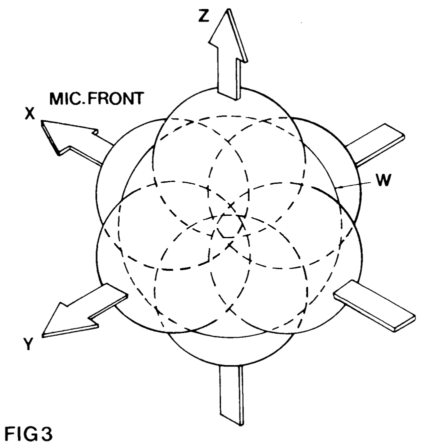


FIG 3

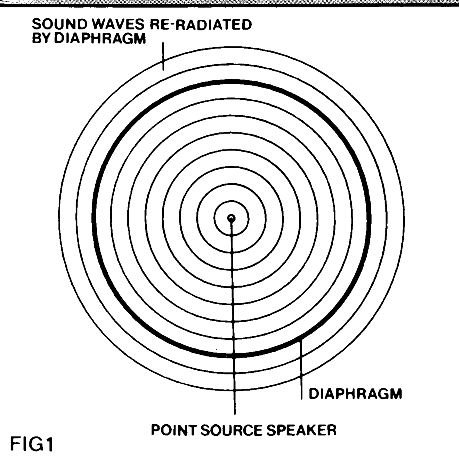
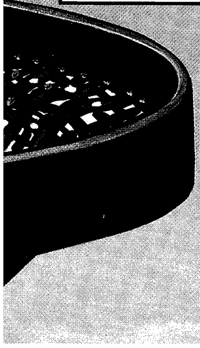


FIG 1

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.
Mains cord.

Optional Accessories : Microphone Mounting Bar.
Anti-vibration mount.
Extension connecting cables.
Splitter cables which allow the microphone to control unit connection to be made via studio XLR-3 balanced tie lines thus placing the control unit in the listening room (2 required, 1 male, 1 female).

(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 OdBu 1KHz signal at the four B-Format Recording Output Lines for tape alignment purposes.
LED (RED)	: Flashes when any of the above conditions are activated.
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 $\pm 45°$ from the four cardinal positions FRONT, LEFT, RIGHT, BACK:— NO BUTTONS = FRONT; BOTH BUTTONS = BACK.
***Soundfield Elevation**	: Allows $\pm 45°$ 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.

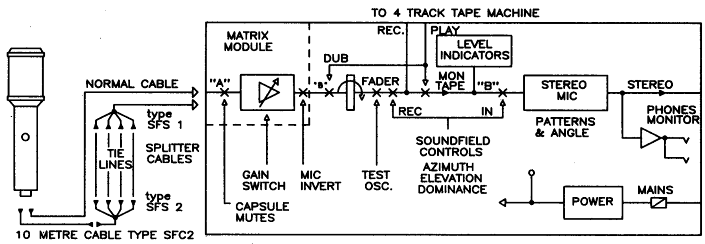
Specification :

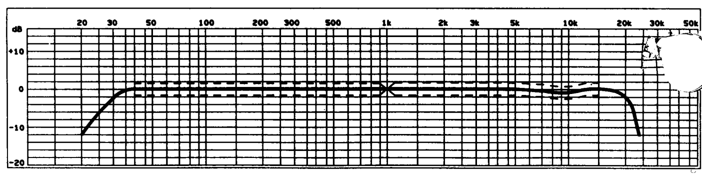
Microphone Front Sensitivity at OdB gain or OdB Faders	: 105dB SPL
B-Format (W) and Stereo (90°, cardioid) outputs at OdBu level. Sensitivity Range at OdBu level	:
Maximum : (45dB gain)	: 60 dB SPL
Minimum : (15dB atten.)	: 120dB SPL
Max. Input for less than 0.5% THD	: 140dB SPL
Frequency Range	: 20-20, 000Hz
Equivalent Self-noise, IEC 179	: Approx. 20-22 dB-A (depends on pattern selection).
Microphone Outputs: A-Format, 48 volt phantom powered, balanced.	
Source impendance	: 100 ohms approx.
Sensitivity	: approx 1mV/microba
Control Unit Outputs at line-up	: OdBu, 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.

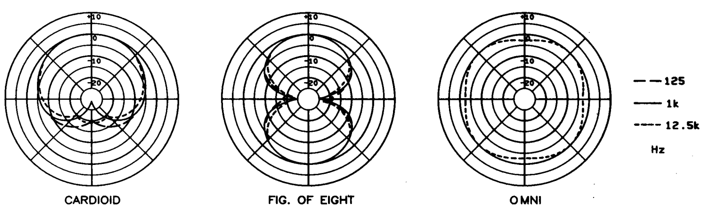
Weights : Microphone : 500 grammes (18oz.)
Control Unit : 3500 grammes (8lbs.)

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

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