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A NEW SOUND MIXER FOR TELEVISION

INTRODUCTION

Over the last few years the number of facilities required on Sound Control Equipment has gradually become more stabilized, and there is now a recognized maximum which any one operator can control. As a result of experience with the modular-type construction described previously in these pages[1] much knowledge has been gained of requirements and this has led to the design of a standard control panel which can readily be expanded to meet today's maximum demands.

It must be appreciated that custom-built equipment, with its unique circuit diagrams, cable layouts and case assemblies, becomes rather costly on a 'one off' basis. This new concept of a larger "major" module system offers a more economic unit through quantity production, and yet it is flexible enough to give a choice in the number of available channels together with various auxiliary facilities as and when required.

This new standard range is known as the Type B1103 Sound Control Equipment. The design is suitable for studio or mobile use and can be extended from 12 to 24 or 36 channels, by coupling basic 12-channel units together. This increases all the auxiliary facilities and the available outputs, as well as the number of channels. The mechanical design is simple and easily mounted into flat wooden desk surfaces. Suitable metal cases are also available.

BASIC FACILITIES

The basic item in this design range is a 12-channel unit which can handle up to 12 microphones of low or medium impedance and 4 high-level inputs (Fig. 1). The latter can be routed to either of the two group or sub-master controls. Channels 1 to 6 are permanent low-level channels and connected to Group A control fader. Channels 7 to 10 can each accept either a low-level input or a high-level input and they can be individually routed to either Group A or Group B. The remaining two channels 11 and 12 are permanently low level and connected to Group B. Every channel has its own input impedance selector key on the front panel, together with an operational pre-set gain control which enables the channel amplifier to accept signal peaks of up to −30 dBm. A miniature quadrant-type fader permits the smooth adjustment of channel gain (Fig. 2).

Each group is electronically divided into two isolated output paths. One goes to the Main Control fader and then to two separate output amplifiers. The other path can be selected by a switch to the "Clean Feed" output, this is a third isolated output amplifier with its own level control permitting the separate transmission or recording of any group of channels without affecting the main programme. Eurovision requirements provide a typical application of the "Clean Feed" facility, where all the effects of an international event can be transmitted separately from the local programme which would include its own language commentaries. All these three programme outputs are of the best broadcast quality and have optional output impedance characteristics, 600 ohms being available for matching high-grade music-quality lines and 75 ohms being offered for feeding lower-grade 600-ohm lines.

Important as these main programme facilities are, no sound equipment can produce good results without certain vital auxiliary circuits. Pre-listening enables each source to be checked for level and quality before fading up on transmission. This is effected by pressing the quadrant fader gently against its stop in the "Off" position. An over-ride micro switch within the fader connects the channel output to another amplifier. The

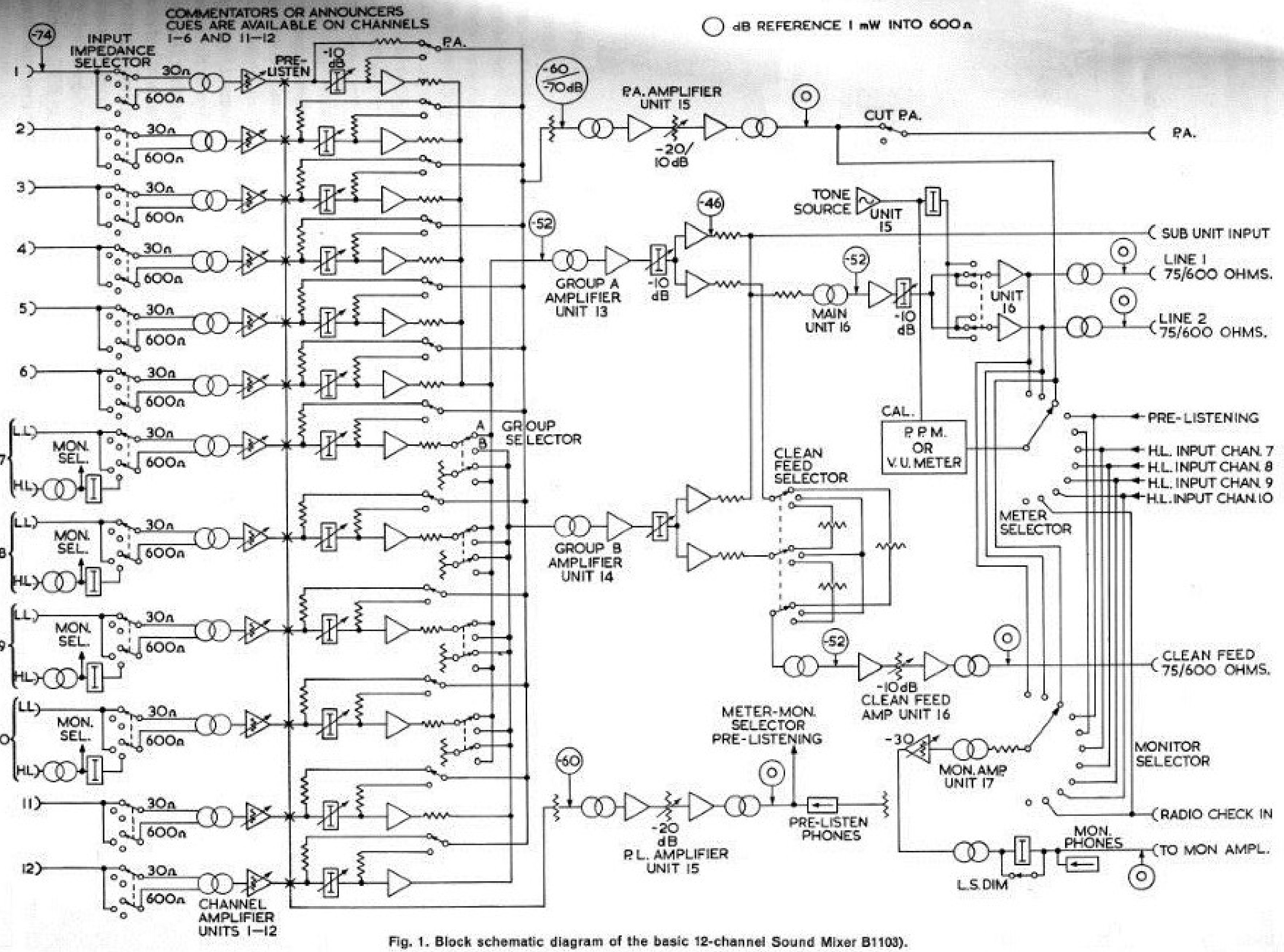

Fig. 1. Block schematic diagram of the basic 12-channel Sound Mixer B1103).

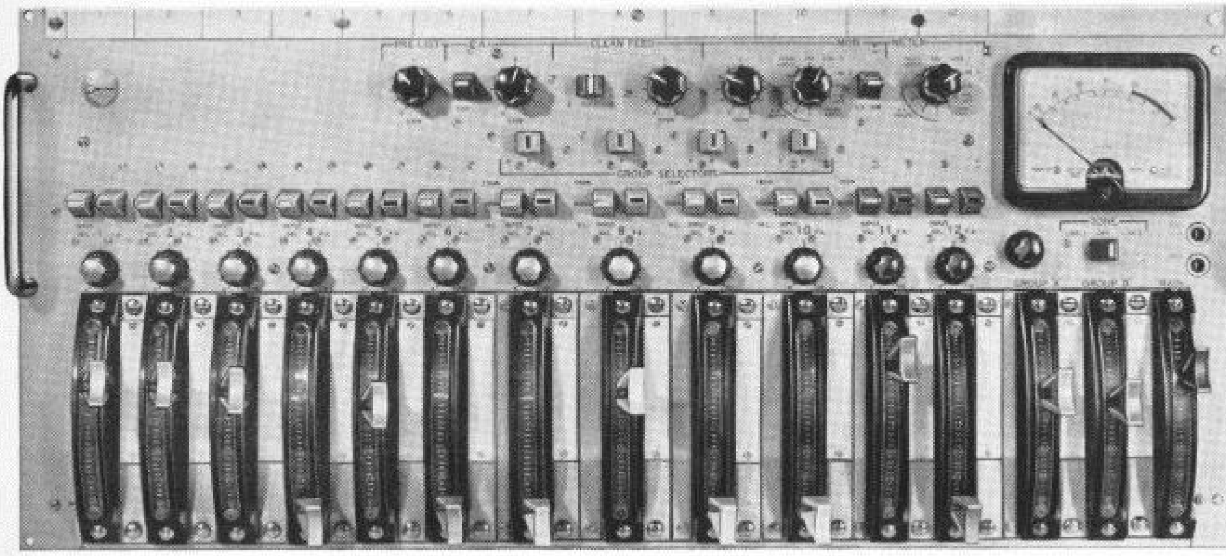

Fig. 2. The control panel of the 12-channel Sound Mixer.

operator can then check the circuit on headphones, loudspeaker or programme meter. This facility is available on every quadrant fader including the group controls.

Public Address (P.A) facility is also essential for many programmes. By pressing a key above each channel fader, it is possible to provide selective sound reinforcement to separate power amplifiers and loudspeakers. This enables an audience in the studio to appreciate fully the programme and react in the correct manner. Acoustic feedback can be prevented by a gain control and "Cut" key on the front panel. There is a second use for this particular output; it can be used for feeding loudspeakers on the studio floor to cue artists or to permit miming, or to feed separate studios with selected sound for complicated programmes such as opera, where the orchestra is in a different studio from the artists. This application is generally termed "Foldback".

The outputs of all these programme and auxiliary outputs have to be monitored both aurally and visually. The human ear is still the best judge of actual volume level and the B1103 Control Panel incorporates a special bridging isolation amplifier with its own input selector switch, gain control and "L.S Dim" key. This enables the operator to switch a loudspeaker around all the outputs and high-level inputs without disturbing those lines. There is also one position on the switch for an external input, normally the "Radio Check" receiver, which permits the cueing-in of live programmes. The "L.S Dim" key is to quieten

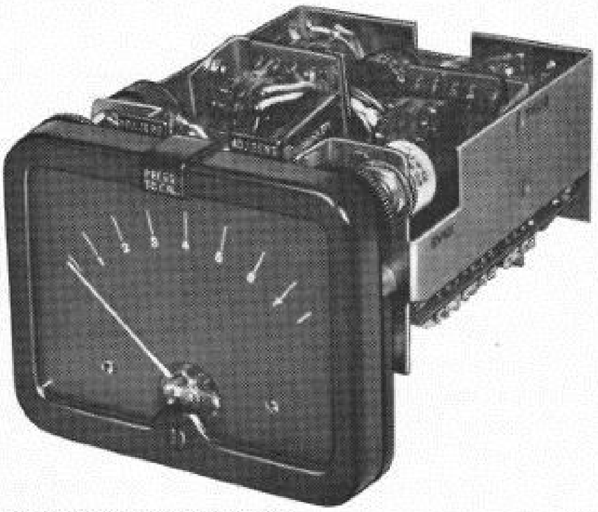

Fig. 3. The peak programme meter.

quickly the loudspeaker temporarily when necessary, but to return immediately to the same sound listening level.

To help the operator establish correct sending levels a programme meter is provided with the same selection facility as the loudspeaker monitor. In this unique design, a standard V.U meter can be fitted or a transistorized Peak Programme Meter, type B1756, which has been specially designed to fit in the same space. The Peak Programme Meter, described in detail later, conforms to the stringent BBC specifications and can drive several remotely mounted slave meters if necessary (Fig. 3).

A stable tone source is incorporated and provides two useful features. It can be used for calibrating the sensitivity of the Peak Programme Meter, and it can be switched to either of the main output lines to keep them "engaged" while a rehearsal can proceed utilizing the second output.

This completes the basic unit. Where more complex programmes have to be produced an auxiliary unit is available which fits neatly across the top edge of the main unit. It is joined with a hinge so that it can be adjusted to mount at any suitable angle. Its facilities include echo hybrid amplifiers and their mixture switches together with an "Echo return" level control (Fig. 4). By the use of external reverberation equipment "Echo" effects can be added to eight of the twelve channels. Two variable equalizers, controlling high, low, and mid-frequencies can be connected into any four channels, normally 7 to 10, but easily adjusted to other channels if required. Finally, a second "foldback" output is provided from any channel, with its own output amplifier, completely independent of the main unit.

Both the main and auxiliary panels are permanently wired together. They also have a suitable input socket available for coupling in a sub-mixer unit, normally another 12-channel unit.

HIGHLIGHTS OF CIRCUIT DESIGN

Channel Amplifier (Fig. 5)

The channel amplifiers are printed wiring board assemblies which plug into the front panel, being easily removable and completely interchangeable. In the unlikely event of a failure, a new amplifier can be fitted in seconds. A test board is also supplied to facilitate fault finding and provide access to the input and output circuits. The design of this amplifier has been carried out with special attention to noise, bandwidth and distortion. To achieve a minimum amount of noise generation, the first stage utilizes a low-noise silicon planar NPN transistor. This has a mean wideband noise figure of 2 dB and is operated as near as possible to its optimum conditions for minimum noise. Nyquist's equation $E_N^2 = 4KTBR$ gives a thermal noise figure of approximately -128 dBm referred to the input for a bandwidth of 15 Kc/s. Noise levels of -125 dB to -123 dB are achieved on this amplifier, giving a noise figure of 3 to 5 dB. Better transistors are available with narrow band noise figures of less than 1 dB, but as they are rather expensive it was considered uneconomic to use them at present. Due to the large numbers used in the equipment, a disproportionate increase in cost would result for the slight improvement in performance.

The amplifier has three directly coupled common emitter stages incorporating overall d.c feedback and d.c stabilization for each stage. Fixed series feedback reduces distortion in the second stage, while variable series feedback in the first and third stages gives a pre-set gain control of 30 dB, allowing a wide range of input levels to be accommodated without distortion. The transistor chosen for the second stage had to be a silicon planar type due to the low bias current available from the collector resistor of the first stage. This is followed by a silicon-grown junction NPN transistor presenting a 600-ohm unbalanced output impedance to match the quadrant-type fader.

After the fader, each amplifier has a buffer stage

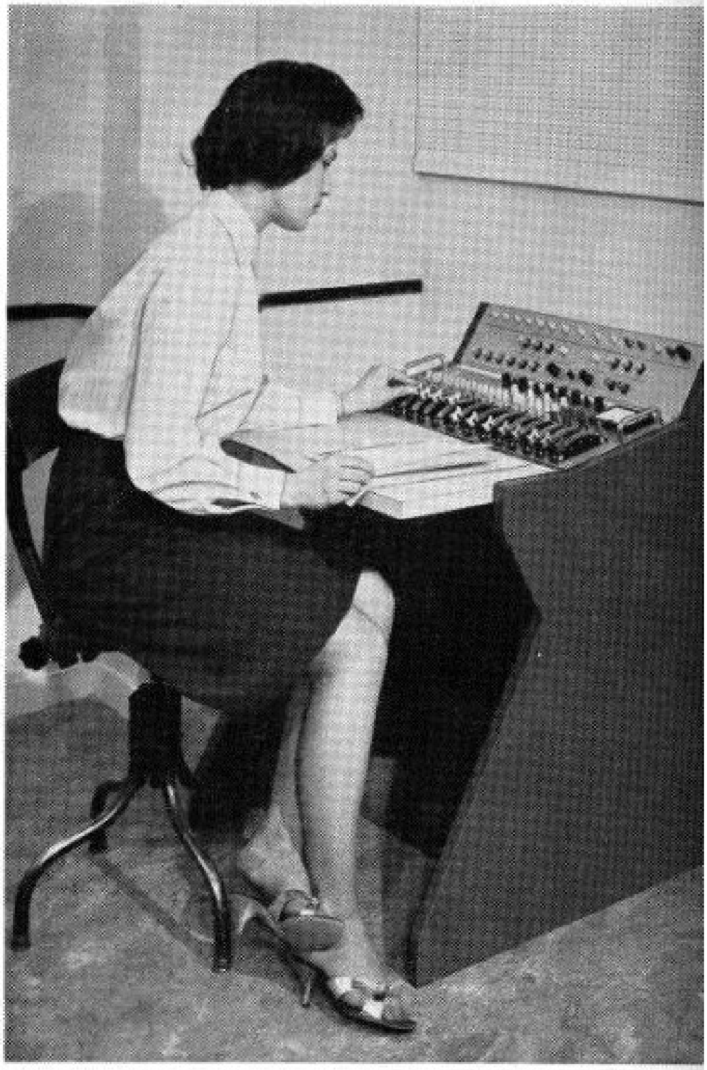

Fig. 4. The basic panel with ancillary echo unit in a typical desk.

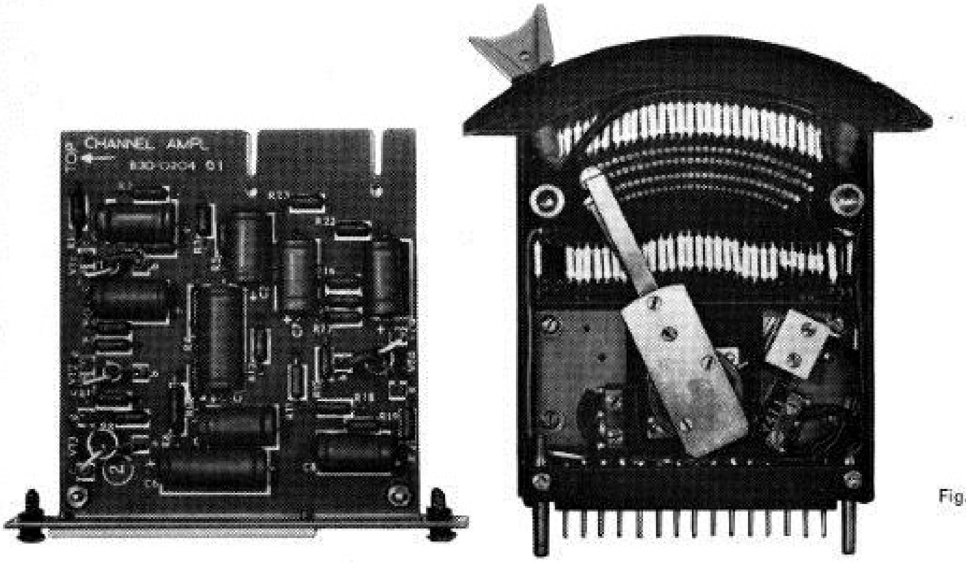

Fig. 5. Channel amplifier and quadrant fader.

before mixing to the common group circuit. This gives good channel isolation for the "P.A" and "Foldback" circuits where it is important to achieve at least 50 dB protection between the wanted and unwanted channels. The P.A take-off point can be before or after the channel fader as required.

Group Amplifier

The requisite number of channels are resistively mixed and fed to two group amplifiers. These are also plug-in printed board assemblies similar to the channel amplifiers, but a barrier key-way prevents them being fitted incorrectly. As this amplifier is an intermediate stage, its main design considerations are low distortion and stability. This is achieved by two stages of directly coupled common emitter amplifiers. Distortion is minimized by overall feedback and individual feedback circuits. D.C feedback is employed for bias stability. These stages feed the 600-ohm quadrant faders which are identical to the channel faders. The output from the fader is electronically split by three stages of directly coupled transistors, their bias being set by the first stage. Two common collector output stages are driven from the emitter of the first stage. The isolation between these two outputs is at least 40 dB, enabling the groups to be mixed or transmitted separately as required.

Main Programme Amplifier

The main design requirements here are stable amplification, suitable output impedance characteristics and power to feed various line requirements. Similar circuits are utilized to feed the two "Main Programme" outputs and the "Clean Feed" output. They have a balanced input feeding a two-stage amplifier, again a large amount of a.c and d.c negative feedback is applied overall, as well as to each stage. This feeds the Main Control fader which is followed by the output amplifiers. These each consist of four stages, one common emitter, one common collector and two push–pull stages, using silicon transistors throughout. The first stage here utilizes a low-noise transistor to give the lowest possible noise output when the Main fader is closed. This is directly coupled to the second stage and is biased from the emitter of the second stage, a.c and d.c feedback being applied over the same path. The low-output impedance of the second stage is coupled via a capacitor to the low-input impedance of a hybrid cross-coupled "short-tailed pair" phase splitter. This drives a push–pull class A output stage using two high-power diffused junction silicon transistors, very much underrun. Matched pairs of transistors in the push–pull stages give a balanced d.c output to the transformer, which avoids saturating the core and minimizes distortion.

By varying the connection to the secondary of the transformer it is possible to have either a 600-ohm output impedance or a 75-ohm output impedance. The latter, with its reactive component kept below 20% at frequencies up to 10 Kc/s, is specially designed to feed low-grade lines, where the line impedance may vary between 100 ohms and 1,000 ohms. It can also be utilized in studio applications where very long lines can cause high-frequency losses if 600-ohm circuits are used.

Pre-listen and P.A Amplifiers

These auxiliary circuits are identical and use low-cost PNP silicon transistors. The amplifiers have a gain of 86 dB and a maximum output of +10 dBm which is adequate for all normal applications. The first stage, which has a transformer input, incorporates the normal stabilization and series negative feedback. It feeds a moulded track low-noise rotary potentiometer gain control which is followed by a three-stage output amplifier. As the collector load of the final stage consists of the transformer primary, provision is made to adjust the collector current by means of the bias control to give minimum third harmonic distortion while maintaining correct bias conditions.

"ECHO" CIRCUIT

The amplifier necessary to create echo effects has several interesting features. It is necessary to divide the channel circuit in order to cross-fade between "direct" sound and "echo effect" sound, and in order to avoid positive feedback around the "Echo Return" circuit, there must be maximum isolation between the two outputs. This is achieved here by an electronic version of the hybrid transformer. However, unlike the transformer this amplifier has zero power loss, flat frequency response and extremely low distortion due to the large amount of feedback.

It has a common emitter first stage, directly coupled to the two output stages. These have 100% feedback as they are connected in the common collector configuration. These collectors are grounded by a decoupling capacitor, which is necessary in order to achieve an isolation of at least 45 dB between outputs. These are both 600-ohm impedance and are fed into an "Echo Mixture" fader. When this control is in mid-position, it has 0 dB loss in both paths, therefore equal "Echo" and "Direct" signals are transmitted. Turning the control clockwise, maintains full "Echo" signal while reducing the "Direct" signal, resulting in a fuller echo effect. Similarly turning the control anti-clockwise will not affect the "Direct" signal level but will reduce the "Echo" signal. This, used in conjunction with an "Echo Return" gain control in the circuit from the echo facility, will produce any required effect, from the slight increase of "liveness" in a dead studio to the full reverberation of a castle dungeon.

It should be appreciated that it is almost essential to have this echo effect, for most television studios are designed to be very "dead" acoustically. They are normally multi-purpose studios and have to handle drama, classical music and light entertainment, all of which require different degrees of reverberation. They also occasionally require adjustment to their frequency response, and this is why the two variable equalizers are fitted.

EQUALIZERS

These are electronic devices with 0 dB insertion loss at 1,000 c/s. The gain at 60 c/s and 10 Kc/s can be varied from +10 dB to −10 dB. To replace "presence" lost on certain types of microphones a variable boost is provided at 3 Kc/s with +10 dB maximum. A.C feedback from the emitter of the second stage is varied in response at either end of the audio spectrum by two continuously variable moulded track resistor and fixed capacitor networks. Attenuation of the feedback provides the boost at 3 Kc/s by means of a tuned circuit. The Q of this can be varied by switched resistors giving 5 steps of 2 dB each.

PEAK PROGRAMME METER (Type B.1756)

This 24-V peak programme meter, which uses an instrument possessing the ballistic characteristics specified by British Broadcasting authorities, is 4¼ in. by 3¼ in. and governs the front panel dimensions of the amplifier (see Fig. 2).

The amplifier designed to drive the meter uses 8 PNP low-cost silicon transistors, 2 germanium diodes and 3 silicon diodes. An isolated input is provided with an impedance of 10 K ohms. To regain the loss in the input circuit there is a common emitter voltage amplifier working in class A push-pull. Directly coupled to this amplifier is a common collector amplifier. Bias for the first stages is obtained from the emitter circuit of the current amplifier; this circuit also providing overall d.c feedback and stabilization. A low-impedance input to the solid tantalum timing capacitor is provided by the current amplifier and the transformer.

The signal is full-wave rectified by two germanium diodes. These also control the first part of the meter law with their logarithmic low-voltage forward characteristics. A discharge time constant of 3 sec is provided by a 220-K ohm resistor and the back resistance of the diodes. The voltage loss is regained by a two-stage "long-tailed pair" d.c amplifier, voltage gain being provided by the first pair of transistors and power gain by the second pair. As the first is a voltage amplifier, it is sensitive to differential changes in base to emitter voltage (Vbe) that can be caused by a temperature gradient being established across the amplifier, due to any nearby sources of heat. This is minimized by thermally coupling the transistors together in a standard copper cooling clip.

Pairs of transistors can be obtained in one transistor case for the same purpose, but the cost makes their use uneconomic. Matched pairs of transistors are used, however, any discrepancies in matching being accommodated by differentially adjusting the bias with the control labelled "Adjust Zero" on the front panel.

A.C gain is set by the front panel adjustment marked "Sensitivity", this controls the first part of the law in conjunction with the rectifier diode characteristics. The remaining part of the law is obtained by shunting the output impedance of the amplifier with diodes having logarithmic forward characteristics. Deviations in the diode characteristics are corrected by four pre-set controls to give 4 dB steps over the range 2 to 7.

MECHANICAL DESIGN

The B.1103 Sound Mixer is a flat control panel, designed to match the same series of equipment as the B.3714 Vision Mixer. It fits a simple rectangular cut-out in a desk. This is usually designed to suit the studio décor and can often be built on site, however a standard design of desk is also available. Where compatibility with old-style equipment is required, the basic mixer will fit into a similar case to the BD.580 Sound Mixer and the BD.841 Vision Mixer.

All connections are by plug and socket on the rear panel. One point of interest is the use of multi-pin 17-way connectors for the low-level inputs. These are used in conjunction with composite 8-pair sound cables which greatly reduce installation or setting-up time, whether in fixed installations or mobile control rooms on outside broadcasts. For the latter application simple splitter boxes are available, making the equipment instantly compatible with the customer's standard microphone connectors. The rear panel also carries certain heavy components such as output transformers, together with the printed wiring board assemblies for the main amplifiers.

INSTALLATION

The equipment is admirably suited for clean installations where a good line of sight is essential. A low profile is easily achieved as illustrated in Fig. 6. It

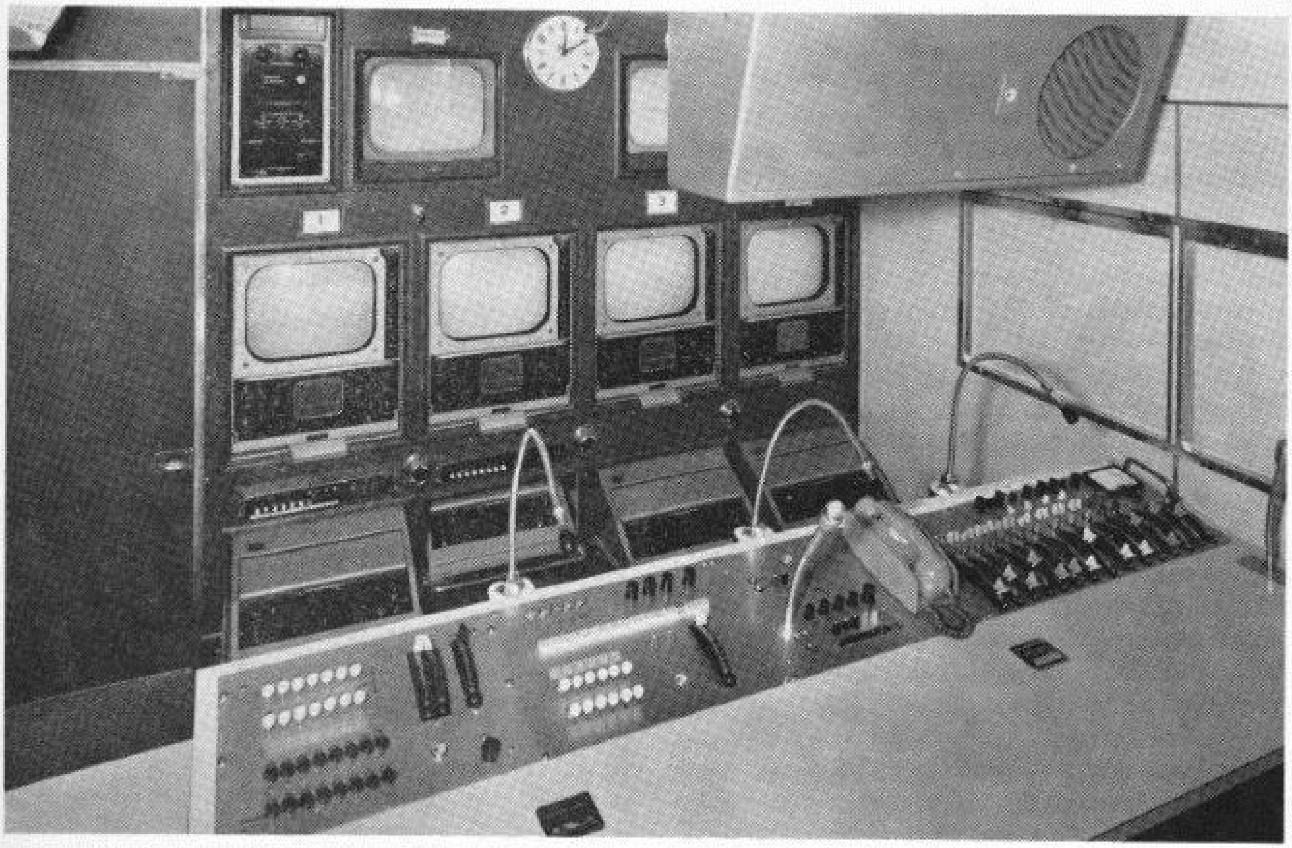

Fig. 6. A 12-channel Sound Mixer installed in an O.B van for Yugoslavia.

requires a 24-volt d.c supply which is normally supplied by the Type B.4203 regulated supply unit. The consumption of a basic unit is approximately 700 mA and this increases to 1 A when the Auxiliary Unit is added. A separate d.c supply is recommended for the signal light circuits. These include illumination of the quadrant faders, together with voltages available from the fader cueing switches for remote warning lights and muting relays.

REFERENCE

D. B. Manning: A New Design of Sound Control Desk for Television; *Sound and Vision broadcasting*, Vol. 4, No. 1, Spring 1964.