EELA AUDIO SRM USERS MANUAL

This manual provides the aspirant user of an EELA AUDIO SRM RADIODESK with the information, needed for operating and maintaining the equipment. Chapters 1 and 4 have to be read by anyone, who works - with the equipment, chapters 2 ,3 and 5 are important for people with technical responsibilities like instillation and maintenance, although is does no harm to others to read this information.

The chapters are:

1 INTRODUCTION

The philosophy, based upon which the EELA AUDIO SRM RADIODESK is designed is that of offering a full function radio studio installation for a budget friendly price, without compromises for facilities and technical qualities.

The result is a control desk with not all, but certainly the most important functions built in, in a way that anyone, after a short instruction, can work with it, also people for whom the program to be transmitted is many times more important than all technicalities around it.

The primary function of the console is the use direct on air or for recording “live” programs for later transmission.

A secondary function is to do not to complicate editing work outside broadcast hours.

The SRM mixer is built in a way, that operation is bath possible by the presenter himself or an engineer.

This is realised by bringing up front only those functions of direct importance and by hiding end automating difficult handling as much as possible.

The SRM mixer is suited for operation in one (control-) room or with a separated booth or studio or a combination of both. The status can be set simply by a few switches to inform the logic of the desired operation.

RELIABILITY was design criteria number 1

This had to be so, because for radio only one slogan is valid: “THE SHOW MUST GO ON”, and that under all circumstances. That is why the console is fully modular, to be able to take out a faulty part without the need for completely taking the installation out of operation.

That is why voltage controlled faders are used end electronic switches instead of ”direct” components.

By being innovative in the construction, EELA AUDIO was able to keep the price at an affordable level, end still making the SRM a fully professional control desk.

A standard SRM is equipped with:

3 microphone input channels

1 telephone in/out channel

4 stereo line input channels without equaliser.

2 stereo line input channels with equaliser

2 free positions for extension, suited for any kind of input module.

Stereo line inputs with internal RIAA preamps for gramophones are available as an option.

This configuration has shown adequate for the majority of applications. The two free positions allow adaptation to special needs of certain customers. The advantage of a standard desk is the favourable price to quality ratio end the possibility to incorporate many typical radio functions without bringing up the cost unnecessary or making operation more complicated than needed.

Important for a problemless function in radio operations are especially facilities for foldback end signalisation. These facilities are automated as much as possible end are user friendly in a way that even an operator with little practical experience is able to bring his program on air without mistakes. He can spend almost all his attention to the program instead of to (often not necessary) technical operations. That’s why the number of controls is so small compared to so called “universal” mixers.

LIMITER

A unique system for over modulation protection is built in the SRM mixer:

Because all channels are controlled by VCA's (Voltage Controlled Amplifiers), it is possible to influence the gain by an external control voltage.

In the SRM the output level is measured and compared with a reference voltage. Is the level higher than the reference, a control signal is generated and fed to all channels to reduce the gain so far that the over- load is compensated.

SCRIPTSPACE

In the centre of the mixer there is a free space of the size of an A4 farm, underneath which all mix- and line amplifiers and the foldback-, monitor- and talkback circuits are located. This is an ideal place to put scripts, records etc. without interfering with controls. Also a visual separation is created this way between microphone- and telephone channels on one side and the stereo inputs on the other side, making easy access possible without mistakes.

INDICATORS

Near the meters four STATUS LED's are located which indicate:

- ST one or more microphones in the studio are open
- CR one or more microphones in the control room are open
- CUE one or more channels are switched to CUE and AUTOCUE is engaged
- LIM the limiter circuit is active

SIGNALISATION

In the SRM inputs are available for driving signalisation lamps in the studio and control room for indication of open microphones for presenter and guests and as a warning against entering the room when it is not allowed.

FADERSTART

A typical radio facility is starting machines (recorders, gramophones etc.) on opening the fader of the dedicated channel. The SRM stereo channels have this possibility standard. As an extra a separate output is available for stopping the machine on closing the fader.

Both outputs are isolated from the audio electronics by optocouplers to avoid any interference with the grounding system by the machine controls.

Adaptation to several types of machines is possible by jumpers in the modules for selecting a pulsed or a continuous contact.

SPECIAL TELEPHONE MODULE

A very popular signal source in radio is the telephone. It is also a complicated one, because on the two wire connection both the signal from the caller and that from the studio is combined. If the is signal is direct coupled to the desk, also the microphone signal from the studio will be heard with the low telephone quality. That is why radio studio's use a TELEPHONE HYBRID, a circuit that separates the combined signal on the two wire connection and converts it to a 4 wire with individual send- and return ways.

The return signal sent to the caller has to be a mix of the program MINUS the contribution of the telephone connection, otherwise the system would oscillate because of the not complete damping of the studio signal.

Using a hybrid can be difficult. For each connection, the hybrid has to be lined up for maximum reduction, a handling that can give problems or cannot be done because of lack of time, so the result is not the optimum.To avoid these line up's, the SRM has some “tricks” built in to ease the operation.

2 INSTALLATION

MOUNTING

The SRM mixer is constructed in a way that it can be built easily into a table or dedicated desk. To do this, make a hole in the desktop to slide the mixer into it, in a way that it is supported by its wooden side panels. The slope of the control panel gives good visibility over controls and indicators and the wiring is partially hidden for a neat appearance.

WIRING

Several systems are in use for signal transport in audio installations.

In large, professional studio's the connection technique is mainly fully balanced and floating, using transformers in in- and outputs. The screen of the cables can then be cut on certain places to separate the grounding systems of parts of the installation.

Why is this separation so important?

In every piece of audio equipment the signal is related to the ground of the power supply. That means that when more systems are coupled, eventual difference voltages between two grounds are added to the audio signal.

For reasons of safety all metal parts of the equipment have to be connected to a safety ground. Also the reference ground of the systems is in most cases connected to the chassis and so to the safety ground.

The safety ground can be "polluted" for several reasons, creating voltage differences between individual connections. When using a 'Wild" grounding system, these voltages can be added to the audio signal and heard as hum or noise.

The most professional way for creating both a good safety system and maximum suppression of interference is then a balanced, floating wiring. A balanced connection also has a good suppression of electrical strayfields by compensation in the in- put stages.

This method of signal transport is for normal people not affordable and often not possible, especially when consumer type equipment is used, a normal practice in small scale radio stations.

The EELA AUDIO SRM radio desk is specially designed for connection of HiFi- or semi professional equipment. These are normally unbalanced, often with high impedance's

.

TAKE CARE WHEN MAKING CONNECTIONS

A good grounding system, using ONE CENTRAL GROUND is vital and no sins are allowed against it. The SRM has all line level inputs unbalanced on HiFi standard RCA/PHONO connectors. All outputs are also unbalanced and on PHONO with exception of the main outputs, which have line amps with transformer outputs and XLR's. The reason for this is that those outputs have to be connected to destinations outside the room like a transmitter or racks room, direct or via PTT lines.

MAKE CONNECTIONS AS SHORT AS POSSIBLE

A long, especially an unbalanced line is more sensitive for interference than a short one. Do not position any interfering line near the signal connections. These can be power lines and telephone- and signalling connections.

3 CONNECTIONS

All connections on the SRM are located on three panels on the back of the frame. Most are RCNPHONO types, balanced connectors are XLR's.

MASTER CONNECTOR PANEL

POWER

This is the input from the power supply, which is an external rack mount unit. The connector is a 4 pole XLR, wired:
1: audio ground
2: +18Volt
3: -18Volt
4: + 48 Volt phantom power

OUTPUTS LEFT and RIGHT

These are the balanced main outputs, meant for connection of the transmitter, direct or via PTT lines. Connections are on male XLR's end wired:
1: audio ground
2: + signal (in phase)
3: - signal (out ot phase)

A good way to make a connection to an other installation is to disconnect the screening where the cable leaves the room to avoid any possibility for a ground loop.

CR LSP Land R

To these sockets the control room monitor loudspeakers have to be connected via a power amplifier or direct in the case of active monitors. The loudspeakers have to be of a good quality to give a reliable reproduction of the signal to be transmitted.

CR HPH

This is a ¼" jack connector for use by a pair of stereo headphones for the presenter or engineer in the control room. The output is suited for common types with the best results from units with impedance's from 200 to 600 Ohms.

MTR Land R

These outputs are meant for connection of an external meter, which follows the selected monitor source, additional to the internal meter reading always the main output.
The external meter can be used to pre-set channels for gain via the CUE system.

CUE LSP

These are two parallel connected sockets for use of a small loudspeaker (via an amplifier), serving CUE-end intercom functions. Also small active units are suited.
It is important that the sound of this small speaker is different from the main monitors to get a good indication of where one is listening to. Also the positioning of the speaker can be important.

CONNECTOR PANEL STEREO INPUTS

On this panel one finds the connections to the 6 stereo channels, each with 4 PHONO connectors.

IN Land R

These inputs are unbalanced, the sensitivity depends on the type of stereo input module.

The normal stereo inputs, with or without equaliser have a linear frequency response and a sensitivity adapted to normal HiFi- or semipro equipment.

The channels adapted for gramophones have a sensitivity allowing direct connection of the pick up cartridge. Also the RIAA compensation is available.

Direct connection of pick ups has to be done with same precautions:

The connection is very susceptible for interference because of the low signal level and the high impedance and the unbalanced character. For these reasons the link has to be kept as short as possible and far away from any interfering source. Most gramophones have two PHONO plugs and a separate ground wire from the frame or tone arm. Tie this wire to the frame of the mixer, using one of the screws nearby the inputs.

In case the interference level is still too high, the solution is using a separate RIAA preamp (e.g. the EELA EA 811) very close to the gramophone and rebuild the input for normal line level.

START

This connector supplies a START signal on opening the fader for remote control of CD players, MD players or other machines.

The output is separated from the audio circuits via an opto coupler and behaves as an NPN transistor coming in conduction. The collector is wired to the pin, the emitter to the ground sleeve. Take care to choose the right polarity and do not short any point to the frame.

The character of the START SIGNAL can be selected internal in the module with a jumper for a pulsed- or a continuous signal. On the selection pulsed, the transistor conducts approx. 0.5 seconds after opening the fader, when continuous as long as the fader is open.

STOP

This remote control output works in conjunction with the above mentioned START output and can in fact only be used when the pulsed option for start is selected. It is a separate output, also via an opto coupler, generating a signal when the fader is closed.

TELEPHONE LINE A/B and TEL A/B

These 4 mm connections are for linking the (POTS) PTT line and a telephone to the telephone module. Use a good quality 4 mm plugs and connect them as follows:

- LINE A A wire of the PTT, often red
- LINE B B wire of the PTT, often blue
- TEL A A wire of the telephone, red
- TEL B B wire of the telephone, blue

This wiring works in most cases. A number of modern telephone installations have more wires, used for digital control of functions like dialling etc. Leave these connected to the telephone and link only the speech circuit (a and b wire) through the telephone channel.

The telephone module links the telephone to the line in the OFF position, allowing the telephone to be used for making a connection, which can be switched to the module after that. Listening to and talking to the subscriber can be done via the CUE- and TALKBACK systems of the mixer, given that the fader of the channel is closed.

For practical reasons a second telephone can be paralleled to the one in the control room and placed in the studio, allowing the presenter to have conversation with the caller prior to the transmission. The presenters microphone has to be closed then of course.

Disconnect the bell circuit of the telephones or replace it by a light indication not to disturb the broadcast when it rings.

SPARE

This space is reserved for extension of the number of channels.

POWER SUPPLY

The power supply of the mixer is an external unit, linked to it via a cable supplied with it.

AUDIO GND and CHASSIS GND

These binding posts can be used for linking the system ground to the safety ground of the mains. This is the best point to form the centre of the STAR-GROUND SYSTEM of the installation.

Locate the power supply not to close to the mixer, because the mains transformer generates a stray field, causing interference on sensitive equipment, microphone inputs etc. The length of the cable is a good indication of the distance.

4 OPERATION

The number of controls on the SRM mixer is very small, but they all have their own function and rules for setting. An easy way to train someone for using the mixer is a description of each of them, grouped per type of module included criteria for the settings.

S 71 MICROPHONE INPUT

This module is meant for connection of microphones, The input is balanced and supplies + 48 Volt phantom power, allowing almost any type of professional microphone to be used. The input connector is a female XLR.

The following controls will be found on its front panel:

GAIN

This rotary control sets the gain of the input stage in such a way that nominal modulation occurs with the fader on the 0 line. This means that:

- The output level of the mixer is the same as the level one hears or measures using the CUE system This means that a channel can be pre-set very accurate with the fader closed.
- The signal level in the channel is optimal for headroom end signal to noise ratio.
- The fader is in its working area for accurate final adjustments of the gain.

EQUALISER

The following three knobs are tools for correction of the sound to adapt it to personal taste or for compensation of wrong acoustics in the studio.
The control range has been kept small (+ /- 6 dB) to prevent presenter or engineer to make such severe corrections that the result for the listener might become unpleasant.

HIGH PASS FILTER

An extra possibility for correction of the frequency response is available internal in the module:
a low frequency cut filter. In general this is switched IN to keep low frequency signals (below 80 Hz) out, which have no function in the intelligibility of the speech, together with low frequency noise from traffic, air movements and “plopping” of the microphone.

Switch the filter OUT only when the microphone has such a filter built in.

PAN CONTROL

This pot enables the (mono) microphone signal to be placed on any desired position in the stereo picture. A good application can be found in interview situations by positioning the presenter end the guest e.g. left end right to improve the separation end the understanding, especially when they speak at the same time.

COUGH LED

This LED indicates the activity of the COUGH/COMMAND function in the module. This activation happens in the studio by means of the COUGH/COMMAND push button there end has as effect:
- the (electronic) fader will be closed end
- the prefade signal is sent to the CUE system.

The action of the COUGH/COMMAND function depends of the fader setting, but is only enabled when the microphone is in the ST (studio) status. The LED indicates also clearly which microphone channel is activated.

S 72 STEREO LINE INPUT

This module is meant for connection of stereo sources like MD and CD players, tape or cassette recorders, line connections etc.

The standard module has linear frequency response inputs with a sensitivity suited for most common units. By means of plug in boards the following extensions are possible:

-Addition of an EQUALISER with the same characteristics as the one in the microphone module.
- Addition of a RIAA preamp, allowing gramophones to be used direct from the pick up.

These extensions cannot simply be added afterwards, the best is to add them to the initial order.

On the module the following controls can be found:

GAIN

This control determines the input sensitivity of the channel, to be set for nominal modulation with the fader on the 0 mark. This setting can be done with the aid of the CUE system, the best if an external meter is available. The setting is mainly dependant on the piece of equipment used, not from the material played, so making the adjustment once is in many cases sufficient. Small gain variations, caused by the “software” can than easily be corrected by the fader, also during the transmission.

EQUALISER

In practice this facility is not really needed for an ON AIR desk, because all played stereo material has to be produced and corrected in advance. There are however cases where a possibility for tonal corrections can be useful, e.g. processing of incoming line- or OB connections or reporters tapes that have not been edited. That is why two stereo channels of the SRM can have an EQ section built in.

The equaliser has the same curves and (restricted) range as the one in the microphone modules.

BALANCE CONTROL

This control has to be used to make corrections for an eventual imperfect left to right balance of the two stereo signals. Also here a small control range to avoid mistakes. Should this range (+ /- 3 dB) not be sufficient, the fault may be found in the source equipment!

CUE SWITCH and LED

This switch can be used for monitoring (in stereo) the source via the channel with the fader closed to adjust evt. gain and EO.

A second, and may be the most used function is the possibility for cueing recordings prior to bringing them in the mix.

This last function can be done in two ways, using the CUE speaker or via the main monitors. Which route will be used depends on the way of working.
When the installation is ON AIR, the CUE speaker will be used, because it is already switched on for intercom with the studio and the main monitors are in use for checking the transmission.
In production the main monitors are often used for cueing by depressing the AUTO switch in the master/script area, changing over the monitoring to the prefade signal of the respective channels on depressing the CUE switch there.

Remark the CUE function can only be active when the fader is closed. The CUE status is indicated on the channels by the CUE LED.

S 73 TELEPHONE IN/OUTPUT

This module can be used to bring telephone conversations in the broadcast.
The procedures to follow are:

CALL ON INITIATIVE OF THE STUDIO:

. Leave the ON switch in the OFF position.
  The PTT line is now direct connected to the telephone.

. Use the telephone to make the connection.
  When the connection is established, it is possible to change over to the console by depressing the ON switch.
  Leave the fader closed, otherwise the call will be brought in the mix immediately.

Now we have the possibility to listen to the caller on depressing the CUE switch and to talk to him by engaging the TB button.

After the initial conversation outside the broadcast, the connection can be brought ON AIR by simply opening the fader. The signal heard by the caller is a total program mix MINUS his own contribution to it, eventually interrupted by talkback when the fader is closed.

Making a connection is also possible in the studio or booth, using a parallel telephone there, however bringing the call ON AIR can only be done in the control room.

ANSWERING AN INCOMING CALL

This can be done in the control room in two ways, using the telephone or direct on the mixer.
An incoming call is indicated by the telephone bell if not disconnected, and with an LED alongside the ON switch, flashing in the rhythm of the bell signal.

The first method of answering is simply lifting the receiver, a second one is depressing the ON switch after which conversation can take place as described earlier using the CUE- and TALKBACK facilities.

The call can also be answered in the studio, although this may not be so simple because often there is no indication.

Individual knobs on the telephone module are:

ON

This switch changes over the PTT line from telephone to the hybrid in the channel. With this switch a circuit is coupled that keeps the fader closed in the OFF position and about 1 second after switching ON. This has been done to avoid the noise generated by the DC on the line on change over to enter the mix if this was done with open fader, deliberately or by accident.

An other function of the ON switch is disabling those functions of the channel, that are not useful with a not activated telephone channel like CUE- and TALKBACK facilities. By this inhibit we have two possibilities less for loosing the signal, enhancing the workability of the installation.

CALL LED

This LED replaces or completes the indication of incoming calls and flashes with the bell signal, pointing at the line that is calling.

GAIN

This is the control for the input sensitivity of the channel, to adapt to the level on the telephone line.
There is a centre detent for a nominal position. Correction will be made in general during the transmission or can be done via the CUE system, the latter only useful when an extra meter is available.

S 76 MASTER/SCRIPT MODULE

This module is positioned in the centre of the desk and can be used for putting a script, records etc without obstruction of controls. It also gives a separation between two groups of inputs. In the standard version microphone and telephone channels are located at the left and stereo line channels at the right of the script space.

Below this module all mix- and line amplifiers find a place. together with the monitor circuits for control room and studio.

In the top part of the master/script module are placed a meter, status indicators and the controls for control room and studio monitoring.

METERS

The meter used here is a stereo LED instrument with two times 19 LED's, with an indication like the well known DIN bargraph PPM. The range is - 40 to +3 dB with respect to the reference level, normal +6 dBu.

This means that an indication 0 equals an output level of + 6 dBu, the standard peak level in European radio studio's

The TIMING of the meter is also standardised with an attack time of 10 msec. and a decay of 1.5 seconds over 20 dB.
The area below 0 indication is green, the over modulation area is red, to give the operator a clear indication of the allowed levels.

Because of the built in limiter, over modulation is in principle impossible, although the amount of over modulation can be judged by the number of red LED's that are on. This is an indication of the overshoot of the limiter, caused by its not extremely fast attack time.

A very practical way of working is to modulate such, that the first red LED is only flashing on short peaks, which means that the limiter only takes away small parts of the signal and is not continuous in action with some side effects on the signal quality.

The internal meter is fixed connected to the MAIN DESK OUTPUT, giving a continuous control over the signal sent to the transmitter or tape.

For those people who want to see more, a pair of sockets is available on the connector panel for connection of an external meter, following the control room monitor selection. This meter can than also be used for pre-setting channels for gain and EO using the CUE system.

STATUS INDICATORS

To the left of the meter four LED's are placed, indicating the status of the mixer.

- LIMIT
  An indication of the activity of the limiter circuit. The LED lights for 2 dB gain reduction with full brightness for 10 dB. The timing of the limiter will also be visible.

The best way of modulation is such that the limiter comes only in action on relative short peaks. Continuous, deep limiting has some audible side effects like bringing up noise and ambience when there is a period of silence or modulation of other signals in the mix by activation of the limiter by e.g. a microphone. So use the limiter as a safeguard and not as an excuse for unprofessional operation of the mixer.

- CUE

  This LED indicates that the monitoring is switched to the channels selected in CUE instead of the selection DESK/AIR. This change over is only possible when the AUTOCUE switch is depressed and one of the channels is really in CUE, so with the fader closed.

STUDIO MONITORING

In this part the following items are present:

SPEAKERS STUDIO

This control is for setting the level of the foldback speakers in the studio.

As described before, these speakers may not be aimed direct at the microphones to avoid unwanted coloration of the mixed signal. Also the level has to be kept as low as practical, not to low however to be able to listen comfortable to the signal.

The signal reproduced by the speakers is a total mix of the broadcast MINUS the contribution of the studio microphones, allowing the guests to follow the program and to react on e.g a telephone conversation or a contribution from tape or an external connection.

When the microphones are not live, talkback is possible to the studio via the speakers by depressing the TB button in this studio monitor area. The standing signal will than be reduced by about 20 dB and the talkback signal added to it for reasons of continuous track ft the broadcast.

The safeguard against unwanted talkback with open microphones is fully automatic.

PHONES STUDIO

This is the level control for the headphones of the presenter in the studio.

The signal on these headphones is always the total mix, included the contribution of the studio microphones, to allow the presenter to hear his guests.

This signal will be replaced by a mix of the total signal at low level and the talkback audio on depressing the TB button. This function is independent of the position of the microphone faders. The presenter can also be reached at any time, also during a contribution from the studio in the program.

TALKBACK (TB)

This push button serves for giving commands to the studio and works by lowering the level of the standing signal by about 20 dB and overriding it with the talkback signal at full level.

This function will be inhibited for the studio loudspeakers when one or more microphone channels assigned to studio are opened.

On depressing the TB button, the level of the control room monitors will be lowered by about 20 dB for a better understanding of the commands.

As SOURCE for the talkback signals a flush mounted microphone is used, located between the PHONES control and the TB button. This microphone is followed by a limiting amplifier in order to deliver a constant output level, independent of the voice and the speech distance.

It is possible to replace this signal by the direct output of a microphone channel used by a presenter / engineer in the control room when making DJ type of programmes. This microphone is always lined up and close to the operator and delivers a better sound quality than the internal one.

This option can be activated by setting a jumper on the monitor PCB from INT to EXT and by linking the TB socket on the master connector panel to the DIRECT OUTPUT of the microphone channel involved.

EA853 HEADPHONE AMPLIFIER / STUDIO INTERFACE

Using the EELA EA853 allows you to connect 4 headphones in the studio with individual level control. With just one multi cable connected to the mixing desk all functions are directly available in the studio

5 EXTENSIONS, OPTIONS, LINE UP

EXTENSIONS

Depending on configuration some modules of your mixer will be covered by blank panels. They can be replaced by additional input modules of any type if the standard configuration is not sufficient. The wiring for power, mixing busses and control rails is present, the wiring to the connector panels will be delivered with the modules.

Some remarks have to be made for the connections:

- If a MICROPHONE module is mounted, only the one closest to the TELEPHONE channel has a switch for setting the status to CR or ST. A next microphone channel is always assigned to STUDIO.
- Extension of the number of STEREO CHANNELS is possible simply by adding the modules and the connector panels.
- Addition of a TELEPHONE CHANNEL is a bit more complicated.
  The wiring of the mixer is suited for maximum two telephone modules with the possibility for cross conversation between them when they are ON AIR.
  To realise this, the modules have a 4 pole DIL switch for selecting the clean teed busses to assign the modules as #1 or #2. The first has to be set as #1 by switching 1 and 3 to ON, the second telephone module has to have the switches 2 and 4 in the ON position.

A LINK, marked R2 has to be removed on the MASTER PCB. This link shorts the CF2 bus to ground when not in use to avoid crosstalk.

Also the TALKBACK AUDIO signal has to be linked internally on the connector boards from the one of the first telephone module.
This is the single wire, coming from the TB Cinch connector in the master section.

OPTIONS

The construction of the input modules of the SRM mixer is such, that same internal options can be added, both at delivery or at a later date. This last possibility is a bit more complicated.

These internal options are:

TRANSFORMER BALANCED MICROPHONE INPUTS

In some cases it can be necessary to have fully floating microphone inputs, e.g for use of the mixer coupled to a PA set with combined microphones. The transformers allow for a ground lift to avoid unwanted loops in the grounding system.

It this is done, the phantom power for condenser microphones has to be supplied by the “other” installation!

RIAA PREAMPS for STEREO LINE CHANNELS

This allows direct connection of pick up cartridges to the input channels. Take good care of the wiring in these cases because of the high sensitivity for interference of this type of in- puts, as described in the chapter CONNECTIONS.

METER PANEL

An optional meter panel is also available with built-in Pfl amplifier and speaker and a choice of (extra) meters.

LINE UP

The SRM input modules have a number of internal trimpots. These are used for eliminating tolerances in components and are labelled GAIN and DISTORTION. They are factory set and of a lifetime stability. So if you are not a competent service engineer or you don't have the required instruments, never touch these controls.

The only necessity is there when one or more VCA's have to be replaced because of a defect. Then leave this to the dealer or manufacturer for best results. There are however two line up's that have to be done to adapt the mixer to its environment:
These are the level setting of the OFF AIR inputs and the INITIAL BALANCE of the TELEPHONE HYBRID.

OFF AIR LEVEL

The principle is to match the listening level of the OFF AIR position of the control room monitor selector to the DESK setting.
Do this by alternate the two sources and trim the pots near the OFF AIR sockets for equal level compared to the direct desk output.

This can be done with advantage via an external meter connected to the EXT.METER outputs.

HYBRID BALANCE

This is a more complicated line up, although no instruments are needed.

This trim lines up the bridge circuit in the telephone hybrid for maximum attenuation of the SEND signal in the RECEIVE chain The setting depends on the line impedance of the connection to the first PTT exchange, which is more or less constant as long as the routing is not altered.
For this reason it is the best to use a direct telephone line for the radio studio instead of going through a house exchange giving both better results by more predictable and especially more constant impedance's.

The line up has be done as follows:

> Remove the TELEPHONE module from the mixer with the power supply switched oft, leave all cables connected and position the module in a way that the trimpot close to the two transformers can be accessed
> Take care not to make a short circuit to the frame by putting a piece of cardboard underneath. Switch on the power supply again after a second check for short circuits.
>
> Make a connection, preferably not a local call, switch on the channel with the ON switch and listen to it via the CUE system, of course with the fader closed, otherwise that function will not work.
> Ask the caller, for practical! reasons someone you know, to keep quiet and if possible to cover the mouthpiece of his receiver by a cloth to minimise the signal to the studio.
>
> Now start a recording, preferably with speech signals and open the fader of that machine channel for nominal! modulation, as can be seen on the meters.
>
> Now you hear the crosstalk of the hybrid, that part of the SEND signal that returns from the telephone line via the hybrid and that is what has to be minimised. Use the earlier mentioned trimpot for that purpose.

A "new" desk is lined up on an artificial line, which looks very much like a practical one, but in most cases a local line up has far more better results.
It has advantages to try more connections and find a good mean value and set the trimpot to that value for a good average result.