

c) PEAK

A red LED indicates the peak signal level at the insert send point. It illuminates at a level of approximately 4dB below clipping.

d) Channel Fader

The channel fader has a slide length of 100mm and an exceptionally smooth feel. Infinity cut off is greater than 90dB.

The 8012 Recording Output Module contains a group output section and 2 tape return monitor channels. The lower monitor section is used to monitor either its associated group output, or one of the tape return tracks 1 to 8, depending on its position in the console. The upper monitor section is used to monitor one of the tape return tracks 9 to 16. Additionally, this section can also be used as a line level input during remixing, with its signal either panned into the main stereo mix, or re-routed into its associated group.

1. Upper Monitor Section (Tracks 9 to 16)

a) Auxiliary Sends 1 and 2

Auxiliary Sends 1 and 2 are normally post-fader, but can be switched pre-fader by selecting PRE. In both cases, the auxiliary signal is derived after the monitor ON switch. These auxiliary sends can alternatively be re-routed to Auxiliary busses 3 and 4 by selecting the 3-4 button.

b) Auxiliary Sends 5 and 6

Auxiliary Sends 5 and 6 are permanently post fader and can alternatively be re-routed to Auxiliary busses 7 and 8.

c) PAN

The pan control allows the monitor signal to be panned to the required position within the stereo mix.

d) VOL

The monitor volume control enables the monitor contribution to the stereo mix to be adjusted in level, to allow a satisfactory monitor balance to be achieved during recording or playback of the multitrack tape.

e) Tape Return

The tape return switch selects both the monitor and meter source between either tape send (group output) or tape return (monitor input).

f) ON

The monitor channel and associated auxiliary sends are activated by depressing the "ON" button. Operation is indicated by an adjacent green LED.

g) PFL (Prefade Listen)

The monitor signal can be soloed, independant of the "ON" switch. An adjacent red LED and a master solo warning LED on the Master module indicates PFL operation.

h) LOC (Local)

Under normal circumstances, the monitor signal is routed into the stereo mix. However, if local is selected, the monitor signal will be re-routed into the associated group summing bus. This is especially useful during remixing, when the monitor return can be used as an effects return channel, allowing the effects return to be returned to a sub group. The LOC switch does not allow the group output (tape send) to be rerouted to itself.

2. Lower Monitor Section (Tracks 1-8)

The lower monitor section is similar in operation to the upper section, and provides monitoring of tracks 1-8.

3. Group Output

Each group output controls the signal level to 2 tracks of the multitrack recorder during the recording process. ie. group 1 will feed tracks 1 and 9, group 2 will feed tracks 2 and 10 etc. The group output signal is then normally monitored by the relevant upper or lower monitor section.

NB: The PFL switches in the monitor sections solo the actual signal level at that point in the circuit, which is 4dB lower than the tape send or return level. If desired, changing the PFL summing resistors from 12K to 7K5 will add back this 4dB loss so that the level on the tape and/return meters will be the same as on the master meters when PFL is depressed.

a) SUB

During remixing, the group can be used as a sub-group of a number of input channels to provide a sub mix perhaps destined for additional external processing, via the group insert patch point. Operation of the SUB switch allows the group signal to be re-routed directly back into the stereo mix. When SUB is depressed the monitor volume control and tape return switch have no function. Additionally the ON switch controls the group signal path, and the PFL source is the pre fade, pre ON switch group level.

The 8014 Auxiliary Master module contains the master level controls for the 8 auxiliary outputs, a sweep frequency oscillator, and a stereo headphone drive amplifier. The auxiliary and oscillator outputs are electronically balanced.

1. Auxiliary Master 1-8

Each of the 8 auxiliary busses has an overall master level control. An associated AFL (after fade listen) button allows the signal at the auxiliary to be monitored and metered.

2. Oscillator

a) PAD

The oscillator output can be reduced by a 40dB attenuator. This allows the signal at the oscillator jack to be used for testing microphone channels.

b) Frequency

The oscillator frequency is continuously variable between 45Hz and 15kHz in 2 ranges; 45Hz to 900Hz and 700Hz to 15kHz.

c) ON

Switches the oscillator on.

d) Level

Adjusts the level of the oscillator. When used in conjunction with the 40dB pad, a wide range of output levels can be obtained, suitable for testing microphone and line level inputs.

e) GRP (Group)

The oscillator can be routed to all output groups for alignment and test purposes.

f) Aux

The oscillator can be routed to all 8 auxiliary busses.

3. Headphone Output

The headphone output allows monitoring of the main stereo mix output, and also of any soloed signal. It will drive headphones of all high impedances, though use of 8ohm phones at high listening levels may cause slight headphone crosstalk into the stereo mix.

The 8015 Master Module is common to all versions of the Series 800B and Series 1600 consoles. It contains the main electronically balanced stereo output, monitoring facilities and the talkback system.

1. Monitor Section

a) Monitor Source

Four interlocking push buttons, 2Ta, 2Tb, 2Tc, and Mix, route the signals from 3 external stereo sources or the main stereo mix output to the Control Room, Studio and Headphone monitoring systems. The inputs are electronically balanced.

b) Studio Level

The signal from the monitor source selection is fed via the Studio On switch to the Studio Level control.

c) Control Room Level

The Control Room monitoring signal can be adjusted in level. The source signal is either the output of the monitor source selection or the signal from any soloed (AFL/PFL) source. Because the stereo meter drive is derived from the signal to the Control Room monitor pot, any signal which is monitored will also be metered. This allows any auxiliary, or channel pre fader signal level to be metered.

d) MNO (Mono)

To provide a mono compatibility check of the stereo signal, the left and right monitor channels can be summed together by the MNO switch. This is operative on both the Control Room and Headphone monitoring systems, and does not affect the main stereo output.

e) DIM

The Control Room and Headphone monitoring systems can be attenuated by a set amount (20dB).

2. Talkback Section

a) Comms Select

The Talkback signal can be routed into any or all pairs of auxiliary outputs, to allow free use of the auxiliaries for either foldback or effects send purposes.

b) Mic Level

Adjusts the level of the talkback signal.

c) Slate

Depressing Slate routes the talkback signal to all group outputs, with the addition of a 30Hz tone. This is useful for identifying sections of the multitrack tape recording, as the low frequency tone can be heard during rewinding of the tape even though the tape is not in contact with the replay head.

d) Comm (Communicate)

The talkback signal can be routed to any or all of the auxiliary busses (selected by the Comms Select switches), to enable communication to the foldback systems.

e) Talkback

This is similar to "Comm" but with the signal also being routed to the studio monitoring system, independantly of the level or ON switch.

2.05 8018 24 TRACK MONITOR/EFFECTS RETURN MODULE

This double width module contains 8 identical sections of monitoring for tape returns 17-24. No tape send/return switching is provided since it is assumed that all 24 track machines already contain this facility, usually under remote control. Each of the monitor sections are identical.

1. Effects Return Input Sockets

8 sockets are provided to allow use of each monitor section as an effects return channel in mix-down. Normally each monitor section will receive the relevant tape return signal from tracks 17-24 of the 24 track machine. However, if a plug is inserted into a socket, the tape return signal will be replaced by the signal on the plug.

2. Monitor Section (1 of 8 identical sections)

a) Auxiliary Sends 1 and 2

Auxiliary Sends 1 and 2 are normally post-fader, but can be switched pre-fader by selecting PRE. In both cases, the auxiliary signal is derived after the monitor ON switch. These auxiliary sends can alternatively be re-routed to Auxiliary busses 3 and 4 by selecting the 3-4 button.

b) PAN

The pan control allows the monitor signal to be panned to the required position within the stereo mix.

c) VOL

The monitor volume control enables the monitor contribution to the stereo mix to be adjusted in level, to allow a satisfactory monitor balance to be achieved during recording or playback of the multitrack tape.

800/1600 STEREO INPUT MODULE

1. Channel Input Section

The Channel can be operated using either Line Input A OR Line Input B OR both. Both Line Inputs are balanced, with an Input Impedance of greater than 10kOhms, which is high enough to interface to any normal professional peripheral equipment without loading the source.

a) B

Line Input B may be selected by pressing the button marked B.

b) $\emptyset$ (Phase)

Pressing the Phase button will invert the phase on the left-hand input only to correct for any input mismatch.

c) INPUT GAIN

The Input Gain can be varied between -10dB and +30dB of gain using the Gain Trim control.

d) L & R.

i. With both these switches out the module works in stereo mode.

ii. With either L or R switched in, both channels of the module are fed by either the left or right input.

iii. With both switches depressed, both channels of the module are fed by a mono sum of the left and right input.

2. Equaliser Section

The Equaliser is a versatile unit, allowing 5 areas of control over the audio spectrum. All amplitude pots are centre detented for easy zeroing.

a) (High Pass Filter)

The High Pass Filter operates at 100Hz with an ultimate slope of 12dB/Octave. This will effectively remove low frequency stage rumble, and other extraneous signals.

b) HF (High Frequency)

15dB of boost or cut is available at 10kHz, with a "shelving" characteristic, ie. the slope of the EQ curve does not keep rising with frequency, but having reached the desired amount, flattens out or "shelves" from that frequency on.

c) HI MID

15dB of boost or cut is available at 2kHz, with a "bell" characteristic, ie. having reached maximum amplitude, (or minimum in the case of cut) the amplitude response returns to zero on either side of the frequency. The shape of the curve, when plotted shows a characteristic "bell" shape. The Q of the net-work (a measure of bandwidth) is 1.5.

d) LO MID

The Lo Mid section is identical to the Hi Mid section, with the exception that the frequency is 300Hz.

e) LF (Low Frequency)

15dB of boost or cut is available at 60Hz, with a "shelving" characteristic.

f) EQ

The Equaliser circuitry can be switched in and out of the signal path, independent of the high pass filter.

3. Auxiliary Section

There are 4 Auxiliary send controls available. Each control can be routed to 2 auxiliary busses, to give a total of 8 Auxiliary sends for use as echo, foldback or other auxiliary effects units.

a) SENDS 1 and 2

Auxiliary sends 1 and 2 are normally post-fader, but can be switched Pre-fader by pressing the appropriate PRE button. In both cases they are post equaliser. Auxiliaries 1 and 2 can be selected mono or stereo and dependent/independent of the channel mute by push-on links.

b) SENDS 3 and 4

Auxiliary sends 1 and 2 may be routed to auxiliary busses 3 and 4 by pressing the 3-4 button. Auxiliaries 3 and 4 can be selected mono or stereo by push-on links.

c) PRE

Auxiliary sends 1-4 may be routed Pre-fader by pressing the PRE button.

d) SENDS 5 and 6

Auxiliary sends 5 and 6 are permanently post-fader.

e) SENDS 7 and 8

Auxiliary sends 5 and 6 may be routed to auxiliary busses 7 and 8 by pressing the 7-8 button.

4. Routing Section

The channel input signal may be routed to any of the pairs of Group Outputs (1-2, 3-4, 5-6, 7-8) and the Stereo Mix, by selecting the relevant routing button.

a) BALANCE

The balance control corrects or deliberately creates any errors in stereo matching before the fader and the auxiliary sends.

b) ROUTING

Selection of any routing button assigns the channel signal to a pair of output groups, or the stereo mix, via the balance correction.

5. Channel Status Section

a) ON

The channel "ON" status is indicated by a green LED. When a channel is switched off the auxiliary sends 1 and 2 are not necessarily cut and depend on the the set up of the push-on links (jumpers).

b) PFL

Pre-fade Listen Soloes the Pre-fader independently of the ON switch. This gives a mono check of signal before the on/off switch. PFL operation is indicated by a red LED on the channel, and a master warning LED on the master module.

c) PEAK

A red LED indicates the peak signal level. This gives visual warning that the higher signal Right or Left before the insert points, is within 5dB of clipping.

5. Channel Status Section

a) ON

The channel "ON" status is indicated by a green LED. When a channel is switched off, all auxiliary sends are also switched off, with the exception of the signal to the insert send jack.

b) PFL

Pre-fade Listen Soloes the Pre-fader, post insert jack signal independently of the ON switch. This gives a mono check of signal, after the insert points but before the on/off switch. PFL operation is indicated by a red LED on the channel, and a master warning LED on the master module.

c) PEAK

A red LED indicates the peak signal level. This gives visual warning that the higher signal Right or Left before the insert points, is within 5dB of clipping.

d) CHANNEL FADER

The channel fader is an accurately matched stereo fader of slide length 100mm and an exceptionally smooth feel. Infinity cut off is greater than 90dB.

2.06 PATCHBAY

The Patchbay system installed in the 24 channel Patchbay version of the Series 1600 greatly enhances the versatility and ease of use by providing convenient, standardised connection points to approximately 200 salient sections of the console circuitry and to 32 pieces of external equipment. This allows rapid insertion of external signal processing equipment into the system, by means of patchcords, and is also very useful for fault-finding.

The Patchbay is clearly divided into 5 sections, thereby making it a simple matter to locate the required connection point in the console.

1. Masters

The first row of the patchbay is devoted to the main stereo outputs and to the stereo tape returns. A signal interrupt point is provided by the Left and Right Mix Insert Send and Return Jacks, which allow a limiter or other device to be inserted into the stereo mix immediately before the stereo fader.

The insert send is normally internally connected (normalled) to the return jack, to maintain signal flow under normal circumstances. When a patch cord is plugged into the insert send, the signal flow is not affected, merely appearing in parallel on the patch cord. However, insertion of a patch cord into the return jack will break the normal signal flow, and replace it with the signal on the patch head. In this way, the signal on the insert send jack can be connected via one patch cord to the input of an effects device, while the output of the effects device can be inserted into the insert return jack to replace the normalled signal.

Patch points are also provided to allow access to the outputs of 3 stereo tape recorders, and to the inputs of the stereo tape recorders.

The second row contains the 8 auxiliary outputs. These are used to patch the auxiliaries into effects devices such as echo and reverberation units, etc. Also on this row are 2 sets of 4 jacks which are connected in parallel with each other. These are used when an output is required to be connected to a number of inputs. They must not be used to connect a number of outputs together.

The oscillator jack enables the console's internal oscillator to be used for fault finding and tape alignment purposes.

2. Input Channels

Each input channel is provided with 4 patch points, allowing access to the channel line input, insert send and return (immediately preceding the channel fader), and the channel direct output, which can be used to provide an additional effects send from a channel. All 24 input channels have identical patch points, split into 4 channels per row, with a total of 6 rows.

3. Groups/Monitors

Each group module is equipped with 4 patch points for access to the group insert send and return, the group output and the upper monitor return section of each group module.

Each group output is normalled to the input of 3 tracks on the multi-track recorder eg. Group 1 normals to tracks 1, 9 and 17. The output of the tape recorder is normalled to the relevant monitor section, ie. tracks 1-8 normal to monitor return 1-8 (not available on the patchbay); tracks 9-16 normal to monitor returns 9 to 16 (available on the patchbay for use as additional line level inputs when not being used for monitoring), and tracks 17 to 24 normal to the monitor inputs of the 24 track Monitor Module, if fitted. Patch points are situated at the top of this module for use as line 1 level effects returns.

4. Tape Tracks

Patch points allow access into the input of each tape track. Insertion of a patch cord into the tape send will interrupt the normal signal from the relevant group output and replace it with the signal on the patch cord. Inserting a patch cord into the tape return jack will not interrupt the signal, but allows tape return to be re-routed to alternative input channels or monitor return inputs. Each tape return is normalled to the relevant monitor return, and also to the relevant channel hire inputs.

5. Effects

32 pairs of patch points are provided for connection of external peripheral equipment. Interface is by 2 multiway Elco connectors on the underside of the console. Each pair is nominally labelled send or return, although they are all connected identically, and can be used to suit the individual purposes of the studio.

3.00

OPERATION: RECORDING CONSOLE

3.01 INTRODUCTION

The Series 1600 console, when fitted with standard input and recording output modules has been designed to provide the varied facilities required in a modern 8 or 16 track recording studio.

The recording process can be broken down into 4 basic sequences:-

1. **Record Mode:** Recording direct from microphone or line input on to the multitrack tape recorder.
2. **Multitrack Playback:** Listening to what has been recorded.
3. **Overdubbing:** Building up the track complement while listening to what has already been recorded. Basically, this is a combination of one and two.
4. **Mixing:** Combining all the recorded tracks, various effects, echo etc. to form the final stereo mix.

An understanding of the signal flow through the mixer is best gained by a study of the block diagram. A detailed explanation of all controls and switches has been given in Section 2, and it is assumed that the user is familiar with basic multitrack recording methods.

3.02 Record Mode

1. Record Mode

This is the basic starting point in making a recording. Input channels are placed in the microphone mode by placing the mic/line switch to the "up" position. The signal is routed to the desired console group output by depressing the relevant group routing button on each channel. The signal can be routed to more than one group if desired and panned between any odd and any even numbered groups by using the channel pan pot.

The signal now appearing at the console group output to the tape recorder can now be metered on the relevant VU meter, if the Tape Return button is in the "up" position. A monitor balance is achieved using the appropriate monitor level control, and depressing the monitor "ON" button. The monitor pan pot will pan the signal between left and right speakers.

The various gain and level controls throughout the signal chain may now be adjusted to set the operating levels for optimum signal to noise ratio and headroom conditions.

2. Channel and Group Faders

In general the channel and group faders should be set to approximately the "0" position. This will enable the engineer to increase the level by 10dB, or fade out completely, while normally operating in the fine resolution area of the fader travel. The absolute position of the fader is not too critical, but situations where for example, the channel fader is operated at -30dB, with the associated group fader at +10dB should be avoided, as this will degrade the normally excellent signal to noise ratio of the console. Similarly, the opposite case runs the risk of distortion.

3. Microphone Gain

Having set both channel and group faders as above, the microphone gain is set to give the required level at the group output. The amount of gain required depends on the type of microphone, the sound pressure level developed by the instrument or voice and the distance between the sound source and microphone. In the case of high output microphones and high sound pressure levels, it may be necessary to switch in the -20dB attenuator pad, to prevent overloading the input stage.

4. Insert Patch Points

In situations where the dynamics of the input signal are very wide, or unpredictable, or where a particular effect is required, it is often necessary to patch in a limiter/compressor, equaliser or effects device into the signal chain. This can be done either via the channel insert jack points or the group insert jack points. Which insert point is used will depend on the actual effect required. If only an individual instrument is to be controlled, the channel insert point would be used. However, if a number of microphone channels have been mixed to a single group, such as backing vocals for instance, then it will be necessary to patch in an overall limiter, using the group insert point.

5. Headphone Mix

During recording, it is of course essential for the musicians to hear what they and everybody else are playing. Headphone mixes are derived using any or all of auxiliary sends, either direct from the input channels, or alternatively from the group monitor channels. Deriving the headphone mix from the monitor channels has the advantage of giving musicians a mix when replaying the recording just made, which is useful when overdubbing to enable them to hear their performance. However, a mix derived from the monitor channels during recording will be affected by any "gain riding" of the channel or group faders, which may cause problems. This can be avoided by using the input channel auxiliary sends 1 and 2, switched to Pre. The headphone signal is thus derived from before the channel fader, and is unaffected by any change in the position of that fader.

6. Echo Sends

Similarly to headphone mixes, echo sends may also be generated from either the input channels or the group monitor channels. However, unlike headphone mixes the echo send is usually required to be derived after the fader, so that it is proportional to the fader level. This ensures that the percentage of direct signal to echo signal remains constant, regardless of fader position. This condition is met on Auxiliary sends 1 and 2, if Pre is not selected, and also on sends 3 and 4.

The individual situation determines whether the echo send is derived from the input channel or the monitor channel. Normally if the echo return signal is to be recorded on the multitrack tape recorder, then the echo send will be derived from the input channels. However, if it is an echo mix for monitoring purposes only, then it would be more relevant to use the monitor channel sends to derive the echo send mix.

7. Echo Returns

The output of the echo or reverberation device, ie. the echo return signal, patched into either unused input channels if this signal is required to go to tape, or alternatively patched into unused monitor channels (9 to 16 only) if it is only monitor echo and no input channels are available.

If an input channel is used, the return signal is patched into the line input jack of the relevant channel. The channel must then be placed in the line mode by depressing Ll. Line gain and fader position can now be set up as described for microphone inputs.

The echo return signal can now be routed to the stereo mix, or to the group outputs if it is required to record the echo signal, by depressing mix or any of the group routing buttons.

Naturally, when an input channel is used as an echo return, the echo return signal may also be sent to any of the auxiliary sends. This allows echo to be added to the headphone mix if desired. It is also possible of course, to send the echo return to the echo send, and create a loop in the echo system. When used with a reverberation device the effect will be to increase the reverberation time. When used with a tape delay, it will create multiple delays, decaying in amplitude in a manner controlled by the send and return levels. This effect must be handled with care. If the echo send level from the echo return channel is set too high, the entire system will start to feed back, and go into oscillation, instead of decreasing in amplitude with each successive loop.

If it is not required to record the echo return signal, but merely to route it direct to monitors, select Mix. This will route the signal directly to the stereo mix buss, which is also the monitor signal in this mode.

3.03 Multitrack Playback

Having actually recorded something on the multi-track recorder, it is necessary to be able to listen to the recording. This is achieved by simply selecting Tape Return. The effect of this is to connect the individual group monitor to the output of the relevant track of the multitrack tape recorder. Therefore the signal from the recorder now follows the same monitor signal path as the group output signal had been using. This means that all levels, and panning, and therefore the monitor mix, will remain the same as they were when the recording was being made. This is assuming, of course, that the tape machines are correctly aligned.

The facility of monitoring the playback of the multitrack in this way enables the operator to perform a quality control check even while the recording is being made, providing that the following precautions are observed.

1. Isolation between control room and studio is good. If not, the musicians will hear a delayed version of what they are playing.
2. Headphone mix is derived from the channel faders, otherwise the musicians will again hear a delayed version.
3. Noise reduction system is either a simultaneous encode-decode system, or not used. If it is not a simultaneous encode-decode system, then while recording, it will be in the encode mode, and the signal presented to the console monitor return will probably be the tape recorder line input signal ie. the signal from the console group output. In this case, depressing Tape Return will appear to have no effect, as it will be the same signal in both cases. This particular effect can in some cases be put to good use, however, when overdubbing as it allows the monitor switching to be achieved automatically, depending on the mode of the tape recorder. This will be dealt with more thoroughly in Section 3.04 which describes the overdubbing process.

3.04 Overdubbing

Overdubbing is the process of building up a recording track by track, while listening to all the previously recorded tracks. As such it will be clear that this is a combination of the previous 2 modes, with some channels in the microphone (Recording) mode, and some monitors in the Tape Return (Multitrack playback) mode.

Any tracks which have already been recorded are monitored by selecting the relevant Tape Return buttons. A headphone mix may be set up using the auxiliary sends on these monitor channels. Monitor echo can also be similarly set up, as already described in Section 3.02.

Input channels in the microphone mode, which will form the overdub signal can now be routed to the output group corresponding to the track to be recorded, in exactly the same way as described in Section 3.02.

At this stage, a decision must be made as to the source of the headphone mix signal from the overdub channels. It can be derived from either the input channel or the group monitor channel or both.

1. Input Channel

In this case, the musician always hears himself. However he will not be able to hear his previously recorded signal off the tape recorder, if he needs to "drop in" in the middle of a take.

2. Monitor Channel

If the headphone mix is derived from the monitor channels, the engineer can decide whether the musician hears himself live or the previously recorded signal off tape (sync playback), by depressing the Tape Return button. However he will not be able to hear both, and a vocalist for instance may find it difficult to match levels, and intonation.

This switching can sometimes be achieved automatically if a noise reduction unit is in use, and with some tape recorders. If the tape recorder or noise reduction unit is arranged to switch its sync output to line input whenever the tape recorder is in stop, fast forward, rewind, or record, and to only switch to sync playback off tape when the tape recorder is in the play mode, the automatic switching condition will be met.

All that is now usually necessary when overdubbing is to select Tape Return on the relevant monitor channel, and let the tape recorder and/or noise reduction system perform the work of monitor switching.

The musician now hears himself live at all times except when the tape recorder is actually in the sync play mode, when he will hear his previous signal off tape. This method saves the engineer from continually switching monitor sources, but can only be used with certain tape recorders.

3. Both

If derived from both, then the musician is able to hear himself live, via the input channel, and his previous recording via the monitor channel, if it is switched to monitor Tape Return, until the moment of entering the record mode. At that point, most tape recorders will switch from sync playback to line input. The effect of this will be a slight increase in the level of that signal in the headphone mix, due to the addition of the input channel signal and tape recorder line input signal, which is effectively the same signal.

3.05 **Remix Mode**

When all recording and overdubbing has been completed, the console is placed in the remix mode.

Remixing is the process of combining all the previously recorded tracks, together with any special effects devices such as harmonisers, flangers and delay lines. It is in this mode that the outstanding flexibility of the Series 1600 becomes most apparent.

The remix mode is entered by selecting line input on each input channel (L1), which connects the output of the multitrack tape recorder to the input of each channel, via the line trim control. This is normally set approximately to the "0" position if the recorder is operating at a nominal +4dBv line level. However, like the mike trim, it can be adjusted to allow the channel fader to work near its nominal "0" position.

By selecting MIX, the channel can be routed directly to the stereo mix, via the channel pan pot, to allow positioning of the signal within the stereo perspective.

Alternatively the signal can be routed to any of the output groups, to enable the formation of sub groups. Up to 8 mono or 4 stereo sub groups can be used, which can then be re-routed back into the stereo mix.

Additionally, the upper monitor section on each output module can be used as a line level input which can be routed either into the stereo mix by selecting SUB, or into its associated sub group, by selecting LOC (Local).

If the 8018 24 track monitor module is fitted (24 input patchbay consoles only) the outputs from 8 effects devices can be plugged into the 8 jack sockets at the top of the double width module from any of the 32 tie lines on the patchbay.

Limiters or other effects devices can be connected to any input channel or output group by patching into the relevant insert points.

All 8 auxiliary sends can be used to feed echo or effects devices, as headphone mixes are not required during remixing.

The composite stereo mix will be controlled in level by the stereo master fader and the level metered on the stereo mix meters. These meters will also indicate the level of any solo signal, or any of the stereo tape returns selected on the monitor source buttons. The monitor source selection buttons allows quality and level checking of the stereo recording in the same way as described for multitrack playback.

4.00 Installation

4.01 Applying Power

Before switching on the Series 1600 check that the mains voltage selector on the power supply unit is set to the correct mains voltage for your area, and that the fuse is of the correct rating.

For operation on voltages between 220 and 240VAC, the fuse should be rated at 3.15Amps, 20mm Anti surge.

For operation on voltages between 100 and 120 VAC, the fuse should be rated at 6.3 Amp, 20mm.

Do not replace the fuse with any other type, as this could become a safety hazard, and will void the warranty.

4.02 Interface Levels

The Series 1600 is normally supplied to provide compatible level interfacing with standard professional equipment. ie. +4dBv (ref 0.775 Volt). However, provision has been made to allow the user to modify the tape interface levels for use with semi-professional equipment, operating at -10dBV (Ref 1.00 Volt), or other levels.

The change in tape interface operating level is accomplished by reducing the console group output level by the required amount, and increasing the console monitor return gain by an identical amount, and can be done by the addition of 3 resistors and the removal of a link on each group output circuit board.

Group Output Level

1. Remove Link 18.
2. Add resistor in position RX1.

Monitor Return Gain

Add resistor in position RX2 and RX3.

The value of the resistor must be the same for RX1, RX2 and RX3 for correct operation. For interface to the normal Teac/Tascam operating level of -10dBv, a value of 1Kohms is suitable.

Control Room, Studio And Auxiliary Outputs

Power amplifiers are often rated at 300 mV sensitivity, for full output. In such cases, an attenuator should be installed at the input of the power amplifier, to attenuate the +4dBv (1.228V) signal from the console, by approximately 10 to 15dB. This can be achieved by using a 2.2K ohm series resistor, and a 680 ohm, shunt resistor across the amplifier input.

4.03 Connector Conventions

All XLR type connectors are normally wired to the following standard:

Pin 1	Signal Earth (Screen)
Pin 2	Signal Low
Pin 3	Signal High

With the exception of the Recording Output Module, all other main outputs are electronically balanced. The Soundcraft output stage allows either the Signal High or the Signal Low to be grounded, without any change in output level or other adverse effects, in the same way as a transformer coupled output. This allows much greater freedom when interfacing to a variety of unbalanced and balanced equipment and a minimisation of earth loop problems.

In most cases, the output can be viewed in the same way as a transformer coupled output, but without the signal degradation inherent in transformer design. However, if the output is driving a long cable run into an unbalanced input, it is usually beneficial to unbalance the output directly at the console to avoid instability.

NB: During the currency of this manual this XLR standard may be changed to the European (DIN) convention of:

PIN 1	Signal Earth (Screen)
PIN 2	Signal High
PIN 3	Signal Low

If in doubt, check with your dealer or the manufacturer at time of purchase.

4.04 General Wiring Procedures

To take full advantage of the excellent signal to noise ratio and low distortion of Soundcraft consoles care must be taken to ensure that incorrect installation and wiring does not degrade its performance. Hum, buzzes, instability and Radio Frequency Interference (RFI) can usually be traced to earth loops and inferior earthing systems. In some areas, especially heavy industrial areas, the incoming mains earth will not be adequate, and a separate technical earth for all the audio equipment must be supplied. However, check with your local electricity supply company to ensure that safety regulations are not infringed or negated.

The successful, hum free, installation of a systems requires careful forethought, and the establishment of a set of ground rules, which must be consistently adhered to at all stages of the installation.

1. Initial Wiring Considerations

- a) For optimum performance, it is essential for the earthing system to be clean and noise free, as all signals are referenced to this earth. A central point should be decided on for the main earth point system, and all earths should be "star fed" from this point. It is common electrical practice to "daisy chain" the earths to all electrical outlets but this method is unsuitable for audio installations. The preferred method is to run an individual earth wire from each outlet, back to the system star point to form the safety earth and screen reference for each piece of equipment. A separate earth wire should also be run from each equipment rack and area, to the star point. This may or may not be used depending on circumstances, but it is easier to install in the first place, than later, when problems arise.

The location of the star point should be a convenient, easily accessible place preferably at the rear of the console, or in the main equipment rack.

- b) Install separate "clean" and "dirty" mains outlets, wired individually back to the incoming mains distribution box. Use the "clean" supply for all audio equipment, and the "dirty" supply for all lighting, vending machines, etc. Never mix the two systems.
- c) If necessary, to provide sufficient isolation from mains borne interference, install an isolating transformer for the "clean" supply. The isolation transformer should be provided with a Faraday Shield which must be connected to earth.
- d) Never locate the incoming mains distribution box near audio equipment, especially tape recorders, which are very sensitive to electro-magnetic fields.
- e) Ensure that all equipment racks are connected to earth, via a separate wire back to the star point.

- f) Equipment which has unbalanced inputs and outputs may need to be isolated from the rack to prevent earth loops.

2. Audio Wiring

Having provided all equipment with power and earthing connections, consideration must be given to the method of providing audio interconnection, and adequate screening of those interconnections. This must be done in a logical sequence to avoid problems, and assist in the localisation of problem equipment.

- a) Connect Control Room Monitor system to the console, and check for any hum, buzz or Radio Frequency Interference. Only when you are satisfied with the quietness of the console and monitor system should you proceed to the next step.
- b) Connect multitrack tape recorder, via noise reduction system if applicable, and again check that the system is still clean.
- c) Connect stereo tape recorders, studio monitors, echo and foldback sends, one at a time, checking and isolating any connection which degrades performance.
- d) Connect all peripheral effects devices.
- e) Connect all microphone lines.

By following this sequence much time and future trouble will be saved, and the result will be a quiet, stable system.

3. Shielding

Audio equipment is supplied with a variety of input and output configurations, which must be taken into consideration when deciding where the screen connections should be made. There are three sources of unwanted signal being impressed on the screen, which are as follows:

- i) Extraneous electrostatic or electromagnetic fields.
- ii) Noise and interference on the earth line.
- iii) Capacitive coupling between the screen and the signal wires.

To minimise the adverse affects of the unwanted coupling to the signal wires, it is important that the screen is connected at one end only. ie. the screen must not carry any signal current. Any signal on the wires within the screen will be capacitively coupled to the screen, and this current will ultimately be returned to the source of the signal, either directly, if the screen is connected at the signal source end, or indirectly via the entire earthing system, if the screen is connected at the signal destination end. The indirect connection will cause an increase in high frequency cross talk, and should be avoided wherever possible. Therefore, in general, always connect the shield only at the signal source end. In high R.F. areas, the screen can also be connected to earth via a 0.01 micro farad capacitor. This will present a short circuit at

R.F. frequencies, thus lowering the effective shield impedance to ground. However, at low audio frequencies the reactance of the capacitor will be sufficiently high to not cause an earth loop problem.

Combinations of unbalanced, balanced, and electronically balanced (differential) systems mean that there are nine interconnection permutations. The optimum connection of the screen in each case is shown in Table 1.

TABLE 1

	OUTPUT	INPUT	SCREEN
1)	Unbalanced	Unbalanced	Source
2)	Unbalanced	Balanced	Source
3)	Unbalanced	Differential	Source
4)	Balanced (Note 1)	Unbalanced	Destination
5)	Balanced	Balanced	Source
6)	Balanced (Note 2)	Differential	Destination
7)	Differential (Note 3)	Unbalanced	Source
8)	Differential	Balanced	Source
9)	Differential	Differential	Source

Note 1 The shield is connected to the destination earth point, which is opposite to normal practice, because the signal wires being shielded are referenced to the input earth, not the output earth.

Note 2 If the output transformer is centre tapped to earth, the screen should be connected at the source.

Note 3 When an active differential output is operated in unbalanced mode, it is very important that the output current returns to earth via the shortest, least reactive route. Check for instability at the output.

Nb. a) In all cases, use good quality twin screened audio cable. Do not use single screened cable.

b) Always connect both conductors at both ends, and ensure that the screen is only connected at one end.

c) Do not disconnect the mains earth from each piece of equipment. This is needed to provide both safety and screen returns to the system star point.

d) Equipment which has unbalanced inputs and outputs may need to be electrically isolated from the equipment rack and/or other equipment, to avoid earth loops.

It is important to remember that all equipment which is connected to the mains is a potential source of hum and interference, and may radiate both electrostatic or electromagnetic radiation. In addition, the mains will also act as a carrier for many forms of R.F. Interference, generated by electric motors, air-conditioning units, Thyristor light dimmers etc. Unless the earth system is clean, all attempts to improve hum and noise levels will be futile. In extreme cases there will be no alternative but to provide a completely separate and independent "technical earth" to replace the incoming "noisy" earth. However, always consult your local electricity supply authority to ensure that safety regulations are not being infringed.

5.00 MAINTENANCE

Although all Soundcraft Consoles have been designed with long term reliability in mind, it is inevitable that occasional maintenance will be required. However, due to the amount of attention given to the problems of maintenance during the design stages of this console, and the modular construction, servicing tends to be extremely simple to carry out, with the minimum of test equipment needed to isolate and rectify faults.

5.01 General Fault Finding

With the exception of the electronically balanced microphone amplifier, all signal electronics are configured around high slew rate, low noise integrated circuits. The microphone amplifier is a proprietary design, utilizing a discrete transistor, noise cancelling front end, differentially summed via a low noise integrated circuit.

The use of integrated circuits means that the majority of audio faults can be repaired by simply replacing the I.C., having first isolated the fault to a particular stage in the signal chain. The isolation can often be done without even removing the module from the console, by judicious use of the insert points, and/or switching the module to various modes. As with all servicing, a good knowledge of the basic signal flow is necessary for best results. Each module should be viewed as a number of signal blocks, through which the signal must flow. If the signal appears at the input to a block, but not at the output, then the fault lies within that block. By dividing a module into individual sections, what at first appears to be an extremely complicated piece of equipment can be simplified into a series of simple sequential stages. This is the basic first move in all types of fault finding, and usually requires nothing more than a certain amount of logical thought. Servicing a mixing console is more a matter of clear thinking and having an understanding of what should be happening, than having a highly developed technical knowledge.

To illustrate the method of logical fault finding, let us assume that we have a non functioning input module, in both microphone and line modes.

The first step is to ensure that a fault really does exist! Check that the module is in the correct mode of operation, and that no jacks are inserted in the insert points, which may be interrupting the signal flow.

If in doubt about the module operation, set up an adjacent module in exactly the same way, which will allow a direct comparison between a working and possibly non-working module. Route the channel directly to mix, so that the channel may be monitored in the normal way. Using the internal oscillator set to approximately 1kHz, patch the oscillator signal into the channel line input. If all is well, an undistorted signal should now be heard. More likely, because of the fault, it won't.

Large sections of the module circuitry can be bypassed by switching the Phase Reverse, or switching out the Hi-pass filter, and the Equaliser.

If switching out a section causes the signal to re-appear, then the fault is located in that section, which can then be traced at component level, by removing the module from the console frame, and reconnecting it via extender cables.

With the module installed on extender cables, access is now available to all parts of the module, and the signal may be traced through the various stages, using an oscilloscope, millivoltmeter, or even high impedance headphones. Refer to the Block Schematic which shows the

signal flow through the modules. When a point is reached where the signal is not present, or is distorted, the probable faulty components can then be checked and if necessary replaced. Integrated circuits, due to their internal complexity, are the most likely cause of problems, followed by mechanical components such as switches and faders, which are susceptible to physical contamination from oxidation, dust, and liquids.

5.02 Removing Modules

Remove the 2 module retaining screws, which will allow the module to be carefully withdrawn from the console. The ribbon cable will now be exposed, and may be detached from the module. The module will still have some cables attached, but these are sufficiently long to allow the module to be completely withdrawn from the console. Extender cables can now be plugged into the main ribbon cable, and the module, taking care not to twist the extender cable. Although damage to the module will not be caused by plugging and unplugging the module with the power still switched on, this is not recommended for the inexperienced, as it is possible to bend the connector pins if care is not taken.

5.03 Removing Patch Cards

The entire patchbay is made up from rows of plug in patch cards, constructed with printed circuit mounting jack sockets. Each row can be easily removed for cleaning and replacement if this should prove necessary.

Remove the 4 patch bay retaining screws, located at the corners of the patch bay. This will allow the entire assembly to be lifted up and tilted to one side. Take care to avoid scratching the paint.

Locate the card which is to be removed and unplug the ribbon cables connected to it, and if necessary, to the adjacent cards. Remove the 2 screws holding the card to the panel and withdraw the card from the assembly.

Each individual jack socket has a dust cover which can be unclipped for cleaning or adjusting the contacts.

Reassembly of the patch bay is the reverse of the above procedure.

5.04 Meter Alignment

Each V.U. meter has its own individual drive card attached to the rear of the meter. This card also contains the detection and drive circuitry for the Peak LED which is pre-set to indicate a peak level of 8dB above OVU.

OVU is normally adjusted to indicate a line level of +4dBv, ie. a level of 1.228 volts. However, it can be re-adjusted to indicate a different line level if required by the pre-set potentiometer on the drive card.

Connect a millivoltmeter to the group output. Route the oscillator set to 1kHz to the group and adjust the group output level to read the

required level on the millivoltmeter. (Normally this would be +4dBv). Adjust the VU drive pre-set to indicate OVU on the VU meter and repeat for all other groups and the Stereo Mix meters.

Note that the Peak LED will always indicate a level of 8dB above whatever the OVU level has been set to.

5.04 Lamp Replacement

Illumination of the VU meters is provided by 2 wire ended lamps in each meter. These are 9 volt lamps wired in parallel. The lamps in each group of 4 meters are wired in series and powered by the ± 17 volt audio supply. A series resistor provides turn on surge current limiting to prolong the lamp life.

The stereo mix meters have an additional series resistor to simulate the voltage drop of the missing pair of meters.

To replace the lamps, first remove the VU meter. This is best done by applying upward pressure to the underside of the meter front cover and then pulling the top of the cover forward. The 2 lamps will now be visible and can be unsoldered and replaced.

It is recommended that both lamps be replaced even if only one has failed, as the remaining lamp will have been overstressed and its life substantially reduced.

5.05 Power Supply Servicing

The Series 800B and 1600 power supply provides the following regulated supply rails;

- i) ± 17 volts, Audio
- ii) ± 24 volts, (not used)
- iii) ± 7.5 volts, Logic
- iv) ± 48 volts, Phantom Power

If a power supply fault is suspected, first ensure that it really is the P.S.U. which is at fault, and not a short circuit in the console. This can be checked by disconnecting the P.S.U. from the console, and measuring the voltage at the connector. A load across the supply should be provided, to simulate the normal load conditions imposed by the console.

A 10ohm, 20 watt resistor across each of the audio supply rails and a 20ohm 5 watt resistor across the +24 volt rail is suitable. The phantom power supply can be loaded with a 2.2kohm, 1 watt resistor.

The ripple and noise value of the various supply rails can now be measured, using a millivoltmeter or an oscilloscope, and a value of at least -80dB (ref 0.775v, DIN audio should be obtained on the audio) and the phantom supply rails.

If a fault is found to exist in the P.S.U., disconnect the mains supply and remove the cover. Check visually for any obvious problems, such as

blown fuse, burnt components, etc. If nothing obvious is observed, reconnect the mains, and measure the voltages across the various electrolytic smoothing capacitors, which should be as follows;

Audio Supply C12, C14 = +26 volts
 C13, C15 = -26 volts

+24 Volt Supply C1, C2 +36 volts

Logic Supply C32 = +13 volts
 C33 = -13 volts

Phantom Supply C8 = +59 volts

Differences of + 10% are acceptable, due to variations in the incoming mains voltage. If satisfactory, the problem lies in the regulator section. If not however, check the bridge rectifier, smoothing capacitor, and transformer for failure.

Audio Supply

Theory of Operation (numbers in brackets refer to negative regulator). The bipolar audio supply is a dual, tracking regulator, with overvoltage protection, capable of supplying up to 5 Amps at ± 17 volts.

The amount of current drawn is sensed by the voltage drop across the parallel resistors, R33, R34, R15 for the positive rail, and R35, R36 for the negative rail. As the current drawn increases to cause a voltage drop approaching 0.6 volt, TR17 (TR18) will begin to turn on, which in turn, will turn on TR15 (TR16). This will starve TR5 (TR7) of base current, from the current source, TR6 (TR8) and therefore start to turn TR19 (TR20), (the main pass transistors) off, thereby reducing the output voltage.

Output voltage regulation is by means of TR9, and TR10. (TR11, TR12). TR9 (TR11) compares the voltage across zener diode ZD2 (ZD3) with the voltage across the output, via the potential divider R41, R39 (R42, R40), and attempts to keep them equal. For example, if the output voltage start to rise, the voltage at the base of TR9 (TR11) will start to exceed the emitter voltage set by the zener diode and the transistor will start to turn on. This will turn TR10 (TR12) on which will starve TR5 (TR7) of current, turning TR19 (TR20) off, and therefore reducing the output voltage accordingly.

The positive and negative regulators are made to track each other by means of the cross connection diode D7, and TR13 (TR14). Under normal circumstances, when both output voltages are equal, the base of TR13 (TR14) will be approximately at the mid point of the supply rails, ie. at 0 volts + the diode voltage drop.

However, if for example the negative supply should start to fall, the mid point voltage will now move positive with respect to the 0V rail, and will start to turn TR13 on. This will starve TR5, TR7 of current and therefore reduce the positive supply voltage to regain equilibrium between the supplies.

Servicing

As the bipolar audio supply is arranged as a tracking regulator, a fault on one half of the supply will usually affect the other half in the same manner.

The tracking facility can be disabled for test purposes by connecting each end of diode D7 to ground.

All transistors and diodes should be checked for normal operation, and replaced if necessary. Bear in mind that a failure of one semiconductor will often cause the failure of other associated components.

NB: During the currency of this manual, the PSU circuit diagram may change. However, the theory of operation and the component numbers referred to above will remain the same.

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SERIES 800B/1600 CUSTOMER SPARES KIT - COMPLETE KIT PART NO: RZ2235

AZ2201	THERMISTOR RA53	1
AZ2202	RES NETWORK 10Kx6	1
AZ2204	8P S.I.L. 4x4K7 RES NETWORK	2
AZ2206	8P S.I.L. 4 x 7K5 RES NETWORK	2
BA0001	DIODE IN4148	3
BB0106	ZENER DIODE 400MW 11V	1
BC0204	BDG RECT KBP02 200V 1.5A	1
BC0207	BDG RECT KBL02 200V 4A	1
BD0301	PNP TRANS 2SA842GR/2SA970GR	1
BD0302	NPN TRANS 2SC1681BL/2SC2240BL	5
BD0306	NPN TRANS BC300	1
BD0311	PNP TRANS BD136	1
BD0312	PNP TRANS PN4 355	2
BD0315	PNP TRANS TIP2955	1
BD0316	NPN TRANS TIP3055	1
BD0317	NPN TRANS BD135	1
BE0403	QUAD OP AMP IC TL074	1
BE0404	SGL OP AMP IC TL071	2
BE0407	SGL OP AMP IC TDA1034/NE5534	1
BE0413	DUAL OP AMP IC TL072	5
BE0428	DUAL OP AMP IC NE5532	3
BE0430	V.REG +1.2/37V 1.5A (T0220)	1
BE0431	V.REG -1.2/37V 1.5A (T0220)	1
BE0438	V.REG LM338K+1.2/32V 5A (T03)	1
BF0502	QUAD 21/P NAND 4011	1
BF0505	QUAD ANGL SWITCH 4016	1
BF0517	QUAD ANALOGUE GATE DG308/CJ	1
DA0001	PREH S/SP C/D + BRAKT 4K7LIN x 2	1
DA0002	PREH S/SP 41P/D 10K + LOG	3
DA0003	PREH S/SP 41P/D 10K + LOG x 2	1
DA0004	PREH S/SP 41P/D + BRKT 10K - LOG	1
DA0005	PREH S/SP C/D 47K LIN	1
DA0014	PREH S/SP 10K LIN	1
DD0316	ALPS 90MM MONO FADER SA012C	1
DE0401	CERMET TRIMMER HORIZ 90H 2K2	1
DE0409	CERMET TRIMMER 90V 1K	1
DF0512	C&K PCB MNT MICRO SWITCH	1
DF0515	RAFI ILLUM PUSH SWT LATCHING	1
DF0516	ALPS (SUN TYPE) 2 POLE PUSH SWT	4
DF0517	ALPS (SUN TYPE) 4 POLE PUSH SWT	2
DL1001	NON-ILLUM ROCKER SWT DPST BLK	1
DJ0801	MARQUARDT TAP CHANGE	1
HB0124	S2400/S3B MNS TX TR7228	1
JA0001	TOSHIBA MINI LED GREEN TLG102	1
JA0002	TOSHIBA MINI LED RED TLR102	1
JA0003	TOSHIBA LED RED TLR104	1
JB0110	RS WEDGE BULB 12V T5.5	1
KA0027	ALPS SA012C FDR KNOB WHITE	1
KA0028	ALPS SA012C FDR KNOB RED	1
KA0029	ALPS SA012C FDR KNOB YELLOW	1
KA0030	11MM/LINE 4MMBRN KNOB RE-AN	2
KC0227	C&K 0.45"SQ SWT CAP RED	1
KC0228	C&K 0.45"SQ SWT CAP GREY	1

Series 800B/1600 Spares List cont.

KC0229	C&K 0.45"SQ SWT CAP GREEN	1
KC0230	RE-AN 11MM KNOB CAP BLACK	2
KC0231	RE-AN 11MM KNOB CAP BLUE	1
KC0232	RE-AN 11MM KNOB CAP GREEN	1
KC0233	RE-AN 11MM KNOB CAP GREY	1
KC0234	RE-AN 11MM KNOB CAP ORANGE	1
KC0235	RE-AN 11MM KNOB CAP RED	1
KC0236	RE-AN 11MM KNOB CAP YELLOW	1
ZD0305	3.15A ANTI-SURGE FUSE	1
ZD0309	FUSEHOLDER (SCHURTER)	1

SOUNDCRAFT RECOMMENDED WARRANTY

(This warranty applies to sales within the UK and should form the basis of the warranty offered by the overseas vendor of Soundcraft products).

1. 'Soundcraft' means Soundcraft Electronics Ltd.
'End User' means the person who first puts the Equipment into regular operation.
'Dealer' means the person other than Soundcraft (if any) from whom the End User purchased the Equipment, provided such person is authorised for this purpose by Soundcraft or its accredited Distributor.
'Equipment' means the equipment supplied with this manual.
2. If within the period of twelve months from the date of delivery of the Equipment to the End User it shall prove defective by reason only of faulty materials and/or workmanship (but not faulty design) to such an extent that the effectiveness and/or usability thereof is materially affected the Equipment or the defective component should be returned to the Dealer or to Soundcraft and subject to the following conditions the Dealer or Soundcraft will repair or at its option replace the defective components. Any components replaced will become the property of Soundcraft.
3. Any Equipment or component returned will be at the risk of the End User whilst in transit (both to and from the Dealer or Soundcraft) and postage must be prepaid.
4. This warranty shall only be available if:-
 - a) the Equipment has been properly installed in accordance with instructions contained in Soundcraft's manual; and
 - b) the End User has notified Soundcraft or the Dealer within 14 days of the defect appearing; and
 - c) no persons other than authorised representatives of Soundcraft or the Dealer have effected any replacement of parts maintenance adjustments or repairs to the Equipment; and
 - d) the End User has used the Equipment only for such purposes as Soundcraft recommends, with only such operating supplies as meet Soundcraft's specifications and otherwise in all respects in accordance with Soundcraft's recommendations.
5. Defects arising as a result of the following are not covered by this Warranty: faulty or negligent handling, chemical or electro-chemical or electrical influences, accidental damage, Acts of God, neglect, deficiency in electrical power, air-conditioning or humidity control.
6. The benefit of this Warranty may not be assigned by the End User.
7. End Users who are consumers should note their rights under this Warranty are in addition to and do not affect any other rights to which they may be entitled against the seller of the Equipment.

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