Input Channel
48v mic phantom switch
Line & mic input gain pots
Peak light and phase switch
Tape input gain trim

The 'input enable flip' button
[allows the global remix button to switch all inputs to the main fader]
Routing
Large/small fader routing switch
with LED
Large fader routing: Direct
centre, L/R1 and L/R2, with LED
Aux sends
5, 6 and stereo 7-8
Small fader section
Hi and Low EQ
Solo and Mute
Small fader routing: Direct
centre, L/R1 and L/R2
'From main' allows a feed from the large fader (switchable pre or post)
to input the small, fast direct centre
output or 5.1 positioning
Signal present LED and pan pot

Aux Sends
Large fader sends 1 to 5
with pre/post switch and
individual mute switches

Classic EQ
An exact reproduction of the
original Series 40 EQ
Switchable Hi-shelf: 12k/3k
Sweep Hi-stop: 1kHz-15kHz
Sweep Low-stop: 100Hz to 1kHz
Switchable Low shelf: 60Hz/120Hz
50Hz Low-cut switch
EQ switch with LED

Large fader pan
Small/large fader flip, with LED
Signal present LED
Large fader Solo and mute buttons
[soft switch control with centre
csection global override

100mm Large fader
[motorised optional]

Finalising section
The Series 80-5.1 features a finalising section that includes a stereo 'Hi-Def' 8 band
EQ, with variable Hi and Low filters and a stereo Sonicomp - Optical/solid state
compressor/limiter. Both can be routed to any output or the patchbay for individual
channel patching

Master section
Oscillator (50Hz, 5kHz, 1kHz)
and Talkback, routable to: external, Aux 7/8, sub outputs or 5.1
Playback Stereo returns 1 to 4 and global 5.1 monitor button
Master Remix mode switch [flips all tape return inputs to large fader and gain line
inputs to small fader]
Sub 1 output mixer (L/R1, L/R2 & centre) with master pot. Console Mono switch
Monitor control section can be switched between 5.1 setup and main/earfields or aux
Headphone output with gain control
Monitor level control is an accurate stepped rotary with '0' position set at reference level
The BEQ PRO24 is a professional 5.1 audio console from the Father of British EQ John Oram. The Oram BEQ PRO24 has been designed for the studio that requires the classic warm sound that Oram Sonics delivers, together with today’s sound-around mix functionality. It is the result of John Oram’s vision of a truly professional 8-buss console for the Pro Studio owner.

**KEY FEATURES:**

- 4 band EQ-magic™ Equaliser
- Low and High filters
- 2 band EQ on small fader
- 24 track outputs, with direct out switch
- 8 group faders / sub mix faders
- 8 AUX sends with mute and pre/post switching
- 8 more AUX sends via the small fader routing (total of 16)
- Large/small fader flip
- 5.1 routing with sub-output mix matrix
- Automation ready
- Global ‘ solo kill ’ and ‘ remix ’ mode
- Signal present- large and small faders
- 8 stereo returns with EQ
- Stereo Hi-Def EQ
- Stereo Sonicomp limiter
- LED channel metering
- External ‘ silent run ’ PSU

Oram Professional Audio has a reputation for designing and building quality analogue consoles. The experience to fully understand what great audio is exists at ORAM, our designers all being musically and technically qualified alike. The ORAM BEQ Series 8 console is now replaced by the PRO24 professional analogue 8-buss console. The facilities required of a console today are incorporated into this design and are highlighted above.

**OPTIONAL EXTRAS**

At ORAM we like to put the customers requirements first, to facilitate this we offer the new PRO24 console with a host of optional extras. This enables the customer to purchase the perfect console for their room. Listed below are some of the extras available but at ORAM we will customize to your requirements.

- ORAM BODY KIT
- ORAM RACK PATCH BAY
- ORAM MOVING FADERS
- ORAM METAL KNOBS (Any colour)
- ORAM STAND
- ANY COLOUR POLYCARB (Limited to 2000 different colours)

**FEATURES**

**Channel**

The first control is the MIC gain associated with this is the +48 phantom switch, the MIC input selector, and the phase switch. These three switches enable the +48v phantom to be individually switched on each channel; the MIC input to be disabled when not be used and the phase of the MIC input to be reversed.

The MIC pre is based around an Analog Devices amplifier used by John Oram in all his low noise designs. The design architecture is unique to John Oram in that by using a special potentiometer designed and made by him, the preamp has a controlled gain swing from +70dB gain to unity. This enables full line levels to be driven into the MIC
pre, in fact up to +22dBu. This enables to line sources to be mixed together in the channel. The MIC pre is paddless and transformerless for high dynamic handling and transparency.

Below the MIC control is the line level control this gives a swing of + and – 15dBu. It also has an input selector to enable the signal. Between the MIC and line controls is the peak LED this comes after the MIC and line inputs have been mixed and starts to turn on at +18dBu leaving 10dBu headroom before clipping.

Next is the tape level control as with the line input this gives a range of + and – 15dBu. Alongside the tape control is the I/P flip switch, this will set the channel up in REMIX mode shown by the illumination of the green LED. When the REMIX switch is pressed in the master section all channels with the green I/P flip LED illuminated with have the MIC/line input swapped with the tape input. The LED then turns red to show that the channel is in remix mode.

The signal going to the subgroup routing can come from the small or large fader via the route switch. This signal can then be sent to 1 and/or all the buss pairs, the pan pot can be used to send the signal to either of the buss pairs.

Below the subgroup routing is the DIR switch, this is used for 24-track recording of the large signal. The console is essentially an 8 buss console this means that channel 1,9,17 etc and 2,10,18 etc have the same subgroup output, but with the DIR switch pressed the direct large signal form that channel goes to the subgroup output.

Below the DIR switch is the large fader routing, this enables the large signal to be sent to left/right 1, left/right 2 and center. These buttons can be used for 5.1 recording as well as stereo recording and mono.

To the left of the routing section are the small fader AUX sends, AUX5, AUX6, AUX 7 and AUX 8. AUX 5 is factory set to post fader via a jumper on the channel, there is a rotary control which adjusts the amount of signal sent to the master AUX send on the master section. The AUX 5 signal is mixed with the AUX 5 signal from the large fader. AUX 6 has a similar level control but also has a pre/post switch which selects the signal pre or post the 60MM fader. AUX 7 and 8 are a stereo pair, with a pan control to position the signal in the stereo mix as well as a level control, as with AUX6 there is a pre/post switch.

Note: All AUX’s are post EQ at all times.

Next is the small EQ control, there is a high and low shelving EQ with + and – 15dBu of level control. The high EQ shelf is set to 10kHz and the low is set to 100Hz.

The next section is the small signal control functions, there is a soft mute and solo both of these controls are momentary which enables the global solo kill and the solo distruct (this is explained in more detail under the master section solo description.)

Below this is the routing for the small signal as with the large signals it can be sent to left/right 1, left/right 2, and the center buss. There is also a DIR switch that mixes with the large DIR signal before being sent out the back of the console. The from main switch disconnects the tape signal from the just before the input to the EQ and places the large signal into the small EQ. The large signal can be pick up pre or post the large fader via the switch.

The 60mm fader controls the level of the small signal (except when the fader flip switch is pressed) and has an additional 5dBu of gain. Next to the fader is a signal present indicated which is pre the fader, this enables you to know that there is a signal on the small fader before it is pushed up.

Last in this section is the pan control, this enables you to move the small signal from left to right, the signal is 3dBu down in the middle.

The next section is the large AUX sends, AUX 1 to AUX 5, each AUX as a level control and its own mute and pre/post fader switch, again all AUX’s are post fader at all times.
Below the AUX’s is the large EQ section

It is the equaliser section of each channel that actually shapes the tonality of your sound. Whether you are reproducing the sound of a violin or a tractor, the realism of that sound is entirely dependant on how you have adjusted the EQ. Therefore, the EQ section can be your best friend or your worst enemy in a difficult audio situation. Be careful not to over-EQ your sound system. More tweeters and woofers are destroyed from over use of an EQ than all the other reasons combined. If you feel that you need more than 10-12dB of boost in you low or high EQ, you need to look elsewhere in your system for a problem. In a live sound system, this might include readjusting the output levels of your active crossover, output levels and equalisation of your signal source devices, adding subwoofers to your system, etc. Are your power amplifiers clipping? Maybe you’ve run out of power or your system just isn’t big enough for the room. Expand to a bigger system instead of damaging your equipment by over-EQ’ing it.

The BEQ-pro24 mixers have a very active, musical sounding EQ section that also features some useful tools. At the bottom of the EQ section is the EQ in switch, this enables the EQ to be switched in and out easily for comparing the difference in sound. To the left of this switch is a low filter, this control rolls off any objectionable, low frequency audio you don’t want in your mix such as stage rumble, room boominess, pops, and breath or wind noise. The great thing about the low cut filter is that its cut off frequency can be swept from 5Hz to 200Hz. You can set the filter to remove only the objectionable part of the signal and leave the rest untouched. There is also a high filter that is useful for removing any unwanted high frequencies signals such as tape hiss or other externally generated noises. The filter ranges from 1k5 to 50kHz, this makes it a very useful tool.

The EQ itself is a 4 band with two mid sweep. The mid sweep ranges from 65Hz to 2kHz and 1k5 to 15kHz. As you can see from the broad range, the mid controls can become a low or high EQ, as well. The EQ is incredibly flexible. Plug in your favourite tape or CD and experiment with the different settings to familiarise yourself with the ways it can help you. Try it with good headphones to appreciate the subtle nuances of the circuit. Remember: don’t over-EQ, as it can be dangerous to your sound system. Sometimes, less gives more. Try reducing the mids down to render more apparent highs and lows. This is called subtractive equalisation. Try it. It works.

**HIGH FILTER**
Selects the frequency at which high frequency attenuation begins. The signal rolls off at 12dB per octave above the selected frequency. The filter ranges from 1k5 to 50kHz.

**HIGH SHELF**
Provides plus or minus 15dB of boost or cut of high frequencies shelving at either 7kHz or 12kHz.

**HIGH MID SWEEP CONTROL**
Selects the center frequency to be boosted or cut in the peak/dip high midrange EQ filter section. The control ranges from 1k5 to 15 kHz

**HIGH MID LEVEL CONTROL**
Provides up to 15dB of boost or cut at the center frequency of the high mid sweep EQ filter section

**LOW MID SWEEP CONTROL**
Selects the center frequency to be boosted or cut in the peak/dip low midrange EQ filter section. The control ranges from 65Hz to 2 kHz

**LOW MID LEVEL CONTROL**
Provides up to 15dB of boost or cut at the center frequency of the low mid sweep EQ filter section

**LOW SHELF**
Provides plus or minus 15dB of boost or cut at low frequencies shelving at either 50Hz or 150Hz
LOW FILTER
Selects the frequency at which low frequency attenuation begins. The signal rolls off at 12dB per octave below the selected frequency. The filter ranges from 5Hz to 200Hz.

Below the EQ section is the large fader pan control this determines the amount of individual channel signal sent to the left and right master outputs and allows you to place that signal anywhere in the left to right stereo panorama. It also functions as a routing control for the odd and even numbered subgroups.

Alongside the pan is the fader flip this sends the large signal to the 60mm fader and the small signal to the 100mm fader.

Below them is the large faders mute and solo, both being soft to enable the solo kill and solo destruct function to work.

Between the mute and solo is the large signal present, this picks the signal up post EQ but pre the fader.

The 100mm fader control the large signal being sent to the master section and is automation ready.

MASTER SECTION
The master section is split up into three sections the group module of which there are eight; this controls the master AUX sends, the stereo AUX returns, and the master subgroups.

Next is the EQ/COMP module, there are two of these, they control the master mixing of left/right1, left/right2 and center. These modules also control the positioning of the SONICOMP compressor and HI-DEF EQ in the mix.

The final module is the monitor this controls the speaker output, talkback, solo, and console mode.

GROUP MODULE
At the top of the group module is the master AUX send control, these controls the amount of mixed AUX 1 sent out the rear of the console. Each AUX has its own solo and mute.

Below the AUX sends are the stereo AUX returns, each one has a trim control that will give an adjustment of + and – 15dB. Below the trim control is the routing, this enables to AUX return to be sent to the left/right1, left/right2, center and any of the subgroups.

Next is the stereo EQ, this gives a high and low shelving EQ with + and – 15dB of cut and boost. The high shelves at 10kHz and the low at 100Hz.

The mono switch will take the signal coming into the left side of the AUX return and feed it to both left and right, any signal on the right side will be disconnected when this switch is in.

The balance control will adjust the amount of signal that goes to the left and right and is 3dB down in the middle.

Each AUX return has a solo and mute switch and a stereo 60mm fader that controls the main level of the AUX returns being routed.

The last section is the master sub group control; each group once mixed can be routed to left/right 1 and/or left/right 2. There is a pan control to position the signal in the left/right panoramic, and each group as a mute and solo. In the panel below is a 100mm fader that controls the final amount of signal set out the rear or back to the channel.

EQ/COMP
This module is split into three sections, first is the world acclaimed HI-DEF EQ, below this is the routing section and finally the SONICOMP compressor.
The HI-DEF is the ultimate EQ mastering tool and has some great features

Eight bands of overlapping adjustment
Unique combining of shelf and bell response curves
Bandwidth parameter adjustment on two mid bands
Separate equalizer and filter bypass

Ultra low noise and high musicality with original **Oram sonics**

The first control is the high filter this sweeps a steep high frequency cut from outside the audio band at 80kHz to within at 1500Hz perfect for: the removal of RFI, super-sonics, system hiss and noise optimising band limiting. Below this is the high sweep frequency and level control. This controls a bell shaped response in the high audio spectrum from 5kHz to 18kHz with a cut and boost control of 18dB, there is also a bandwidth control that switches between narrow and wide.

Below these controls is the first of the shelving function, the high shelf. This has switchable turnover points of 5kHz and 16kHz and a cut and boost control with a 36dB range.

The next four controls are bell shaped responses in the middle of the audio spectrum. The high mid sweep covers frequencies from 1kHz to 9kHz, with a cut and boost control of 18dB; there is also a bandwidth control that switches between narrow and wide. Next is the low mid with a frequency range from 250Hz to 2.5kHz, with a cut and boost control of 18dB, there is also a bandwidth control that switches between narrow and wide.

Below the mids is the other shelf control, the low shelf. This has switchable turnover points of 150Hz and 40Hz and a cut and boost control with a 36dB range.

The next two controls are the low frequency sweep and level. This stage controls a bell shaped response at the low audio spectrum from 35Hz to 500Hz with a cut and boost control of 18dB; there is also a bandwidth control that switches between narrow and wide.

The next two switches are, filters in and EQ in these enable the filters and EQ to be individually switched in and out for easily comparison.

The next control is the low filter this sweeps a steep low frequency cut from outside the audio band at 5Hz to within at 300Hz perfect for the removal of sub-sonic, rumble and microphone popping and for tightening kick drums and effects.

The last control is the input gain; this is at the front end of the EQ and provides level adjustment of 15dB either side of unity gain (central point)

The next section is the HI-DEF and SONICOMP routing. This allows either or both the EQ and compressor to be placed in the signal path of the subs, left/right 1, left/right 2 and patch. The switches are a radio type so only one from each section can be routed at any given time. The patch position will make the EQ and/or compressor available at the patchbay so that they can be used on individual channels when not used for mastering.

The final section is the SONICOMP compressor dual-attenuator limiter-compressor; this represents the combination of the old and the new. The compressor features a vintage optical attenuator and state of the art semi-conductor attenuator and can give immediate A-B comparison between round knee and precision level adjustment. The control parameters are infinitely adjustable between wide range limits: added to the dynamic and exciting analogue feel of **Oram sonics**.

The first control is the make up gain or output level, this is used to bring the compressed signal back to the correct output level. The gain adjustment goes from 0dB to +15dB.

Next is the release control this gives a release time from 0.05 seconds to 3 seconds. Below this is the threshold control which gives between –25dBu and +20dBu and then off. The ratio control is next and varies between 1.4:1 and 30:1 (full limit ), Below the ratio is the attack control which can a time control from 0.1 milli-seconds and 40 milli-seconds. The final control is the input level this will control the amount of signal being sent to the compressor.

The bottom two switches change the compressor from solid state to LDR mode and switch the compressor in and out.
On this PCB there are 7 presets these are used for the calibration of the compressor. During testing these presets are set for optimum operation, but can vary over time and may need adjusting periodically. The list below shows the operation of each preset.

VR1  LDR bias adjust
VR2  LDR make up gain
VR3  Ratio limit
VR4  Meter adjust
VR5  Meter calibration
VR6  Ratio Null
VR7  Gain reduction adjust

MONITOR

The monitor module is split into 6 sections, talkback/oscillator, and playback returns, remix, solo, sub-band, and speaker selection.
At the top of the talkback section is a 3 pin XLR for the talkback MIC (PIN1=GND, PIN2=HOT/+ , PIN3=COLD/-), to activate the talkback the talk button must be pressed, this is situated near the master faders.

Below the MIC input is the routing for the talkback and oscillator, this enables the talkback and oscillator to be sent to the 5.1 mix, subs, AUX 7/8 and an external output at the rear of the console. The level control is used to adjust the amount of signal for both the talkback and oscillator. The oscillator has an enable switch and a rotary selector to change the frequency from 50Hz, 1kHz and 15kHz.

The next section is the playback returns; this enables the console to switch between listening to the console output mix and playback. The first switch is playback when this is pressed the control room output changes from what the console is sending out to the signal being played back to the console. There are 5 different inputs available, 4 stereo returns RTN1 to 4 and a 5.1 switch that will playback all 6 sources. All the playbacks go via the master control room level at the bottom of the monitor module, and are displayed on the VU meters.

The next section is the console remix mode. When this switch is pressed the MIC/line inputs will be swapped with the tape input on any channel that has the I/P flip switched de-pressed.

The solo section contains the master solo level; this sets the monitoring level of any soloed signal. To the right of this level control is the solo kill switch, when this is pressed any solo on the entire console will be deactivated.

The solo mode of the console can be changed from destructive to non-destructive by using the mode switch. In destructive mode when any solo is activated everything that is not soloed will be muted (except for the AUX send and returns), the dim level control will have no effect in this mode. This mode or operation is useful for listening to sources with any effects. (As a result of using AUX send controls). In non-destructive mode the level control will bring everything that is not soloed back into the mix, eventually at the maximum level the whole mix will be audible.

The sub-band/Mono mix produces the sub woofer signal for the 5.1 mix and also a mono mix of the 5.1 sent to left/right 1.

Each input, left/right 1, left/right 2, and center to the to the sub-band mix comes pre the main faders and has its own level control to adjust the amount of signal being sent to the sub-band mix. Once all the signals are mix there is a master sub-band control that adjusts the final amount of signal sent to the speakers and the recording device.

To the right of the sub-band is the mono switch, this takes all the 5.1 signals and mixes them together into a mono signal, this signal is then sent to left/right 1. Mono signal id then presented to the front left and right monitors for easy evaluation of phase coherency.
The final section is the speaker selection; the Pro24 has a wide variety of speaker outputs that can be individually selected. The 5.1 switch enables the left/right1, left/right2, center, and sub to be sent to the speakers. The main switch puts the left/right signal to the left and right speakers of the 5.1 group. The nearfields are a separate output and derive their signal from the left right. The AUX switch enables another set of speakers to be connected if required. The level control adjusts the amount of signal going to the AUX output as well as the headphone output.

Finally there is the master control room level control; this is a precision stem control that adjusts the output level of the control room. This control will adjust all the 5.1 signals at the same time and keep there relative levels the same.

In the panel below is the master 100mm faders for left/right 1, left/right 2, and centre, each one has its own mute. The C to L/R1 will take the signal that post center fader and mix this into the left/right 1 buss. The L/R2 to L/R1 will take that signal post the left/right2 fader and mix it with the left/right1 signal.

To the right of these controls is the talk switch, when the button is press the talkback is activated.

**METERBRIDGE**

**CHANNEL**

The channel is monitored by a 12 LED bar meter; this signal follows the large (100mm) fader signal path. It also has a peak LED which is set to start coming on at +22dBu, there is also a signal present LED which can be set to pre or post fader by way of a jumper. (The factory setting is pre)

**MASTER SECTION**

The master section has two metering sections, the LED bar meters monitor the subs and derives their signals post the sub faders. The six VU meters monitor the 5.1 output that is being sent to the recording device or the playback returns. In addition, any soloed signal will appear on the front left and right VU meters.

**INPUTS AND OUTPUTS**

The inputs and outputs are by way of 90way EDAC connectors, 1 EDAC will have 4 channels of in and outs on it. The master section has 4 EDAC’s to handle all the inputs and outputs. The EDAC connectors are on the bottom of the console towards the rear. Hence if the console was ordered without a stand (for desk top or table mounting) provision should be made for the cable exits.

**WIRING CONNECTIONS**

**CHANNEL**

**FOR EACH SET OF 4 CHANNELS (1 x 90 WAY EDAC)**

<table>
<thead>
<tr>
<th>SIGNAL</th>
<th>EDAC PIN IDENTIFIER</th>
<th>SIGNAL</th>
<th>EDAC PIN IDENTIFIER</th>
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<tbody>
<tr>
<td>0V SIGNAL</td>
<td>A</td>
<td>0V SIGNAL</td>
<td>M</td>
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<tr>
<td>MIC1 IN -</td>
<td>B</td>
<td>RETURN1 -</td>
<td>N</td>
</tr>
<tr>
<td>MIC1 IN +</td>
<td>C</td>
<td>RETURN1 +</td>
<td>P</td>
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<tr>
<td>0V SIGNAL</td>
<td>D</td>
<td>0V SIGNAL</td>
<td>R</td>
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<td>E</td>
<td>SEND1 -</td>
<td>S</td>
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<tr>
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<td>F</td>
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<td>RTN3 LEFT +</td>
<td>BR</td>
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<td>RTN4 RIGHT -</td>
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<td>RTN4 RIGHT +</td>
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<td>RTN3 RIGHT -</td>
<td>BT</td>
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<td>5.1 RTN LEFT FRONT -</td>
<td>CD</td>
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<tr>
<td>RNT3 RIGHT +</td>
<td>BU</td>
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<tr>
<td>5.1 RTN LEFT FRONT +</td>
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<td>0V SIGNAL</td>
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<tr>
<td>5.1 RTN RIGHT FRONT -</td>
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<tr>
<td>5.1 RTN CENTER -</td>
<td>CU</td>
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<tr>
<td>5.1 RTN RIGHT FRONT +</td>
<td>CJ</td>
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<tr>
<td>5.1 RTN CENTER +</td>
<td>CV</td>
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<tr>
<td>0V SIGNAL</td>
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<tr>
<td>0V SIGNAL</td>
<td>CW</td>
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<tr>
<td>5.1 RTN LEFT REAR -</td>
<td>CL</td>
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<tr>
<td>5.1 RTN SUB BAND -</td>
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<tr>
<td>5.1 RTN LEFT REAR +</td>
<td>CM</td>
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<tr>
<td>5.1 RTN SUB BAND +</td>
<td>CY</td>
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<td>0V SIGNAL</td>
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<tr>
<td>5.1 RTN RIGHT REAR -</td>
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<tr>
<td>5.1 RTN RIGHT REAR +</td>
<td>CR</td>
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</table>

**THE POWER SUPPLY**

The power supply for the BEQ series is housed in a 2U high rack case and provides outputs for bipolar rail (plus and minus 18 volt), the 5 volt line for LED and switching functions and a 48 volt line for phantom powering of capacitor microphone. The line input, along with the d.c. output, are fused and accessible on the panel of the supply.

**Remember the importance of good ventilation at all times**

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**Operation Hints**

Now you are getting familiar with the function of each control and connection of your BEQ-PRO 24 mixer, it's time to make music. The next section will show you how to connect your mixer and use it effectively and efficiently. Before beginning, there are a few precautions that should be noted.

**OPERATING PRECAUTIONS**

Your Oram BEQ-PRO 24 mixing console is well protected from any external faults. However, we recommend following these common sense precautions.

1. **Safety Instructions**
   
   Read and follow all of the safety warnings of this manual and on the separate safety precautions page enclosed with the mixer.

   **CAUTION:**
   
   Do not expose the unit to water or other liquids.

   Always unplug the unit if water is present. Failure to do so can result in injury or death from electric shock.

2. **Grounding**
Your BEQ-PRO 24 mixer is supplied with a three conductor, grounded power cord and plug. Connect the unit only to a properly grounded mains outlet. Do not use a ground lift adapter or otherwise attempt to defeat the ground on the plug. Failure to properly ground the unit can result in damage to the mixer or other equipment connected to it and represents a dangerous safety hazard.

3. Line Voltage

Operate from AC mains not more than 5% above or below the specified line voltage. Failure to comply may invalidate your warranty.

4. Pre-Connection Caution

Always switch off the power and set all output level control(s) to minimum before making any connections. This will eliminate any chance of unexpected, loud audio transients that could damage your speaker systems.

A. OPERATION

Please read the entire section on operation before you start.

Output levels

Before proceeding, set all the output level controls to minimum and make sure that the power to the mixer and all devices being connected to it are switched off. After making all your connections to the mixer and related equipment, switch the power on.

Phantom Power Switch (48 Volts DC)

The advantage of this switch is that there is one on every channel. Did you know that 48 Volts DC applied to the balanced output of a keyboard, effects device or other sound module could possibly destroy that device's memory circuits. It is very dangerous to connect anything except a MIC into the MIC input on a mixer that has only one phantom power switch for the whole mixer. Because your BEQ-pro24 mixer has a phantom switch on each channel, you can plug any kind of device into any channel. No restrictions! Just make sure the phantom power is switched off (up position) on that channel before making any connections. Switch on the phantom power only if the MIC for that channel requires it.

Input select switch

Set the input select switch to match the input source or sources you will be using for each channel, remember lines are fine in MIC inputs.

Vocal Microphone

Start by plugging the MIC you intend to use into a mixer channel. Set the MIC trim control at half (12 o’clock). Move the channel fader to – (OFF). If you are the person that will be speaking or singing into that MIC, please proceed. If not, that person will have to repeatedly speak or sing into the MIC at the loudest possible level that they will use during the performance or session. Turn the MIC trim control clockwise until the peak LED begins to flash. Back the control off (counter clockwise), just a little, so the LED barely begins to flash on the loudest words or singing parts. You will now have the proper gain setting and the best possible signal to noise ratio on that channel. Remember that you may have to reset the gain for different vocalists that have stronger or weaker voices.

Line or Tape Inputs
Use the same method as the MIC except connect your tape deck, keyboard or other line level source into the appropriate input. Play the loudest possible passage of music on the tape or instrument being used and set the line trim so the peak LED only begins to flash on the loudest signal peaks. Remember your BEQ-pro24 mixer has two trim controls: one for the MIC and one for the line inputs.

Peak Overload LED

This is a warning light to tell you that you are within 18dB of distortion on that individual channel. If the LED comes on most of the time during use of that channel, your input gain control may be set to high. If the light never comes on your input gain is probably to low. Setting the gain to high will result in distortion that you can hear. Setting the gain to low may force you to operate the channel fader at near the end of its travel, at full volume, and you still may have enough output level from that channel. Additionally, a low setting at the input requires that you raise the level of other devices in the signal path higher than normal, increasing the noise level in your sound system or recording.

Phase Reverse Switch

This is a very handy tool for any live soundman or engineer in the studio. It very simply reverses the polarity or phase of that channels MIC input. Why would you want to do that? Here are a few examples of when you might wish to take advantage of this feature. If you have a MIC, or more often, a MIC cable that is wired incorrectly, you can fix it without having to pull out the soldering pencil. Hit the phase reverse switch and the problem is solved. When using two overhead cymbal MIC’s on a drum set, some engineers will purposely switch one MIC out of phase. This causes the low and mid frequencies to cancel each other out and delivers only the cymbal sizzle. In studio works you may have the need for an “ AM radio mix “ of a mutli-track master tape. A well-recorded master will usually have a stereo guitar, keyboard or chorused vocal tracks that may phase cancel themselves and disappear completely in a mono mix. Thus the need for a separate, in phase mono mix. Simply use the phase switch on either the left or right channel (but not both) of the mystical disappearing stereo track and it may suddenly reappear in the mix. Audio magic! Not really, just good use of a great feature on your BEQ-pro 24.

The “absolute phase” discussion is also easily solved with this switch. Many audiophiles feel it is important for the speaker cone on your woofer to exactly mimic the motion of the bass drumhead as struck by the drummer. In other words, if the drumhead is moving forward, towards the listener, the speaker cone should do the same. While that may be a good idea, your entire sound system must also be in perfect phase. That means the polarity of pins 2 & 3 of the XLR connectors and the tips and sleeves of all ¼ inch phone plugs and jacks MUST be identical throughout your entire system. Unfortunately, there is no world wide standard to follow. Many Asian manufactures use pin 3 of the XLR connectors as the hot or positive pole. But European manufactured equipment usually uses pin 2 as hot. US manufacturers are caught in the middle and produce both! What to do? Hit the phase reverse switch. Your BEQ-pro24 mixer will never be out of phase. In the switches up position, pin 2 is hot. In the down position, pin 2 is cold or negative while pin 3 is hot or positive. The phase reverse switch allows instant world-wide compatibility.

Now that you have correctly connected your inputs and correctly set the gain and phase, lets move to the EQ section.

Equaliser Section

It is the equaliser section of each channel that actually shapes the tonality of your sound. Whether you are reproducing the sound of a violin or a tractor, the realism of that sound is entirely dependant on how you have adjusted the EQ. Therefore, the EQ section can be your best friend or your worst enemy in a difficult audio situation. Be careful not to over-EQ your sound system. More tweeters and woofers are destroyed from over use of an EQ than all the other reasons combined. If you feel that you need more than 10-12dB of boost in you low or high EQ, you need to look elsewhere in your system for a problem. In a live sound system, this might include readjusting the output levels of your active crossover, output levels and equalisation of your signal source devices, adding subwoofers to your system, etc. Are your power amplifiers clipping? Maybe you’ve run out of power or your system just isn’t big enough for the room. Expand to a bigger system instead of damaging your equipment by over-EQ’ing it.
Fader

The BEQ-PRO 24’s channel faders should be operated –15 to +0 on the fader scale in most normal uses. If you are at the extremes of the faders travel, you are probably not setting the gain trim controls and/or the EQ controls properly. Remember, start with the trim controls as high as possible without audible distortion. It is OK to run your levels 2/3 or ¾ of the way up. You should have more output level than you need. There is a writing strip near each fader for labelling each channel’s input and sub output source. Write in wax pencil so you can change labels easily.

Some common sound engineer abbreviations:

- **VOX**: BACKUP VOCALIST
- **L VOX**: LEAD SINGER
- **BASS**: BASS GUITAR OR STRING BASS
- **L GUIT**: LEAD GUITAR
- **G-1)**: ADDITIONAL GUITARS ON STAGE LEFT TO RIGHT
- **G-2)**: ADDITIONAL GUITARS ON STAGE LEFT TO RIGHT
- **G-3)**: ADDITIONAL GUITARS ON STAGE LEFT TO RIGHT
- **KICK**: BASS DRUM
- **SNARE**: SNARE DRUM
- **HAT OR H/H**: HI-HAT
- **CYM**: OVERHEAD DRUM MICS
- **SAMP**: A DRUM SAMPLER
- **RACK 1 OR TOM 1**: SMALL MOUNTED TOM
- **RACK 2 OR TOM 2**: LARGE MOUNTED TOM
- **FLOOR**: FLOOR TOM
- **PERC**: BELLS, GONGS, WOODBLOCK, ETC
- **RIDE**: LARGE RIDE CYMBAL, TYPICALLY THE BELL PORTION
- **KEYS**: USUALLY A MIXED SIGNAL OF KEYBOARDS
- **B-3 TOP**: TREBLE HORN OF A ROTATING (LESLEI) SPEAKER FOR ORGAN, TYPICALLY A HAMMOND B-3
- **B-3 LOW**: THE WOOFER OF THE LESLEI SPEAKER FOR THE ORGAN
- **DI**: DIRECT INJECTION. A DIRECT SIGNAL FROM AN INSTRUMENT, BASS-DI, WOULD BE A DIRECT SIGNAL FROM BASS GUITAR, BYPASSING THE BASS GUITAR AMPLIFIER

Using these standard abbreviations, a guest artist’s engineer could easily mix on your system.

The Master section

You have established a mix throughout all of the input channels and you have sent that mix to subgroups and/or left and right masters. As with the channel signals the whole point of mixing signals is to be able to send them somewhere. Our complete mix is really a combination of several mixes that will either end up on a tape or in someone’s ears. Both destinations are away from the mixer. The master section has all the final controls for the various types of outputs on the back of the console.

Master Auxiliary Sends

Before we send the completed mixes out for recording or amplifying, we may wish to add some effects using the AUX sends. After these signals have been processed they are returned to the console via the AUX Returns to become a part of the main mix. You have already selected to amount of signal from each channel going to the individual AUX’s which have been mixed and are present at the Master AUX Send control. Each Master AUX has a
solo button so you can listen to the sum of these signals individually. You will not hear the processed signal because you are listening to the signal being sent to the effect. The signals on AUX’s if set to pre EQ and fader will most likely be for use on a live sound application for monitor sends, in this case you will not get the signal back for further processing. You can therefore use the convenient solo buttons to check these signals before they are sent out the back of the console. This is very useful for hunting down a feedback whistle in the stage monitors with headphones on.

Master AUX Returns

The Master AUX Returns are used for getting signal back into the stereo or mono mix that have previously been sent out for processing to an effect unit such as a digital reverb unit. Because you will not know how strong the return signal will be when they return to the mixer all the AUX returns have a level control. This is used to set the input level of the returned signal. With each AUX return there is also a pan control which is great for use with true stereo effects devices.

Panning stereo effects really opens up a mix in both live and studio applications. If you are using AUX’s on a pre EQ and fader mode for monitors, you don’t need to return the sent signal to the main mix. You can then use the returns for these AUX’s as true stereo returns by panning one hard left and the other hard right.

Output connection

Your mixed and processed signals are all available at the rear underside of the console and are waiting to be sent to their respective destinations. The outputs of your BEQ mixer are all active at the same time. Don’t be afraid to use any or all of them. If you need to send signals to more than just a few destinations, you have plenty of options. The numerous outputs on the BEQ Pro24 mixer proves this point, Oram’s BEQ mixers are extremely flexible.

Power requirements

Your BEQ mixing console is capable of 110-120 VAC or 220-240 VAC operation allowing worldwide usage. It is prewired at the Oram factory for the correct voltage in your country and is furnished with the appropriate power cord and line fuse. Should the voltage setting need to be changed, it can be accomplished by changing the voltage selector found on the rear of the PSU.

Care and maintenance

Your Oram BEQ mixing console is built to provide years of dependable service under demanding circumstances. It requires no internal maintenance but a common sense approach to its use will help you enjoy long and reliable operation. Here are some tips:

Periodic cleaning

Keep the unit clean by wiping frequently with a damp cloth, soft cloth. Use a mild detergent cleaner if necessary, applied to the cloth, but not directly to the mixer. Do not use solvents or other chemicals to clean the unit. A large paintbrush is useful to remove accumulated dust from between the many control knobs on the mixer. If liquid is accidentally spilled onto or into the unit, disconnect the power cord and allow the unit to dry thoroughly before attempting to use it.

Connecting cables

Use only high quality connecting cables with your Oram BEQ mixer. Faulty or suspicious cables should be replaced to avoid possible deterioration of your sound quality.
**TECHNICAL INFORMATION**

**BRITISH EQUALISATION SERIES PRO24 CONSOLE**

**Input Impedance**
- Microphone: >1.2 kΩ electronically balanced
- Line & Tape: >10 kΩ electronically balanced

**Output Impedance**
- All outputs: <100 Ω electronically balanced

**Gain**
- Microphone: +75dB (input to group output)
- Line: +25dB (input to group output)
- Tape: +25dB (input to group output)

**Noise**
- Microphone: <-128 dBu e.i.n. ref 200Ω, 20Hz to 22kHz
- Line: <-89 dBu (Eq in, direct output) wideband

**Maximum Levels**
- Microphone input: +24 dBu wideband
- Line input: +24 dBu wideband
- All outputs: +28 dBu balanced, +22dBu unbalanced

**Distortion**
- Microphone: <0.008% THD (-50 dBu input, +14 dBu output)
- Line: <0.008% THD ( +14 dBu input, +24 dBu output)

**Frequency Response**
- Microphone: 20Hz to 20kHz within 1dB, -3dB @ 45kHz
- Line: 20Hz to 20kHz within 1dB, -3dB @ 45kHz

**Crosstalk**
- Adjacent channel: -85dB @ 1kHz, -80dB @ 15kHz
- Adjacent Group: -85dB @ 1kHz, -80dB @ 15kHz

**Nominal Level**
- 0 VU: +4dBu
**SONICOMP COMPRESSOR**

Output gain control ranging from 0 to +15dB of gain

The release control ranging from 0.05 seconds to 3 seconds

The threshold goes from -25dBu to +20dBu then off

The ratio limio is adjustable from 1.4:1 to 30:1 (full limit)

The attack gives a range from 0.1mS to 40mS

The input level will give + and - 15dB of of control

**EQ ROUTING**

Individual routing of the EQ to the subgroups, left/right 1, left/right 2 and external unit via the patchbay

**HI-DEF EQ**

6 bands of EQ including a high and low shelf and 4 swept mid bands

High and low filters ranging from 5Hz to 80kHz

All 6 bands of the EQ have + and - 18dB of cut and boost

Individual bypass switch for the filters and EQ

Input gain control giving + and - 15dB level

**COMPRESSOR ROUTING**

Individual routing of the compressor to the subgroups, left/right 1, left/right 2 and external unit via the patchbay

The meters read the signal gain reduction for each side

The link switch enables the release, threshold, ratio and attack of the right to be controlled by the left

The optical switch allows the compressor to work in solid state or LDR mode

There is an individual compressor bypass for left and right

This side controls left1, left2, subgroup 1,3,5,7 and patch left

This side controls right1, right2, subgroup 2,4,6,8 and patch right
MASTER AUX SEND
MASTER AUX LEVEL CONTROL
WITH SOFT MUTE AND SOLO
(IN DESTRUCT MODE AUX SEND IS NOT MUTED)

STEREO MASTER AUX RETURN
INPUT TRIM CONTROL WITH + AND - 15dB OF GAIN
INDIVIDUAL ROUTING TO CENTER, LEFT/RIGHT 1 AND LEFT/RIGHT 2
ROUTING TO SUBGROUPS
HIGH AND LOW SHELVING EQ WITH + AND - 15 dB OF GAIN
MONO FUNCTION PICKS UP THE SIGNAL ON THE LEFT AND FEED IT TO BOTH LEFT AND RIGHT
THE BALANCE CONTROL ENABLES THE SIGNAL TO BE ADJUSTED BETWEEN LEFT AND RIGHT
SOFT MUTE AND SOLO (IN DESTRUCT MODE AUX RETURN IS NOT MUTED)
60mm STEREO FADER

SUBGROUP
ROUTABLE TO LEFT/RIGHT 1 AND LEFT/RIGHT 2
SUB PAN TO POSITION THE SIGNAL IN THE LEFT/RIGHT MIX
SOFT SOLO AND MUTE (SUB IS MUTED WHEN IN DESTRUCT MODE)
THE MASTER SUBGROUP OUTPUT IS CONTROLLED BY THESE 100mm FADERS.

LEFT/RIGHT 1 AND LEFT/RIGHT 2 ARE CONTROLLED BY 2 STEREO 100mm FADERS. THE CENTER IS CONTROLLED WITH A MONO 100mm FADER.

THERE ARE INDIVIDUAL MUTES FOR LEFT/RIGHT 1, LEFT/RIGHT 2, AND CENTER.

C TO L/R1 WILL MIX THE SIGNAL GOING TO THE CENTER WITH THE LEFT/RIGHT 1 SIGNAL.

L/R2 TO L/R1 WILL MIX THE SIGNAL GOING TO L/R2 WITH LEFT/RIGHT 1.

TALK WILL ENABLE THE TALKBACK.
**Talkback & Oscillator**

3 pin XLR for talkback mic (PIN1=GND, PIN2=POS, PIN3=NEG)

Routing buttons allow talkback and oscillator to be sent to 5.1, subs, aux7/8 and an external output.

The oscillator has three set frequencies of 50Hz, 1kHz, and 15kHz and an on/off switch.

The level control allows the talkback and oscillator level to be adjusted.

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**Playback Returns**

When this button is pressed any channel that has the I/P flip switch pressed will flip the mic/line input with the tape input.

**Solo**

Solo level controls the amount of solo signal sent to the left/right buss. Solo kill will remove any soloed switch.

The mode switch will select either destruct or non-destruct.

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**Sub-Band**

The sub-band is made up of left/right 1, left/right 2, and center. The amount of each can be individually controlled.

There is a master sub level to control the final sub output.

The mono switch will combine the 5.1 signal (being sent out or returned) and give a stereo mix of it.

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**Speaker Selection**

The output can be sent to various speakers each individually selectable.

The aux output and the headphones have their own level control.

All outputs go via the master control room level over 40dB of attenuation.