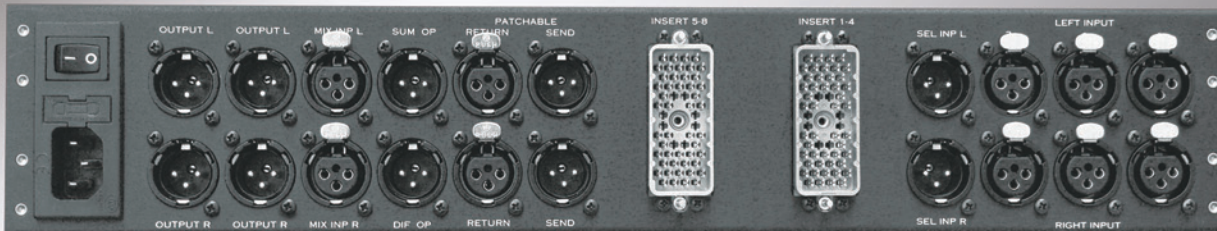




# THE MANLEY MASTERING BACKBONE

## OWNER'S MANUAL



13880 Magnolia Ave.  
Chino, California 91710  
www.manley.com

emanley@manleylabs.com  
Tel: +1 (909) 627-4256  
Fax: +1 (909) 628-2482

**MANLEY**  
LABORATORIES, INC.

# CONTENTS

INTRODUCTION.....	3
MAINS CONNECTIONS.....	4
THE FRONT PANEL.....	5
SIGNAL FLOW.....	6-9
Input Selection.....	6
Input/Output Level Trims.....	6
Inserts 1-3 (SUM-DIFFERENCE processing).....	7
Inserts 4, 5 (SWAP).....	8
Inserts 6-8.....	8
Mix/Fade Knob.....	8
Output Level Switches.....	9
THE BACK PANEL.....	10, 11
THE GUTZ.....	12
SOME THOUGHTS FROM OUR LAB.....	13-16
LABELING YOUR SWITCHES.....	16
TROUBLESHOOTING.....	17
PATCHABLE SENDS/RETURNS.....	18
ELCO PINOUT.....	19
DB-25 PINOUT.....	20
Y-CABLE DIAGRAM.....	21
BLOCK DIAGRAM.....	22
PARALLEL PROCESSING DIAGRAMS.....	23
DC-COUPLING I/O MODIFICATION.....	24
SPECIFICATIONS.....	25
TEMPLATE FOR STORING SETTINGS.....	26

## INTRODUCTION

### THANK YOU!...

...for purchasing the Manley Laboratories Mastering BACKBONE. This product is the culmination of many custom Manley mastering consoles and mastering processors we have built over the years. When we started building custom consoles in 1990 the only option for mastering houses was to commission purpose-built systems from rare engineering shops like ours. We began to attract top mastering engineers who would approach us and request audiophile grade consoles and dedicated mastering versions of our compressors and EQs. We were happy to listen and oblige - and frankly, somebody had to support that neglected side of the business.

Over the next dozen years, mastering businesses flourished and branched, and many more audio companies developed products aimed specifically for mastering. In fact, a whole new sub-category of mastering grew out of this, most often called “project mastering”. We were getting deluged with requests for custom projects at a point in time when it was difficult to keep up with orders for standard products. We had to limit the number of custom projects we could accept, and even those were beginning to take too long to build and test. Eventually we stopped accepting custom work altogether. Meanwhile, several smaller companies were beginning to market non-custom off-the-shelf mastering consoles (some remarkably similar to ones we had built).

However, even with several companies building reasonable mastering consoles, we were still getting requests for something similar; not custom, but “Manley, like the famous guys use.” The Manley BACKBONE is one part of our solution. In fact, it represents an evolution in performance from where we left off in our custom designs.

The Manley BACKBONE is built with a unique electronic topology that maintains a fully balanced signal path through 95% of the console. All other designs require the balanced signals to be converted to unbalanced before sections that, for example, allow you to boost or cut level, then inversely need to be converted back to balanced to feed the outside world. In other words, this new topology not only does it in one stage rather than three or four stages, but also does it as cleanly and quietly as any single stage could that we know of, including fully discrete approaches. This topology also provides better common mode rejection and power supply noise rejection than conventional approaches. This is significant especially in the “Sum-Difference” functions.

Each of the 10 gain blocks and all of the switches are on connectors, which means if the unit ever requires service, repair can likely be done quickly without the need for a technician or soldering iron – just a medium-small Philips screwdriver. No expense was spared in the quality of the parts, so one should expect many years of continuous professional use.

Please note that this is not a “Monitor Control Box.” The Manley BACKBONE is simply a device that routes various analog processors with maximum fidelity and allows you to maintain the shortest possible signal paths while providing some flexible central control functions. This unit is not supposed to color your sound in the least (excepting the SUM-DIF feature, of course); it just exists to let you use your favorite outboard gear in more creative ways.

# MAINS CONNECTIONS

Your BACKBONE has been factory set to the correct mains voltage for your country. The voltage setting is marked on the serial badge, located on the rear panel. Check that this complies with your local supply.

Export units for certain markets have a moulded mains plug fitted to comply with local requirements. If your unit does not have a plug fitted the coloured wires should be connected to the appropriate plug terminals in accordance with the following code:

GREEN/YELLOW	EARTH
BLUE	NEUTRAL
BROWN	LIVE

As the colours of the wires in the mains lead may not correspond with the coloured marking identifying the terminals in your plug proceed as follows:

The wire which is coloured GREEN/YELLOW must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol or coloured GREEN or GREEN and YELLOW.

The wire which is coloured BLUE must be connected to the terminal in the plug which is marked by the letter N or coloured BLACK.

The wire which is coloured BROWN must be connected to the terminal in the plug which is marked by the letter L or coloured RED.

**DO NOT CONNECT/SWITCH ON THE MAINS SUPPLY UNTIL ALL OTHER CONNECTIONS HAVE BEEN MADE.**

**Note: This unit has been factory wired for your country. If you plan to take the unit to countries with a different mains voltage you will need to send the BACKBONE to a qualified service technician or to the Manley Labs Service Center for the correct transformer primaries wiring conversion. Alternatively, you may use outboard AC voltage step-up or step-down transformers as appropriate.**

**TRANSFORMER WIRED IN PARALLEL for 100V and 120V operation. Uses a 1A SLO-BLO fuse.**

**TRANSFORMER WIRED IN SERIES for 220-240V operation. Uses a 0.5A SLO-BLO fuse.**

**\*\*\*Fuses for ALL VOLTAGES are 5x20mm GDC packaged fuses.\*\*\***

## Waste Electrical and Electronic Equipment (WEEE)

### Information for customers:

The European Parliament and the Council of the European Union have issued the Waste Electrical and Electronic Equipment Directive. The purpose of the Directive is the prevention of waste of electrical and electronic equipment, and to promote the reuse and recycling and other forms of recovery of such waste. As such the Directive concerns producers, distributors and consumers.

The WEEE directive requires that both manufacturers and end-consumers dispose of electrical and electronic equipment and parts in an environmentally safe manner, and that equipment and waste are reused or recovered for their materials or energy. Electrical and electronic equipment and parts must not be disposed of with normal household wastage; all electrical and electronic equipment and parts must be collected and disposed of separately.

Products and equipment which must be collected for reuse, recycling and other forms of recovery are marked with the following pictogram:



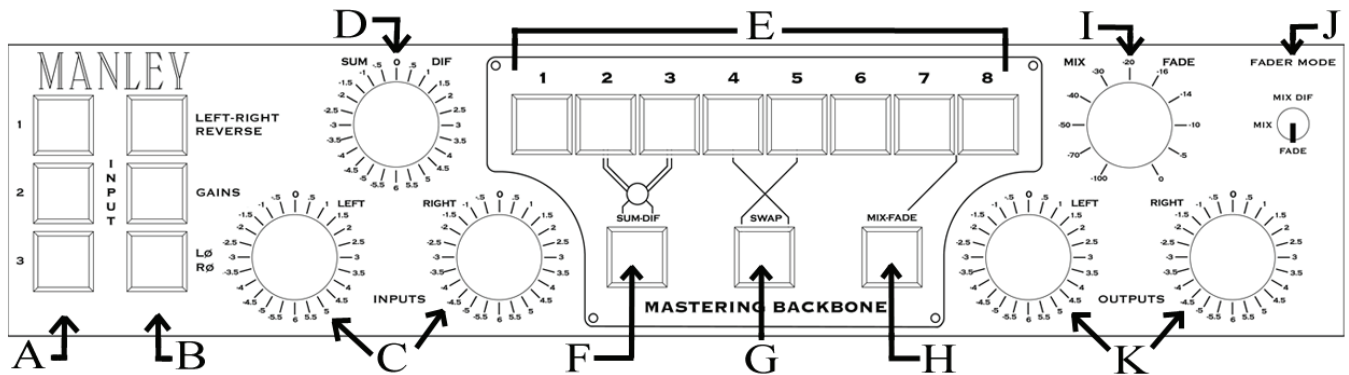
Small products may not always be marked with this pictogram in which case this is present in the instructions for use, on the guarantee certificate and printed on the packaging.

When disposing of electrical and electronic equipment by use of the collection systems available in your country, you protect the environment, human health and contribute to the prudent and rational use of natural resources. Collecting electrical and electronic equipment and waste prevents the potential contamination of nature with the hazardous substances which may be present in electrical and electronic products and equipment.

Your MANLEY or LANGEVIN retailer will assist with and advise you of the correct way of disposal in your country.



## THE FRONT PANEL



- A. INPUT 1, 2, 3:** Choose which of the three inputs will drive the BACKBONE audio chain. Whichever source is selected here will also come out of the “SELECTED INPUT” XLRs on the back panel.
- B. LEFT-RIGHT REVERSE:** Allows you to swap left and right channels.  
**GAINS:** When lit, engages all active electronics for the input and output gains.  
**LØ RØ (polarity reverse):** Changes the absolute polarity of the signal on both channels. For more on this, see page 6.
- C. INPUTS:** Input signal level for each channel can be selected on these 24-position, 1/2 dB stepped controls. Range is from +6 to -5.5 dB.
- D. SUM-DIF Rotary Switch:** Alters the level of the DIFFERENCE signal sent to Inserts 2 and 3. Activated only when the SUM-DIF button (F in this diagram) is engaged. For more info, see pages 7 & 13.
- E. Inserts 1-8:** Push to activate the outboard processor that is patched into that insert point. See page 7-8 for details.
- F. SUM-DIF:** Press to engage SUM-DIF circuit for processing of Inserts 2 & 3.
- G. SWAP:** Swaps the order of Inserts 4 & 5 so that Insert 5 comes before Insert 4 when engaged (illuminated). See pages 8 and 13 for info and suggestions.
- H. MIX-FADE Button:** Activates the MIX-FADE knob (I in this diagram).
- I. MIX-FADE Knob:** Acts as a master fader or mixer, depending on the position of the FADER MODE SWITCH (J in this diagram). For a detailed explanation, see pages 8 and 15.
- J. FADER MODE Switch:** This is a 3-position switch (I in this diagram) that determines the operating mode of the MIX-FADE knob. The three modes are MIX DIF, MIX, and FADE. See pages 8 and 15 for more info.
- K. OUTPUTS:** Output signal level for each channel can be selected on these 24-position, 1/2 dB stepped controls. Range is from +6 to -5.5 dB.

# SIGNAL FLOW

## INPUT SELECTION

The signal-flow through the BACKBONE is simple enough to describe and mostly follows the front panel left-to-right layout. Three pairs of XLRs on the back panel are balanced line inputs associated with the client's sources. Typical sources might be a superb DAC (or two) or an analog tape machine.

Next come a pair of XLRs that are direct outputs of the chosen input. Typically these outputs would feed a pair of monitor control box inputs and are used for comparing the raw tracks that were brought in versus the processed signal. Mastering engineers may use this as a "confidence check" and as an immediate way to show the client the "before and after pictures". There are other outputs that can be used to check the final output after processors, and for a mid-point in the processor chain if desired.

The chosen input's Left and Right channels can be swapped just for those occasional sessions when either the mixing studio accidentally swapped cables or the mix seems to sound better in the opposite perspective. Absolute polarity can also be reversed. This reverses the "phase" (polarity is the correct term) of **both** the left and right channels. Usually one wants the kick drum pushing the speaker out rather than sucking the speaker inward. Most of the time, "Polarity Reverse" is a subtle effect and with some mixes the effect is inaudible. Strangely, the effect is often most pronounced on reed instruments and some types of vocal character. It is usually worth a quick check to verify which way seems to sound best and/or seems to have the kick pushing the woofer outward. If in doubt, leave the absolute polarity alone.

The Input Selection, Left-Right Reverse and Polarity Reverse are accomplished purely passively, and are highly unlikely to damage the signals at all. This is achieved with simple sealed gold-contact relay closures. In fact, the Left channel and Right channel signal paths are mainly kept on separate circuit boards and/or widely separated physically to maintain maximum channel separation and low cross-talk. Where a signal meets a connector, two gold contact pins and two wires are used for extreme reliability. That feature is called "redundancy" so that if one expects one failure in one thousand with only one connector, one might expect one failure in a million with doubled contacts (1,000 X 1,000).

## INPUT LEVEL SWITCHES

Next up are the two INPUT LEVEL rotary switches that provide 24 precise ½ dB gain steps. The range is from 6 dB of boost to -5.5 dB of attenuation. We select ultra-low distortion thin film 0.1% tolerance resistors and sealed gold contact Greyhill switches for near perfect tolerance and matching. As described earlier, the gain stage is a fully balanced circuit followed by a pair of unity gain high current buffers. "Nearly inaudible" is the goal here.

There is a GAINS switch that engages active electronics for both the INPUT LEVEL and OUTPUT LEVEL control circuits. This is one more form of confidence check that allows you to compare whether two sets of gain stages have any audible footprint, whether detrimental or beneficial. It also allows one to run the console in a 100% passive mode.

Generally the INPUT and OUTPUT LEVELS are meant to correct for any losses or unintentional gains from the source machines or in any of the outboard processing gear. However, these controls really open a lot of possibilities in terms of driving gear hotter or softer especially when one considers combining that concept with setting up outboard gain settings away from a standard unity gain setting. The issue here is “the sweet spot” or “gain staging” and the fortunate or unfortunate aspect of many devices that just seem to sound best when driven a little harder or softer than the designer intended. One can think of it a bit like a guitar amp with its input pot and master volume to get the right amount of distortion at a comfortable stage volume. Or, one might be more familiar with analog mixing consoles and liken it to finding that magic combination of individual fader levels and mix-bus fader level.

*\*\*\*In some cases, for “absolute purist fanatics”, even these buffers can be bypassed/removed (by removing the chips and inserting small jumper wires between pins 3 and 8 on the dip socket) for one less gain stage in the path – BUT in general they serve a number of beneficial purposes (including isolating the cable and destination electronics from the gain stage’s feedback loop) whereas buffer-ectomy may have more of an imaginary benefit . If you can’t hear any change of quality bypassing ALL the electronics in the console (and so far that is the unanimous opinion), then don’t even think of possible mods and variations (especially a mod that might be a detriment in practice). Keep in mind, it also isn’t the kind of change that can be truly A-B compared...\*\*\**

### **INSERTS 1-3...SUM-DIFFERENCE PROCESSING**

OK, back to the idiosyncrasies of this Insert Box. Next in the signal path is INSERT #1. Pushing the button marked INSERT 1 engages whatever box might be connected there. When the button is NOT pushed and lit the main audio path is hard-wire bypassed, of course. However, a little isolation amplifier is engaged to feed that box. This is to pre-drive a compressor so that it is already doing appropriate gain reduction at the moment that you do insert it. This prevents momentary level changes.

INSERT 2 and INSERT 3 are special. Below these button is another button labeled SUM-DIF which engages a complete set of Sum / Difference encoders and decoders (or is it the other way around?). Some people refer to this as M-S (or Mid-Side) but there are also those who insist that these terms are reserved for mic techniques.

These circuits allow Insert 2 and 3 to act on both the mono (sum) and the side (difference) parts of the original stereo signal. The left side becomes sum and the right side becomes the difference. So if you just want to narrow the width of the extreme lows to help cut a lacquer, just cut lows on what was the right channel of your outboard EQ and is now the “Difference Channel”. After passing through Insert 2 and 3 the signals are decoded back to left and right. If the SUM-DIF button is not in, Insert 2 & 3 just operate as the other inserts do, as normal stereo left and right.

There is also a 24 position switch that adjusts the level of the difference signal over a -5.5 dB (more mono) to +6.0 dB (wider) range. This switch is located BEFORE Insert 2 in the signal path, so it will affect the signal sent to Inserts 2 and 3 if SUM-DIF mode is engaged. Alternatively, one can mix the difference signal either treated or untreated into the stereo mix near the end of the chain (more on that later). In other words, you should have more than enough options to mangle the stereo image especially if you also have yet more sum-diff tools in your digital arsenal. Using it as a cheap trick to impress a client may have less desirable consequences later. Just beware of mono compatibility and FM radio transmission issues, difficulties in cutting vinyl records, diminished lead vocals, bass and drums in the mix (which might be more important to record sales) and reverb balances.

## **INSERTS 4 & 5...SWAP!**

INSERT 4 and INSERT 5 are also special. You can reverse the order of these two using the button below marked SWAP. With the button pushed, Insert 5 precedes Insert 4. Whatever processors you choose for these inserts, verify that with both units NOT bypassed and “flat” (e.g. zero compression), that there are no unwanted effects related to the patch order, including other processors potentially in the chain (especially Inserts 3 and 6 which are nearest 4 & 5).

It might be interesting to put compressor/limiters on these inserts. Depending on settings and the order, there can be benefits in how each responds dynamically, especially if one unit tends to overshoot a bit. Of course, the SWAP button will be appreciated by those who are tormented with the decision of whether EQ should precede compression or vice versa. Usually, “it just depends.” With the gentle processing typical in mastering, there isn’t much difference.

## **INSERTS 6-8**

Next in the signal flow are INSERT 6 and INSERT 7, which are not special (like their neighbors are, at least). Some engineers prefer these inserts for their main processors and some for their alternative or “occasional” processors. We know that was an ambiguous sentence, but what we mean is, use these inserts for whatever you’d like to. We don’t mind either way.

INSERT 8 is special if for no other reason than being the last process in the chain. This, of course, is likely the most appropriate place for your final limiter. We might suggest that this processor insert be used to sometimes push the level, after keeping the signal at more reasonable and conventional levels earlier in the chain.

Between INSERT 7 and INSERT 8 is a multi-purpose function. Pushing the button labeled MIX FADE below them engages the large Penny and Giles rotary knob to the right of INSERT 8. This is especially handy for “Producer Fades”. Most of the time, fades are simply done graphically in the workstation, but there are occasions when one wants a manual fade and big knob to ride the volume down (or up). This rotary knob is called the...

## **MIX/FADE KNOB**

There is a small toggle switch associated with this circuit that sets up alternative functions. With the switch down, the FADER is a fader. Simple. With the switch in the middle, the knob becomes a precision stereo mixer. Lastly, in the “up” position, the mixer uses the “DIF” signal (unprocessed by Inserts 2 and 3) from that earlier part of the console so one can blend in a little extra width.

The basic MIX function is typically used for several purposes. One is the so-called “parallel compression” which generally employs a limiter set for somewhat drastic settings. When it is blended into the main path, it has the effect of raising lower level signals without squashing transients, which sort of translates to less quiet dynamics and increased ambiance. It can be considered a technique for manipulating quiet sections rather as opposed to normal limiting that affects the louder passages.

One could also try lightly mixing in extreme EQ or compression settings on the SUM or DIF parts of the signal, via the back panel XLRs. If you’re interested, more details on MIX/FADE techniques follow later on in the manual...



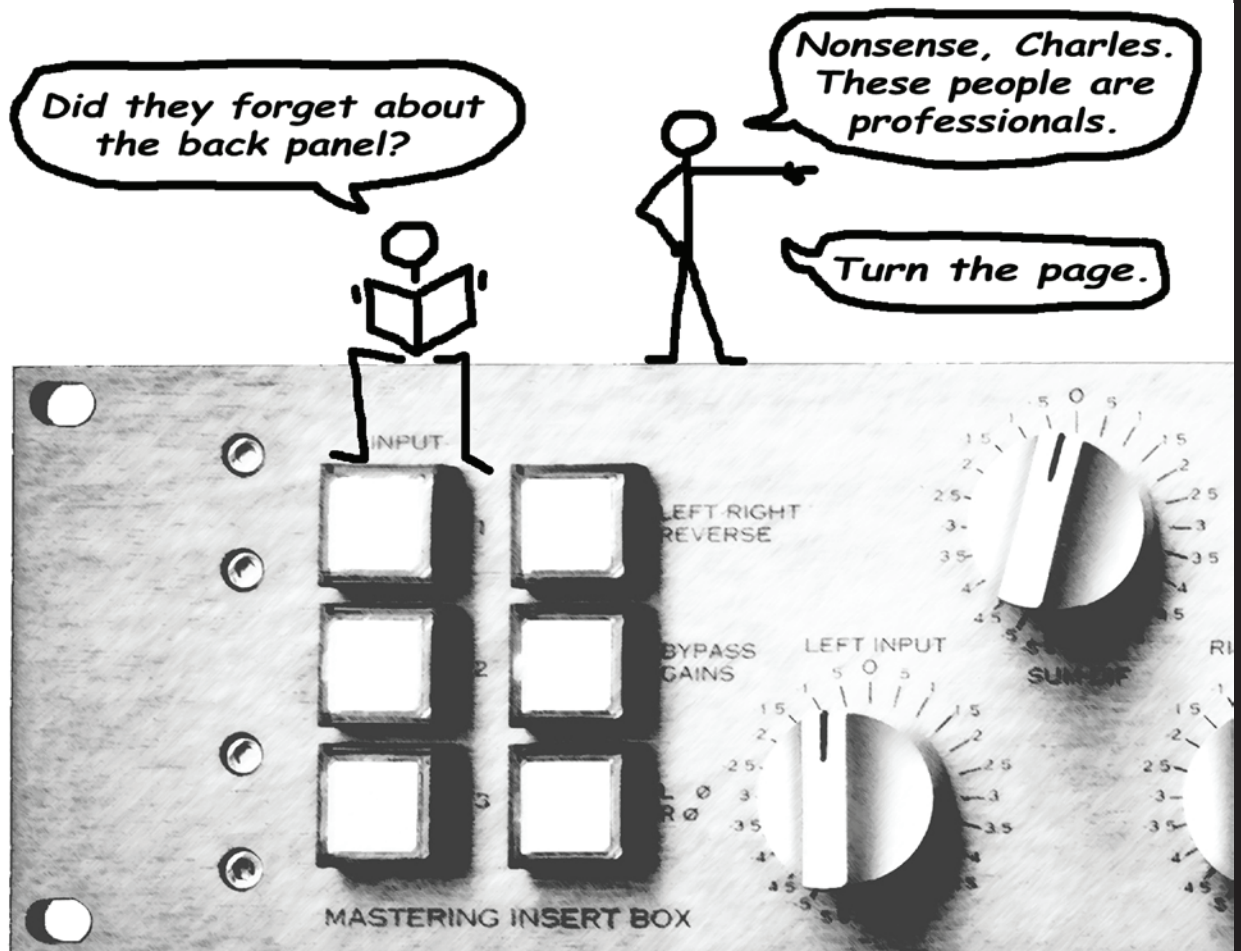
## OUTPUT LEVEL SWITCHES

Right after the FADER/MIX stage is the final active stage, which are the OUTPUT LEVEL switches. Like the INPUT LEVEL switches, you have 24 ½ dB steps filling a -5.5 dB to +6dB range. And this stage together with the INPUT LEVEL controls can be bypassed/engaged with the button marked GAINS.

Now, it's time for an...

### *\*IMPORTANT PARAGRAPH\**

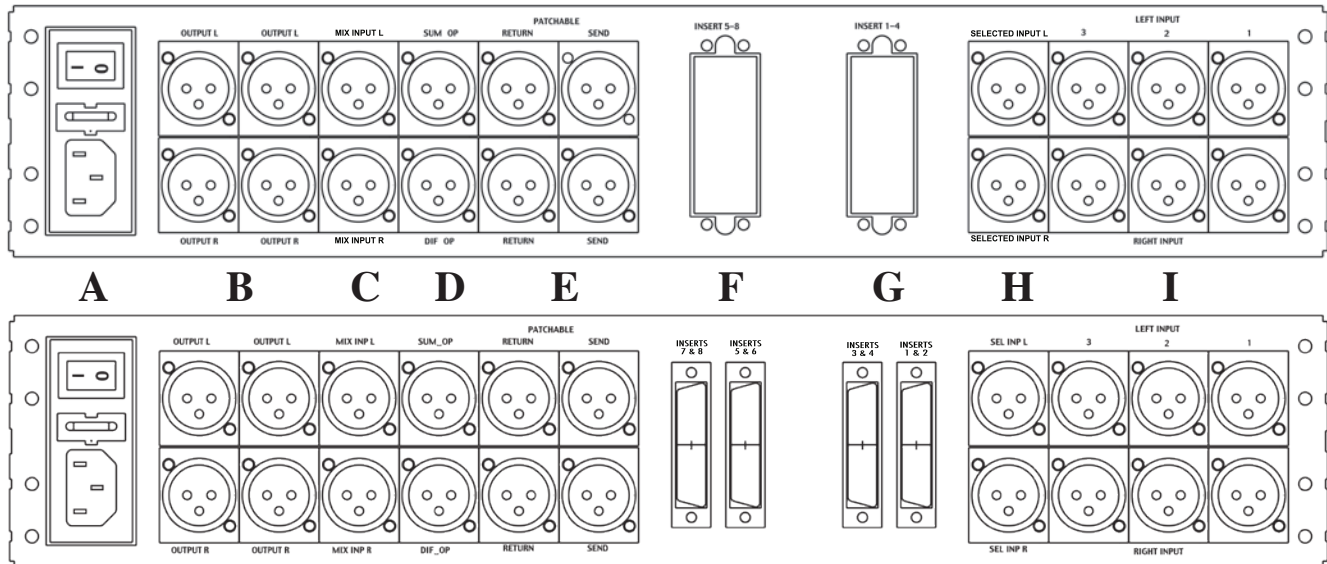
Note that both the FADER/MIX and OUTPUT LEVEL controls precede INSERT 8. This way you can patch your final limiter in 8 and semi-permanently set thresholds and gains, and use the OUTPUT LEVEL controls as a very handy way to set how hard the signal is feeding the limiter and thusly how many dB it is typically limiting. Sometimes this even negates the need for a special “mastering version” limiter, because the most critical controls, threshold and output level, stay “calibrated” while the Mastering Switcher OUTPUT LEVEL controls become something like a “drive” or “input level” control.



# THE BACK PANEL

Top drawing: ELCO configuration

Bottom drawing: DB-25 configuration



- A. *Power Switch*: Where the IEC power cable is inserted, the fuse is placed, and the power is switched off and on. Most users should be familiar with this feature set.
- B. *Output L/R (x2)*: Two sets of balanced outputs, typically going to your favorite A/D converter and a monitor controller/VU meters.
- C. *Mix Input L/R*: These inputs feed the FADER in MIX mode. See page 15 for more details.
- D. *SUM-DIF Outputs*: The outputs for the raw encoded signals (unprocessed by Inserts 2 & 3). SUM signal is attenuated 6dB, DIF is boosted 6 db to bring them to “normal” levels that are easier for most outboard gear to process.
- E. *PATCHABLE Sends>Returns*: Factory-wired to patch into Insert 6, but internally user-adjustable. See page 18 for details.
- F. *Inserts 5-8 connectors\**: ELCO/DB-25 connectors for the last 4 inserts.
- G. *Inserts 1-4 connectors\**: ELCO/DB-25 connectors for the first 4 inserts.
- H. *Selected Input L/R*: The selected input’s dry signal comes out here.
- I. *Inputs*: 3 sets of inputs for whatever you want to plug into the BACKBONE. We recommend audio sources; you can try plugging in your toaster, but it probably won’t sound very good.

\*In the ELCO configuration, two 56-pin ELCO connectors handle all of the inserts. In the DB-25 configuration, four 25-pin connectors do the same job. For the ELCO pinout, see page 19; for the DB-25 pinout, see page 20.

## **IF DRIVING ANY UNBALANCED LOADS, READ BELOW:**

*If you are planning on driving any unbalanced equipment with your BACKBONE, you must make sure to **FLOAT PIN 3**. If Pin 3 is grounded, there is a possibility of damaging the BACKBONE, or at least introducing unwanted noise into the signal path. Lifting Pin 3 will allow you to drive unbalanced equipment, but please note that this scenario will lead to a 6 dB drop in level. This applies to ALL outputs, including those on the ELCO/DB25 snakes.*

## BACK PANEL

As previously described, you have 3 Input XLRs per side followed by an Output XLR that reflects the chosen input. These are factory set-up as AC coupled (the inputs are protected by capacitors, which may prevent some clicks or pops when switching), which implies the signal has to flow through capacitors. If these capacitors are unwanted, there are jumpers (no soldering required) to bypass them and provide DC coupled inputs (similar jumpers exist for the outputs). The diagram describing the jumper locations is on page 23.

Next are Multi-Pin connectors that handle all of the INSERT inputs and outputs. The BACKBONE uses EDAC/ELCO 56-pin connectors. There is also an option for DB-25 connectors, if one so desires. Both of these systems are very reliable and allow for pretty bulky audiophile wires that many mastering engineers prefer. Are these wired to the “industry standard”? Close, but not exactly. The big difference between the industry standard and what the BACKBONE uses has to do with the XLR “sex”. The standard may have either all male or all female XLRs, while what you need here is half male, half female. So, worst case scenario, buy one of each ‘standard’ cable, and spend an hour or two swapping half of the XLRs. Better case scenario, find pre-made ones that fit the bill (make sure to thoroughly investigate first!). Best case scenario, have a good tech build some audiophile wire (maybe silver conductor) cut-to-length, tailor-made ELCO or DB-25 cables. (*ELCO pinout is on p. 19, DB-25 pinout is on p. 20*)

If you wish to purchase custom cabling for your setup, we highly recommend Rapco Horizon. If you go to their website ([www.rapcohorizon.com](http://www.rapcohorizon.com)), there is more information on their cables, line testers, and various other products. They build their cables VERY well, they have excellent customer service, and we’ve found their prices to be quite reasonable.

Next comes an interesting pair of XLRs marked “PATCHABLE”. There are a number of points internally in the console where these XLRs can be patched. Of course, this means opening up the BACKBONE, opening up the manual (which is probably open already if you’re reading this), attempting to interpret the map on page 18, and moving some ribbon cables, but dealing with ribbon cables seems to be par for the course if you’ve had a PC in your life. The three intended purposes for these are:

- An alternative to the multi-pins for one pair of inserts. You might use those for “producer-processors” or new gear you are evaluating. In that case, you probably don’t want to have anything plugged into the corresponding XLRs off the multipins. For example, the Manley factory presets that “PATCHABLE” to INSERT 6, so use either the multi-pin Insert 6 XLRs or these. Actually, just watch out for the female XLRs or inputs because you don’t want the outputs from two different boxes feeding into each other. The alternative set of male XLRs can be considered a parallel output, so it can feed something. Which brings us to...
- A feed to the monitor controller for checking the processing half-way into the chain. Typically a mastering engineer may want to compare the dry signal (“Selected Input” XLR), the analog processed signal (Main Output 2), and maybe somewhere part way through the processing, which these jacks allow.
- A feed for some of the unusual processing tricks described in the MIX/FADE section on page 15.

Next on the back panel are outputs for SUM and DIF. These are the raw encoded signals (unprocessed by Insert 2 & 3). These XLRs could be used to send to processors, in which case probably either but not both would be used. You can also send these signals to VU meters. *\*\*Note that the SUM signal is attenuated 6 dB and the DIF signal is boosted 6 dB, so that with typical stereo music mixes these outputs have similar levels to normal left and right signals (easier for outboard gear to process).\*\**

After that, you get MIX INPUTS. These feed the FADER in MIX mode. As described on page 15, if you use these as a return from processing the DIFF signal (but not the SUM), you should get a special Y cable where the right XLR is wired out-of-phase. There is a diagram of this cable on page 21.

Last is a pair of stereo outputs. One pair generally goes to your main Analog to Digital Converter and the other goes to the monitor controller or VU meters.

## THE GUTZ

### GUTZ

There is a 4-deck sandwich arrangement of printed circuit boards that make up the active electronics that process audio. Two long boards are so-called motherboards that hold 5 small PCBs each. There are separate motherboards for left and right, and the left is the top board, while the bottom takes care of the right. Each mother has about 36 relays and a dozen connectors, plus about 4 ICs that feed a fake signal to bypassed inserts. There are a number of connectors on the bottom the motherboards to join with each other.

The 5 small daughter boards on each mother can be considered gain blocks and there are 10 total. Each is labeled to its calibrated location but there is some minor inter-changeability for troubleshooting. For example, on each mother, #1 and #5 (INPUT TRIM, OUTPUT TRIM) can be swapped. #2 and #3 can be swapped as well, but in the case of such a swap *you must move the red jumpers off #3 (now in #2's place) and put them on #2 (now in #3's place)*. However, you cannot swap daughter boards from the bottom to the top mothers because of connector inversions. Because of the precision calibrations, especially in the SUM-DIF circuits, we advise that after a temporary swap the boards should go back to their original locations.

If one board or switch develops a problem and needs to be replaced, Manley has a support program and boards in stock intended for swapping in the field (rather than sending the entire BACKBONE back to the factory and losing time).

Similarly, front panel boards associated with EAO buttons (which are also on sockets) and Greyhill switches, back panel boards associated with XLRs and multi-pin connectors and power supply boards are all on connectors and can be easily field serviced. The Mastering switcher should be operable for decades.

Each daughter board has several trimmers and some of them should be tweaked on a routine basis (like once a year) for best performance. These would be trims associated with gains and DC offset. About half the trims may rarely need tweaking or perhaps can be left at factory settings and these are only associated with common mode rejection (hum & buzz immunity). Because the trimming procedure for the BACKBONE requires special equipment generally not in the hands of end users, we strongly suggest leaving this process to the Manley Labs service center, or to a qualified technician.

## SOME THOUGHTS FROM OUR LAB

*\*\*NOTE\*\* We recommend leaving one rack space open above and below your BACKBONE for cooling purposes (as opposed to cooling porpoises, which we don't know anything about). It's always good to keep some air flowing around your gear. This helps avoid damage to delicate components, as well as to your delicate hands.*

### GEAR SEQUENCE

One interesting quality of much analog equipment is the optimum sequence of gear. While there is some validity to ideas of “EQ before compression” or “EQ after compression”, the practical reality is trickier. For example, Box #1 may seem to sound best only if it drives into another box that does not have a transformer input – it likes to see a relatively simple load and maybe really shines driving into just one particular box for unknown reasons. Box #2 may be quite the opposite, and it just loves to drive a transformer input. So the optimum order may just take a day or two of experimentation and listening, rather than thinking about the function of the box. This is the biggest single reason for not including the option which allows one to arbitrarily change the order of processors – there is almost always an optimum order sound-wise.

While we are on that topic, there are some other interesting and not very obvious potential sequences of processors. For example, engineers might only consider a brick-wall limiter as appropriate as a final or next-to-final process. And while that is generally a good place for one, you might find that a limiter early in the chain helps compressors and de-essers further down from over-reacting and pumping, and might prevent EQs from clipping. We are not suggesting trying it for the sake of an idea, but suggesting a listening experiment because, in the end, it is more about the particular collection of gear that you have chosen and the collection of input-output idiosyncrasies that few are aware of. This hints at why some people hate gear that others love. The “sound” of a processor has to do with what device is patched to what, and settings of input and output levels. One might hear serious raves about some piece and expect something dramatic, when it was actually the subtlety of the box that was so sweet, so one's expectations play a part in user reviews. And not everybody has the same goals, styles and requirements.

### SOME SUM-DIFFERENCE MUSINGS

One school of thought suggests we use our gentlest and smoothest processors for mid-side tweaks, mostly because there can be inherent dangers with moderate alterations of those channels. But this is mastering, and a few dB of this or that is already verging on drastic, right? The other school of thought suggests that sometimes a narrow notch or steep filter on either the sum or difference can effectively treat problems that can't be fixed any other way. Perhaps you will find that these two inserts are better served with your best EQ's than with dynamic processors. Equalizing in the sum-diff domain relaxes some of the demands of precise left-right matching.

### “WIDTH”

We've just talked about how subtle SUM-DIF processing can be very useful. With that said, be very careful not to mangle the stereo image too much - this is a tool that is often over-used, abused, and very poorly understood. Certainly you have heard speakers wired out-of-phase and heard a stereo pair and checked out what happens if one side is phase-reversed. It is not pleasant and can be rather disconcerting. Increasing the stereo width is employing the same effect but to a lesser degree.

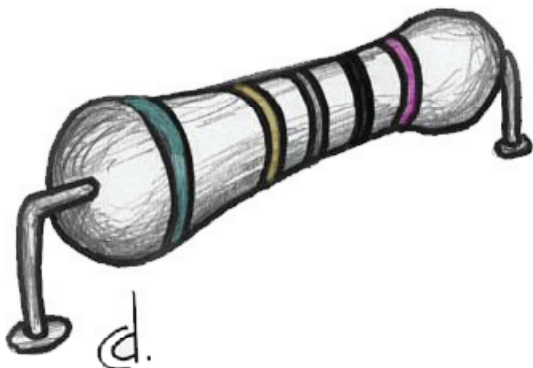


## “WIDTH” cont’d...

So we’ve all heard some “big” sounds before that result from mid-side tweaks. But did you know this? Let’s say you increase the width by just 1 dB (which is pretty subtle). What begins with maybe 100 dB of separation between left and right diminishes quickly. Now a sound that was panned hard left will also be on the right side but out-of-phase and down about -20 dB. So, what seems only a hair wider actually reduced left-right separation from 100 dB to 20 dB. Maybe this could be a concern if you expect some people will listen on headphones. Maybe mixes with cool dramatic hard panned tracks might actually LOSE some width along with losing center. Hmmmm.

To do what we want, an analog circuit requires that 2 signals get subtracted (and added) accurately, and this happens twice. It relates to a common spec called “common mode rejection ratio” or CMRR. In general, good CMRR is spec’d for a low frequency because the numbers are best there and get worse as the frequency gets higher. For example, a typical op-amp like a MC5534 can claim a best case scenario CMRR of 90 dB at 60 Hz. Looking at the graph, however, that number drops to 40 dB at 20 kHz. Not very impressive, if we hope for 80 dB across the entire spectrum as a base-line. A typical high quality resistor may have a tolerance rating of 1% which means it is likely we only get 40 dB of separation in practice. Even a superb resistor might be 0.1% tolerance which translates to 60 dB. So to make a long story short, it ain’t easy to get 80 dB of separation from 20Hz to 20KHz in a high performance sum-diff and back circuit in analog. It is easy in digital when the operations are just a few additions and subtractions, and 24 bit math virtually guarantees numbers in the range of 144 dB. But these are just numbers and the previous paragraph suggests that once one actually adjusts or processes sum-diff signals those numbers quickly become low anyways.

Perhaps some of the feeling that increasing the width is practically necessary in mastering might be due to a few common modern studio techniques that may accidentally strip some subtle localization cues. For example, the ear uses the initial onset of transient peaks as well as phase and level differences as localization cues, and now we have more dynamic processors (compressor plug-ins on every track, group and mix bus) in more studios than ever before in history. We also have more tracks and denser arrangements. We time shift tracks and use more high-pass filters (messing with phase). Not that these things are bad, but one should be aware that there are always trade-offs, and some say that “Engineering is the art of balancing trade-offs”.



Vive la  
Résistance!

## MORE ON THE MIX/FADE KNOB

Traditionally, mastering engineers used a mix function when cross-fades between two analog tape machines were used to segue between songs. Now that is much easier done in the workstation. But in recent years, with the availability of convolution reverb programs and sizable libraries of impulse responses (IRs) on the internet, some mastering engineers are finding that a touch of convolution reverb helps gel drier mixes. The effect should not be like “reverb” in the traditional sense but more like a naturally complex flavor. Not surprisingly, the most likely IR candidates are fast decaying real rooms somewhat on the “dry” side. The bad news is that it can be unpredictable, which IR works best with each session or even each song, and auditioning IRs can be time-consuming and tedious. In any event, it is a dangerous technique and one to be used sparingly and subtly, unless the client requests to put reverb on a whole mix. But there are plenty of free “colors” to add to your palette if you have the inclination to experiment on your own time.

There are other possible sources you may want to try. For example, both the sum and difference signals appear on the back panel XLRs and you may find that you can do some really dramatic equalization or compression on either of those and mix it lightly back into the stereo mix, similarly to what was described in the “Sum-Difference Musings” section earlier. You would need a special XLR “Y” cable to do this right. This cable would start with a female XLR, with two balanced lines coming out of it to two male XLRs. *VERY IMPORTANT NOTE:* One male XLR gets wired conventionally and the other male XLR needs to be wired reverse phase and that XLR should get boldly marked “RIGHT”. There is a diagram of this cable on page 21.

Another possibility is introducing a processor that might ordinarily be considered too colored or distorted for mastering but if mixed in lightly (as opposed to running the full signal through it), can be brought in under the main signal. This technique may tend to vary dynamically depending on the nature and depth of the distortion. For example, if it is set up to be a conventional clipping effect, during quiet passages, there may be no clipping, during moderate levels where there is some clipping, the mixed-in effect might seem significant. However, with even hotter passages, the clipping is preventing that effect from gaining more output level, while the dry signal has had no such restraint, so the relative mix will lean more toward dry. Pre-limiting that color-box can prevent the distortion from becoming too nasty.

Yet another combination that can be excellently mixed in is the limiter-EQ, where you have the EQ set for drastic high boost or drastic “smiley face” boosted low & highs & dipped mids. What happens is that as the signal level drops, with this low threshold limiter-EQ, the proportion of “wet” signal becomes relatively stronger. In the first case, it would give more highs and air at low levels without becoming harsh and biting once the song gets loud. The second case acts a bit like an automatic Fletcher-Munson EQ where both highs and lows seem bigger at lower levels (opposite of what our ears normally do, so it sort of “compensates”) and this happens without the usual drawbacks of just boosting lows and highs. It is probably best to have the limiter set up to be digging in at relatively lower levels, which is quite the opposite of typical limiter settings. Note, that normal transients are passed by the “dry” part of the chain. Also, these kinds of tricks can bring a new motivation for keeping gear that may have been collecting dust, because it doesn’t require the ultra-high performance that you need for the primary chain.

*For some easy-to-follow parallel processing diagrams, check out Page 23...*

## EXPLORING OTHER USES

We've had a few interesting questions regarding different applications of the BACKBONE. Some customers have decided it would make an excellent teaching tool. The ability to simply and easily demonstrate the importance of gear sequence (especially when making use of the SWAP button) makes the BACKBONE well-suited for educational purposes. Still others are planning to use their BACKBONE for tracking needs, rather than the mastering applications for which it was designed.

In terms of tracking, another somewhat more complex use of the BACKBONE has been suggested by a few people - using it to route three different mics through many different microphone preamplifiers. We didn't design the BACKBONE for this purpose, but we DO build a product called the Manley MicMAID that is designed to do just that very thing. Check out our website ([www.manley.com](http://www.manley.com)) for more information!

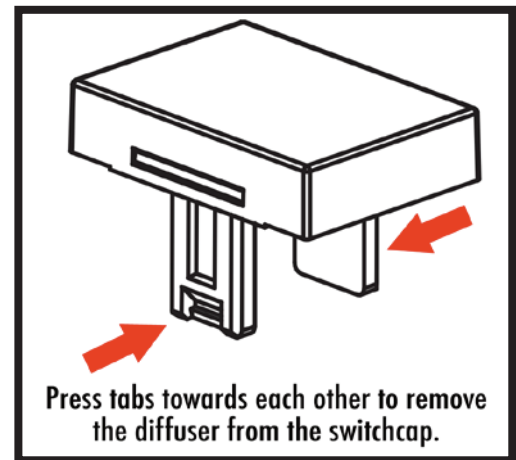
As mentioned before, the BACKBONE was not designed for this purpose. Because of the way the unit's signal flow is structured, using a mic pre as an insert has one major complication - **UNDER NO CIRCUMSTANCES CAN YOU USE +48V PHANTOM POWER WITH ANYTHING CONNECTED TO THE BACKBONE.** Turning on +48V on your mic pre would mean sending those 48V into your BACKBONE, which would potentially (read: "almost certainly") damage crucial components. Also, the BACKBONE was designed for line-level signals, not mic-level ones. Since signal from a microphone has about 1/100th the level of a line-level one, and the BACKBONE has not been optimized for such a signal, it simply wouldn't function correctly for this application.

With that said, once a signal has been brought up to line level it can be used in many ways with the BACKBONE. Tracking, mixing, mastering - use your imagination!

## LABELING YOUR SWITCHES

The EAO switches for each input and insert on the front panel can be labeled, e.g. "EQ #1", "COMP #4", etc... If you desire the label to be on the INSIDE of the button (perhaps slightly more permanent than just a sticker on the outside), we advise that you go to a local office supply store and purchase a P-Touch machine (or a similar label-printing device) and some "black on clear" tape for it. This will let the light shine through around the letters, enhancing legibility.

To remove the switchcaps, you'll just need a pair of pincer-style pliers - you have to be able to grip the top and bottom sides of the switchcap and pull it away from the socket. You will notice some small indents on these sides of the switchcap, which make it easier to remove. Be very gentle when you grip and pull the switchcap - the more delicate the pliers, the better. Once you have the switchcap in hand, you can easily pull out the diffuser by pinching the two tabs towards each other and pulling away from the clear cover. Print your label and stick it to this diffuser (observing the orientation of your lettering so that it is not upside-down and/or backwards), then put the diffuser back inside the clear switchcap. Re-installing the switchcap is as simple as pushing it back into the socket. Now you're done!



## TROUBLESHOOTING

**NO POWER, NO LIGHTS, NADA** - Probably something to do with AC power. Is it plugged in? Is the mains voltage set correctly for your country? Also, check the fuse on the back panel. A blown fuse often looks blackened inside or the little wire inside looks broken. A very blackened fuse is a big hint that a Very Bad Thing occurred. Try replacing the fuse with a good one of the same value and size. If it blows too then prepare to send the unit back to the dealer or factory for repair. The fuse is a protection device and it should blow if there is a problem. If the unit works with a new fuse, fine.

**LIGHTS BUT NO SOUND** - First, if you have gear plugged into the BACKBONE's inserts, take each one out of circuit by pressing whichever insert buttons are lit to turn them off. Remember, the BACKBONE's inserts are wired in series, which means that if one piece of gear is muted AND selected into the signal path, everything after it won't get any signal, including your speakers. This also means if you have an empty insert (no piece of outboard gear connected to it at all) and select this insert on the front panel, your signal will disappear...

Next, try disengaging all of the BACKBONE's active stages: this means GAINS, SUM-DIF, and MIX/FADE. If there is a problem with one of these active boards, it could cause signal to disappear or be otherwise altered.

Also try checking the signal from the output labeled "SEL INPUT" on the rear of the BACKBONE - this output is hard-wired directly from your selected source. If this is muted as well, you might want to check your source into the BACKBONE and make sure there's a signal going in.

*\*\*\*IMPORTANT NOTE\*\*\* Your Manley Mastering BACKBONE was designed to interface primarily with fully balanced equipment. It is happiest that way. It is also fine with receiving signal FROM an unbalanced unit, but matters are complicated when the BACKBONE tries to output signal TO an unbalanced unit.*

*If you intend to use any unbalanced gear with your BACKBONE, you must ensure that Pin 3 is floating, not grounded. Grounding Pin 3 can potentially cause damage to the BACKBONE, and will likely introduce unwanted noise into the signal path. By floating Pin 3, you can safely utilize unbalanced equipment - but remember that in this scenario, signal level will drop by 6 dB.*

*(Note: this applies to ALL outputs of the BACKBONE, including the ELCO/DB25 snakes.)*

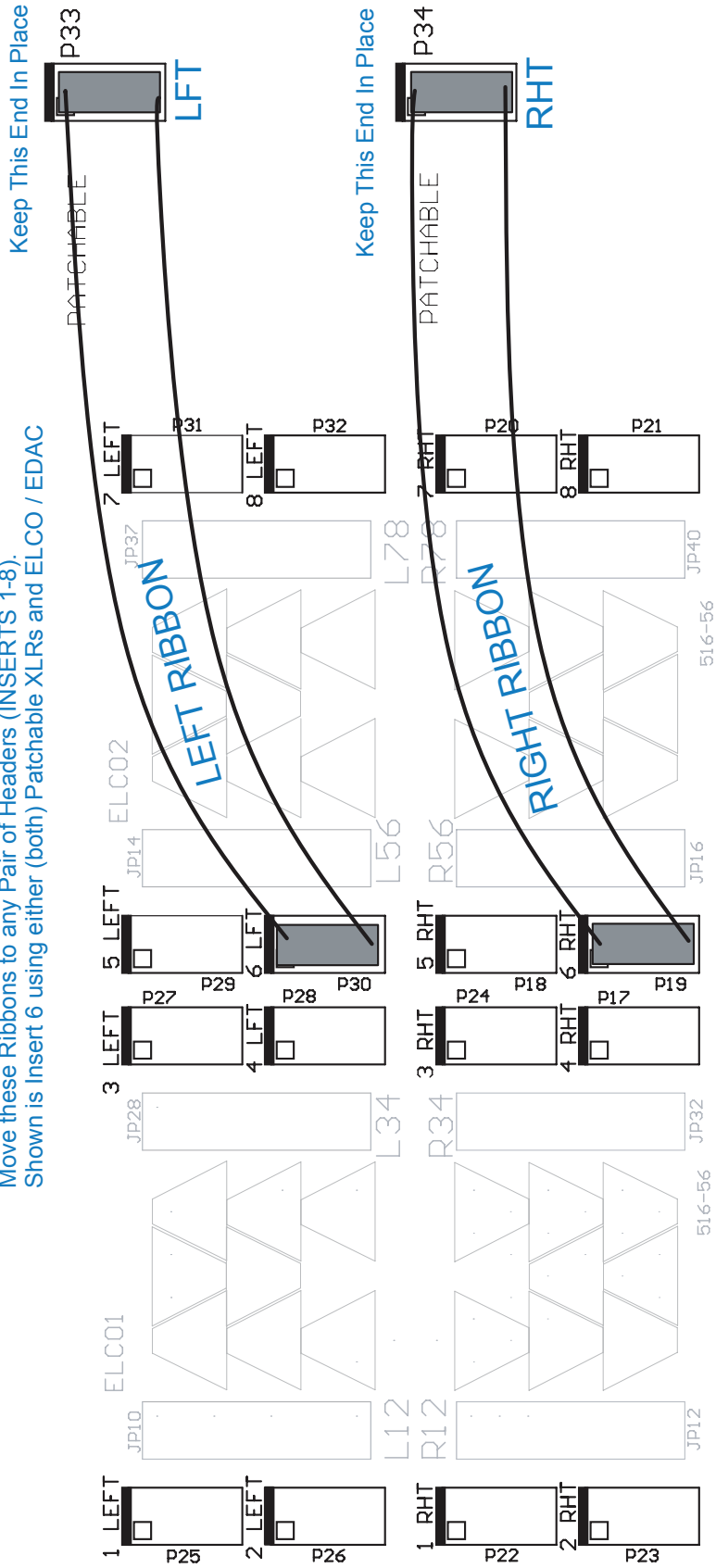
*There is one situation where you can always drive an unbalanced input from your BACKBONE. Because Insert 8 is positioned AFTER your output gains, if your Insert 8 has transformer outputs (or idiot-proof transformer-like outputs), that processor would effectively act as the output amplifier and be much more tolerant of unbalanced lines connected to your BACKBONE's output XLRs.*

*All things considered, we recommend sticking with balanced gear after your BACKBONE to avoid complications.*

# INSERT PATCHING

## Interior View of Back Panel

Move these Ribbons to any Pair of Headers (INSERTS 1-8).  
Shown is Insert 6 using either (both) Patchable XLRs and ELCO / EDAC



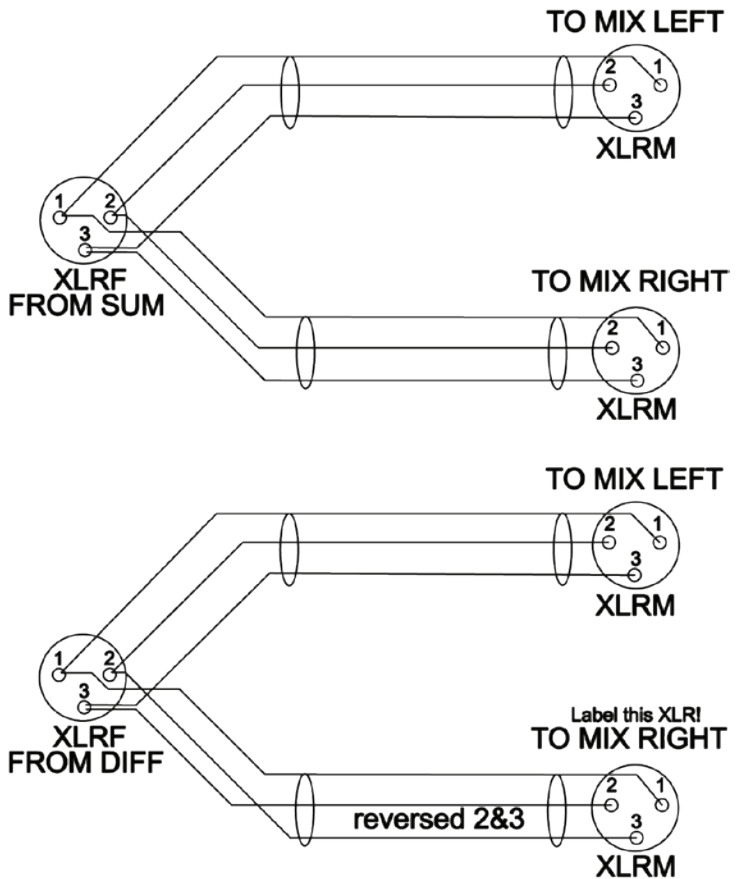




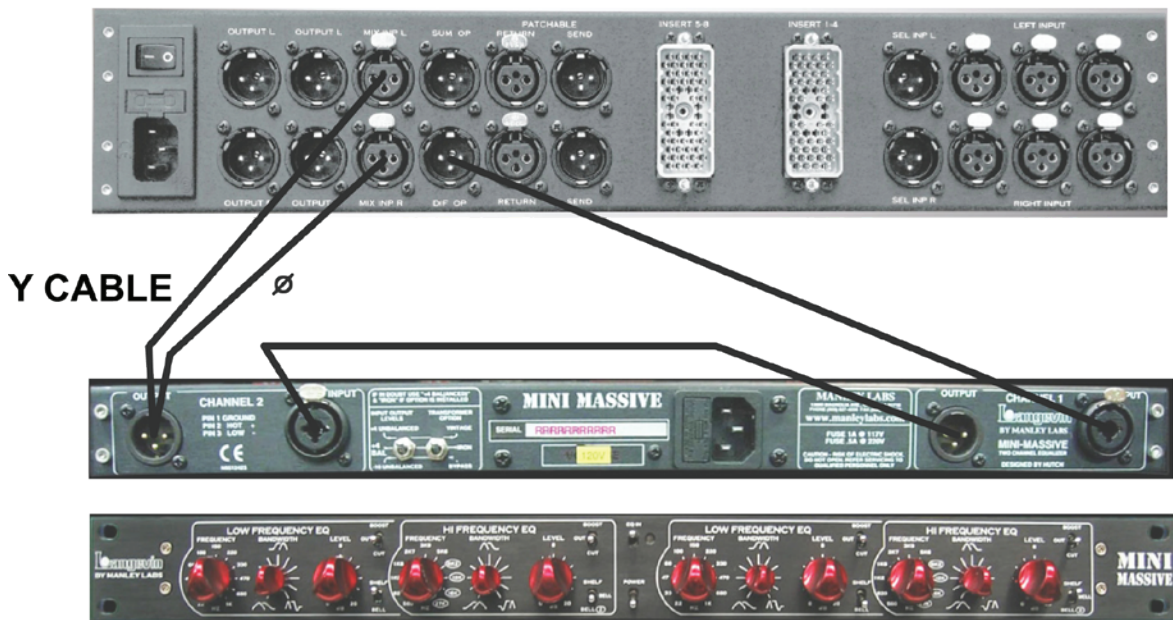


# Y-CABLE DIAGRAM

Note that the Y-cable coming from the SUM output should be wired differently from the Y-cable coming from the DIFF output.

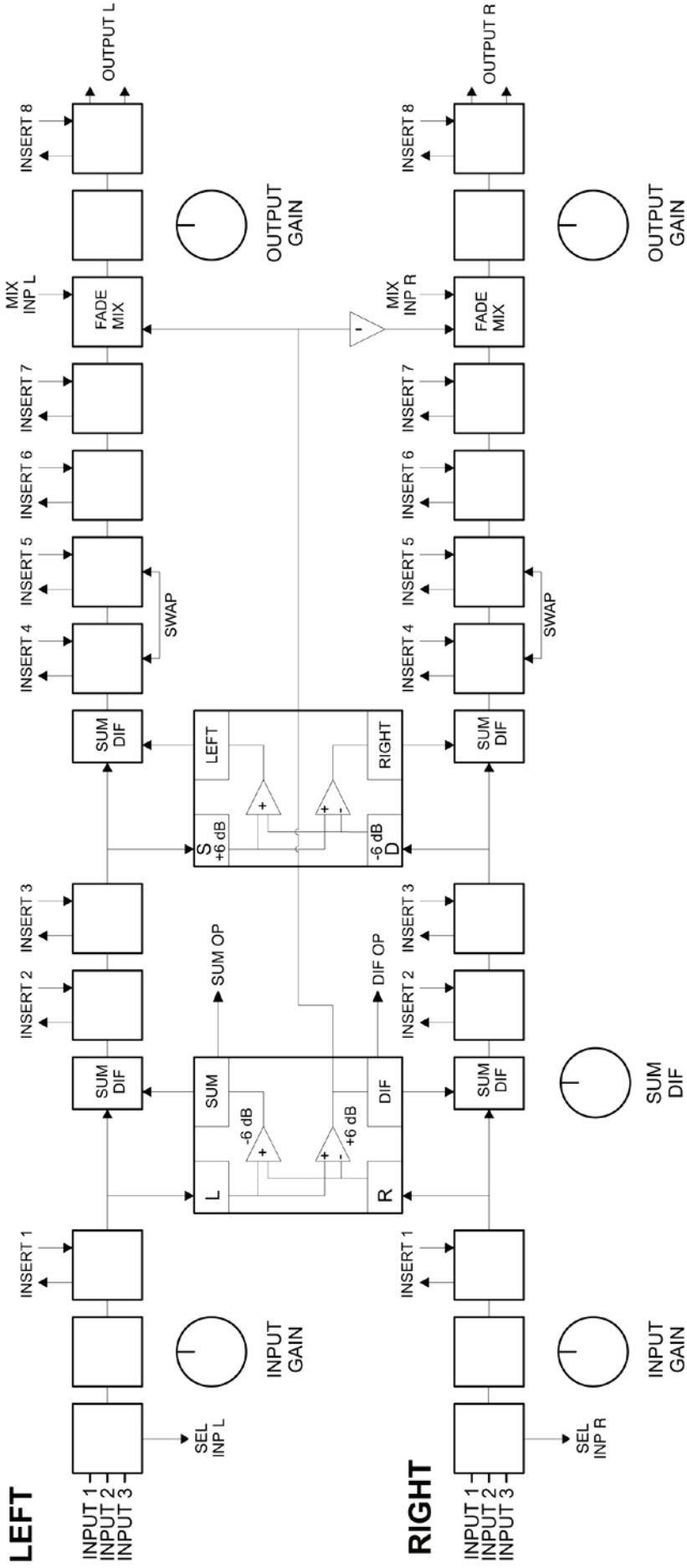


Example showing cascaded EQ (channel 1 into channel 2) processing DIF signal and patched to be subtly mixed into main path. In this case, the mastering engineer has 4 bands of EQ available.



It really is unlikely that simply patching SUM into Left Mix and DIF into Right Mix would ever be useful but the above example might be by virtue of subtlety and L/R separation. Mixing a small amount of DIF into the main path involves less electronics than using SUM-DIF to decode, processing using Inserts 2&3, then re-encoding to Left and Right.

# BACKBONE BLOCK DIAGRAM



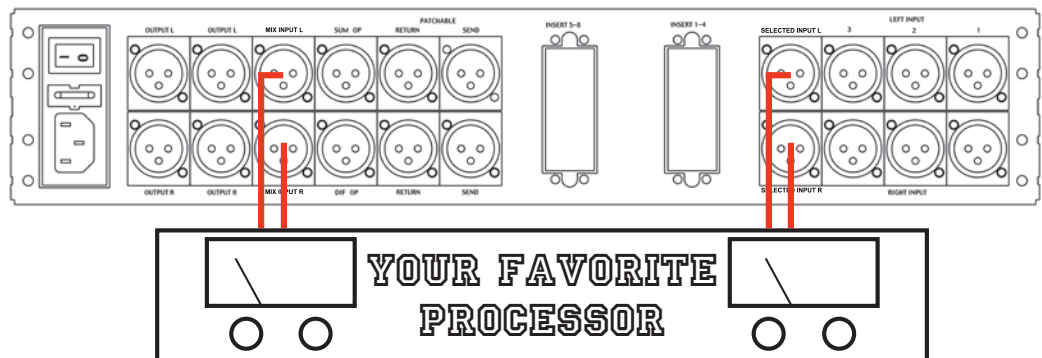
# PARALLEL PROCESSING DIAGRAMS

One term you may have heard over the years is “parallel processing” - but what exactly IS that, you may ask? When you duplicate a signal path and modify the duplicate in some way (compress it, add reverb to it, EQ it, or use any sort of processing), then mix it back in with the original, THAT is parallel processing. It allows you to add an effected sound to a mix (or a submix, or a single track) without losing the character or dynamic range of the original mix.

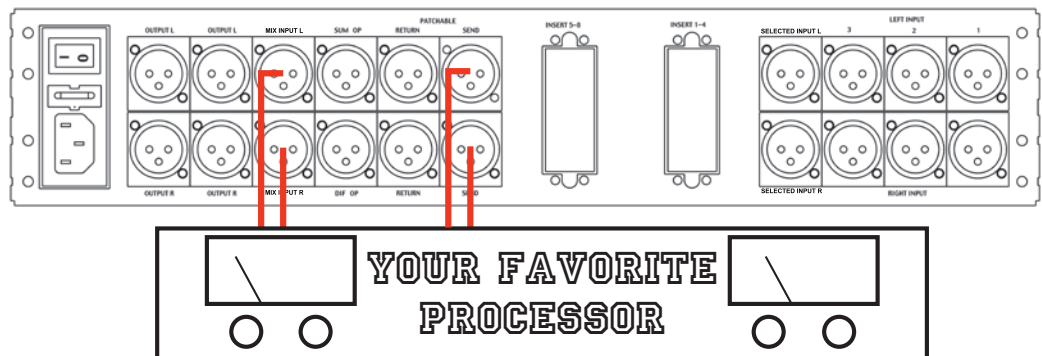
Imagine a really great, larger-than-life rock drum sound - thick, compressed, but still punchy and dynamic. If you heavily compress a drum mix, you have the potential to squeeze all the transients out of it. No more sizzle. It also loses dynamic range - now the cymbal crash on the downbeat of the chorus is barely louder than the high-hat during the verse! Instead, try parallel processing. Duplicate your drum mix onto two other tracks, then send those two new tracks through your favorite compressor. Squash them, beat them up, make them sound gritty and aggressive. Then, in your mixer - with your original sound up in the mix already - bring up the parallel mix. You’ll hear a weight and punchiness added to the sound, but you still retain all the transient response and dynamic range of the original signal. Find a balance between the two mixes, and...voila! You are now a master of parallel processing.

As stated on page 15, the BACKBONE’s MIX/FADE knob can be used to mix the main signal path with whatever is sent to the “MIX INPUT” jacks on the rear panel. Page 15 discusses using the SUM and DIF outputs to feed the MIX/FADE knob; in the cases shown below, the knob is being used to mix the BACKBONE’s main signal path with the the duplicated signal sent to the MIX INPUTS.

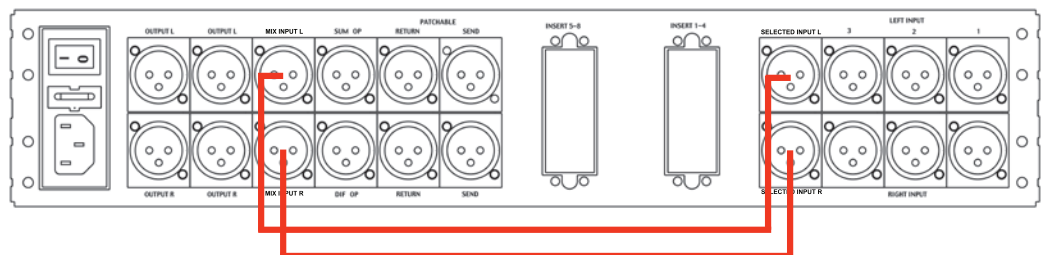
Here, the parallel chain is fed by the selected input on the BACKBONE.



Here, the parallel chain is fed by the PATCHABLE SEND, which can be set to send from anywhere in the signal path. See page 18 for instructions on patching.



Here, your main BACKBONE signal path is the “wet” signal, and the direct routing shown is the “dry” signal with no processing to mix in with the main signal path.



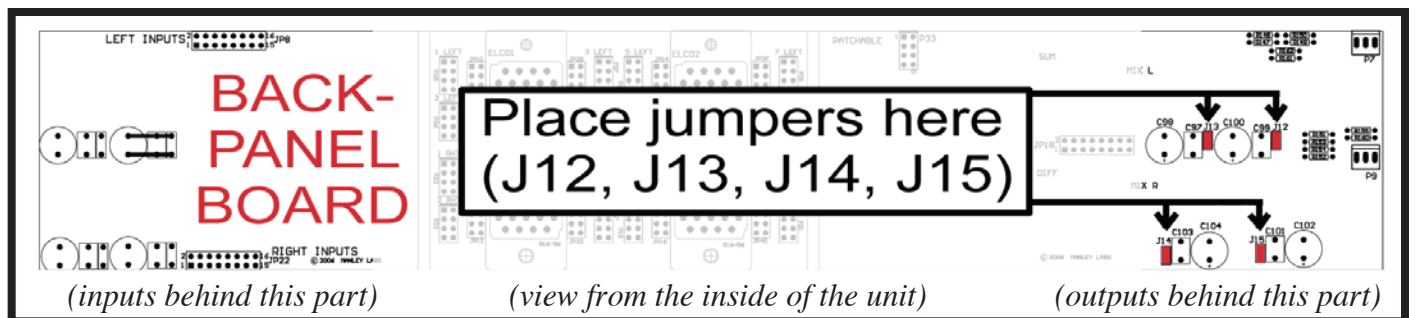
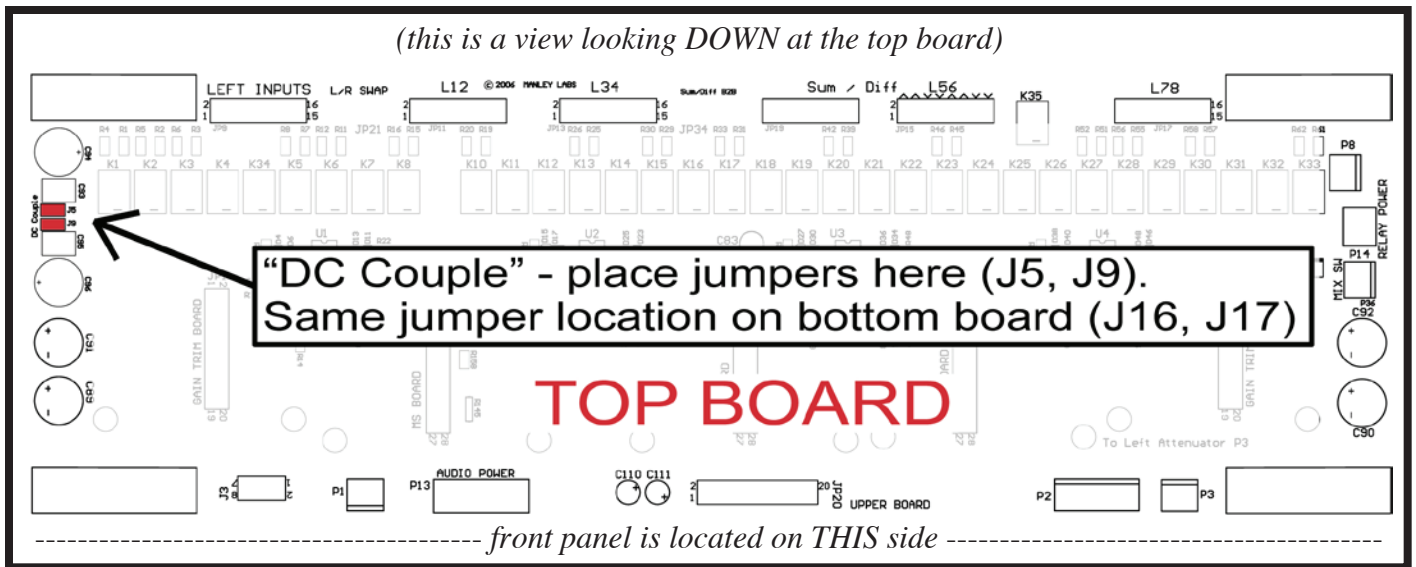


# DC-COUPLING I/O MODIFICATION

Your Manley BACKBONE left the factory with AC-coupled inputs and outputs. The coupling capacitors in this circuit serve to prevent any pops and clicks that might occur when changing inputs or selecting inserts. We at Manley don't believe that these high-quality capacitors have much (if any) noticeable effect on audio passing through the BACKBONE, but some purists may disagree. We've designed the unit so that these capacitors can be bypassed easily, with no soldering required. Easy as pie\*!

The jumpers that control the input coupling are located on the main boards. If you're looking head-on at the BACKBONE with the top and bottom covers removed, they'll be located near the left side of the top board facing *up*, towards the back of the board. They are labeled J5 & J9, and are for the left channel. Jumpers for the right channel inputs live directly below these, on the bottom board facing *down*, and are labeled J16 & J17. The jumpers are factory-installed to be only half connected, thus leaving one of the two pins open. To make your BACKBONE's inputs DC-coupled, simply remove the jumper from the one pin it is seated on, and put it back so that it connects the two pins together.

Now we modify the outputs the same way. The same type of jumpers are located just behind the output XLR connectors on the back-panel board of the BACKBONE. These will be labeled J12, J13, J14, and J15. Use the red jumpers in the same way to connect each of these pairs of pins. Now your BACKBONE's outputs are DC-coupled too! Yes, it really IS that easy.



\*Pie not included.

# SPECIFICATIONS

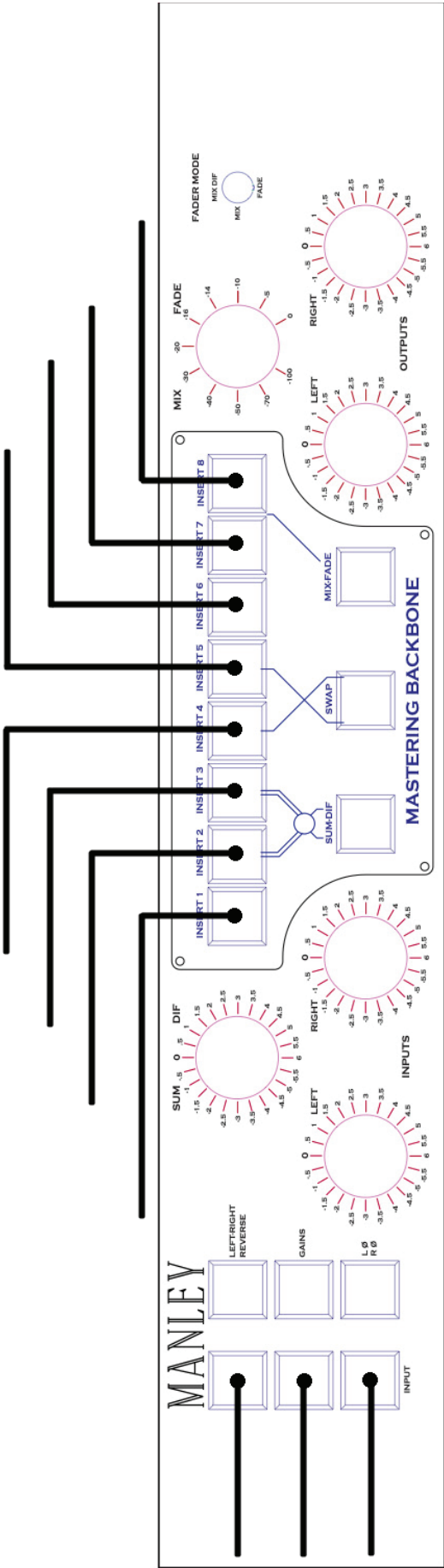
Frequency response.....	+/- 0.1 dB, 10Hz-100KHz
THD + Noise.....	0.0015% @ +4 dBu input level 0.0011% @ +22 dBu input level
IMD60 4:1.....	0.0017% @ +4 dBu input level 0.0035% @ +22 dBu input level
Crosstalk Rejection.....	≥ 95dB @ 1 KHz between Gains and Mix-Fade Stages ≥ 70db @ 1KHz at SUM-DIF Stage
Common Mode Rejection	
100Hz.....	≥ 100dB
1KHz.....	≥ 95dB
10KHz.....	≥ 80dB
Noise floor.....	-95 dBu, Bandwidth 22Hz-22KHz
Maximum signal level.....	+29dBu
Dynamic Range.....	124dB
Input impedance.....	30k ohms
Output impedance.....	150 ohms
Input Or Output Stepped Gain accuracy.....	+/- 0.05dB
Power Consumption.....	60 watts
Mains Fuse.....	120VAC= 1A; 240VAC= .5A Use 5mm x 20mm SLO-BLO glass fuses, 250V AC rating

NOTE: Operating mains voltage: Units are purpose built for original destination country's mains voltage (100V, 120V, or 220-240V) as indicated on the serial number badge. The power transformer must be **rewired** in order to change mains operating voltage.

Unit Size: 19" x 10" x 3.5" (occupies 2u) *Note: add 5" behind unit for cable allowance*  
Unit Weight: 13 lbs.  
Shipping Weight: 19 dim. (international); 17 lbs. (USA)  
Shipping Box Dimensions: 23" x 15" x 9"

Specifications subject to change because they just might.

# SETTINGS TEMPLATE



Notes:

- \_\_\_\_\_
- \_\_\_\_\_
- \_\_\_\_\_
- \_\_\_\_\_
- \_\_\_\_\_
- \_\_\_\_\_
- \_\_\_\_\_
- \_\_\_\_\_

