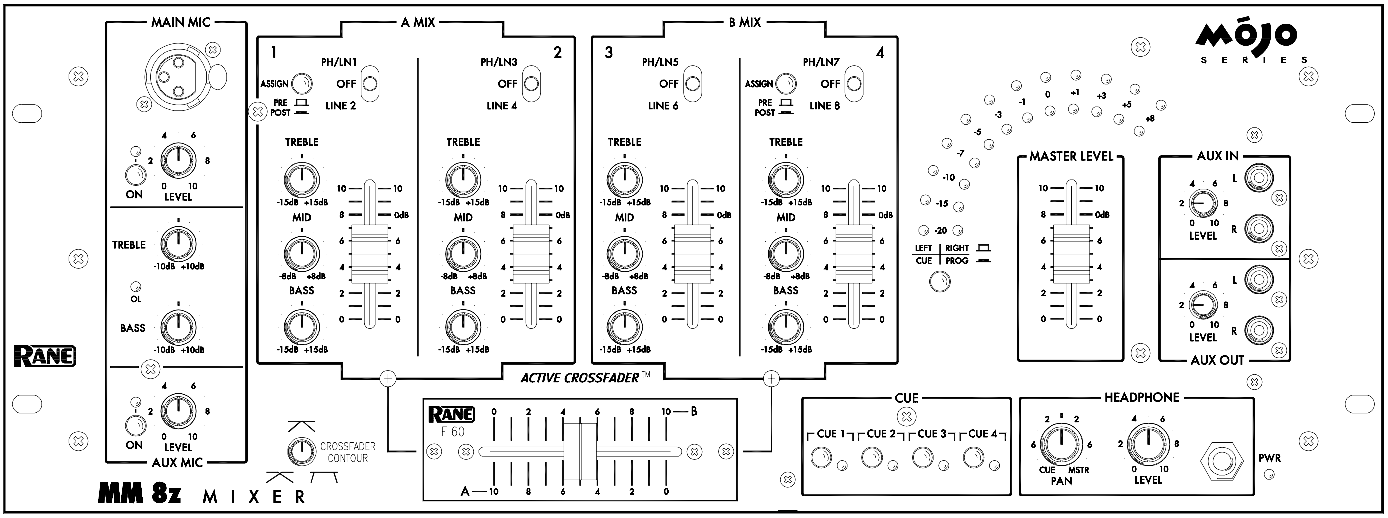
OPERATORS MANUAL MM 8z

MOJO CLUB MIXER



QUICK START

Here are a few basic “plug and play” steps to get you performing. If you are a first-time DJ mixer user, please do yourself a favor and at least read this section.

Begin by making sure your amplifier is off before making connections. The MM 8z has all unbalanced connectors, except for the ¼" balanced Main Outputs and XLR Mic Inputs. Plug a source component, such as a CD player, into one of the four **LINE** Inputs on the rear. Connect the **MAIN OUTPUTS** to your amplifier. Set all front panel controls to the middle of their travel and all pushbuttons to their *out* position. Slide the **MASTER LEVEL** down to **0**. Set the **INPUT SELECT** switch to your active Source Input. Plug in the MM 8z’s power supply and see the **PWR** indicator illuminate. Turn on your source and amplifier. Slowly turn up the **MASTER LEVEL,** as the meters light and music is heard from the speakers.

You can get tripped up at two places. If you have a phono signal into one of the four **PH/LN** Inputs, be sure the **PH/LN** switch is in the **PHONO** position (*in*); likewise, when using a tape deck or CD into these Inputs, be sure the switch is in the **LINE** (*out*) position. If you plug into the **PROGRAM LOOP**, the internal signal path is broken. Be sure that a complete loop is made, to and from an outside device.

Couldn’t be easier, right?

With the **CROSSFADER CONTOUR** set to full counter-clockwise position, it operates as a typical constant-power crossfader. Set to clockwise position, the crossfader has the very steep slope as indicated in Figure 1 to the right. Now that was a pretty quick start, right?

Never connect anything except a Rane RS 1 to the thing that looks like a red telephone jack on the rear of the MM 8z. This is an AC supply and requires some special attention if you do not have an operational power supply ***exactly*** like the one that came with your unit. Consult the Rane factory for a replacement or substitution.

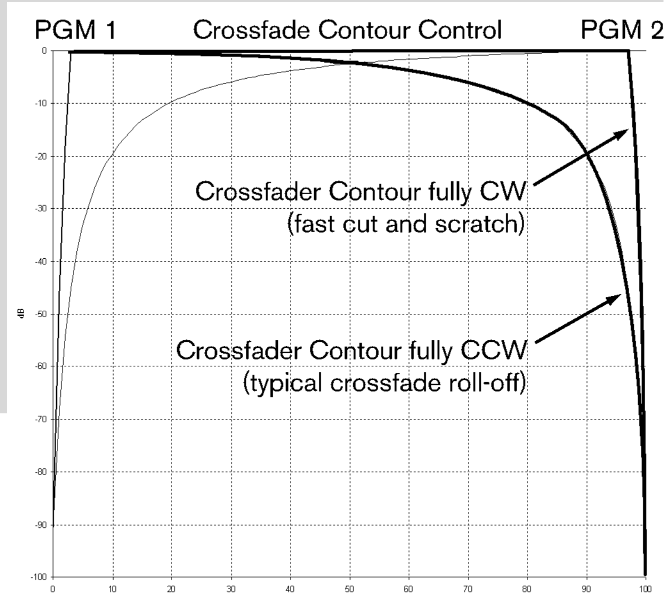
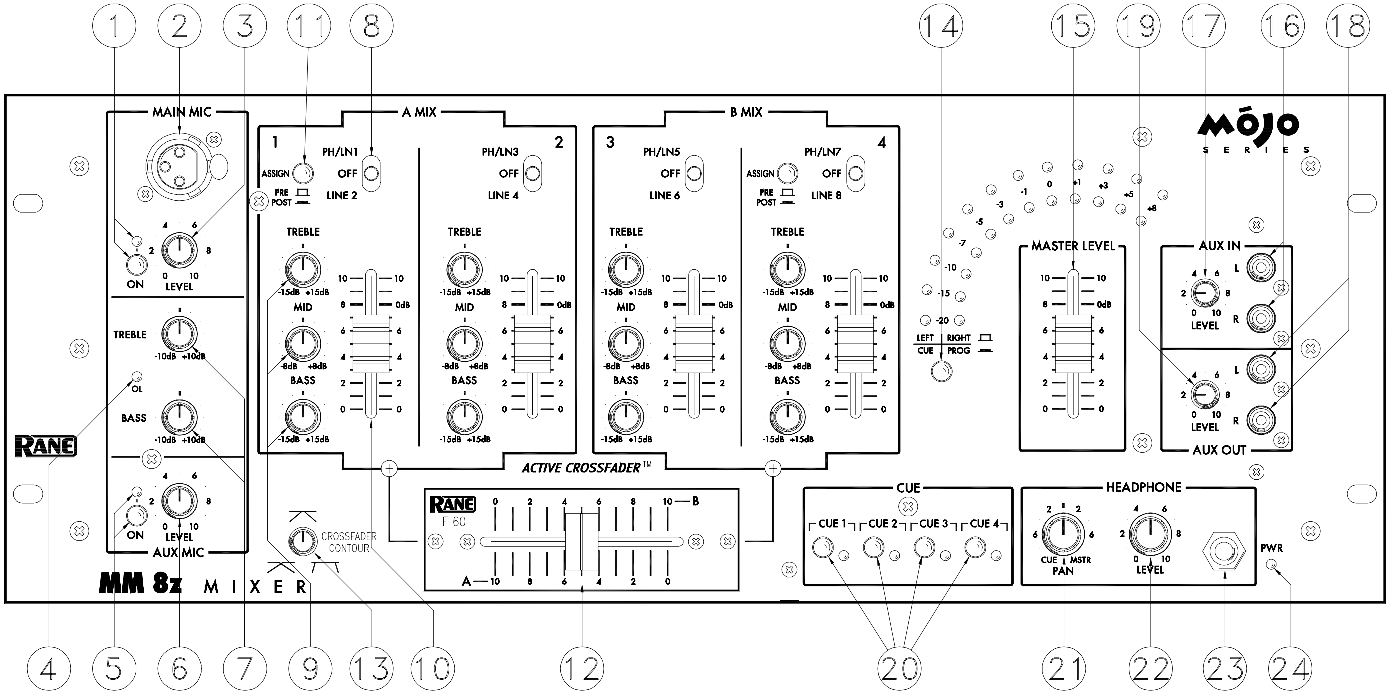


Figure 1. Crossfader Contour control

WEAR PART

This product contains the following wear part subject to the ninety (90) day warranty period described on page Service-1: *(1) Active Crossfader Assembly F 60.*

FRONT PANEL DESCRIPTION



① **MAIN MIC ON switch:** puts the Main Mic signal into the mixer signal path. When the switch is pressed *in*, the Main Mic is *on* and the adjacent red indicator blinks.

② Front panel **MAIN MIC input:** accepts a balanced microphone with an XLR connector. A parallel Main Mic Input is on the rear of the unit. *Do not connect microphones to both front and rear Main Mic inputs.*

③ **MAIN MIC LEVEL control:** adjusts the level of the Main Mic Input.

④ **MIC OVERLOAD indicator:** monitors both Main and Aux Mic Inputs, before and after the Mic EQ. It lights when the signal exceeds the Mic section's output capability (3 dB below clipping). Occasional flickering is acceptable; however, steady lighting requires a reduction in the Mic LEVEL control to prevent distortion.

⑤ **AUX MIC ON switch:** puts the Aux Mic signal into the mixer signal path. The adjacent red indicator blinks when the switch is pressed *in*, indicating that the Aux Mic is *on*.

⑥ **AUX MIC LEVEL control:** adjusts the level of the Aux Mic Input.

⑦ **MIC EQ controls:** contour the frequency response of the summed Main and Aux Mic Inputs.

⑧ **Source Input Select switch:** selects either a Phono/Line or Line Input for the Source. In the center position the Source Input is muted.

⑨ **Source EQ controls:** contour the frequency response of the selected Input.

⑩ **Source Input fader:** controls the level of the selected Input.

⑪ **Source ASSIGN PRE/POST CROSSFADER switches:** are provided for Inputs 1 and 4. In the PRE (*up*) position, this Input is routed to the Crossfader. In the POST (*down*) position the Input bypasses the Crossfader.

⑫ **ACTIVE CROSSFADER:** controls the relative output level from the summed A and B Mixes. When the fader is at its far left, only the A Mix is heard from the Outputs. As the fader is moved toward the right, the amount of B Mix is increased and the amount of A Mix is decreased. When the fader is centered, equal amounts of A and B Mixes are routed to the Outputs. Fully right is all B Mix at the Outputs.

⑬ **CROSSFADER CONTOUR control:** Allows adjusting the "shape" of the Crossfader response from a gentle curve for smooth, long running fades, to the steep pitch required for top performance cut and scratch effects. See Figure 1 on page Manual-1.

⑭ **Meter and Mode switch:** This peak-dBu reading meter displays one of two modes, depending on the switch position. In the LEFT/RIGHT (*out*) position, the meter indicates the stereo level in the LEFT and RIGHT Main Outputs. In the CUE/PRG (*in*) position, mono CUE level is displayed on the Left meter and mono PROGRAM level is displayed on the Right meter.

⑮ **MASTER LEVEL fader:** adjusts the level at the rear panel Main Outputs.

⑯ **AUX INPUT:** This stereo pair of RCA connectors is an extra line level Input which sums with the other signals before the Master Level fader.

⑰ **AUX IN LEVEL control:** adjusts the level of the front panel Aux Inputs.

⑱ **AUX OUTPUT:** This stereo pair of RCA connectors provides an extra line level output mix. This is the same signal as the Main Outputs, but with a separate Level control.

⑲ **AUX OUT LEVEL control:** adjusts the level of the front panel Aux Outputs. This control is independent of the Master Level fader and Main Outputs.

⑳ **CUE switches:** Pressing *in* any or all of the CUE pushbuttons routes the respective Source to the Headphone and Meter Cue sections. The adjacent yellow indicator illuminates when the switch is depressed.

HEADPHONE PAN control: adjusts the relative levels of Cue and Master signals mixed together in stereo for the headphones. Counterclockwise rotation increases the amount of Cue signal; clockwise rotation increases the amount of Master signal.

HEADPHONE LEVEL control: adjusts the volume for the headphones.

HEADPHONE Output jack: This stereo ¼" TRS (Tip-Ring-Sleeve) jack accepts ¼" TRS stereo headphone plugs (do *not* use mono plugs).

POWER "ON" indicator: When the yellow indicator is lit, the MM 8z is ready to go.

Fader Cleaning

With heavy use in harsh environments, the faders may need lubrication. This treatment extends longevity and can make used faders as good as new. The fader assembly must be removed from the MM 8z for proper cleaning. We recommend any of the following cleaning solutions:

Caig Cailube MCL 100% spray lubricant
Caig Cailube MCL 5% spray cleaner
CRC 2-26

Order CaiLube MCL® from:
CAIG Laboratories, Inc.
12200 Thatcher Ct.
Poway, CA 92064
Phone 619-486-8388
Fax 619-486-8398
Web *http://www.caig.com*

REPLACING THE ACTIVE CROSSFADER

The Crossfader may be removed without any disassembly of the MM 8z itself, and may be performed while the unit is operating with no interruption of the audio signal.

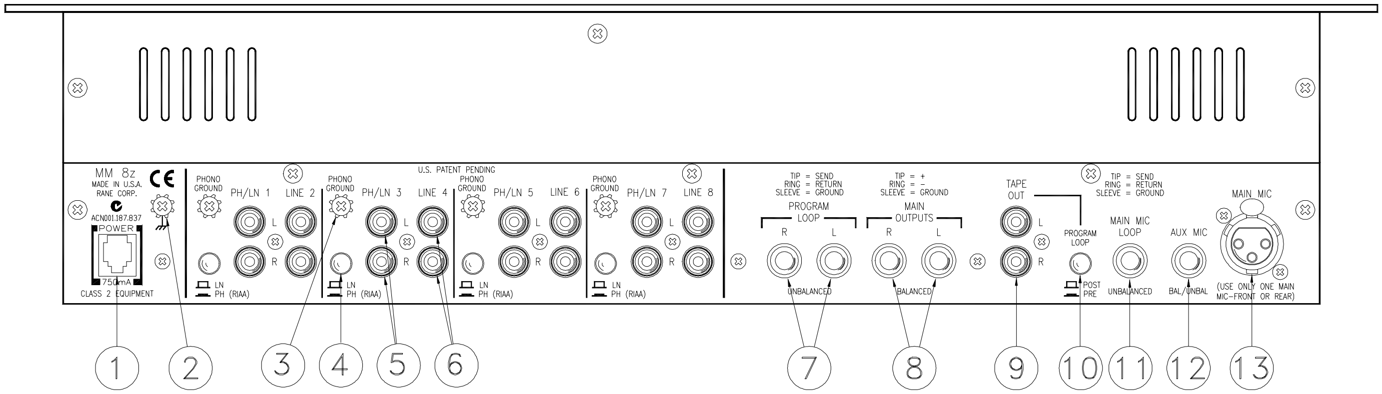
F 60 and F 45 Fader Kits are available from your local retailer or the factory. The kit includes full assembly including knobs, ribbon, and face plate.

1. Remove the two (2) outer screws attaching the crossfader assembly to the front panel.
2. Pull the Crossfader Assembly forward and unplug the ribbon from the connector on the bottom board.
3. Proceed with Cleaning Instructions, or install the replacement assembly by reversing the above instructions.

CLEANING INSTRUCTIONS

1. Hold the fader assembly away from the mixer.
2. Position the fader at mid-travel.
3. Spray cleaner/lubricant into both ends of the fader.
4. Move the fader over its full travel back and forth a few times.
5. Shake excess fluid from the fader assembly.
6. Wipe off excess fluid.

TOP / REAR PANEL DESCRIPTION



① **Remote POWER Supply Input:** The unit is supplied from the factory with a Rane Model RS 1 remote power supply suitable for connection to this input jack. The power requirement of the unit specifies an 18 volt AC center-tapped transformer only***. This is not a telephone jack. Never use a power supply with your unit other than the one supplied or a replacement approved by Rane Corporation.*** Using any other type of supply may damage the unit and void the warranty.

② **Chassis ground point:** This screw is provided for grounding purposes. This unit comes with an outboard power supply which *does not ground the chassis through the line cord*. The MM 8z can be grounded either to another chassis which is earth grounded, or directly to the grounding screw on an AC outlet cover with a wire connected to this chassis screw.

③ **PHONO GROUND screw:** These screws provide a place to connect the ground wire from a turntable.

④ **PHONO/LINE switch:** These switches change the Input from a PHONO (pushbutton *in*) to a LINE (pushbutton *out*).

⑤ **PHONO/LINE INPUTS:** These stereo pairs of RCA connectors are an Input for a PHONO (RIAA) stage for magnetic cartridges or a LINE stage suitable for any device, such as a CD player.

⑥ **LINE INPUTS:** These stereo pairs of RCA connectors are an Input for any LINE level device.

⑦ **PROGRAM LOOP jacks:** These ¼" TRS (Tip-Ring-Sleeve) jacks allow stereo external processing of the Program signal. The tips connect the sends to the processor inputs, and the rings connect the returns from the processor. These are switching jacks—always complete the loop when connecting a send and return, or no sound will be heard.

⑧ **MAIN OUTPUTS:** These ¼" TRS jacks provide a balanced line level output.

⑨ **TAPE OUTPUTS:** This stereo pair of RCA connectors is a line level Output of Mix program (without Mic or Aux Inputs). The signal is unaffected by the MASTER LEVEL fader. It is intended for use with a tape recorder, but is not restricted to that purpose.

⑩ **PRE/POST LOOP switch:** With the switch pressed in the PRE *(in)* position, the TAPE OUTPUTS have the signal before processing by an external device connected to the PROGRAM LOOP. Releasing the switch to POST *(out)* provides the TAPE OUTPUTS with the signal that has been processed by an external device connected to the PROGRAM LOOP. If no plugs are inserted in the PROGRAM LOOP, this switch has no effect.

⑪ **MAIN MIC LOOP jack:** This ¼" TRS jack is for inserting external mono signal processing in the MAIN MICROPHONE circuit path only. The tip connects the send to the processor inputs, and the ring connects the return from the processor. This is a switching jack—always complete the signal loop when connecting a send and return, or no sound will be heard.

⑫ **AUX MIC jack:** This ¼" balanced TRS Input accepts wireless mics or another line-level mono source.

⑬ Rear panel **MAIN MIC input:** This XLR Input connects a balanced microphone. The same MAIN MIC Input is on the front of the unit. *Do not connect microphones to both front and rear MAIN MIC Inputs.*

MM 8z CONNECTION

With the MM 8z's ability to accommodate a wide variety of systems, these basic guidelines will assist the user in incorporating this mixer into their equipment setup.

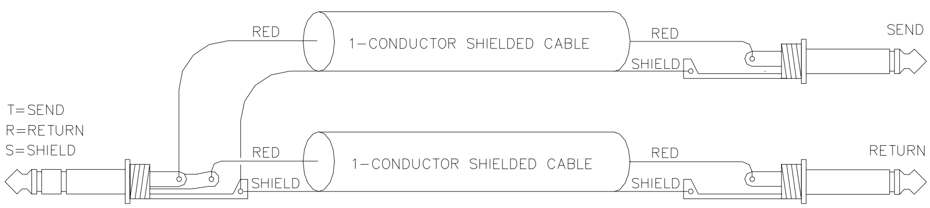
Since most source components (e.g., turntables, disc players, tape decks) used with the mixer are consumer grade, the MM 8z features unbalanced RCA Source Input connectors.

When using the PH/LN inputs, be sure the adjacent switch is in the correct position for the connected device: pushbutton *in* for phono, *out* for line level. Ground screws attach the turntable ground wires, which help eliminate hum or buzz.

The front and rear panel Main Mic Inputs are XLR connectors, for use with a balanced output microphone of any impedance. Use only one mic at a time in the front *or* rear jack. Effects can be inserted in the Main Mic signal path thru the ¼" unbalanced TRS MAIN MIC LOOP jack. Use a special TRS send/return cable: tip = send, ring = return, sleeve = ground. Refer to the wiring diagram below. The Auxiliary Mic input is ¼" balanced TRS jack, useful for wireless mics and other high-impedance sources.

The MAIN OUTPUTS are ¼" balanced TRS connectors, which provide good hum rejection and allow long (greater than 10 feet) lengths of interconnect cable without significant losses (see *Sound System Interconnection* on page Manual-10 for proper wiring of connector/cable). The PROGRAM LOOP has left and right ¼" unbalanced TRS connectors for sending the Source program signal to an external effects device and returning the signal back to the mixer. Use a special TRS send/return cable: Tip = send, Ring = return, Sleeve = ground. Refer to the wiring diagram below. Remember, these are switching jacks—always complete the loop when connecting the send to and return from an external device.

Two sets of outputs can provide convenient connections for recording equipment. The TAPE OUTPUT unbalanced RCA jacks provide an output for recording program material. If you want signal processing via the Program Loop to have an effect on the recording, let *out* the PRE/POST Record push-button to the POST position. The TAPE OUTPUT *does not contain any signal from the Mic or Aux In sections*. If you need to record the mics, use the Aux Out for your recording signal; this output is a composite of the Program, Microphone, and Aux Inputs.



Send-Return Cable Wiring

OPERATING INSTRUCTIONS

SOURCE SELECTION

The MM 8z is able to mix together four Sources. Each Source is switchable between two line Inputs, one of which may be switched to a Phono Input. The center OFF position of the Source Input Select switch mutes it.

The ASSIGN switch for Source 1 in the PRE position combines PH/LN 1 or LINE 2 with PH/LN 3 or LINE 4 for the A Mix. Likewise, the ASSIGN switch for Source 4 in the PRE position combines PH/LN 5 or LINE 6 with PH/LN 7 or LINE 8 for the B Mix. These Mixes are routed to each side of the Crossfader, A & B. Either ASSIGN switch in the POST position routes Source 1 or 4 directly to the Master Level control, and routes Source 2 or 3 through the Crossfader before going to the Master Level control.

SOURCE EQUALIZATION

Each selected Source's frequency spectrum can be contoured with the Treble, Mid, and Bass controls. These are intended to provide EQ between varying program material. The Treble and Bass controls offer 15 dB of boost or cut; the Mid control offers 8 dB of boost or cut. Positioning any control to the "12 o'clock" position turns that equalization band off.

SOURCE FADERS

The Source faders control the level of the selected Input. They also provide a means to set relative levels for each Input to A and B Mixes. Set the Source faders near their maximum levels (0 dB) instead of increasing the gain later at the Master Level control or power amp. You achieve optimum noise performance by having the majority of the gain at the input stages. Taking the least amount of gain at the output ensures that the system doesn't have to amplify the unavoidable noise generated by the input buffers and summing amplifiers. Unity gain for Line Inputs is achieved with the faders positioned at the "0 dB" marking.

HEADPHONE SYSTEM

Depending on the position of the Pan control, a mix of stereo Cue or stereo Master is heard through the headphones. Fully counterclockwise, the *sum* of *selected* Cue Sources is heard. This allows previewing of the full equalized Source signal even while the fader is down. Clockwise rotation increases the amount of Master signal. The Level control adjusts the headphone output.

ACTIVE CROSSFADER

With the ASSIGN switches in the PRE position, Sources 1 and 2 create the A Mix and Sources 3 and 4 create the B Mix. The output of these Mixes are under control of the Active Crossfader. When the appropriate ASSIGN switch is in the POST position, only Source 2 and/or 3 is routed to the Crossfader.

When the Crossfader is in its left-most position, only A Mix (PRE) or Source 2 (POST) appears at the Output. In the center, both Mixes are present in equal levels, and only B Mix (PRE) or Source 3 (POST) is heard once the Crossfader reaches its far right position. With the ASSIGN switches in POST, Source 1 and 4 are still active, unaffected by the Crossfader.

With the CROSSFADER CONTOUR set to the full ccw position, the sound pressure level does not change as this transition progresses. See Figure 1 on page Manual-1 for response curves.

Active Crossfader technology dramatically increases the service life of the crossfader. In the unlikely event of crossfader failure, there is no loss of signal—all Sources are present in equal levels, as if the crossfader was in its center position. Use the Source Input faders to set the audio levels while the Crossfader is out of service. If a Crossfader becomes rough or noisy, it is possible to remove and replace ("hot swap") the control during a performance with no interruption of the audio signal.

MAIN OUTPUT

The output signal from the Crossfader and Source Input faders is routed to the Program Loop jack. The signal can be fed to external effects units and returned to the MM 8z. The Program Loop return signal is combined with the Mic and Aux Input signals and presented to the Master Level fader, Aux Out Level control, headphone amplifier, and peak meter. The Master Level fader should be set at the lowest position while still achieving overall desired sound output level. The least amount of gain in the output stage will avoid amplifying unavoidable noise and provide the cleanest output. For unity gain in the output stage, set the Master Level fader at the "0 dB" marking.

MICROPHONE OPERATION

Connect the microphone to the appropriate connector. The Main Mic Input on the front panel allows a gooseneck-mounted microphone. The connector is rotated such that a right angle connector may be used when connecting via mic cable. Use only one of the front or rear Main Mic Inputs, both are not operable simultaneously. Leave the Master Level fader in roughly the same position as it was for music. Press *in* the Mic On switch, lighting the adjacent indicator, and adjust the Main Mic (or Aux Mic) Level. The tonal balance of the Main and Aux Mic Inputs may be adjusted via the Mic EQ controls. Modifying the sound of the mic in this way won't affect the EQ of the music. When the mic is not in use, release the Mic On switch to its *out* position to disconnect the mic signal. Should the mic preamp become overloaded, the Overload indicator will light. By reducing the appropriate Mic Level control and increasing the Master Level fader, the desired microphone level may be restored without overload distortion.

The TAPE OUTPUT does not contain any signal from the microphone section. If you need to record the mics, use the Aux Out for your recording signal; this output is a composite of the Program, Microphone, and Aux Inputs.

AUXILIARY IN/OUT

The Auxiliary Input is an insertion point for added signals. This Input combines with the program mix and microphone signal to provide a final mixer output. Leave the Master Level fader in roughly the same position as it was for program music. Adjust the Aux In Level control for the desired sound output.

The Aux Out provides a separate final mixer Output, unaffected by the Master Level fader, for external devices such as tape recorders and video cameras, or for additional zone feeds. The Aux Out Level control varies the output signal level.

METERING

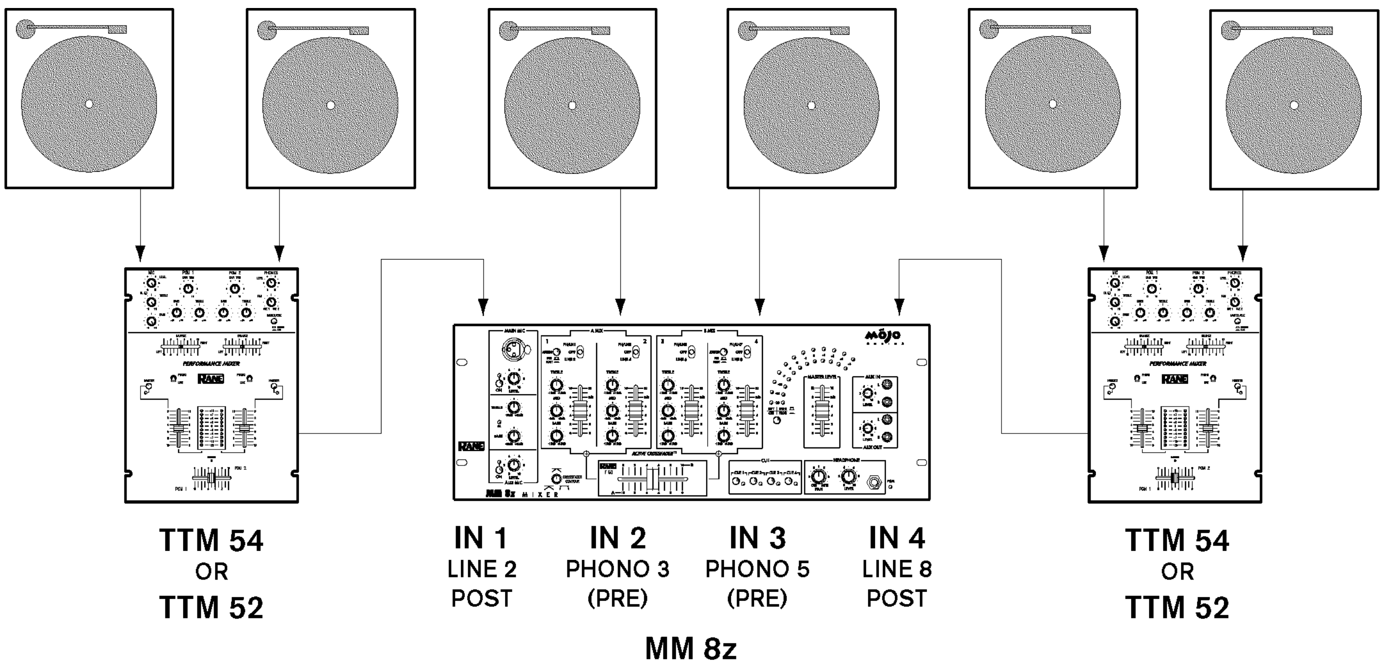
The MM 8z's meter displays signal level in peak dBu. Two display modes are provided, Stereo Master and Mono Cue/Mono Program. The mode is selected with the meter mode switch. With the switch in the *out* position (LEFT/RIGHT), the meter indicates the level of the left and right Main Outputs as measured at the output jacks (what you see is what you hear). The Master signal is the sum of Program, Microphone, and Aux Input signals.

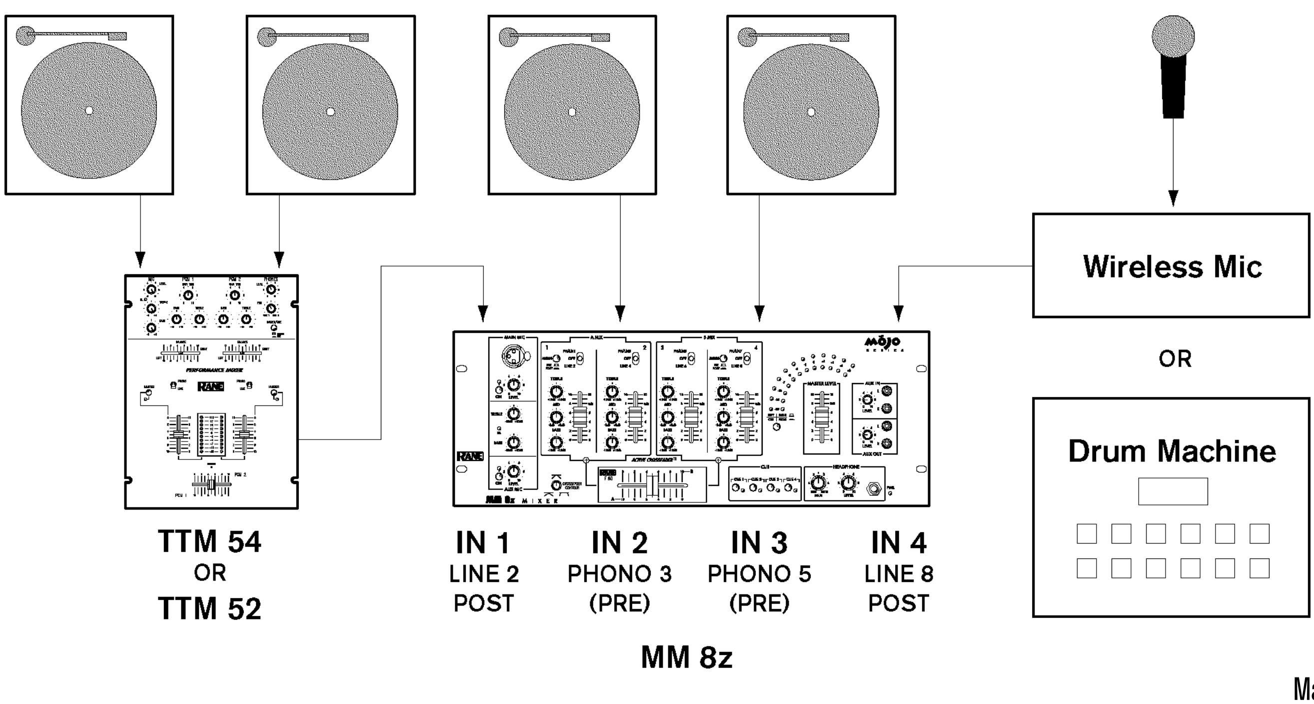
With the switch in the *in* position, the left meter indicates the *sum* of *selected* Mono Cue levels and the right meter indicates Mono Program level. Mono Cue levels are measured at the output of the Source Input fader (just before the Crossfader). The Input fader of a cued Source will need to be advanced to see level indication on the meter. Mono Program level is measured at the output of the Crossfader and Source Input faders (Pre Master summing and Master Level fader). This arrangement allows matching of Source levels and beat prior to Crossfading. Note that if two Sources are to be in an A or B mix, both need to be cued for the meter to reflect the combined levels.

MM 8z APPLICATIONS

Two Pre/Post Crossfader ASSIGN switches add to the flexibility of the MM 8z. By placing these in the POST position, the Crossfader is bypassed, allowing the Source 1 or 4 to be a separate mixer or other device, while still using the MM 8z's Crossfader with Sources 2 and 3. This powerful new feature greatly increases the flexibility of the MM 8z, making it one of the most capable mixers in it's price range.

It is now possible to have a pair of booth turntables on Source 2 and B Mix Source 3, each sent to the Crossfader, while having the output of a 2nd Performance Mixer connected to Source 1 and summed post Crossfader. Source 4 could be summed Pre-crossfader for use with a 3rd booth turntable, or Post-crossfader to allow use of a 3rd mixer, a wireless microphone, a drum machine, etc... you get the idea. The examples shown below illustrate some of the many possibilities. Any of the turntables can be replaced with CD players by switching the Phono Input to Line level.





SOUND SYSTEM INTERCONNECTION

Rane's policy is to accommodate rather than dictate. However, this document contains suggestions for external wiring changes that should ideally only be implemented by trained technical personnel. Safety regulations require that all original grounding means provided from the factory be left intact for safe operation. No guarantee of responsibility for incidental or consequential damages can be provided. *(In other words, don't modify cables, or try your own version of grounding unless you really understand exactly what type of output and input you have to connect.)*

THE ABSOLUTE BEST RIGHT WAY TO DO IT

Use balanced lines and *tie the cable shield to the metal chassis (right where it enters the chassis) at both ends of the cable*.

A balanced line requires three separate conductors, two of which are signal (+ and –) and one shield. The shield serves to guard the sensitive audio lines from interference. Only by using balanced line interconnects can you *guarantee* (yes, *guarantee*) hum-free results. Always use twisted pair cable. Chassis tying the shield at each end also *guarantees* the best possible protection from RFI [radio frequency interference] and other noises [neon signs, lighting dimmers].

THE NEXT BEST RIGHT WAY TO DO IT

The quickest, quietest and most foolproof method to connect balanced and unbalanced is to **transformer isolate all unbalanced connections**. Your audio dealer can recommend such a transformer.

The goal of transformer adaptors is to allow the use of *standard cables.* With these transformer isolation boxes, modification of cable assemblies is unnecessary. Virtually any two pieces of audio equipment can be successfully interfaced without risk of unwanted hum and noise.

Another way to create the necessary isolation is to use a *direct box.* Originally named for its use to convert the high impedance, high level output of an electric guitar to the low impedance, low level input of a recording console, it allowed the player to plug "directly" into the console. Now this term is commonly used to describe any box used to convert unbalanced lines to balanced lines.

THE LAST BEST RIGHT WAY TO DO IT

If transformer isolation is not an option, special cable assemblies are a last resort. The key here is to prevent the shield currents from flowing into a unit whose grounding scheme creates ground loops (hum) in the audio path (i.e., most audio equipment). Do not be tempted to use 3-prong to 2-prong "cheater" adapters to lift grounds. This is a dangerous and illegal practice.

It is true that connecting both ends of the shield is theoretically the best way to interconnect equipment – though this assumes the interconnected equipment is internally grounded properly. Since most equipment is *not* internally grounded properly, connecting both ends of the shield is not often practiced, since doing so can create noisy interconnections.

A common solution to these noisy hum and buzz problems involves disconnecting one end of the shield, even though one can not buy off-the-shelf cables with the shield disconnected at one end. The best end to disconnect is a matter of personal preference and should be religiously obeyed; choose inputs or outputs and always lift the side you choose (our drawings happen to disconnect the outputs). If one end of the shield is disconnected, the noisy hum current stops flowing and away goes the hum — but only at low frequencies. A one-end-only shield connection increases the possibility of high frequency (radio) interference since the shield may act as an antenna. Many reduce this potential RF interference by providing an RF path through a small capacitor (0.1 or 0.01 microfarad ceramic disc) connected from the lifted end of the shield to the chassis. The fact that many modern day installers still follow this one-end-only rule with consistent success indicates this and other acceptable solutions to RF issues exist, though the increasing use of digital and wireless technology greatly increases the possibility of future RF problems.

See the following page for suggested cable assemblies for your particular interconnection needs. Find the appropriate output configuration from either your mixer output or the MX 22 output (down the left side), and then match this with the correct balanced or unbalanced input to the MX 22 or the amplifer (down the right side.) An "off-the-shelf" cable may be available or modifiable. Soldering should only be attempted by those trained in the art.

SUMMARY

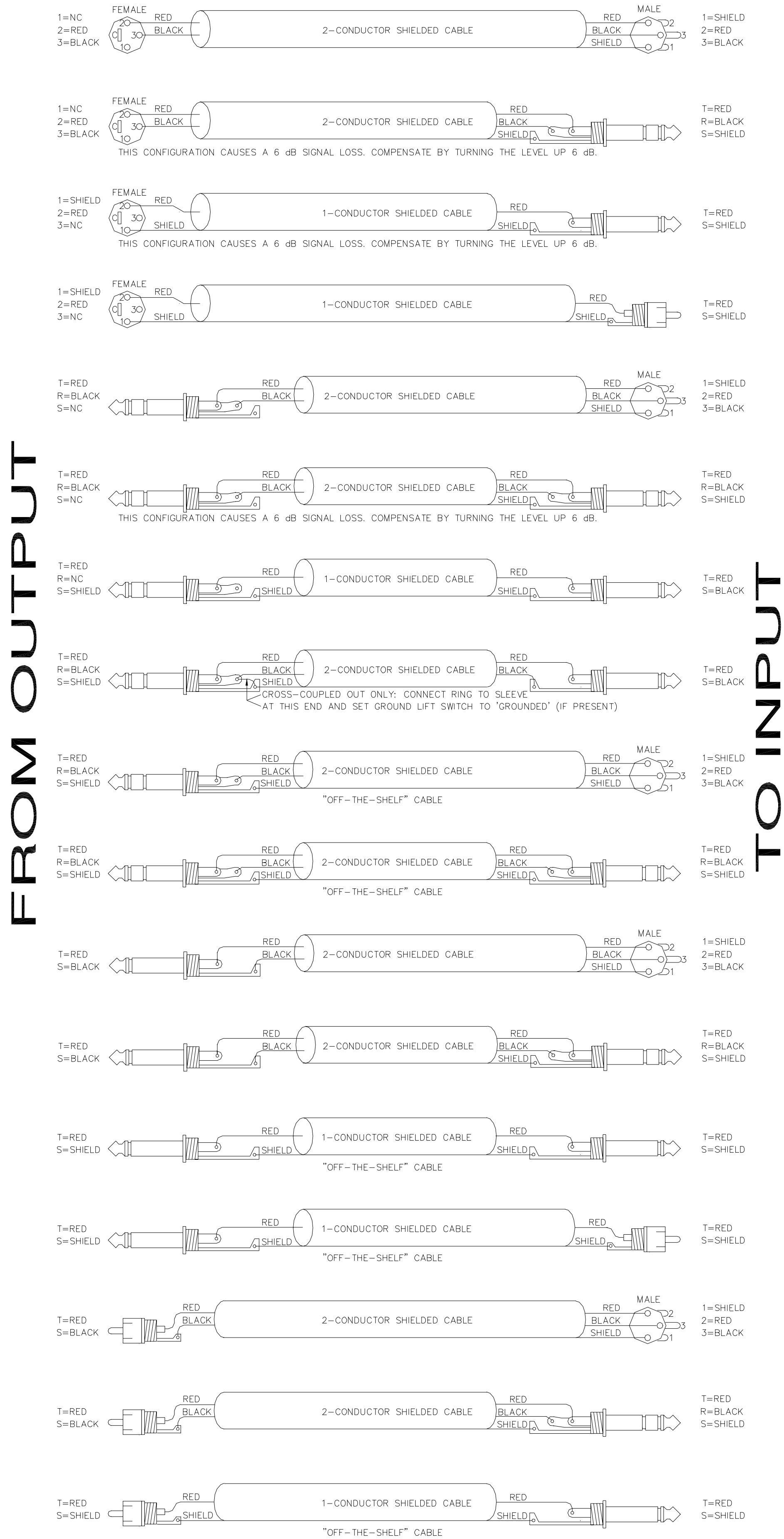
If you are unable to do things correctly (i.e. use fully balanced wiring with shields tied to the *chassis* at the point of entry, or transformer isolate all unbalanced signals from balanced signals) then there is no guarantee that a hum free interconnect can be achieved, nor is there a definite scheme that will assure noise free operation in all configurations.

WINNING THE WIRING WARS

- Use balanced connections whenever possible.
- Transformer isolate all unbalanced connections from balanced connections.
- Use special cable assemblies when unbalanced lines cannot be transformer isolated.
- Any unbalanced cable must be kept under ten feet (three meters) in length. Lengths longer than this will amplify the nasty side effects of unbalanced circuitry's ground loops.

This information was condensed from Rane Note 110, "Sound System Interconnection". If you would like the complete note, call or email the factory, download it from Rane's web site (addresses on page Manual-12), or ask your dealer for a copy.

VARIOUS XLR, RCA & ¼" CABLE ASSEMBLIES



MOJO GLOSSARY

balanced line The recommended method of interconnecting audio equipment. A balanced line requires three conductors: a twisted-pair for the signal (positive and negative) and an overall shield. *The shield must be tied to the chassis at both ends for hum-free interconnect.*

bandwidth *Abbr.* **BW** The numerical difference between the upper and lower -3 dB points of an audio band.

clipping What occurs when a unit tries to produce a signal *larger than its power supply*. The signal takes on a flat-topped, or *clipped* shape. When an amplifier tries to go above its max power, it *clips*.

compressor A signal processing device used to *reduce the dynamic range* of the signal passing through it. For instance, an input dynamic range of 110 dB might pass through a compressor and exit with a new dynamic range of 70 dB. The modern usage for compressors is to turn down (or reduce the dynamic range of) just the loudest signals. Other applications use compressors to control the *creation* of sound. When used in conjunction with microphones and musical instrument pick-ups, compressors help determine the final timbre by selectively compressing specific frequencies and waveforms.

connectors Audio equipment uses different styles:

RCA An *unbalanced* pin connector commonly used on consumer and some pro equipment; aka *phono plug*
XLR A 3-pin connector common on pro audio equipment. Preferred for *balanced line* interconnect; aka *Cannon plug*
¼" TRS 1. *Stereo* ¼" connector consisting of *tip* (T), *ring* (R), and *sleeve* (S) sections, with T = *left*, R = *right*, and S = *ground/shield*. 2. *Balanced* interconnect with the pos & neg signal lines tied to T and R respectively and S acting only as an overall shield. 3. *Insert loop* interconnect with T = *send*, R = *return*, and S = *ground/shield*. [Think: *ring, right, return*]
¼" TS *Mono* ¼" connector consisting of *tip* (T) [signal] and *sleeve* (S) [ground & shield] for *unbalanced* wiring.

constant-Q equalizer (also **constant-bandwidth**) The bandwidth remains constant for all boost/cut levels. Since Q and bandwidth are interrelated, the terms are fully interchangeable.

decibel *Abbr.* **dB** (named after *Alexander Graham Bell*). The preferred method and term for representing the *ratio* of different audio levels. Being a ratio, *decibels have no units*. Everything is relative. So it must be relative to some *0 dB reference point*. A suffix letter is added to distinguish between reference points:

0 dBu A reference point equal to 0.775 V
+4 dBu Standard pro reference level equal to 1.23 V
0 dBV A reference point equal to 1.0 V
-10 dBV Standard reference level for consumer and some pro audio use, equal to 0.316 V. *RCA* (phono) connectors are a good indicator of units operating at -10 dBV

dynamic range The ratio of the loudest signal to the quietest signal in a unit or system as expressed in *decibels* (dB).

expander A signal processing device used to *increase the dynamic range* of the signal passing through it. Expanders complement compressors. For example, a compressed input dynamic range of 70 dB might pass through a expander and exit with a new *expanded* dynamic range of 110 dB. Modern expanders usually operate only *below a set threshold point*, i.e., they operate only on low-level audio. The term *downward expander* describes this type of application.

ground Any electrical reference point for measuring voltage levels. Usually a large conducting body, such as the earth or an electric circuit connected to the earth. Chassis should always be at earth potential. ***WARNING: SHOCK HAZARD***
Never use an AC line cord ground-lift adapter or cut off the 3rd pin. It is illegal and dangerous.

headroom The level in dB between the typical operating level and *clipping*. For example, a nominal +4 dBu system that clips at +20 dBu has 16 dB of *headroom*.

hum Unwanted sound contaminating audio paths due to EMI (electro-magnetic interference) caused by AC power-lines & transformers getting into unbalanced, poorly shielded, or improperly grounded connecting cables. Hum has a definite smooth (sine wave) repetitive sound based on the harmonics of 50/60 Hz such as 100/120 Hz and 150/180 Hz.

interpolating Term meaning to insert between two points. If a graphic equalizer's adjacent bands, when moved together, produce a smooth response without a dip in the center, they are *interpolating* between the fixed center frequencies.

levels Terms used to describe relative audio signal levels:

mic-level Nominal signal coming directly from a microphone. Very low, in the microvolts, and requires a preamp with at least 60 dB gain before using with any *line-level* equipment.
line-level Standard *+4 dBu* or *-10 dBV* audio levels.
instrument-level Nominal signal from musical instruments using electrical pick-ups. Varies widely, from very low *mic-levels* to quite large *line-levels*.

limiter A compressor with a fixed *ratio* of 10:1 or greater. The dynamic action prevents the audio signal from becoming larger than the *threshold* setting.

Linkwitz-Riley crossover The most preferred active crossover design. It features steep 24 dB/octave slopes, in-phase outputs, and flat amplitude response. Due to the in-phase outputs the acoustic lobe resulting when both loudspeakers reproduce the crossover frequency is always on-axis (not tilted up or down) and has no peaking.

noise 1. *Interconnect*. Unwanted sounds contaminating audio paths. RFI (radio frequency interference) caused by broadcast signals leaking into unbalanced, poorly shielded, or improperly grounded connecting cables. Also by light dimmers, motor controls and computers. 2. *Music.* A random mix of audio frequencies not harmonically related, sounding like radio static.

polarity A signal's electromechanical potential with respect to a reference. For example, a microphone has *positive polarity* if a positive pressure on its diaphragm results in a positive output voltage. **polarity vs. phase shift:** *polarity* refers to a signal's *reference* NOT to its *phase shift*. Being 180 degrees *out-of-phase* and having *inverse polarity* are DIFFERENT things. We wrongly say something is *out-of-phase* when we mean it is *inverted.* One occurs over a period of *time*; the other occurs instantaneously.

Q (upper-case) Quality factor. Defined to be the ratio of the center frequency *f* divided by the bandwidth *BW* for a bandpass filter.

signal-to-noise ratio The ratio in dB between a reference level and the noise floor. For example, a signal-to-noise ratio of 90 dB re +4 dBu, means the noise floor is 90 dB below a +4 dBu ref.

unbalanced line An audio interconnect scheme using one wire with an overall shield. The shield must perform two functions: act as the return signal path (*ground*) and to protect the conductor from noise (*shield*). Consequently this method is vulnerable to hum & noise problems.

unity gain A gain setting of one. The level out equals the level in.