HI-LEVEL PM10000 PRODUCTION MIXER - Instruction Manual

The HI-LEVEL PM10000 is an extremely compact and easy to handle modular 19" rack mountable mixing console which incorporates all the necessary facilities for public address and discotheques. The PM10000 is the answer to all those situations where a versatile high quality mixer is needed.

To become familiar with all the facilities of the PM10000 we suggest you to read this manual very carefully. It will give you important information about the operation, installation and service of this high quality product.

INTRODUCTION

A choice of two modules is available: A mono Mic/Line module and a stereo Line/Line (Phono)/Mic module. The 19" rack frame can accept a maximum of 10 input modules.

2.0 MONO INPUT CHANNEL

The PM10000 channels are either mono or stereo. The front panel controls only differ in a few functions. We shall describe the mono and stereo channels separately.

2.1 MIC/LINE INPUT

The channel can operate in either the microphone or line input mode. The microphone input is an electronically balanced design. The input impedance is greater than 2 kOhms which will not cause any loading effects (signal loss) on today's studio microphones. The line level input has an input impedance greater than 10kOhms which is high enough to interface with all available peripheral equipment, keyboards and CD players.

2.2 MIC/LINE SWITCH

The line input is selected when the LINE button is depressed. The line input is now connected to the balanced line input jack. In the up-position of this switch the balanced microphone amplifier is connected to the XLR input on the back of this module.

2.3 MIC/LINE GAIN

The microphone input can be varied between -12 dBu and -65 dBu of gain with a good overload margin. The line input gain (the same control but with selected line input) can be varied between -40dBu and +0dBu.

2.4 EQUALIZER

The equalizer of the PM10000 is a very musical and versatile section with three controls to span the entire audio spectrum.

2.5 HIGH

16dB of boost or cut is available at 10kHz with a shelving curve, which means that when the desired amount of boost or cut is achieved all frequencies from 10K and above are boosted or cut equally.

2.6 MID

This control has a range of ± 16dB with a bell curve. Having reached its maximum/minimum at its centre frequency (1kHz) the amplitude response returns to zero on either side of that frequency. A plot from that response shows a bell shape. The bandwidth of that bell curve is fixed.

2.7 LOW

The low control has a shelving characteristic with ± 16dB control from 80Hz down.

3.0 AUXILIARY SECTION

There are two AUX send controls on each module. The second AUX control is on a concentric knob on top of the already fitted control.

AUX SENDS

The standard auxiliary sends are set pre and post-fader. AUX 1 is post fader and AUX 2 is pre fader. The AUX sends are designed to be post equalizer and post insert point.

3.1 TALKOVER

The Talkover switch sends a post-fader signal to the control circuitry located in the master section to attenuate music signals in a controlled amount for compact presentation.

4.0 THE PANPOT

This control (with a 4.5 dB loss at its centre point) pans the signal between the left and right master buss.

5.0 CUE/ON

The Cue switch (pre fader listen) enables you to listen to the channel signal before the fader without effecting the normal signal path to the master outputs. The ON switch activates the input module when depressed. The ON switch also closes contacts at the start jack at the back of the console.

6.0 CHANNEL FADER

The channel fader has a slide length of 100mm and is manufactured to give an exceptionally smooth feel in operation.

7.0 CHANNEL IN/OUTPUTS

Located on the back of the console.

7.1 MICROPHONE INPUT

This is the balanced XLR input for condenser or dynamic microphones. Pin 1 is ground, pin 2 is hot (in phase) and pin 3 is cold (out of phase).

7.2 LINE INPUT

This is a 1/4" tip ring sleeve jack which is balanced. The tip is hot, ring is cold and the sleeve is wired to ground. This input has a sensitivity of -40dBu to 0 dBu. The impedance is 10kOhms. It will accept any Line output of tape recorders, keyboards, CD players and so on.

7.3 INSERT

This is the channel insert (placed ahead of the channel fader). The tip is the input and the ring of the stereo jack is the output, both at a level of 0dBu.

7.4 START CONNECTOR

On the back of the console is a stereo jack labelled START to control start function of mono signal sources such as jingle machines. It can also be used as a red light indicator for broadcast purposes.

8.0 STEREO INPUT CHANNEL

Basically there is not much difference in use between the mono and the triple input stereo channel in controlling the sound. The Line B input amp can be converted into a stereo phono input with R.I.A.A. correction amp (optional). All the other electronics in the channel are doubled to achieve a full stereo output with a minimum of crosstalk. The panpot has been changed into a balance control to justify incorrect balanced signals. This module is capable of mixing stereo line signals, stereo phono signals and balanced mic signals, all in one module.

8.1 LINE A/B SWITCH

The Line A/B switch selects which input jack is activated. The optional R.I.A.A. correction amp can be placed on the pcb very easily by taking out the J2 jumpers and putting the small board up side down on the connectors J2 and J4.
By doing this the Line B input becomes an R.I.A.A. corrected phono input. The sensitivity and frequency response is now changed to amplify phono cartridge s correctly. The Line A/B switch is now marked by a red push button. In the up position of the Line A/B switch the triple input module can accept the signal from Line A input. In the down position the Line B jack can now accept phono cartridges. The tip is wired to the left channel amps and the ring to the right channel amps. The phones amp has a built in correction amplifier which follows the standard R.I.A.A. frequency curve within 0.5 dB. Connect the turntable ground wire to the large ground terminal of your PM10000.

8.2 GAIN

The gain control of the stereo channel adjusts the incoming level of all connected sources to the line A/B and Mic XLR connectors.

8.3 EQUALIZER

The equalizer of the stereo channel has the same characteristics as the one designed in the mono channels. It consists of two shelving controls for high and low frequencies and a bell curve control for the mid frequencies. (see description in mono channel)

8.4 AUX SENDS

The AUX sends are factory set pre and post fader. The AUX 1 is post-fader is mono and receives a summed left right signal. AUX 2 is mono and wired pre fader and receives a summed stereo signal.

8.5 CROSSFADER ASSIGN SWITCHES A/B

The two switches labelled A and B are routing the output of the stereo module to the left(A) or Right (B) side of the crossfader. In this way it is possible to crossfade between any module that is assigned to the crossfader. It is possible to have multiple modules assigned to the crossfader section at the same time.

8.6 BALANCE CONTROL

This control is able to correct incorrect stereo balanced signals. It can even cancel one signal either left or right to create special effects.

8.7 ON SWITCH

The ON switch naturally switches the channel's audio ON but it can also serve start functions of equipment.
There are several possibilities to start equipment connected to the start jack.
1. The ON switch closes contacts
2. The Cue switch can start equipment
3. A fader start can do the job
4. The ON switch in series with the fader
5. The ON switch in series with the fader and the Cue switch in parallel

It is up to you how you want this function to be active in your set up. In the technical section we will explain how to achieve all these individual possibilities.

8.8 CUE SWITCH

The Cue switch brings the stereo signal pre-fader to the phones section for setting up channel gain levels and checking out signal sources. The Cue switch is also able to start equipment when cueing only.

8.9 FADER

The stereo channel fader has a length of 100mm and is manufactured to give an exceptionally smooth feel in operation. A start switch is optional and can be fitted when ordered in advance.

9.0 STEREO CHANNEL INPUTS

These are located at the back of the console and are accessible through jack sockets and an XLR connector.

9.1 LINE A

This is a stereo input for line level input signals with a sensitivity between -20dBu and +20dBu.

9.2 LINE B

The line B input is a stereo jack socket accepting a jack stereo plug. Tip is left, ring is right and sleeve is ground. If a mono signal has to be connected to the stereo channel it is necessary to short circuit both tip and ring with each other. Note: If a mono jack plug is inserted only the left channel will function. The right channel path input is shorted to ground.
The B input can also be converted to a Phono input as described before. The tip will be the left input and the ring will be the right input.

9.3 START

The start stereo jack connector is for connecting start functions together with external equipment. If a fader start switch is fitted the tip of the stereo jack socket will short to the ring when the fader is up or the ON switch is activated or the Cue switch.

10.0 MASTER MODULE

The master module of the PM10000 contains all the electronics for the summing of the left/right signals, the AUX signals, the crossfader signals, the Cue signals, the Talkover signals, and the control room monitoring as well as the power supply. The width of this module is four times the width of the channel module.

10.1 VU-METER

The VU-Meter is a peak reading instrument with attack and release times confirming world standards. The 0dB led lights when the output reaches +10dBu. This also conforms to studio standards. The peak reading VU-Meter is calibrated -6dB down from the output of the console. The VU-Meter will indicate all levels heard in the PHONES signal path. If a different calibration is requested, two small holes below the VU-Meter allow for adjustment of the ledbar sensitivity.

10.2 AUX SENDS

The AUX sends are the master controls of the outgoing level to the AUX outputs. Nominal level is +4dBu ground compensated balanced at 100 ohm output impedance. There are two CUE switches to check the AUX outputs post fader.

10.3 AUX RETURN

The AUX return is a stereo line input for the program mixing amps, the Talkover does not have any influence on these because the AUX return is intended to return effects coming from mics from the mono channels that control the Talkover circuitry.

All signals are fed to the program mix amp and adjusted in leve l by the AUX return control. There is a CUE switch to listen pre fader AUX return.

10.4 TALKOVER SECTION

The Talkover section controls the level dynamically of the signals coming from the stereo channels. Any Mic/Line channel where a Talkover switch is activated is able to lower the program level (which are the stereo signals coming from the stereo modules directly or through the crossfader section), in a controlled way. This feature is used when announcing needs to be done without having to control all sorts of levels at the same time. It also gives a very compact way of combining music with speech. The amount of attenuation of the music signal can be controlled by the ATT control. (A maximum of 20dB is possible). The Talkover switch activates the circuit. The REL control below the ATT control adjust how fast the music returns to its original level after the control voltage from the mic/line modules has stopped. In large buildings with back ground music and regularly announcements it is wise to set the REL(ease) control halfway or even further to avoid having music between each word or line spoken.

10.5 LIGHT OUTPUT

The light output of the PM10000 is a transformer isolated output to control light equipment. There is a source selector for choosing between Program (which is all that is send to the master) or Music only. A level pot controls the output.

10.6 MASTER OUTPUT

The PM10000 has three individually adjustable main outputs apart from the Light, AUX, Subbass and Phones outputs.
The MASTER output is intended to be the main output of the console. It is a ground compensated balanced XLR +4dBu output, which can be switched to mono.
The MASTER output will give all signals send to the output through direct routing from the channels pan-pots and balance controls, as well as all signals coming from the crossfader section. Both left and right outputs have inserts for limiters or graphic equalizers.

10.7 MONITOR OUTPUT

The MONITOR output is a parallel output of the MASTER section and has the same signal sourcing. The signal is taken pre inserts and fed to the MONITOR stereo fader.
The MONITOR FOLLOWS PHONES switch is designed to give the same functions as the is in the PHONES section. Some DJ's prefer not to work with headphones and this is an ideal way of mixing through external loudspeakers (in a closed DJ booth or radio station). The output of the MONITOR section is unbalanced on one XLR type connector with a level of +4dBu.

10.8 ZONE OUTPUT

The ZONE output is a +4dBu unbalanced output on a jack connector. The ZONE level can be controlled and it is also possible to control the content of music and microphone signal amount with a rotary knob.

10.9 PHONES

The Phones section has many useful possibilities which we will explain in detail. The input source can be either the cue signals from the channels or the MASTER or 2 outputs. A Cue led indicates that anywhere in the console a Cue switch is activated which will in turn disconnect the Phones section from the main output signal and redirect it to the selected Cue switch. All stereo signals will be heard in stereo. The SPLIT switch directs the program signal (everything that is coming from the channels) to the left and the summed stereo cue signal to the right. This circuitry makes synching of two input signals an easy task. The program signal can be mixed into the Cue signal (if necessary), so the main signal will not disappear completely when using the autocue function.
Caution: the Phones amplifier is capable of driving headphones extremely loud and can handle headphones as low as 8 ohms.

10.10 CROSSFADER

The linear crossfader is fed by the assign switches in the stereo channels and makes smooth transitions possible between any of the assigned input channels. The crossfader can easily be replaced or disconnected by lifting the Master module out of the 19” frame.

10.11 CUT SWITCHES

The CUT switches control Soft Fet switches which silently mute one or two of the crossfader outputs. This feature creates intermittent rhythmic effects needed in todays live DJ performances.

10.12 POWER SUPPLY

The power supply, which is located in the master section, is a highly regulated circuit with a torroidal transformer to minimise hum pickup.
The phantom power supply switch sets +48 volt on every microphone input, when the individual jumper setting on the mic/line channels is set to receive the phantom voltage.

11.0 MASTER IN/OUTPUTS

Located on the back of the master module the master in/outs are 6.3mm jack sockets and XLR type connectors.

11.1 MASTER OUTPUT

These outputs are ground compensated balanced which means that they behave like a balanced output but without the doubling of output power related with servo balanced output amps. The XLR type connector wiring is conform international standards, pin1 is ground, pin2 is hot, pin3 is cold.

11.2 ZONE OUTPUT

This output is an unbalanced output which is wired as follows pin1 is ground, pin2 is left signal, and pin 3 is the right output signal.

11.3 SUBBASS OUTPUT

The Subbass output of the PM10000 is a +4dBu ground compensated balanced output which has a frequency roll off above 80Hz by a gentle 6dB/octave slope avoiding phasing errors. The subbass output is directly fed from the MASTER output. The MASTER output controls the level of the subbass output at the same time.

11.4 LIGHT OUTPUT

The transformer balanced light output has a level of +4dBu and is wired on a stereo jack plug with the transformer winding between the tip and ring. It is important not to wire the ground of this jack to the light equipment. By only wiring the tip and ring to the light equipment a separation of ground is guaranteed, which is important for your safety.

11.5 TAPE OUTPUT

The tape outputs are derived just ahead of the master faders and the inserts so the signal is unaffected by adjustments of the master faders. The tape output level is -10dBv with an impedance of 100 ohms. This output can also be used to drive any other equipment necessary if there is a need for an extra output.

11.6 AUX RETURN

The AUX return is a stereo jack socket. The tip goes to the left program buss and the ring to the right program buss. A mono signal source has to feed both tip and ring simultaneously.
Warning: If a mono jack is used in the AUX return jack, the right input will be shorted to ground and there will be no signal in this return.

11.7 AUX SENDS

The AUX sends are ground compensated output jacks wired as balanced outputs. The tip is hot, the ring is the ground compensation, the sleeve is ground.

11.8 INSERTS

The inserts are send and return signals on one jack. The ring is the send at 0dBu level at 100 Ohms. The tip is the return with an input impedance of 10 kOhms which will slightly vary with the master fader settings.

12.0 POWER/FUSE

The power supply is primarily fused between the power supply cable and the power supply transformer. For 230 Volt use the value is slow 3.15 Amp. For 115 Volt use the value is slow 6.30 Amp.

13.0 OPERATION

The PM10000 is designed to be the perfect answer to all stereo output mixing situations. The console can accept a maximum of 10 input modules these can be mono and/or stereo modules.

13.1 STANDARD CONTROL SETTINGS

Before you switch on the PM10000 check whether you have a 115 Volt version or a 230 Volt version. This has to match with your local voltage. Before you apply voltage to the PM10000 put the switches and controls in the following settings:

Channels

Input/CUE switches	Up/down dependent upon connected signal sources.
Gain controls	Fully counter clockwise.
Equalizers	12 o'clock position.
AUX sends	Fully counter clockwise.
Panpots/balance controls	12 o clock position
Faders	Fully down.

Master section

All controls	Fully counter clockwise.
Faders	Fully down.

13.2 CONNECTIONS

Before you apply power to the PM10000 you have to wire up your system first. To be of help with this job we will summarise all type of connectors with their associated wiring. Be very careful with this wiring procedure. Use professional soldering equipment to achieve professional results. The quality of the solder points and their isolation is tremendously important for the reliability of the whole system.

MONO INPUT CHANNEL

XLR inputs	level : -65 dBu to -0dBu. pin 1 : signal ground (shield) pin 2 : signal high (in phase +) pin 3 : signal low (out of phase-)
Line inputs	level : -20 dBu to +20dBu. tip : signal high (in phase +) ring : signal low (out of phase -) sleeve : signal ground.
Inserts (only mono mic channels)	level : 0dBu (0.775V) tip : return signal. ring : send signal sleeve : ground.

STEREO INPUT CHANNEL

XLR inputs level : -65 dBu to -0dBu.
pin 1 : signal ground (shield)
pin 2 : signal high (in phase +)
pin 3 : signal low (out of phase-)

Line B inputs with R.I.A.A.	Phono inputs level : 2-5 mV. tip : left. ring : right. sleeve : ground.
Stereo line inputs	level : -20 dBu to +20dBu tip : left. ring : right. sleeve : ground.
START connector	level 24V 500mA max. Shorts between tip and ring when activated
MASTER outputs	level + 4dBu (1.22V). XLR connectors. pin 1 : signal ground (shield) pin 2 : signal high (in phase +) pin 3 : signal low (out of phase-)
MONITOR LEFT/RIGHT	Level +4dBu (1.22V) XLR connectors. pin 1 : signal ground (shield) pin 2 : Left pin 3 : Right
SUBBASS OUTPUT	jack +4dBu at 100 Ohms tip: in phase ring: ground compensation sleeve: ground
LIGHT OUTPUT	Jack +4dBu at 100 Ohms tip: in phase ring: out of phase Sleeve: ground
ZONE OUTPUT	Jack +4dBu at 100 Ohms tip left. ring : right. sleeve : ground.
INSERTS	level : 0dBu (0.775V) tip : return signal. ring : send signal. sleeve : ground.
TAPE OUTPUTS	level -10dBv (300 mV). tip : left output signal. ring : right output signal sleeve : ground.

AUX RETURN	level : 0dBu-(775 mV) tip : left input. ring : right input. sleeve : ground.
AUX SENDS	Jack connector level : + 4dBu (1.22V). tip: in phase. ring: ground compensation sleeve: ground
PHONES	Stereo jack, level : + 4du (1.22 V) tip: left output ring: right output sleeve: ground

14. INSTALLATION

Applying power. Before switching on the power supply of the PM10000, check the main voltage of the supply by looking at the sticker on the back of the console, and see whether the power cord matches the ones you are used to work with. This also indicates that your PM10000 has been wired for correct voltage.
This should be 115 Volt for area's with voltage from 100 Volt to 120 Volt and 230 Volt for area's with voltages between 220 and 240 volts. Main voltages: The main fuse should be 3.15 Amp. slow blow for 230 Volt, and 6.3 Amp 20 mm. slow blow for 115 Volts.

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

14.1 INTERFACE LEVELS

The PM10000 is prepared for interfacing with almost all available equipment. One point of attention has to be made concerning the output. The outputs deliver a nominal +4dBu level, which is sometimes too high for power amps rated at 300mV sensitivity for full output. In those cases you should install an input attenuator at the power amps input to reduce this + 4dBu level by approximately 12dB. Use a 2K2 series resistor and a 680 Ohm shunt resistor across the amplifier inputs.

14.2 GENERAL WIRING PROCEDURES

To take full advantage of the excellent signal to noise ratio of the PM10000 it is necessary to carefully read this part of the manual.
Hum, radio frequency interference buzzes and instability are often caused by improper wiring and inferior grounding systems! Sometimes the incoming mains ground is not adequate for studio and a separate technical ground has to be made for all the audio equipment.
Your electricity supply company will give you all the details to avoid insufficient safety regulations.
There are some ground rules to be followed.
All signals in a studio are referenced to ground. This ground has to be clean and free of noise. *ONE* central point should be decided for the main ground point system and all grounds should be started from this point.

The way your electricity company has daisy chained the ground in your situation is unsuitable for your studio. The best way is to run a separate ground wire from each outlet to the system starpoint ground. This the safety ground earth and screen reference for all your equipment. A separate wire from all the equipment racks to the starpoint is nice to have in cases where the ground via main plugs is not satisfying.
The starpoint should be located at the rear of the console or equipment rack. All equipment has to be located as far as possible from the incoming mains distribution boxes. Unbalanced equipment may need to be isolated from the rack to avoid ground loops.

14.3 SETTING UP THE INITIAL WIRING

First connect the power supply of the PM10000 to the console. All faders must be down and the Phones control at 12 o'clock with a headphone set connected.

Connect the power amps or headphones to the Phones outputs and check for any hum buzz or interference. If this is already proceeded, now the inputs can be wired up. Check every channel that is wired to your equipment individually. Carefully listen for noise and/or hum. Connect stereo tape recorders microphones and all other equipment one by one and all signal processors one at the time and keep checking that your system stays clean. If not carefully check for a ground loop.

14.4 SHIELDING/EARTHING OF AUDlO EQUIPMENT

The shield of any audio connection should be connected at one end only. If not, ground loops and high frequency crosstalk will be the result.
Connect the shield as a general rule to the signal source end. In high R.F. area's it is wise to ground the other end of shield via a 0.01 uf capacitor. This will be a short circuit at high frequencies but not at low frequencies.

Typical shielding situations:

Output	Input	Shield at
Unbalanced	Unbalanced	Source
Unbalanced	Balanced	Source
Unbalanced	Differential	Source
Balanced	Unbalanced	Destination
Balanced	Balanced	Source
Balanced	Differential	Destination

It is essential to study the signal flow chart carefully. This will help to isolate problems in the PM10000. If you follow the signal from input to output jacks it is possible to locate a fault. If a fault is located, inform your dealer or us and we will assist you by phone. If this will not help return the channel or master to your dealer, or the factory and we will be happy to repair it within short time. Many faults can be found by logical thinking and replacing integrated circuits, which is very easy. They are all on sockets.

15.0 WORKING WITH THE PM10000

After you have wired up the PM10000 properly as described in earlier, it is time to switch the unit on. All the control settings are as described under the heading control settings. The ledbar will light up partly and fall down slowly leaving the ON led on. Now your PM10000 is ready to operate.

Push down the CUE button and adjust the gain control until the VU-Meter are reaching the zero dB position. Do this for every channel where a signal is connected . Now that all basic adjustments are made you can mix all the signals together.

Bring up the channel faders and the master fader with the right amount of level needed for the perfect mix. You can now adjust the equalization until the right coloration of the sound has been made. Please note that levels can increase if you boost the equalizers because of this, it may be necessary to go back and push CUE on each input and check the input gain as described before.

Optimum level in the channel is around 0db, this means a headroom of more than 22dB and a signal to noise ratio of more than 78 dBr can be achieved . If levels are too high in the channels, you are giving up headroom and improving signal to noise ratio. This is a trade off . On the other hand, too low of a level in the channel will increase the headroom and decrease the signal to noise ratio. When all your levels are set correctly you will maintain the excellent signal to noise ratio the PM10000 offers. This is the most important thing in producing a clean, clear and professional sound.

15.1 AUX SENDS AND RETURNS

The master AUX send controls has to be turned fully clockwise and the AUX return turned clockwise to a desired level. Now turn the AUX send controls on the individual channels until you hear the effect level you require. You must watch the input and output level of the connected processing equipment as well.

15.2 PAN-POT/BALANCE

These controls let you set the position of the signal in the stereo image. The pan pot has an attenuation of -4.5 dB in the middle to achieve a good panning range between left and right. To make a proper level set-up, set the pan pot fully left or right, only then you can check the 0dB positions on the channel and master faders. On the stereo channel the centre position on the balance control is the calibration position.

15.3 VU-METER

In order to achieve optimum results we choose a peak-reading VU-Meter design. This means the meter is calibrated 6dB down from the measured output . The 6db down calibration is an international standard and is a good compromise between peak and average levels. If any other setting is required we have designed the calibration trimmers close to the front panel so you can adjust them yourself.

15.4 PHONES

The headphone output is a stereo jack (tip-left ring-right) which is capable of driving 8-600 ohm headphones. The output normally gives the stereo master signal but as soon as a CUE switch is activated this master signal is automatically switched to this activated channel. This signal can be stereo or mono depending upon the chosen channel (mic/line or stereo/line).

15.5 OUTPUTS

Apart from the normal MASTER/MONITOR/ZONE/Subbass and light outputs with their +4dBu level there are also tape outputs. These tape outputs are not effected by the master faders and have a level of -10dBV and are in phase with the main outputs.

16.0 REMOVING A MODULE

Switch off the power supply first. Remove the metal strip on top of the modules by simply taking them from the front panels. They are hold down by magnetic tape. Now remove the two module retaining screws which will allow to carefully withdraw the module from the console. First move the module towards you to take the jack connectors out their mounting holes and remove the flat cable connector and XLR wiring by unplugging its 3 pin connector. Now the module can be lifted out of the chassis. The same applies for the master module although there are more screws and more flat cable connectors.

17.0 PRODUCT SAFETY

The product you just unpacked is manufactured with safety in mind and it is double checked in the quality control department for reliability in its high voltage section.

CAUTION

* Never open your equipment yourself there are no users serviceable parts inside.
* Opening the unit should be done only by a trained and qualified service technician, who is fully aware that it can be dangerous to service a mains powered unit.
* Always GROUND the unit.
* Only make use of the product in a way as is described in the manufacturers brochures and manuals, never use it for other purposes than intended by the manufacturer.
* Never use this equipment in an environment with high humidity or expose it to water.
* Do not use this equipment in the rain, snow, or equivalent type of weather.
* Check your mains cord regularly and see if it is in safe condition with properly connected mains plugs on one side and securely tightened in the equipment on the other side.
* Return your product yearly to your dealer to give it a safety check.
* The hazard of an electrical shock can be avoided by careful ly following the above mentioned rules.

PLEASE CAREFULLY READ THE FOLLOWING INFORMATION

Especially in sound equipment on stage the following information is essential to know. An electrical shock is caused by voltage and current, actually it is the current that causes the shock. In practice the higher the voltage the higher the current will be and the higher the shock. But there is another thing to consider and it is resistance. When the resistance (in Ohms) is high between two poles the current will be low and vice versa.
All three of these; voltage current and resistance are important in determining the effect of an electrical shock. However the severity of a shock is primarily determined by the amount of current flowing through a person. A person can feel a shock because the muscles in a body respond to electrical current and because the heart is a muscle, it can be affected, when the current is high enough.
Current can also be fatal when it causes the chest muscles to contract and cause you to stop breathing.
At what potential is current dangerous? Well the first feeling of current is a tingle at 0.001 amp of

current. The current between 0.1 and 0.2 amp is fatal. Imagine that your home fuses of 20 amp can handle 200 times more current than is necessary to kill.
How does resistance effect the shock a person feels? A typical resistance between one hand to the other in a "dry" condition could be well be over 100.000 ohm. If you are playing on stage your body is perspiring profusely and your body's resistance is lowered by more than 50% ! This is a situation in which current can easily flow. Current will flow when there is a difference in ground potential between the housing of the mics and the guitar amps which will be linked by your body on stage.
Imagine a guitar in your hand and your lips close to the mic! A ground potential difference of above 10 Volts is not unusual. In improperly wired buildings it can possibly be as high as 240 Volts. Although removing the ground wire sometimes cures a system hum, it will create a very hazardous situation for the performing musician.

ALWAYS GROUND all your equipment by the grounding pin in your mains plug. Hum loops should only be cured by proper wiring and isolation input/output transformers. Replace fuses always with the same type and rating after the equipment has been turned off and unplugged. If the fuse blows again you have an equipment failure do not use it again and return it to your dealer for repair.

DO NOT TOUCH a person being SHOCKED because you could also be shocked! Once removed from the shock have someone send for medical help immediately.

ALWAYS KEEP THE ABOVE MENTIONED INFORMATION IN MIND WHEN USING ELECTRICALLY POWERED EQUIPMENT.

SUMMARY

In this manual we have tried to give you an oversight of all the possibilities the HI-LEVEL PM10000 offers you. If there are any questions left do not hesitate to contact us or your dealer.

We wish you many years of enjoyable music.

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