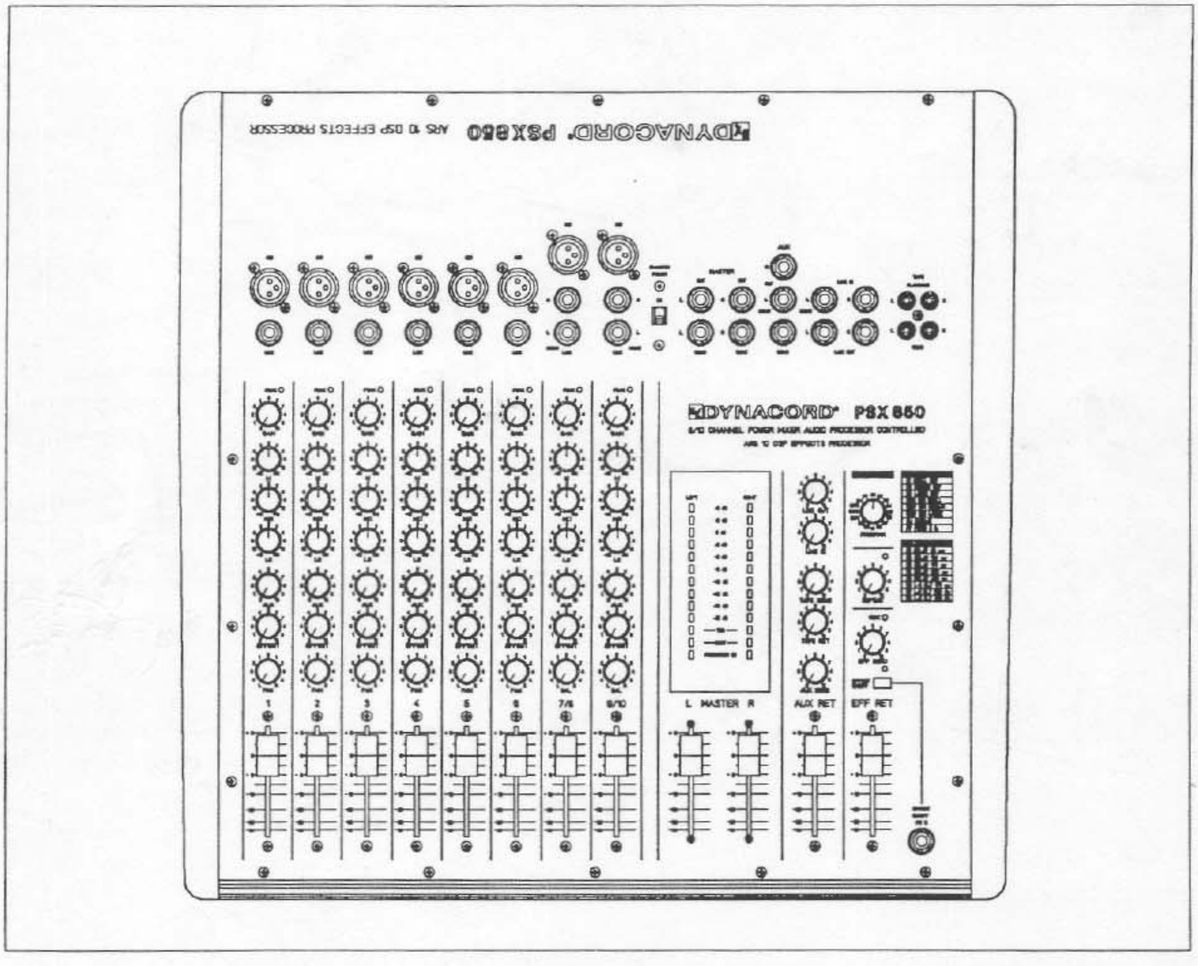
DYNACORD®

USER MANUAL



PSX 850

8/10 CHANNEL POWER MIXER

PSX 850 - GENERAL OVERVIEW

- 8 Input channels = 6 x Mic/Line + 2x Mic / 2x Line Stereo
- 3 band equalizer in each input channel
- Digital 16 bit stereo effect unit with 6 reverb, 8 delay programs and 2 special effects
- 48 V Phantom Power
- 1 effect path
- 1 AUX/Monitor path
- Controllable Line In and Line Out path
- Controllable Tape Record and Tape Playback path
- Big 2 way LED Level Meter
- 2 x 250 Watts (RMS/4 Ohms) PCA processor power amplifiers

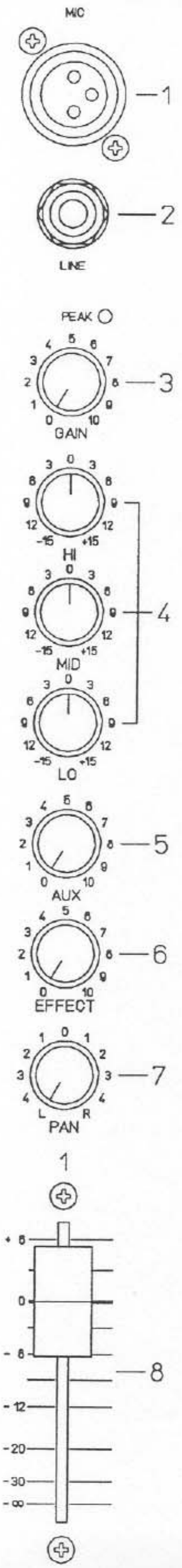
The mixer of the PSX compact power mixers is equipped with a wide range of features. The stereo input channels allow the connection of stereo program sources like keyboards, drum machines, tape recorders and additional mixing desks. Thus a lot more sound sources can be connected than with conventional power mixers. The stereo input channels can also be used as regular mic input channels. All mic inputs are electronically balanced and equipped with XLR sockets. 48 volt Phantom power can be switched to the mic sockets.

The effect section is equipped with a digital 16 bit stereo reverb/delay module. The effect unit generates different, extremely natural sounding stereo reverb programs, special stereo programs which combine the reverb programs with additional echoes, and a stereo chorus program especially structured for vocals, brass and woodwinds. In addition 8 different delay and echo programs with excellent quality are available from the ARS 10 DSP effects board. The number of echo repeats is front-panel controllable.

The master section is equipped very comprehensively with separate control functions, e.g. for tape playback and record, connection for an additional external stereo effect unit, separately controllable AUX/LINE output/input and big LED level meter for the power amplifiers.

The PCA processor power amplifiers are designed in advanced Hi-Power MOS technology. The frequency and phase response of the loudspeaker cabinets connected is linearized in the low-frequency region with a 2nd order shelving eq. The corner frequencies of the 2nd order shelving eq have been designed to match the characteristics of modern high-efficiency loudspeaker cabinets. A built-in fast acting limiter prevents excessive overdrive.

The power outputs of the power amps are equipped with speakon adapters. These connectors were developed especially for the peak performance of modern power amps and guarantee a safe and loss- free connection of loudspeaker cables with the greatest possible cable cross-sectional areas.

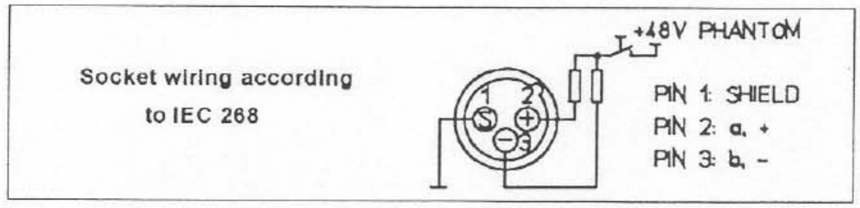


INPUT/MONO

1. Mic

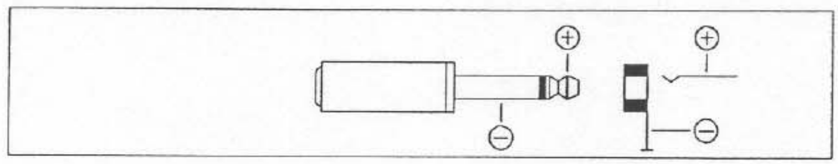
Electronically balanced XLR inputs 1-6 for connection of low- impedance microphones. Also phantom-powered microphones (condensor microphones) can be connected to these sockets.

For further information see: **38. Phantom Power.**



2. LINE

Unbalanced inputs 1-6 for instruments and other high level signal sources. The insertion of a plug into this socket will switch off the XLR input.



Please make sure that the respective channel fader or at least both master faders are closed before connecting signal sources, to protect your audience from annoying click noise.

3. GAIN + PEAK LED

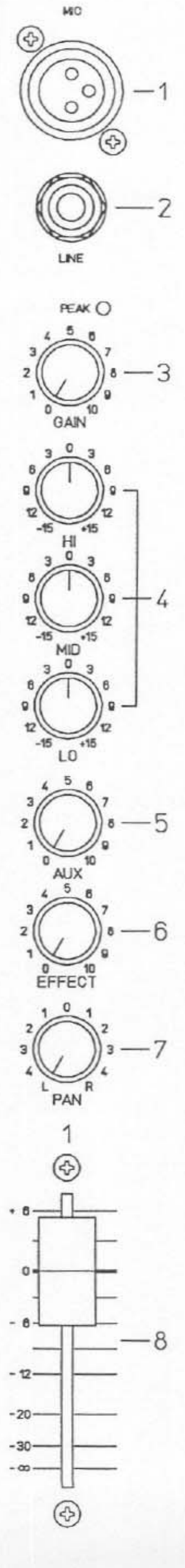
Control for adjusting the input sensitivity between -56 dBV (1.5 mV) and -20 dBV (95 mV).

This control should be adjusted so that the PEAK LED lights up only briefly at level peaks. This will result in an optimum S/N ratio. If the PEAK LED lights up, the available headroom is 10 dB before the input signal is distorted audibly. Please note that the sound corrections with the EQ controls influence the input signal level as well. Check the correct setting of the GAIN control again after sound adjustment.

4. EQUALIZING

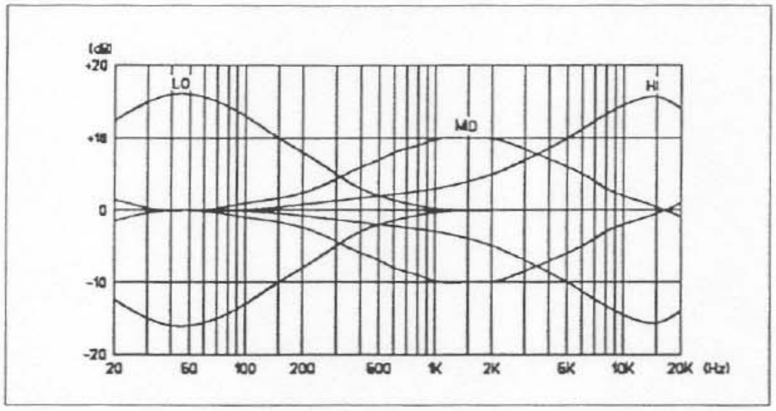
The EQ section allows for a very comprehensive and effective control of the input signal for the different frequency regions. Turning the control to the right increases the respective frequency region.

When adjusting you should always start in a neutral position, i.e. all controls are in central position. If possible, do not use extreme control positions; normally a small sound correction is enough and delivers the best sound quality. Take as orientation the naturalness of the reproduction and depend on your musically trained ear for sound checking. You can effectively avoid acoustic feedback by operating the mid control (MID) gently. Avoid too much gain especially in this frequency region; a light lowering allows an amplification of microphone signals with little danger of feedback.



4. **EQUALIZING** (continuation of page 4)

Control	Variation	Frequency	Type
HI	+/-15 dB	15 kHz	shelving
MID	+/-10 dB	1.5 kHz	boost/cut
LO	+/-15 dB	50 Hz	shelving



5. AUX

Control for adjusting the AUX level. This control is electrically arranged after (post fader) the channel fader (8), so that the signal level depends on the position of the channel fader.

The AUX path can be used to send a signal to a separate external effect unit or to drive a separate monitor power amp.

For further information see : **26-29. AUX.**

6. EFFECT

Control for sending a signal to the built-in digital effect unit (reverb/delay). This control is also arranged after (post fader) the channel fader (8); the effect signal level depends on the position of the channel fader as well.

Please control the send signal to the integrated effect module carefully. The PEAK indication of the effect module (22) may only light up briefly at dynamic signal peaks. If the LED is lit continuously, the unit is being overdriven.

For further information see: **20-25. EFFECT**

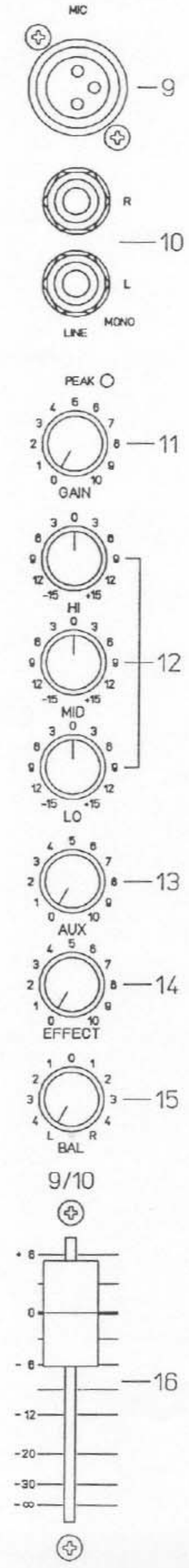
7. PAN

This control determines the stereo position of the input signal. In central position the signal is divided into equal parts on both master channels left and right.

8. Channel Fader

The channel fader is used to adjust the volume of the single channel and the volume balance between the individual channels.

Try to adjust the fader near the 0 dB position. This will also enable you to adjust the volume with sufficient control displacement even if you have great level differences between the different input channels. The master volume of the complete unit is controlled by the master faders (19).

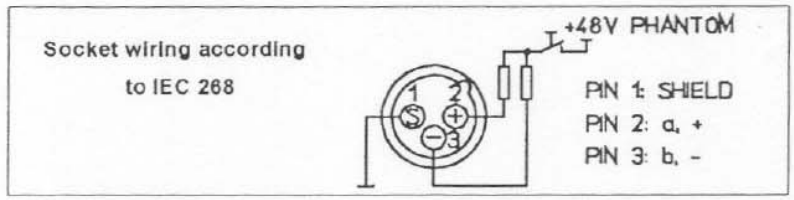


INPUT/STEREO

The input channels 7 and 8 are designed completely in stereo. You can connect all types of stereo signal sources (e.g. drums, synthesizers, samplers or submixers).

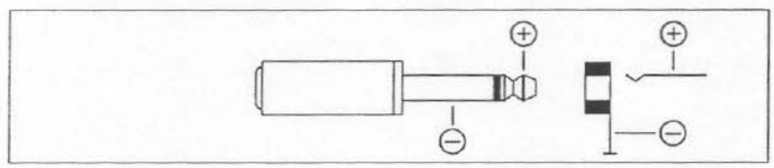
9. Mic

The stereo input channels can, of course, also be operated in mono. With low impedance microphones you should use the XLR socket MIC and with higher level signal sources the jack LINE L/MONO (10).



10. LINE

Unbalanced inputs L + R for instruments and other high level signal sources. The inserti-on of a plug into these sockets will switch off the XLR input.



Please make sure that the respective channel fader or at least both master faders are closed before connecting signal sources, to protect your audience from anno-ying click noise.

11. GAIN + PEAK LED

Control for adjusting the input sensitivity between -56 dBV (1.5 mV) and -20 dBV (95 mV).

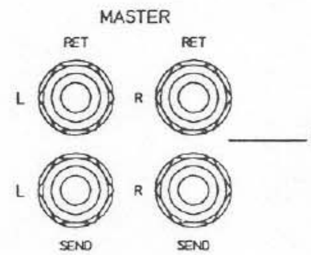
This control should be adjusted so that the PEAK LED only lights up briefly at signal peaks. This will result in an optimum S/N ratio. If the PEAK LED lights up, the available headroom is 10 dB before the input signal is distorted audibly. Please note that the sound corrections with the EQ controls influence the input signal level as well. Check the correct setting of the GAIN control again after adjusting the sound.

12. EQUALIZING

The EQ section allows for a very comprehensive and effective control of the input signal for the different frequency regions. Turning the control to the right increases the respective frequency region.

When adjusting you should always start at a neutral position, i.e. all controls are in central position. If possible, do not use extreme control positions; normally a small sound correction is enough and delivers the best sound quality. Take as orientation the naturalness of the reproduction and depend on your musical trained ear for sound checking. You can effectively avoid acoustical feedback by operating of mid control (MID) gently. Avoid too much gain especially in this frequency region; a slight lowering allows an amplification of microphone signals with little danger of feedback.

MASTER

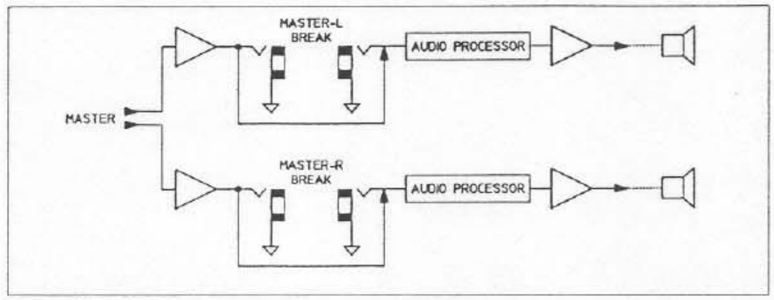


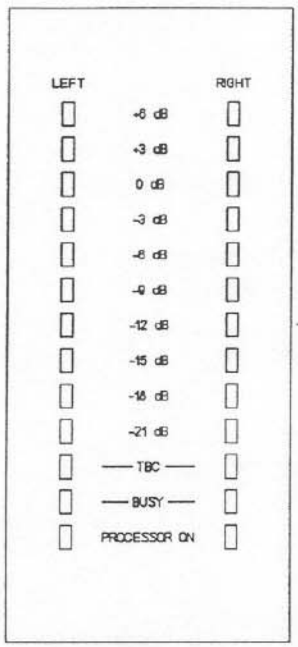
17

17. MASTER RET/SEND

These jacks are used for looping in an equalizer etc. into the master signal path.

The return jacks RET interrupt the master signal. The SEND jacks can also be used as master out e.g. for additional power amps.





18

18. LEFT + RIGHT LED Level Meter

The two LED Level Meters show the power modulation of the two power amplifiers. The area from +3 dB to +6 dB indicates risk of overdriving.

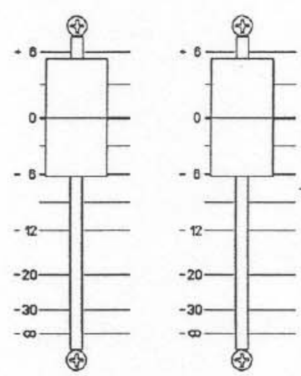
Please avoid overdriving. The unit or connected loudspeaker cabinets could be damaged.

TBC

The short-term peak output power of the PCA power amps is considerably higher than the rated output power in order to give you excellent dynamic behaviour. The "dynamic headroom" (IHF-A) is 1.5 dB which is equivalent to approximately 350 Watts/4Ohms output power. The TBC circuit contains a simple 1st order voice coil model to simulate the thermal behaviour of a typical woofer. At continuous overdriving or modulation with square wave signals this part of the processor reduces the power output to the rated output (250 W / 4 ohms), to protect the connected loudspeaker system against thermal overload of the woofer's voice coil .

Please note that speakers with less power capability than the rated power cannot be protected completely by the "Thermal Brain Circuit".

L MASTER R



19

BUSY

This indicator lights up if the limiter part of the processor is activated. Continuous lighting of the BUSY LED indicates danger of overdriving the amp and should be avoided by reducing the output volume.

PROCESSOR ON

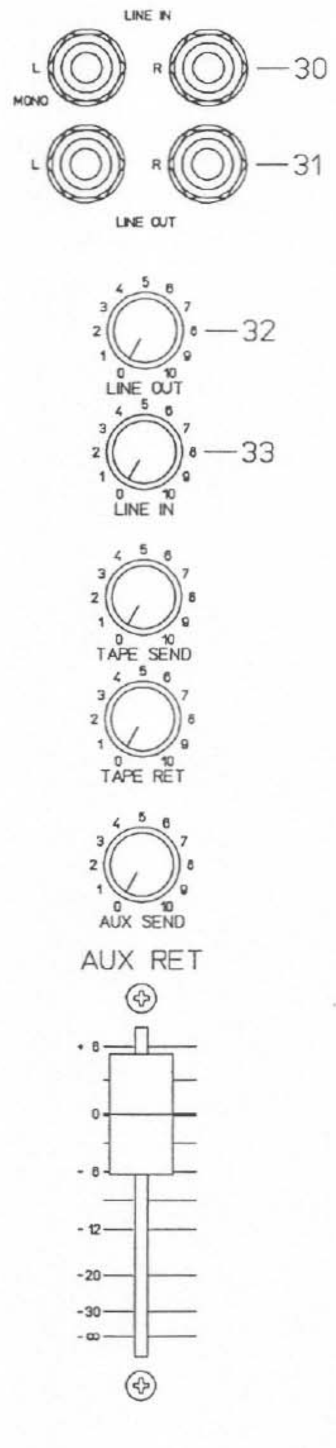
These LED's indicate that the unit is ready for operation.

19. MASTER L + R

Master volume control for the left and right master output.

For further information see: **40. SPEAKER OUTPUTS**

Please make sure that the respective channel fader or at least both master faders are closed before connecting signal sources, to protect your audience from annoying click noise.



LINE

30. LINE IN L/MONO + R

You can feed in a stereo signal, e.g. from submixers via these jacks. This signal is fed to the master bus (like the other input channels).

31. LINE OUT L + R

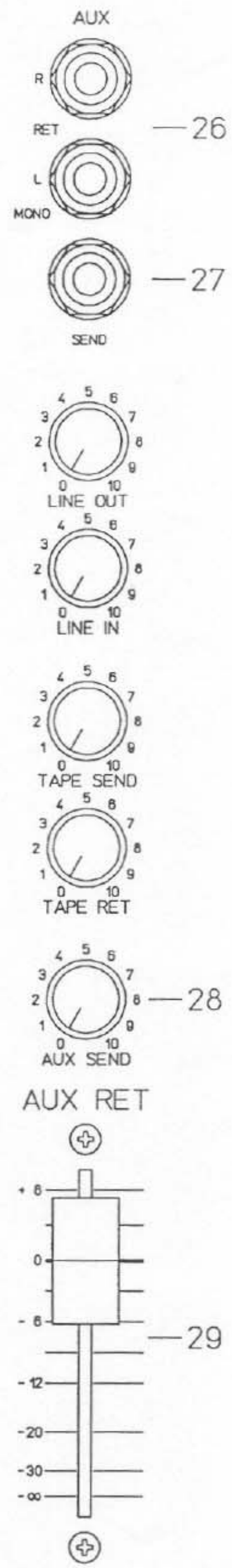
You can take the master bus signal (pre master fader) from these sockets. The LINE OUT signal therefore does not depend on the position of the MASTER L + R faders (19). You can feed via these jacks a separately controllable master bus signal e.g. to a master mixing desk or into an separate amplifier / speaker circuit for monitor purposes.

32. LINE OUT control

With this stereo control knob you can adjust the output signal for the LINE OUT jacks (31).

33. LINE IN control

With this stereo control knob you can control the input signal, coming from the LINE IN jacks (30) and the mixing in of this signal into the master bus. The master volume depends on the position of the MASTER L + R faders (19).



AUX

26. AUX RET R + L/MONO

Jacks for feeding a stereo signal into the master bus. When using e.g. an external stereo effect unit, connect the output socket of this unit to the jacks AUX RET R + L. If you have a mono unit, use the jack AUX RET L/MONO.

The control AUX RET (28) controls the volume of the signal added to the master bus.

27. AUX SEND

28. AUX SEND CONTROL

An AUX signal can be fed e.g. to an external effect unit or a separate monitor power amp via the AUX SEND jack (27).

The master send level is controlled by the AUX SEND control (28).

The AUX controls of the input channels (5, 13) allow a separate adjustment of the aux signal level of the respective channel.

29. AUX RET

Stereo volume control for mixing the AUX return signal into the master signal (e.g. effect volume).

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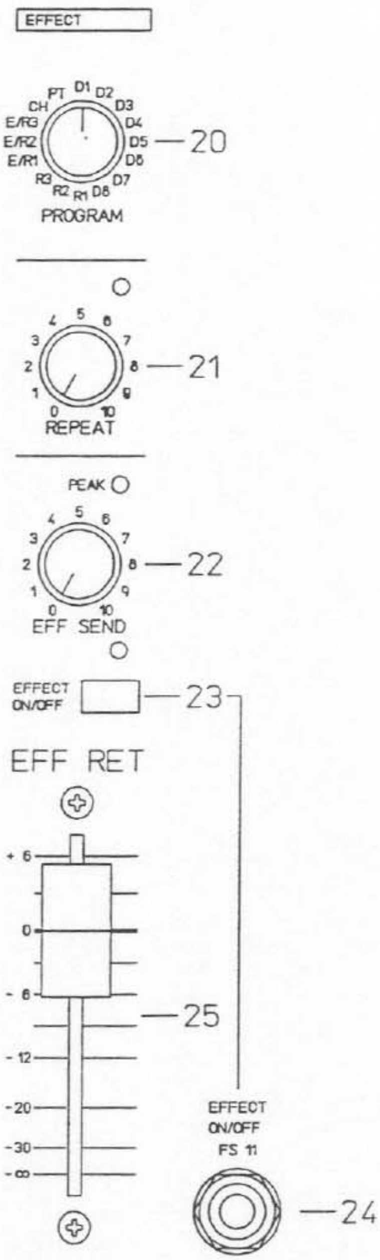


EFFECT

20. PROGRAM

Program switch for 16 effect programs

REVERB	CHARACTERISTICS:
R1 SMALL REV	SHORT REVERB TIME, SMALL ROOM
R2 MEDIUM REV	MEDIUM REVERB TIME, MEDIUM HALL
R3 LARGE REV	LARGE REVERB TIME, LARGE HALL
E/R1 E/R SMALL	ECHO/REVERB, SMALL ROOM
E/R2 E/R MEDIUM	ECHO/REVERB, MEDIUM HALL
E/R3 E/R LARGE	ECHO/REVERB, LARGE HALL
CH CHORUS	FOR VOCALS, BRASS, WOODWINDS
PT PITCH	DUAL PITCH SHIFTER

DELAY
D1 DELAY 60 ms
D2 DELAY 100 ms
D3 DELAY 170 ms
D4 DELAY 280 ms
D5 DELAY 460 ms
D6 L-R DLY 170 ms
D7 L-R DLY 280 ms
D8 L-R DLY 460 ms

21. REPEAT + LED

With this control the amount of echo repeats will be adjusted.

The green LED indicates that this control is active.

22. EFF SEND + PEAK LED

With this rotary control you can adjust the input level for the built-in effect unit. The PEAK LED indicates risk of overdriving the effect unit. Please adjust the control EFF SEND so that the LED only lights up breifly at dynamic signal peaks.

23. EFFECT ON/OFF + LED

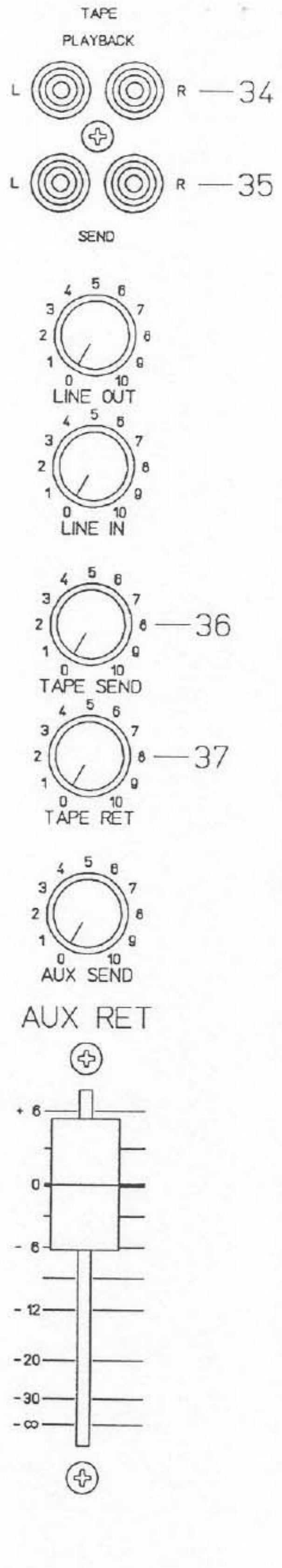
Pushing this button (green LED lights up) will switch on the effect module.

24. EFFECT ON/OFF FS-11

The effect can be switched on and off by the foot switch FS-11. The button EFFECT ON/OFF (23) must be pushed and a foot switch FS-11 has to be connected to the jack (24). The red LED in the foot switch lights up if the effect is ON.

25. EFF RET

Stereo fader for adding the effect signal to the master signal.



TAPE

34. TAPE PLAYBACK

You can playback a tape or cassette recording via these sockets. The volume is controlled by the TAPE RET control (37) and is not dependent on the position of the master faders (19).

35. TAPE SEND

From these sockets you can take the master bus signal for tape recording. The recording level is controlled by the TAPE SEND control (36) and is not dependent on the position of the master fader (19).

36. TAPE SEND control

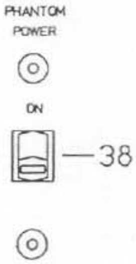
With this stereo control knob you can control the output signal of the TAPE SEND sockets (35). This control is for driving a connected tape recorder or cassette player.

37. TAPE RET control

This stereo control adjusts the volume of the tape playback. This tape signal is fed in after the Master L + R faders (19) and is therefore not dependent on the position of the MASTER L + R faders (19).

You can play back tape signals at any volume without altering the master volume.

MASTER



38. PHANTOM POWER

Central switch for the 48 V phantom power supply for the MIC sockets (1, 9).

If you are using phantom powered microphones (e.g. condensor micropho-nes), they can be supplied by the PSX power supply. Separate batteries for the microphones are not necessary.

Please only switch the phantom power supply on and off if the PSX 850 is switched off.

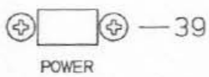
With PHANTOM POWER ON you must not connect unbalanced signal sources (keyboards, mixers) to the XLR sockets. These units could be damaged or destroyed by the phantom voltage.

ATTENTION! Important Note!

Basically speaking, phantom-powered microphones and balanced dyna-mic microphones can be operated simultaneously.

However, there are some balanced dynamic microphones, which are especially sensitive and could possibly be damaged by the phantom voltage. Please read the operating manuals of your microphones care-fully.

For reasons of safety, always ensure that the PSX 850 is switched off (39. POWER), if balanced dynamic microphones are to be connected to the mic input sockets. You will thus avoid possible damage of these especially sensitive dynamic microphones.

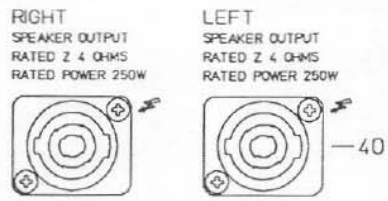


39. POWER

Mains switch for switching the unit on and off.

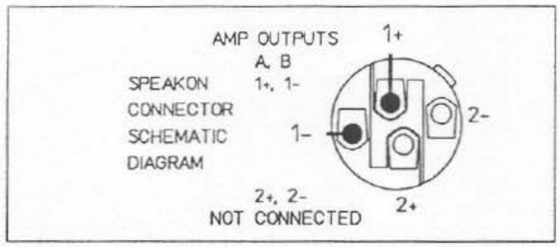
The unit is ready for operation if both PROCESSOR ON LEDs (18) are lit and the power relays have switched the output stages to the speaker outputs.

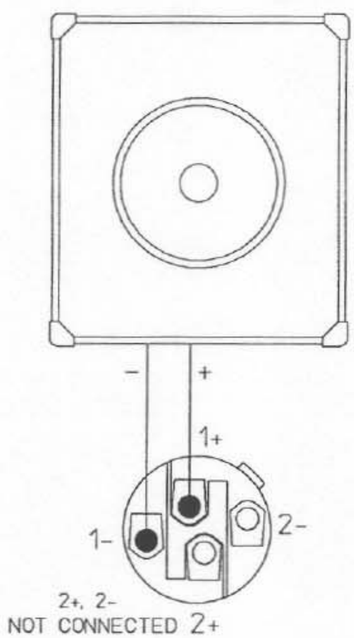
Make sure that both master volume faders are closed when you switch the unit on. You will protect your audience and your equipment from un-necessary inconvenience caused by unwanted amplification and possi-ble feedback.



40. SPEAKER OUTPUT RIGHT + LEFT

The PSX power mixers are supplied with professional SPEAKON high power connectors. This mechanically and electrically safe connection complies with all safety demands and regulations and allows the use of high-powered loudspeaker cables up to a cross-sectional area of $4 \times 2{,}5\ mm^2$.





Check the polarity of the connected speaker cabinets

To prevent acoustic cancellation problems, the polarity of the loudspeaker cabinets connected to the outputs must be equal. Otherwise the bass can sound muddy, supresssed and unprecise and weird midrange lobing problems can occur.

A very simple checking method involves a 9 V battery. If the + pole of the battery is connected to the + pole of the loudspeaker connector, the cone of the woofer should move outwards.

The correct polarity of mid-range and high-range speakers cannot be checked in this way, because some crossover networks change the polarity of speakers in the mid-range and high-range region.

E-V loudspeaker cabinets are internally wired correctly and do not need any polarity check procedure.

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SPECIFICATIONS

SINGLE CHANNEL OUTPUT POWER

(measured with "Dynamic Headroom" test signal according IHF-A: 1 kHz Tone burst, 20 ms ON, 480 ms OFF)

SPEAKER (L or R), 4 ohms 355 watts

DYNAMIC HEADROOM 1,5 dB

C. FREQUENCY RESPONSE

-3 dB referenced to mid-band level

1. MIC — SPEAKER : 8 Hz - 55 kHz
2. LINE — SPEAKER : 8 Hz - 30 kHz

D. AMPLITUDE NON-LINEARITIES

1. Rated Total Harmonic Distortion $k \leq 0,3$ %
2. Norm Total Harmonic Distortion $k \leq 0,03$ %
   (power amp only : measured from
   BREAK RETURN to SPEAKER OUT)
3. Norm Total Harmonic Distortion $k2 < 0,018$ %
   (mixer only: measured on BREAK SEND)

all higher distortion products lower than measuring limit (measured with spectrum analyzer)

E. NOISE LEVEL

- R(Q) = 200 Ohms between pin 2 and pin 3 of the XLR input socket
- U(F) = Noise voltage, unweighted with B = 20 Hz ... 20 kHz, quasi peak-weighted (IEC 268-1)
- U(G) = Noise voltage, frequency weighting filter according CCIR, quasi peak-weighted (IEC 268-1)
- U(A) = Noise voltage, dB(A) frequency-weighted, RMS (IEC 268-1)
- S/N ratio ref. rated output voltage (power)

1. Rated noise level (typ.)

	Noise-Voltage	S/N-Ratio	equiv. input noise voltage	equiv. input noise level
1.1 U(F)	40 mV	58 dBq	1,9 µV	-114 dB
1.2 U(G)	82 mV	52 dBqp	3.9 µV	-108 dB(G)
1.3 U(A)	16 mV	66 dBp	0,76 µV	-122 dB(A)

2. Residual output noise

2.1 U(F) = 1,7 mV (85 dBq)

2.2 U(G) = 3,1 mV (80 dBqp)

2.3 U(A) = 0,7 mV (93 dBp)

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SPECIFICATIONS

OVERALL TEST DATA PSX 850

Standard specifications: IEC 268 part 3

0 dB = 1 V (RMS)

A. POWER SUPPLY

1. Mains:	AC
2. Rated voltage:	120 V
3. Rated frequency:	50 - 60 Hz
4. Rated power consumption:	1100 watts
5. Normal power consumption:	320 watts
6. Tolerance of mains voltage:	± 10%

B. INPUT CHARACTERISTICS

Input sockets	Rated Input Level	*1 Max. Input Level
MIC	-56dB (1.5 mV)	-2dB (780 mV)
LINE (Mono)	-38dB (13 mV)	+18dB (7.6 V)
LINE (L + R)	-38dB (13 mV)	+18dB (7.6 V)
TAPE-PLAYBACK (L+R)	-15dB (180 mV)	+12dB (4.0 V)
LINE-IN/MASTER (L+R)	-10dB (300 mV)	+11dB (3.8 V)
AUX-RETURN (L + R)	-8dB (400 mV)	+11dB (3.8 V)
MASTER BREAK/RETURN	0dB (1.0 V)	n.V.

C. OUTPUT CHARACTERISTICS

Output sockets	Rated Load Impedance	Output Level *2	
		Rated Value	Max. Level before Clipping
SPEAKER (L + R)	4 ohms	250 watts	n.v.
	8 ohms	180 watts	n.v.
MASTER BREAK (L+R)	10 kohms	0dB (1.0 V)	[+13dB (4,5 V)]
AUX SEND	10 kohms	+8dB (2,6 V)	+17dB (7,5 V)
LINE OUT (L+R)	10 kohms-2dB (800 mV)	+12dB (3,8 V)	
TAPE RECORD (L+R)	47 kohms-2dB (800 mV)	+ 12dB (3,8 V)	

Output Sockets	Stabilizing
SPEAKER (L + R)	2% (0.17dB)

BLOCK DIAGRAM