TAPCO®

TECHNICAL AUDIO PRODUCTS CORPORATION

Panjo

SERIES 72

OWNER'S MANUAL

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Panjo

SERIES 72

OWNER'S MANUAL

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INTRODUCTION

Panjo is the name we've chosen to identify TAPCO's Series 72 and 74 Mixers. No.... Panjo is not an acronym for a series of audio terms, but it does have a very real meaning for us.

We started considering our next entry into the mixer market. We knew that TAPCO had a reputation to uphold, so we looked back to our beginnings in the music business. In doing so, we identified what we still believe are the four most important qualities of a professional audio product:

* performance - the new unit must make the whole system sound better.
* reliability - it must hold up both in normal use and abuse.
* features - improved control of the sound must be provided.
* price - the price must be a fair one for both the buyer and the manufacturer.

In other words, we had to design it so that each characteristic would appear in the finished unit in proper balance. This is not an easy task, as evidenced by the many products on the market which seem to have all the right features and a low price, but which sound bad and fall apart, even in normal road use, or the fine performers that are priced out of the reach of most of us.

The design process started with a meeting of TAPCO's management team. It was our desire to challenge our Engineering Department to meet Marketing's features and performance goals and the cost and reliability goals of Manufacturing.

About a month later, Engineering came back with their proposal. "We'll utilize the latest integrated and discrete circuits and components," they said, "so that noise, distortion and signal interaction are virtually eliminated. The unit must be as compact as possible to reduce circuit boards, internal wiring and sheet metal to a bare minimum."

"We will still get all of the features you want without exceeding the compact size limits by using specially designed connectors and controls with increased component density. At the same time, hand wiring will be nearly eliminated by wave soldering just about everything to the printed circuit boards. And if that's not enough, the unit will be made of sheet metal modules for a super rugged package. This will also give Manufacturing some flexability in creating a variety of end products by marrying several different modules. We'll use standard, reliable electronic components that are virtually 'bullet-proof', and all of the printed circuit boards will be interconnected by plug-in cables that can be replaced or repaired in a flash.... and you won't believe the styling 'til you see it!", they grinned.

Well, their logical design ideas and enthusiastic determination overcame any possible doubts as to the project's eventual success, so I gave it the "go-ahead". Panjo, by the way, is the name of the Pizza Parlor where the design team had lunch the day they dreamed up their answer. After the project began, we continued to refer to it as Panjo to remind us that we had committed ourselves to bringing out a brand new series of mixers that optimized performance, reliability, features and price. Now the name 'Panjo' seems to exemplify all of the care and effort we put in, in reaching those goals....and it stuck!

DAVE MERREY
GENERAL MANAGER
TAPCO

DESCRIPTION

LINE INPUT

A standard ¼" phone jack for line level signals. Examples of line level signals include some electric and most electronic keyboards, synthesizers, turntables (with appropriate preamps), tape decks and the line outputs from other mixers. All input channel controls, including the variable gain Trim control, affect the Line Input. Maximum input level before preamp clipping is 15V or +26 dBV. Minimum input level to set a nominal channel send level of -3dBV is 55 mV, or -23dBV. The input impedance is 33 k Ohms. If in using the line input, an acceptable level is not possible with the gain trim in its furthest clockwise position, the signal must then be treated as a mic level signal. If necessary, use an appropriate balancing transformer (EV Model 502CP or equivalent) and the microphone (XLR) input. The Line Input can also be used for an effects return.

MIC INPUT

A 3-pin XLR connector for balanced low impedance Microphone Inputs. The Mic Input is actively balanced with 24V phantom powering available. Active balancing allows elimination of the input transformer (with its limitations) while maintaining the RF and hum rejections of a good transformer coupled input. The maximum input level is 1.5 V, or +6dBV. Pin 3 is "hot". Phantom powering applies 24V dc to pins 2 and 3, referred to ground. Any low impedance <u>balanced</u> output microphone (dynamic, ribbon) may be used regardless of the position of the phantom power switch. If the switch is set to on, most phantom powerable condenser mics may be used simultaneously with other non-powered mics. The mic input will also accept direct box outputs. In this manner, the mic input may be used with low level, high impedance sources.

CHANNEL PATCHING, OUTPUTS

These connectors are used for the insertion of specialized external processing gear in an individual channel, and/or for direct outputs. The Send and Return jacks are normalled, so the signal path will not be interrupted unless a plug is inserted in the Return jack.

The Send/Return insert point is immediately ahead of the channel fader. This point is <u>after</u> the channel EQ. The send jack may be used as a direct output (for tape recording or other use) but it will not be affected by the slide fader. All level adjustments are made with the channel's gain trim control. When used in conjunction with the return jack, this offers the distinct advantage that any processor device used will have its output noise attenuated by the channel fader. In addition, the processor operates at constant level (that is, it is unaffected by the slide fader) and thus it is at its optimum signal to noise ratio. As a result, the return jack can also be used as an additional line level input. It will not be affected by the EQ controls and cannot be accessed by the monitor bus.

PEAK INDICATOR, TRIM CONTROL

Trim adjusts the gain of the first preamp stage in the input channel, for both the Mic and Line imputs. The Peak indicator LED lights when an overload condition occurs, in <u>either</u> the first stage or in the EQ and fader stage.

The trim control is used to match the gain of the first preamp stage to the Signal strength of the source being run through the channel. To get the cleanest, quietest operation from the board it is important that the Trim control be properly set. To set up a mix, first put all the input and sub-group slide faders at "12". Then adjust the Trim controls for a rough mix, and do the fine tuning with EQ and faders as necessary. Whenever possible, it's best to try to maintain that "straight line" relationship between all the faders. When this is done, all the levels within the console are very close to being optimized for the best noise and distortion performance. Of course it's not always possible to adjust the Trim controls during a mix because they cause the levels at the Monitor, pre-Aux and channel patch point to change as well.

There's no real harm in having a widely varying set of levels on the sliders, unless the Peak indicator is being lighted. But the sensing circuitry for the Peak LED monitors levels after both the first stage preamp, and the EQ and fader stages. So signals that are set up as described above, then boosted with either lots of EQ or drastic level increases at the fader, may cause the Peak LED to flash even though the first stage is OK. In any case, reducing the overall gain with the Trim control will almost always eliminate the overload condition. If not, it may be necessary to use an external pad to reduce levels before they enter the channel.

AUX

The Aux control can be used either pre or post fader to provide an additional mix for monitors, cue, effects, reverb, etc.

The Aux bus can get its signals from two points in the input channel -- one point is just after the first stage like the Monitor send, and the other just after the channel fader, like the Effects send.

The Aux send control is off in the center of its rotation (the pot has a center detent). To the right, clockwise, the signal is drawn from the post-fader point. To the left, or counter-clockwise, the signals are drawn from the pre-fader point in the circuit. This dual purpose pot arrangement saves a switch on each channel without depriving the board of a very necessary and useful function.

The Aux controls may be used in any combination of pre and post sends simultaneously, on as many channels as the situation requires.

MONITOR

The Monitor control is a pre-fader, post EQ send that provides a separate mix for stage monitoring, headphone cue, or what have you. The monitor send is not affected by the action of any of the controls on its respective channel except the Trim and the Mic/Line switch. The Monitor send signal is derived before all other controls, so the stage monitors or headphones do not need constant attention.

EFFECTS

The Effects control is a post-fader send most often used in conjunction with the Effects Returns in the output section to provide a mix for external effects devices. Because it is usually desirable to maintain a specific straight signal to effects ratio for any fader setting, the Effects send is derived after the channel fader.

EQUALIZATION CONTROLS

The EQ section consists of ±18 dB shelving type bass and treble controls, and a switchable frequency, ±12 dB peak/dip type midrange control. The midrange frequency to be equalized may be set at 600 Hz or 3.5 kHz. You can see that in the extremes of their range the bass and treble curves are overlapped by the midrange, giving rise to some very interesting EQ possibilities for some of those difficult situations.

CHANNEL FADER

The slide fader controls the output level of the channel as it's fed to the sub-groups. The control should be normalized at the "12" mark. With all controls set to their designated normalized operating points, all circuits in the board are optimized from both noise and distortion standpoints. In other words, the signal levels are high enough to keep noise from creeping in and low enough to insure plenty of headroom and freedom from slew-induced distortion. If the fader must run wide open to get enough level, turn up the Trim control. Conversely, if the fader must be pulled way back to get the right level, the Trim control should be turned down. For optimum performance, the channel faders should always be run as close to the "12" mark as possible.

STEREO OUTPUT SECTION

The stereo output section is where everything comes together. The L and R Stereo Masters control the overall level of the main mix. They also affect the mono mix as it is a composite of left plus right. The output of the stereo master controls then go to several different places:

1. The left and right main output
2. The mono output
3. The left and right Output Level Meters
4. Headphones, via solo switching circuitry.

The mono output is a 50-50 mix of the left and right outputs. It has its own master control and may be metered by depressing the meter select switch.

The three send busses have their master gain controls in the output section. Each of the busses has a gain control and a solo switch. They may be metered by selecting solo for that bus and depressing the meter select switch. Since the sends are post-fader and before the patch point, levels will always reflect the fader settings.

EFFECTS RETURNS

There are two effects returns provided. Each has 1 input (guitar level to line level), a level control, pan pot to the stero master bus, and a "to monitor" control. The effects to monitor control allows any effects returned to the mix via the effects returns to be mixed into the monitor bus. This control is post fader, that is, it is affected by the main effects return level control.

Additionally, effects return 'A' doubles as the reverb return when the <u>internal reverb option</u> is ordered. The reverb is defeated if a plug is inserted into the effects return 'A' input jack.

When used with "guitar type" effects boxes, the Effects send "lo" jack should be used and the effects return may have to be turned up considerably past half way. This is normal and is the optimum way to operate these devices.

SOLO

In addition to its usual solo listening function, the solo bus may be used to observe levels, at any of the solo points, on the right output meter. Normal operating levels (OVU) will be seen if all the controls on the input channel, or in the particular bus that's being soloed, are being run in their proper operating range. If something's amiss, levels shown by the meter will be either too high or too low, as the case may be. It is not necessary to adjust the gain trim control for OVU on the solo VU meter. Adjusting the gain trim for "0" will not optimize the channel for maximum headroom.

The solo Status indicator is lighted whenever a Solo button is depressed. Because the Solo buttons are all locking type switches (so you can have both hands free while soloing something), a red LED indicates that the Solo system has taken priority over normal signals.

The rear panel solo stacking connectors are used when stacking with another TAPCO mixer with an internal solo system (other Panjo mixers, C-12, C-8E). The solo audio stacking connector connects to the solo output of the slave mixer, and the solo control jacks are patched together.

HEADPHONE LEVEL

This control affects only the level of the Headphone Output signals. It will usually be operated between 9 and 12 o'clock as there's plenty of extra gain to help you find those very weak, "lost" signals.

The Stereo/Mono switch affects the headphones only. It is useful when using the 4 subgroups for a mono mix. This can allow subgroup pan assignments that might facilitate some other use, for instance, simultaneous recording and PA.

APPLICATIONS

USING THE PATCHING CONNECTORS AND DIRECT OUTPUTS

The Panjo series products are equipped with a number of access jacks that greatly enhance their flexibility in both recording and sound reinforcement situations. Each input channel and subgroup has a pair of Send/Return jacks. The Send jack can also do double duty as a direct output.

The Send/Return jacks are located on the rear panel. They allow the insertion of an outboard signal processing device (compressor, equalizer, etc.) into the normal signal flow of any input or submaster. Note on the block diagram that the Send/ Return jacks are located before the fader and after the EQ. The submaster patch point is located after the submaster fader and before the pan pot. Again, refer to the block diagram.

All signals that appear at the Send jacks will be at line level because they've already been processed through the channel's preamp at that point. The Send/Return jacks are "normalled", that is, the normal signal flow through the input channel will be interrupted only when a plug is inserted into the return jack. A plug inserted into the Send will not disturb the regular signal flow in any way. This arrangement has several benefits.

Outboard processing gear can be connected to the Send jack without affecting the channel's operation. The mixer's gain Trim and fader settings can be optimized at the same time the controls on the external processor are set. For instance, if a compressor were completely patched into the channel before the gain Trim and fader settings were adjusted, on both the mixer and the compressor, it's likely that some sort of level mismatch would occur. The result could be either clipping distortion, or excessive noise. If the channel gain is too high, the compressor's input might be overloaded, and if the compressor's output level is too high the compressor might run out of output headroom. Likewise, if levels are too low, especially at the compressor's input, noise could cause more trouble than the compressor does good.

In any case it's always best to set up the input channel for normal operation, as if no outboard gear were to be used. Patch the Send to the processor's input, and adjust pertinent controls as much as possible. Then, patch the processor's output to the Return jack (breaking the normal, so the signals are now routed through the processor), and everything should be close to optimal operation.

INPUTS, OUTPUTS AND SIGNAL LEVELS

The input is the beginning of the signal path, and the output is the end. Signals always enter the unit at inputs and come out of outputs, so outputs are always connected to inputs and vice versa. Intermediate points are usually designated Send and Return, with the Send jack serving as the Output, and the Return jack as the Input.

Signals exist at many levels or strengths. In audio, weak signals are generally referred to as "mic" level. Not so weak signals are considered line level and strong signals are called speaker level. To give you an idea of what the actual levels are; mic levels are generally considered to range from around -60dBm to -20dBm (.000000001W to .00001W), line levels range from approx. -20dBm to +30dBm (.00001W to 1W), and speaker levels from 1W up. Bear in mind that these numbers are approximations and it is possible for a microphone to put out line level signals.

Both inputs and outputs have specific maximum and minimum operating levels, and outputs usually have a specified minimum impedance into which they will work. The minimum operating level for outputs as well as inputs, is related to one thing; noise, because all electrical circuits generate some unchanging amount of noise when they operate. When audio circuits are forced to handle signals that are too weak, their inherent noise becomes evident and you hear hiss. The hiss is at a level usually referred to as the noise floor of the circuit. When signals are kept well above the noise floor the hiss is masked by the desired signal, and the circuit is operating well within its normal range of levels. Until the maximum is reached, of course.

A circuit's maximum output level is determined by the voltage on which the circuit is designed to operate, and by the minimum impedance into which the circuit is designed to work. So it'll only put out so many volts into some given load, like 600 Ohms. When the circuit is pushed beyond its limits you hear harsh clipping distortion. Likewise, inputs have inherent level limitations, also determined in part at least by the maximum number of volts available within the circuitry. But inputs don't suffer from any impedance problems other than those incurred by whatever is plugged in, i.e. some kind of output. Of course, they may be 50% responsible for a mismatch if the preceding output has an impedance greater than the input's impedance.

As a general rule, impedance will match if the input impedance is greater than the output impedance of the preceding device. How much greater is a matter of some debate. When a device is asked to work into a load greater (in other words, an impedance lower) than it should, distortion increases dramatically, and headroom is reduced. Most manufacturers specify minimum operating impedances on the order of ten times the actual output impedance of the circuit. All outputs on the Panjo series products will deliver +20dBm (7.8 Vrms) before clipping.

STEREO AND MONO

The minimum requirements for a monaural sound reproduction or reinforcement system are:

1. one microphone
2. one amplifier
3. one loudspeaker
4. listener with one working ear

In some larger systems there may be many microphones, amplifiers and speakers. The important thing is that in a mono system all the speakers are reproducing exactly the same program material.

The minimum requirements for a stereo sound system are:

1. two microphones, in the same acoustic space
2. two amplifiers
3. two loudspeakers, in a common acoustic space
4. listener with two working ears

What's important here is that there are two completely separate amplification channels, all the way from the source to the listener. If anywhere in the chain the two signals get mixed together the result is mono.

Most stereo systems, recording or reinforcement, rely on multiple monophonic sources that are panned into some stereo perspective, with panpots. As an example, a guitar is recorded in an isolation booth using one microphone -- it's a mono source. We can pan that source to the left, for instance, so the guitar seems to be localized in a particular spot. You can tell which direction the sound is coming from, but you can't tell anything about the room in which it was recorded because there isn't any ambience. The point is, even though the signal has been positioned with the panpot, it's still a mono source. 16 mono tracks panned in 16 different positions in a stereo panorama is still mono, albeit 16 track mono. Stereo would be created if those same 16 tracks were recorded as 8 stereo pairs, in other words eight sources each picked up by two mics, one feeding the left channel and the other the right.

PHANTOM POWER

Your Panjo mixer is equipped with a +24 VDC Phantom Power supply which is switchable on and off. The switch is located adjacent to the AC power ON/OFF switch on the rear of the mixer. An LED on the front panel indicates the presence of Phantom Power. The DC voltage is applied to all Inputs simultaneously. A general discussion of Phantom Powering follows:

> CAUTION: Use of an external Phantom Power supply which is NOT DC isolated from your mixer could result in damage to the Input circuitry. Please consult the manual of your external supply.

All condenser microphones have one thing in common: They all require some kind of electrical power. This power is needed to operate the mic's preamp circuits, and in some cases to charge the capacitive plates that constitute the actual transducer elements. In the newer electret condenser mics the power is used only for the internal preamp because the plates are permanently charged when the mic is built.

Early condenser mics contained tube type preamps. The tubes required an external AC power supply, which was usually connected to the mic through a multi-conductor cable that also housed the audio lines. Transistors have now virtually eliminated the use of tubes in condenser mics because they offer lower power consumption, greatly reduced size, and improved noise performance.

The newer, solid state mics get their power from either internal batteries, or an external supply that is fed to the mic via the audio cable. Many of the battery powered mics may also be externally powered - check the manufacturer's literature for specifics. External power may be applied two ways:

1. Phantom or Simplex powering DIN standard 45 596

2. "T" system powering (also called modulation lead powering, or AB powering), DIN standard 45 595

The difference between the two systems is the way the power is applied to the mic, through the audio cable.

Phantom powering imposes a positive voltage on both audio conductors, using the shield for the power ground. The "T" system imposes a positive voltage on one audio lead, using the other as the power ground. In the "T" system the shield functions, as it usually does in a balanced system, only as a shield. THE TWO SYSTEMS ARE NOT COMPATIBLE WITHOUT SPECIAL ADAPTERS.

The term "PHANTOM POWER" specifically means one thing: +DC applied to the microphone on both pins 2 and 3 thru a current limiting network.

The following microphones are "T" system powered and are not phantom power compatible under any circumstances:

Neuman:	FET 70 series, KTM
Sennheiser:	MKH 110 (-1)
	MKH with T suffix
	MKH 435 U
	MKH 415, 815 (compatible with suitable adapter)

The following microphones use a remote powering system unique to them and are not compatible without their unique power supply. They will work in a phantom powered system regardless of whether phantom power is on or off:

Altec:	M20, M21, M50, M51
AKG:	C 60, C 12, C 24
Neuman:	KM 56, KM 64, U47, U67, M49, M269
Sony:	C37, C37A

The following microphones are phantom power compatible but do not require power. Some of the microphones listed are condenser types that are battery powered only:

All EV pro series (RE-15, etc.)
All EV PL series (except PL77, which is battery or phantom power)
All EV 600 series low impedance
Shure SM series
Shure 500 series wired low impedance
Sure PE series (low impedance only)
Beyer dynamic and ribbon mics
RCA broadcast mics
AKG D series
ANY MICROPHONE WITH A BALANCED LOW IMPEDANCE OUTPUT

The following microphones are phantom power compatible and do require power for operation:

AKG:	C series, CE series, except C452E
Beyer:	Electret condenser series
EV:	PL77, PL77A, 1777, 1778, CS15 series, PL76 (when used with 506 adapter)
Sony:	ECM series, C74, C76, C38B, C37P

The following microphones (note exception) are 48V microphones and will not work in a 24V system without external power. See caution note at head of this section:

Neuman:	KM83, KM84, KM85, KM86, U87, U89
	U87 will work provided internal batteries are installed and internal switch set to "battery".
AKG:	C452E series
Sony:	C500, C47P

If your microphone isn't listed here, consult the data sheet or the manufacturer.

CAUTION: A "T" powered mic may be damaged by Phantom powering. Always check manufacturer's specifications before applying power to any microphone.

Some dynamic and ribbon mics may exhibit random noise (crackling, sputtering, or even humming) when used in a Phantom powered system. The problem is that the transformer inside the microphone has developed leakage from the winding to the microphone case (pin 1). It's the leakage that causes the noise, not the power. There are three solutions:

1. Turn off the Phantom Power.

2. Insert a 1:1 isolation transformer in series with the bad mic.

3. Get the mic repaired.

Note: With the exception of "T" powered mics, the mics listed above can be used with the Panjo mixer. However, check with us or the mic manufacturer if you aren't sure about a mic.

STANDARD PATCHES

MONO PA

The standard mono PA patch is quite simple; the normal signal flow routes through the mic or line inputs, and out through the mono output (via the Stereo Master control). Additional mixes are generated through the Effects, Monitor and Aux busses, as necessary.

THE PA FEED

Taking a closer look at this hookup, the mics (or perhaps a pre-mixed line level signal from a keyboard mixer) are connected to the input channels, 1 - 12. The Panpots are used to route the individual channels to the left and right Master level control and the left and right busses are combined to form the mono signal which is available at the Mono Output through the Mono Output Master. The Mono Output is used to drive the power amp signal chain, which is perhaps headed up by an equalizer and/or electronic cross-over.

Additional mixes are created with the three sends available on each input channel: Effects, Monitor and Aux.

EFFECTS

When an effects bus is used for external effects devices, like the TAPCO 4400A Reverberation System, the signal path begins at the individual channel Effect send control, goes through the Effects Master, to the Effects Outputs, through the effects device, then back into the mixer through the Effects Return(s). In the case of the 4400A, the Hi level effects output would be used because the 4400A is a line level device. If guitar level effects were used, like the Roland Space Echo, the Lo level Effects output would be used.

Both the 4400A and the Space Echo are series type effects units -- they can be used without the need for any effects mixing capability, because both units already have controls that allow the operator to set the ratio between the effects and straight signal. But when an effects device is used in the console's effects to send and return loop, the controls on the console are used to set this ratio, because Effects send/return system is designed for parallel operation of effects generators. So, the mix or blend controls on the effects unit must be set for maximum effects, allowing no straight signal to be returned to the mixer. If straight signal is present in the effects signal, the loudness of the entire PA will be increased when the effect is used. And, if the effects unit just happens to cause a phase reversal, there will be a point at which the signals from the effects unit will tend to cancel the main signal. (Half the Effects devices available in the world invert phase!) This will not happen if no straight signal is present.

The Effects Master control should be used to set an appropriate level for the effects unit, as indicated by the unit's meter or LED's. If there are no level indicators on the effects unit, the output level should be observed on meter #2. A normal level indicated here should provide approximately the right level to the input of the effects unit. The effects Return level should be set so the headroom of the effects device is not stressed, but low enough so noise is not a problem. The exact level would be different for every device and every situation. You can usually determine what the best send and return levels are by listening to the effects return signal by themselves, on headphones with the gain turned up. The Effects Return Panpot setting is not important in a mono setup.

MONITORS

The Monitor bus is used to generate a separate mix for stage monitors, using an additional amplifier and speaker(s). As with the effects bus, the signals flow from the input channel Monitor controls, through the Monitor Master and to the Monitor output. The relative balance in the monitor mix can be heard in the headphones by soloing at the Monitor Master. Output levels are again observed only through the Solo bus, on the right hand meter.

AUX BUS

The Aux bus, with its selectable pre/post function may be used for effects, monitors, or to provide a separate mix for any other purpose. Signal flow is the same as the other two busses, but levels are observed only through the solo bus, on the right hand meter.

SIMULTANEOUS RECORDING

With a stereo recorder, just connect the machine's line inputs to the Stereo Left and Right outputs. The two tracks will be positioned in the mix as they are panned. If you are doing mono PA, you can pan everything as you want for the recording and the mono output will be the mono sum of the left and right channels.

SOME OTHER WAYS TO DO IT

Since the object is a mono PA mix, and since that mix is merely the sum of the left and right stereo signals, you can do a good stereo mix for recording and still have the same mono PA. The settings of the Pan pots will not affect the PA mix at all.

There are many ways to do stereo as well as 4 track recording, while still doing the usual mono PA.

STEREO PA

The setup for stereo PA is much the same as for mono, except that the power amp signal chain is run off the Left and Right Stereo outputs. For stereo, you must have two of everything - microphones, mixer outputs, equalizers, crossovers, power amps, speakers.

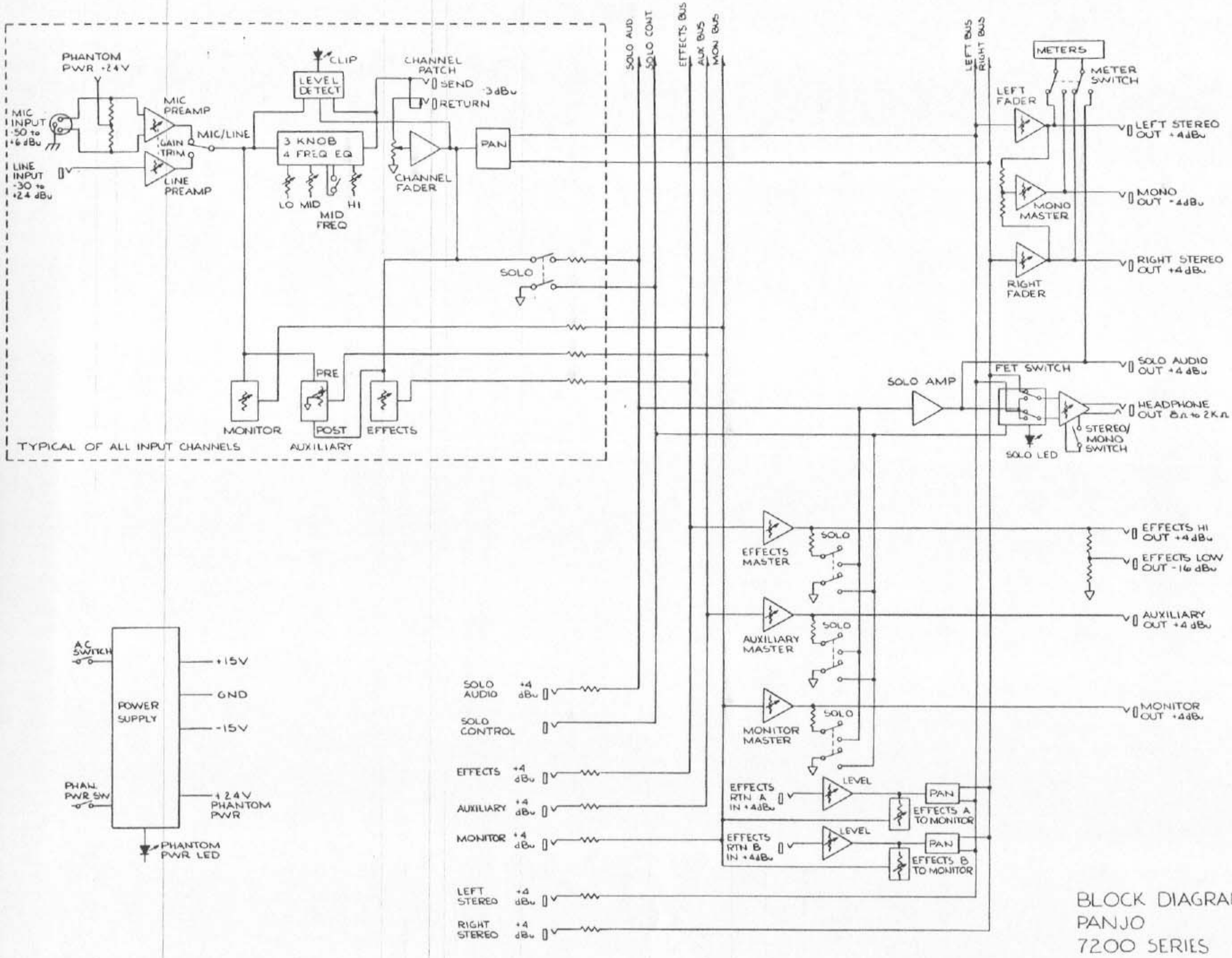
MONITOR AND AUX BUS

The Monitor and Aux busses are used the same way as described in the mono PA section, when they are used for their usual duties. Additionally, either or both of these busses may be used to provide "fill" signals for the stereo PA or recording mix.

There are several reasons why PA mixes often don't sound so good on tape. One, as mentioned in the mono PA section, is that the acoustical contribution to the recorded mix from stage sounds is often insufficient for good balance. Another is that the gross frequency response irregularities produced by some speaker/room combinations require almost radical EQ on some inputs, just to get an acceptable level of intelligibility on the PA. PA mixes are normally not as "tight" as recording mixes; the vocals are usually more out front in PA work than they might be on a recording.

The Monitor, Aux, or Effects bus may be used to enhance the recording mix. As an example, let's assume that all of the above problems apply and that the resulting recording sound is just not acceptable, even though the PA sounds OK. A "stereo" recording could be put together using the Monitor and Aux busses as substitute left and right busses. With the pre/post function of the Aux bus, those inputs that sounded better with their PA EQ could be fed to the tape machine post, while those that sounded better flat could be derived "pre". With careful mixing, a good stereo panorama and well balanced sound can be created this way, completely separate from the PA mix.

A stereo recorder with mic/line input mixing enables you to use the Effects bus to "fill" vacancies that may exist in the recorded PA mix. If, for instance, there are not enough drums in the recorded sound, a drum mix could be put together on the Effects bus and fed to the mic input on the tape machine. The Lo level Effects output would have to be used to feed the mic input on the tape machine, and the levels would have to be watched carefully to avoid overloading the machine's input. If necessary, the drum mix created in this manner could be fed to both channels of the tape recorder. An appropriate level for the drums in the recording is then set with the tape machine's mic level controls.



BLOCK DIAGRAM
PANJO
7200 SERIES

WARRANTY

LIMITED WARRANTY

(a) TAPCO warrants the materials, workmanship and proper functioning of its products for a period of one full year from the date of original purchase. If any defects are found in the materials or workmanship of TAPCO products, or if the product ceases to properly function within one year from the date of first purchase, TAPCO will repair or replace any non-conforming materials through the nearest TAPCO authorized warranty service center.

(b) Purchaser must return the product to the TAPCO authorized warranty service center, freight prepaid. A list of authorized warranty service centers is available at all TAPCO authorized dealers. Claims must first be sent to any TAPCO authorized warranty service center. If claims are not resolved by the TAPCO authorized warranty service center, any warranty claim should be sent to:

TECHNICAL AUDIO PRODUCTS CORPORATION
3810-148th Avenue N.E.
Redmond, WA 98052 Phone (206) 883-3510

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(d) This warranty is extended to the purchaser and to any purchaser from him for value.

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(g) TAPCO does not authorize any third party, including any dealer or authorized warranty service center to assume any liability of TAPCO or make any warranty for TAPCO.

Warranty registration cards must be completed and mailed to TAPCO within 30 days of purchase.

FACTORY SERVICE

Tapco has a staff of highly qualified service personnel who can assist with any field problems which may arise, and are able to answer questions concerning any aspect of the use and performance of our products. Our telephone number is area code 206 883-3510. If you wish written information, replacement parts, or factory service, our address is:

TAPCO
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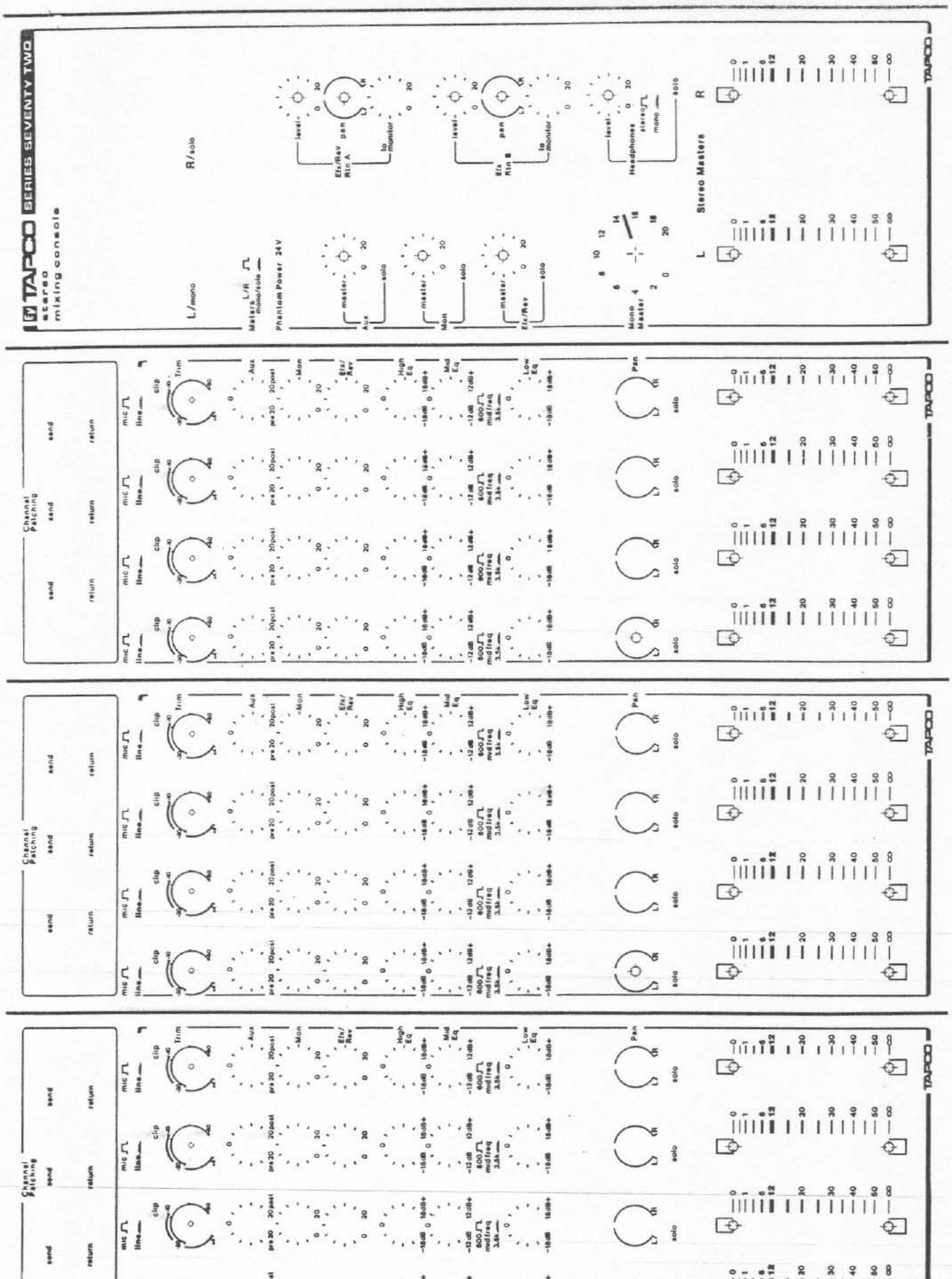
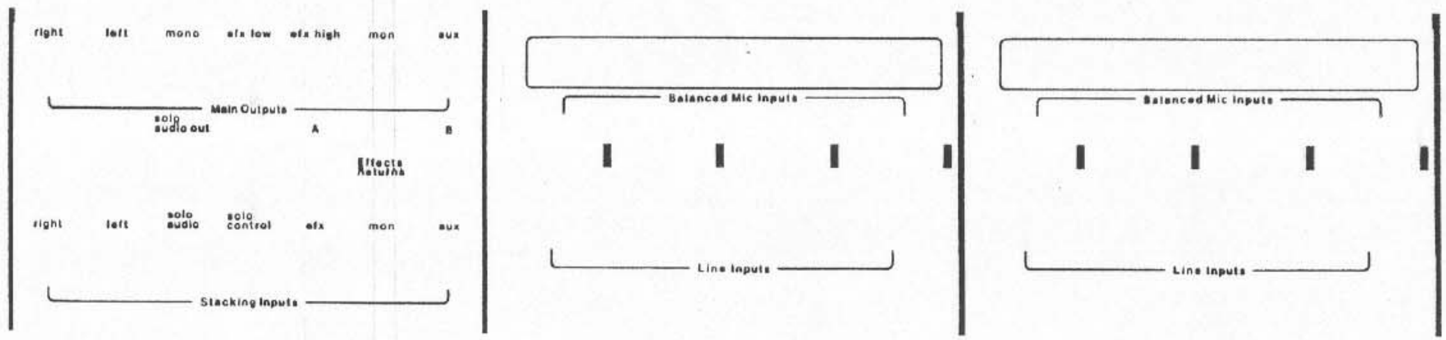
Redmond, WA 98052



FRONT PANEL LAYOUT

REAR PANEL LAYOUT



PANJO SERIES 72 SPECIFICATIONS

INPUTS	12, 16, 24
OUTPUTS	2x1
EQ	3-knob/4-frequency
FREQUENCY RESPONSE $\pm$1dB	20Hz - 20kHz
TOTAL HARMONIC DISTORTION	.02%
INTERMODULATION DISTORTION	.02%
EQUIVALENT INPUT NOISE	-130dBV
SIGNAL TO NOISE (Ref. to clipping output noise)	100dB
MAXIMUM MIC INPUT LEVEL	+6dBu
MAXIMUM LINE INPUT LEVEL	+24dBu
MAXIMUM PREAMP GAIN	50dB
MAXIMUM TOTAL GAIN	84dB
MAXIMUM OUTPUT LEVEL (Main channel)	9V 5kΩ +18dBm
NOMINAL OUTPUT LEVEL (Send busses)	+4dBu
NOMINAL INPUT LEVEL (Returns)	+4dBu
DIMENSIONS	W - 21.5", 26.5", 36.5" D - 15.5" H - 8.25"
POWER REQUIREMENTS	120V AC
SHIPPING WEIGHT (lbs)	36, 43, 58

ADDENDUM #1 - SERIES 72 and 74

A.D.R.

ADJUSTABLE DECAY REVERB

Your Series 72 or 74 mixer may have been equipped with ADR (Adjustable Decay Reverb), an exclusive TAPCO feature which allows you to vary the reverberation decay time as well as the amount of reverb. Now you can vary the decay time of the reverb by simply turning a knob.

The ADR is easy to operate. In ADR equipped mixers, the reverb return control is effects return A. As with any other effect, the return A level control will make the reverb louder or softer, the pan control controls direction or stereo placement, the decay time control varies the decay time, and the 'to monitors' control allows you to add reverb to the monitor send. The decay time control is located in the rear of the mixer.

Set up your reverb mix as usual on the mixers effects send. You can monitor levels by soloing the effects send and observing the levels on the solo VU meter. Zero level peaks seen here will give you the proper drive level.

The decay time control is used to tailor the decay time of the internal reverb to more closely suit musical content and the listening or performing environment. Shorter decay times can be used to keep the reverb from "getting in the way" on faster up-tempo material while the longer times can be used for effect on a solo instrument. On vocals let the music dictate your choice of reverb time. Larger rooms might benefit from a shorter reverb time than smaller ones and so on.

Changing the reverb decay time is a somewhat subtle effect. Don't expect the sound to change from that of a very small live room to that of the Taj Mahal. Plug a single microphone into the mixer and set it up to have reverb. Listen to this one microphone while you or someone else talks into it and then vary the decay time control. Listen for the change in the length of time that the reverb hangs on after a word or between syllables. That's ADR.

The mixer effects send is still available on the rear panel. It can be used for additional effects that are driven from the effects bus. It can be used simultaneously with the reverb.

If you desire to defeat the ADR, bring the external effects back into the mixer using effects return A. Remember, if you need more effects returns than A and B, you can use channel line inputs or the stacking jacks for additional effects returns.

ADDENDUM #2 – SERIES 72 and 74

PRE FADER SENDS

The pre fader sends (monitor and ½ aux) in the Series 72 and 74 mixers may be modified so either or both sends are pre fader and pre EQ. The factory supplied configuration is pre fader, post EQ.

Modification is a matter of moving 1 or 2 jumpers per input circuit board. This is a job for a qualified service technician. Consult your dealer, nearest service station or the factory for help in accomplishing this. The cost of the modification is not covered by the warranty.

SOLO (Units after serial 021XXXX)

The solo button is post fader, post EQ. This allows monitoring of multiple sources in the same musical balance that they are in the main mix. For some applications, the prefered configuration is PRE fader, post EQ.

This may be accomplished by moving 1 jumper on each input circuit board. It is a job for a qualified technician. Consult your dealer, nearest service station or the factory for help in accomplishing this. The cost of the modification is not covered by the warranty.