EV TAPCO

# SERIES 74

## SUBGROUP MIXING CONSOLE

OWNER'S MANUAL

a gulton company

DESCRIPTION

Line Inputs, Mic Inputs, Channel Patching and Outputs, Peak Indicator and Trim Control 2

Aux, Monitor, Effects 3

Equalization Controls, Channel Assign Switches, Channel Fader 4

OUTPUT SECTION CONTROLS, CONNECTORS

Sub-Groups 4

Talkback Section, Stereo Output Section, Effects Returns 5

Solo, Headphone Level 6

APPLICATIONS

Use of Sub-Groups 7

Using the Patching Connectors and Direct Outputs, Inputs, Outputs and Levels 8

Stereo and Mono 9

Phantom Power 10

Standard Patches
Mono PA, The PA Feed, Effects 12

Monitors, Aux Bus, Simultaneous Recording 13

Some Other Ways To Do It, Stereo PA, Sub-Group Assignments 14

Monitor and Aux Bus, Pre-Fader Sends, Other Options 15

Adjustable Decay Reverb 16

7400 SERIES BLOCK DIAGRAM 17

# 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 +26dBV. Minimum input level to set a nominal channel send level of -3dBV is 55mV, or -23dBV. The input impedance is 33k 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 1.5V, or +6dBV. Pin 3 is "hot". Phantom powering applies 24V DC to pins 2 and 3, referred to ground. Any low impedance balanced 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 normalized, 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 after 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. In a pinch, 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 inputs. The Peak indicator LED lights when an overload condition occurs, in either 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 Return 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 ±18dB shelving type bass and treble controls, and a switchable frequency, ±12dB 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 at the extremes of its 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 ASSIGN SWITCHES, PAN POT

The Channel Assign Switches and the Pan Pot work together to route the signals from the input channels to the sub-groups. Any input channel may be fed to any or all submasters by using the proper combination of switches and Pan Pot positioning.

When the upper switch, marked 1-2 is depressed, the Pan Pot will swing the signal from that channel between sub-group one (or left) and sub-group two (right). Likewise, when the lower switch is depressed, the Pan Pot will place the signals between sub-groups 3 and 4. With the Pan Pot centered and all switches depressed, the channel will appear on all four subs. Pan Pot fully clockwise, the channel appears only on groups 2 and 4, and so on. The only thing you can't do is directly route an input channel to subgroups 2 and 3, or 1 and 4.

## CHANNEL FADER

The Slide Fader controls the output level of the channel as it's fed to the subgroups. 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.

# OUTPUT SECTION CONTROLS, CONNECTORS

## SUB-GROUPS

The four Sub-group level controls determine the overall loudness of their respective group. The control system used here is active instead of passive, so the best noise performance and headroom are preserved throughout the system. The sub-group level controls are normalized at the "12" mark, and should be run as close to that as possible.

Each sub-group has a pan pot that controls its placement in the stereo mix. The sub-group solo switch allows the engineer to monitor the output of that submaster in the phones. Sub-group levels are indicated by the fluorescent bar graph display at the top of the module. The display is peak-weighted-average-responding which allows a more accurate interpretation of real signal levels than does an ordinary average responding display. The four pan pots are then mixed to stereo and then sent to the LEFT and RIGHT stereo master controls.

The Stereo Tape Input is also found in the submaster section. This is a line level stereo input that mixes to the L and R stereo masters. A level and balance control is provided. This input can (as the name implies) be used for an external stereo source, like a tape machine, or as another aux input, like a feed from an external stereo mixer. Levels from -30dBV to +20dBV can be accommodated.

## TALKBACK SECTION

The Talkback section allows the engineer at the console to communicate with those on stage, or to the house. The overall level of the talkback mic is controlled by the Level Control and three push switches control the routing of the talkback signal. The choices are:

1. Mains. Talkback routed to L and R stereo busses before the master. Because the mono output is a derived (L+R) output, it will have the talkback signal on it simultaneously.
2. Monitor. Talkback routed to the monitor bus, before the master.
3. Aux. Talkback routed to the aux bus, before the master.

The talkback input will accept any low impedance microphone. The microphone should be wired for a normal balanced input. (Pin 1 to 1, 2 to 2 and 3 to 3.)

It is not possible to talk to the Effects Bus or the Mono Output independent of the Stereo Output or vice versa.

## STEREO OUTPUT SECTION

The Stereo Output section is where everything comes together. The L and R Stereo Master controls the overall level of the main mix. They also affect the mono mix as it is a composite of left plus right. The various functions that come together at the stereo masters are:

1. The four sub groups.
2. The effects returns.
3. Talkback.
4. Stereo tape input.

The output of the Stereo Master controls then go to several different places:

1. The left and right main outputs.
2. The mono output.
3. The left and right Output Level Meters (fluorescent bar graph in 74 Series).
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 have a gain control and a solo switch. They may be metered by selecting solo for that bus and depressing the meter select switch. On sub-masters 1 through 4 the solo pick off point is post fader and before the patch point; levels will always reflect the fader setting.

## EFFECTS RETURNS

There are two Effects Returns provided. Each has 1 input (guitar level to line level), a level control, pan pot to the stereo 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 internal reverb option 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. At normal operating levels approximately OVU peaks will be seen if all the controls on the input channel, or in the particular bus that's 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 "O" 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 prioriety over normal signals.

### 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 preferred 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.

The rear panel Solo Stacking Connectors are used when stacking with another TAPCO mixer with an internal solo system (other 7400 Series 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 Sub-groups for a mono mix. This can allow sub-group pan assignments that might facilitate some other use, for instance, simultaneous recording and PA.

## APPLICATIONS

### USE OF SUB-GROUPS

The Sub-Groups are used to divide the mix into smaller, more managable parts. These parts are then mixed, as groups rather than as individual inputs, to form the final composite mix.

As an example, take the following set-up for a medium sized group doing mono PA. Input signals are run through individual input channels (one through twelve, in this case), then assigned to sub-groups. (We're assuming that keyboards are pre-mixed by the keyboard player, then fed to a line input.)

| INSTRUMENT | INPUT CH. | SUB-GROUP | CHANNEL ASSIGN | PAN |
|---|---|---|---|---|
| Base | 1 | 1 | 1-2 | L |
| Kick | 2 | 1 | 1-2 | L |
| Snare/hat | 3 | 1 | 1-2 | L |
| Rack toms | 4 | 1 | 1-2 | L |
| Floor toms | 5 | 1 | 1-2 | L |
| Elec. guit. | 6 | 2 | 1-2 | R |
| Rhy. guit. | 7 | 2 | 1-2 | R |
| Keyboards | 8 | 3 | 3-4 | L |
| Lead vocal | 9 | 4 | 3-4 | R |
| Vocal (guit.) | 10 | 4 | 3-4 | R |
| Vocal (key) | 11 | 4 | 3-4 | R |
| Vocal (bass.) | 12 | 4 | 3-4 | R |

These channel assignments give sub-group control over the four major components in the mix (in this case; Bass and Drums, Guitars, Keyboards, Vocals), making your job a lot easier once the mix within each sub-group is established. The sub-groups are mixed together, through the stereo master control, to form the final composite mix at the mono output.

Now that you have your 4 sub-group assignments, you can use the pan pots to place the 4 sources in the stereo field. Remember that this is really 4 channel mono as a mono source is always a mono source. (See later section on stereo and mono.) If you're doing stereo PA or recording, a more pleasing or realistic sound may be obtained by using two of the sub-groups to make up a stereo sub-group. This is done by panning one of the subs hard left and the other hard right. Any panning for this group is done at the inputs. In this way, you have the same number of apparent positions possible in the stereo mix as you have inputs assigned to the stereo sub-group. The remaining two subs can be another stereo sub-group or 2 mono sub-groups.

Using the patching connectors provided, each sub-group may now be processed (compressed, limited, flanged, phased, gated, expanded...) as a group, so you don't have to have one of every kind of signal processor for each input channel. Besides making the mixing job easier and saving you all that money you could have spent on processing gear, four bus subgrouping allows you to use your four track machine to record the performance. Of course, you can always record off the stereo outputs, but the level of flexibility and control you retain with multi-tracking is obvious.

## USING THE PATCHING CONNECTORS AND DIRECT OUTPUTS

The 7400 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 sub-group 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 setting 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 (bracking the normal, so the signals are not routed through the processor), and everything should be close to optimal operation.

It's important to note that any and all of the patching jacks may be used to provide, or accept, signals from other sections of the mixer. Just a little clever re-routing can provide additional sub-groups, stereo sub-groups, instantly switchable EQ, more returns and sends, etc. There's essentially no way to cause damage to any of the 7400 Series circuitry by inter-patching, so don't be afraid to try it. Just don't plug the output of a power amplifier into any of the mixer's jacks......please.

## 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 type 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 impedance on the order of ten times the actual output impedance of the circuit. All outputs on the 7400 Series products will deliver +20dBm (7.8 Vrms) before clipping (input channel and submaster patch points (Sends) included).

## 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 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 7400 Series mixer is equipped with a +48 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. Any 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 through a current limiting network.

The following microphones are "T" systems 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, U 47, U 67, M 49, M 269 |
| 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.), PL series (except PL 77, which is battery or phantom power), 600 series low impedance |
| Shure: | SM series, 500 series wired low impedance, 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, do require power for operation and will work in a +48V phantom power system.

| | |
|---|---|
| EV: | PL77 series, 1777, 1778, CS15 series, PL76 (when used with 506 adaptor), CO90 when used with CO9PM |
| AKG: | All C series (C451E, C452E, C412, C414, etc.), CE series |
| Beyer: | CV710 series, CV720 series with modification |
| Neuman: | KM83, KM84, KMS85, KM86, KM88, U47 FET, U87, U89 |
| Sony: | ECM series except ECM220, C series (C37P, C38, etc.) |

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 +48 V 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 in the table above can be used with the 7400 Series mixer, but they require their own power supplies to be used. However, still 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, to the sub-groups, 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 - 16. The Assign Switches and Pan Pots are used to route the individual channels to the sub-groups. Since this is a mono mix, the pan pots will be either full left or full right, so the assignments will be to only one sub-group. The four subs are internally mixed into the stereo bus, via the submaster pan pots. After passing through the left and right Master level control and the left and right busses are combined to and from the mono signal, which is available at the Mono Output through the Mono 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 Effects 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 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 will 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 signals by themselves, on headphones with the gain turned up. With the input channels, the Effects Return Pan Pot setting is not important because this is a mono setup.

## MONITORS

The Monitor Bus is used to generate a separate mix for stage monitors, when using 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

There are several ways to do mono, stereo, multi-track recording at the same time as mono PA.

A 4 track tape machine may be connected directly to the Sub-group Outputs. This will put four discrete tracks on the tape. Later the four tracks may be mixed down to stereo or mono, through the mixer's line inputs.

With a stereo recorder just connect the machine's line inputs to the Stereo Left and Right outputs. The two tracks will be composed of the four sub-groups, positioned in the mix as they are panned. A more convincing stereo effect may be had by using two of the sub-groups for a stereo pair and using the channel pan pot. If you are doing mono PA, and the independence of the sub-groups is not that important, 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.

To check the PA mix, a mono recording may be made by Y-cording the Mono output to the line input of the tape machine.

## 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.

The easiest way to get a stereo recording is to run the Stereo Outputs to the tape machine. Set the Pan Pots for a pleasing stereophonic balance. Since the Pan Pots are not hard left or hard right, the signals from some inputs will appear on more than one sub-group. Some of the flexibility available from the sub-group system will be lost. To retain some sub mixing capability a stereo pair may be created on two of the subs; composed of the rhythm section for instance, while the other two subs might be used for leads and vocals. To get the vocals and leads to appear centered in the stereo mix (on both sides, that is) cross patch their Sent to the stacking input of the sub-group that will appear on the opposite side of the stereo composite. So, if the rhythm stack stereo pair were on subs 3 and 4, and vocals were on 1, with leads on 2, patch sub 1 Send (Direct Out) to sub 4 Stacking, and sub 2 Send to sub 3 Stacking.

As you can see, routing the signals this way puts subs 1 and 2 on both sides of the stereo mix, so they will appear centered. At the same time a good sense of stereo perspective can be created with the rhythm tracks. This track assignment technique could be reversed so the vocals would be in stereo if the music were more vocally oriented.

## 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.

## SUB-GROUP ASSIGNMENTS

In any case, care must be taken when any input channel is panned to the extreme. With most stereo PA's half of the audience will not be able to hear any sound that is placed in the stereo image furthest away from their listening position. The negative effects of stereo PA may be greatly reduced by limiting the range of the Pan Pots and the available gain before feedback of the system may be increased with clever panning arrangements, resulting in a cleaner overall sound. Even when more gain is not needed, the reduced mic to mic and speaker to mic leakage will tend to make the total mix sound better.

## 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 that PA mixes often don't sound so good on tape. One, 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 recorded sound is unacceptable, 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 Outputs 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.

## 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 <u>and</u> 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.

## OTHER OPTIONS

The options for producing a stereo recording while doing stereo PA are essentially the same as for stereo recording with mono PA. A PA Mix may be set up on two of the Sub-groups, while two others are used for recording. Dummy plugs may be used to keep certain subs out of the PA, and so on. Again, don't hesitate to try something that may at first seem strange - if it gets you where you want to go, do it.

## ADJUSTABLE DECAY REVERB

Your 7400 Series 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.

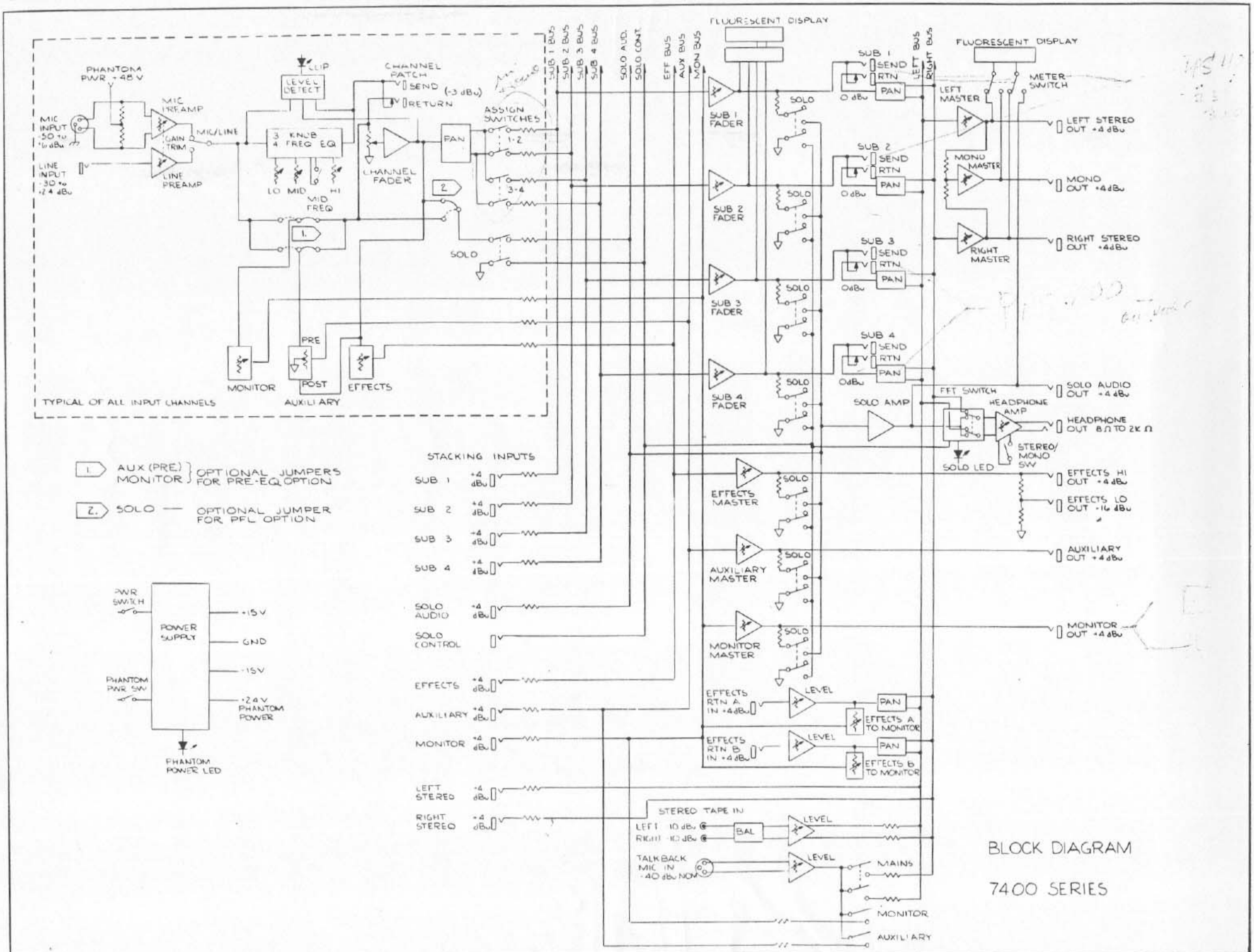
Set up your reverb mix as usual on the mixers effects send. You can monitor levels by soloing the effects send and observing the levels on the solo VU meter. Zero level peaks seen here will give you the proper drive level.

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.

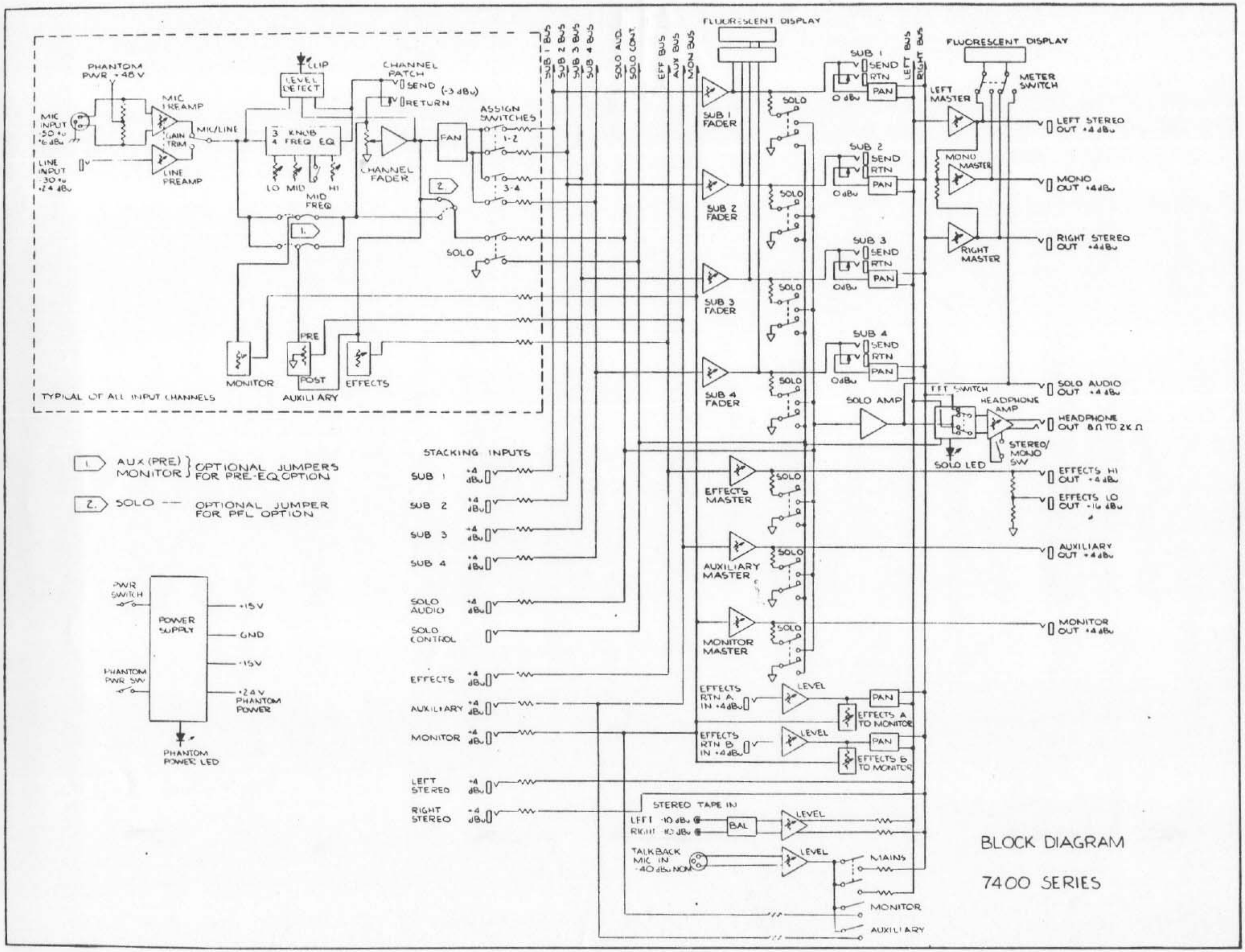
The mixer effects send is still available on the rear panel. It can be used for additional effects that are driven from the effects bus. It can be used simultaneously with the reverb. The additional effects devices can return to the mixer via effects return B, a channel line input or the stacking jacks.

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.




BLOCK DIAGRAM

7400 SERIES



PHANTOM PWR +48V
MIC PREAMP
MIC INPUT -50 to +6 dBu
LINE INPUT -30 to +24 dBu
GAIN TRIM
LINE PREAMP
MIC/LINE
CLIP
LEVEL DETECT
3 KNOB 4 FREQ EQ
LO MID HI
MID FREQ
CHANNEL PATCH
SEND (-3 dBu)
RETURN
CHANNEL FADER
PAN
ASSIGN SWITCHES
1-2
3-4
SOLO
MONITOR
PRE
POST
AUXILIARY
EFFECTS
TYPICAL OF ALL INPUT CHANNELS
SUB 1 BUS
SUB 2 BUS
SUB 3 BUS
SUB 4 BUS
SOLO AUD.
SOLO CONT.
EFF BUS
AUX BUS
MON BUS
FLUORESCENT DISPLAY
SUB 1 FADER
SUB 2 FADER
SUB 3 FADER
SUB 4 FADER
SUB 1
SUB 2
SUB 3
SUB 4
SEND
RTN
0 dBu
LEFT BUS
RIGHT BUS
LEFT MASTER
MONO MASTER
RIGHT MASTER
METER SWITCH
LEFT STEREO OUT +4 dBu
MONO OUT +4 dBu
RIGHT STEREO OUT +4 dBu
SOLO AMP
TFT SWITCH
HEADPHONE AMP
SOLO LED
STEREO/MONO SW
SOLO AUDIO OUT +4 dBu
HEADPHONE OUT 8 Ω TO 2K Ω
EFFECTS HI OUT +4 dBu
EFFECTS LO OUT -16 dBu
AUXILIARY OUT +4 dBu
MONITOR OUT +4 dBu
EFFECTS MASTER
AUXILIARY MASTER
MONITOR MASTER
EFFECTS RTN A IN +4 dBu
EFFECTS RTN B IN +4 dBu
LEVEL
EFFECTS A TO MONITOR
EFFECTS B TO MONITOR
STACKING INPUTS
SUB 1 +4 dBu
SUB 2 +4 dBu
SUB 3 +4 dBu
SUB 4 +4 dBu
SOLO AUDIO +4 dBu
SOLO CONTROL
EFFECTS +4 dBu
AUXILIARY +4 dBu
MONITOR +4 dBu
LEFT STEREO +4 dBu
RIGHT STEREO +4 dBu
STEREO TAPE IN
LEFT -10 dBu
RIGHT -10 dBu
BAL
TALKBACK MIC IN -40 dBu NOM
MAINS
MONITOR
AUXILIARY
1. AUX (PRE) MONITOR } OPTIONAL JUMPERS FOR PRE-EQ OPTION
2. SOLO — OPTIONAL JUMPER FOR PFL OPTION
PWR SWITCH
PHANTOM PWR SW
POWER SUPPLY
+15V
GND
-15V
+24V PHANTOM POWER
PHANTOM POWER LED
BLOCK DIAGRAM
7400 SERIES

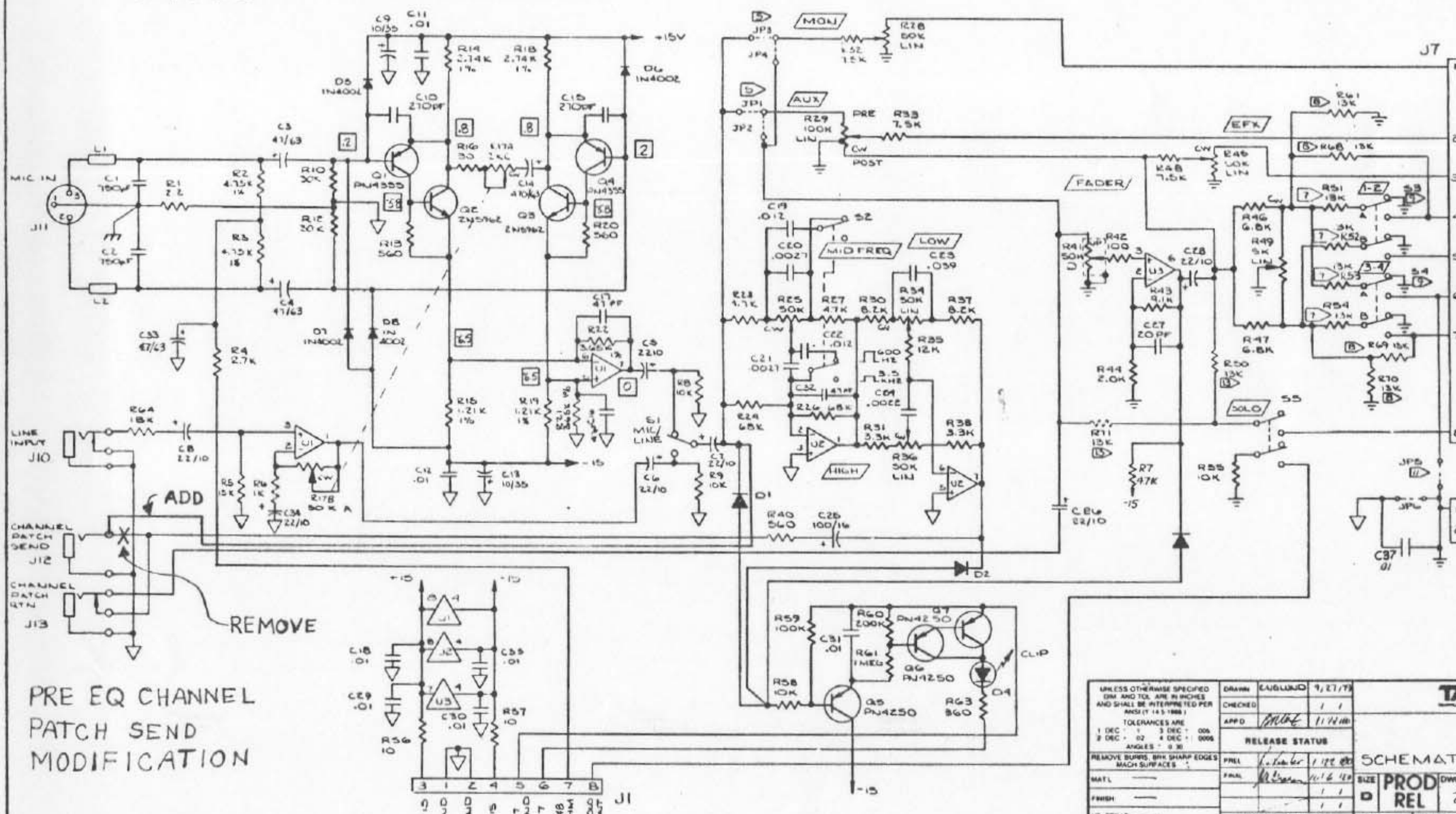
NOTES

1. [illegible] ARE LF355 [illegible] LF351
2. ALL RESISTORS ARE CARBON FILM, 1/4 W, 5 %, EXCEPT AS NOTED
3. L1 & L2 ARE FERRITE BEADS (FAIR RITE [illegible] 274-[illegible]-22)
4. ALL LYTICS ARE RADIAL
5. JP2&4 ARE FACTORY INSTALLED MAY BE MOVED TO JP1 & JP3 POSITIONS TO SELECT PRE EQ PRE FADER FOR MON & AUX SENDS
6. LAST REF DES USED: C36, D4, R[illegible], U3, L2, JP4, Q7, S5, J13

   REF DES NOT USED:

7. R51, 52, 53, 54, 800110 ONLY
8. R67, 68, 69, 70 800101 ONLY
9. S3 & 4 800110 ONLY
10. 1% RESISTORS ARE 1/4 W MF
11. JP5 800120 ONLY
12. J8 800126 ONLY
13. R50 IS FACTORY INSTALLED, R71 MAY BE FIELD INSTALLED INSTEAD TO PROVIDE A PRE FADER SOLO.

| REVISIONS | | | | |
|---|---|---|---|---|
| LTR | DESCRIPTION | BY | DATE | APPROVED |
| A | REVISED MIC/LINE CIRCUIT | | | |
| B | UPDATED SCHEMATIC | [illegible] | [illegible] | |
| C | UPDATED FOR PILOT RUN | [illegible] | [illegible] | |
| D | [illegible] | [illegible] | [illegible] | [illegible] |
| – | RELEASED TO PILOT RUN | [illegible] | [illegible] | [illegible] |
| E | R40 WAS 68 SEE ECN 223 | [illegible] | 4-3 | |
| F | ADDED BUS GND/48V [illegible] | [illegible] | [illegible] | |
| G | INCORP ECN 241 & 246 | [illegible] | [illegible] | |
| – | RELEASED TO PRODUCTION | DM | [illegible] | [illegible] |




PRE EQ CHANNEL
PATCH SEND
MODIFICATION

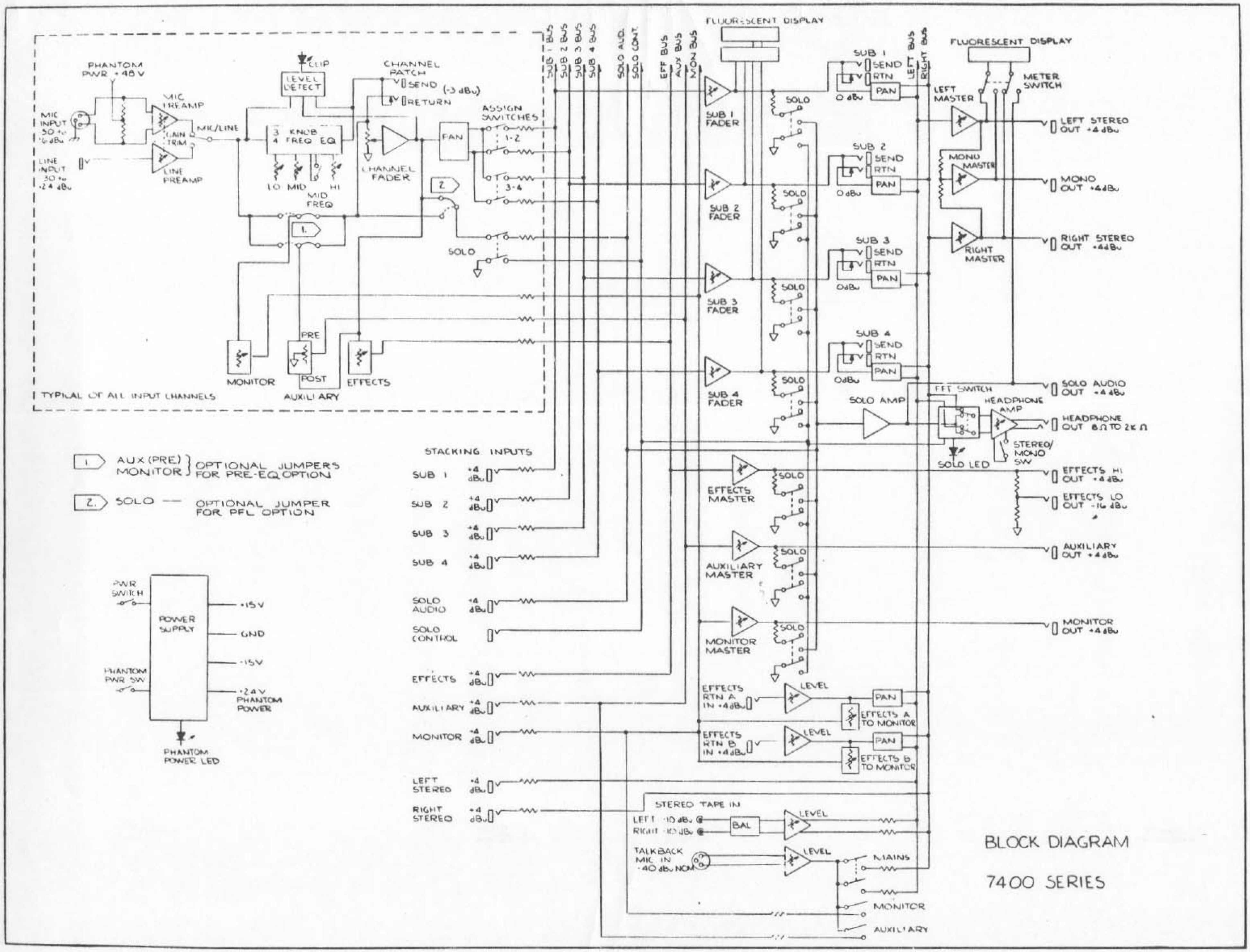
| | | | |
|---|---|---|---|
| UNLESS OTHERWISE SPECIFIED DIM AND TOL ARE IN INCHES AND SHALL BE INTERPRETED PER ANSI Y 14.5-1966 | DRAWN | EUGLUND | 9/27/79 |
| TOLERANCES ARE: 1 DEC ± .1, 2 DEC ± .02, 3 DEC ± .005, 4 DEC ± .0005, ANGLES ± 0 30 | CHECKED | | |
| REMOVE BURRS, BRK SHARP EDGES, MACH SURFACES | APPD | [illegible] | [illegible] |
| MAT'L | RELEASE STATUS | | |
| FINISH | PREL | [illegible] | [illegible] |
| HT TREAT | FINAL | [illegible] | [illegible] |
| | DO NOT SCALE DWG | | |

TAPCO

SCHEMATIC, INPUT

SIZE D | PROD REL | DWG NO 24859 | REV G

SCALE — | UNIT WT | SHEET 1 OF 1




BLOCK DIAGRAM

7400 SERIES