

TASCAM

TEAC Professional Division

M-106

Mixing Console



OWNER'S MANUAL

5700056601

The guarantee of performance that we provide for the 106 must have several restrictions. We say that the 106 will perform properly only if it is adjusted properly and the guarantee is that such adjustment is possible. However, we cannot guarantee your skill in adjustment or your technical comprehension of this manual. Therefore, setup is not covered by the Warranty. If your attempts at internal adjustment are unsuccessful, we must make a service charge to correct your mistakes.

Recording is an art as well as a science. A successful recording is often judged primarily on the quality of sound as art, and we obviously cannot guarantee that. A company that makes paint and brushes for artists cannot say that the paintings made with their products will be well received critically. The art is the province of the artist. TASCAM can make no guarantee that the 106 *by itself* will assure the quality of the recordings you make.

Your skill as a technician and your abilities as an artist will be significant factors in the results you achieve.

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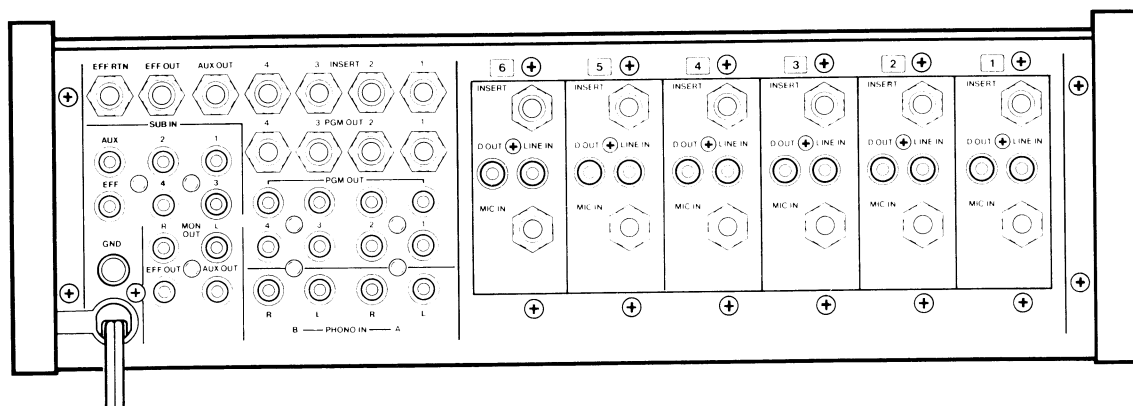
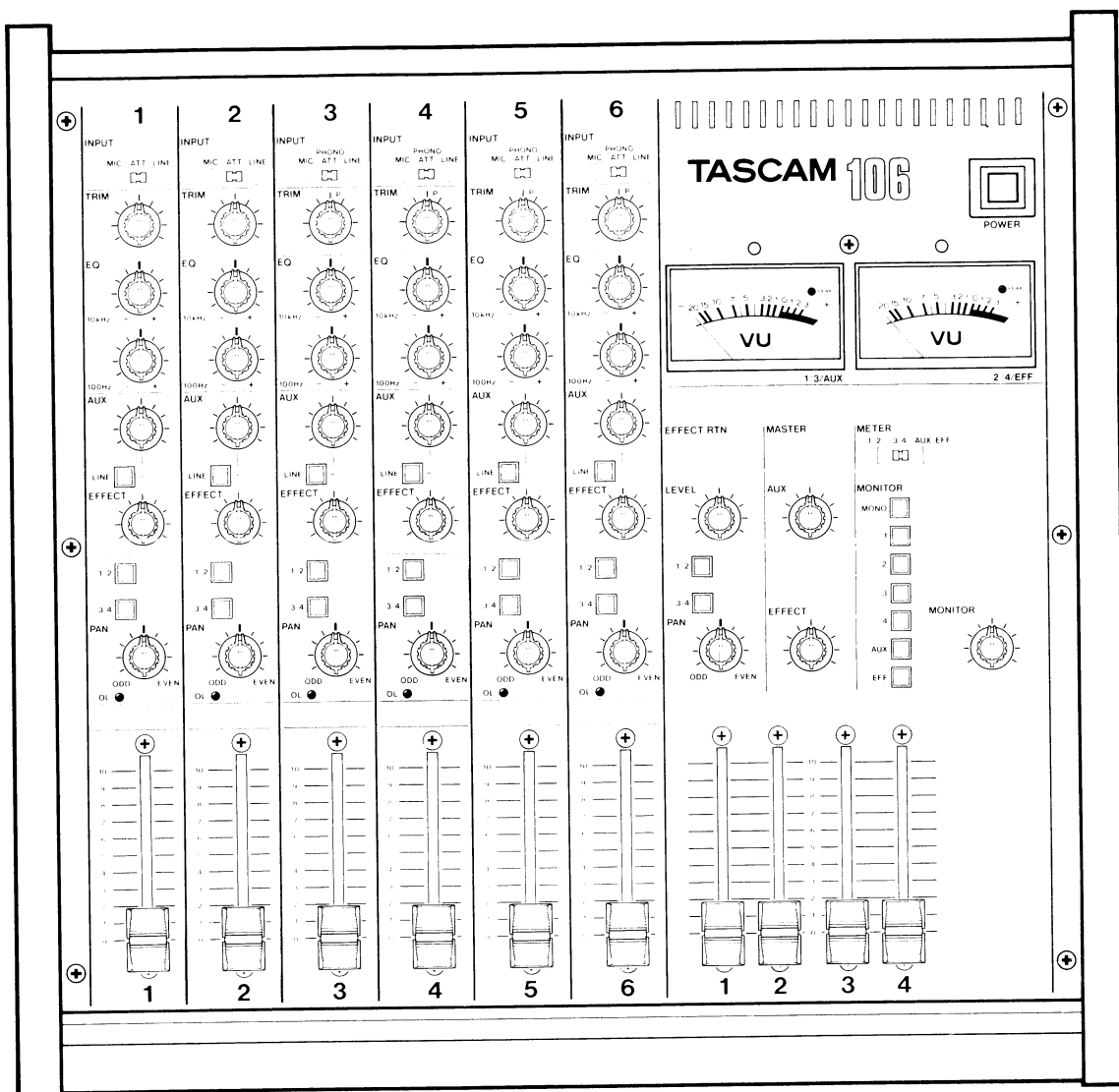
WARNING:
TO PREVENT FIRE OR SHOCK
HAZARD, DO NOT EXPOSE THIS
APPLIANCE TO RAIN OR
MOISTURE.

This apparatus has a serial number located on the rear panel. Please record the model number and serial number and retain them for your records.

Model number _____
Serial number _____

CAUTION RISK OF ELECTRIC SHOCK DO NOT OPEN	CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.
	The lightning flash with arrowhead symbol within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure, that may be of sufficient magnitude to constitute a risk of electric shock to persons.
	The exclamation point within an equilateral triangle is intended to alert the user of the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

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Introduction

Understanding what is going on inside your equipment will help improve your sound. Think of this manual as a reference handbook. You won't need all of what is here to begin, and it is certainly not necessary to memorize it, but do try to find the time to read it thoroughly at least once, that way you will be familiar with its contents and if you need answers they will be here waiting.

Good luck with your sound.

The TASCAM 106 mixing console was designed for maximum flexibility, without compromising performance in any way.

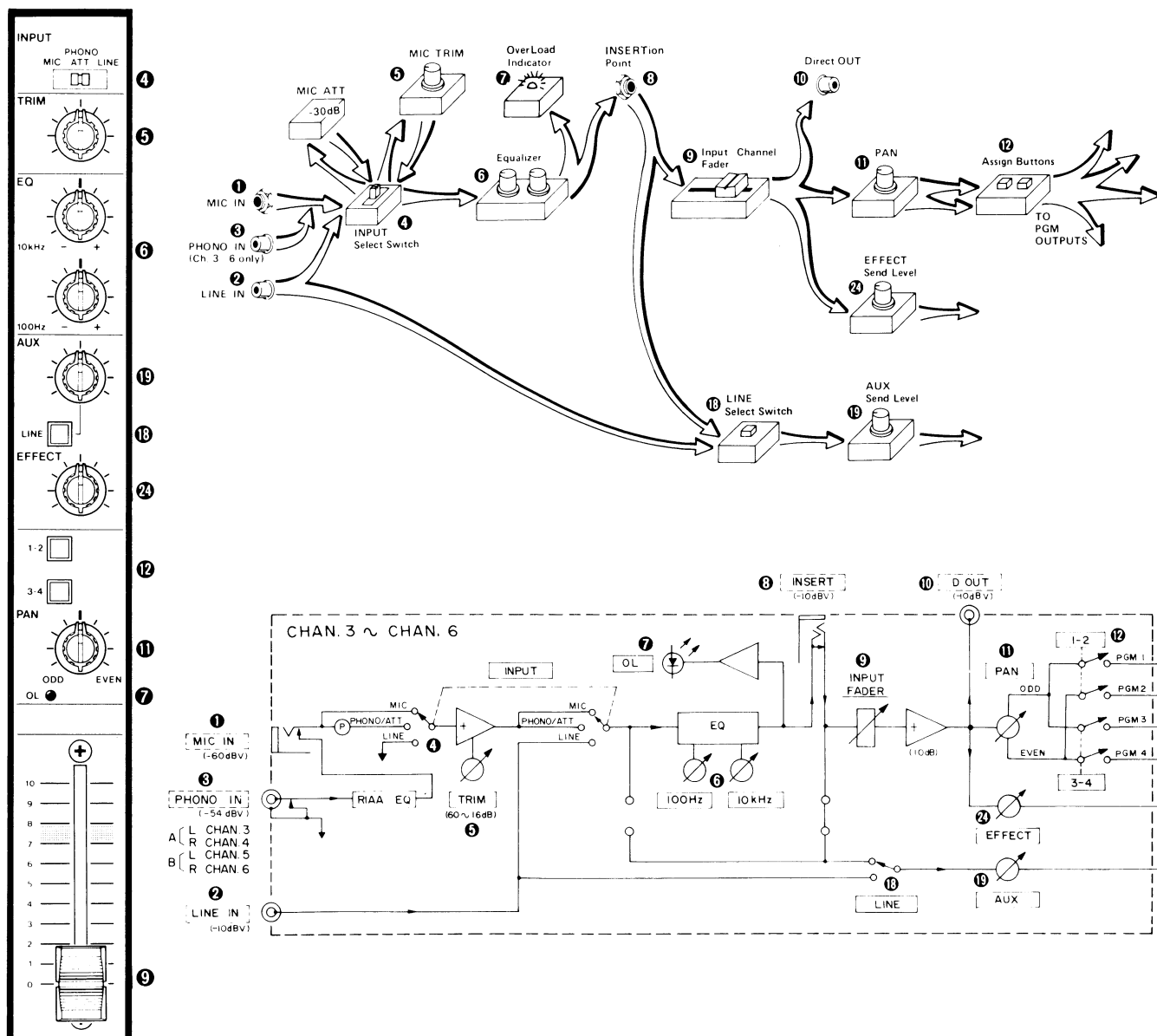
It is equipped with a flexible combination of functions so that it can be used in a wide range of applications, from small studio recording, to mixing in PA and disco systems and mixing audio for video productions.

A mixer's value and usefulness can be determined by its flexibility. The more flexible a mixer, the greater its value to you. This flexibility, however, can make the mixer hard to understand at first. With a little study and experience, you will find the 106 understandable and easy to use.

To understand the mixer and its capabilities, it is important to realize that the 106 is a mixer system comprised of 5 submix systems. The 5 submix systems in the 106 are designed to perform specific functions and yet still retain flexibility that makes the 106 adaptable to many applications. These submix systems are the MAIN MIX, the AUX SUBMIX, the EFFECT SUBMIX, the EFFECT RETURN (EFF RTN) SUBMIX, and the MONITOR SUBMIX systems.

Each submix system performs the same basic job; it takes signals into its inputs, performs a function (processes them), and sends them out its outputs.

These 3 illustrations will help you understand the 5 submix systems. The first is an illustration of the front panel controls. The second a pictogram; shows these controls laid out according to how they are wired up (their *Functional* sequence). Note, that the functional sequence is NOT the same as how the controls are laid out. The third, a simplified electrical sequence illustration called a *Block Diagram*, it is the same thing as Illustration 2, except it has a little more detail and it uses different symbols. Using the numbers on the illustrations, you can see how a control knob on the first drawing relates to a function on the second drawing, and a symbol on the third, this will help you learn to read the *Block Diagram*. This is a valuable skill that will help you get the most out of your 106. Experienced engineers can operate a mixer by using the block diagram alone.



Each function is explained in the Features and Controls section. Whenever possible, an explanation of when you would use it will be given.

The MAIN MIX system is the most complex of the five, and is capable of accepting a number of different types of signals at its inputs and processing them for a wide variety of applications. It is the primary mixing system. It is used to mix sound sources such as mics or tape playback and send that mix to tape recorder inputs, video recorder audio inputs, P.A. speaker systems, or disco speaker systems.

On channels 3 through 6 there is a PHONO INPUT. This allows the 106 to mix up to 2 stereo turntables (2 channels are required for each stereo signal). These inputs include RIAA phono pre-amplifiers, so you can connect your turntable directly to the 106. After processing, the signal is then assigned to an output buss or busses using the PAN pot and assign buttons.

The AUXiliary SUBMIX system takes its input signal from either the LINE IN or the MAIN MIX input channel. This is selectable by the LINE switch next to the AUX level control. The

signal from the input channel can come either pre-EQ (before the equalizer) or pre-fader (after the EQ and INSERTion Point), as selected by an internal jumper. The pre-EQ take-off point would be used if you wanted the AUX signal to be unaffected by the equalizer and any processing being done at the INSERTion Point. The post INSERTion Point take-off is used if you want the AUX signal to be affected by the equalizer and any device connected to the INSERTion Point. This is the most common use, so the 106 comes with the jumper wired in this position. The AUX SUBMIX system then combines all the AUX signals from each input channel and sends the composite signal to the AUX MASTER. This control allows you to adjust the level of the composite signal. It is then sent to the AUX OUT jacks. The AUX SUBMIX system can be used as a talent cue, monitor or tape cue system when overdubbing.

The LINE select switch allows you to feed the signal at the LINE INput to the AUX SUBMIX system independently of the INPUT select switch. This feature lets you use the AUX SUBMIX system as a tape cue system when overdubbing. It can also be used as a pre-fader effects send. "Pre-fader" means that when you change the level of the MAIN MIX linear fader on the input channel, the signal going to the AUX SUBMIX system does not follow the changes of the MAIN MIX fader.

The EFFECT SUBMIX system is used primarily for adding processing such as reverb or echo to the MAIN MIX. It can also be used as a cue submix if desired. It is important to note that the EFFECT signal is post-fader, meaning that any changes on the MAIN MIX linear fader will be reflected in the EFFECT signal level. You will not be able to get a completely independent mix. This is desirable in certain applications such as adding reverb to the MAIN MIX. Unlike the AUX SUBMIX, the EFFECT SUBMIX derives its input only from the MAIN MIX input channel.

The EFFECT RETURN (EFF RTN) submix system takes its input from the EFF RTN jack. It allows the signal level coming into it to be adjusted and assigned to the 4 output busses using its PAN control and assign buttons. This submix system is usually used as a way to bring the output of an effects device such as a reverb back into the MAIN MIX, hence the name

EFFECT RTN (Return). It can also be used as a line level input and can be used as additional input channels to the main mix.

The MONITOR SUBMIX system takes its inputs from the 4 output busses, the AUX OUTput and the EFFECT OUTput. Using the switch matrix, you are able to select which of the inputs you want to listen to. There is also a MONO switch that connects the 2 monitor channels together so that you can detect any problems that may arise when the stereo signals are reproduced in mono.

From the switch matrix, the selected signal goes to the MONITOR level control and then to the MON L and R output jacks, and the Headphones jack.

These submix systems are all interconnected and configured so as to provide a great amount of flexibility.

Features and Controls

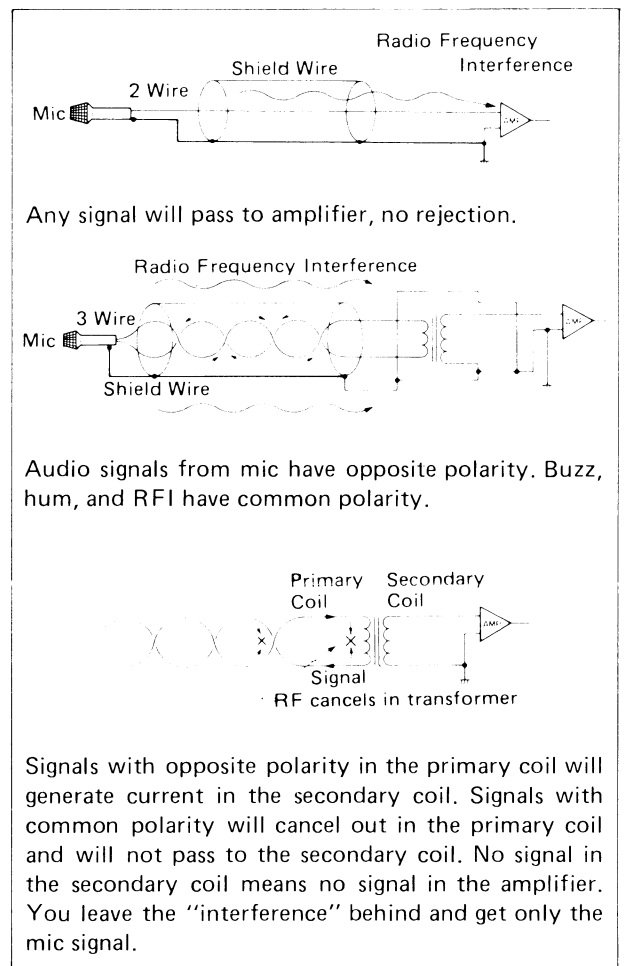
MAIN MIX SYSTEM

① MIC IN Jack

This 1/4" phone jack accepts unbalanced signals from any type of microphone having any impedance from 150 ohms to 10,000 ohms. You can also connect any magnetic instrument pick-up (electrical guitar or bass), an electronic keyboard or any line level source below 0.4 volts. There is no need for a "direct box" or transformer, but such devices may be used if desired.

NOTE: When anything is connected to the MIC IN jack on channels 3 – 6, the PHONO IN is automatically disconnected. If you are using balanced professional mics, use #109B adaptor to connect it to the MIC IN jack.

Why use a transformer if it is not absolutely necessary? There may be a good reason, and it applies primarily to mics used with cable runs exceeding 10 feet. The low power signal that mics (and some instrument pickups) generate must frequently be protected and isolated from other low power signals. Radio, power line hum, buzz, crackles and switching noise when motors start up (do you have an air conditioner on your AC line or maybe an old fridge?) – all these unwanted signals must be kept out of the very high gain amplifiers that are needed to raise the mic signal to a working level. The balanced or three-wire mic circuit and input isolation transformer are the only sure way to deal with the problem. Here's how it works:

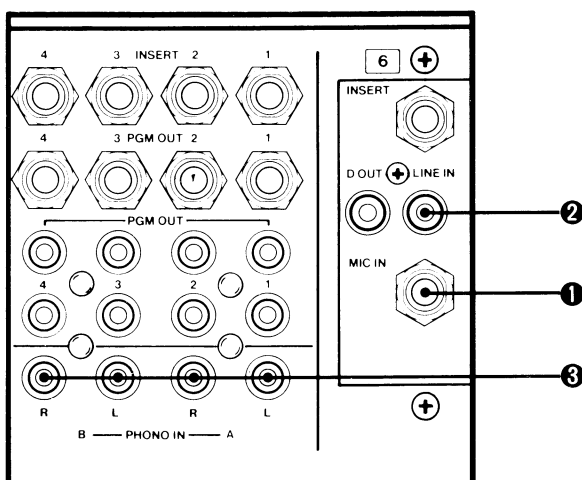


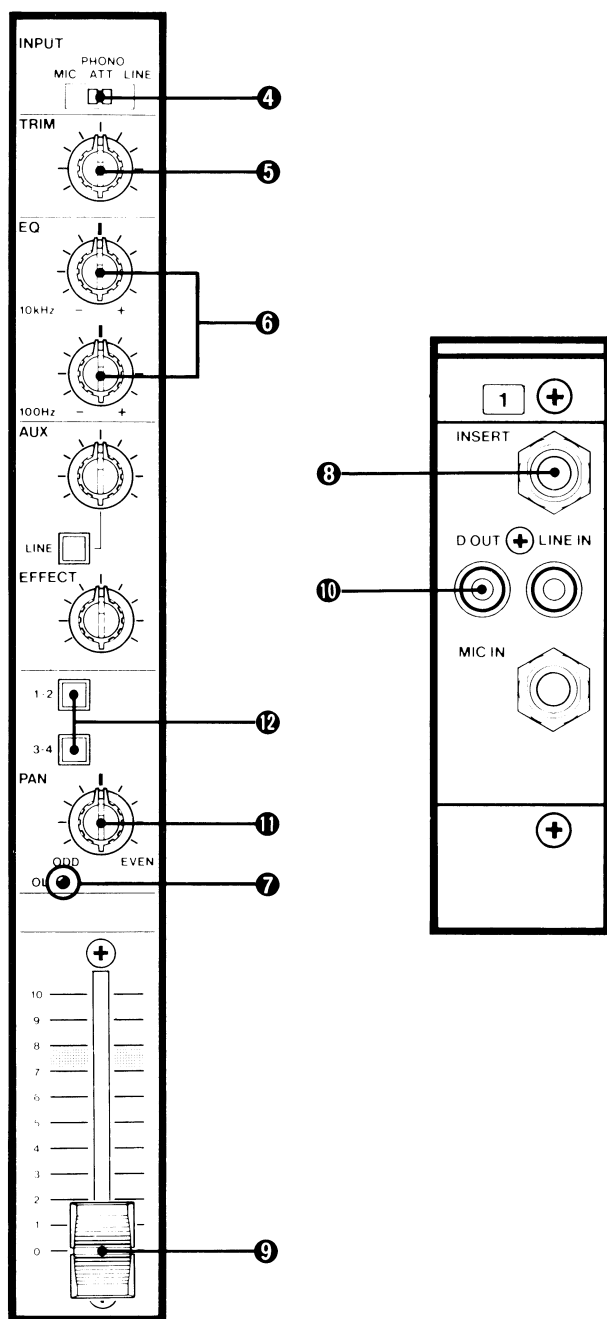
② LINE IN Jack

This RCA jack accepts line level (-10 dBV/0.316 volt) signals. From this jack the signal is fed to both the INPUT select switch and the LINE input select switch on the AUX SUBMIX system. This input can also be used for connecting electronic keyboards that have a line level output. The LINE IN jack bypasses the MIC IN amplifier and ATTenuator.

③ PHONO IN Jacks

These RCA jacks accept signals from phono cartridges and feed them through an RIAA phono pre-amplifier to the input channel via a connection in the MIC IN jack. Remember that when anything is connected to the MIC IN jack, the signal coming from the PHONO IN jack is disconnected.





4 INPUT Select Switch

Each of the six input channels has an INPUT select switch located at the top of its vertical row of controls. This switch selects which of the input signals will be routed through the input channel. In the LINE position, the signal coming from the LINE IN jack is selected. In the MIC position, the signal coming from the MIC IN jack is selected. With the switch in the ATT (ATTenuate) position, the signal from the MIC IN jack is selected and its level is dropped by 30 dB. On channels 3 – 6, this position also selects the PHONO IN jacks.

The electronic circuit that is used to attenuate the mic signal is called a PAD. This position should be used when the MIC IN signals make the OL (OverLoad) indicator light even when the TRIM control is turned down (counterclockwise).

5 Mic TRIM Control

The signal coming from the MIC IN jack goes through a high gain amplifier to bring it up to line level. This amplifier is called the Mic Preamp. The amount of increase or gain that the Mic Preamp gives the mic signal is determined by the TRIM control. The TRIM control allows you to adjust the Mic Preamp to handle a wide variety of mic levels. Turning the TRIM control clockwise causes the Mic Preamp to give more gain (level increase) when working with softer sound sources. Turning it counterclockwise reduces the amount of increase or gain when working with louder sound sources. This control affects only the MIC IN and PHONO IN signals. The "P" mark on the TRIM scale represents the nominal setting position for the PHONO IN signals.

6 Equalizer

The Equalizer, or EQ, is the circuitry that allows you to adjust the tonality of the signal going through the input channel. It is a two-knob type, with the top knob allowing a boost or cut of 15 dB at a 10 kHz for the high frequencies, and the bottom knob allowing a boost or cut of 15 dB at 100 Hz for the low frequencies. They work similarly to the bass and treble knobs on other audio equipment.

We've included, on page 22, a chart of the frequency characteristics of some musical instruments. So you can get a better idea how these tone controls can be used with the best artistic advantage. Of course, using them and hearing the results will tell you exactly how they work.

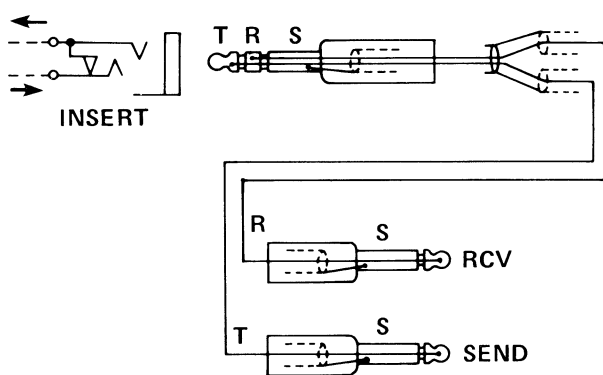
7 OL (OverLoad) Indicator

When signals high enough to make the INSERTION jack output exceed +15 dBV are applied to an input channel, this LED will light. If you are working in the MIC position of the input switch you should first adjust the TRIM. If the LED does not go out, switch the INPUT selector to ATTenuate to engage the 30 dB PAD and readjust the TRIM until the LED remains unlit or blinks briefly when high level signals are present. If, when recording extremely percussive

transient material the LED continues to flicker on strong peaks, changing MIC placement or using a less sensitive mic will help.

⑧ INSERTion Jack

This 3 conductor phone jack provides electronic access to the input channel signal path. This access allows you to add external signal processing to any of the input channels. As an example, if you want more equalization on a particular channel than provided, you can connect a graphic equalizer's input and output to the INSERTion jack on that channel. INSERTion refers to the idea of "inserting" or connecting a piece of equipment (such as the graphic equalizer) in the input channel's signal path of the 106. When there is nothing connected to this jack, there is an automatic internal connection, or "normal", that allows the signal to continue flowing through the mixer without interruption.



T — Tip, send signal
R — Ring, receive signal
S — Sleeve, ground

⑨ Input Channel Fader

This linear, or slide, fader varies the amount of signal going from the input channel to the Program (PGM) Output busses via the PAN control and assign buttons. This also affects the amount of signal going to the EFFECTs send level control. The channel fader is the main mixing control for adjusting how much of the input appears at the selected output(s). The input channel faders, as well the program master faders ⑭, are compatible with P&G fader #3020.

⑩ Direct OUT Jack

This jack provides an extra output on the input

channel so that its signal can be routed to outside equipment independently of the PGM Output busses. This is useful when the PGM Output busses are all being used for combined signal from the other input channels, and you want to get this one channel to one input, such as one track on a multitrack tape machine.

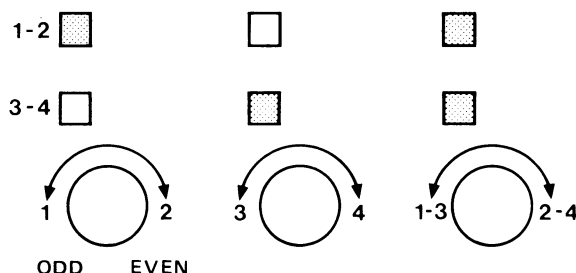
⑪ PAN Control

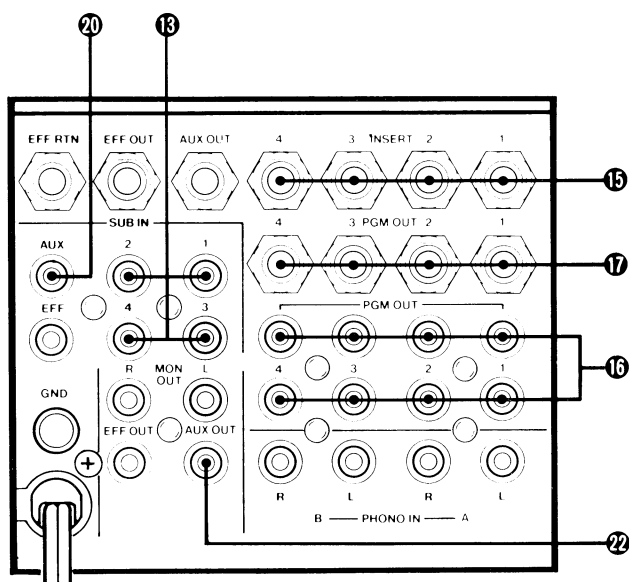
This control is used to assign the input channel's signal to the PGM Output busses. The PAN provides continuously variable assignment to the odd numbered PGM Output busses (full counterclockwise rotation) and the even numbered PGM Output busses (full clockwise rotation). This allows you to make stereo mixes and locate an input channel's signal anywhere in the stereo panorama.

⑫ Assign Buttons

The assign buttons work in conjunction with the PAN control to assign an input channel's signal to the PGM Output busses. When the top button is depressed, the full counterclockwise (left) position of the PAN control assigns the signal to PGM Output buss 1, and the full clockwise (right) position assigns it to PGM Output buss 2. Likewise, when the bottom button is depressed the full counterclockwise (left) position of the PAN control assigns the signal to PGM Output buss 3, and the full clockwise (right) position assigns it to PGM Output buss 4. Both buttons may be depressed at the same time to assign the signal to PGM Output busses 1 and 3 with PAN full left, and to PGM Output busses 2 and 4 with PAN full right.

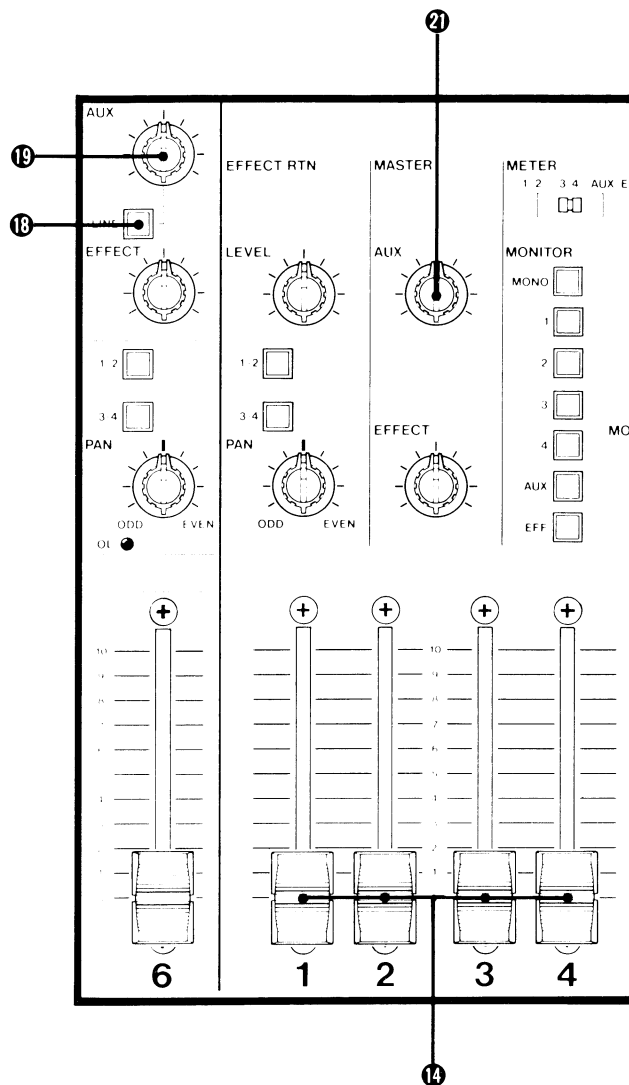
To assign an input channel to just one of the PGM Output busses, press the assign button that has the desired buss as part of its pair of busses and rotate the PAN control full left or full right to select the desired buss from the pair.



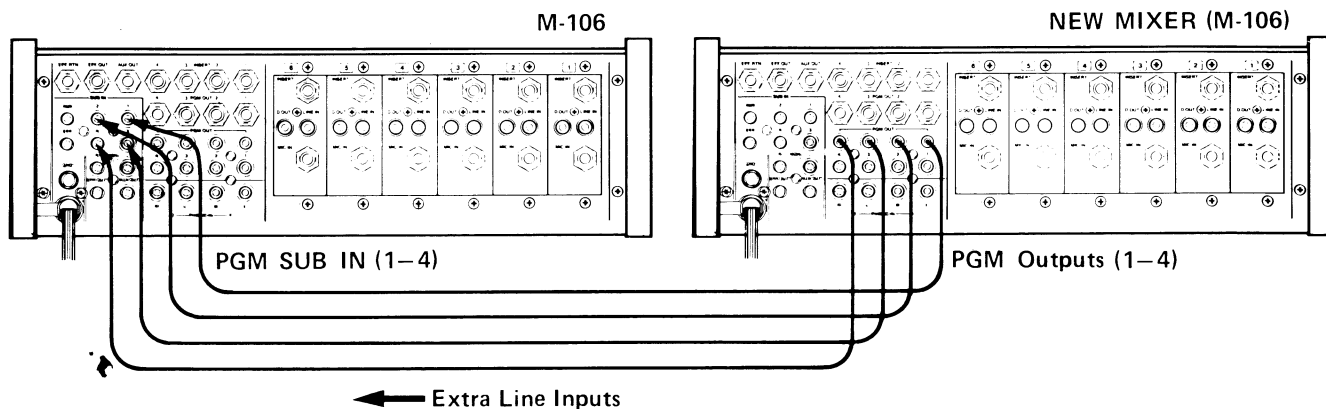


13 Program (PGM) SUB IN

The PGM SUB IN jacks (1 – 4) provide access to the PGM Output busses. The access point is located at the input of the PGM Output buss after all the signals assigned to that buss have been mixed together. This provides you a set of extra line inputs and the ability to “stack” mixers. If you use the PGM SUB INs as line inputs, note that there are no controls on these inputs. All level control, etc., must be done externally to the 106. If you want to expand the number of inputs on the 106, you can use the PGM SUB INs to connect the outputs of the two mixers together. To connect them together so that they function as a single mixer, connect the PGM Outputs of the new mixer to the SUB-INS of the 106. The PGM MASTERS on the new mixer will still control the channels assigned to it on that mixer, but the PGM MASTERS in the 106 will control all the levels of both mixers.



You can set the new mixer’s masters so that you only need to use the PGM MASTERS on the 106 and not have to adjust them again. See the section on calibration for information on how to do this.



⑭ Program (PGM) Master Faders

These 4 linear (slide) faders control the level of the mix of the signals assigned to the PGM busses 1 – 4. They simultaneously adjust the signal level sent to the:

1. PGM OUTput jacks
2. METER select switch
3. MONITOR switch matrix

⑮ PGM INSERTion Jacks

Like the input channel INSERT jack, the PGM INSERT jack provides electronic access to the PGM signal. You can insert signal processing such as a graphic equalizer or a compressor/limiter. Any processing added at this point will affect all the signals assigned to that particular PGM Output buss. When there is nothing connected to the INSERT jack, an automatic internal connection allows the signal to continue through the 106 uninterrupted.

For cable wiring, see page 11, INSERTion Jack diagram.

⑯ PGM OUT Jacks (RCA)

These jacks provide a low level signal, nominal -10 dBV (0.316 volt) unbalanced output for each PGM Output buss, for use with equipment with industry standard high sensitivity inputs (tape recorders, audio amplifiers, etc.). There are two jacks for each PGM Output buss. The extra jack is provided to allow connection to 2 inputs.

⑰ PGM OUT Jacks (1/4" Phone)

These jacks provide a nominal 0 dBu (0.775 volt) unbalanced output, high level signal for each PGM Output buss for use with equipment with industry standard low sensitivity inputs (public address, sound reinforcement amplifiers and broadcast equipment).

AUX SUBMIX SYSTEM

⑱ LINE Select Switch

This switch selects the input for the AUX send on each input channel. Depressing this switch selects the signal at the LINE Input jack independently of the setting on the input channel's INPUT switch. This special feature allows you to make a totally independent mix from the sources connected to the LINE IN jacks, such as

a headphone cue mix from the sync playback of a multitrack recorder. You can still use the rest of the input channel for mixing other inputs. In broadcast applications, this feature allows you to create a "mix-minus", and in a disco application, you can use this feature to cue records or tape machines independently of the MAIN MIX system.

With the LINE switch in the up position the signal after the INSERT jack on the MAIN MIX systems is selected. This is a PRE-FADER point. That means that any changes made with the EQ and at the INSERT jack will affect the signal going to the LINE switch on the AUX SUBMIX system, but any movement of the input channel's slide fader will NOT affect this signal's level.

⑲ AUX Send Level Control

This controls the amount of signal selected by the LINE select switch that is sent to the AUX MASTER control.

⑳ AUX SUB IN Jack

This jack provides access to the input of the AUX Submix MASTER. This access point is at the input of the AUX MASTER control. When "stacking" mixers, this jack allows you to operate all the AUX sends as a single system. Connect the output of the second mixer's Aux Submix system to the 106's AUX SUB IN jack. The second mixer's Aux Master can be set so that you never have to use it. See the section on calibrating the 106. This lets the two AUX Submix systems function as one. You then can use the 106 AUX MASTER to control the combined AUX System.

You can also use the AUX SUB IN as a separate input to the AUX MASTER to return such things as reverb sends so that you have reverb in the headphones. All level control, however, must be done on the device that is connected to the AUX SUB IN.

㉑ AUX MASTER Level Control

This controls the level of the AUX Submix going to the AUX out jacks.

㉒ AUX OUT Jack (RCA)

This jack provides a low level, nominal -10 dBV (0.316 volt) output for the AUX Submix system. The maximum level before clipping is +10 dBV (3.1 volts), and the lowest impedance that should be connected to this output is 9.6 k ohms.

23 AUX OUT Jack (1/4" Phone)

This jack provides a high level, nominal 0 dBu (0.775 volt) output for the AUX Submix system. The maximum level before clipping is +20.2 dBu (8 volts), and the lowest impedance that should be connected to this output is 2 k ohms.

EFFECT SUBMIX SYSTEM

24 EFFECT Send Level Control

This rotary control takes a post-eq, post-fader (after the equalizer and slide fader on the input channel) signal from the MAIN MIX system, and controls the amount of that signal going to the EFFECT MASTER control. Because signal to the EFFECT Submix is subject to any adjustment to the input channel fader, it is preferred for effects and echo mixes.

25 EFFECT SUB IN

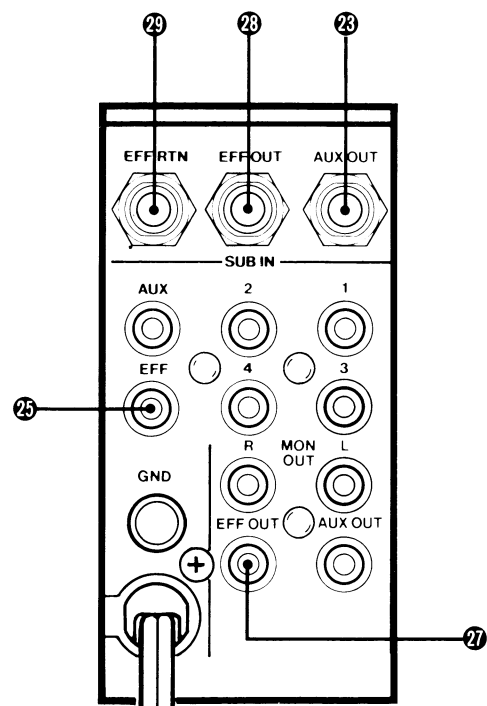
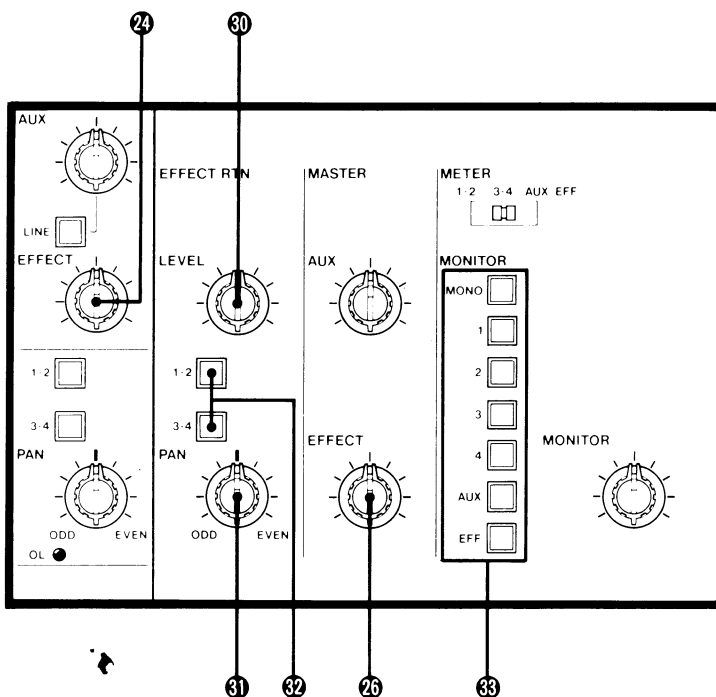
This jack provides access to the input of the EFFECT MASTER control. This access point is between the EFFECT send controls on the input channels and the EFFECT MASTER. When "stacking" mixers, this jack allows you to operate all the EFFECT Submixes as a single system. All you do is connect the EFFECT OUT

of the new mixer to the EFFECT SUB IN of the 106. The overall level of the new mixer's EFFECT Submix will be controlled by its EFFECT MASTER, and the 106's EFFECT MASTER will control the overall level of both its EFFECT Submix, and the level of the first mixer's EFFECT Submix. You can then have both mixer's EFFECT Submix signals at the 106's EFFECT OUT. See the section on calibration for information on how to set-up the new mixer so that you don't need to use its Effect Master.

You can also use the EFFECT SUB IN as a separate input to the EFFECT MASTER. DO NOT connect the output of a device to the EFFECT SUB IN if its input is connected to the 106's EFFECT OUT. Since doing this will create a loop and create feedback and distortion, all level control must be done externally to the 106 on the device that is connected to the EFFECT SUB IN.

26 EFFECT MASTER Control

This controls the level going to the EFFECT OUT jacks of the mix of the signals from all the EFFECT send level controls, and the EFFECT SUB IN.



27 EFFECT OUT Jack (RCA)

This jack provides a nominal -10 dBV (0.316 volt) unbalanced output for the EFFECT Submix system. The maximum level before clipping is +10 dBV (3.1 volts), and the lowest impedance that should be connected to this output is 9.6 k ohms.

28 EFFECT OUT Jack (1/4" Phone)

This jack provides a nominal 0 dBu (0.775 volts) unbalanced output for the EFFECT Submix system. The maximum level before clipping is +20.2 dBu (8 volts), and the lowest impedance that should be connected to this output is 2 k ohms.

EFFECT RETURN SUBMIX SYSTEM

29 EFFECT Return (RTN) Jack

This 1/4" phone jack accepts line level inputs. Its primary use is to return the output of a signal processor (such as a reverb) to the PGM Output busses. It can also be used as a line level input for other signal sources. Since this input has a PAN control and assign buttons, you can route signal to any of the PGM Output busses in the same way as the input channel.

30 EFFECT Return (RTN) LEVEL Control

This control adjusts the amount of signal from the EFF RTN jack that is sent to the EFFECT RTN routing circuitry, and subsequently added to the PGM Output busses.

31 EFFECT RTN PAN Control

Like the PAN controls on the input channels, this is part of the routing circuitry for the EFFECT RTN Submix system. The EFFECT RTN PAN provides continuously variable assignment to the odd numbered PGM Output busses (full counterclockwise rotation) and the even numbered busses (full clockwise rotation). This allows you to assign the EFFECT RTN signal to any point in the stereo spectrum.

32 EFFECT RTN Assign Buttons

These assign buttons are the rest of the EFFECT RTN routing circuitry. When the top button (1-2) is depressed the full left (counterclockwise) position is assigned to PGM Output buss 1, and the full right (clockwise) position is assigned to PGM Output buss 2. The signal can be routed to both channels in varying proportions by turning the PAN control to a position

between full left and full right. When the bottom button (3-4) is depressed the full left position is assigned to PGM Output buss 3, and the full right position is assigned to PGM Output buss 4. Both buttons may be depressed at the same time to assign the full left position to PGM Output busses 1 and 3, and the full right position to PGM Output busses 2 and 4.

To assign the EFFECT RTN signal to just one PGM Output buss, depress the assign button for the desired buss, rotate the PAN control full left or full right to select the desired buss.

MONITOR SUBMIX SYSTEM

33 MONITOR Select Switch Matrix

This switch matrix selects the output of the 4 PGM Output busses, the AUX Submix system, and the EFFECT Submix system, and allows you to select between these signals for the MONITOR OUT and Headphones jacks.

If only one of the PGM OUT buttons is depressed, that signal will appear on both the Left and Right MONITOR OUTs and the Headphones jack. If two PGM OUT buttons are depressed, PGM OUTs 1 and 3 will show up in the Left channel, and 2 and 4 will show up in the Right channel. Both AUX and EFF will show up on both Left and Right channels (mono).

The selections available are restricted so that you can select no more than 2 signals at a time. PGM 1 and 2 buttons have priority over PGM 3 and 4 buttons, and 3 and 4 have priority over the AUX and EFF buttons. If PGM 1 and 3 buttons are depressed at the same time, for example, only PGM 1 will be heard.

There is also a MONO button which adds the Left and Right signals together in the MONITOR Submix system, it does not affect the PGM Output busses. The MONO switch will show you what the mix will sound like when the final format is mono, such as a radio spot. Rather than force you to "center" all the PANs, we provide this switch. It will sum together the busses you've selected (1 & 2, 3 & 4) in the MONITOR Select switch matrix and will show you what your mix will sound like before it is broadcast.

In stereo recording for release, it is useful to

know in advance what will happen when a stereo sound is combined to mono, even though no mono mix is planned. Much difficulty with mics wired "out of phase" or effects return added to the mix unwisely can be avoided by listening to mono in the monitor while you still have a chance to change the approach. Since two loudspeakers are truly independent systems, you can get a terrific sounding tape that makes a very poor mono if you don't check for compatibility. What to listen for? A mix that doesn't lose most of its high or low frequency content when you select mono.

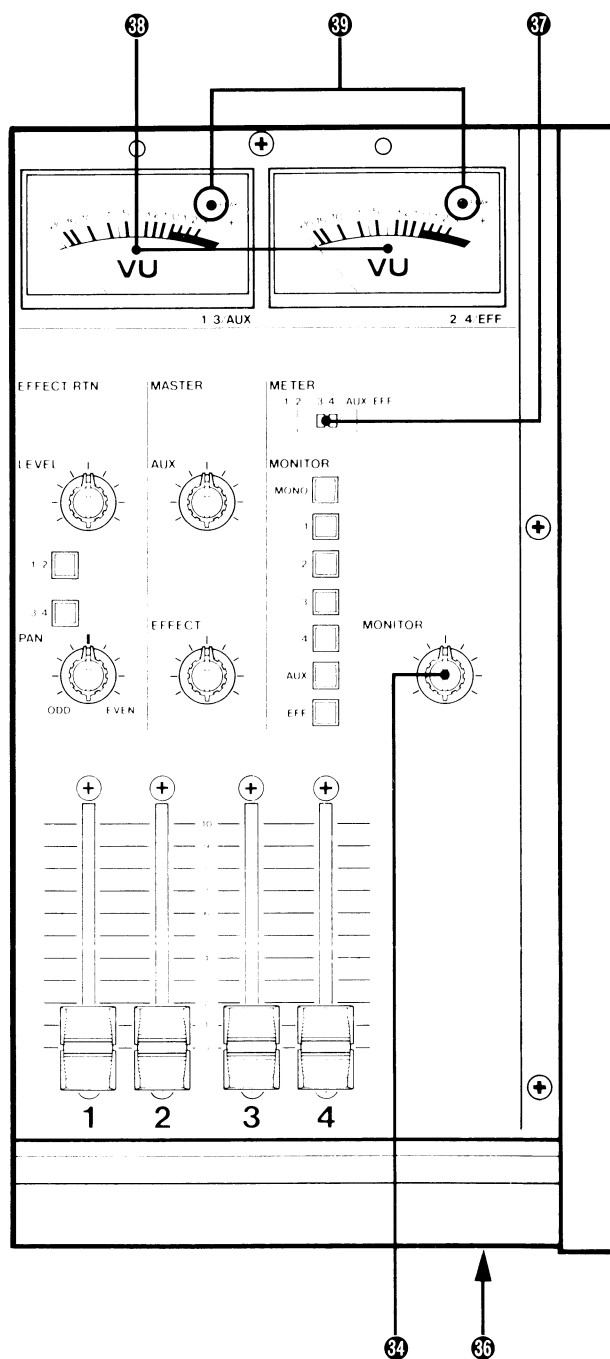
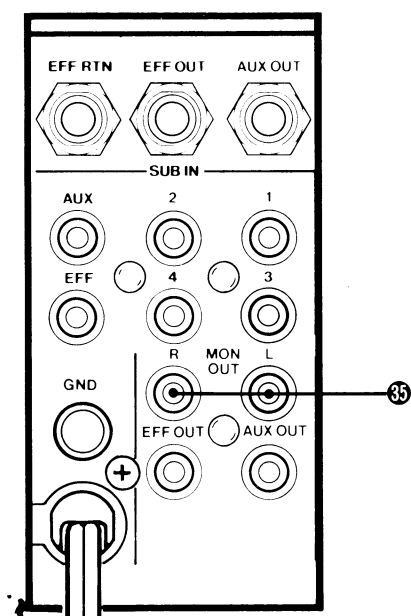
Phase is a difficult subject and there are no simple solutions that we can guarantee. You will have to experiment to find solutions one at a time. Moving the mic 1/2 inch may change everything.

34 MONITOR Level Control

This adjusts the signal selected by the MONITOR select switch matrix that shows up at the MONITOR OUT jacks and the Headphones jack.

35 MONITOR OUT Jacks

These RCA jacks provide the output for the MONITOR Submix system. These should be connected to your stereo amplifier and speakers for control room monitoring.

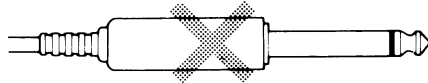


36 Headphones Jack

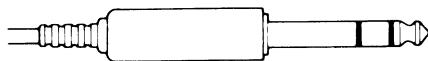
This 1/4" phone jack is where you connect your stereo headphones. The signal at this point is the same as the signal at the MON OUT jacks. This output is designed for 8 ohm stereophones.

CAUTION;

MONO (2 WIRE) HEADPHONES WILL CAUSE CIRCUIT FAILURE. If your headphones have this connector, don't use them



(1/4" phone 2-connector)



(1/4" phone 3-connector)

Your headphone connector must have 3 sections to be safe. While accidents do happen, and protection circuits have been built in, use of mono/2 wire headphones will eventually cause circuit failure (2 to 3 minutes). Using the 2 wire connector shorts out one of the amplifiers driving the headphones, which will cause it to burn out.

METERS

37 METER Select Switch

This selects which signals the 2 VU meters will display. With the switch in the left (1-2) position, the left meter displays the level of PGM Output buss 1 and the right meter displays the level of PGM Output buss 2. If the switch is in the center position, the left meter displays the level of PGM Output buss 3 and the right meter displays the level of PGM Output 4. With the switch in the right position, the left meter displays the level of the AUX Submix output, and the right meter displays the level of the EFFECT Submix output.

38 Meters

These two meters display the signal level of the outputs selected by the METER select switch. They respond to the average signal level using standard volume unit ballistics, and do NOT show peak levels.

39 PEAK Indicators

These light emitting diodes will react much more quickly than the meters, and are set to "flash" 10 dB above "0 VU". They will show you the difference between average and peak levels. On most percussive material (kick drum, latin percussion such as castanets) you will see these LEDs flash long before the VU meters read anywhere near zero. Short term peak distortion may be hard to detect. Use discretion and experiment with the final meter level when you see these lights flash. They are telling you the truth about the REAL level that is being sent to the final output. "Average" is not always a safe concept. For example, castanets should be recorded with no more than -20 indication showing on the averaging VU meter. Even when the meter reads this low you may still see the LED flash. Take care and avoid overload.

It is normal for the 106 meters to jump when AC power is first applied, and the headphone amplifier may produce a substantial transient "pop" even if the MON control pot is rotated fully leftwards (off).

Calibrating the 106 Mixer for Recording

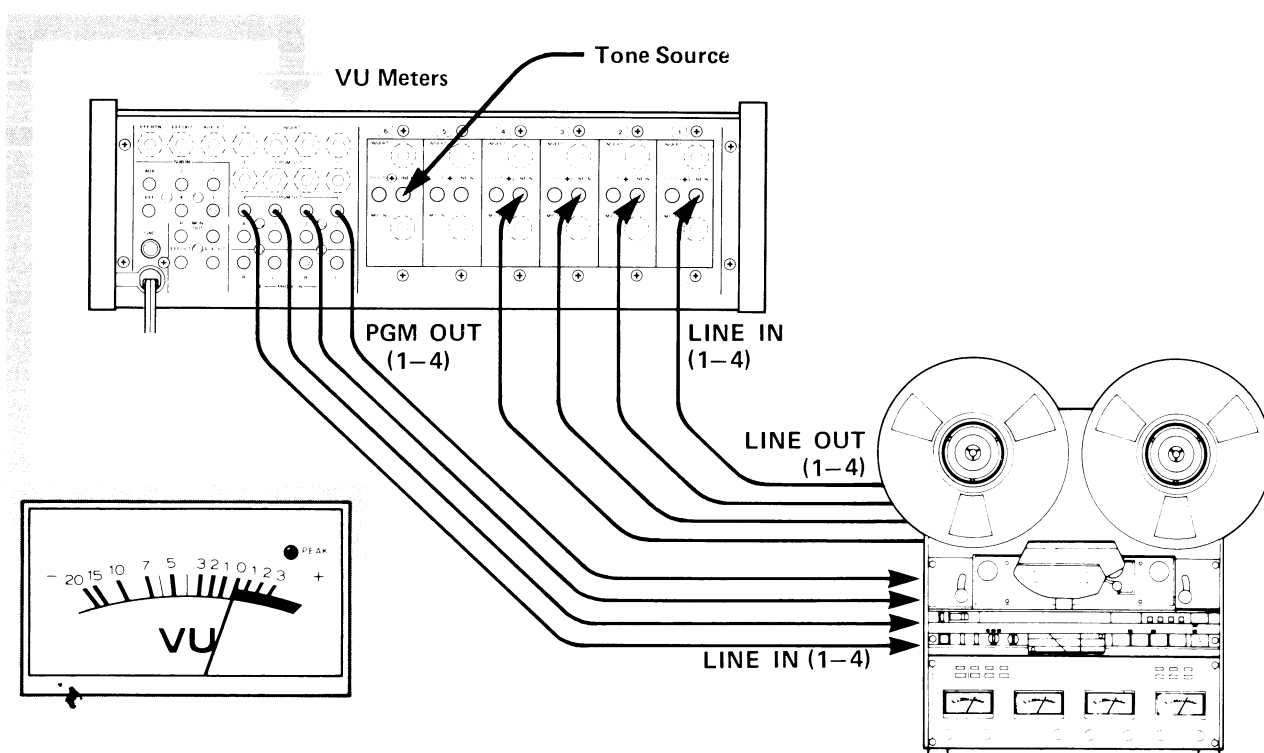
Calibrating the 106 mixer is simply the process of matching its output to the rest of the equipment being used. In this example, we will assume you are using a 4 track recorder. The method we use here can be used with virtually any type of equipment you are likely to encounter.

The first step is to connect the outputs from the 106 to the inputs of the 4 track. Once the equipment is connected, you can begin the calibration. A test tone generator or test tape is handy to have for this procedure. Assuming you have a tone generator or constant tone source, set the frequency to 1 kHz then proceed, as outlined below:

1. Connect the tone source to the channel 6 line input. Switch the INPUT select switch to LINE.
2. Set the EQ and PAN to the 12 o'clock position. Depress both assign buttons.
3. Raise the channel 6 fader to the shaded area, about 7 – 8 on the fader scale.
4. With the METER switch set to 1–2, raise PGM 1 & 2 faders until the meters read

0 VU. Switch the meters to PGM 3 – 4 and set the levels to 0 VU.

5. Raise the input level controls on the recorder until its meters read 0 VU.
6. Record the 0 VU tone, on all four tracks, for about a minute. Disconnect the tone source.
7. Connect the outputs of the recorder to the first four LINE inputs of the 106.
8. Set inputs 1 – 4 to the LINE position with the EQ flat (12 o'clock position).
9. Rewind the tape and play the tone. Set the recorder's output level controls so the recorder's meters read 0 VU.
10. Pan channel 1 full left, 2 full right and depress assign buttons 1–2. Pan channel 3 full left, 4 full right and depress assign buttons 3–4.
11. Raise the channel 3 & 4 faders until the meters read 0 VU. Switch the meters to 1 – 2. Raise the channel 1 & 2 faders until the meters read 0 VU.

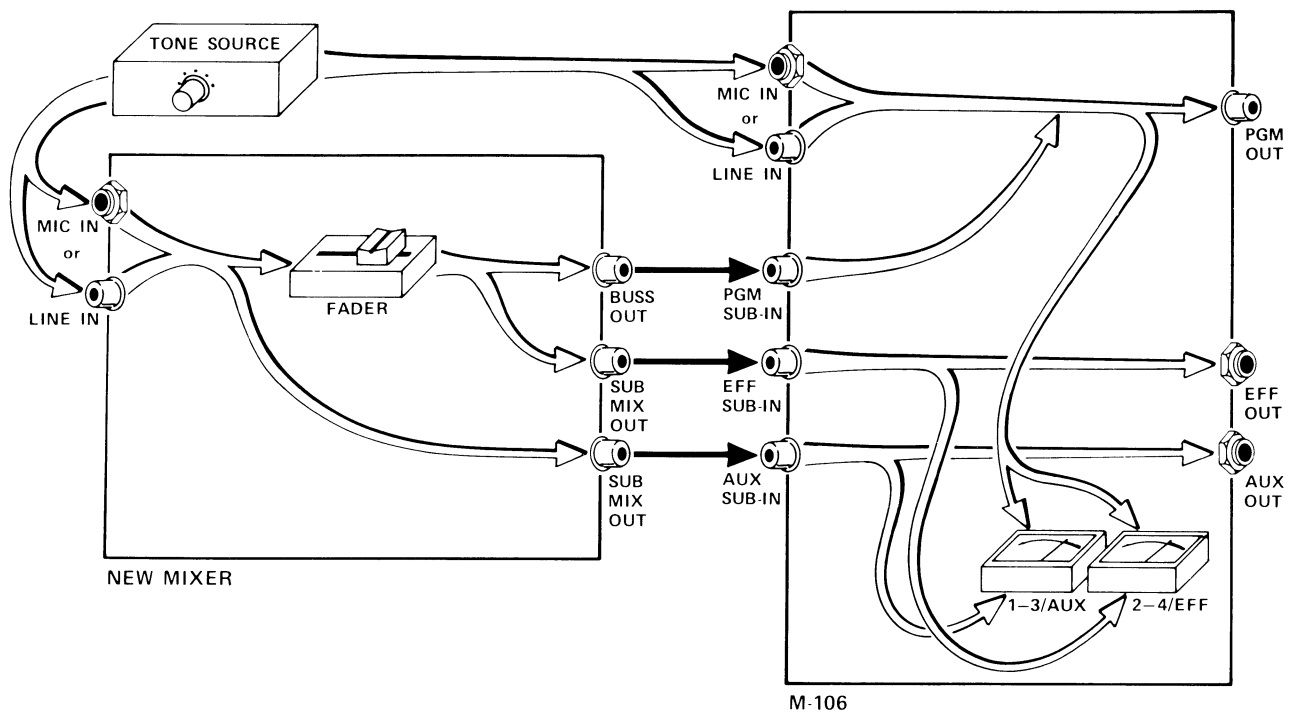


Calibrating “Stacked” Mixers

“Stacking” mixers allows you to expand your mixing capability by increasing the number of inputs available. It also allows you to expand your mixing capability without obsoleting the equipment you already own — you simply add on to what you already have. The SUB INs provide the access points to make this possible.

Note that there is no standard terminology among mixer manufacturers. Since we have no way of knowing what make of mixer you are stacking with the 106, we can’t tell you the exact names of a given feature on that mixer. The names used will be chosen to best describe the feature in as universal terms as possible.

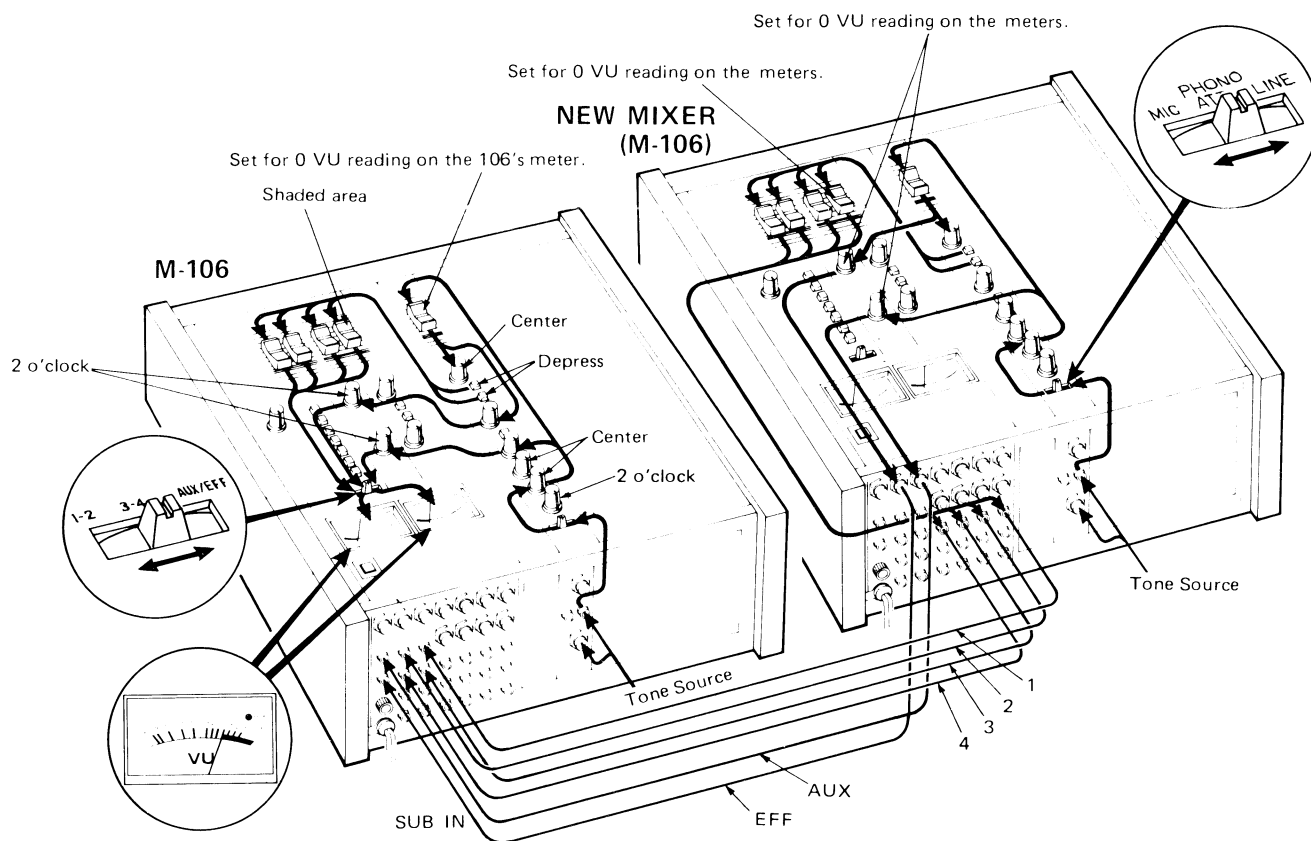
To stack two mixers, you need to connect the new mixer’s Program Outputs to the 106’s PGM SUB INs. If the new mixer is a 4 buss (output) board like the 106, connect its Program Output 1 to the 106’s PGM SUB IN 1, Program Output 2 to PGM SUB IN 2, and so forth until all the Program Outputs are connected to the 106’s PGM SUB INs. If your new mixer has only two outputs, connect the first to PGM SUB IN 1 and the second to PGM SUB IN 2. Next, connect the mixer’s Pre-Fader submix output to the 106’s AUX SUB IN, and the mixer’s Post-Fader submix output to the 106’s EFFect SUB IN.



You can calibrate the Program Outputs, Pre-Fader submix system, and Post-Fader submix system on the new mixer so that they operate at equal levels with the 106's submix systems. Calibration also allows you to operate the two masters, disregarding the masters on the new mixer. To calibrate the two mixers:

1. Set the PGM Master faders on the 106 to the same point in the shaded area.
2. Set the METER switch to 1-2.
3. Connect a tone source to an input channel on the 106. If you are using a line level tone source, connect it to the LINE IN jack. If you are using a lower level tone source, connect it to the MIC IN jack, and adjust the TRIM control to the "2 o'clock" position.
4. Set the INPUT switch to MIC or LINE, depending upon which jack your tone source is connected to.
5. Set the PAN control in the center, and depress both assign buttons. Be sure all EQ knobs are centered.
6. Make sure the input channel fader is fully down.
7. Turn on the tone source.
8. Slowly bring up the channel fader until the meters read 0 VU. If you are using the MIC IN jack and you can't get a 0 VU reading even with the fader up full, turn up the TRIM control until you do. If you are using the LINE IN jack and have this problem, turn up the output of the tone source or move its connection to the MIC IN jack.
9. Adjust PGM Masters 1 and 2 to compensate for any slight differences between the two busses. Set the METER switch 3-4 and adjust PGM Masters 3 and 4 to compensate for any slight differences. All 4 PGM Output busses should read exactly 0 VU.
10. Set the 106's AUX and EFF MASTERS to the "2 o'clock" position.
11. Set the METER switch to AUX/EFF.
12. Slowly bring up the input channel's AUX level control until you get a reading of 0 VU on the left meter.
13. Slowly bring up the input channel's EFFECT level control until you get a reading of 0 VU on the right meter.
14. Turn off the tone source and connect it to an input channel on the new mixer. Use the same type of input (MIC IN or LINE IN) on the new mixer as you used on the 106.
15. On the new mixer, adjust the settings on the input channel, including the Pre-Fader and Post-Fader sends, to match those on the 106, and assign the channel to all the Program Output busses. Be sure all EQ knobs are centered.
16. Turn on the tone source.
17. Set the 106's METER switch to 1-2. Bring up Program Master fader 1 on the new mixer until you get a 0 VU reading on the 106's left meter. Bring up Program Master fader 2 until you get a 0 VU reading on the 106's right meter.
18. Set the 106's METER switch to 3-4. Bring up Program Master fader 3 on the new mixer until you get a 0 VU reading on the 106's left meter. Bring up Program Master fader 4 until you get a 0 VU reading on the 106's right meter.
19. Tape the new mixer's Program Master faders so that they won't move from that position. Drafting tape is the best choice.
20. Set the METER switch on the 106 to AUX/EFF.
21. Slowly bring up the Pre-Fader submix Master until you get a 0 VU reading on the 106's left meter.
22. Slowly bring up the Post-Fader submix Master until you get a 0 VU reading on the 106's right meter.
23. Tape the new mixer's Pre-Fader and Post-Fader Masters so that they won't move from that position. Drafting tape is the best choice.

Your two stacked mixers are now calibrated.



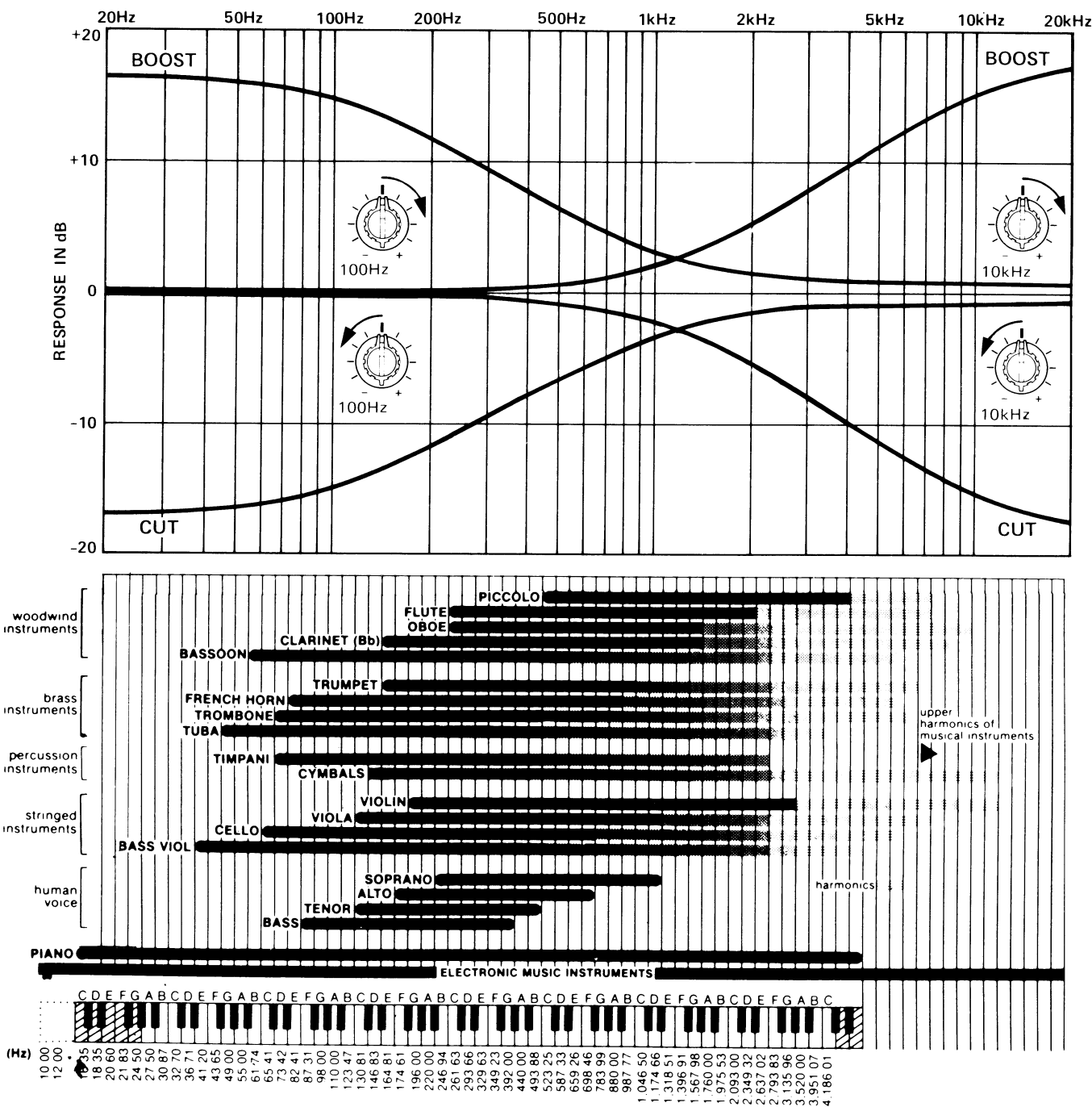
How to Use the 106's Equalizer

EQ can be used to change the tonality of an individual instrument. The 10 kHz control affects the "brightness" or "brilliance" of an instrument, and the 100 Hz control affects the "boominess" or "bassiness" of an instrument.

It is important to remember that there are two ways to make a given tonal change. If you want to add 10 kHz, for example, you can get a similar effect by turning down the 100 Hz.

you want more in the low frequency range, you can turn down the 10 kHz and get a similar tonality change.

In general terms, you get a desired tonal change in two ways. Either make the appropriate change on the control that affects the range you want to alter, or make the opposite change on the control that affects the opposite frequency range.



A Word of Mixing Advice

The equalizer on the 106 is a two-knob shelving type, and its range of control covers the low and high frequencies. To use an equalizer, it is important to understand the frequency range of the sound sources you will be processing and the control range of the equalizer. Refer to the illustration on the preceding page to see the relationship between the frequency range of various instruments, and the range of control of the 106's equalizer.

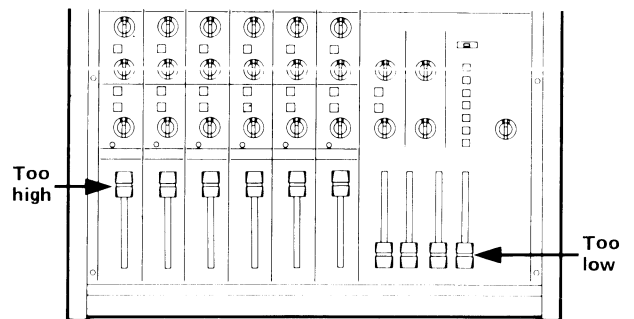
This illustration shows how different instruments will be affected by changes in the EQ settings. You can see how certain instruments will be unaffected by changes on one of the EQ controls. Cymbals and flutes, for example, will not be affected very much by the 100 Hz EQ. This is because these instruments have little or no frequency content in that range. On the other hand, you can boost or cut a certain part of an instrument without boosting or cutting other parts of that instrument. On drums, for example, you can bring out the kick drum relative to the other drums by turning up the 100 Hz control. On the other drums that have a little frequency content in that range there will be some effect. Because the kick drum has a large amount of energy in the 100 Hz range, it will be the most affected by changes on the 100 Hz control.

Likewise, if you want to accentuate or diminish the cymbals in a drum kit, you can do so using the 10 kHz control. Turn it up for more and down for less. This, too, will affect any other source (signal) that has frequency content in that range. Experience will show you that there is a limit to how much cut or boost you can do before its effects on the other signals become desirable.

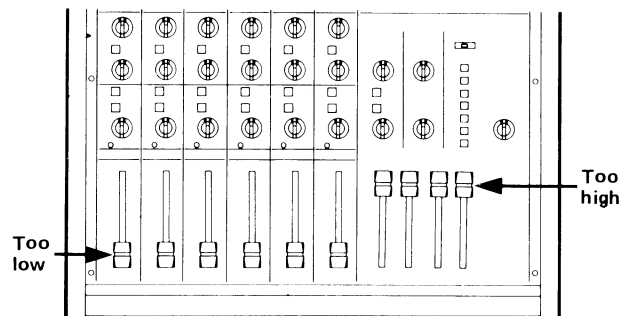
As before, experience will help you learn the capabilities and limits of this method.

All finished tapes must be balanced — one sound and its tone judged by blending with others. Don't depend on EQ to set up a "perfect" tone, because the minute you add your perfect sound back to the "mix" the tone may not be so "perfect". Always try to get the levels as close to "right" as possible before using EQ. If the mix is close, you will know which tracks need fine EQ tuning to be heard. Less EQ means less distortion and full boost on every pot will also boost the noise in your mix as well as the signal.

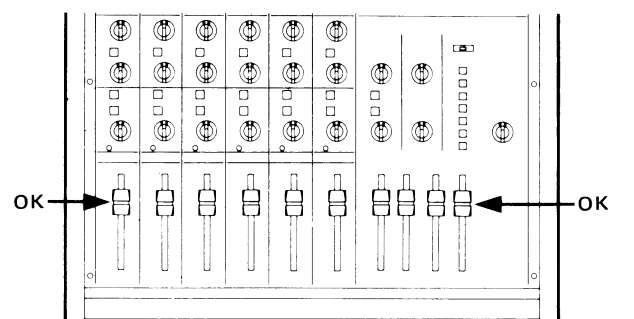
1. If your console faders always wind up like this, you are likely to be over-loading your outputs. Pull down the inputs and raise the master.



2. Conversely, if this is what you usually have, you are getting too much gain from your master. Your mix is clean, but noisy.



3. This picture is a reasonable compromise and is probably better all around.



Applications

This is the section of the manual designed to get you started mixing with your TASCAM 106. We provide several examples to aid you in creating quality audio, but these examples should not be considered as the "only" way to accomplish the task at hand. As you learn the various systems of the 106, you will discover alternate methods which may suit your particular mixing situation better.

BASIC PA (PUBLIC ADDRESS)

The 106 is ideal as a small live sound mixer for P.A. applications. It is capable of mixing up to 6 mics, 6 line level signals, and 2 stereo turntables (although not all at the same time!) In our example, we will show a simple yet very typical mixing situation using 4 mics and 2 line level signals. As you can see by our setup, the 106 is being used to mix the performance of a trio consisting of an acoustic guitar, a bass player and a keyboard player (a synthesizer). All three musicians have vocal mics (input ch's 1, 3 & 4) while input ch. 2 is being used to mic the acoustic guitar. Inputs 5 & 6 are handling the signals from the synth via the line in jacks. All the signals are being mixed to the 1 – 2 busses to create a simple stereo output mix feeding the main power amp and speaker systems. On-stage monitoring is being controlled by the AUXiliary system. The MONITOR signal is derived pre-fader. This allows the monitor and house system levels to run independently. The monitor signal is then sent to the monitor amp and speaker system from AUXiliary output jack. Special audio effects devices such as reverb, echo or compression could be patched into the signal path at three different points in this set-up:

1. Through the individual channel INSERTion points, to process a single signal.
2. Through the Program 1 & 2 INSERTion points, for continuous processing of a mix of signals (i.e., E.Q., limiters, etc.), or
3. Through the EFFECTs OUTPUT and EFFECTs return (RTN) jacks, for top panel control of level and distribution (PAN and assignment).

When setting up a system of this type, the first thing to do is to put the speakers and amplifiers in their proper positions. Then it's time to wire

each system, audience and monitor, to the 106. At this time make sure all the faders on the 106 are down; then, begin plugging in the mics and line connectors. With the faders still down, begin turning the system on by first turning on the 106, then the power amps. Set all the equalizers to the twelve o'clock or flat position. Set the channel input select switches to the appropriate position; MIC or LINE. Assign the inputs to the desired outputs (Program 1 & 2 in this example) and raise the output (buss) fader to 7. Set the CH PAN to the center or twelve o'clock. Set the first input trim to the twelve o'clock position. Begin setting your channel levels by speaking into the first mic while slowly raising the corresponding channel fader. The level will be correct if the output meters are near or at 0 VU and the fader is in the shaded area, from 7 to 8 on the fader scale. If the fader is out of this range, adjust the fader and the input trim control until the level is about right.

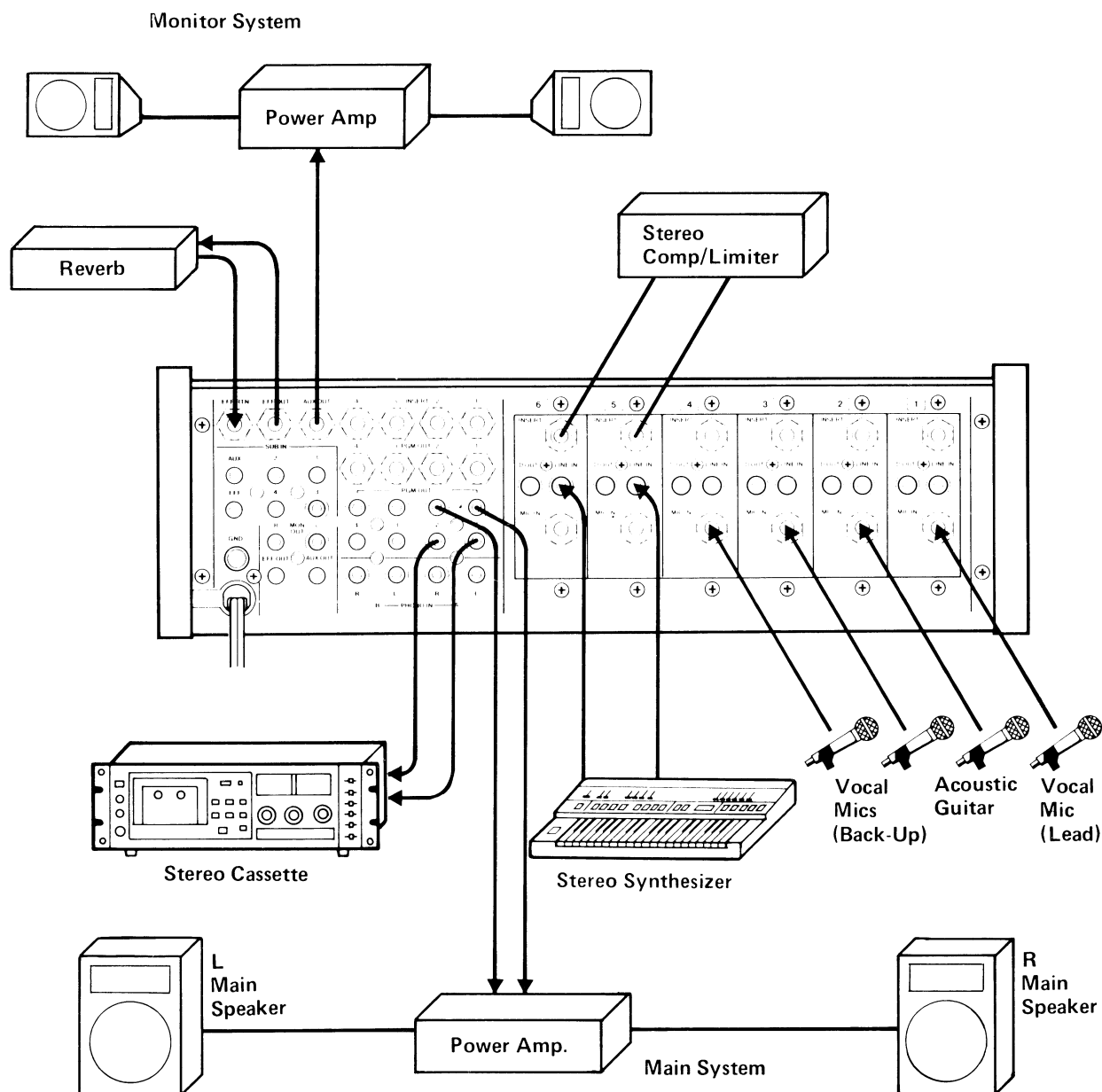
If the mic input level is too high or strong, the channel input overload LED will light. If this indicator is on constantly, move the input selector switch to the ATTenuation position. This puts a 30 dB pad into the input, reducing the signal strength and possibility of overload and distortion. If the overload LED only lights periodically, but the channel fader must be kept in a low position (below 5) to attain the proper output meter reading, then you need to adjust the input trim control downward (counterclockwise) until the fader can be raised to the proper position (7 – 8) and the signal level is correct.

Line level signals such as those generated by a synthesizer as in our example, may be set in a similar manner as the mic levels. The difference is the line input bypasses the input trim control. Since most line level signals are generated by electronic devices, the level can normally be adjusted at the output of the instrument. The set-up remains the same as the mic section.

Setting the level for on-stage monitors is very similar to setting the program levels, except different controls are used. In our example, the signals are derived pre-fader from the inputs by selecting the normal AUXiliary position and opening the AUXiliary control on each channel. Begin by setting these controls to the twelve o'clock position. Next, move the meter select switch to the EFFECTs-AUXiliary position. By watching the left hand meter (1-3/AUX), you can raise the AUXiliary MASTER control

while signal is present in each input until the meter reading is about the same as the program level. When setting monitor levels, several precautions should be taken. If howling or feedback occurs, reduce the AUX MASTER level and either reposition the monitor speakers or the mics. This should allow you to raise the level to the desired working volume. Often, several

various combinations of speaker and mic positions must be tried before the most useable solution is found. Keep in mind, monitor output levels usually do not need to be as high as the program level. The object of on-stage monitoring is to allow the performers to hear the various key components of the mix in order to improve their individual sound and timing.



BASIC RECORDING PROCEDURE

Previously, we described live sound mixing and P.A. uses of the 106. Here we will concentrate on using the 106 in a small recording studio situation. Many of the techniques mentioned previously may be applied here as well, so we will not repeat the basics of setting levels, etc.

Basic Connections

In our example for recording, we will assume you are using a 4 track recorder. Later, we will discuss the basic mixdown procedure and we will assume you would be using a 2 track recorder.

The first step is to connect the 4 track machine. Since the 106 has 4 outputs, the simplest method is to connect Program (PGM) out 1 to the track 1 input of the recorder, Program out 2 to track input 2, etc. Next, connect the output of the tape recorder to the line inputs to the 106. Again, the simplest way is to connect the output of track 1 to line in 1, etc. Once these connections are complete, the next step is to calibrate your system (see System Calibration pg. 18).

You are now ready to begin the basic mix session.

Recording the Basic Tracks

Using the example group we so successfully mixed in the live session, we will now begin to record a multitrack tape for mixdown later. Remember in the live session, we mixed all the inputs down to a stereo left and right pair. Here we will use all four program busses and tracks, but we don't have to record all four at the same time. By saving some tracks, we can record more information overall than we could if we recorded all four at once. The first track we want to record is the basic rhythm or beat tracks consisting of the bass and the "left hand" of the synthesizer. Since the group doesn't have a drummer, the easiest way to insure proper timing is to record a "click" track which is basically an electronic metronome or electronic drum unit. This is done by plugging the device into the Program SUB IN of the track or via the EFF RTN routed to the track, for our example, track 3, setting the proper level and recording the beat quite a bit longer than the

finished tracks length. Once this is done, rewind the tape to the beginning.

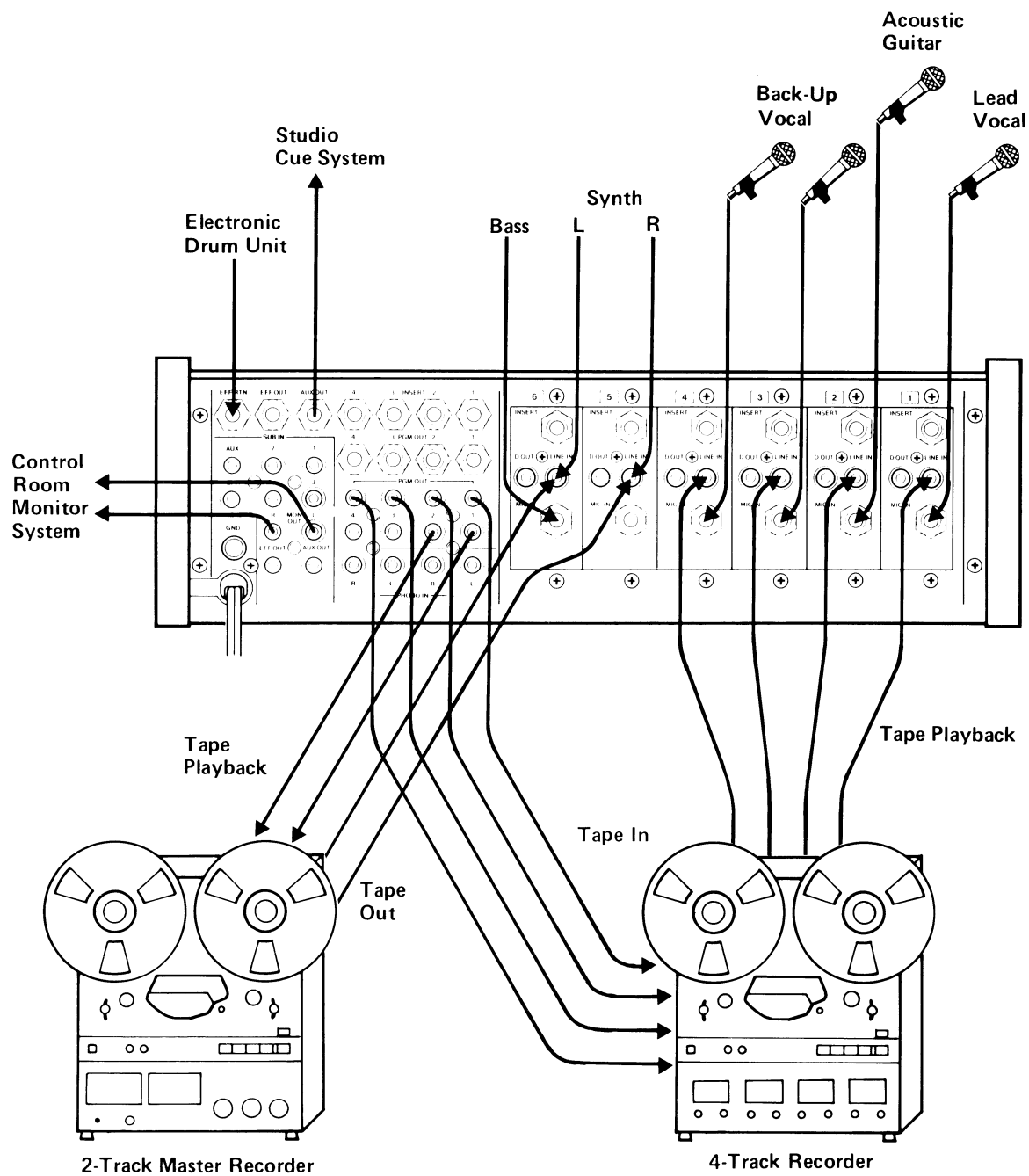
Now we can set up the studio cue feed so the bass and keyboard players can hear it and play in time. This is done by depressing the AUX Line Select Switch on input 3, track 3's playback channel. You can leave the channel 3 fader down, or off, so the click can't bleed into the mix. The studio cue headphone amp (we are using a TASCAM MH-40B Headphone Distribution Amp) can then be connected to the AUX output of the 106 and the overall cue level controlled by the AUX master control.

Now you can begin the recording process. Connect the synth line inputs to the mixer, and using either a direct box or a microphone, add the bass guitar signal to the synth signal. Assign these signals to one track of the multi-track recorder by using the channel assign switches and the PAN control. For our example, let's put them on track 1. Now, play track 3 (the "click" track) while recording on track 1. Once you are satisfied with the first track, you can continue to build the tape one track at a time. Again for example, the next track could be lead guitar and right hand synth on track 2, then lead vocals on track 3 (this erases the click track which you no longer need).

Finally, all the "extra" instrumentals and the background vocals can be recorded on track 4. Keep in mind, as each track is being recorded, the previous tracks can be sent to the cue system for reference. It's a good idea, after each pass to playback the previously recorded tracks alone to hear if any audio problems are emerging, such as instruments out of tune, voices off key, etc. If problems are found, correct the cause and re-record the offending track. While certain tonal characteristics can be re-equalized later, many problems are impossible to "fix in the mix".

Mixdown Procedures

Once a multi-track master tape has been created, the next step is to mix it down to a standard two track stereo format. The procedure for this is very similar to the original recording process, however, there is often no need for the studio musicians unless overdubbing is going to take place (overdubbing is the process of replacing a portion of the original material with new



signal or the blending of original material with new material). The only additional equipment required is a 2 track mastering recorder and any auxiliary effects devices you may feel are needed.

The basic hook-up is fairly straight forward: The Program 1 & 2 outputs connected to the inputs of the 2 track; the output of the 2 track connected to channel 5 (L) and channel 6 (R) line inputs. Any of the effects can be patched to the 106 in a number of ways. Specialty or limited use effects such as a flanger should be patched directly to the input which requires it by way of the channel patch point. Similarly, if an effect is desirable on most of the signals being mixed, it can be patched through the Program patch points. If an effect is to be used on a changing number of signals throughout the mixdown process, it can be connected to the 106's EFFECTS output jack and brought back into the mix via the EFFECTS return system. This method allows any signal to be sent to the device simply by opening the channel EFFECTS send controls, while the overall output and return levels can be accurately set by the master send and return controls. A fourth method is a bit trickier, but may come in handy if the effect and/or the signal being processed is subject to level variations or needs equalization. The input of the effect can be driven by a signal taken from a channel's direct output (Post EQ-Post Fader), from a channel or program patch or access output, from the AUX or EFFECTS master output, or from a program output, provided it is not going to the mastering 2 track. The output from the effect can then be connected to the line input of an unused channel. In this way the returning signal can be controlled as if it were a regular line input. Caution must be exercised to prevent this returning signal from being mixed with the original outgoing effect signal or a feedback loop will be created. This will cause a loud howling sound and could damage your equipment.

Once you have connected all the equipment, the rest of the procedure is straight forward: calibrate the machines and the mixer outputs, set the levels, rehearse or preview the mixdown, make any changes and roll tape. As always, once you have completed a mixdown, review the results before you repatch or put away the equipment. If the results indicate the need to make changes, go back and do it again.

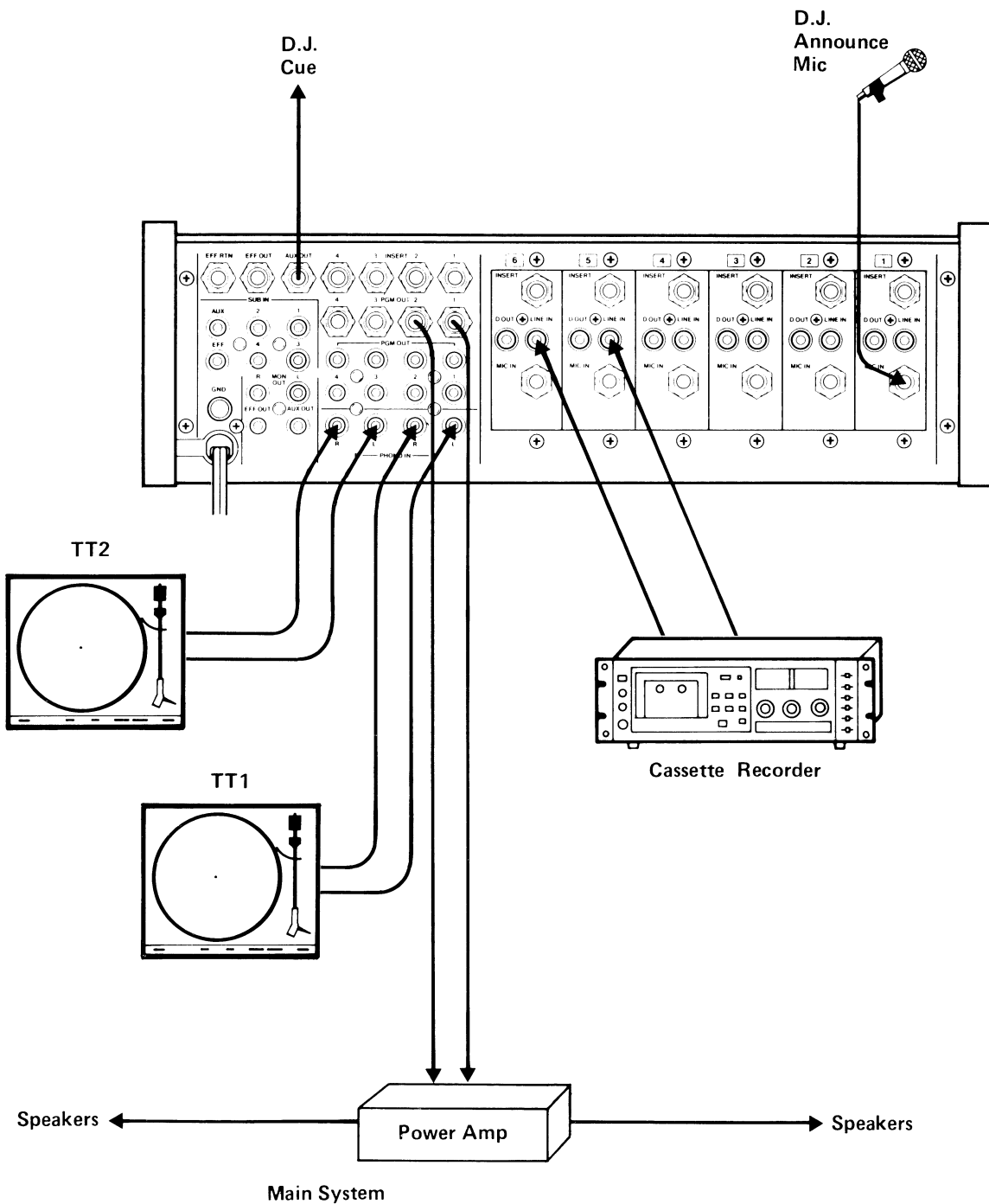
DISCO AND NIGHTCLUB MIXING WITH THE TASCAM 106

Mixing the outputs from multiple pre-recorded sources is a difficult task which can be made easier by using the TASCAM 106 mixer. The features and design of the 106 are ideal for this type of application. The 106 can mix the signals from 2 stereo turntables while also connected to 3 stereo tape decks. The switching and patching capabilities allow further signal input and selection if required, but in most cases, 5 stereo sources are enough. The 106 has two pairs of dedicated PHONO inputs which are wired directly to the last 4 input channels. These do not, however, prevent the mixing of either MIC or LINE signals on these channels if necessary, although using a mic input on these channels does over-ride the corresponding PHONO input signal.

A typical set-up for disco would find the 2 turntables running through inputs 3 – 6, while stereo tape playback decks would be connected to the line inputs of all six (6) channels. Segues between turntables can be quickly done by cross-fading channel pairs (the easiest, most consistent method to accomplish this is to use fader-link knobs). Once a given turntable is off-air, its inputs can be quickly switched to the line position for a tape segue from the on-line source. The AUXiliary send system of the 106 is designed to allow the D.J. to have access to all the signal sources without having to change the input switch settings. If the input source to be cued is connected to the line inputs, the channel AUX send selection switch can be depressed to send just the line signal to the AUX Submix System. This signal can be monitored directly from the AUX output or through the monitor section.

Similarly, if the source to be cued is on the PHONO or MIC inputs, the AUX send can be switched to send those signals pre-fader. In this way, records can be back-cued and slipped while tapes can be rocked into position. All the while allowing the D.J. to punch-in the announce mic for intros. The four outputs can be used to drive two separate stereo systems, plus the parallel connectors on each output can be used to feed multiples of each output pair. The Program patch points and the EFFECTS send and return networks provide access for introducing special effects devices such as reverb or for adding final

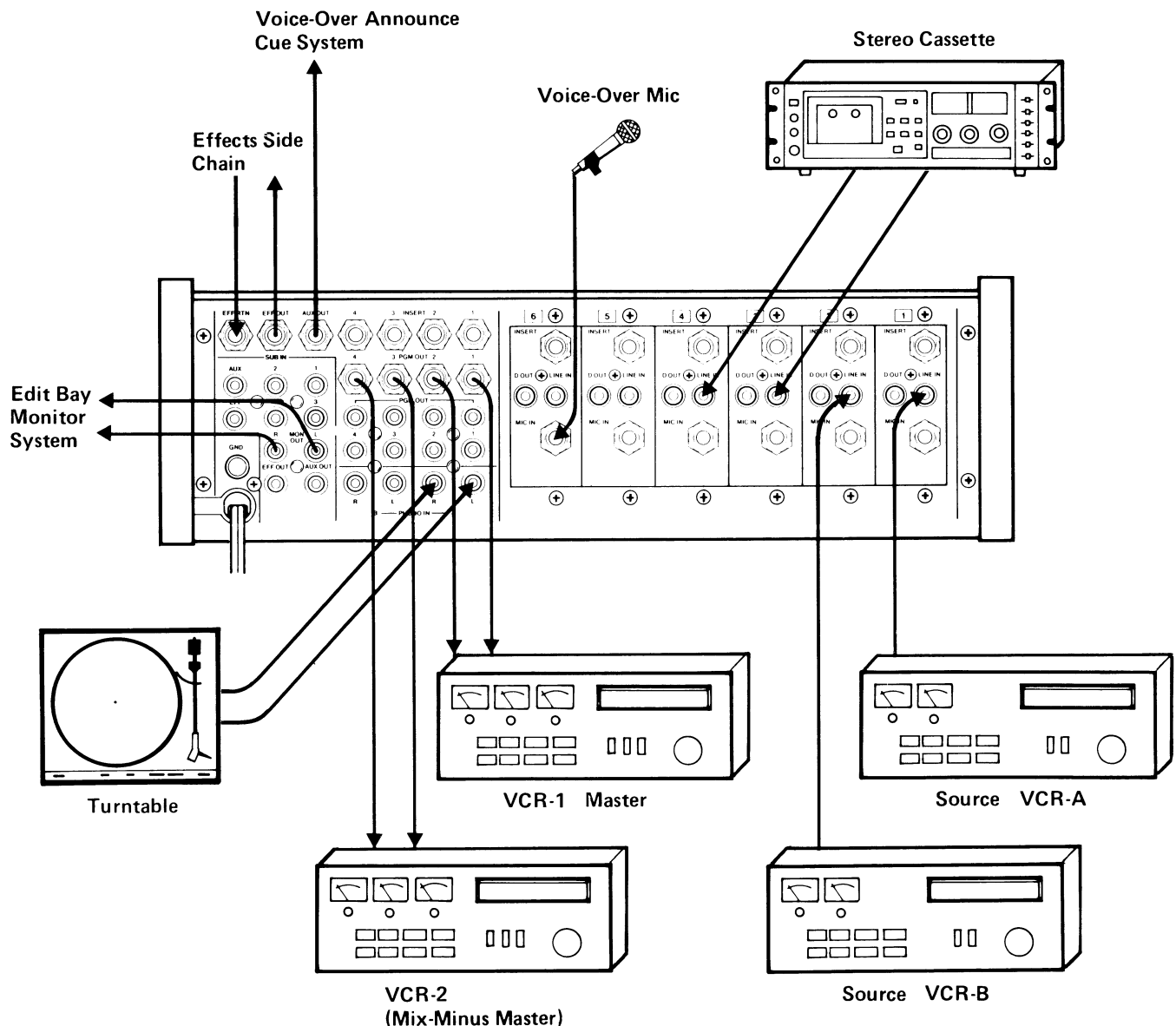
signal processing equipment such as limiters for overall level control. Equally important is the capability to monitor all the separate busses without interrupting the signal flow. Thus continuous entertainment can be assured even if some adjustments to the signal must be made.



VIDEO POST-PRODUCTION AND AUDIO SWEETENING WITH THE 106

Mixing audio for video has become a demanding field due to the rising sophistication of both the production facilities and the viewing audience. With the advent of cable T.V. systems offering stereo audio feeds as well as the coming stereo broadcast systems for over-the-air programming, the awareness of the importance of quality audio production for this field will continue to

grow. Another area of rapid growth in the field of video is in non-broadcast applications. As corporate or industrial telecommunications expands, the need for improved audio will keep pace. The TASCAM 106 has all the features necessary to fulfill the audio control needs of the small independent production house or off-line editing/sweetening facility. Virtually all popular brands and models of 3/4" and 1/2" VCR's will interface directly with the 106.

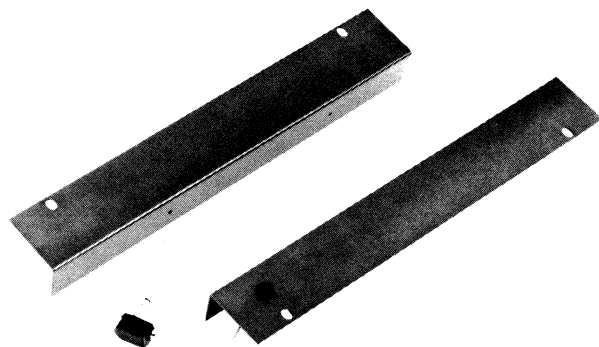


Typically, mono or stereo audio signals from 3 to 6 source machines could be mixed simultaneously. These signals could be combined with various other audio sources such as voice-over mics, turntables, cart machines, etc., and assigned to the program outputs feeding the edit master record VCR. Since the 106 has 4 program busses, a mix-minus of the various ambient and wild track sources can be created simultaneously, without adding the narration and voice-over material. This can then be dubbed later in a different language in a typical audio-follow-video session. The AUXiliary sends on the 106 can be used for cueing the voice-over announcer, while the EFFects sends and return system can be used as a side-chain for special effects.

In the example shown, two stereo source VCR's are shown in a typical A — B roll editing situation while a voice-over mic is running through input 6. A cart or cassette machine could be connected to input 5 if needed for hot rolling effects on to the mix. Monitoring of the mix is done through the MONitor output section while cues are sent via the AUX out to the V.O. booth. The master record machine is being fed signal from Program Outputs 1 and 2, while an international dub (mix-minus-narration and V.O.) is being recorded from Program outs 3 and 4. Additional dubs could be cut simultaneously from the parallel output jacks of the Program outputs.

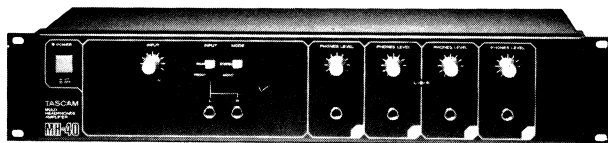
Accessories for the 106

RM-106 Rack Mount Kit



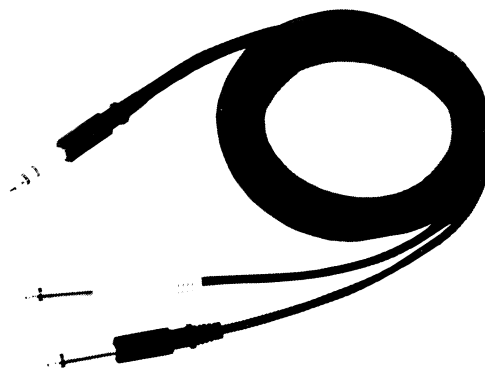
The RM-106 is designed especially for the TASCAM 106 and allows it to be mounted on an EIA standard 19" rack.

MH-40B Headphone Amplifier



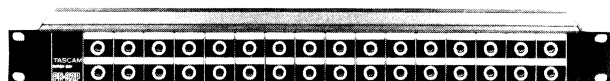
The MH-40B is a headphone distribution amplifier that can be used to feed four sets of studio cue headphones and can be mounted in a 19" EIA rack.

PW-2Y/PW-4Y Insertion Cable



The TASCAM PW-2Y/PW-4Y is a connecting cable that allows signal processing such as a graphic equalizer to be inserted at specific points of the signal path of the 106. Its tip-ring-sleeve plug connects to the INSERT jack while its "Y'ed" end accommodates connection to the input and output terminals of the outboard equipment being used. Available in two lengths — 2 m (PW-2Y) and 4 m (PW-4Y).

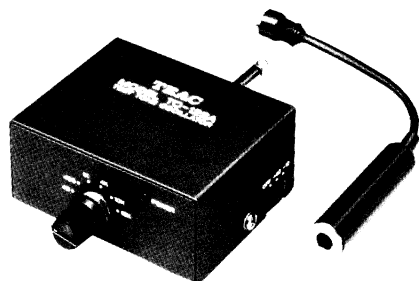
PB-32 Patch Bay



The PB-32 patch bay is today's simplest and most effective way to deal with the ever-increasing tangle of wires necessary for the recording and signal processing you want to do. Instead of leaning over, walking around, or turning consoles, racks, recorders and other equipment, you can bring all those confusing cable ends to one single spot in your rack, or next to your mixer, and get them under control. You can label inputs and outputs on the PB-32, and won't have to guess anymore at what you might be plugging in. A few cords can save endless hours of searching around on the floor behind your console and, as the whole process of patching is made simpler, you'll probably find more flexibility in your recording setup.

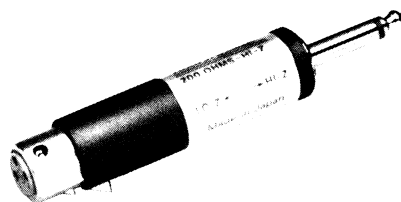
The PB-32 comes in four versions. The PB-32P is equipped with 1/4" phone jacks only, the PB-32R with RCA jacks only, the PB-32H with 1/4" phone jacks on one side and RCA jacks on the other, and the PB-32W with 1/4" phone jacks for 6 channels and RCA jacks for the remaining 10 channels.

TO-122A Test Tone Oscillator



This compact, battery powered unit can be used to check the input/output levels, channel balance, and electronic alignment calibration of the 106. It has an RCA pin jack output, switchable output levels of -10 dBV (line level) or -40 dBV (mic/instrument level). Frequency is switchable to 40 Hz, 400 Hz, 1 kHz, 4 kHz, 10 kHz or 15 kHz. The TO-122A is also useful for calibration and troubleshooting of entire recording systems.

109B Input Transformer



The TASCAM 109B Input Transformer is an adaptor that matches balanced low impedance microphones with XLR connectors to unbalanced high impedance 1/4" phone jack inputs. This adaptor enables a long cable from the low impedance mic to remain balanced for rejection of hum and buzz. The male XLR connector on the end of the cable farthest from the mic is then connected to the 109B, and the 109B's phone plug is connected to the 1/4" phone jack. This approach is far superior to simple wiring a phone plug onto a 3-wire cable from the mic. The 109B not only maintains the noise rejection of the balanced low impedance mic, it also properly loads the mic to preserve correct frequency response. If you have an unbalanced input and a professional mic, the 109B is the ideal transformer.

Specifications

MECHANICAL CHARACTERISTICS

Input Selector	MIC/ATT/LINE (Ch. 1, 2) MIC/ ^{PHONO} /LINE (Ch. 3–6) ATT
Fader Length	60 mm
Assign	Odd/Even 1 / 2 3 / 4
Aux Select	Post Equalizer/Line, Switchable
Monitor Select Matrix	MONO PGM 1, 2, 3, 4 Aux Effect
Meter Select Switch	PGM 1-2 PGM 3-4 Aux/Effect
Dimensions (WxHxD)	395 x 135 x 395 mm (15-9/16" x 5-5/16" x 15-9/16")
Weight	7.0 kg (15-7/16 lbs.) net

ELECTRICAL CHARACTERISTICS

Mic Input	
Mic Impedance	150 ohms to 10k ohms nominal
Input Impedance	45k ohms
Nominal Input Level	-60 dBV (1 mV)
Minimum Input Level	-70 dBV (0.3 mV)
Maximum Input Level	+32 dBV (40 V)
Attenuation	30 dB
Line Input	
Input Impedance	10k ohms
Nominal Input Level	-10 dBV (0.3 V)
Maximum Input Level	+18 dBV (8 V)
Phono Input (Channels 3 to 6)	
Input Impedance	45k ohms
Nominal Input Level	-54 dBV (2 mV) at 1 kHz
Equalizer	
Type	Shelving
Frequency	100 Hz (Low) 10 kHz (High)
Boost/Cut	±15 dB
Channel Overload Indicator	Set to light at 25 dB above Nominal
Insertion	
Output Impedance	100 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+18 dBV (8 V)
Input Impedance	6.8k ohms
Nominal Input Level	-10 dBV (0.3 V)
Maximum Input Level	+18 dBV (8 V)
Direct Output	
Output Impedance	100 ohms

Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+18 dBV (8 V)
Sub Input (PGM/Effects/Aux)	
Input Impedance	22k ohms
Nominal Input Level	-10 dBV (0.3 V)
Maximum Input Level	+18 dBV (8 V)
PGM Insertion	
Output Impedance	100 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+18 dBV (8 V)
Input Impedance	10k ohms
Nominal Input Level	-10 dBV (0.3 V)
Maximum Input Level	+10 dBV (3.1 V)
PGM Output	
RCA	
Output Impedance	100 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+18 dBV (8 V)
1/4"	
Output Impedance	22 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	0 dBu (0.78 V)
Maximum Output Level	+20.2 dBu (8 V)
Aux Output	
RCA	
Output Impedance	480 ohms
Minimum Load Impedance	9.6k ohms
Nominal Load Impedance	48k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+10 dBV (3.1 V)
1/4"	
Output Impedance	22 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	0 dBu (0.78 V)
Maximum Output Level	+20.2 dBu (8 V)
Effect Output	
RCA	
Output Impedance	480 ohms
Minimum Load Impedance	9.6k ohms
Nominal Load Impedance	48k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+10 dBV (3.1 V)
1/4"	
Output Impedance	22 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	0 dBu (0.78V)
Maximum Output Level	+20.2 dBu (8 V)
Effect RTN	
Input Impedance	5k ohms
Nominal Input Level	0 dBu (0.78 V)

Monitor Output	
Output Impedance	100 ohms
Minimum Load Impedance	2k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10 dBV (0.3 V)
Maximum Output Level	+14 dBV (5 V)
Headphone Output	
Load Impedance	8 ohms, stereophones
Maximum Output Level	1.5 W + 1.5 W
Meter	
Peak Indicator Level	2, VU Type
	10 dB above nominal output level
Fader Attenuation	
Power Requirements	Greater than 70 dB
U.S.A./Canada	120 V AC, 60 Hz, 15 W
Europe	220 V AC, 50 Hz, 15 W
U.K./Australia	240 V AC, 50 Hz, 15 W
General Export	100/120/220/240 V, AC 50/60 Hz, 15 W

Performance Characteristics were measured:

- 1) At 1 kHz, nominal input level
- 2) At 1 kHz, 20 dB above nominal input level, 30 kHz L.P.F.
- 3) At 1 kHz, 20 dB above nominal input level, 400 Hz H.P.F., 30 kHz L.P.F.

In these specifications: 0 dBV is referenced to 1.0 Volt; 0 dBu is referenced to 0.775 Volt. Actual voltage levels are also given in parenthesis (0.316 Volt for -10 dBV is rounded off and given as 0.3 Volt, and 0.775 Volt for 0 dBu as 0.78 Volt).

Changes in specifications and features may be made without notice or obligation.

- **Accessories included:** 2 Fader-link knobs

PERFORMANCE CHARACTERISTICS

Equivalent Input Noise ¹⁾	DIN Audio/"A" Wtd
150 ohm source	130 dB/132 dB
Signal-to-Noise Ratio ¹⁾	DIN Audio/"A" Wtd
1 Mic to 1 PGM Output	68 dB/70 dB
6 Mics to 1 PGM Output	59 dB/62 dB
1 Line to 1 PGM Output	85 dB/89 dB
6 Lines to 1 PGM Output	80 dB/83 dB
1 Line to AUX Output	85 dB/90 dB
6 Lines to AUX Output	81 dB/84 dB
1 Line to Effect Output	85 dB/89 dB
6 Lines to Effect Output	78 dB/81 dB
1 Phono to 1 PGM Output	66 dB/70 dB
1 PGM Output to Monitor Output	76 dB/80 dB
Total Harmonic Distortion (THD)	
1 Mic to Direct Output ²⁾	0.02 %
1 Mic to 1 PGM Output ²⁾	0.03 %
1 Line to Direct Output ¹⁾	0.01 %
1 Line to PGM Output ¹⁾	0.01 %
1 Line to AUX Output ¹⁾	0.01 %
1 Line to Effect Output ¹⁾	0.01 %
1 Phono to Direct Output ³⁾	0.01 %
1 Phono to 1 PGM Output ³⁾	0.01 %
1 PGM Output to Monitor Output ¹⁾	0.02 %
Effect RTN to 1 PGM Output ¹⁾	0.01 %
Intermodulation Distortion (IMD), SMPTE	
1 Mic to PGM Output ²⁾	0.06 %
1 Line to PGM Output ¹⁾	0.045 %
Frequency Response¹⁾	
Mic or Line Input to Any Outputs	20 Hz – 30 kHz ⁺¹ / _{-1.5} dB
Cross-Talk ¹⁾	Better than 60 dB

Bescheinigung des Herstellers/Importeurs

Hiermit wird bescheinigt, daß der/die/das

AUDIO MISCHPULT TASCAM M-106

(Gerät. Typ Bezeichnung)

in Übereinstimmung mit den Bestimmungen der

AMTSBLATT 163/1984, VFG 1045/1984

(Amtsblattverfugung)

funk-entstört ist.

Der Deutschen Bundespost wurde das Inverkehrbringen dieses Gerätes angezeigt und die Berechtigung zur Überprüfung der Serie auf Einhaltung der Bestimmungen eingeräumt.

TEAC CORPORATION

Name des Herstellers Importeurs

TASCAM

TEAC Professional Division

M-106

TEAC CORPORATION

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