TASCAM
TEAC Production Products

M-30
Audio Mixer

OPERATION / MAINTENANCE

5700026001

The guarantee of performance that we provide for the M-30 must have several restrictions. We say that the recorder will perform properly only if it is adjusted properly and the guarantee is that such adjustment will be possible. However, we cannot guarantee your skill in adjustment or your technical comprehension of this manual. Therefore, Basic Daily Setup is not covered by the Warranty. If your attempts at such things as rebias and record EQ trim 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 M-30 in 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.

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

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 ________________

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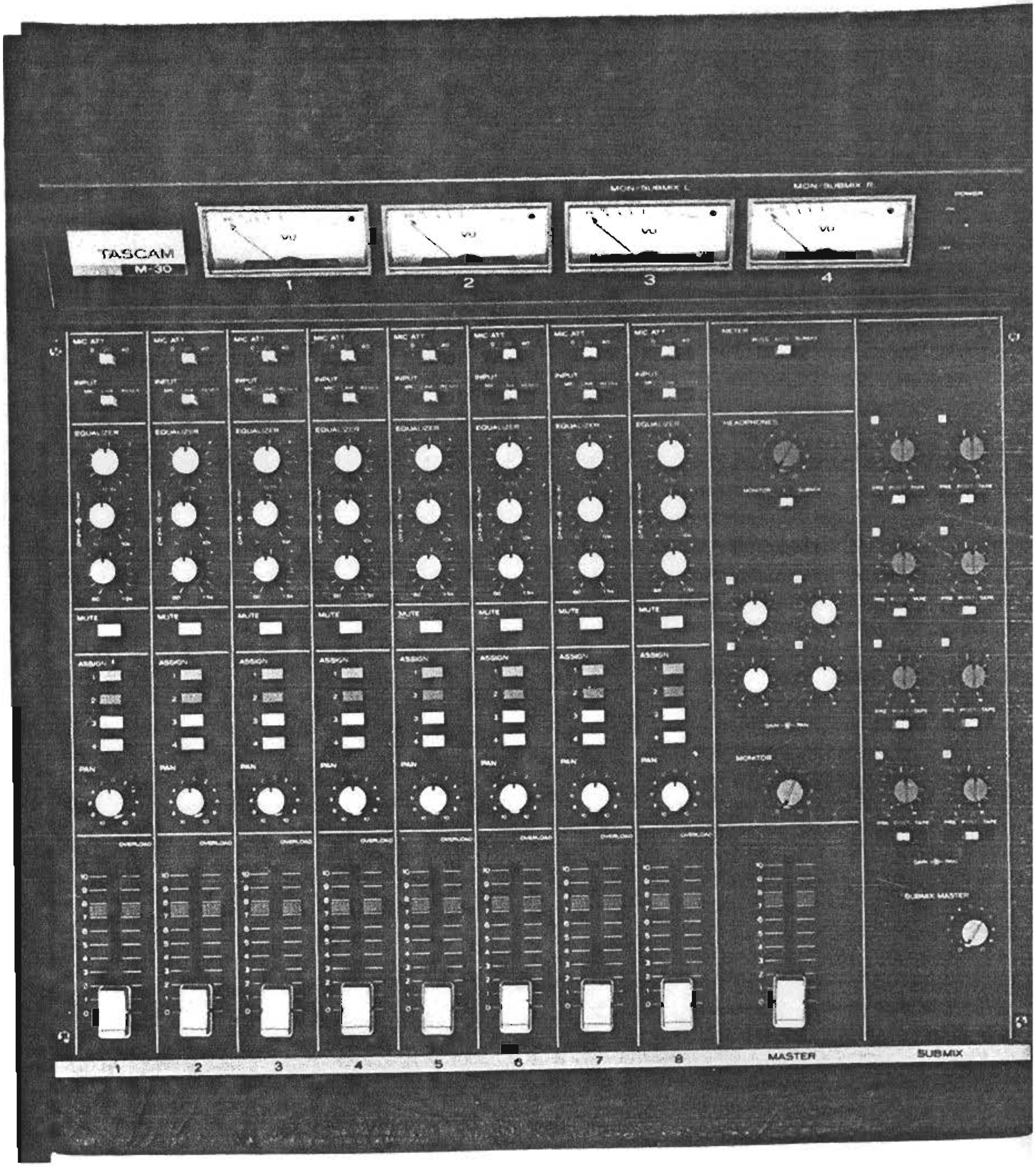
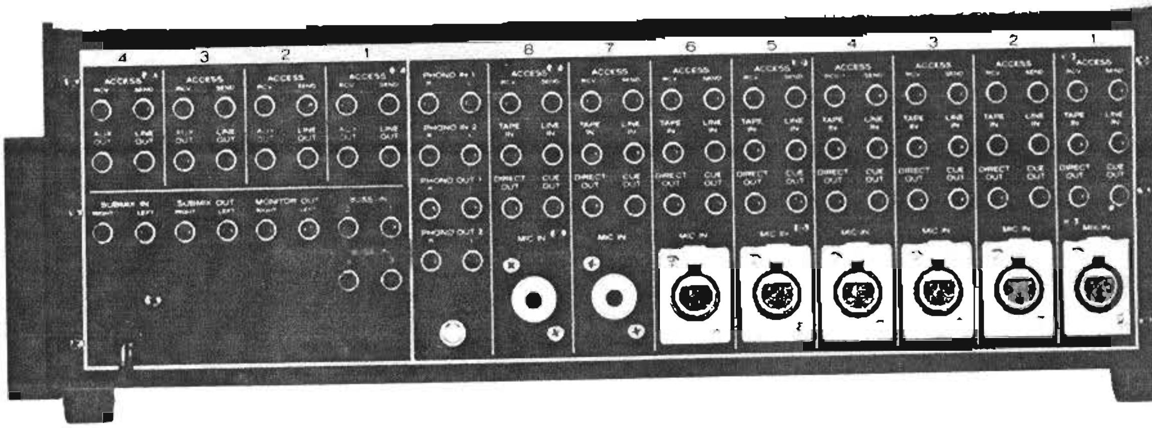
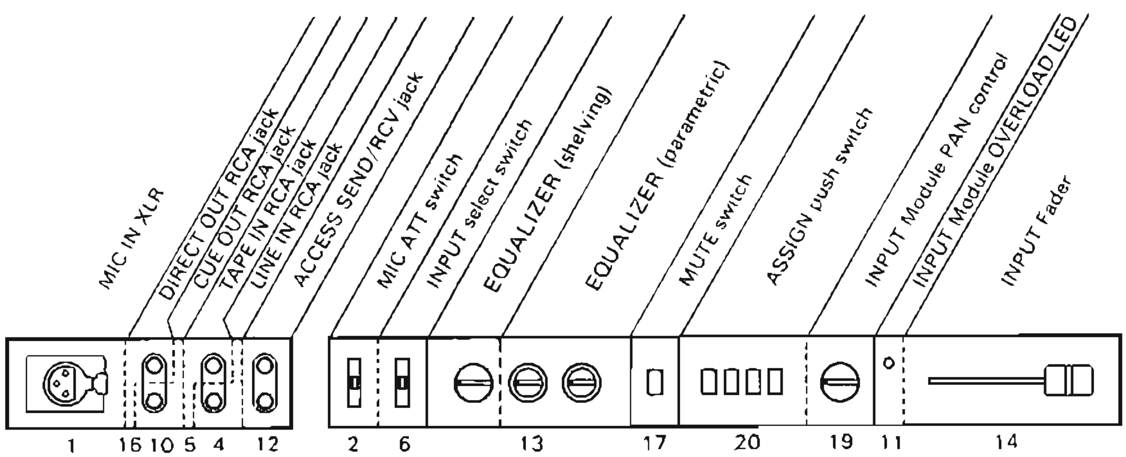
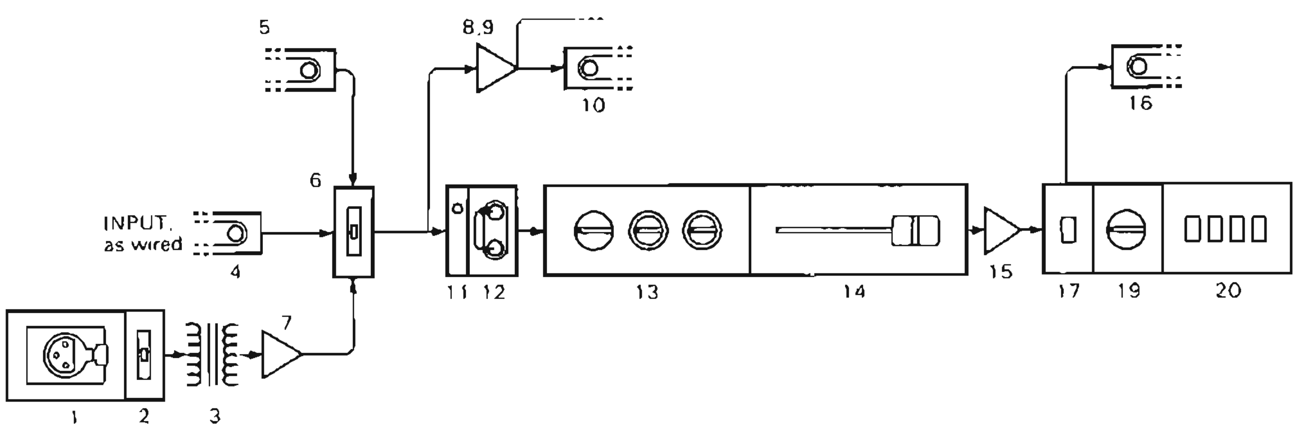

TASCAM
M-30
MASTER
SUBMIX


ACCESS
MIC IN

INTRODUCTION TO THE MANUAL, AND HOW TO USE IT.

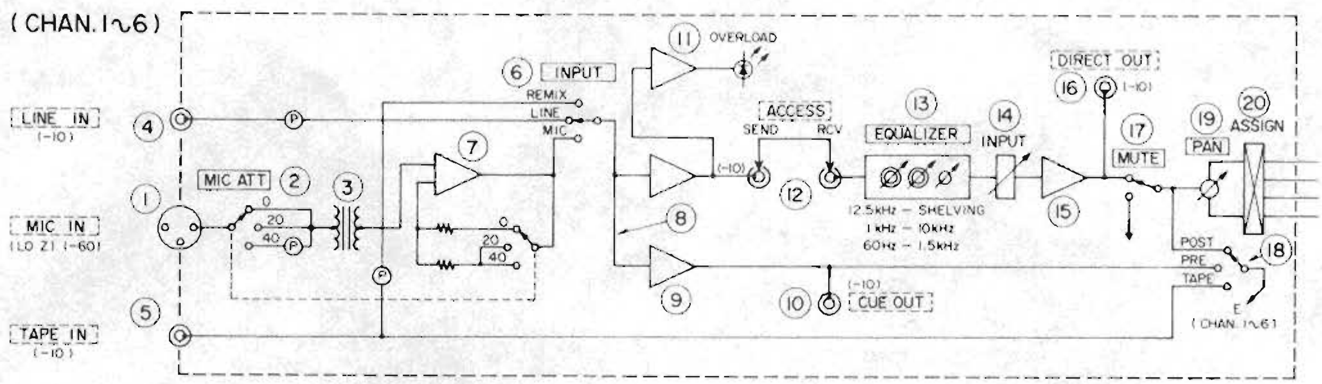
The Model 30 mixing console has been designed to satisfy the requirements of the modern multi-channel recording process. Two auxiliary mixing systems are "built in" and the SUBMIX section can be assigned and re-routed to do more than one task. Complete and convenient multitrack operation can usually be accomplished directly from the top panel, without re-patching.
However, the process of multitrack recording is constantly changing, growing more complex as an art with each advance of technology. No console can ever be built so large that it will be capable of coping with all of the switching and routing problems directly, with a "one button" top panel solution. Someone will always be able to come up with that unique situation requiring "just one more mix".
The design of the Model 30 recognizes the fact that your signal processing needs may require a unique arrangement of subsystems. In order to cope with these changing needs, patch points are provided throughout all signal pathways of the M-30.
As our mixing console becomes more flexible, the amount of time needed to understand the available function increases as well. The main signal path from "mic in" to "line out" is still fairly straightforward, as the requirements have not changed much since the days of "mono", but the routing for effects sends, cue feeds and stereo monitoring can be hard to visualize. The first time user often overlooks the significance of unfamiliar connections that are immediately obvious to the experienced recording engineer.
If you expect to find that "extra mix" quickly, you must be prepared to study the layout of the M-30 thoroughly.
In most instances, the physical arrangement of the controls on the top panel has very little to do with the sequence of electronic parts inside. The actual "wiring order" will determine what goes where and how, so this is the information





you need in order to use the M-30 successfully. As an example, if the controls on an input module were actually placed in the order that they were "wired up" inside, the top panel would look like this. We'll include the patch points from the back panel in their wiring order as well as the faders and switches. Here's the comparison:

While this arrangement of controls might help↗ the first time user to understand the flow of signal in the module, it would be very inconvenient to operate. Still the wiring sequence must be understood before the more complex functions of the M-30 can be used, so along with the documentation you will need for service (schematic diagrams, mother board layouts and mechanical disassembly information), we include a simplified electrical sequence chart called a BLOCK DIAGRAM.



This drawing shows all the controls, switches, amplifying stages and connectors in their actual "as wired" sequence. Learning to read it will provide the answers to any question concerning "what comes where" on the inside, things like "does the direct out jack come before the EQ circuit or after it?" can be answered quickly. (Yes, the direct out jack is shown connected AFTER the EQ section, so you will have EQ on any signal derived from that point.)

If you have no prior experience in the reading of block diagrams, you can use the three illustrations we have provided here as a translator. Compare the reconstructed (as wired) module drawing with the graphic used on the block to find out what each symbol represents.

Even though the block diagram can indicate what is available in the way of extra circuit flexibility, it can't explain WHY a connection or switch has been included, or suggest a standard layout for your initial setup.

In the following sections of this manual, we will do our best to describe the individual functions and controls of the M-30 and how they can be arranged in more than one sequence. In the final analysis, your mixing needs may be best served by an arrangement of inputs and sub-systems that you work out for yourself.

Some reference to the scientific terms used by our engineers will be necessary. The M-30 does nothing useful unless it is connected properly to quite a lot of sophisticated equipment. Mics, tape recorders, power amps and loudspeakers all play a part in the process of mixing/recording and each piece of gear has its own technical vocabulary. We have tried to make this reference manual as simple as technology will allow. Each section and topic will give you some basic instruction in the terminology as well as a list of "what plug" goes into "which jack".

Even though there is a substantial amount of information available to the recording engineer, much of it assumes that the reader already has an engineering or scientific background and is comfortable with "THE MATH". Practical "rules of thumb" for the musician are not generally available, and in fact, to operate a mixer no degree in science is necessary. You don't have to build a mixer "from scratch" you just need to know how to find the right control function to get the job done.

To begin our manual, we'll start with some basic information about SOUND and the numbering systems used to describe energy levels in and out of the system, IMPEDANCE – what the term means and how to deal with the details when you must connect the M-30 to other equipment. Many aspects of scientific terminology will be discussed in the most basic terms we can use.

Whenever possible, the scientific terms will be related to understandable common references. 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. If you need the numbers they will be here waiting.
Good luck with your sound.

THE dB; WHO, WHAT AND WHY

No matter what happens to the signal while it is being processed, it will eventually be heard once again by a human ear. So the process of converting a sound to an electrical quantity and back to sound again must follow the logic of human hearing.

The first group of scientists and engineers to deal with the problems of understanding how the ear works were telephone company researchers, and the results of their investigations form the foundation of all the measurement systems we use in audio today. The folks at Bell Laboratories get the credit for finding out how we judge sound power, how quiet a sound an average person can hear, and almost all of the many other details about sound you must know before you can work with it successfully.

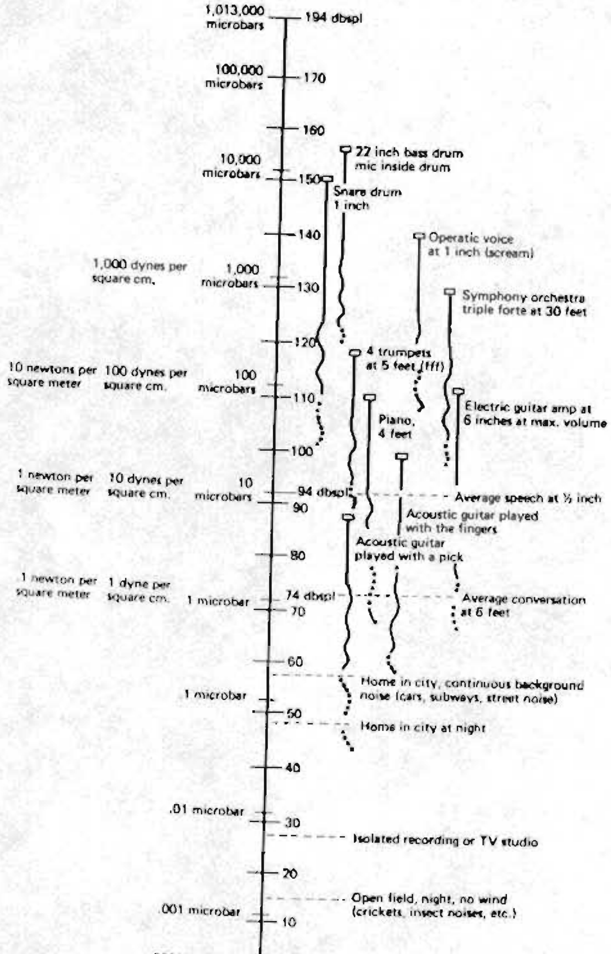
From this basic research, Bell Labs developed a system of units that could be applied to all phases of the system. Sound traveling on wires as electrical energy, sound on tape as magnetic energy, sound in air; anyplace that sound is, or has been stored as energy until some future time when it will again be sound, can be described by using the human ear-related system of numbers called "bels" in honor of Alexander Graham Bell, the inventor of the telephone.

What is a bel and what does it stand for?

It means, very simply, twice as loud to the human ear. Twice as loud as what? An obvious question. The bel is always a comparison between two things. No matter what system of units of measure you are working with at the time, you must always state a value as a reference before you can compare another value to it by using bels, volts, dynes, webers — it doesn't matter, a bel, or ear-related statement of "twice as loud" is always a ratio, not an absolute number. Unless a zero, or "no difference" point is placed somewhere, no comparison is possible.

There are many positive and definite statements of reference in use today. But before we go over them, we should divide the "bel" into smaller units. "Twice as loud" will be a little crude to be used all the time. How about one tenth of a bel? Okay, the decibel it is, and 0 means "no difference, same as the reference". It seldom means "nothing". Now, if you double the power, is that twice as loud? No, it is only 3 dB more sound. If you double an electrical voltage, is it twice as loud? No, it is only 6 dB more sound. The unit quantities must follow non-linear progressions to satisfy the ears' demand.

Remember, decibels follow the ears. All other quantities of measure must be increased in whatever units necessary to satisfy the human requirements, and may not be easy to visualize. Sound in air, our beginning reference, is the least sound the human ear (young men) can detect at 1000 to 4000 Hz. Bell Labs measured this value to be .0002 microbar, so we say 0 dB = .0002 microbars and work our way up from the bottom, or "no perceivable sound to humans" point. Here is a chart of sounds and their ratings in dB, using .0002 microbar pressure change in air as our reference for "0 dB" spl (Sound Pressure Level).



Since the reference is assumed to be the lowest possible audible value, dB spl is almost always positive, and correctly written should have a + sign in front of the number. But it is frequently omitted. Negative dB spl would indicate so low an energy value as to be of interest to a scientist trying to record one cricket at 1,000 yds. distance, and is of no significance to the multi-channel recordist. Far more to the point is the question "What is a microbar?" It is a unit of measurement related to atmospheric pressure and although it is extremely small, it must be divided down quite a lot before it will indicate the minimum pressure change in air that we consider minimum audible sound. This will give you a better idea of the sensitivity of the human ear.

One whole atmosphere, 14.70 pounds per square inch, equals 1.01325 bars. So one whole atmosphere in microbars comes out to be 1,013,250. One microbar of pressure change is slightly less than one millionth of an atmosphere, and you can find it on our chart as 74 dB spl. It is not terribly loud, but it is certainly not hard to hear. As a matter of fact, it represents the average power of conversational speech at 6 feet. This level is also used by the phone company to define normal earpiece volume on a standard telephone. Now think about that minimum audible threshold again:

.0002 microbar.

That's two ten thousandths of a millionth part of one atmosphere!

This breakdown of one reference is not given just to amaze you, or even to provide a feel for the quantity of power that moderate levels of sound represent. Rather it is intended to explain the reason we are saddled with a ratio/logarithm measurement system for audio. Adding and subtracting multi-digit numbers might be easy in this age of pocket calculators, but in the 1920's when the phone company began its research into sound and the human ear, a more easily handled system of numbers became an absolute necessity. Convenience for the scientist and practical engineer, however, has left us with a system that requires a great deal of complex explanation before you can read and correctly interpret a "spec sheet" for almost any piece of gear.

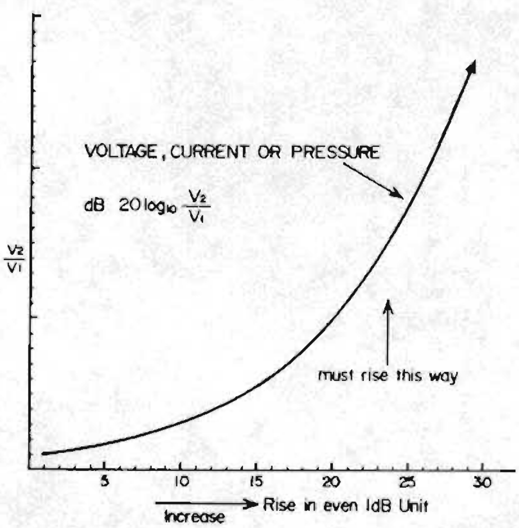
Here are the formulae for unit increment, but they are necessary only for designers. And unless you build your own gear, you won't have to deal with them. For power (watts) increase or loss, calculate by the following equation:

$$10 \text{ LOG}_{10} \frac{P2}{P1} = N \text{ (dB)}$$

For voltage, current or pressure calculations:

$$20 \text{ LOG}_{10} \frac{V2}{V1} = N \text{ (dB)}$$

Once we have this chart, we can see the difference between the way humans perceive sound and the amount of force it takes to change air pressure. Unfortunately, the result is not a simple "twice as much pressure" of sound to be heard as "twice as loud". If you plot decibels as the even divisions on a graph, the unit increase you need is a very funny curve.



This is how the ear works, and we must adapt our system to it. We have no choice if we expect our loudspeaker to produce a sound that resembles the original sound we begin with. The high sensitivity to sound of the human ear produces a strong "energy" illusion that has confused listeners since early times. How powerful are the loudest sounds of music in real power? Can sound be used as a source of energy to do useful work, such as operating a car? For any normally "loud" sound the answer is, regrettably, no! Perhaps not so regrettably, consider what would happen if one pound of pressure was applied not to your head, but directly to your inner ear. One pound of air pressure variation is 170 dB spl! This amount of "power" might do some useful work – but not much, it's still only one pound and to make use of it you will have to stand one mile away or you will go deaf immediately.

If we reduce our sound power to realistic musical values, we will not be injured, but we will have almost nothing (in real power terms) to run the mic with! This low available energy is the reason that high gain amplifiers are required for microphones.

When we take a microphone and "pick up" the sound, we do have some leeway in deciding how much energy we must have in order to operate the electrical part of our system. If we can decide that we don't have to truly hear the signal while we are processing it from point to point and we can wait until the electronic devices have done all their routing and switching before we need audible sound, we can lower the power of the signal. What is a good value for a reference here? Well, we need to have enough energy so that the signal is not obscured by hiss, hum, buzz or other unpleasant things we don't want, but not so high that it costs a fortune in "juice" or electrical power. This was a big consideration for the telephone company.

They now have the world's biggest audio mixing system, and even when they started out, electricity was not free. They set their electrical power signal reference as low as was practical at the time, and it has lowered over the years as electronic equipment has gotten better. In 1939 the telephone company, radio broadcasting, and recording industry got together and standardized 1 milliwatt of power as 0 dBm, and this is still the standard of related industries. Thus, a 0 dBm signal at a 600 ohm line impedance will present a voltage of 0.775 volts.

Once again, we owe you an explanation. Why does it say ZERO on the meter? What is an ohm? Why 600 of them and not some other value? What's a volt? Let's look at one thing at a time.

1. The logic of ZERO on the meter is another hangover from the telephone company practice. When you start a phone call in California, the significant information to a telephone company technician in Boston is – did the signal level drop? If so, how much? When the meter says ZERO it indicates (to the phone company) that there has been no loss in the transmission, and all is well. The reference level is one milli-watt of power, but the gain

or loss is in the information the meter was supposed to display, so the logic of ZERO made good sense, and that's what they put on the dial. We still use it even though it's not logical for anything else, and the idea of a reference level described as a "no loss" ZERO, no matter what actual power is being measured is so firmly set in the minds of everyone in the audio world that it is probably never going to change.

2. One ohm is a unit of resistance to the passage of electrical energy. The exact reasons for the choice of 600 ohms as a standard are connected to the demands of the circuits used for long distance transmission and are not simple or easy to explain. Suffice it to say that the worst possible thing you can do to a piece of electronic equipment is to lower the resistance it is expected to work into (the load). The lower the number of ohms, the harder it is to design a stable circuit. When you think about "load", the truth is just the opposite of what you might expect! 0 ohm is a "short circuit", no resistance to the passage of signal. If this condition occurs before your signal gets from California to Boston, you won't be able to talk – the circuit didn't "get there", it "shorted out". Once again, telephone company logic has entered the language on a permanent basis. Unless the value for ohms is infinity (no contact, no possible energy flow) you will be better off with a higher value, and many working electronic devices have input numbers in the millions or billions of ohms.

3. A volt is a unit of electrical pressure, and by itself is not enough to describe the electrical power available. To give you an analogy – that may help, you can think of water in a hose. The pressure is not the amount of water, and fast flow will depend upon the size of the hose (impedance or resistance) as well. Increase the size of the pipe (lower the resistance, or Z) and pressure (volts) will drop unless you make more water (current) available to keep up the demand. This analogy works fairly well for DC current and voltage, but alternating current asks you to imagine the water running in and out of the nozzle at whatever frequency your "circuit" is working at, and is harder to use a mental aid. Water has never been known to flow out of a pipe at 10,000 cycles per second.

This reference level for a starting point has been used by radio, television, and many other groups in audio because the telephone company was the largest buyer for audio equipment. Most of the companies that built the gear started out working for the phone company and new audio industries, as they came along, found it economical to use as much of the ready-to-hand stuff as they could, even though they were not routing signals from one end of the world to the other.

Must we use this telephone standard for recording? Its use in audio has been so widespread that many people have assumed that it was the only choice for quality audio. Not so.

A 600 ohm, 3-wire transformer-isolated circuit is a necessity for the telephone company, but the primary reason it is used has nothing to do with audio quality. It is noise, hum and buzz rejection in really long line operation (hundreds and hundreds of miles).

Quality audio does not demand 600 ohm, 3-wire circuitry. In fact, when shielding and isolation are not the major consideration, there are big advantages in using the 2-wire system that go well beyond cost reduction. It is, as a system, inherently capable of much better performance than 3-wire transformer-isolated circuits.

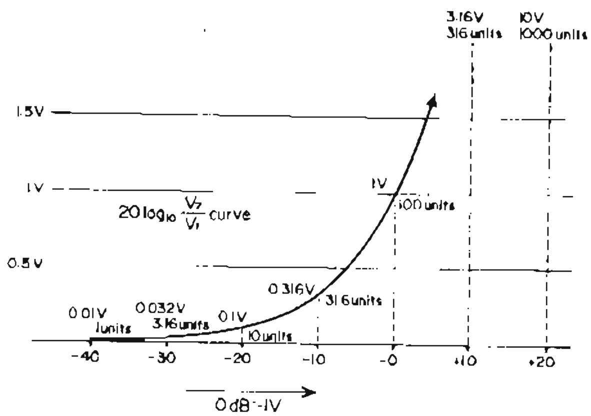
Since TASCAM M-30 mixing console is designed to route a signal from a mic to a recorder, we think that the 2-wire system is a wise choice. The internationally accepted standard (IEC) for electronics of this kind uses a voltage reference without specifying the exact load it is expected to drive. The reference is this:

0 = 1 volt

This is now the preferred reference for all electronic work except for the telephone company and some parts of the radio and television business. Long distance electronic transmission still is in need of the 600-ohm standard.

If your test gear has provision for inserting a 600 ohm load, be sure that load is not used when working on TASCAM equipment.

Now that we have given a reference for our "0" point, we can print the funny curve again, with numbers on it, and you can read voltages to go along with the changes in dB.



IMPEDANCE MATCHING AND LINE LEVELS

All electronic parts, including cables and non-powered devices (mics, passive mixers and such), have impedance, measurable in ohms (symbol Ω or Z). Impedance is the total opposition a part presents to the flow of signal, and it's important to understand some things about this value when you are making connections in your mixing system. The outputs of circuits have an impedance rating and so do inputs. What's good? What values are best? It depends on the direction of signal flow, and in theory, it looks like this:

OUTPUTS——→Plug into——→INPUTS

It is generally said that the output impedance (Z) should be as low as possible. 100 ohms, 10 ohms. The lower, the better, in theory. A circuit with a low output impedance will offer a low resistance to the passage of signal, and thus will be able to supply many multiple connections without a loss in performance or a voltage drop in any part of the total signal pathway. Low impedance values can be achieved economically by using transistors and integrated circuits, but other considerations are still a problem in practice.

1. The practical power supply is not infinitely large. At some point, even if the circuit is capable of supplying more energy you will run out of "juice".

2. Long before this happens, you may burn out other parts of the circuit. The output impedance may be close to the theoretically ideal "ohms" but many parts in the practical circuit are not. Passing energy through a resistance generates heat and too much current will literally burn parts right off the circuit board if steps are not taken to prevent catastrophic failure.

3. Even if the circuit does not destroy itself, too high a demand for current may seriously affect the quality of the audio. Distortion will rise, frequency response will suffer, and you will get poor results.

Inputs should have very high impedance numbers, as high as possible (100,000 ohms, 1 million ohms, more, if it can be arranged). A high resistance to the flow of signal at first sounds bad, but you are not going to build the gear. If the designer tells you his input will work properly and has no need for a large amount of signal, you can assume that he means what he says. For you, a high input impedance is an unalloyed virtue. It means that the circuit will do its job with a minimum of electrical energy at the beginning. The most "economical" electronic devices in use today have input impedances of many millions of ohms. Test gear, for example, voltmeters of good quality must not draw signal away from what they are measuring, or they will disturb the proper operation of the circuit. A design engineer needs to see what is going on in his design without destroying it, so he must have an "efficient" device to measure with.

SOURCE (output) $\xrightarrow{\text{Plugs into}}$ LOAD (input)

The classic procedure for measuring output impedance is to reduce the load's impedance until the output voltage drops 6 dB (half the original power) and note what the load value is. In theory, you now have a load impedance that is equal to the output impedance. If you gradually reduce the load (increase the input impedance), the dB reading will return slowly to its original value. How much drop is acceptable? What load will be left when an acceptable drop is read on the meter?

Traditionally, when the load value (input Z) is approximately seven times the output impedance, the needle is still a little more than 1 dB lower than the original reading.

Most technicians say, "1 dB, not bad, that's acceptable". We at TASCAM must say that we do not agree. We think that a seven-to-one ratio of input (7) to output (1) is not a high enough ratio, and here's why:

1. The measurement is usually made at a mid-range frequency and does not show true loss at the frequency extremes. What about the drop at 20 Hz or 30 kHz?

2. All outputs are not measured at the same time. Most people don't have twently meters, we do. Remember, everybody plays together when you record and the circuit demands, in practice, are simultaneous. All draw power at the same time.

Because of this widely misunderstood rule of thumb — the seven-to-one ratio — we will give you the values for output impedance.

True Output Impedance
Even though the true output impedance may be low, say 100 ohms, it takes a lab to check the rule of thumb, so for the practical reasons we have explained, the use of the ratio method of impedance calculation must be changed to a higher ratio. We prefer 100:1 if possible and we consider 50:1 to be the minimum ratio that we think safe. Because of this, we will give you a number for ohms that you can match, Minimum Load Impedance. No calculations, we have made them already.

Minimum Load Impedance
MAKE CERTAIN THAT YOU CONNECT NO TOTAL LOAD IMPEDANCE LOWER (numerically) THAN THESE FIGURES.

Line Output	5k ohms
Monitor Output	5k ohms
Submix Output	5k ohms
Cue Output	5k ohms
Direct Output	5k ohms
Access Send Output	5k ohms
Phono Output	5k ohms
Aux Output	5k ohms

Nominal Load Impedance
Our specifications usually show 10,000 ohms as a Nominal Load Impedance. This load will assure optimum performance. Remember, any impedance lower than 10,000 ohms is more load.

Input Impedance
Input impedance is more straightforward and requires only one number. Here are the values for the M-30.

Mic Input (Channel 1 – 6)	600 ohms
Mic Input (Channel 7, 8)	10k ohms
Line Input	20k ohms
Tape Input	50k ohms
Access Receive Input	200k ohms
Submix Input	10k ohms
Buss Input	10k ohms
Phono Input	45k ohms

If one output is to be "Y" connected to two inputs the total impedance of the two inputs must not be lower than the minimum load impedance, mentioned above, and if it becomes necessary to increase the number of inputs with slight reduction of the load specifications, you must check for a drop in level, a loss of headroom, low frequency response, or else suffer from a bad recording. If one input is 10,000 ohms, another of the same 10,000 ohms will give you a total input impedance (load) of 5,000 ohms. To avoid calculations you can do the following when you have two inputs to connect to one output.

Take the lower value of the two input impedances and divide it in half. If the number you have is greater than the minimum load impedance, you can connect both at the same time. Remember, we are not using the true output impedance we are using the adjusted number, the minimum output load impedance.
If you must have exact values here is the formula for dissimilar 2 loads or inputs:

$$Rx = \frac{R1 \times R2}{R1 + R2}$$

When you have more than two loads (inputs), just dividing the lowest impedance by the number of inputs will not be accurate unless they are all the same size. But if you still get a safe load greater than the minimum load impedance by this method, you can connect without worry.

If you must have exact values, here is the formula for more than 2 loads or inputs:

$$Rx = \frac{1}{\frac{1}{R1} + \frac{1}{R2} + \frac{1}{R3} + \ldots + \frac{1}{Rn}}$$

Rx = Value of Total Load

Finding Impedance Values on Other Brands of Equipment

When you are reading an output impedance specification, you will occasionally see this kind of statement:

Minimum load impedance = X ohms

or

Maximum load impedance = X ohms

These two statements are trying to say the same thing, and can be very confusing. The minimum load impedance says: please don't make the NUMBER or ohms you connect to this output any lower than X ohms. That's the lowest NUMBER. The second statement changes the logic, but says the exact same thing.

Maximum load impedance refers to the idea of the LOAD instead of the number, and says: please don't make the LOAD any heavier. How do you increase the load? Make the number lower for ohms. Maximum load means minimum ohms, so read carefully.

When the minimum/maximum statement is made, you can safely assure that the manufacturer has already done his calculations, and the number given in ohms does not have to be multiplied. You can MATCH the value of your input to this number of ohms successfully: but as always, higher ohms will be okay (less load).

Occasionally, a manufacturer will want to show you that 7 times the output Z is not quite the right idea and will give the output impedance and the correct load this way, they will call the output impedance the True Output Impedance and then will give the recommended minimum LOAD impedance. It may be a higher or lower ratio than 7 times and will be whatever the specific circuit in question requires.

REFERENCE LEVELS

We should talk about one more reference, a practical one.

Anyone who has ever watched a VU meter bounce around while recording knows that "real sound" is not a fixed value of energy. It varies with time and can range from "no reading" to "good grief" in less time than it takes to blink. In order to give you the numbers for gain, headroom and noise in the M-30, we must use a steady signal that will not jump around. We use a tone of 1000 Hz and start it out at a level of –60 dB at the mic input, our beginning reference level. All levels after the mic input will be higher than this, showing that they have been amplified, and eventually we will come to the last output of the M-30 – the line-out and the reference signal there will be –10 dB, our "line level" reference.

From this you can see that if your sound is louder than 94 dB spl or – your mic will produce more electricity from a sound of 94 dB spl than –60 dB, all these numbers will be changed. We have set this reference for mic level fairly low. If you examine the sound power or sound pressure level (spl) chart on page 7 you will see that most musical instruments are louder on the average than 94 dB spl, and most commercial mics will produce more electricity than the –60 dB for a sound pressure of 94 dB, so you should have no problems getting up of "0 VU" on your recorder.

We should also make a point of mentioning that the maximum number on the chart on page 7 represents "peak power" and not average power. The reason? Consider if even some momentary part of your recording is distorted, it will force a re-recording and it is wisest to be prepared for the highest values and pressure even if they only happen "once in a while". On this point, statistics are not going to be useful, the average sound pressure is not the whole story. The words themselves can be used as an example. Say the word "statistics" close to the mic while watching the meters and the peak LED level detector. Then say the word "average". What you are likely to see are two good examples of the problems encountered in the "real world" of recording. The strong peaks in the "s" and "t" sounds will probably cause the LED's to flash long before the VU meter reads anywhere near "zero" while the vowel sounds that make up the work "average" will cause no such drastic action.

To allow peaks to pass undistorted through a chain of audio parts, the individual gain stages must all have a large reserve capability. If the average is X then X + 20 dB is usually safe for speech, but extremely percussive sounds may require as much as 40 dB of "reserve" to insure good results. Woodblocks, castanets, latin percussion (guido, afuche) are good examples of this short term violence that will show a large difference between "LED flash" and actual meter movement. When you are dealing with this kind of sound, believe the LED, it is telling you the truth.

If you are going to record very loud sounds you may produce more electrical power from the mic than the M-30 can handle as an input. How can you estimate this in advance? Well, the spl chart and the mic sensitivity are tied together on a one-to-one basis. If 94 dB spl in gives −60 dB (1 mV) out, 104 dB spl will give you −50 dB out, and so forth. Use the number, on our chart for sound power together with your mic sensitivity ratings to find out how much level, then check that against the maximum input levels for the various jacks on the M-30. If your mic is in fact producing −10 dB or line level, there is nothing wrong with plugging it into the line-level connections on the mixer. You will need an adaptor, but after that it will work!

Most mic manufacturers give the output of their mics as a minus-so-many-dB number, but they don't give the loudness of the test sound in dB, it's stated as a pressure reference (usually 10 microbars of pressure). This reference can be found on our sound chart. It is 94 dB spl, 10 microbars, 10 dynes per cm^2 or 1 Newton per square meter. For mics, the reference "0" is 1 volt (dB). So, if the sound is 94 dB spl the electrical output of the mic is given as −60 dB, meaning so many dB less than the reference 0 = 1 volt. In practice you will see levels of −60 dB for low level dynamics, up to about −40 dB or slightly higher for the better grade of condenser mics available today. TASCAM recorders and mixers work at a level of −10 dB referenced to 1 volt (0.316 volt) so, for 94 dB spl, a mic with a reference output of −60 dB will need 50 dB of amplification from your M-30 or recorder in order to see "0 VU" (−10 dB) on your meter. Now, if the sound you want to record is louder than 94 dB spl, the output from the mic will be more powerful and you will need less amplification from your M-30 to make the needles on your recorder read "0 VU".

THE BLOCK DIAGRAM AND GAIN BLOCK DIAGRAM

Before you begin reading the next section of this manual, flip out the extra fold on page **44**. On this page, we have printed the block diagram. It shows the signal flow through the M-30 and it represents in simple form, the actual electronic arrangement of all the jacks, controls and gain stages from mic-in to line-out.

The diagram on page 46 indicates the gain of a reference signal, the noise level, and the available reserve gain or headroom at any point in the signal chain.

An experienced audio engineer would be able to operate the M-30 successfully with just these diagrams and a list of input and output specifications.

Any question about function or gain can be answered by studying the drawings. Will the accessory send signal change in level if the input fader is moved? No, the signal is shown leaving the main line before the input fader. You read these diagrams from left to right, input to output. When printed in its entirety, a block diagram can look formidable, and tracing a signal path is not easy, so to aid you in your initial understanding, we'll continue to use our 3-drawing system shown first in the introduction, but in slightly smaller segments.

1. As laid out for convenience.

2. As wired, but knobs and jacks as they appear on the outside.

3. The block diagram, with the controls numbered to correspond to numbers on the first two drawings.

The usual method of preparing a block diagram is to draw only one of any multiple line and label "how many" there are. For example, only one input module is drawn for a 24-input console. This saves space and reduces the complexity of the drawing.

For the M-30, we think it wiser not to use repeats, and our "block" shows how many "lines" of each type there actually are.
After we have explained what each part does, and where each part is, in position and in electrical terms, we'll start over at the first mic in, and show you some systems that can be "constructed" from the "parts list" available.

MODULE PARTS, WHERE THEY ARE AND HOW THEY WORK.

In multitrack recording consoles, a description of signal "sequence" or flow is made more confusing by the multiple uses of the same circuit, depending on what point in the process you are considering. Broadly speaking, multitrack recording has three categories; basic tracking (recording the initial track or tracks), overdubbing (adding more to the "basic") and finally, when all recording is complete, remixing to the desired final format (stereo, mono, etc.) Obviously, there is no real need to duplicate functions that can use the same parts at a later stage in the process. For example, you don't need a separate EQ section for the line in or tape in functions, a simple re-routing switch to select the current "start" signal, MIC, LINE, or REMIX (TAPE) will be all that is necessary. If we were to follow the logic of the recording process, this description of signal flow in the M-30 would have to consider an extremely large number of different possibilities, with no certainty that your working "method" would be one of the ones explained.

To save space, we will use the wiring sequence as our logical "guide" to organizing this manual, and describe the signal flow "options" as we come to them on a "once through" basis. The practical decisions about a specific "working method" must be left to you, the operator. At first this seems unusual, to say the least, but we must remind you that your total system is unknown to us (what recorder, what location, what jobs, etc.). Some "confusion" about which way is "right" is an unavoidable side effect of flexibility.

This manual presumes nothing about your past experience with the multichannel process except a willingness to learn. An expert in the field might see the significance of a control or jack with just a simple statement of its location, but in this manual we will point out "the obvious" many times just to make sure that you see the point.

Why? Consider these things:

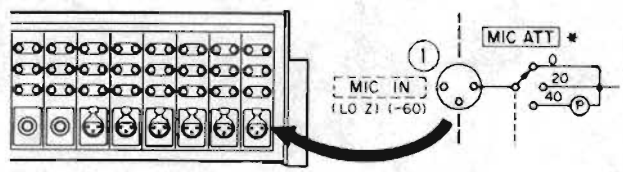
First, this manual may be the only "experience" that you can draw upon. We have said that we will explain the BENEFIT of each control as well as the way to "tune" or set it, and the science behind its function. If you have little or no practical experience, you will not FIND the advice unless we place it in a sequence that does not depend on prior knowledge of the process.

Second, the terms that we use in the index may not relate the use of the Model 30 to the field that you already know. For example, a musician will call the beginning of the tape the "top", because it relates to the top of a piece of music paper. An audio-visual producer would refer to the beginning of the program as "Home" as in "Return Home" when thinking of the rewind function. This variation in the "jargon"makes the use of a conventional index or table of contents difficult, and you have no guarantee of finding the help you need easily. Our JOB RELATED sequence of instruction is SELF INDEXING to a much larger extent.

When You Begin

It is likely that more than one person will be involved in the recording process sooner or later, but we have written this owner's manual with the assumption that the first time user will start "all alone". If you can comprehend the "solo" use of the M-30 you should have no troubles converting the logic of the unit to accommodate extra "studio staff".

1. Mic In XLR

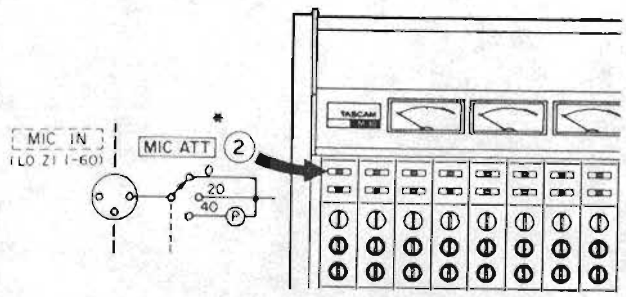


A balanced three conductor transformer isolated circuit is provided on the first six modules. Any mic with an output impedance from 50 to 600 ohms will work. Transformer primary impedance (true impedance) is 600 ohms, 1A. The MIC IN input circuit is somewhat different on the last two modules (7, 8). These INPUT MODULES are given a complete "line" drawn on the block even though the circuit is identical after you pass the first gain stage. A high gain 2 conductor transformerless input is used on modules 7 and 8. In addition to the use of two-wire microphones, this input will also accept the signal "direct" from most electric guitars and basses if the MIC ATT switch is used to lower the input sensitivity. Use the 40 dB (full right) setting as a trial. With some "hot" pickups you may have to "turn down" the instrument volume control a little as well to avoid overload. Input impedance of this circuit is 100k ohms. Max input before overload varies with the setting of the MIC ATT.

Set left – no pad* max is –30 dB (30 mV).
Set center – 20 dB pad* max is –10 dB (0.3 V).
Set right – 40 dB pad* max is +10 dB (3 V).

These specs apply to the MIC ATT on modules 7 + 8. For modules 1 – 6, the circuit is wired differently, and the results are listed in the next paragraph.

2. **Mic Attenuator Switch**



The dotted lines on the block diagram indicate that this switch acts on more than one part of the circuit. Three positions are provided.

A. Set fully left, there is no effect.

B. Set center, a "loss" of 20 dB is inserted. This loss is caused (20 dB)* by changing the gain of the first stage amplifier.

C. Set right, a "loss" of 40 dB is inserted. This loss is also caused in part (20 dB)* by a pad to protect the transformer from excessive input signal, and in part (20 dB)* by changing the gain of the first stage amplifier.

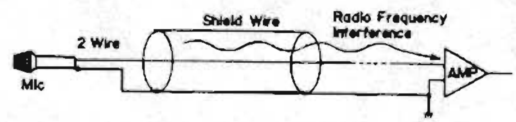
3. **Input Transformer**

This transformer does not show on the outside, but its contribution to the system is considerable. Maximum signal allowable without the use of the MIC ATT is -30 dB (30 mV). With the MIC ATT set fully right, maximum input rises to +10 dB (3 V). This XLR connector, "pad" and transformer are the only balanced "three wire" circuits in the M-30.

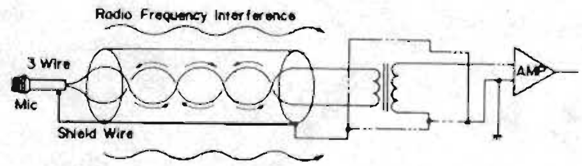
We have talked a lot about the 2-wire circuit being a better way to do the audio job, and mic lines do not run for "miles and miles" in our system. Why do we equip 6 modules with this more expensive design if it offers no improvement in quality?

The low power signal that the mic generates must often be protected and isolated from other low power signals in the real world. Radio, Power line hum, buzz, crackles and switching noise when motors start up (do you have an air conditioner on your AC line?) – 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 circuit and input isolation transformer becomes the only sure way to deal with the problem.

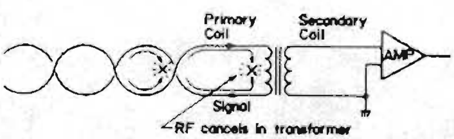
Here's how it works:



Any signal will pass to amplifier, no rejection.



Audio signals from the mic have opposite polarity. Buzz, hum, and RFI have common polarity.



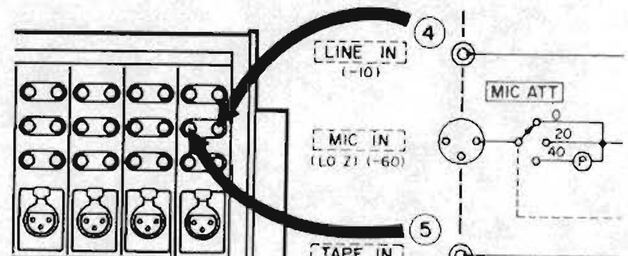
Signals with opposite polarity in the primary coil will generate current in the secondary coil. Signals with common polarity will cancel out in the primary coil and will not pass to the secondary coil. No signal in the secondary coil means no signal in the amplifier, so you leave the "junk" behind and get only the mic signal.

4. **Line In RCA Jack**

The expected level at this input is -10 dB (0.3 V). Maximum input level before clipping is +14 dB (5V), the input impedance is 20k ohms. ohms.

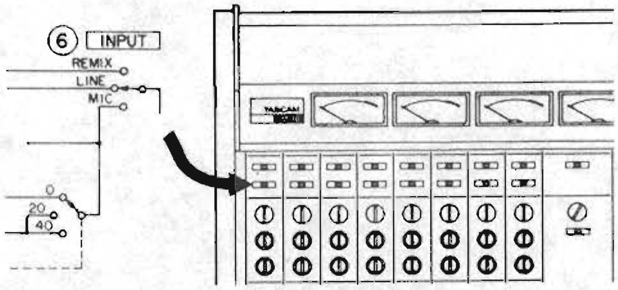
5. **Tape In RCA Jack**

The expected level at this input is -10 dB (0.3 V). Maximum level before clipping is +14 dB



(5V), the input impedance will change if BOTH the module and the SUBMIX are fed from this jack simultaneously. Worst case (both on) is 33k ohms.

6. Input Select Switch



This switch has three positions. The diagram shows a dotted line connecting two areas of the drawing and two representations of the switch. This imaginary "split" (there is only one actual control) is drawn in order to indicate that this switch also affects the gain of the first stage amp as well as switching to a different connector. Since increasing the gain of LINE or TAPE signals is not necessary, the first stage gain is reduced in these 2 modes by 3.5 dB.

The switch positions are as follows:

Set leftwards – Selects the MIC IN.

Set Center (LINE) – Selects the LINE IN connector on the back of the module.

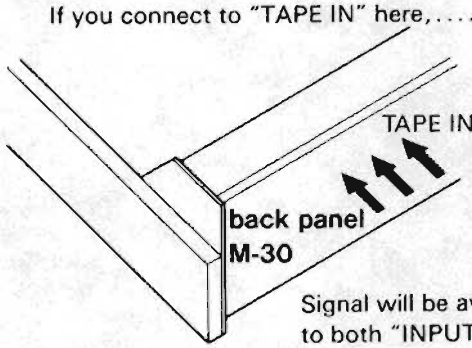
Set Rightwards (REMIX) – Selects the TAPE IN connector on the back of the module.

Since there are two high level (-10 dBV) jacks on each INPUT MODULE, what is the intended purpose of each jack? Are they just extras, or is there some difference that suggests that you use one instead of the other on a given job?

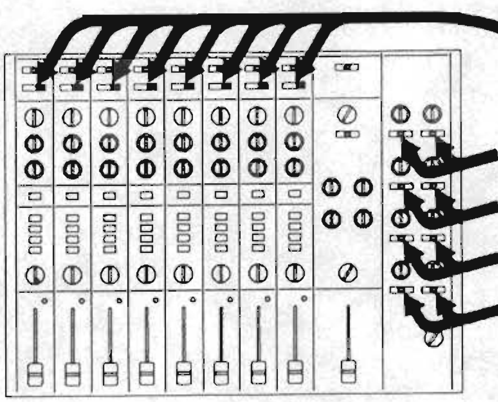
SUBMIX Uses and Functions

Here we have our first major problem in comprehension. The TAPE IN connection is plainly drawn on the block diagram, but what does it provide in terms of function? Why is the INPUT select switch wired to this extra LINE IN when there is another LINE IN on the module? What is the purpose of the line that connects the TAPE IN to the #18 switch marked PRE/POST/TAPE on the block diagram and where is this switch on the top panel? The answers lie in the requirements of a multitrack recording system "in use" and to explain, we'll have to discuss the

If you connect to "TAPE IN" here,......



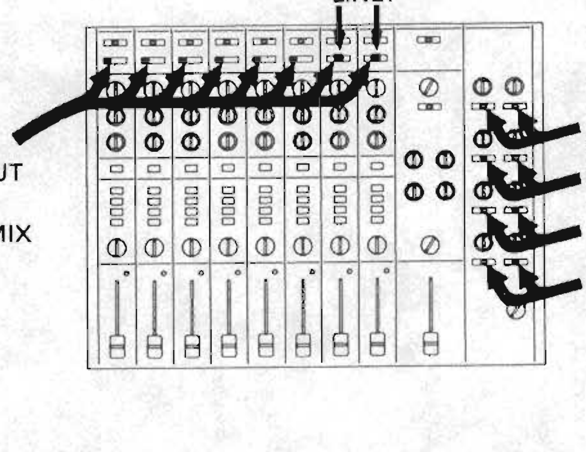
Signal will be available to both "INPUT" via "REMIX", and SUBMIX via "TAPE".



REMIX and......

TAPE IN as well

LINE?



BUT......
You may use MICS or LINES on the INPUT and still have Tape for Cue on the SUBMIX

SUBMIX still on "TAPE"

uses and limits of the SUBMIX system even though we have not yet covered the first complete path to the recorder.

We assume that any multichannel recorder has only one set of reproduce outputs. We will have at least three basic jobs to do that will require the reproduce signal:

A. Simple playback to judge a performance, requiring no corrective EQ. In short, what did you record?

B. Simple playback into a "cueing" system so partially completed tapes can be finished. This function should somehow combine the reproduce signals with "new" mic signals so musicians may hear a balance of both when "overdubbing".

C. Final remix, when the full control capability of the system (EQ, effects, etc.) can be used to "fine tune" the completed master.

Three basic tasks, one tape reproduce signal, so, to avoid resetting all the controls on the INPUT module, and "losing" the EQ and record level

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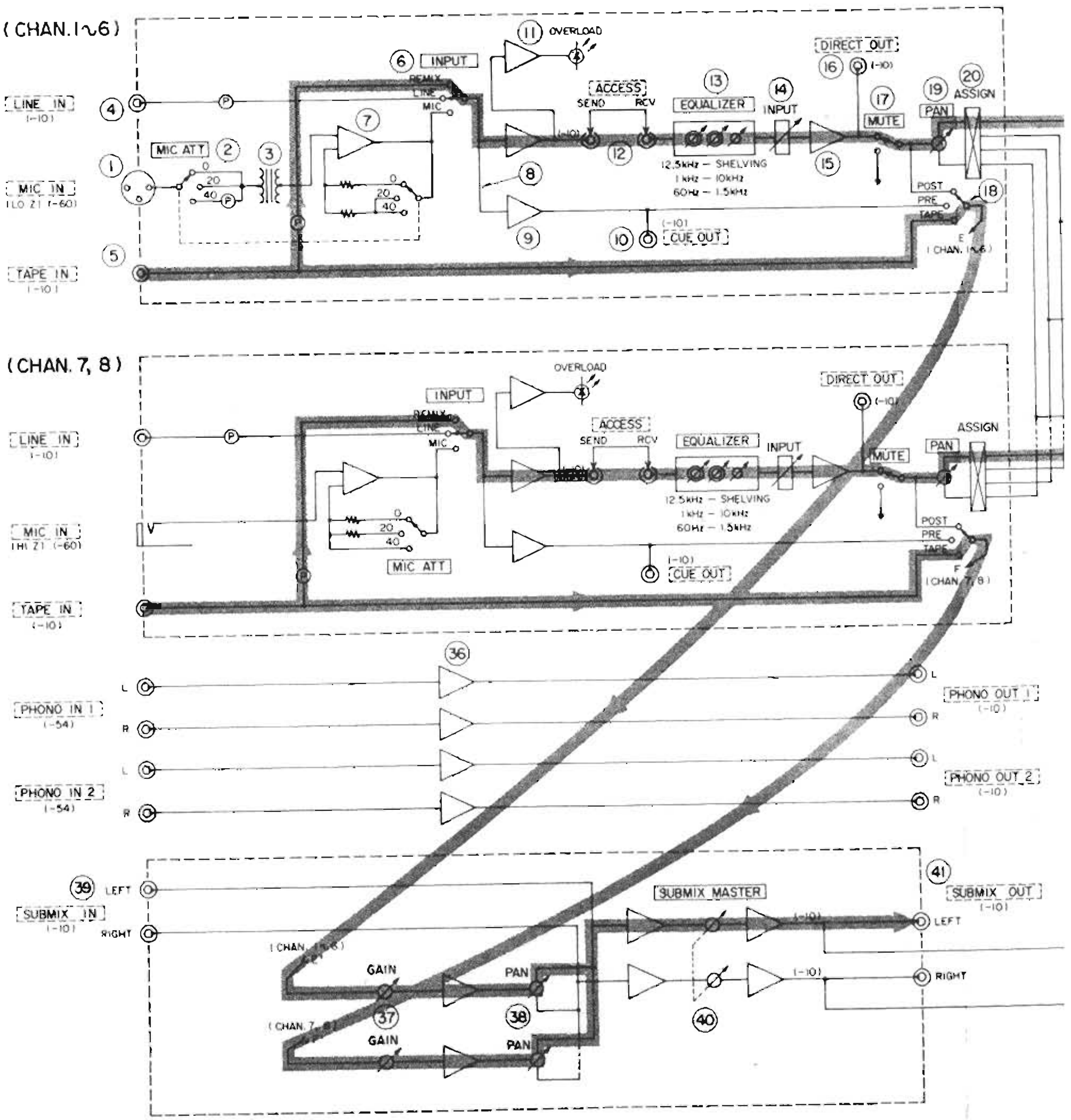
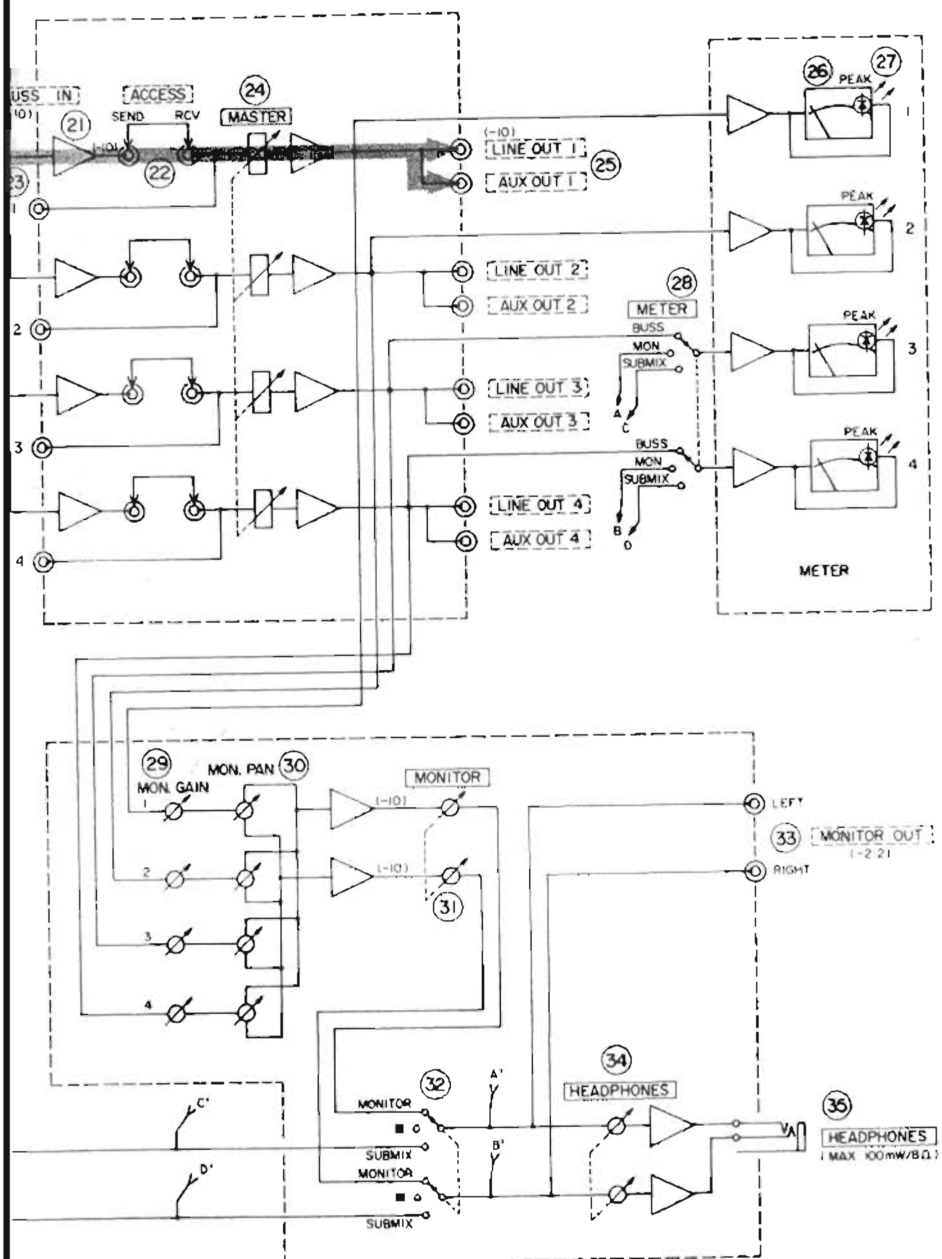


ave taken much time to get "just
e TAPE INs instead of the LINE
PUT MODULES. From this single
he reproduce signal can now be
e to both the INPUT MODULES
lect switch to REMIX) and to the
(set SUBMIX PRE/POST/TAPE
to TAPE). These two independent
v you to select a DIFFERENT
for each of the 16 sections on the
PUT MODULES and 8 SUBMIX
sections).

On the SUBMIXER, only one "route" can be active at a time, but the use of TAPE IN instead of LINE IN will allow you to use MIC IN on the INPUT while you use TAPE IN to "feed" the SUBMIX. If you use LINE IN, you will be forced to switch the INPUT select switch away from MIC to get the reproduce signal to the SUBMIX.



A. **Simple record check**

1. Connect the PLAYBACK outputs of your multitrack recorder to the TAPE IN jacks on the input modules of the M-30.
2. Move the three position SUBMIX select switch (#18) to the rightmost position marked "TAPE".
3. Set the MONITOR/SUBMIX signal select switch (#32) right, to SUBMIX.

The monitor mix will now be derived from SUBMIX, TAPE IN on each section of the SUBMIX so switched.

Many engineers use this "logic" for control room monitor all the time. "Listening" to the tape recorder electronics solves the problem of "where is the signal coming from?"

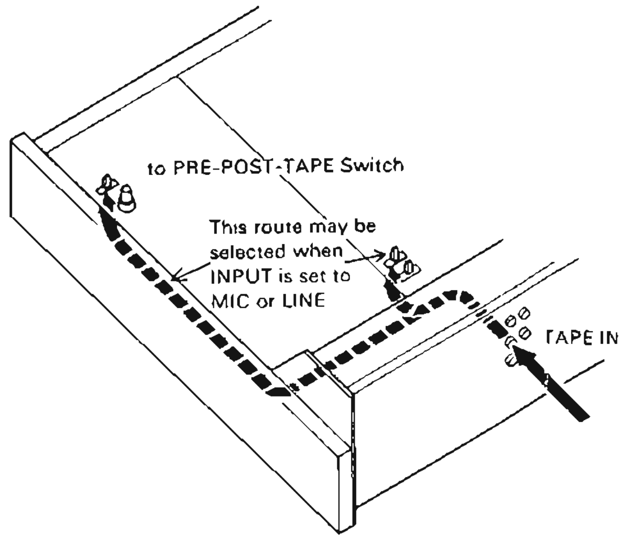
Never mind what output is feeding track three, let's just listen to track three, and we won't have to remember under pressure how we "got there". With this monitor method, ANY line level source that carries a processed signal can be considered as a "feed" to a track, even if it has no monitor capability on the way "out" of the console. You will be monitoring the "return", not the "send". Since many tracks in multichannel work can be described as "one mic per track" recording, a little thought will show you that you can use the #16 DIRECT OUTS (x8) to "shoot" more tracks than you have BUSS OUTs if you use this method.

Another advantage of this "machine monitor" is that you won't have to change ANYTHING to make a reproduction for the musicians, because you are ALREADY listening to "reproduce". The "mix" that you have been working with will be what you get in the monitor when you rewind and play the tape. We recommend this monitor method highly, it saves a lot of time and eliminates confusion.

Drawback

Since the M-30 has only one SUBMIX, and the musicians need a CUE mix. You will have to adjust this "machine monitor" to suit the cue requirement, and use it for BOTH the "control room" AND the CUE function when you use DIRECT OUTs (#16) to feed a recorder. In this mode, the control room volume set by the HEADPHONES master will not relate directly to the headphone volume so take care. If this dual use of the same mix is restrictive, and you need another, we show the addition of another SUBMIX as an accessory on page 36, suggested locations for the Model 1, 8 x 2 line level mixer.

B. **Tape in for remix**



When the INPUT module select switch is set to its rightmost position (remix) the master module TAPE IN jacks are internally connected to the starting point for line level signals on input modules 1-8.

Selecting REMIX on any or all input modules WILL NOT disable the feed to the SUBMIX. A TAPE IN signal will be available on both circuits at the same time. There are several advantages that this multiple feed offers. In remix, the CUE function is not needed, so the stereo SUBMIX section can now be used instead as the effects send. If we are now ready to send signals out to some sort of special effects device (echo, phaser, flanger, etc.) where does the processed result return to the "mix" so it can be recorded? There are TWO possible methods, one complex and another that is more straightforward.

Effects return method

Use the input modules that don't have mics plugged in. Much of multitrack production is done on a "one mic per track" basis and will leave you with unused input modules. You can take advantage of the functions that they provide to do things to the return signal. Separate EQ can be used to improve the "sound" of the effect. The two SUBMIX outputs can be used to feed one device per jack and the "spare" input modules used to control the selection and balance of the two separately. This method is not restricted to the "one mic per track" jobs, even stereo is possible. Remember that in 4 track work you will have a total of four unused modules.

CAUTION:

These complex patches can lead to a circular assignment, or FEEDBACK LOOP. To use these setups successfully, the SUBMIX select switch relating to the RETURN MODULES must not be set to PRE or POST. Make sure that you don't assign the processed signal return BACK OUT TO THE EFFECTS DEVICE by accident.

A word or two of reality

The first time user may say at this point that these extras are SO HARD TO GRASP that the benefits are not worth the risks. The M-30 is new, there are what seems like a thousand knobs, and the first few pages in the manual at first seem to be describing logic that is so sophisticated that it only makes sense to a 20 year "pro". WE AGREE! These mix patches ARE complex and their routings are not easy to visualize. We will not insult your intelligence by saying otherwise. The M-30 is a tool, not a toy. Like any good tool, good results depend on practice and understanding. You will find a use for this "deluxe patch" when your art is in need of the control that it can provide.

In 4 track operation, unused inputs are available to be used as EFFECTS receive back to the mix. In 8 track operation a more complex "return" will be necessary as there will be no "spares". We show two possible solutions to this 8 track problem on page 34 in the STANDARD PATCH section.

Resetting the input module to tape

During the course of normal multitrack production, a good "take" may be acceptable in every way except one; some doubt may arise as to the "mixability" of one track. Since the "remix" function may be selected for one input module at a time, a single track may be routed through the EQ section and a "we'll fix it in the mix" correction tried out BEFORE the musicians "pack up". You can make sure that re-recording is not required. This checkout will only require the readjustment of one module.

7. First Gain Stage

This triangle is the symbol for an amplifier, the first of many that will appear in our diagram.

8. Pre-Fader Line

9. Cue Out Buffer

Rather than load down the first gain stage with the job of driving the submix and the CUE OUT jack, another amp is provided. This amp feeds the SUBMIX select switch when PRE is selected, and the CUE OUT jack.

10. Cue Out RCA Jack

The nominal level at this output is −10 dB (0.3 V). Maximum level before clipping is +14 dB (5 V). The output load impedance (lowest permissable number in ohms) is 5k ohms. This output carries the same signal as the PRE setting of the SUBMIX select switch, and is the preferred point for CUE mixes. A mix made from this point will not change if you move the INPUT fader, use the MUTE function, or adjust the EQ. The only thing that is more frustrating to a player than having the CUE "jump around" in the headphones is to have the sound of a critical part disappear entirely.

11. Input Module Overload LED

When signals high enough to make the ACCESS SEND jack output exceed +14 dB are applied to an input module, this LED will light. Adjust the MIC pad until this LED remains out when signal is present. When working with extremely percussive transient material, you may require an additional outboard pad to prevent this LED from flashing on strong "peaks". Changing to a less "sensitive" mic may help.

12. Access Send-Receive Jacks

The high gain provided by the mic preamplifier allows us to place our first "patch point" in this useful location. The nominal level at this output is −10 dB (0.3 V). Maximum level before clipping is +14 dB (5 V), the output load impedance (lowest permissable number in ohms) is 5k ohms.

A limiter connected to this point in the M-30 can be set to a range of compression that will not be altered when the fader is moved, or the EQ is adjusted.

When no accessory device is bridged from "send" to "receive", the jumpers provided MUST be in place for signal to flow to the EQ amps and on through the M-30. There is no "normal" or automatic internal connection when the jumpers are removed.

Since all the mixing controls lie after the receive jack, it is possible to consider using ACCESS RCV as an input, and bypass the first gain stage. The only functions lost will be the MIC-LINE-REMIX switching, and the overload indicator.

This unorthodox "patch" is suggested for final "remix" when all recording has been completed, and more time for patching is available. Any successful recording will already have "level control" and you won't need the trim and overload indicators. Bypassing amps wherever possible improves signal quality. Max input level will be +14 dB (5V). The input impedance of the RCV jack is 200k ohms.

13. Three Section Semi-Parametric (Sweep) Equalizer

Before we begin, the label itself will need some explanation. What does the term "parametric" mean, and how about "equal". Equal to what? An obvious first question, because the term does not describe what you do with the controls. In multitrack audio, tone controls are almost always used to "make different" and the concept of "make the same" doesn't quite fit, how did the term "equalizer" come to be used in audio tone control?

The telephone company uses it.

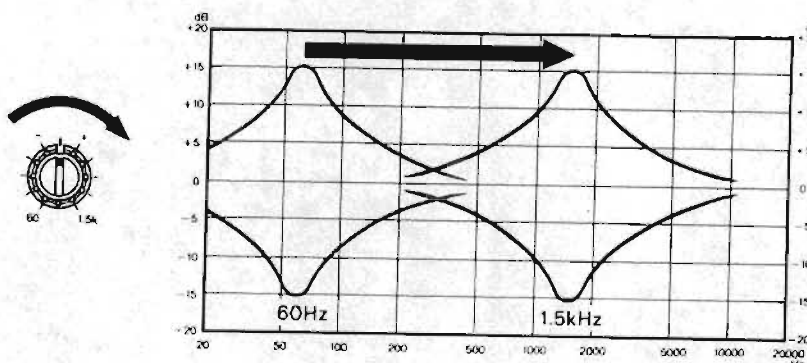
In the early days, the telephone worked well in the lab, short runs of 100 yards or so but . . . .

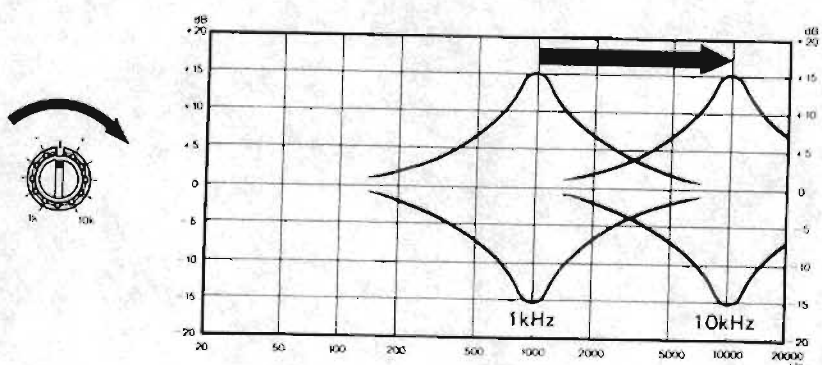
When two "phones" were 10 miles apart, the line between them did not transmit all of the sound representing signal in the same way. Some parts of the frequency spectrum did not pass down the line at all, some parts were different in level or displaced in "time". What came out of the earpiece was definitely not what had gone in 10 miles away and understanding a conversation proved to be difficult. What now? The phone company had to learn how to make the output "sound like" the input.

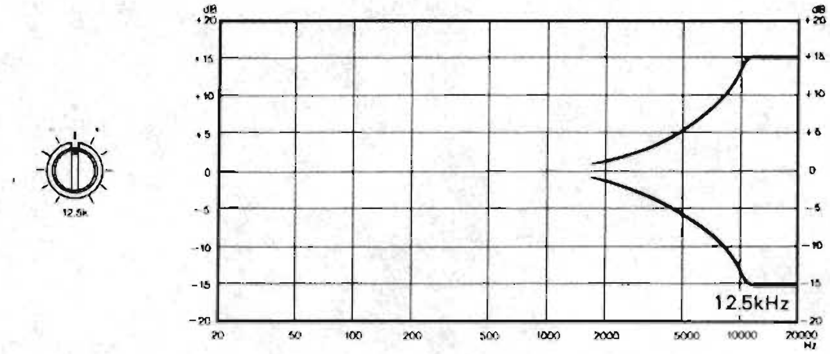
If "output equals input" is the concept, an "equalizer" is a logical name for the device used to fix your problem. Just as in many other concepts in audio, the telephone company language has established the terms we use today. The term "parametric" refers to the adjustable frequency point. The "parameters" or "rules" are not fixed at any specific number, but are continuously variable. Two aspects of the circuit, the frequency center point and the "boost" or "cut" in gain are adjustable without "steps".

Here are graphs showing the control ranges of each of the three sections. Each section provides some "overlap" of the previous section in regards to the frequency range.

Outside Control, rotate right to raise center frequency. Inner Control rotate leftwards to cut rotate rightwards to "boost".







The M-30 EQ section also provides a fixed frequency boost/cut control at 12.5 kHz. The range of control is shown on this last graph.

The great advantage of a parametric or continuously "tunable" equalizer over the more conventional "fixed center frequency" types is that you can adjust the frequency center point to the precise area you need. Now the cut or boost you use will be more effective. You get the result needed with less rotation of the control. This puts less "strain" on the electronics. No matter how many "frequencies" there are on a "set" type EQ, it is unlikely that any one will prove to be "just right" and many more ranges are needed to do the job. More sections means more parts to degrade the signal, and more money to build the EQ. In the long run a "parametric" EQ requires fewer parts to cover the same task, so it costs less, and in many cases will work even better than graphic EQs which may also leave many sections unused on a given job. To achieve a comparable "spec" a graphic has to use superior parts to counteract the effect of all those extra "sections". "Less" is always the best working concept in audio, so use the EQ after all other methods have been exhausted, move the mic, change the mic, and finally – try the "cut" functions of the EQ first.

Even experienced engineers have a tendency to forget that "cutting" the lows will have a similar effect to "boosting" the highs, and puts less of a strain on the electronics. The results are not identical, but they are close enough to warrant trying. Cut bass, raise the overall gain and see if it sounds better than just "boosting" the highs.

On the block diagram and gain chart you can see that the EQ stage has a moderate "boost and cut" range (15 dB) and a substantial reserve, or headroom (9 dB). The reserve in the circuit is necessary to maintain sufficient "headroom" when the EQ is set for maximum effect. Without some extra margin, you would have to use the INPUT ATT switch to lower the sensitivity of the input in order to avoid overloading subsequent sections of the console when using the extreme boost settings of the EQ.

14. Input Fader

The main mixing control for individual signals on the M-30. Faders, also called "pots" (potentiometers) or attenuators always cause loss in order to control signal. Gain stages in an electronic device always run "wide open" at whatever gain they are set for, unless they have provisions for actual gain adjustment. In the M-30, only one of the many amplifiers used actually has "trim", the first stage in the input module. When you advance any straight line fader on the M-30, you are just reducing the loss it causes. The entire signal flows to the next stage only if the fader is "wide open" or up all the way.

After we pass the fader, there are many options so we'll take some time to explain why each route has been included.

15. Input Module Buffer Amplifier

This amplifier has a gain of 8 dB, but its primary purpose is not really signal "boosting". It is here to isolate the fader from the effects caused by the connection and disconnection of the parts that follow; the DIRECT OUT, the MUTE switch, the POST-FADER SUBMIX switch and the PAN POT. In addition to stability, this amp provides the extra current necessary to feed the three-way multiple connection shown on the block diagram.

A. Signal feeds the POST position of the SUBMIX select switch.

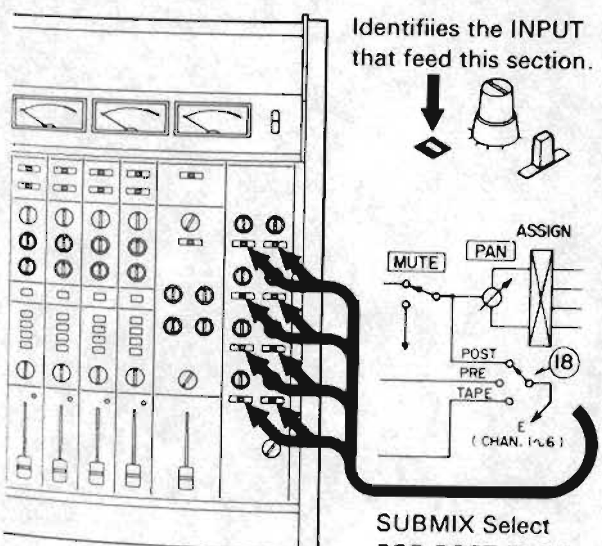
POST FADER is usually used to feed signal to an echo device when a fadeout is planned. As the INPUT FADER is lowered, the POST signal will also "fade" and you won't be left with a "ghost player" in your mix.

B. Signal feeds the DIRECT OUT jack.

16. Direct Out RCA Jack
The nominal level at this output is –10 dB (0.3 V). Maximum level before clipping is +14 dB (5 V), the output load impedance (lowest permissable number in ohms) is 5k ohms. This output can be used in combination with an outboard mixer to make up a POST fader mix, or to feed a one mic per track signal to a recorder. Going "direct" will allow you to bypass many stages and that will improve signal quality.
The MUTE function DOES NOT affect this DIRECT OUT.

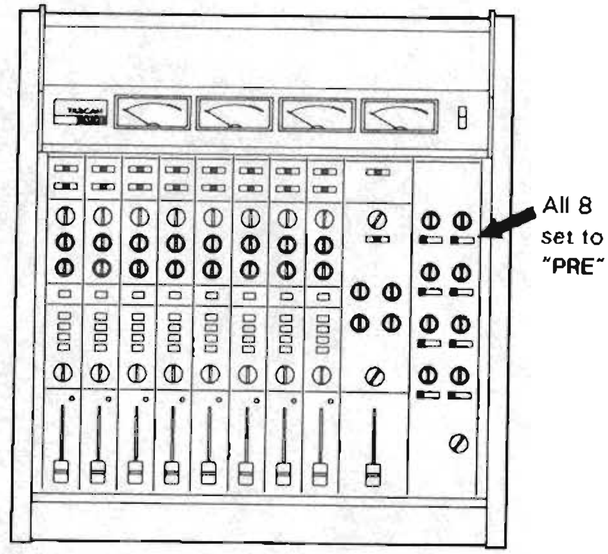
17. Mute Switch
Push to MUTE, push again in restore signal. If all switches are depressed, this mute can be used as a "reverse" SOLO system in remix. Release the one you wish to work on, and you will be able to hear a single signal without "shutting down" all the other critical fader settings just to hear. MUTE may also be used as a "one button" way to add a signal to a mix at a critical point without having to worry about bringing a closed fader back to a "mark". Set the level, use the MUTE, and at the place in the mix that you need the signal, unlatch the mute.

18. Submix (Pre/Post/Tape) Select Switch
Even though this switch is not on the INPUT module "Strip" it is logical to deal with it now because all of the signal options that it controls are derived either from signal points in the INPUT chain or the patches on the INPUT back panel. First, where is it physically?

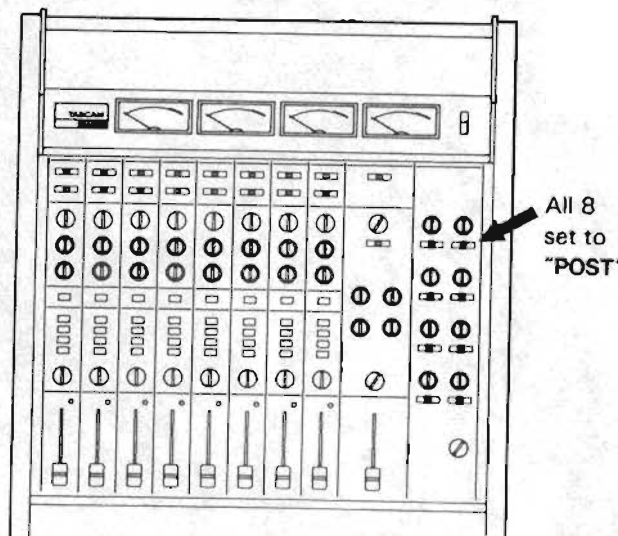


SUBMIX Select
PRE-POST-TAPE
switch(×8)

The numbers on the SUBMIX that are ABOVE each control group correspond to the INPUT section that the signal will be derived from, either internally, or from the back panel TAPE IN jack. To make the function of each SUBMIX select setting clear, we show the controls on the INPUT module that will remain active.
A. When set left to PRE – signal enters SUBMIX BEFORE ACC SEND-RCV.

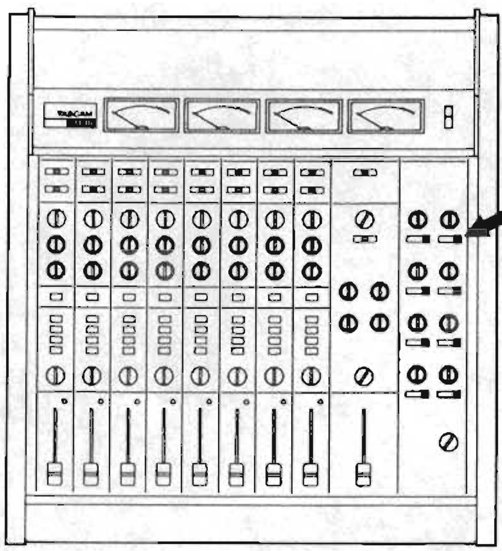


Only the MIC ATT and the INPUT signal select switch are active when PRE is selected. This is the setting we recommend for CUE mixes. Mix action will not disturb rehearsals. This point is the same as CUE OUT (#10).
B. When set center to POST – signal enters SUBMIX after the MUTE function.



If this point is used, EQ and fader action WILL affect the SUBMIX, anything "patched in" to the INPUT ACC SEND-RCV will contribute its effect to the SUBMIX signal as well as to the main line and DIRECT OUT. This signal point is the same as the DIRECT OUT with one function added. MUTE DOES WORK on the SUBMIX signal. MUTE DOES NOT WORK on the DIRECT OUT.

C. When set right to TAPE — signal enters SUBMIX directly from the TAPE IN jack on the back panel.



This drawing is included to re-inforce the point that the TAPE position of the SUBMIX select switch will provide you with a TOTALLY INDEPENDENT source that can be completely different than the signal that appears in the module. The only restriction is that the TAPE IN is a line level input. The SUBMIX section has a maximum gain of only 8 dB and that's not enough to raise a mic to line level.

Before we complete the description of the remaining SUBMIX controls we should return to the INPUT module and finish the route out through the mixer to the main LINE OUTs.

The final signal feed from the BUFFER AMP is to the input module pan pot.

The ASSIGN switches and the PAN POT make up the last wiring order sections on each input module. At this point you have selected signal source, processed and routed it, decided whether or not it will also go from PRE or POST to a SUBMIX section, it is available at the CUE (PRE) and DIRECT OUT (POST) jacks for insertion into a subsidiary mixer, and its "main line" and DIRECT levels have been set with the INPUT FADER. What LINE OUT do you want to feed? LINE 1, LINE 2, LINE 3 or LINE 4? How about stereo? When more than one button is down in the ASSIGN section the PAN POT will work.

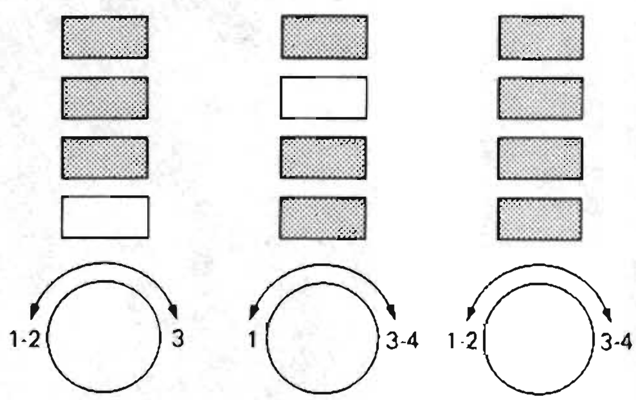
19. Input Module Pan Control

This single knob works two faders that are wired "back to back". As you rotate, one fader is turned up as the other is turned down. At dead center, each control is down just enough to compensate for the fact that two sources will have the same signal. This is primarily an active stereo pan requirement. The compensation allows you to "pan" a signal from left to right without getting an unwanted "rise" in sound power as you pass through "center".

20. Assign Push Switches

Push to enable, push again to release. Each button enables the feed to its corresponding master "group" or BUSS (LINE OUT). Any or all switches may be locked down (assigned) simultaneously. When any TWO buttons are down, the PAN is engaged. The two button logic is always high number, RIGHT, low number, LEFT, high number.

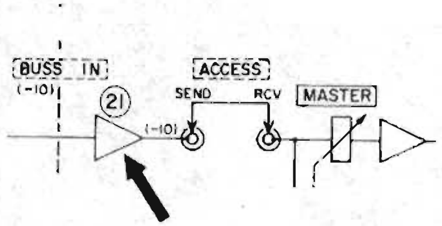
The logic of three or four button "arrays" is best shown by some drawings. If the shaded buttons in the ASSIGN group are assumed to be "down", the logic is as follows:



Typical examples of multichannel panning

21. Master Buss 1 — 4 Combines Network and Summing Amplifiers

These amplifiers don't show on the outside but their contribution to the system is considerable. These devices allow the eight input modules to add their signals together without one module distorting the output of another. When you wish



to "combine" or "sum" two or more varying voltages that are being used to represent sounds, a simple "joining together" of the wires will not work. There is no "one way sign" on a piece of wire. First, you must cause the energy to flow in the direction that you want. A network of resistors can be arranged in such a way as to make the "forward" pathway the easist "way out" of the network and the route back in to the adjacent module very "difficult". Simple, but this network reduces the signal level a lot, so this unavoidable loss must be made up by another amplifying stage. This method of resistive net and gain make-up is referred to by engineers as the "Brute Force" method and is widely used. With no user controls on the outside, you might think that a discussion of "summing networks" would be unnecessary, but understanding these principles will make clear why you can't use a simple "Y" adapter to add two signals together.

22. Master (Buss) Access Send-Receive RCA Jacks

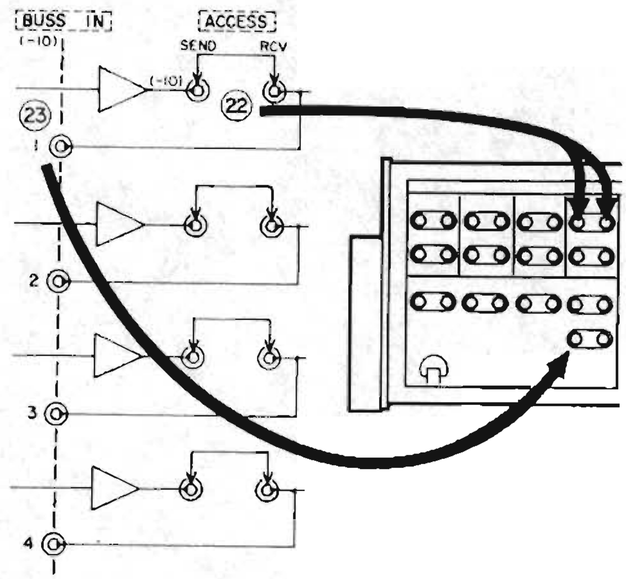
This pair of jacks is used to add an accessory or Effects device (echo, flange, what have you) to the entire group of signals. When no device is "bridged" across these jacks, the jumpers must be in place for signal to flow, as there is no "normal" or internal connection.

The nominal level at this output is −10 dB (0.3 V). Maximum level before clipping +14 dB (5 V), the output load impedance of the SEND (lowest permissable number in ohms) is 5k ohms.

The expected level at the RCV is -10dB (0.3V). Maximum level before clipping is +14dB (5V), the input load impedance is 200k ohms.

23. Buss In RCA Jacks

The normal use of these inputs is to "stack" a pair of mixers. Signal may also be added to the mix with this patch point. The expected level at this input is −10 dB (0.3 V). Maximum level before clipping is +19 dB (8.9 V), the input load impedance is 10k ohms.



24. 4 Section Buss Master Faders (Line 1 – 4)

This quad slide fader controls the output from the buss summing amps. It is a "grand master" and the individual levels of each line are not settable separately. You use it to simultaneously adjust these 3 different "feeds".

A. The level sent to the LINE OUT.
B. The level shown on the meters.
C. The level sent to the MONITOR signal select switch when the #32 MONITOR/SUBMIX select switch is in MONITOR (left setting) position.

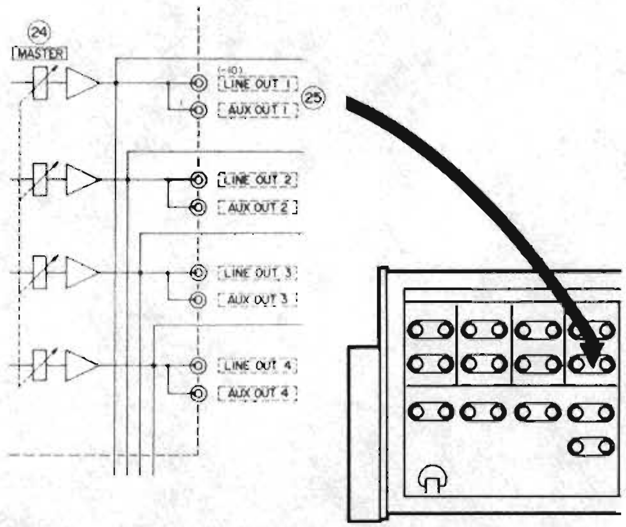
Master fader isolation amplifiers

All these places that the signal must go to represent a lot of "load" so another amplifier is provided to isolate the fader from this multiple "V" connection.

Many electronic devices have only a single control for setting an output power, or a listening level. In short, there is only one "right way" to turn up the effect, whatever that effect might be. The M-30 is not so simple, and its use as a tool for the manipulation of sound requires the adjustment of many intermediate stages of control. If you have a choice, which knob or slider should you use? What do you set first?

The overall performance of the system will improve if you start with the master and input faders set roughly on the "marks". Don't run the master "wide open" and the input faders 1/10 up, and conversely, don't run the master near the bottom and then complain that you don't have enough "gain" to get the meters up to "zero".

25. Line Out and Aux Out (Buss) 1 – 4 RCA Jacks



The final output of your mix. All functions have been applied to the signals. The only controls that remain are the sections of the mixer that allow you to see and hear what you are doing; the meters and the monitor feeds. We'll go "up" on the block diagram and deal with the meters first.

The nominal level at these outputs is -10 dB (0.3 V). Maximum level before clipping is +14 dB (5 V), the output load impedance (lowest permissable number in ohms) is 5k ohms.

Since the LINE OUT and AUX OUT jacks are wired together to provide the same signal out, be sure that the 5k ohms figure is the minimum number of ohms for the loads connected to BOTH JACKS, if you are feeding from both LINE OUT and AUX OUT at the same time.

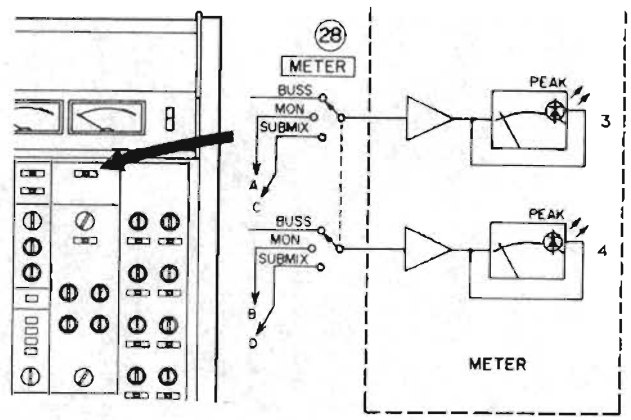
26. VU Meters 1 – 4

These four meters have the standard volume unit ballistic, they respond to the AVERAGE level of the signal, not the PEAK level. The 0 dB point is set to equal 0.3 V (–10 dB referenced to 1 volt).

27. Meter LEDs

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 percussion material (kick drum, latin percussion such as castanets or the Brazilian instrument called an afuche) 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 and "average" is not always a safe concept. For example, castanets should be recorded with NO MORE than a –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 M-30 meters to jump when AC power is first applied, and the headphone amplifier may produce a substantial transient "pop" even if the HEADPHONES master pot is fully rotated leftwards (off). Patching in mics and accessories with the power off will insure that you don't damage your ears, the M-30 or any other equipment that you may be using. Take care.

28. Meter Select Switch (Meters 3, 4 Only)



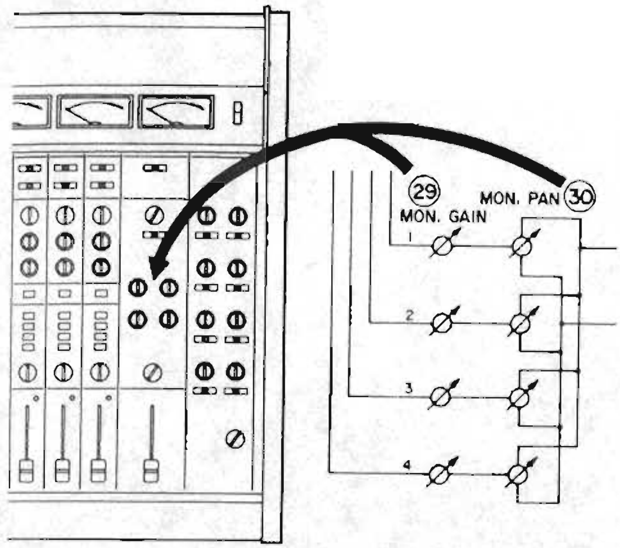
The last two meters on the M-30 can be switched to read three different signal groups.

A. Set LEFT to BUSS – METERS show level of the LINE OUTs 3, 4.

B. Set CENTER to MON – METERS indicate signal level being sent to the MONITOR OUTPUT RCA jacks. Since this signal may be switched from MONITOR to SUBMIX, the meters will indicate whatever you have assigned to this final stage.

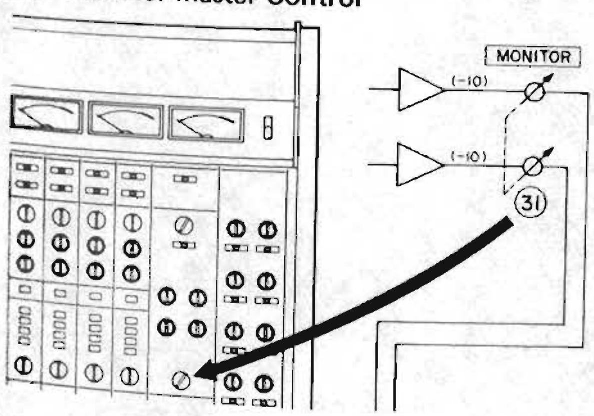
C. Set RIGHT to SUB – METERS indicate signal level being sent to the SUBMIX OUTPUT RCA jacks and to the MONITOR/SUBMIX select switch. This is not a redundant function. Yes, you can "see" SUBMIX level by using the center position, but a separate position will allow you to check levels BEFORE you throw the MONITOR select switch and thus avoid a serious or potentially unpleasant (TOO LOUD) mismatch in power.

29. **Monitor Gain** (upper section of dual concentric control)
30. **Monitor Pan** (lower section of dual concentric control)



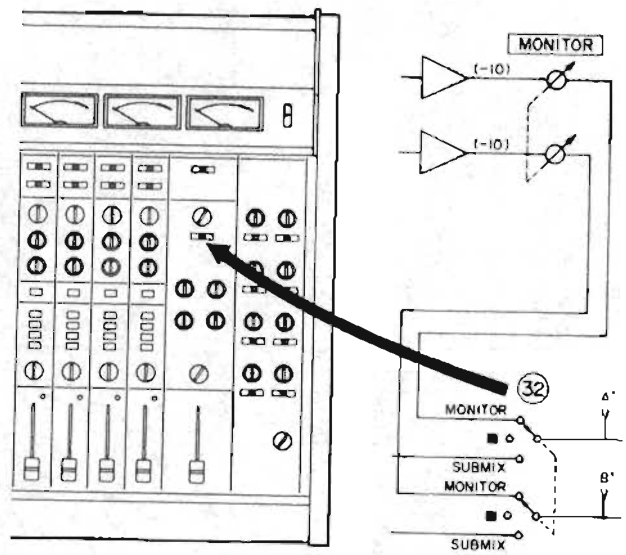
On the block, these two controls are shown separated, but on the top panel they are "stacked", one on top of the other. Each section controls the signal from the line (buss) number 1 – 4 that is marked ABOVE and slightly to the left of the control group. Even though there is no separate adjustment provided for the LINE OUTs (the MASTER FADER #24 is a 4 section control). These discrete controls will allow you to adjust the listening level of each buss (LINE OUT) to more closely approximate a "real" balance of the four signals while you are recording, and PRIOR to final mixdown. Remember, if you adjust some of the lines to lesser levels just to allow for a musical pre-judgement in the monitor, some tracks will not be well recorded (too low) and your recording will be compromised.

31. **Monitor Master Control**



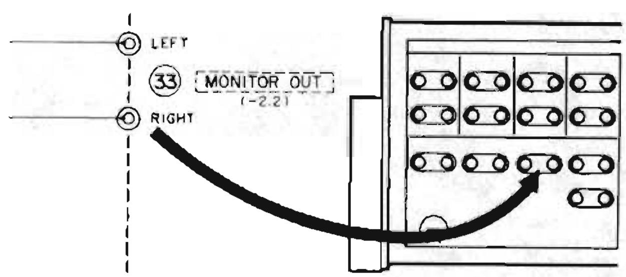
This rotary control adjusts the overall level of the four signals sent to the MONITOR/SUBMIX switch.

32. **Monitor/Submix Signal Select Switch**



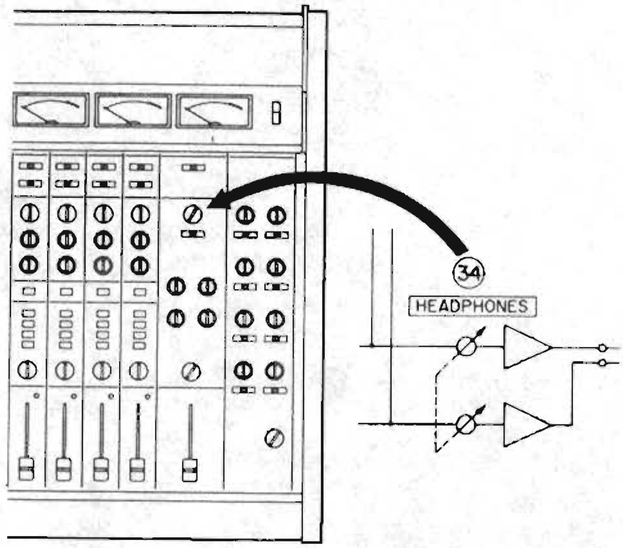
If you examine the block diagram you will see that this signal select determines what will appear at BOTH the L + R MONITOR OUTPUT RCA jacks and the HEADPHONES tip-ring-sleeve final stereo output (#34).
Set LEFT to MONITOR – BOTH outputs will have the 4 section buss (LINEs 1 – 4) monitor signals assigned.
Set RIGHT to SUBMIX – BOTH outputs will be fed the final mix created by the submixer controls and switches.

33. **Monitor Out L + R RCA Jacks**



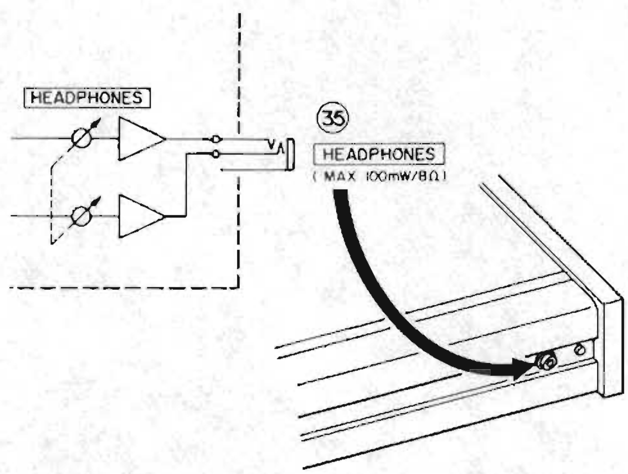
The nominal level at this output is –2.2 dB (0.775 V). Maximum level before clipping is +14 dB (5 V), the output load impedance (lowest permissable number in ohms) is 5k ohms. The expected use of this output pair is to send signal to a power amp and loudspeakers for control room monitoring.

34. Headphone Master Volume Control



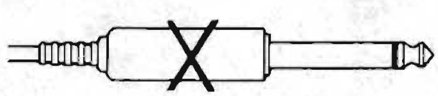
This rotary control will allow independent adjustment of the headphone volume.

35. Tip-Ring-Sleeve Stereo Output Jack

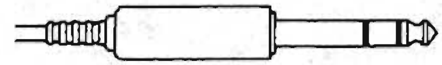


Maximum power before clipping is 100 mW, the output load impedance is 8 ohms.

CAUTION:
MONO (2 WIRE) HEADPHONES WILL CAUSE EVENTUAL CIRCUIT FAILURE. If your "phones" have this connector, don't use them.



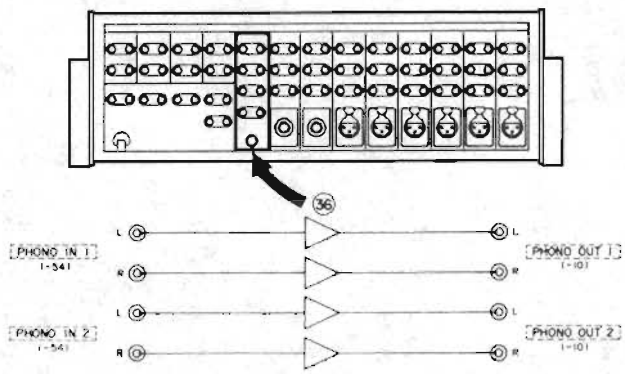
(1/4" phone 2-connector)



(1/4" phone 3-connector)

To be safe, the headset connector must have three sections. We realize that in any patchable system, accidents can happen. We do build protection circuits in to assure that a momentary mis-connection will not cause instant failure, but just because the circuit seems to work OK when you try it for a moment or two, don't assume that we are overly cautious and keep on. Sooner or later it WILL blow (2 to 3 minutes). The reason? When the "sleeve" of the 2 wire phone jack is inserted it will connect both outputs together "head to head" and this is not a usable signal combining method.

36. Phono Amplifiers 1 and 2



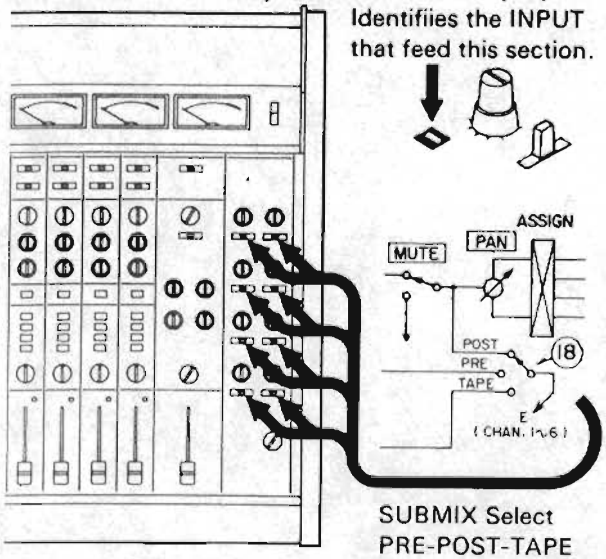
These two separate groups of four RCA jacks each are provided to allow the use of two stereo phono cartridges as signal sources. Each section will equalize and amplify one stereo cartridge to line level so it can be "patched in" to an input on the M-30. Any line level input pair that you find convenient may be used. There is NO internal connection provided, so you WILL have to "patch" to use a phonograph as a signal source.

The nominal level at this input is -54 dB(2 mV) at 1000 cycles. Maximum level before clipping is -30 dB (31.6 mV), the input load impedance is 45k ohms. The GAIN of this circuit is 44dB at 1000 cycles.

The nominal level at this output is -10 dB (0.3 V) at 1000 cycles. Maximum level before clipping is +14 dB (5 V), the output load impedance (lowest permissable number in ohms) is 5k ohms.

Now for the rest of the "parts" in the SUBMIX SECTION.

Submix (Pre/Post/Tape) Select Switch (18)



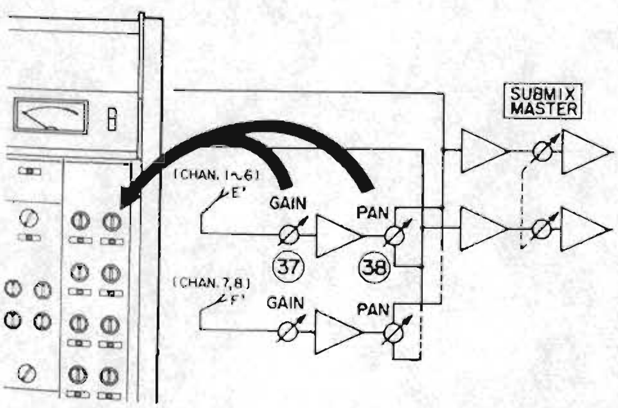
SUBMIX Select
PRE-POST-TAPE
switch(×8)

Even though this switch is located on the SUBMIX section, the fact that ALL the signal options that it controls are on the INPUT MODULE have led us to place a complete description of its function in an earlier part of the manual. See page 25 for the full story.

Submix System Dual Concentric Controls

37. Submix Gain
38. Submix Pan

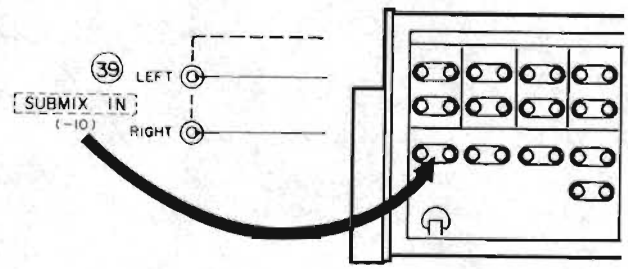
On the block, these two controls are shown separated, but on the top panel they are "stacked", one on top of the other.



The upper section controls the amount of signal. To increase, rotate to the right.
The lower section of this control is a "pan". Full left rotation sends signal to the SUBMIX L output amp ONLY.
Full right rotation sends signal to the SUBMIX R output amp ONLY. Any setting in between will divide the signal proportionally between the two amplifiers. The stereo effect can be auditioned in the monitor circuits by selecting SUBMIX (set switch #32 to the RIGHT).

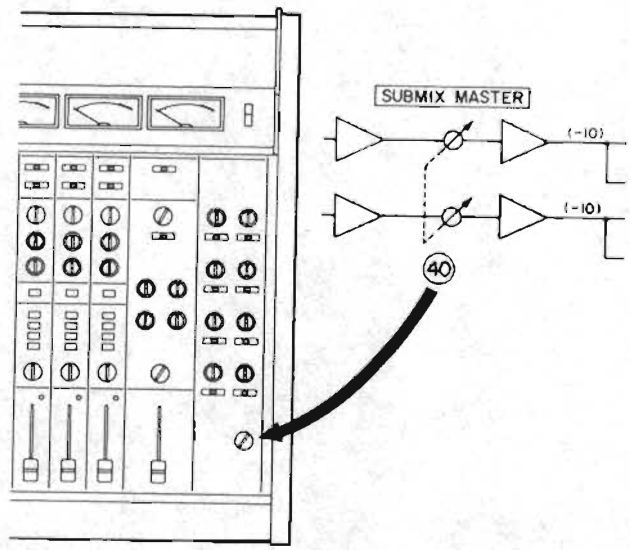
39. Submix In L + R RCA Jacks

The nominal level at this input is −10 dB (0.3 V). Maximum level before clipping is +14 dB (5V), the input load impedance is 10k ohms.



This input is provided in order to "stack" two mixers together, but it may be used to add any appropriate signal to the SUBMIX. Level control of this signal must come from the device, there is no individual gain or pan on this input pair.

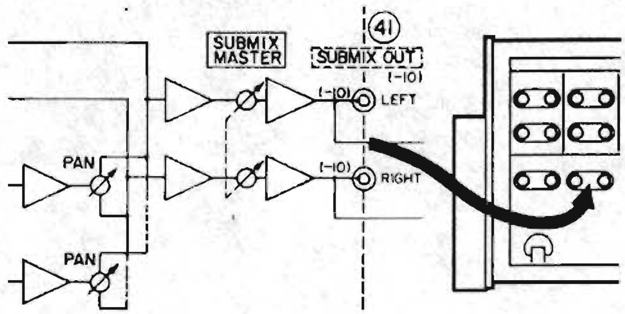
40. Submix Master



This rotary control provides overall level adjustment of the SUBMIX signal sent to the SUBMIX OUT L + R jacks, the METER select switch (#28) and to the MONITOR/SUBMIX select switch (#32).

41. Submix Out L + R RCA Jacks

The nominal level at this output is −10 dB (0.3 V). Maximum level before clipping is +14 dB (5 V), the output load impedance (lowest permissable number in ohms) is 5k ohms.



This output pair can be used for a variety of accessory functions, effects sends, cues, monitor mixes and the like, what you connect here will depend on the job you need done at the time.

We have now reviewed and described every control and control route on the M-30. In the next section of this manual you will find some suggestions for basic system layouts that make use of the controls.

STANDARD PATCHES AND SET-UP ADVICE

The standard patching setups described here are not rigid commands. Rather, they are provided with the hope that they will stimulate your imagination when you have mixing needs that cannot be solved with a standard setup. Line level is line level, whatever the source, and the many line level inputs on the M-30 can offer solutions to your specific problems that we have not addressed directly with a dedicated top panel control, or subsystem. The jacks on the back are there to be used. Patching is not a crime, and may be used to improve the quality of your signal by bypassing unneeded controls, or by making additional control possible in unorthdox ways.

Most people tend to look for a "permanent" set-up of connections in order to reduce the complex logic to something that can be dealt with "under pressure". It is true that the logic of the control functions on the top panel takes some time to become familiar with, but multi-channel recording has many mixing requirements. A permanent "patch" will severely restrict flexibility. If you can learn to examine the system with re-patching in mind, you can achieve significant improvements in system performance. For this reason we suggest that you plan on access to the back panel of the mixer. Don't set up the system in such a way that you "hide all that mess". Leave yourself room to get at all the connectors. You will need all the options you can get.

When unorthodox patches are used and the top panel labels are no longer correct, we strongly suggest that you take the time to re-label each control to correspond to the new function that your re-patch is controlling. Drafting tape labels applied to each control or group will prevent accidents from happening because you have tried to operate the mixer "normally".

It is also wise to label both ends of every cable. When re-patching away from "normal" a label will save endless tracing and re-tracing of the wiring.

In all patching and connecting of two-wire single ended circuits some basic rules are worth mentioning:

1. Keep your cable runs SHORT! – as short as possible. Installing a patch bay behind the engineers chair will require at least 20 foot runs out and back and is not recommended. Left or right side mounting will allow much shorter runs, and wisest of all is to use our PB-64 patch bay accessory mounted on top of the meter bridge itself. This location will

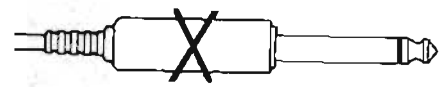
permit the use of the shortest lengths of cable, and will improve your sound. Incidentally, short runs cost less so you will save money as well.

The use of "professional" 3-conductor cable such as Belden B451 should be avoided. Even though it is of excellent quality, it is not the right idea for 2-wire transformerless systems. If you are going to make up your own cables, we would suggest that you consider TEAC LOW LOSS cable in the 500 foot rolls, or cable such as Belden B218. Solid core insulation and low capacitance are the important considerations in the 2-wire system. Some low capacitance cable uses soft foam insulation and is also not recommended. The center conductor will cut through the soft foam with time and the cable will short circuit. Don't use it.

TEAC LOW LOSS cable has less than 15 pF per foot of capacitance and uses a very durable material for its insulator. In the "made up" lengths we offer, the connector is a heavy duty RCA jack made of steel that will stand up to the demands of constant patching and re-patching without breaking down.

2. Multiple output connections always require impedance matching calculations. Make sure you are not asking too much of your output stages. Permanently connecting several cables to a single output may produce poor quality. If you are not using a patch, unplug it! Convenience can cost you quality. Be certain that a multiple connection is well within safe limits. Use the section on impedance calculations in this manual, abide by the rules for 2-wire circuits we have discussed, and you will get better results.
3. Using a "Y" cable to "sum" or join two outputs in order to feed one input WILL NOT BE POSSIBLE. Since there is no "one way" sign on a wire, signal from one side of the "Y" will flow back into the other side as well as on to the input of the next device. Summing, or adding two signals together requires that they be properly isolated, a simple joining together of the "hot" leads will not work. We recognize the fact that with a patchable system, accidents will occur. We have built protection circuits in to insure that a momentary mis-connection will not cause instant failure, but – just because it seems to function when you try it for a moment or two, don't assume that we are overly cautious and keep on. Sooner or later it WILL blow (2 to 3 minutes) and it is definitely not a usable method of expanding mixer flexibility.

When using the STEREO headphone circuit on the M-30, a similar caution applies. The use of "MONO" headphones will cause circuit failure. If your "phones" have this connector, don't use them.



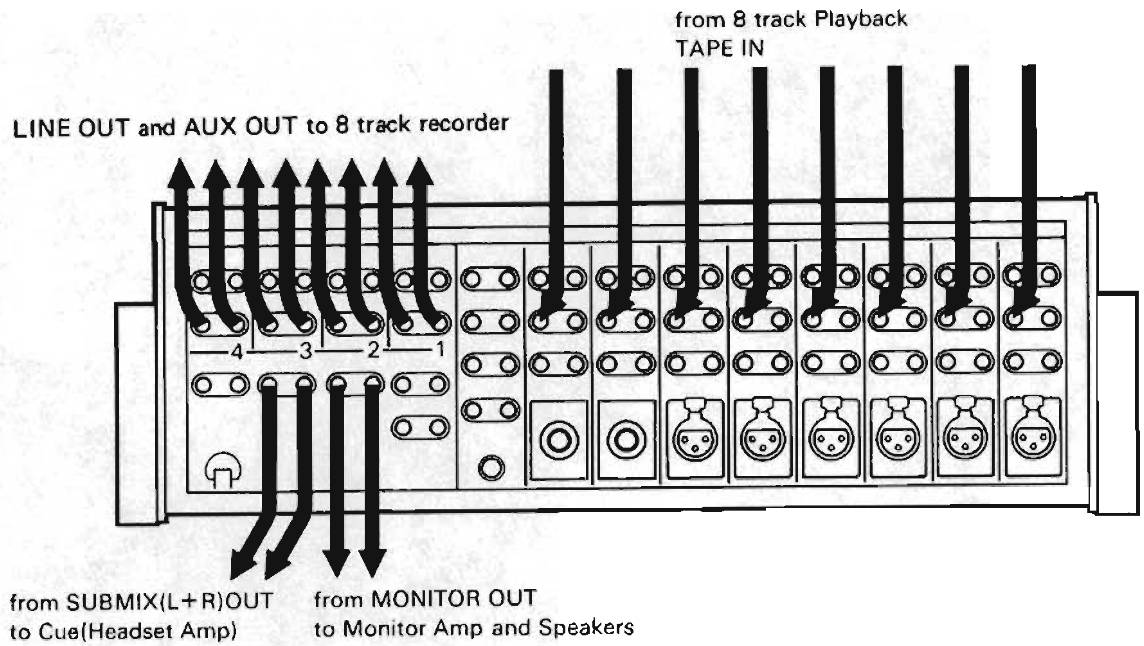
(1/4" phone 2-connector)

(1/4" phone 3-connector)

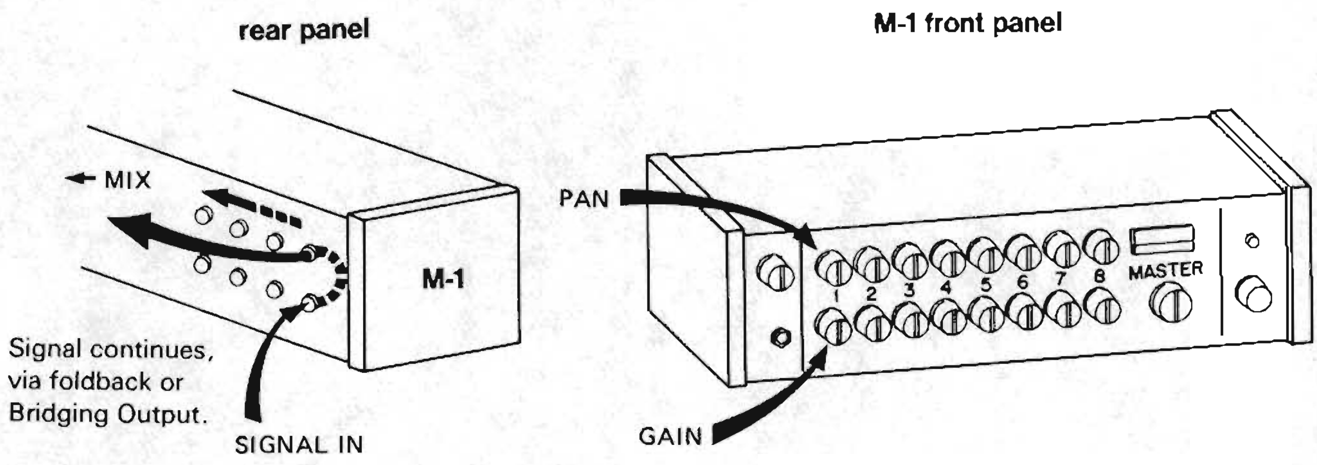
The "sleeve" of the 2 section plug will connect both sides of the stereo headphone amplifier together in the "head to head" mode. To avoid this, you must have THREE bands on the plug. It is also a good idea to check the wiring to make sure that the three sections are actually wired individually. Look for this discrete configuration when you unscrew the protective cover on the connector.

Recommended 8-Track Set-Up

This set-up will require the use of either the main inputs for playback while overdubbing, or the use of the same mix for both cue AND monitor functions. The monitor gain and pan cannot be combined with the submix for a 12 source "feed unless you use an outboard mixer and then you will still have only one cue mix for both jobs".



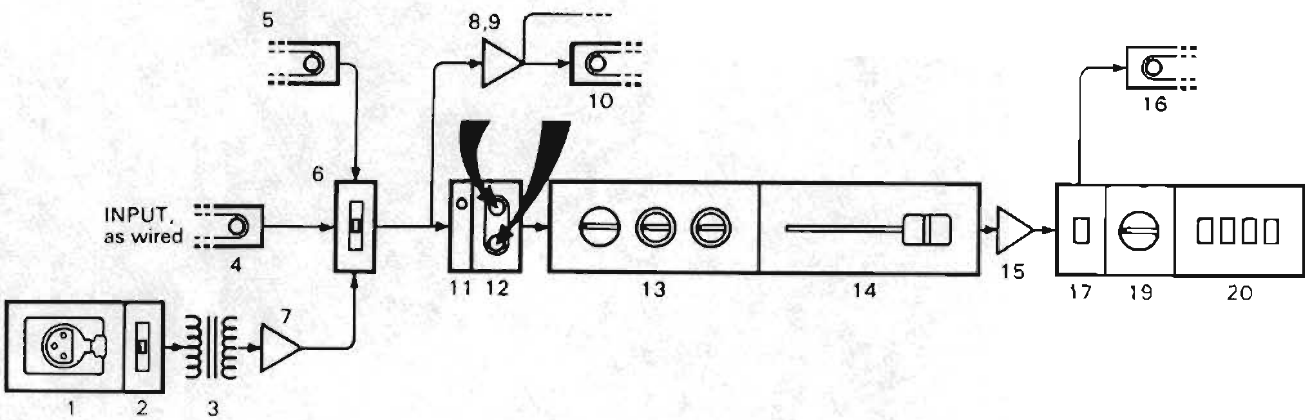
The M-30 has only one submix for cue and effects. If you need to have another accessory system made up of pre- or post-fader signals, we offer an accessory line level device called the Model 1. It will "mix" 8 signals to stereo without permanently "using up" the possibility of a second connection of the signal source, it works this way — .



All 8 inputs use this "pass through" or bridging input method, so you can "pass through" on your way to another necessary connection, and get two mixes or more for one signal source group of 8, two groups of 4, or "what have you". In our illustration we show first:

1. A "pass through" from an 8 track recorder, this "mix" can be used as a stereo cue, or an effects-send without EQ. Since it is "pre" everything, it will stay on and not be affected by any console control. "The cue out" jack will also provide a signal that is "pre fader, pre-equalizer", but from module signals as well as tape.
2. In this second model, one patch point uses "pass through" again, and signal-by-signal is patched through the accessory send-rcv point on each input module. You now have a pre-fader, pre-equalizer STEREO cue mix in addition to the ONE that is built-in.

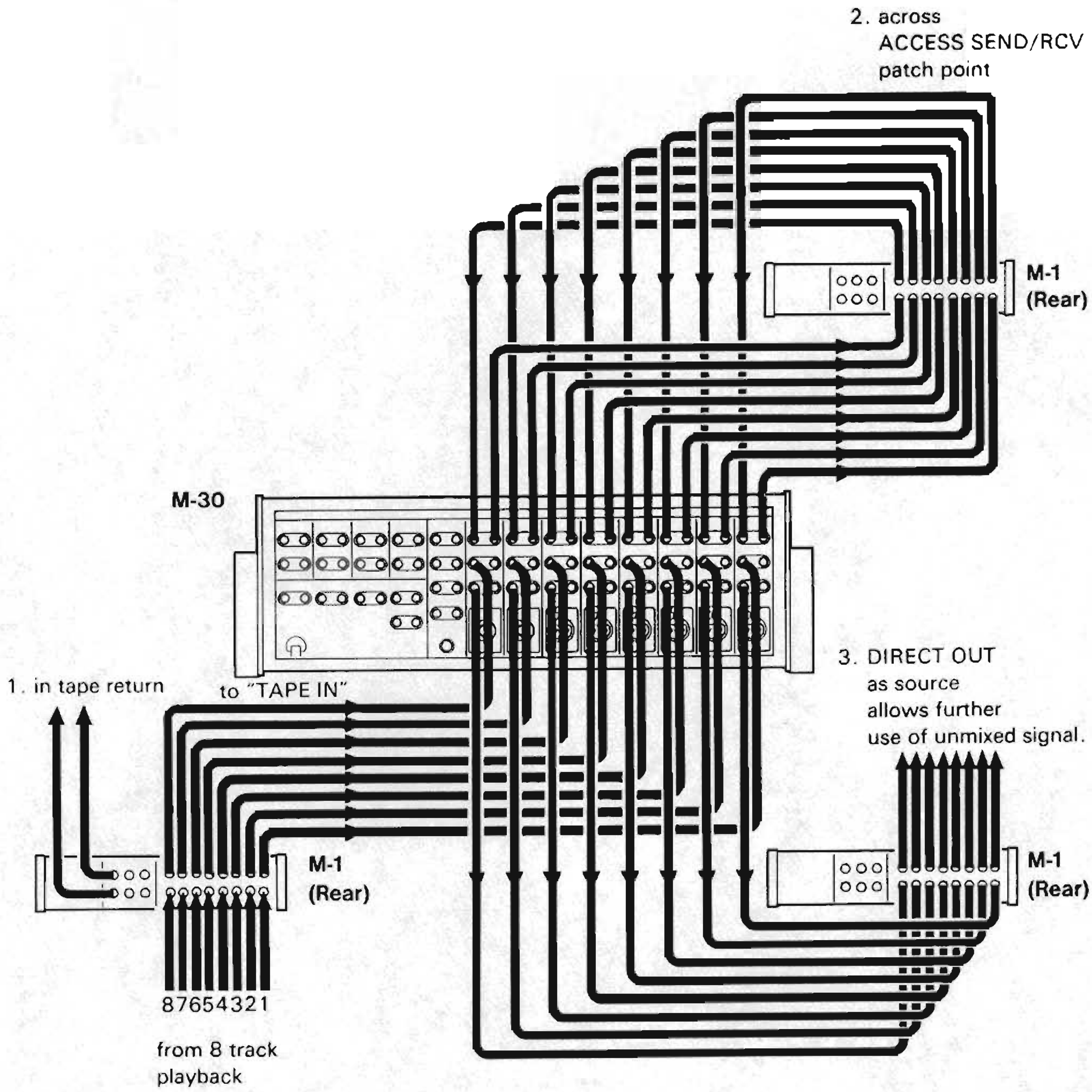
Need two? Cascade a pair this way, from access-send to one, then pass through the second, and then back to access-rcv. Since each input load is 22k ohms, the actual load on the accessory send is 11k and is safe. We don't recommend more than two here. At this point you get anything assigned to the module; "tape tracks", "line ins" and "mic ins" as well.



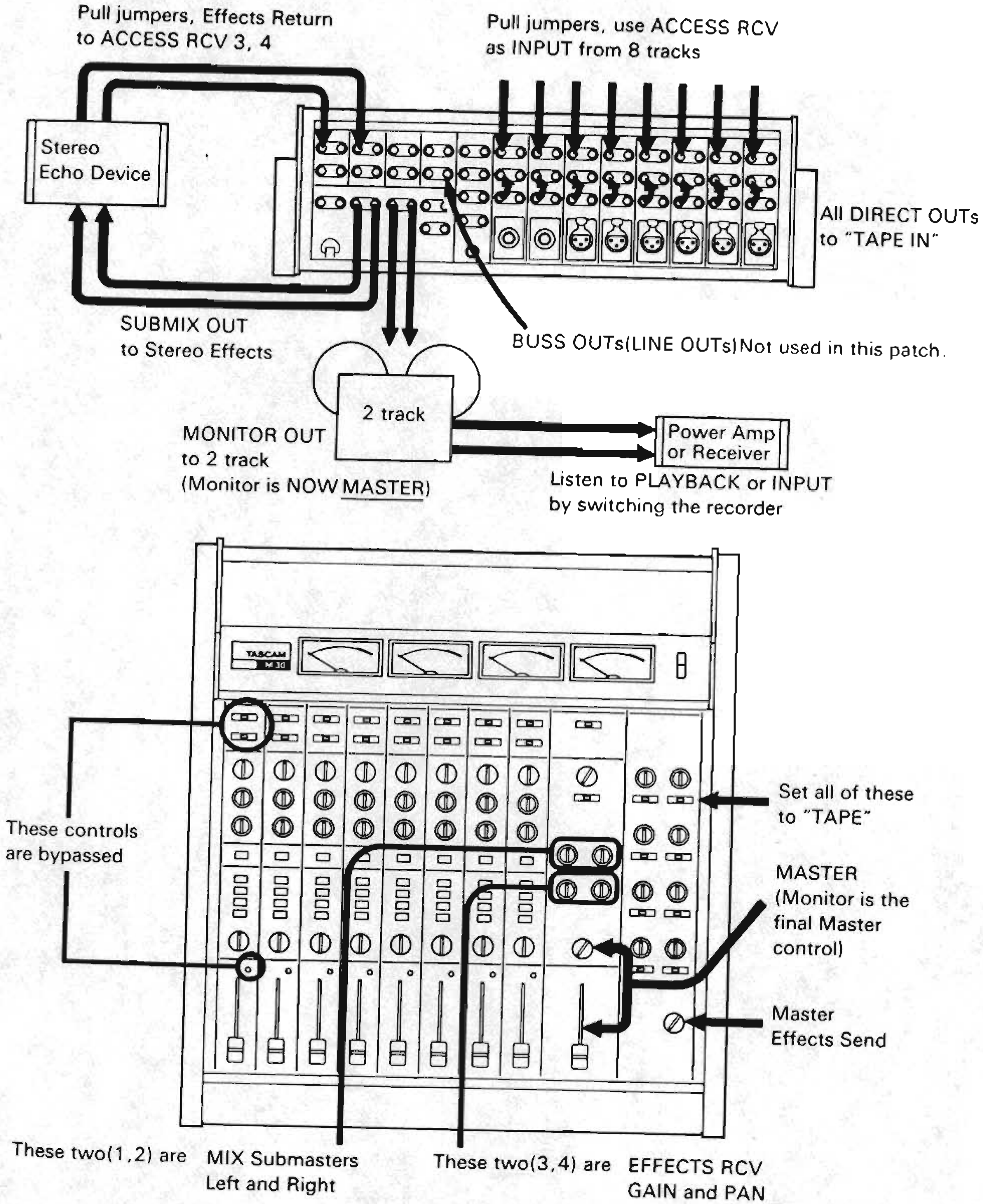
3. If you need an effects-send that contains the results of input fader adjustment and equalization, use the direct-out source shown in example #3.

Two units may be cascaded as in the previous example if necessary.

Recommended Locations for Model 1

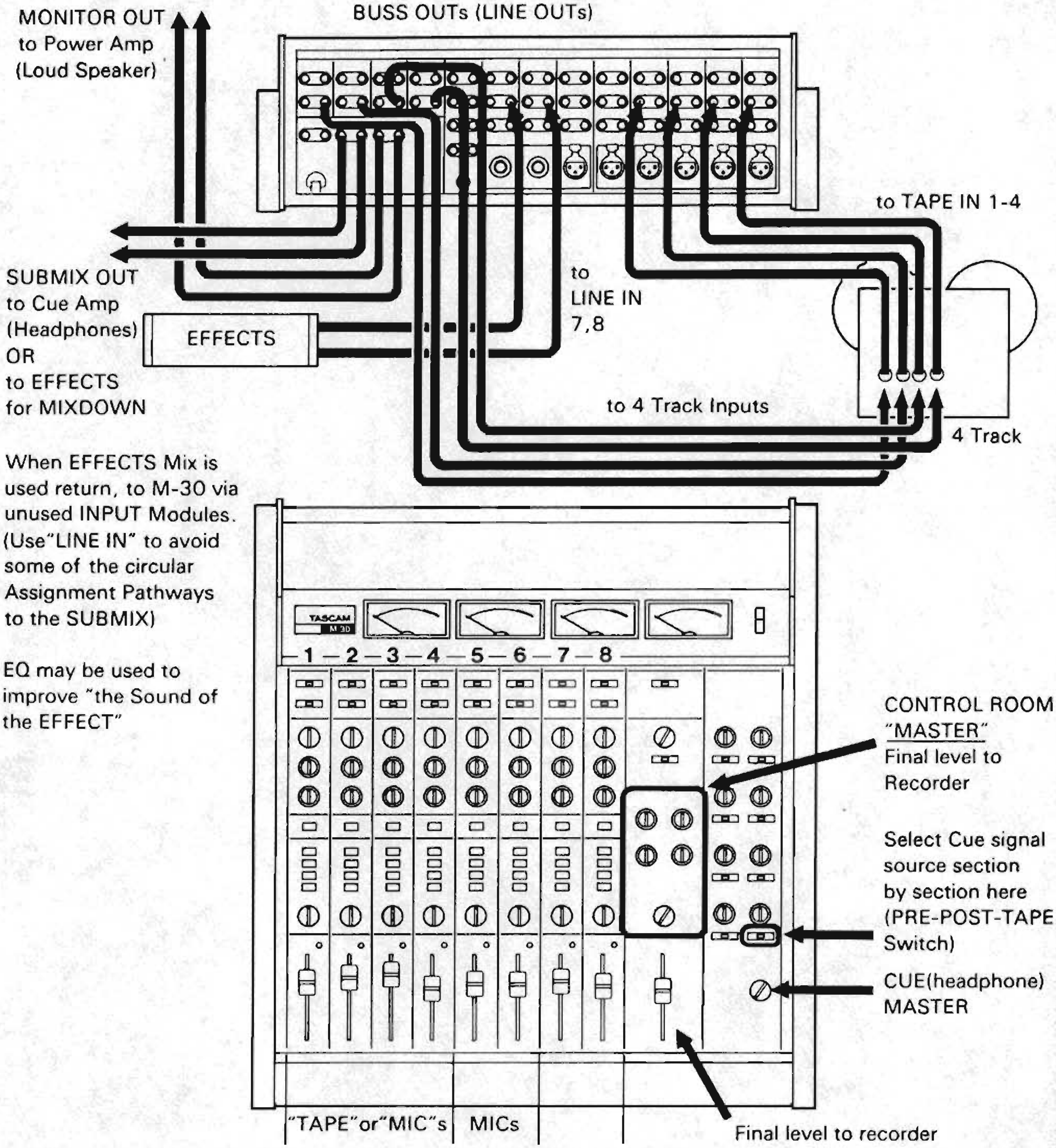


AN UNORTHODOX PATCH FOR REMIX WITH STEREO ECHO CAPABILITY



When unorthodox patches are used and the console top panel labels are no longer correct, we strongly recommend that you take the time to re-label each control to correspond to the new function that your re-patch has provided. Drafting tape applied to each group will prevent accidents from happening because you have tried to operate the M-30 "normally". It is also wise to label both ends of every cable. When re-patching away from and back to "normal", a label will save endless tracing and retracing of cables to find out where they start from.

Recommended 4-Track Set-Up



When EFFECTS Mix is used return, to M-30 via unused INPUT Modules. (Use"LINE IN" to avoid some of the circular Assignment Pathways to the SUBMIX)

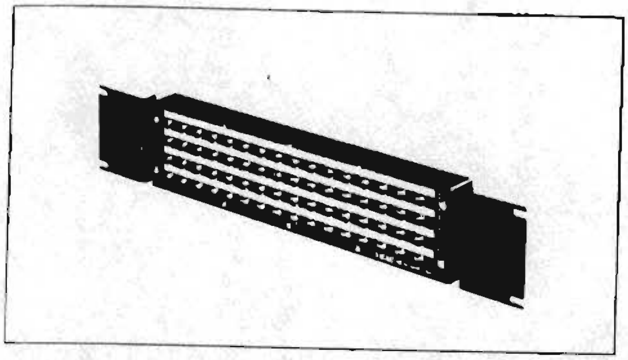
EQ may be used to improve "the Sound of the EFFECT"

....AND, in REMIX
7,8 become EFFECTS RCV with EQ
(select "LINE IN" on INPUT switch)

OTHER USEFUL ACCESSORIES

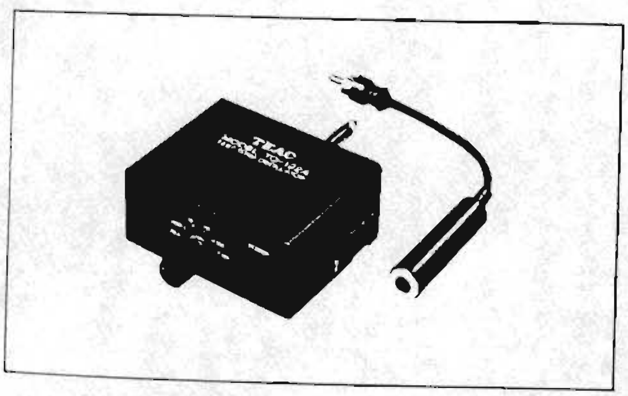
In addition to the Model 1 and the MB 20 Meter Bridge, we also offer these valuable additions to your system.

The PB-64 Patch Bay



When your system begins to expand beyond the basic, sorting out where things go can take much time away from the recording process. This accessory will allow you to speed things up and get back to what you really want to do. Sixty-four RCA pins on a panel. So you can bring all those jacks to where you are. It will get you off the floor and back to recording. Connect all your inputs and outputs to the back, and you can re-route your signals with short jumpers quickly.

The TO-122A



A signal generator that will allow you to set up and adjust many parts of your system. Originally designed to provide a reference signal for tape recorder alignment, it is perfect for tracing out patches and getting levels set prior to recording. It has very low distortion and 2 reference level settings.

With this six-frequency oscillator you can set reference levels, balance gain stages of components, adjust the bias and frequency response of your tape recorder and check overall system response plus the acoustic characteristics of your monitoring room. Powered by a 9-volt battery (not included) – this highly functional frequency generator is suitable for use with a mic input or, with the phono-to-phono plug adaptor, a line input. The output levels are −10 dB and −40 dB and there are six selectable frequencies:
40 Hz, 400 Hz, 1k Hz, 4k Hz, 10k Hz, and 15k Hz.

Lets assume that you have made a complete lay-out and have plugged up a full set of mics, lines and recorders. Now what? Well, how do you know that everything is actually working? You record something and play it back. An obvious statement, but what do you record if there are no musicians around to help out? We have talked about a tone generator to test out a set-up and you can record test tones, but there is another way to go that has some extra advantages. It will test the whole system from mics to speakers.

A portable radio can be placed in front of a mic and the signal will then be something that will actually pass through everything. Speech, music, whatever and you will be able to compare what is coming out of the radio to what you hear in your "studio", both before and after it is recorded. The radio can be moved from mic to mic and you will be able to judge how much signal from one mic location will be picked up in another (leakage). Will the guitar players mic pick up the sound of the piano? The radio test may help you find out before the musicians arrive.

The radio test is useful, but don't expect real musicians to be as easy to control. Remember, engineers have already "mixed" what comes out of a radio and have spent considerable time, money, and effort in tailoring the sound to acceptable levels. What you will get coming in your mics will be "Raw Sound" to start.
If you want to set levels on your recorder more carefully, and source of continuous sound will work better.

1. Try a weight on one key of an electronic organ so it plays continuously.
2. A vacuum cleaner makes a good sound source for this kind of test, or any other motor-operated device that won't be damaged by continuous operation.

Try to budget your system to include at least one really good microphone. In multichannel recording, one mic can be used over and over and will have a large effect on your total sound. Good mics are the key to really good sound.

Professional Low Loss Cable:

There are vast differences in cable design and performance, and those differences can make or break an otherwise excellent sound system. When you're investing in the kind of high quality audio equipment represented by the TASCAM Studio Series, it makes sense to use TASCAM professional audio cables. Anyone who's switched to them will tell you they're worth every cent.

LOW CAPACITANCE

Our cables feature very low capacitance under 15 picofarads per foot, so they don't act as high-frequency roll-off filters as do typical cables of 100 or 300 pF/foot. In addition, our cables use an ultra-high density bare-copper braided shield (99 % coverage), so electrostatic noise (buzz or hum) and RFI (CB or broadcast signals) are kept out of your program.

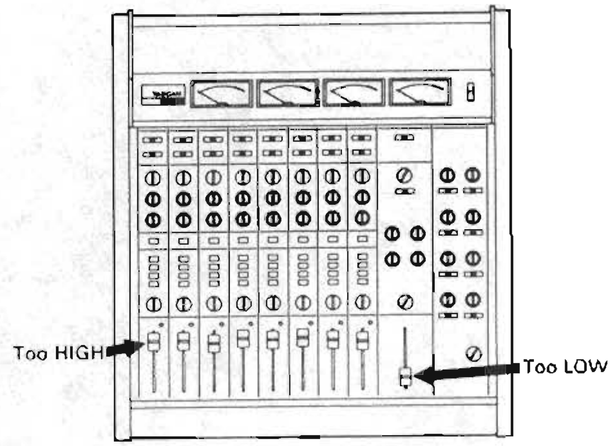
Low capacitance is important, and so is consistent capacitance; that is, you want the electrical coupling of center conductor-to-shield to remain the same throughout the cable, even if it is sharply bent, crushed, flexed, or tugged. Should the local cable capacitance change, noise and/or signal losses often result. We utilize the unique dielectric known as Datalene. This special insulation keeps the stranded signal conductor perfectly centered within the shield. Datalene is about as flexible as foam core dielectrics but far more resistant to extreme heat or cold, and it has a "memory", so it retains its shape after flexing. Datalene also acts as a mechanical shock absorber, guarding against external impacts which, in other cables, might sever the center conductors and cause intermittent contact.

When we join the connector to the cable, we insert the cable's stranded center conductor all the way into the pin and then fill the pin with solder. The braid is wrapped and soldered a full 120° around the shell, not tacked at one spot, so you get maximum shielding and strength.

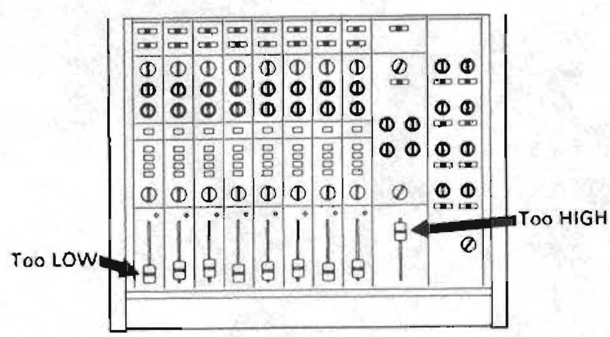
A Final Word of Mixdown Advice:

All finished tapes must be balanced – one sound and its tone judged by blending with others. Don't depend on EQ in "solo" 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 at 5K on every pot will also boost the noise in your mix as well as the signal.

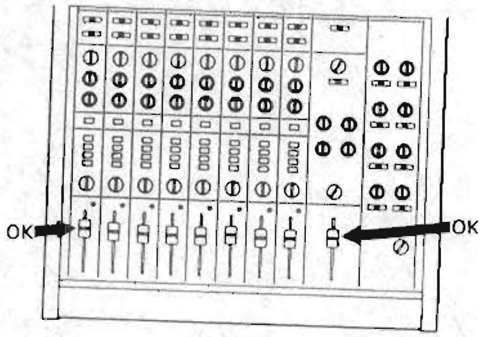
1. If your console faders always wind up like this, you are likely to be over-loading your summing amps. Pull down the inputs and raise the master.



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



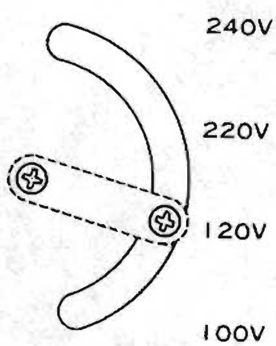
3. This picture is a reasonable compromise, and is probably better all around.



VOLTAGE CONVERSION

This mixer is adjusted to operate on the electric voltage specified on the packing carton.

Note: This voltage conversion is not possible on model sold in the U.S.A. and Canada, U.K., Australia or Europe.
For general export units, if it is necessary to change the voltage requirements of this mixer to match your area, use the following procedures.
Always disconnect Power Line Cord before making these changes.



1. Remove the bottom panel of the mixer by removing the screws.
2. Locate the voltage selector shown in the illustration.
3. Loosen the two screws in the shorting bar and move the bar so that it shorts across the terminals marked with the required voltage (100, 120, 220 or 240).
4. Retighten the screws.
5. Replace the bottom cover.

Note for U.K. Customers

U.K. customers only: Due to the variety of plugs being used in the U.K., this unit is sold without an AC plug. Please request your dealer to install the correct plug to match the mains power outlet where your unit will be used as per these instructions.

IMPORTANT

The wires in this mains lead are coloured in accordance with the following code:

BLUE:	NEUTRAL
BROWN:	LIVE

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows.

The wire which is coloured BLUE must be connected to the terminal which is marked with the letter N or coloured BLACK. The wire which is coloured BROWN must be connected to the terminal which is marked with the letter L or coloured RED.

SPECIFICATIONS

1. 8-Input/4-Line Output/2-Monitor Output/ 2-Submix Output:

2. Input Selector:

1-6 channel
MIC(Low Impedance)/LINE/REMIX

7, 8 channel
MIC(High Impedance)/LINE/REMIX

3. Mic Input:

(Low Impedance) — channel 1-6:

Mic Impedance	200 to 600 ohms nominal mics (matched for mics of 600 ohms or less)
Input Impedance	600 ohms, balanced, XLR type
Nominal Input Level	-60dBV (1mV)
Maximum Input Level	+10dBV (3V) ATT to 40dB

(High Impedance) — channel 7-8:

Mic Impednace	10k ohms nominal mics
Input Impedance	100k ohms
Nominal Input Level	-60dBV (1mV)
Maximum Input Level	+10dBV (3V) ATT to 40dB

4. Line Input:

Input Impedance	20k ohms
Nominal Input Level	-10dBV (0.3V)
Maximum Input Level	+14dBV (5V)

5. Tape Input:

Input Impedance	50k ohms
Nominal Input Level	-10dBV (0.3V)
Maximum Input Level	+14dBV (5V)

6. Line Output/Aux Output:

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10dBV (0.3V)
Maximum Output Level	+14dBV (5V)

7. Monitor Output:

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-2.2dBV (0.775V)
Maximum Output Level	+14dBV (5V)

8. Submix Output:

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10dBV (0.3V)
Maximum Output Level	+14dBV (5V)

9. Cue Output:

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10dBV(0.3V)

10. Direct Output:

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10dBV(0.3V)

11. Access Send Output (Input/Master Section):

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10dBV(0.3V)

12. Access Receive Input (Input/Master Section):

Input Impedance	200k ohms
Nominal Input Level	-10dBV(0.3V)
Minimum Input Level	-18dBV(0.126V)

13. Submix Input —Channel L,R(and PRE, POST, Tape 1-8):

Input Impedance	10k ohms
Nominal Input Level	-10dBV(0.3V)
Maximum Input Level	+14dBV(5V)

14. Buss Input:

Input Impedance	10k ohms
Nominal Input Level	-10dBV(0.3V)
Maximum Input Level	+14dBV(5V)

15. Headphones Output:

Load Impedance	8 ohms
Maximum Output Power	Greater than 100mW Output VR at max.

16. Phono Input:

Input Impedance	45k ohms
Nominal Input Level	-54dBV(2mV) at 1kHz
Minimum Input Level	-60dBV(1mV) at 1kHz
Maximum Input Level	-30dBV(31.6mV) at 1kHz

17. Phono Output:

Minimum Load Impedance	5k ohms
Nominal Load Impedance	10k ohms
Nominal Output Level	-10dBV(0.3V) at 1kHz

18. Frequency Responce:

Line Output	30 to 20,000Hz, ±2dB
Monitor Output	30 to 20,000Hz, ±2dB
Submix Output	30 to 20,000Hz, ±2dB

19. Equalizer:

Type	Parametric Frequency and Shelving
Level	±15dB
Frequency (low)	60 to 1,500Hz
(high)	1,000 to 10,000Hz
(shelving)	12,500Hz

20. Signal to Noise Ratio (A weighted/unweighted 20 to 20,000Hz):

Equivalent Mic(Low Impedance)	116dB/114dB
Mic(Low Impedance) 1 channel	Better than 66/64dB
6 channels	Better than 57/55dB
Mic(High Impedance) 1 channel	Better than 58/57dB
2 channels	Better than 55/53dB
Mic(Low and High Impedance) 8 channels	Better than 53/51dB
Phono Input to Phono Output	Better than 57dB UNWTD (20 to 20,000Hz)

21. Cross Talk:
Better than 60dB (1kHz, Nominal Input Level)

22. Total Harmonic Distortion:
Less than 0.1% at 1kHz, Nominal Input Level

23. Fader Attenuation:
60dB or more

24. Overload Indicator Level:
25dB above Nominal Input Level

25. Peak Indicator Level:
10dB above Nominal Output Level
— Adjustable

26. Dimensions (WxHxD):
465x160x520mm
(18 1/4"x6 5/16"x 20 1/2")

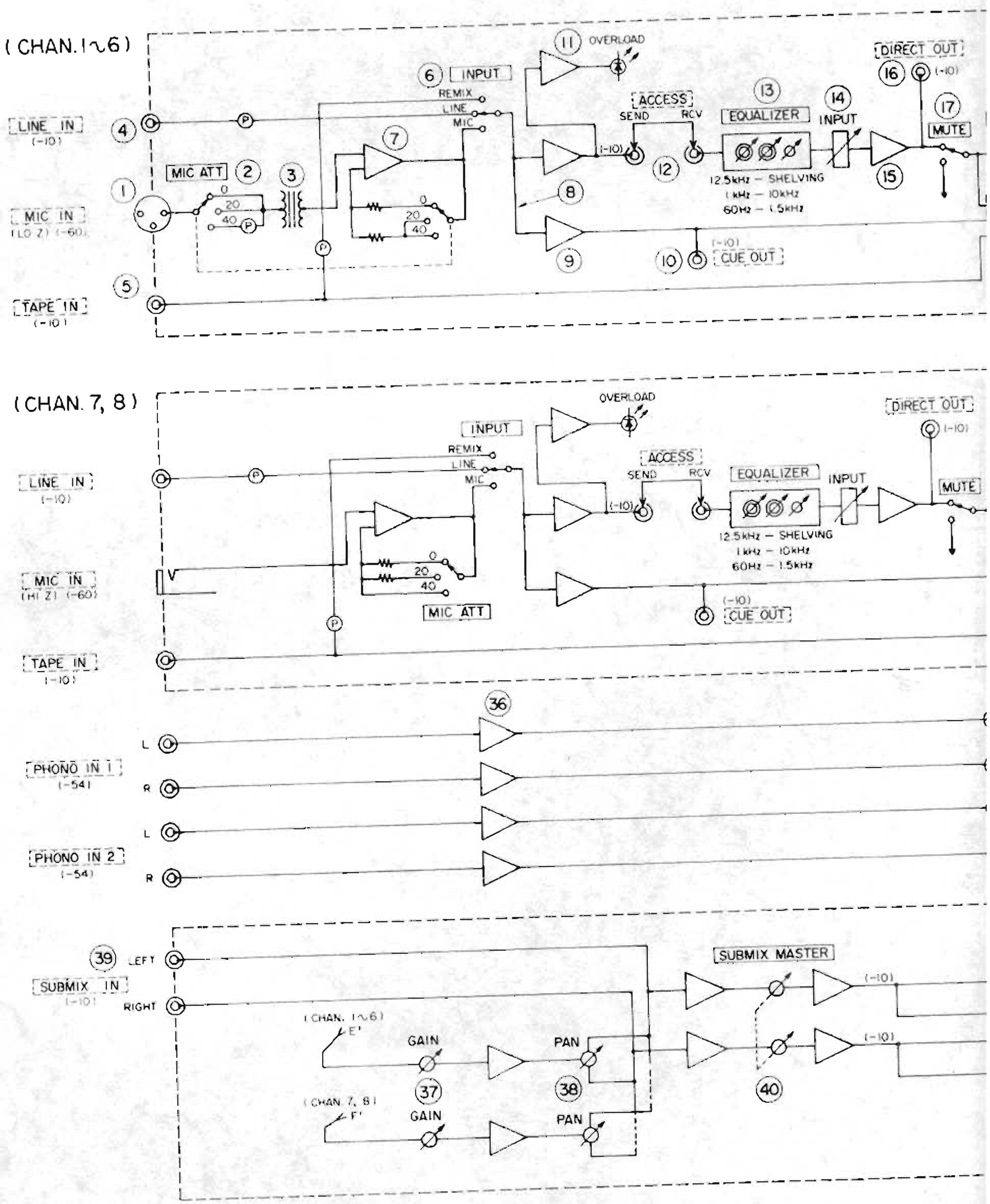
27. Weight:
16kg (35 3/8 lbs)

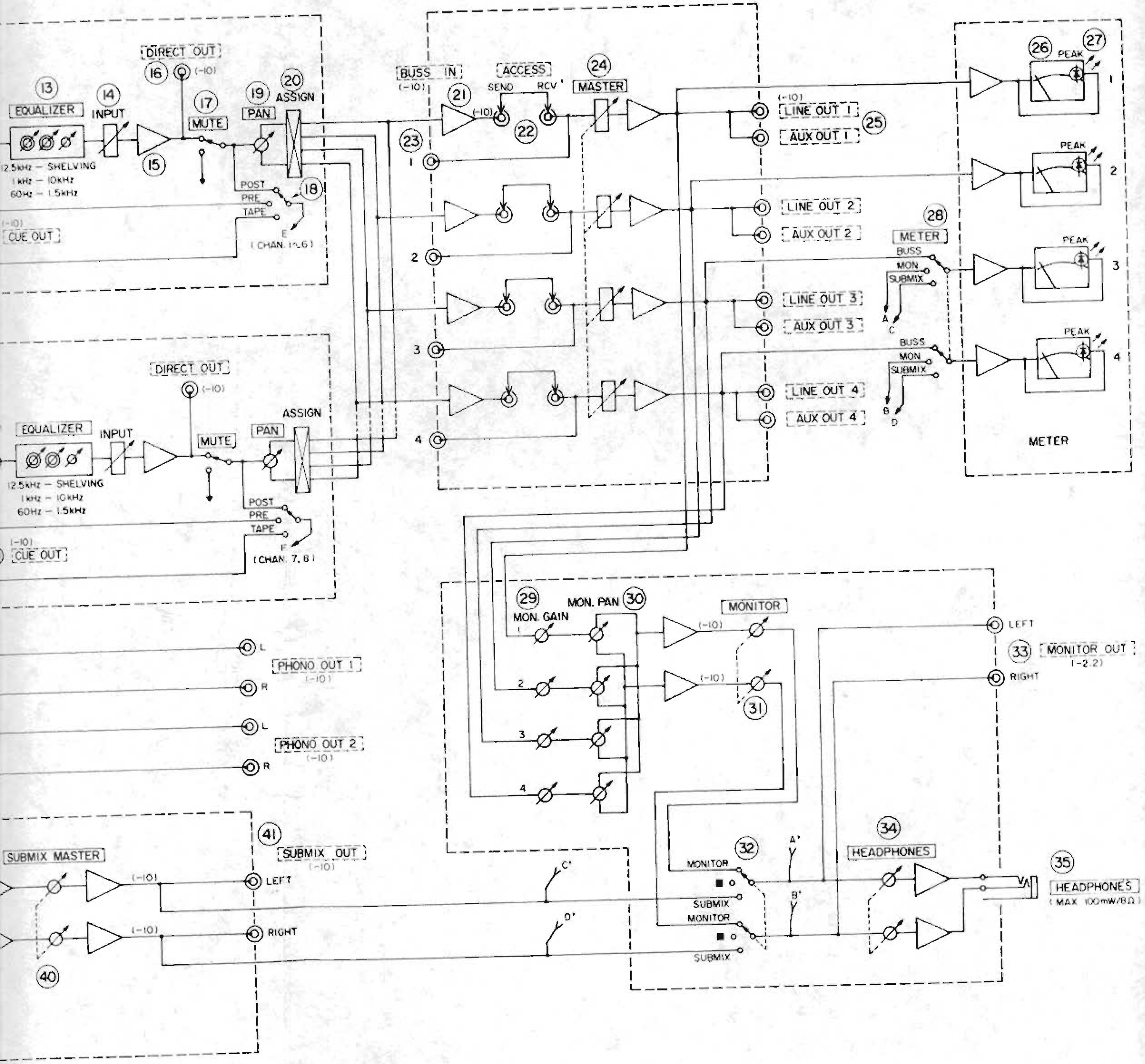
28. Power Requirement:

100/120/220/240V AC, 50/60Hz, 26W	(General Export Model)
120V AC, 60Hz, 26W	(U.S.A./Canada Model)
220V AC, 50Hz, 26W	(Europe Model)
240V AC, 50Hz, 26W	(U.K./Australia Model)

In these specifications, 0 dBV is referenced to 1.0 volt. Actual voltage levels also are given in parenthesis. To calculate the 0 dB/0.775Volt reference level (i.e., 0 dBm in a 600 ohms circuit) and 2.2 dB to the listed dB value; i.e., -10 dB re:1V/-7.8 dB re:0.775V. Changes in specifications and features may be made without notice or obligation.

BLOCK DIAGRAM

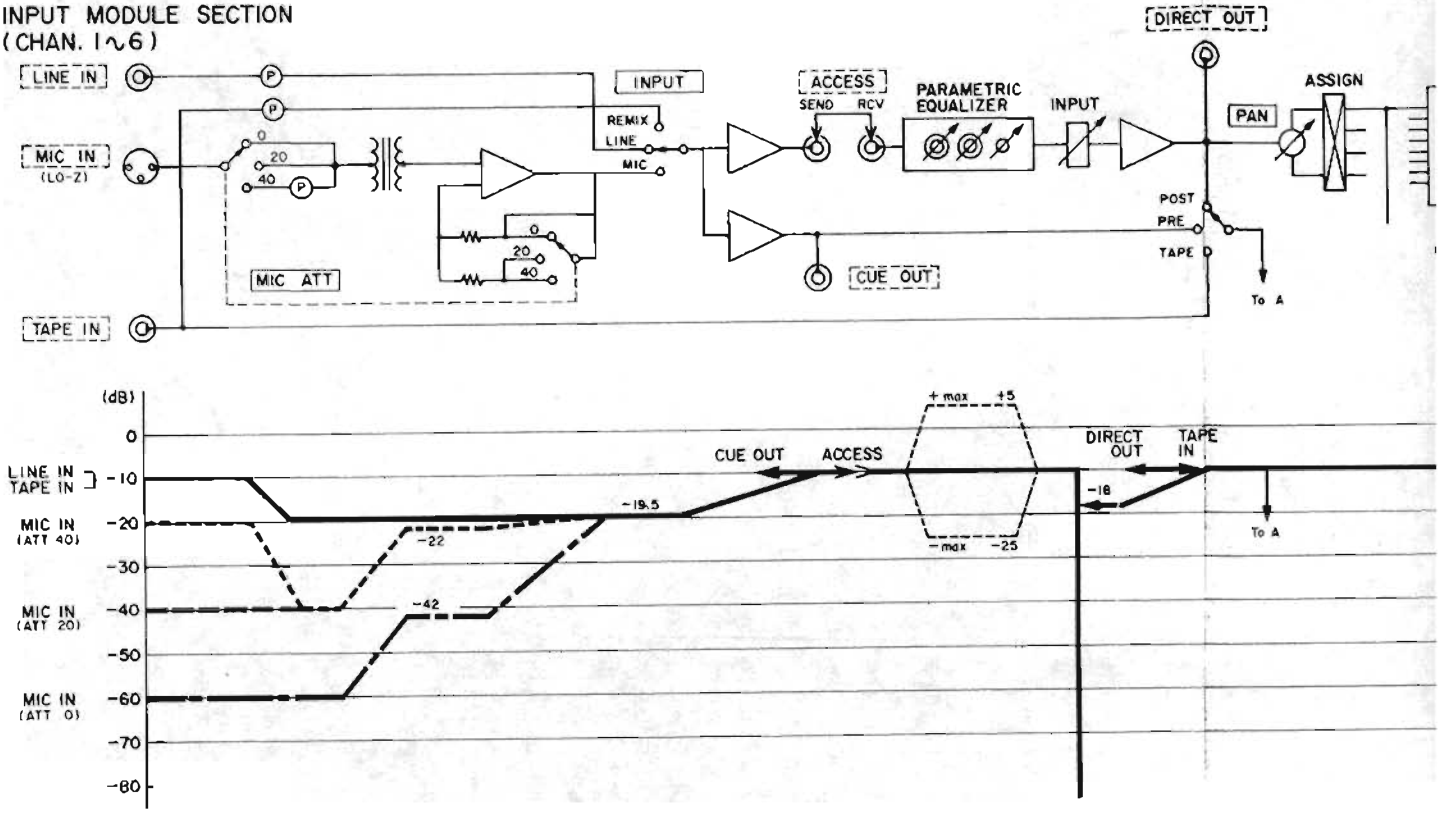




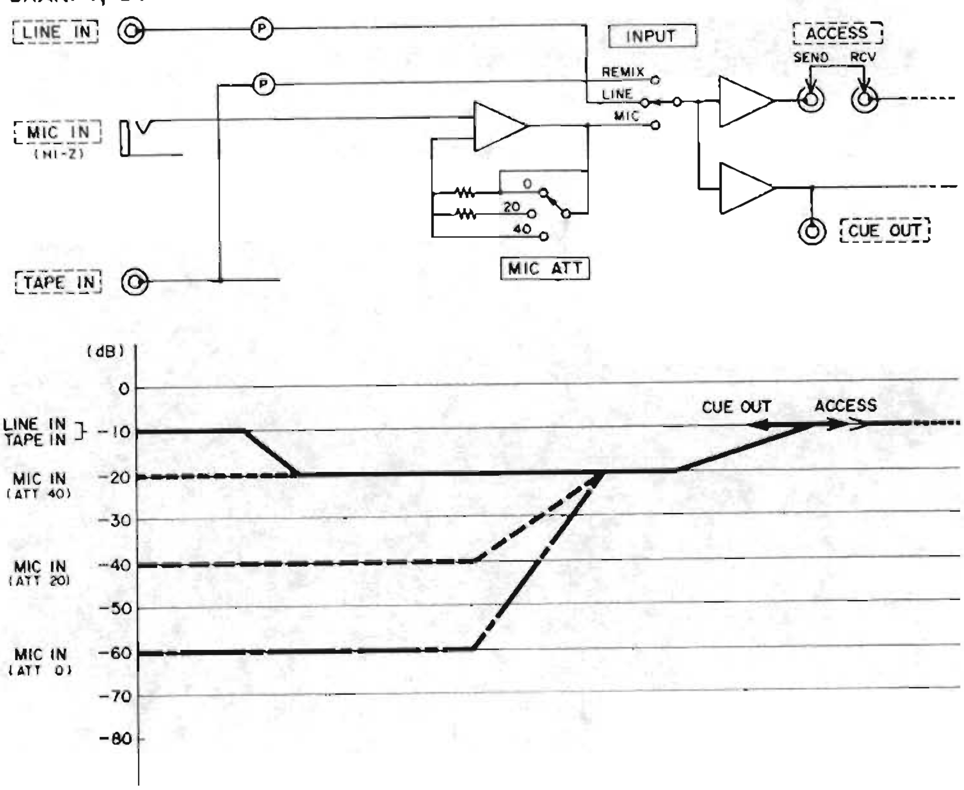
DIRECT OUT
(-10)
EQUALIZER
INPUT
MUTE
PAN
ASSIGN
12.5kHz — SHELVING
1kHz — 10kHz
60Hz — 1.5kHz
POST
PRE
TAPE
(-10)
CUE OUT
(CHAN. 1~6)
(CHAN. 7, 8)
BUSS IN
(-10)
ACCESS
SEND
RCV
MASTER
LINE OUT 1
AUX OUT 1
LINE OUT 2
AUX OUT 2
LINE OUT 3
AUX OUT 3
LINE OUT 4
AUX OUT 4
PEAK
METER
BUSS
MON
SUBMIX
MON. GAIN
MON. PAN
MONITOR
LEFT
RIGHT
MONITOR OUT
(-2.2)
PHONO OUT 1
PHONO OUT 2
SUBMIX MASTER
SUBMIX OUT
HEADPHONES
(MAX 100mW/8Ω)

LEVEL DIAGRAM

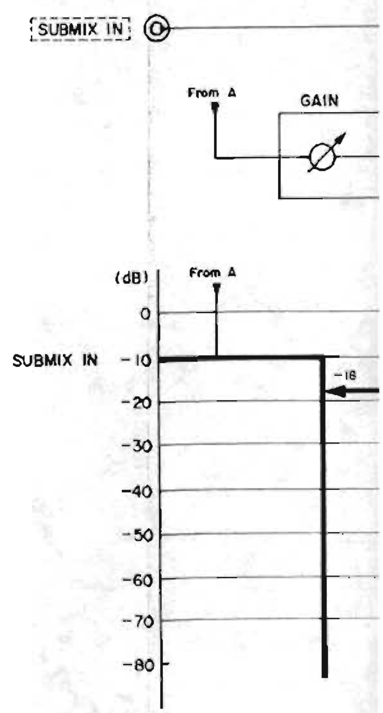
INPUT MODULE SECTION
(CHAN. 1∿6)

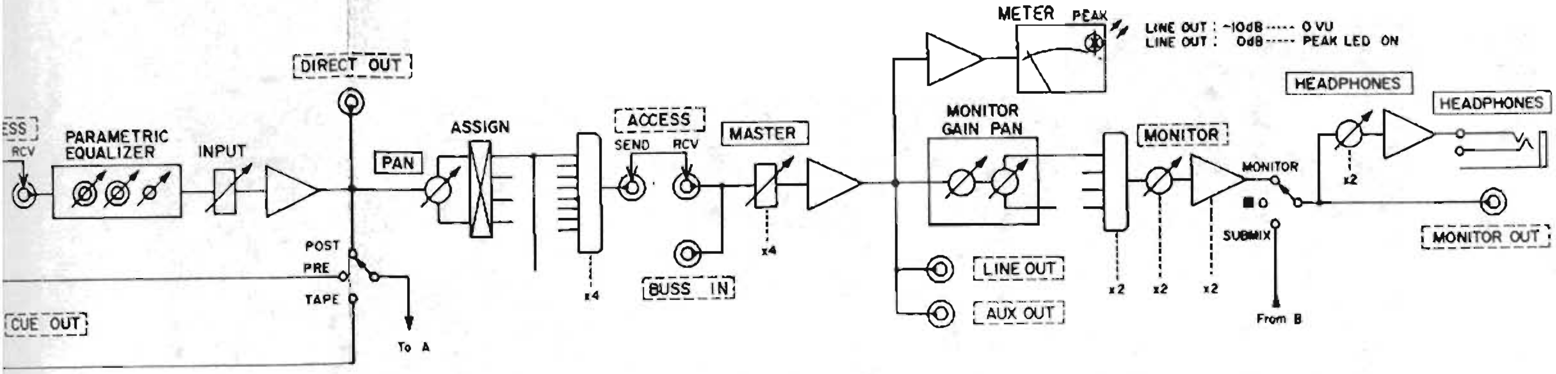


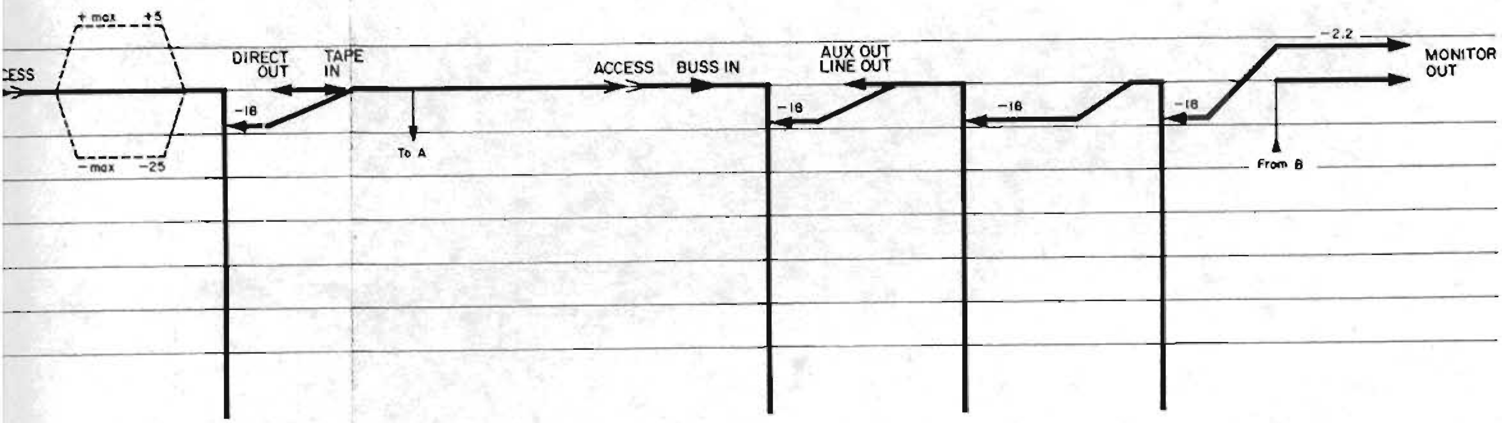
INPUT MODULE SECTION
(CHAN. 7, 8)



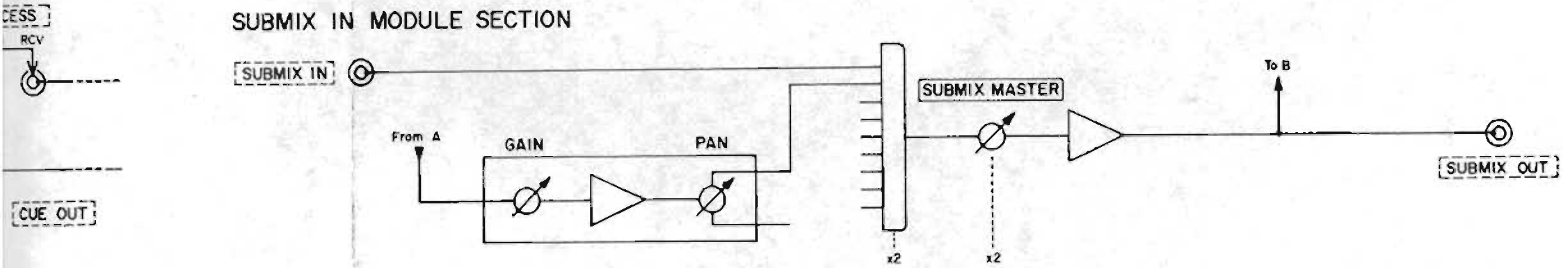
SUBMIX IN MODULE SECTION

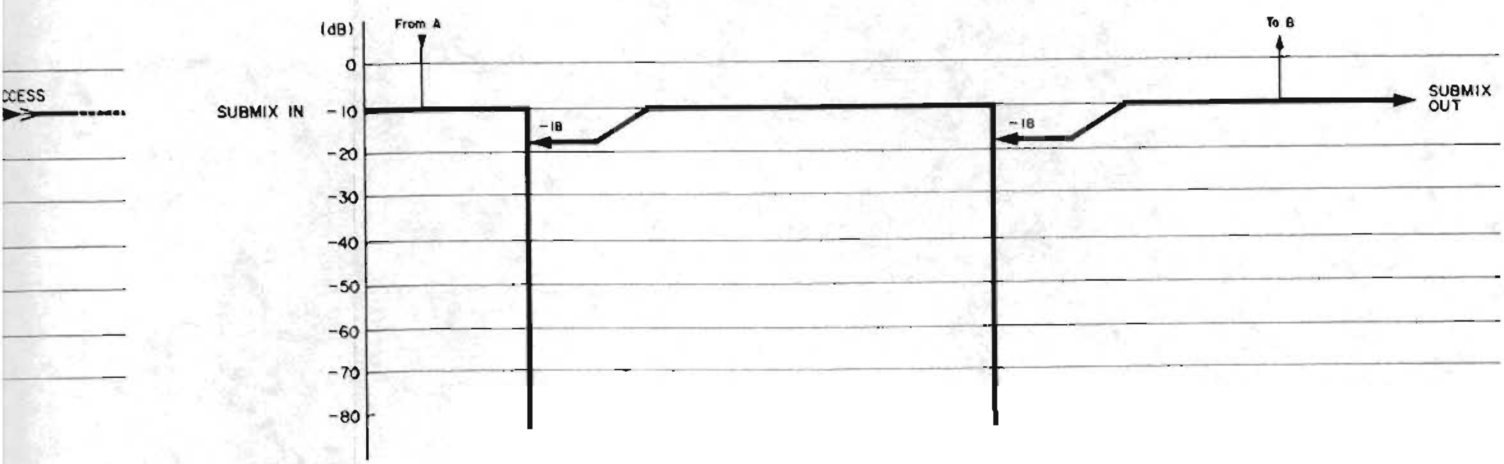






SUBMIX IN MODULE SECTION





MORE INFORMATION IS AVAILABLE

We've tried to give you representative examples of some of the things you can do to get started, and you'll discover many more — some by way of happy coincidence, others after long hours of concentration. If you're just getting into recording and want to expand your knowledge, more information is available.

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1954
More concerned with exact formulae, but still very readable. It is not necessary to do calculations to gain knowledge from this textbook.

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MUSIC ACOUSTICS AND ARCHITECTURE
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New York, N.Y.
1962
A technical survey on concert halls with much documentation. Worth reading. This author has many useful stories to tell about the interface of science and art.

Clifford, Martin
MICROPHONES: HOW THEY WORK AND HOW TO USE THEM
Tab Books
Blue Ridge Summit, Pa.
1977
An excellent low-cost book for the beginner on microphone types, history and construction. The explanations given assume no prior knowledge and are very complete. Recommended.

Everest F. Alton
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Tab Books
Blue Ridge Summit, Pa.
1978
Low-cost basic book. This book on studio acoustics is the easiest to read and understand of all the textbooks on the subject, and comes closest to dealing with the actual problems encountered in the home studio.

Everest F. Alton
HANDBOOK OF MULTICHANNEL RECORDING
Tab Books
Blue Ridge Summit, Pa.
1976
A survey volume containing good information on all topics. Very clearly written and recommended for a beginner.

Nisbett, Alec
THE TECHNICS OF THE SOUND STUDIO FOR RADIO, TELEVISION AND FILM
Hastings House Publishers, Inc.
New York, N.Y.
1976
Although not specifically written for the tape recordist, this 500-page book is well worth its cost. Very useful practical advice if you are working with speech (drama, commercial announcing, etc.).

Nisbett, Alec
THE USE OF MICROPHONES
Hastings House Publishers, Inc.
New York, N.Y.
1976
The author's point of view is basically radio, but his ability to communicate difficult concepts is very good. Well illustrated.

Olsen, Harry F.
ACOUSTICAL ENGINEERING
D. Van Nostrand Company
New York, N.Y.
1957

and

Olsen, Harry F.
MUSICAL ENGINEERING
D. Van Nostrand Company
New York, N.Y.
1959
Anything you can find by this writer is worthwhile, and the latter book in particular will give scientific answers to musical questions (what frequency is the note dB above middle C?) and can be used to translate one "language" into another. Extremely valuable.

Rettinger, Michael
ACOUSTIC DESIGN AND NOISE CONTROL, VOL. 1
Chemical Publishing Company
New York, N.Y.
1977
Although this book is highly technical, the writing is very lucid and many examples are given to go along with the math. This writer is not afraid to draw conclusions and give his reasons for doing so in simple language.

Runstein, Robert E.
MODERN RECORDING TECHNIQUES
Howard W. Sams and Co.
Indianapolis, Indiana
1974
The first low-cost book on studio practice. The equipment dealt with is somewhat outdated, but the theory is still the same. Excellent basic survey.

Tremaine, Howard M.
THE AUDIO CYCLOPEDIA
Howard W. Sams and Co.
Indianapolis, Indiana
1976
This 1,700-page reference work is sure to contain the answer to almost any technical question you can think of. The writing assumes much prior knowledge and this book should be used with others that are more basic in their writing style if you are new to the field of scientific audio.

SOME MAGAZINES OF INTEREST:

"db" — THE SOUND ENGINEERING MAGAZINE
1120 Old Country Road
Plainview, N.Y. 11803

"MODERN RECORDING"
14 Vanderventer Avenue
Port Washington, N.Y. 11050

"RE/P" — RECORDING ENGINEER/PRODUCER
1850 Whitley Street, Suite 220
Hollywood, Ca. 90028

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