TASCAM

TEAC Production Products

M-35

Audio Mixer

OPERATION/MAINTENANCE

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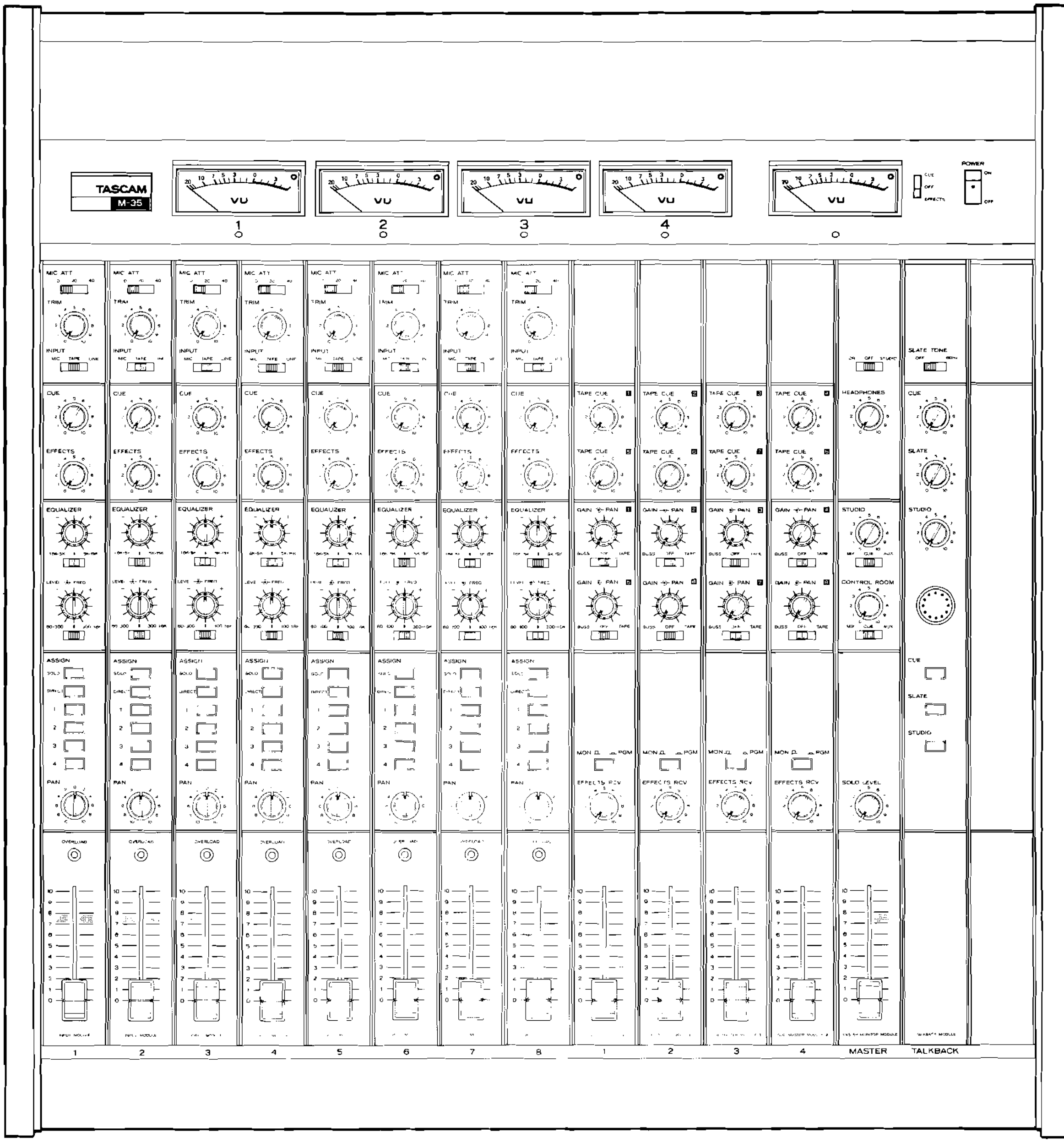
TASCAM
M-35
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POWER
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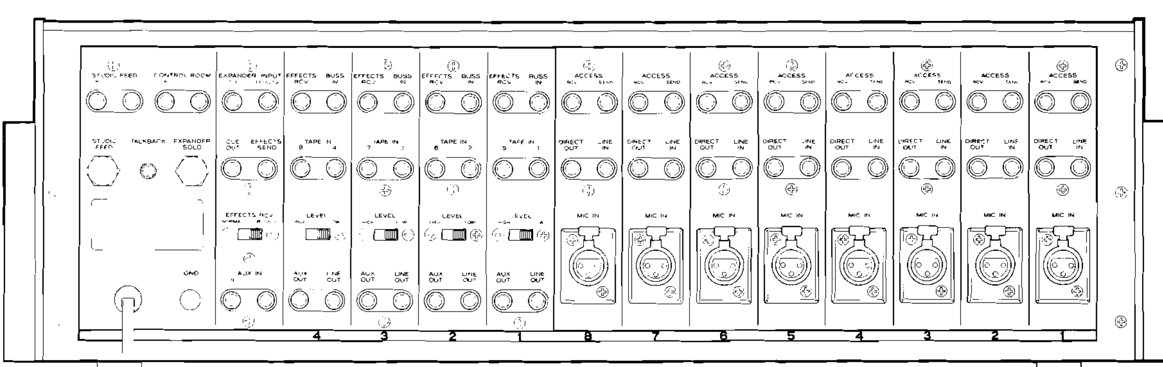
STUDIO FEED
CONTROL ROOM
EXPANDER INPUT
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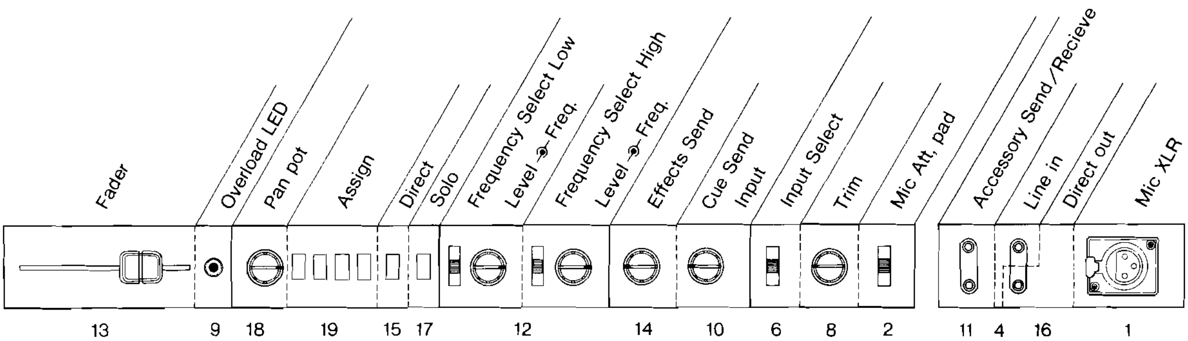
BUSS IN
ACCESS
TALKBACK
EXPANDER SOLO
CUE OUT
EFFECTS SEND
TAPE IN
DIRECT OUT
LINE IN
LEVEL
MIC IN
GND
AUX IN
AUX OUT
LINE OUT

INTRODUCTION TO THE MODEL 35

The Model 35 is an audio mixing console designed to satisfy the requirements of modern multichannel recording. Many of the auxiliary mixing systems needed are built-in and can be re-routed to do more than one job. Fast, convenient and complete operation with 4- or 8-track recorders can usually be accomplished without re-patching. However, the process of multichannel recording is constantly changing, growing more complex as an art with each advance in technology. Your signal processing needs may require a unique arrangement of subsystems. No console has ever been built so large and complete in its routing that it could solve every imaginable problem with one button. Someone will always be able to come up with that unusual situation requiring "just one more mix". In order to cope with these unpredictable requirements, patch points are provided throughout all signal pathways on the M-35.

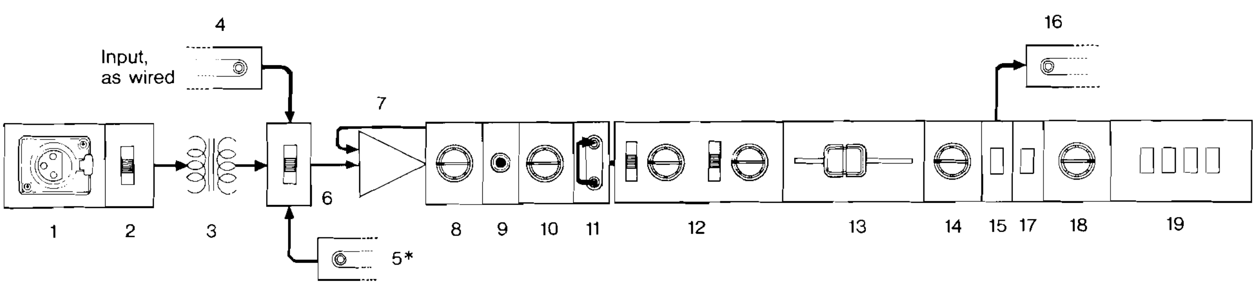
As our mixing console becomes more flexible, the amount of time needed to understand the available function increases as well. The main signal path from mic in to line out is still fairly straightforward. The requirements have not changed much since the days of "mono", but the routing for effects sends, cue feeds, and stereo monitoring can be hard to visualize. The beginner often overlooks the significance of connections that would be 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-35 thoroughly.



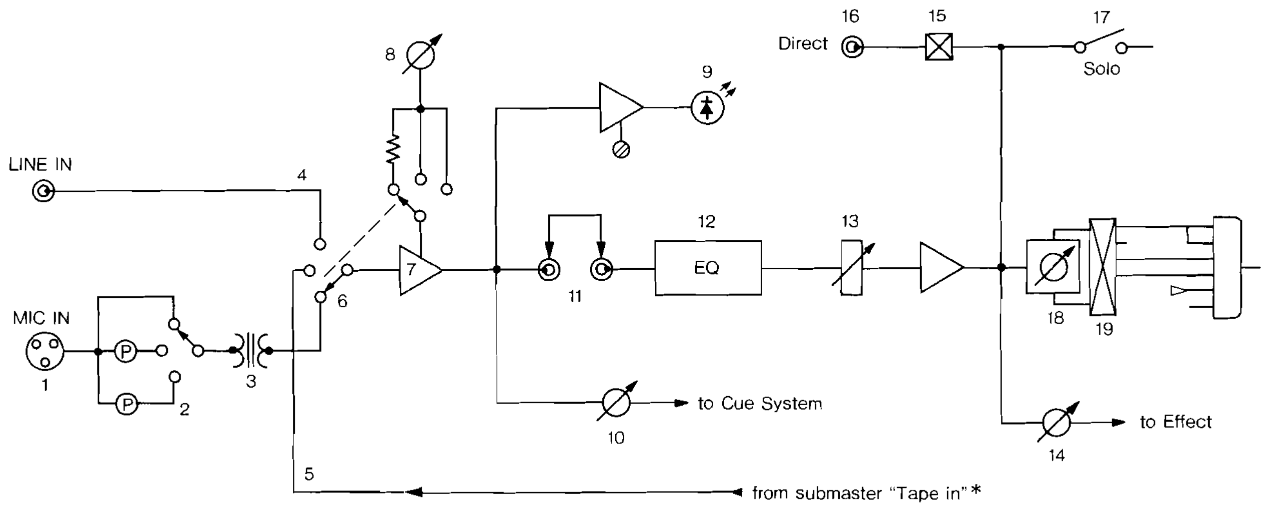
Input module layout including back panel

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 is the information you need to understand to use the M-35 successfully.

As an example, if the controls on an input module actually followed the order in which they are wired, the module top panel would look like this. We'll put the jacks on the top, as well as the switches and faders.



While this arrangement of controls might help the beginner 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-35 can be used successfully. So along with the documentation you will need for service (schematic diagrams, mother-board layouts and mechanical disassembly inforamtion), we include a simplified electrical sequence chart called a block diagram.



This drawing shows all the controls, switches, amplifying stages and connectors in their proper order. Learning to read it will provide the answers to any questions about what comes where on the inside. 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. In the following sections of this manual, we will do our best to describe the individual functions and controls of the M-35, and how they can be arranged in more than one sequence; but, your mixing needs may be best served by an arrangement of inputs and sub-system connections you work out for yourself.

To begin, we'll start with some basic information about sound and the numbering systems used to describe levels in and out of the equipment and impedance—what the term means and how to deal with the details when you must connect from our gear to other equipment. Many aspects will be discussed in the most basic language we can use. There is a vast amount of information available to the beginning sound mixer but much of it is not basic enough to be easily understood, or it assumes that the reader has an engineering or scientific background and will be interested in "the math". Practical "rules of thumb" for the novice are not generally available. Something between a picture of the outside of the unit and a complete mathematical analysis of the circuits inside is needed. You don't have to build a mixer from scratch, you just need to know how to operate one.

However, some numbers are unavoidable. The M-35 mixer does nothing useful without being connected to quite a lot of sophisticated gear. 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 requirements and problems. We have tried to make this manual as simple as technology will allow. Each section or topic will give you some basic instruction in the terminology used in the process of mixing as well as a list of what plug goes into which jack.
Whenever possible, the scientific terms have been related to understandable common references. Understanding what is going on inside your equipment will help you improve your sound. Think of this manual as a reference book. You won't need all of what is here to begin, and it certainly is not necessary to memorize it, but do try to find time to read it carefully at least once. That way you will be familiar with its contents. If you need the numbers, they will be there waiting. Good luck with your sound.

THE DB; WHO, WHAT, 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.

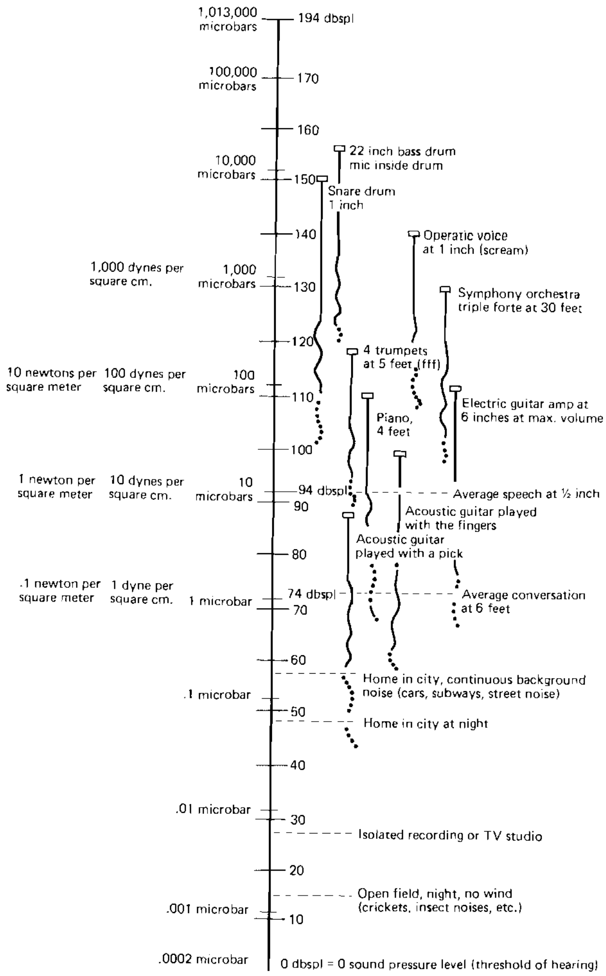
From this basic research, Bell Labs developed a system of units that could be applied to all phases of the system. Sound traveling on wires as electrical energy, sound on tape as magnetic energy, sound in air; anyplace that sound is, or has been stored as energy until some future time when it will again 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 3dB more sound. If you double an electrical voltage, is it twice as loud? No, it is only 6dB more sound. The unit quantities must follow nonlinear 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 4000Hz. Bell Labs measured this value to be .0002 microbar, so we say 0dB = .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 "0dB".



We should also make a point of mentioning that the maximum number on this chart represents "peak power" and not average power. The reason? Consider if even some monetary 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 word "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 than X + 20dB is usually safe for speech, but extremely percussive sounds may require as much as 90dB 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.

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 multichannel 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.

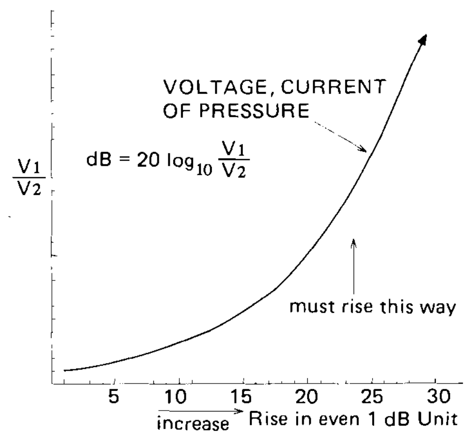
Here are the formulae for unit increment, but they are necessary only for designers. And unless you build your own gear, you won't have to deal with them. For power (watts) increase or loss, calculate by the following equation:

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

For voltage, current or pressure calculations:

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

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 170dB 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 0dBm, and this is still the standard of related industries. Thus, a 0dBm signal at a 600ohm 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 ohms 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.

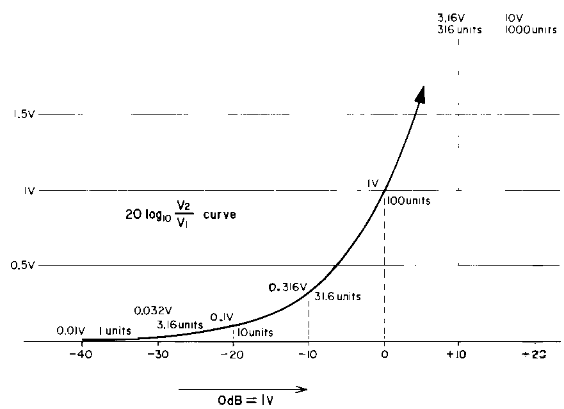
Since TASCAM M-35 mixer is designed to route a signal from a mic to a recorder, we think that the 2-wire system is a wise choice. The internationally accepted standard (IEC) for electronics of this kind uses a voltage reference without specifying the exact load it is expected to drive. The reference is this:

$$0 = 1 \text{ volt}$$

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 the 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, such as:

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 card 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 as a 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.

The classic measurement for output impedance is to load a circuit until the voltage drops 6dB (to half the original power) and note what the load value is. In theory, you now have a load impedance that is the same as the output impedance. If you reduce the load graudally, 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?

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

Most technicians say "1dB, not bad, that's acceptable". We at TEAC must say 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 drop at 20 Hz?

2. All outputs are not measured at the same time. Most people don't have twenty 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 the widely misunderstood rule of thumb – the seven-to-one ratio – we will give you the values for outputs in a complete form.

Even though the true output impedance may be low, say 100 ohms, for the practical reasons explained previously, we feel that the 7:1 ratio is not sufficient. To use this rule of thumb, you must use a higher value. We'll call this value the "output load impedance". For example, in our model M-35:

ACCESS SEND	1.4k ohms × 7 = 10k ohms
LINE OUT	1.4k ohms × 7 = 10k ohms

This is a number that will give good results with the 7:1 method. To go one step further, here are the actual minimum ohmic values we feel are wise. Connect to TOTAL INPUT IMPEDANCE LOAD higher than:

ACCESS SEND	10k ohms
LINE OUT	10k ohms etc.

Our specifications usually show 10,000 ohms as a "Nominal Load Impedance" and you can see that we arrived at the first column above by dividing 10,000 by 7. Any number higher than 10,000 is less load.

Input impedance is more straight forward and requires only one number. Load is load, and here are the values for the M-35:

MIC IN	600 ohms
LINE IN	50k ohms
ACCESS RECEIVE	220k ohms
BUSS IN	12k ohms

If one output is to be "Y" connected to two inputs, the total impedance of the two inputs must not exceed the load impedance, mentioned above, and if it becomes necessary to increase the number of inputs with slight exceeding 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 still 7 times the output impedance, you can connect both at the same time. Remember, we are not using the true output impedance, we are using the adjusted number in group 1, output load impedance.

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 (higher than 7 : 1 ratio) by this method, you can connect without worry.

If you must have exact values, here are the formulae: For more than 2 :

$$RX = \frac{1}{\frac{1}{R1} + \frac{1}{R2} + \frac{1}{R3} + \cdots + \frac{1}{Rn}}$$

RX = Value of Total Load

For 2 loads or inputs:

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

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 of 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 assume that the manufacturer has already done the "seven times is best" ratio calculation. 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 impedance and then will give the recommended lowest 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-35, we must use a steady signal that will not jump around. We use a tone of 1000 cycles and start it out at a level of -60dB 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-35 — the line-out and the reference signal there will be -10dB, our "line level" reference.

From this you can see that if your sound is louder than 94dB spl, or your mic will produce more electricity from a sound of 94dB spl than -60dB, 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 6 you will see that most musical instruments are louder on the average than 94dB spl, and most commercial mics will produce more electricity than the -60dB for a sound pressure of 94dB, so you should have no problems getting up of "0VU" on your recorder.

If you are going to record very loud sounds you may produce more electrical power from the mic than the M-35 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 94dB spl gives -60dB (1mV) out, 104dB spl will give you -50dB 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-35. If your mic is in fact producing -10dB 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 94dB 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 94dB spl, the electrical output of the mic is given as -60dB, meaning so many dB less than the reference 0=1 volt. In practice, you will see levels of -60dB for low level dynamics, up to about -40dB or slightly higher for the better grade of condenser mics available today. TASCAM recorders and mixers work at a level of -10dB referenced to 1 volt (0.3 volt) so, for 94dB spl, a mic with a reference output of -60dB will need 50dB of amplification from your M-35 or recorder in order to see "0VU" (-10dB) on your meter. Now, if the sound you want to record is louder than 94dB spl, the output from the mic will be more powerful and you will need less amplification from your M-35 to make the needles on your recorder read "0VU".

THE BLOCK DIAGRAM AND GAIN BLOCK DIAGRAM

Before you begin reading the next section of this manual, flip out the extra fold on page 42. On this page, we have printed the block diagram. It shows the signal flow through the M-35 and it represents in simple form, the actual electron arrangement of all the jacks, controls and gain stages from mic-in to line-out.

The diagram on page 43 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-35 successfully with just these two diagrams and a list of input and output specifications.

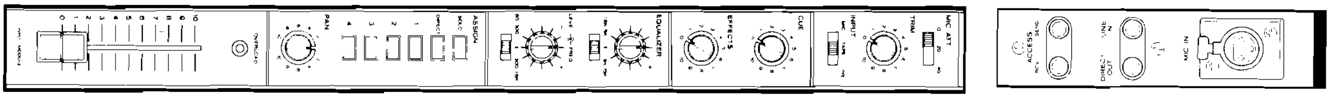
Any question about function or gain can be answered by studying the drawings. Will the accessory send signal change in level if the input fader is moved? No, the signal is shown leaving the main line before the input fader. You read both diagrams from left to right, input to output.

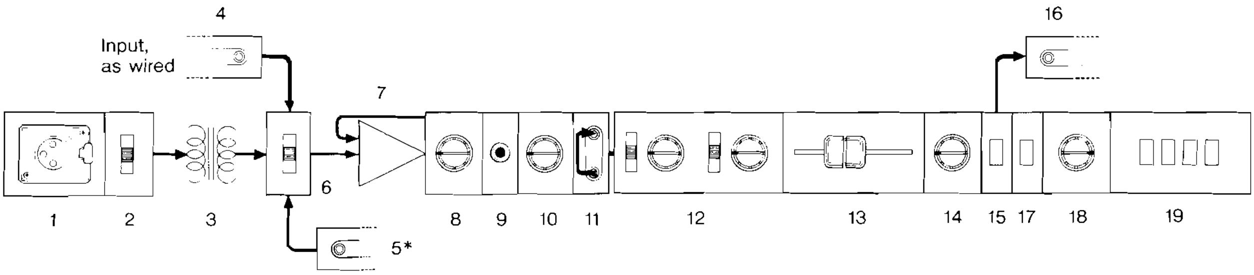
When printed in its entirety, a block diagram can look formidable, and tracing a signal path is not easy, so to aid you in your initial understanding, we'll continue to use our 3 drawing system first shown in the introduction, but in slightly smaller segments.

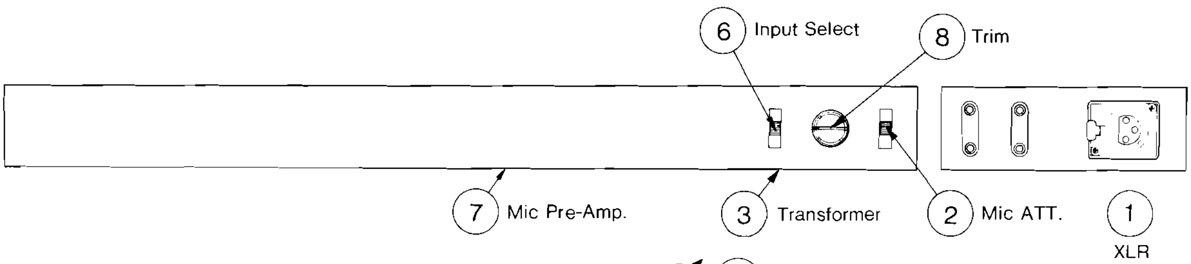
1. As laid out for convenience.
2. As wired, but knobs and jacks as they appear on the outside.
3. The block diagram, with the controls numbered to correspond to numbers on the first two drawings.

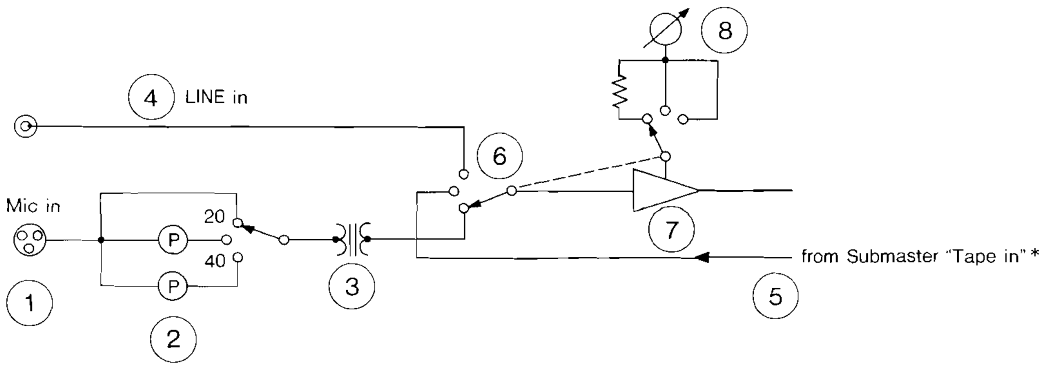
Even with this "translation system" to help, multiple sources and outputs can complicate things, so when necessary; we will also include other types of drawings to help get the point of a subsystem across when we first encounter a source "point" that will be used in a specific way. This may require re-reading if you are not familiar with subsystems, but we think it best to advise you as early as possible.

INPUT MODULE









All 8 input modules are identical and can be interchanged without modification.

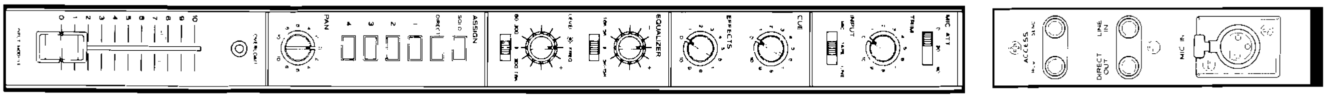
Mic input XLR connector ①

Balanced pad circuit control MIC ATT switch ②
3 positions are provided: off, or no effect, set rightwards one step, a loss in signal of 20dB, set rightwards two steps, a loss in signal of 40dB. Before using the first step, reduce the 8 trim control to minimum or furthest counterclockwise rotation. Since the combination of trim and pad is a maximum of 60dB loss, it is possible to use this mic-in jack as an emergency "line in" if no tools are available to convert a 3-wire circuit to a 2-wire RCA connector – if the line level signal can be reduced to a max of 0dB, (1V).

Input transformer ③
Maximum signal to this internally mounted transformer is -35dB (17.8mV) without using the pad or MIC ATT switch. At 20dB pad, maximum input is -15dB (17.8mV). At 40dB pad, maximum input is +5dB (1.8V).

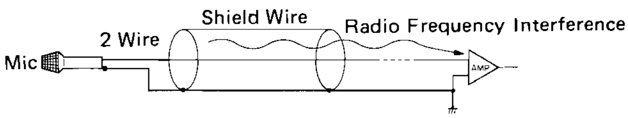
This 3-pin connector, pad circuit, and transformer are the only 3-wire circuits in the M-35.

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 use this more expensive design to begin with if it offers no improvement in quality? The low-power signal that the mic generates must be protected and isolated from other low-

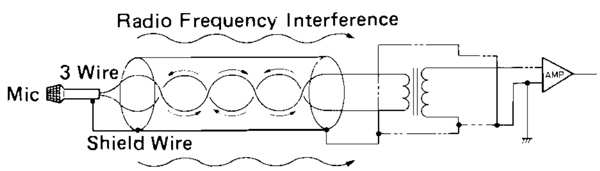


power signals in the real world. Radio power line hum, crackles and switching noise when motors start up (do you have a refrigerator on your AC line?)—all these unwanted things—must be kept out of the very high gain amplifiers that are necessary to raise the mic signal to a working level. So, the balanced or 3-wire, circuit and input-isolation transformer becomes the only sure way to deal with the problem:

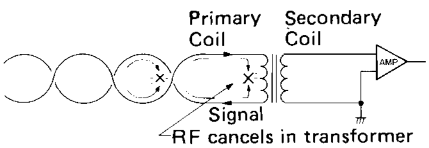
Here's how it works:



Any signal will pass to amplifier, no rejection.



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



Signals with opposite polarity in the primary coil will general 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.

Input Select Switch

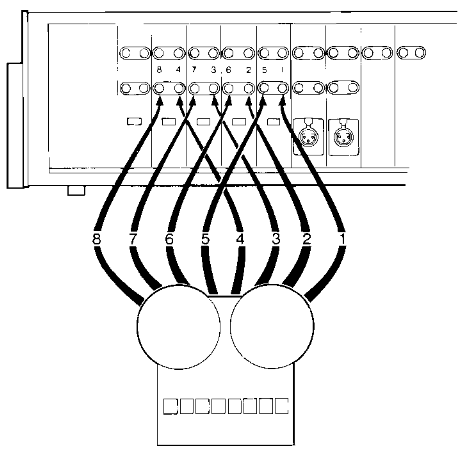
This switch has 3 positions. Left selects the MIC-IN XLR. Right selects the LINE IN RCA jack on the back of the module, and center selects one of the TAPE IN jacks on the buss master modules. Since each input module will receive only one TAPE IN signal, we'll provide a chart to show which signal goes to which module.

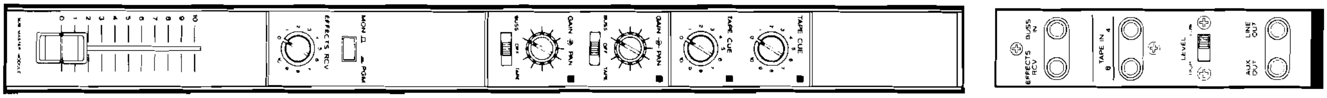
Right here we have our first major problem in comprehension. The connection and its circuit is drawn plainly on the block diagram, but what does it mean in functional terms? Why is the INPUT switch wired to this extra LINE IN when there is another LINE-IN on the module? The answer lies in the requirements of an 8-track system in use, and to explain, we'll have to show the system in its entirely, even though we have not reviewed the first path to the recorder at all.

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

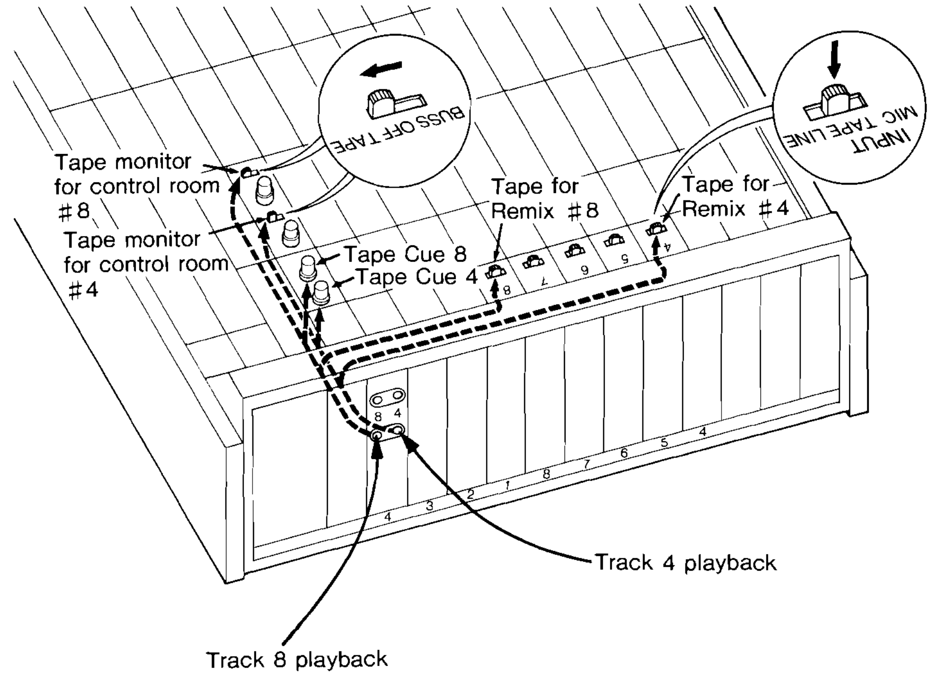
1. Simple playback to judge performance, requiring no corrective EQ. In short, what did you record?
2. Simple playback into a cueing system so partially completed tapes can be finished. This function should somehow combine the signals of simple playback with new mic signals, so musicians may hear when overdubbing.
3. Final remix, when the full control capability of the system (EQ effects mixing, etc.) can be used to "fine tune" the finished tape.

Three tasks, one output. How do you plug in? This special input RCA Jack is on the master module, not the input module, and the 8 sections tions are laid out on the back like this, part of our standard "working patch" for 8-track recording.

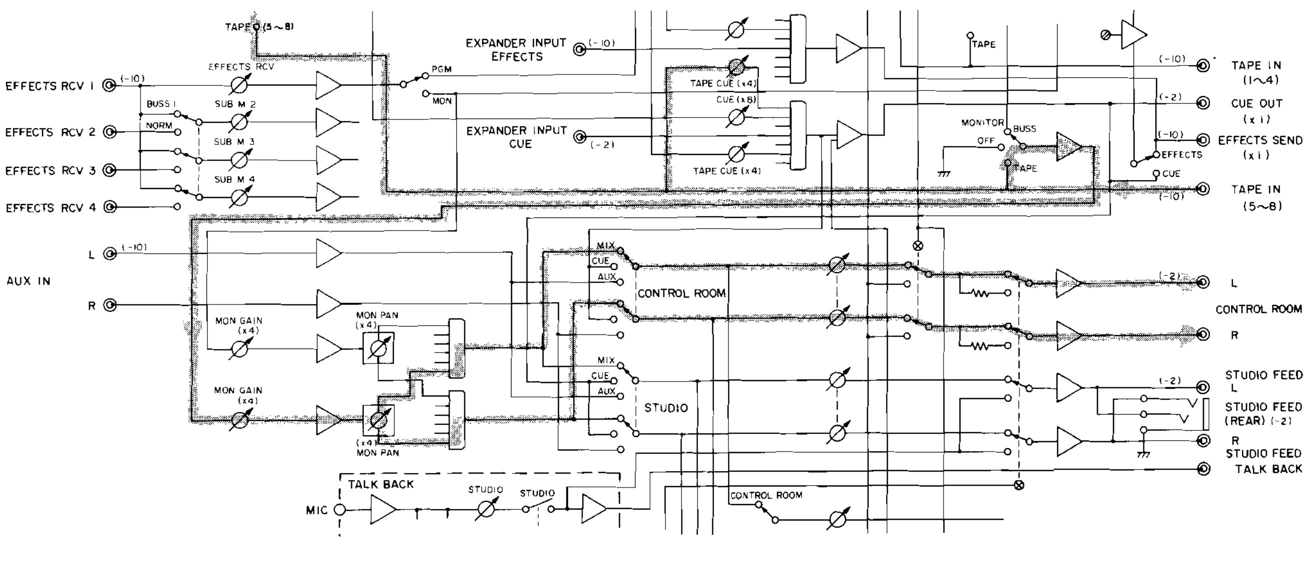


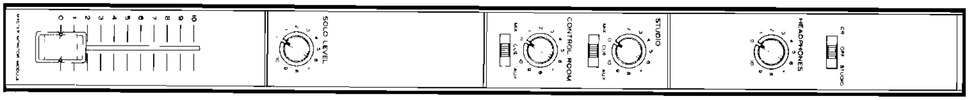


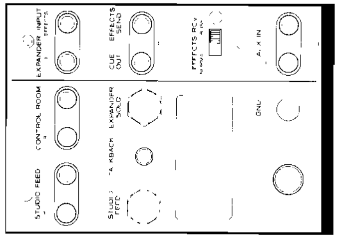
The numbers on the input module now relate to the jacks on the submaster. Track one from tape playback will now be available on three separate systems. If only the "line input" on the input module is used, the signal will only be available for "re-mix" and all module settings for "mic" will have to be disturbed — every time you playback. By using the TAPE-IN jack, resetting is avoided. Another drawing may make the wiring more understandable.



To keep the routing clear, we show only the last submaster module that handles track 8 and track 4. The other groups are similar, routing signal to their respective input module numbers as shown in our first drawing (the one with the 8-track).

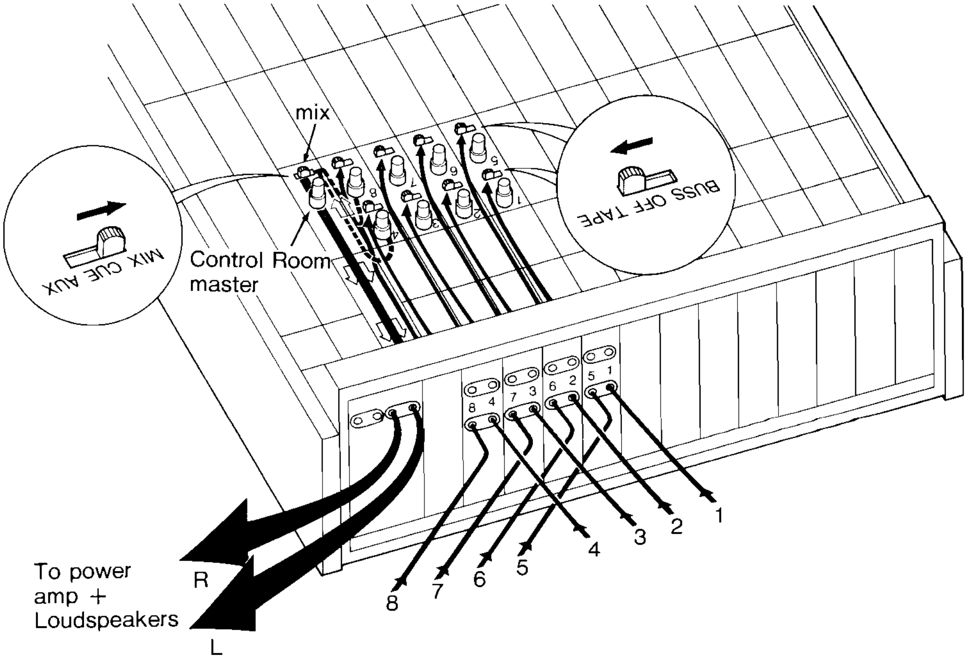


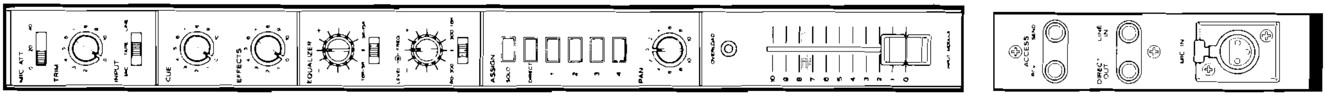




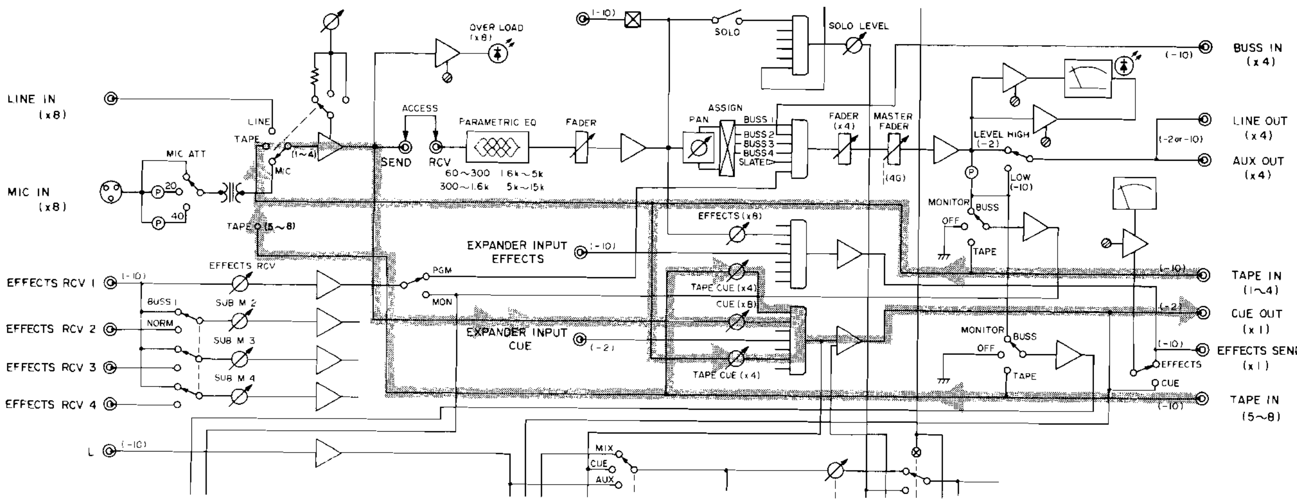
Now for the three requirements of tape playback:

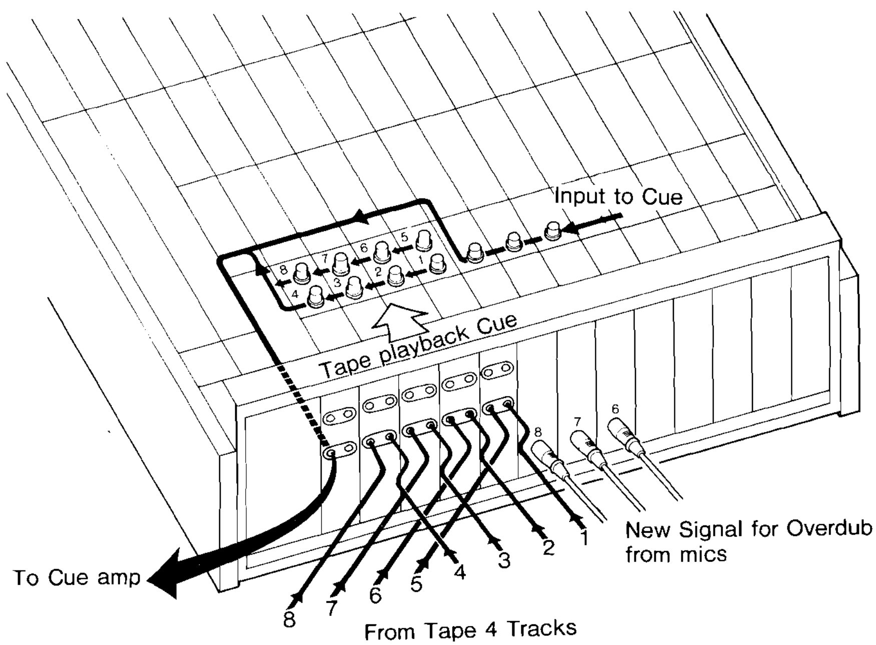
1. Simple record check
   To do this, the tape playback signal is substituted for the monitor output signal on the submaster monitor select, and the monitor mix is now derived from tape playback on each section so switched. Any or all 8 may be selected individually and the control room master will then set a level for the control room loudspeakers.



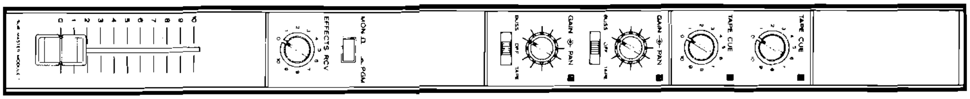


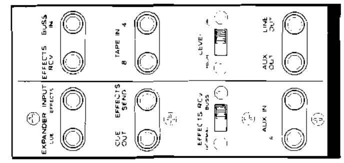
2. Cue System
Tape playback plus mic cueing for overdubs.



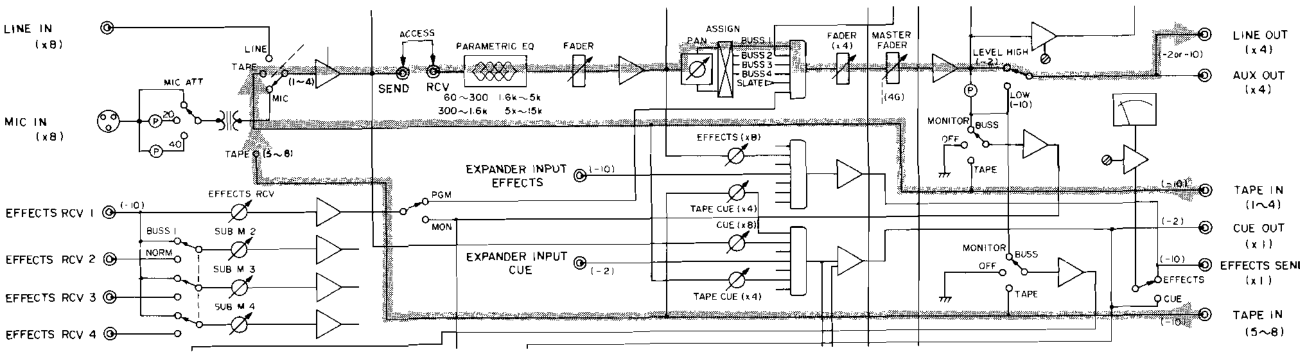


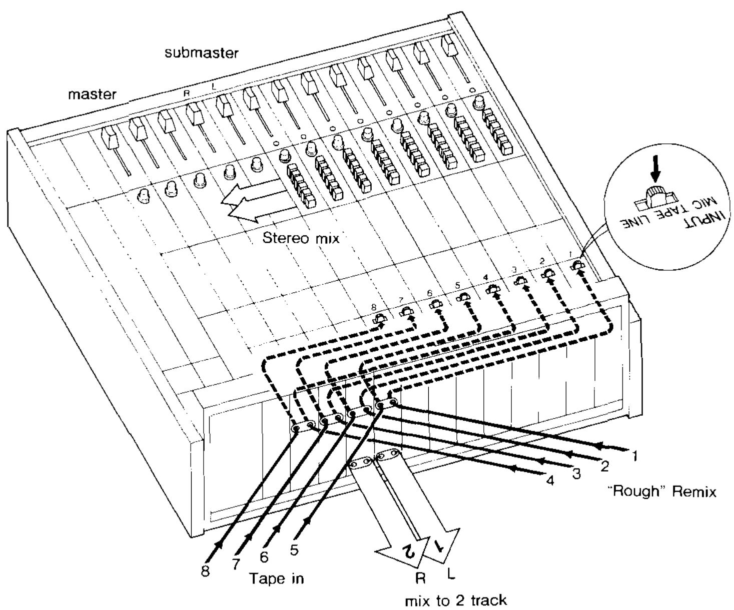
This cue system combines the tape playback controls (x 8) on the submaster modules with the input cue controls (x 8) on the input modules to form a mono sum of all 16 possible signals that might be needed for a musicians cue. Since the monitor system for the control room can be set to audition this signal "mix" you will be able to "hear" it, and a meter shared with the effects buss allows you to set levels; but caution is advised. The headphones volume may not relate directly to your control room volume. You have a master for loudspeakers, but it does not affect this mix in the headphones. The cue mix has no master pot.





3. Tape-in for Rough Remix

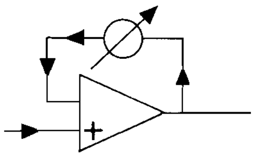




When the INPUT select switches on the modules are set to the center position, the submaster TAPE IN jacks are internally connected to the input modules. Selecting this remix position on the input module will not disable the normal operation of the monitor. Signal will go to both circuits at the same time, allowing separate use of the monitor outputs as extra mixes. True stereo echo is an obvious first choice, and the mixdown machine can be monitored using the AUX IN position on the control room monitor module. Since the remix position can be selected one module at a time, a single track may be equalized and monitored without disturbing MIC IN settings on the whole mix. A decision can be made quickly on the artistic success or failure of an individual part without the need to place the entire console in remix mode just to see the effect of corrective equalization on a single track. Since the TAPE IN jacks on the buss master moduels actually feeds 3 separate mixing positions it will present a more severe load to anything connected to it than "Line in" on the module. TAPE IN's have an input load impedance of 20k ohms and LINE IN's (on the input module) have an input impedance of 50k ohms.

Line In Jack
An RCA jack on the rear of each input module. The maximum signal you can apply here is +15dB. The MIC ATT switch does not affect this input. The input impedance 50k ohms.

Trim



This control will alter the gain of the first amplifier in the console. It will affect the level of any signal, MIC, LINE or TAPE. With this pot rotated fully clockwise (rightward), the maximum gain of the first amplifier is 26dB. In this position, the maximum input signal before overload is -11dB (282mV). When the pot is in its minimum setting (fully counterclockwise or leftward) the gain is reduced to 5dB and the maximum signal that can be handled without overload will be +10dB (3.2 V). Remember, these overload figures refer to the input of the amplifier, not the input plug or connector. Losses occur, and pads can be inserted before this point. The maximum signal that can be applied to the LINE IN jack is +8dB (2.5 V) with the TRIM rotated fully leftwards, and +30dB (31.6 V) with the TRIM rotated fully rightwards. Trim pots control the gain of an amplifier by adjusting the amount of output signal returned to a secondary input control "pot" or input. Because of this reverse control aspect, we consider it unwise to adjust the TRIM while signal is being recorded. Obviously, you must adjust when signal is present, but when serious recording is in progress the possible negative side effects on amplifier stability and distortion indicate that you should "mix" with the straight line input, submaster, and master faders only, and adjust "trim" during rehearsal.

Cue Pot
At this point in our input module, we derive signal for a cue or headphones mix. This function is best served by a mix of signals that will remain constant after being set, so it is drawn off, before the input fader. To raise the level, rotate rightwards or clockwise.

This pre-fader source insures that your mixing decisions will not interfere with the rehearsal in the studio. The only thing that is more annoying to a player wearing a headphones than a sudden change in tone is losing track of the sound of his instrument entirely. Remember, all music execution is in large part a response to what is heard, and if the main source of sound is provided by this headphones "mix", and you turn it off in the course of some other control room action, you will deprive the player of the creative guide to what is going on. Your session may stop cold right then. The cue system routing has been drawn on page 18 and shows the 16 sources of signal and the inputs and outputs on the back panel.

Overload LED
When signals high enough to make the ACCESS SEND jack output exceed +15dB are applied to an input module, this LED will light up. The TRIM, or the MIC ATT pad should be adjusted until the LED remains out when signal is present. When recording extremely percussive transient material, it may require full negative trim and pad (MIC ATT) to prevent this LED from flickering on strong peaks. Changing to a less-sensitive mic may help.

Access Send – Rcv Jacks
The high gain provided by the mic preamplifier allows us to place a "patch-point" in this more useful position. The level at the SEND jacks is -10dB (0.3V) and the output load impedance is 10k ohms. A limiter connected to this point in the M-35 circuit can now be set to a range of compression that will not be altered when either the equalizer (next stage) is adjusted or the input fader is moved (the stage after the EQ amps). This pair of jacks is not "normalled" so, when no device is bridged from SEND to RCV, jumpers must be in place for signal to flow to the EQ amps and on through the console. However, since all the mixing controls lie after the "RCV" jack, it is possible to use "ACCESS RCV" as an input, and by-pass the first gain stage. The only functions that will be lost are the trim and overload indicators. The signal quality will improve slightly but it will not be possible to switch to "MIC IN", "LINE IN" or "TAPE" without repatching. This unorthodox patch is suggested for final remix when all recording has been completed, and more time for patching is available. Maximum level in will be +15dB. Input impedance is 220k ohms.

PARAMETRIC EQUALIZER SECTION

Before we begin, the label itself will require some explanation. What is parametric and, equal to what? A logical 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 we get this label on our tone control?

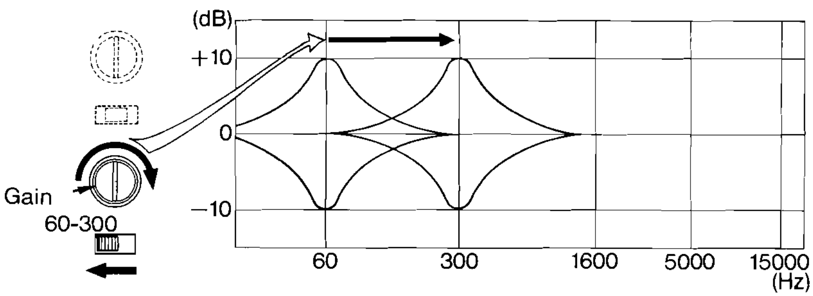
The telephone company uses it. In the early days, the system worked well in the lab, and in short runs of 100 yards or so, but .....

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 set the terms we use today.

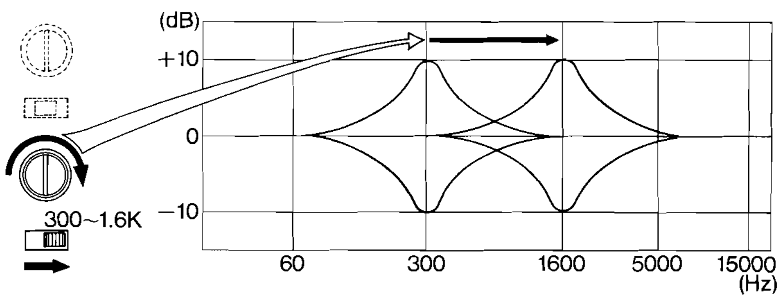
The term "parametric" refers to the adjustable frequency point. The "parameters" or "rules" are not fixed at any given number, but are continuously variable. Both aspects of the circuit, frequency center point as well as gain or loss are continuously adjustable without "steps" but there are limits.

The Model 35 offers a two control, four range parametric equalizer. The lower group of controls offer ±10dB of boost/cut control at any frequency between 60Hz and 1.6kHz in two ranges, selected by the switch below the concentric controls. Set leftwards, the switch selects the range from 60 to 300Hz. The outer knob when see fully leftwards or counterclockwise then sets a center frequency of 60Hz. As the outer knob is rotated rightwards (or clockwise), the center frequency is raised in stepless fashion. When fully right (or clockwise), the center frequency is 300Hz. The inner knob is the boost/cut control. Set fully leftwards, the cut is -10dB. Set fully rightwards, the boost is +10dB.



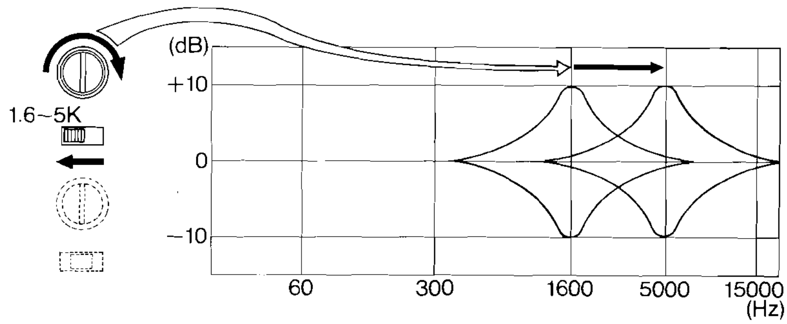
Outside Control,
rotate right to
raise center frequency.
Inner Control
rotate leftwards to cut
rotate rightwards
to"boost"

When the lower switch is set fully rightwards, the action is the same, but the range changes.

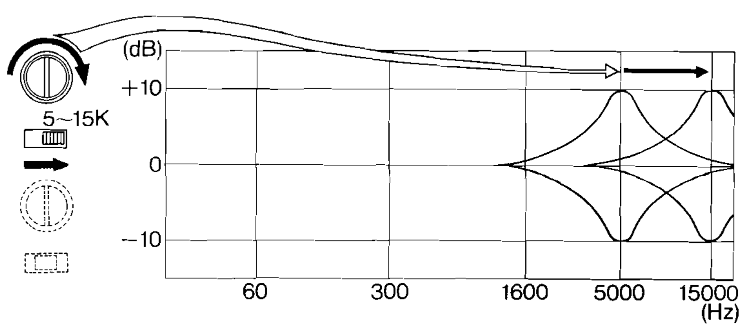


The upper group of controls offer ±10dB of boost or cut at any frequency between 1.6kHz and 15kHz in two ranges, selected by the switch below the concentric controls. Set leftwards, the switch selects the range from 1.6 to 5kHz. The outer knob when set fully leftwards or counterclockwise then sets a center frequency of 1.6kHz.

As the outer knob is rotated rightwards or clockwise, the center frequency is raised in stepless fashion. When fully right, or clockwise, the center frequency is 5kHz.
The inner knob is the boost/cut control. Set fully leftwards, the cut is -10dB. Set fully rightwards, the boost is +10dB.



When the upper switch is set fully rightwards the range once again, is shifted upwards.



The center position on both upper and lower frequency select switches is "off".

The great advantage of a "parametric" or continuously "tunable" equalizer over the more conventional "set" types is that you can tune your frequency center point to the precise area you need, and then cut or boost will have the maximum desired effect on your art. You get the result you want with much less rotation of the boost/cut control, and put less "strain" on your electronics. No matter how many "frequencies" there are on a "set" type equalizer, it is unlikely that anyone will be "just right" and many more ranges are needed to do the job. In the long run a "parametric" type requires fewer parts to do the job so it costs less, and performs better than "graphic equalizers" which may leave many sections unused on a given input, or "set frequency select" types which never seem to have the right "number" available. "Less" as a working concept in electronics is always the safest route, so before using the tone controls on mic signals, it is better to get as close to the sound you want by moving the mic. Even a small change in mic location can make a big difference in the sound quality. Listen to the sound from the actual mic position. Place your head where the mic is, and listen carefully. What do you really hear? Is it what you want? Doing this check may help solve many problems. Too much bass? Not enough? Well, perhaps there is a better location for the mic in order to get the balance you need. When you have gone as far as you can in this fashion, the tone controls will get you the rest of the way.

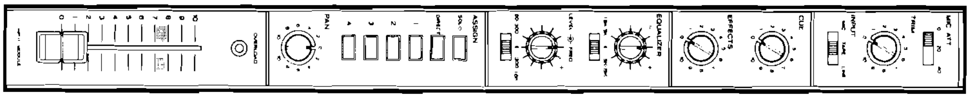
However, the "ear test" may not be wise, if the volume of sound is very high. Don't put your head near to any part of a drum set. Even if only moderate force is used to play drums, at close mic distance, the sound power may be enough to cause permanent damage to your ears.

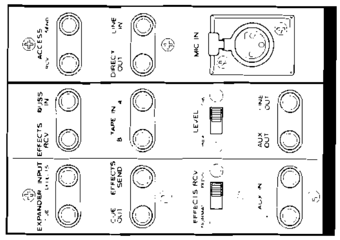
If you have the time and a co-operative musician, experiment with different combinations of mic placement and tone control settings. Although it can be very tiring for someone to play a part over and over again while you "go to school", it's the best way to get the knowledge of mic technique and tonal balance you need to make good mixes. In fact, experience is the only teacher that will work on your specific problem, i.e., your guitar, your voice, your music. All the information we can give you in the manual will only be a starting point. How far you get will be up to you.

On the block diagram and gain chart you can see that the tone control stage has a moderate gain (±10dB) and a very large excess gain capability or "headroom" (25dB). This gain chart is made with the assumption that the tone controls are set to the "flat" or "no boost or cut" position. The reserve in the circuit is necessary to maintain a 20 dB value of "headroom" when the tone controls are set for maximum effect. Without this extra margin, you would have to lower the setting of the input fader when you used the extreme boost or cut settings of the tone controls.

Add it up – if you start with a reserve of only 25dB and you boost +10dB (the maximum) at 10 kHz, your margin of safety is reduced to 15dB. For a steady tone from an organ or a violin, this might be just enough to avoid clipping, or serious distortion, but it is definitely not enough to cope with any percussive peaks from things like piano, guitar or drums.

Even experienced engineers have a tendency to forget that "cutting" the lows will have a similar effect to "boosting" the highs, and is much easier on the electronics (cutting leaves more headroom and consequently causes less distortion). The results are not identical but close enough to warrant trying. Cut bass, raise the overall gain, and see if it sounds better than simply boosting the highs.





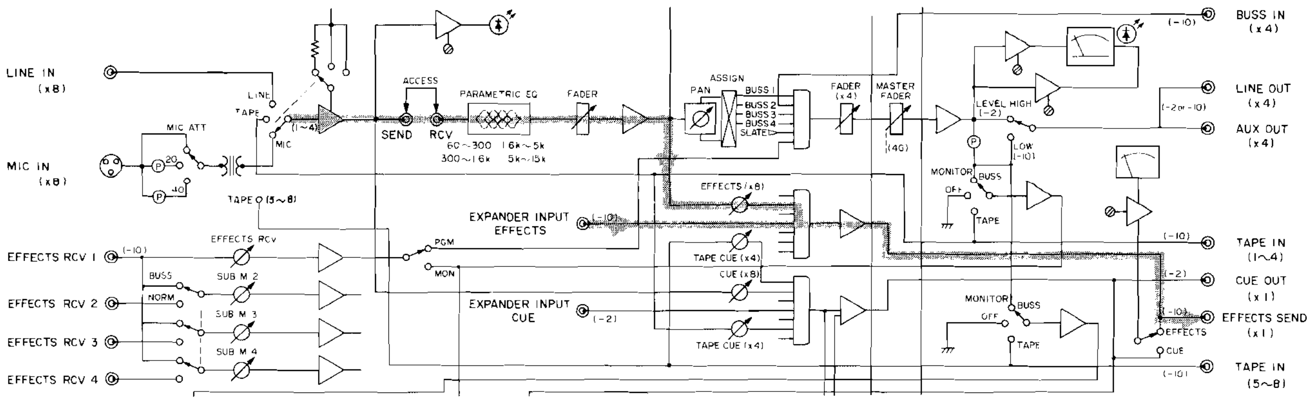
Input Fader

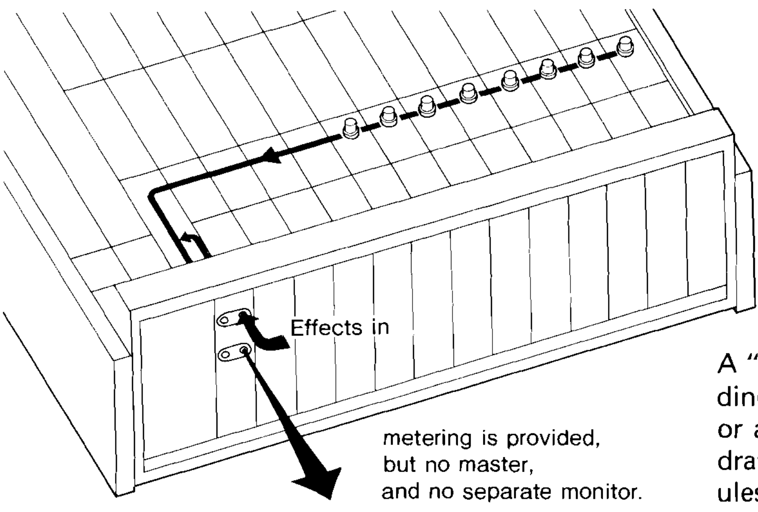
Controls the signal level from all prior stages. Faders, also called pots or attenuators always cause loss in order to control signal level. Gain stages in an electronic device always run "wide open" at whatever gain they are set for, unless they have provisions for "TRIM" or actual gain adjustment. In the M-35 only one of the many amplifiers employed actually has "Trim", the first gain stage or mic preamplifire. When you advance any straight line fader on the M-35 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.

Effects Send Pot

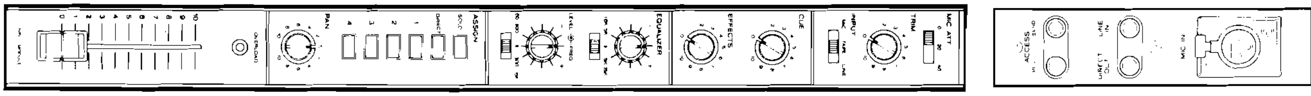
This rotary control is the preferred location to use when a secondary "mix" must reflect the prior mix decisions of level and equalization. "Echo Send" is a good example of such use and this post-fader, post-equalizer signal will then also "fade out" as you "fade out" the regular signal with the input fader.

Eight signals, one from each module can be combined, and metered and will appear on the back-panel output jack marked "EFFECTS SEND". On the M-35, this mix cannot be monitored separately until it is returned after processing by the effects device. This send has no master control on its output.





A "buss in" or piggyback, is provided to allow adding more such "effects" signal from an expender or another mixer. "Effects Return" lines will be drawn later when we get to the submaster modules.

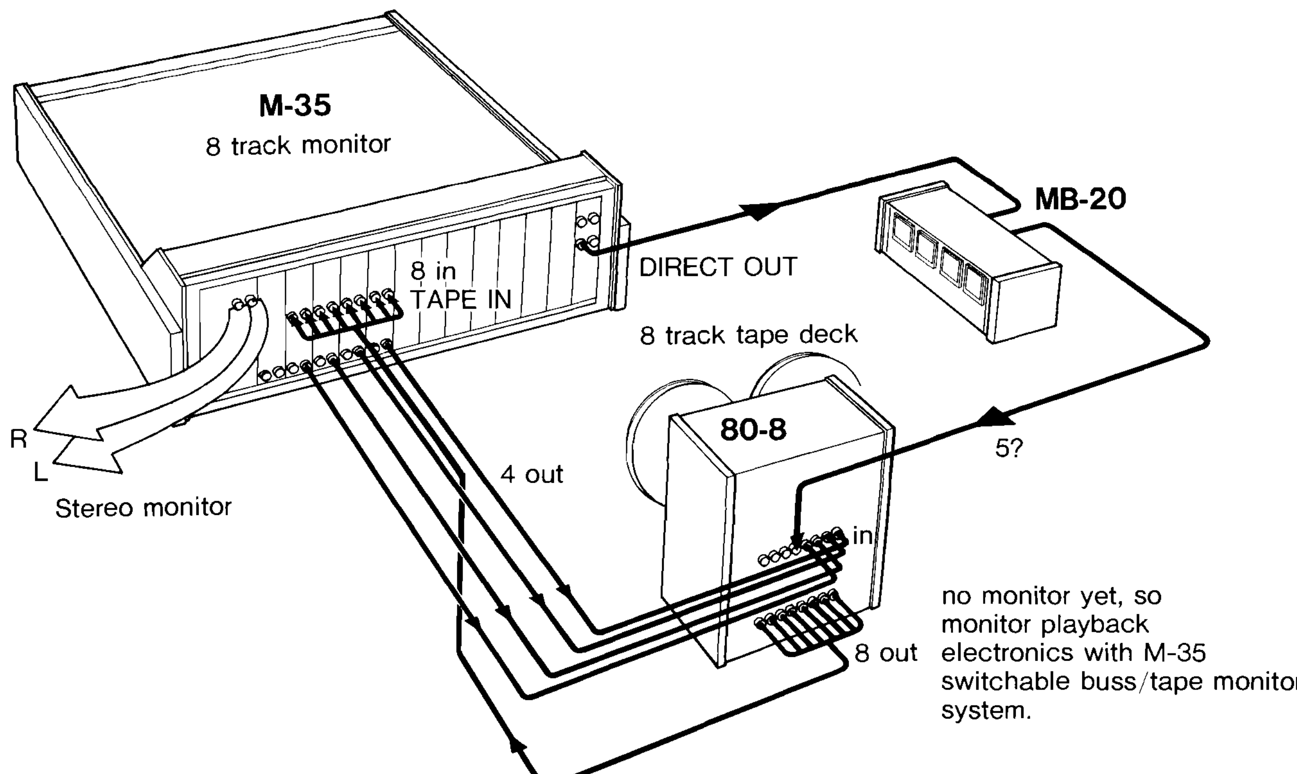


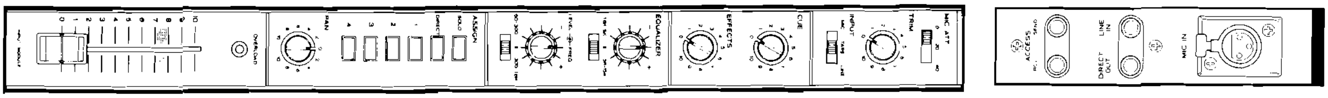
Direct Out Switch
When depressed, allows signal to pass to the direct out jack on the back of the input module. Push to lock, push again to release.

Direct Out Jack
Provides an unmixed single signal output of whatever has been assigned to the module. This direct output can be used for a variety of purpose, such as:

1. A subsidiary mix can be made by using an accessory mixer fed by this output.
2. One-mic, one-track recording happens frequently, and using DIRECT OUT will pass unneeded summing networks and amplifiers. Going "Direct" to the recorder will result in a cleaner signal.

Channel Assign and Pan Pot
On the M-35 monitoring will be possible by switching the appropriate monitor section to "TAPE" and listening to the recorder instead of a buss master. For metering, either the recorder or an accessory meter bridge (MB-20) can be used.
Since the amplifier that feeds this jack also feeds the buss-assign network and the post fader effects system it is wise to calculate the total load carefully. The output load impedance of this jack is 10k ohms. The load of the pre-wired circuits restrict the connection of this output to a single circuit at a time, unless the equipment you plan on connecting here is known to have a very high values for input impedance (100 k ohms or more).





Solo System

This momentary pushbutton will divert input module signal directly to the monitor section, replacing the submaster monitor mix with a single signal. Although these switches do not latch, more than one may be depressed simultaneously, providing a mix of those buttons "held". The signal "mix" in the monitor is "center feed mono". This location as a source for signal on the input module will allow auditioning EQ, but the input fader must be advanced for the signal to flow.

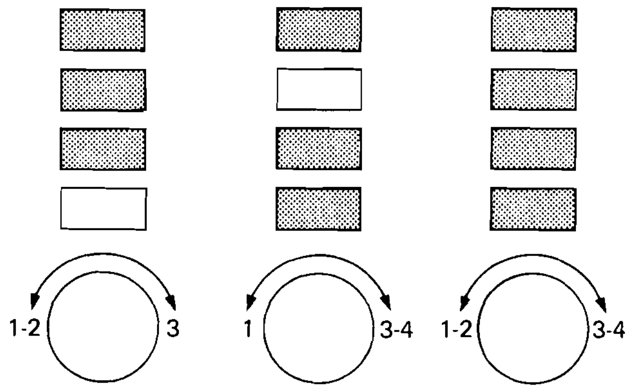
Since no "return type solo" is provided on the submaster modules, effects-returns will be muted as well as monitor when any solo is held down. Since this entire system affects only the monitor output (not the direct out or the master buss-outs or any other minor as well as major output), all recording will continue unaffected while this solo system is in use.

The channel-assign switches and the pan control together make up the last section of the input strip. At this point you have selected a signal, it has appeared at the direct-out jack, its level has been set with the input fader and you have made the necessary changes in its tonal quality.

What master "line out" do you want it to go to? Line one, line two, line three, line four – any or all may be selected by depressing the appropriate buttons on the channel-assign strip. Push to lock, push again to release.

Depressing more than one button will engage the "Pan" control. This single knob works two faders that are wired "back to back". As you rotate the knob, one fader is turned up as the other is turned down. When the control is "dead center" each fader is still reducing the signal slightly so that the signal transition through "center" does not become louder as you "PAN" through it. When both speakers in a stereo pair are producing sound, you don't need as much power to maintain a constant volume. If only one button is depressed, the pan control has no effect on the signal. When any two buttons are depressed, the lower number is "Panned" to full on when the control is turned as far as it will go counterclockwise. The higher is "Panned" to full on clockwise.

The "PAN" logic for 3 or 4 button arrays is easiest to explain with some drawings. If shaded buttons are assumed to be down, the logic is:



Typical examples of multichannel panning

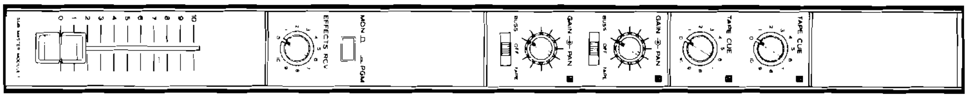
Summing Amplifier

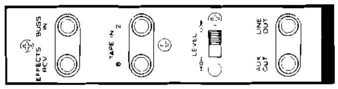
Leaving the Pan/channel assign, the signal is passed through a "summing resistor" and suffers a big drop in level before it is allowed to pass down the summing amplifier. This loss is necessary to prevent the signal from one input going back into another instead of going down the line to the master fader. You can think of the summing resistor as the "traffic cop" that turns the line into the "one way street" you need here.

Buss In Jack

The primary purpose of this final input on the block diagram is to "stack" or run a pair of mixers with one overall master control. The input impedance and signal point are identical to the accessory receive jack. Any electronic device that has a compatible output impedance may be connected here and its contribution to your mix will then be controlled by the submaster fader on the M-35.

SUBMASTER MODULE





Effects receive to "buss" function actually comes first only if the latching pass switch is down, but by the rule of "wiring first" we'll pick up the system now, and mention that we will see it again in the monitor section.

Effects Rcv Jack
An RCA connector on the back panel of the submaster module intended as line level input to either the summing amp (Buss) or the monitor section signal route, depending on the position of the mon/pgm switch.

Effects Rcv Pot
This rotary control adjusts the signal received from the RCA effects-receive jack.

Mon/Pgm Push Switch
When depressed, signal from the effects RCA Jack is assigned to the buss summing amplifier.

When raised, signal from the effects RCA jack is assigned to the monitor system only. You can hear it, but it won't "record".
Let's leave the monitor system and its multiple functions for a while and return to finish our "main line" out.

Submaster Fader
This fader controls the overall level of signals from the input modules and the effects receive section, when effects are selected.

Master Fader
This four-ganged (4 Section) pot provides final level control of all four output sections of the mixer. This signal control operates all four faders at once. Any signal added to a buss from the various jacks on the M-35 will be affected by the setting of this control, if your mixed signal is taken from the last output pair on the block diagram.

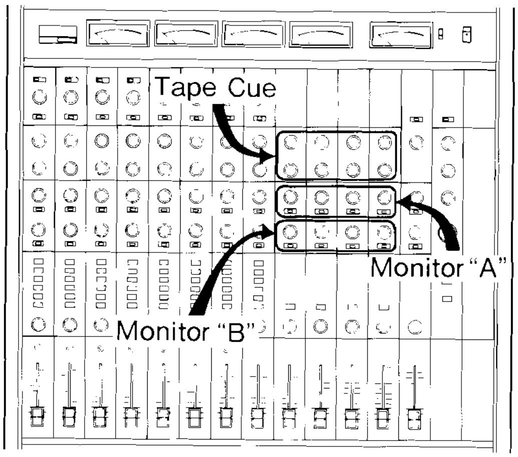
Meter Drives and Meters
Two lines are shown on the block diagram here: one to the monitor circuits, one to the meter circuits. We'll go "up" on the block first, and deal with the visual references, the meters and peak LED. These two circuits can be adjusted internally, and will respond as set. Original setting for the LED ampl is 0dB above 1 volts. Original setting for the meter is 0VU = −10dB.

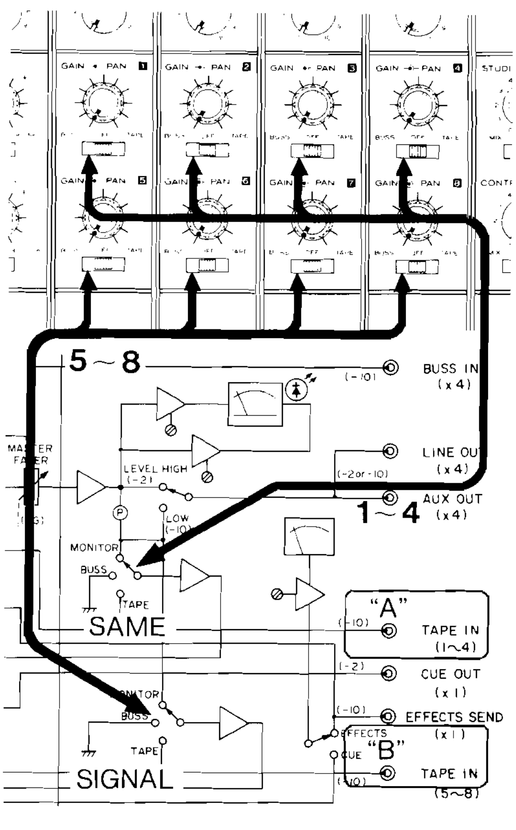
Pad Switch
On the back panel of each submaster module, this switch when set rightwards (when you are facing the rear panel) will lower the −2.2dB (0.3V), output to −10dB in order to match the inputs of most TASCAM recorders. When set leftwards, the output will be higher. There is no need to adjust the meters because they follow the signal at "high" all the time, so the "zero" will be visually correct for either setting.

Line Out — Aux Out
One last gain stage appears after the master fader and just before this double output jack. A small amount of gain is necessary to make up the loses caused by wiring up the master fader, and to give a solid source of signal with a stable and relatively low output impedance to whatever you are mixing to. The final reference level is dependent on the setting of the high/low switch on the back of the submaster modules.

0 = Either −2.2dB (0.78V) High
0 = Or −10dB (0.3V) Low

Since the two output jacks are connected to the same gain stage, any device connected to one pin will affect the output capability of the other pin. To determine the true value of loading on the mixers final stage, the input independence values of both devices must be considered even when only one of them is being used. For this reason we suggest that you unplug anything connected to the final stage that you are not using when you make your most critical mixes. The output impedance of this stage is 100 ohms.





It's always a good idea to use as small an amount of electronic stages as you can. If you don't need a circuit for its control or function, bypass it and your sound will improve.

At this point in the circuit we have a complete Buss Master Mix. All functions of the input module will affect the signal here, all major and minor patch points are behind us, and as far as the recorder is concerned, the signal is ready to record – we are done. The only problem remaining is: how do we hear what we are doing? This signal must go to our monitor circuits.

Monitor Section Function

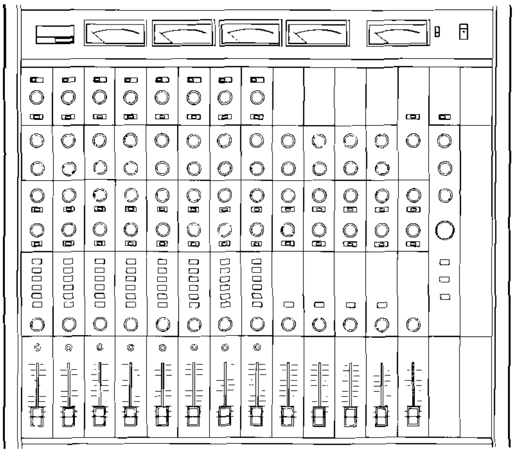
The block diagram shows the controls that influence the monitor signal as a "one module" layout, but all four submaster modules have slightly different labels to account for the TAPE IN jacks and their relationship to the input modules. To make functions more understandable, we'll show the entire top panel as it appears on "the outside". Several aspects of control require the entire section, and the top panel will help.

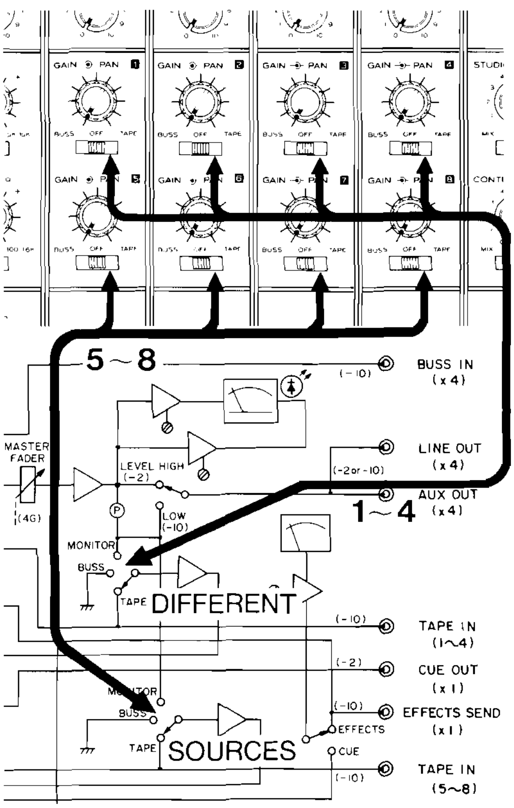
Buss as Source (Diagram)

There are two switches that seem to do the same job, feeding signals to the monitor system. This apparent doubling of function is provided for several reasons. Even though the M-35 has only four buss master modules, it has an 8-track monitor system. Setting the monitor select switch to the left selects "buss master output" as a signal. Resetting to the right-most position will now select "Tape" as a monitor, and we need eight positions to monitor the playback of all 8 tracks. Because of this 8-track monitor requirement, all monitor controls are doubled and there are two apparently identical sets of controls. When used as "Buss Master" monitor, they become redundant, but the two sections are not identical in "Tape" mode. Monitor section "A" is connected only to TAPE IN A, and monitor section "B" only to TAPE IN B.

Tape as Source

If our basic 8 track setup on Page 33 is used, you will get a different track on each section of the monitor or Buss Master Module (one for each). Upper tape select will be track 1, lower tape select will be track 5. Both monitor section will see buss master 1 when Buss Out is selected by their Monitor Select Switches. However, this switch does not control the logic of signal selection for the Tape circuits immediately above, it only works on the rotary fader marked "Mon Gain".





Monitor Gain/Pan

This dual concentric pot (one inside the other) controls both the level and the monitor pan. The inner section is the gain, the outer section is the pan. 8 sections are provided for 8-track playback or 4 buss output monitor. Each section is individually switchable to Buss or Tape.

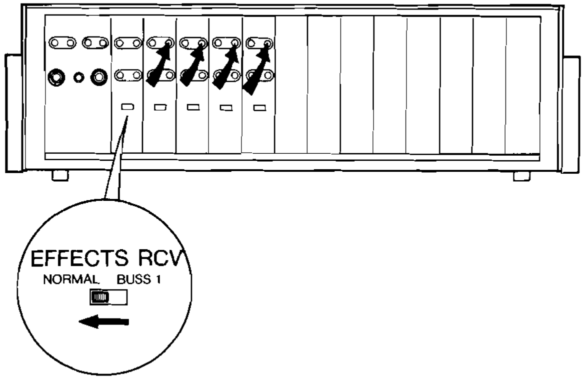
Monitor Pan

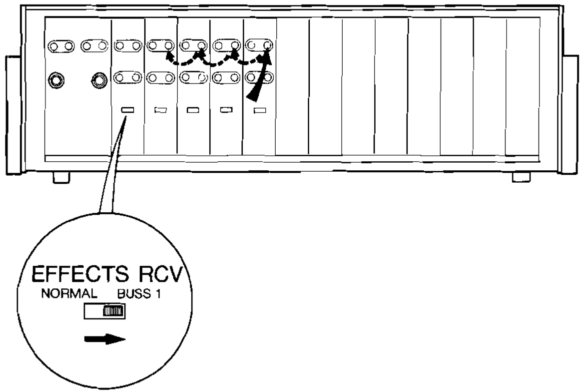
The Monitor Pan affects only the monitor mix. It will not alter the level or pan position on the Buss Master output when input panning is done between two busses; but, if both selected busses are monitor-panned to the same side in your stereo monitor mix, turning the input pan control will produce no audible result. Input panning will, in fact, affect what you record, but you won't have any indication of what you are doing unless the "Monitor Pan" on the two busses are set one to each side. This fact can give you a lot of trouble. Work through this part of the block diagram carefully, it is very complex, and can take sometime to understand thoroughly.

EFFECTS RCV Group

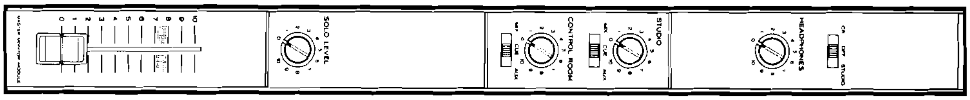
This group of controls enters the monitor at the same point as the Buss/Tape monitor select switch, but slightly before it. If the switch is set to its center position, the Effects return signal will be all that remains in the monitor, and you will be able to hear "effects only". This signal enters the system prior to "pan" so effects signal will appear in stereo as affected by the pan pot on the monitor section. The entire four pot and button group can be made to receive signal from input Jack #1 by setting a back panel switch to the right (as you face the rear of the M-35) the switch is on the Master Module in line with the level Hi/Low switches, set leftwards, each effects receive section is routed to its own module via its own effects in Jack.

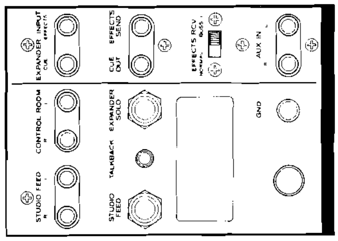
Set rightwards, module 1 effects receive Jack feeds all four sections internally with the same signal.





MASTER/MONITOR MODULE



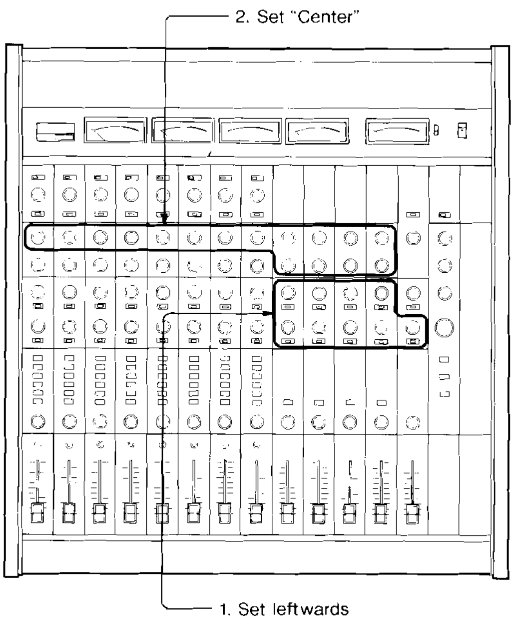


Master Fader
This four-section slide pot appears here physically but the outputs it actually controls appear individually on the submaster modules.

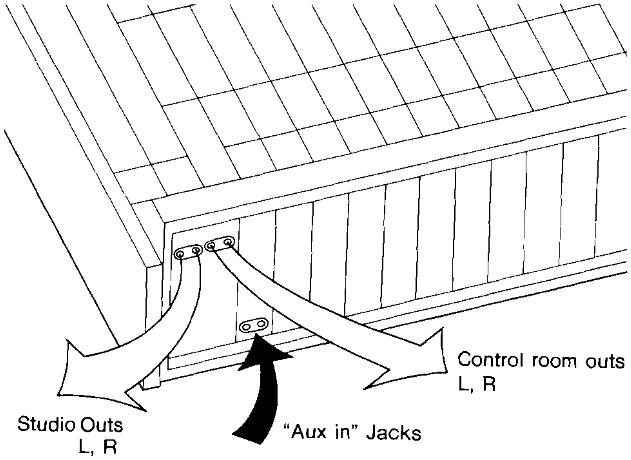
Solo Level
This rotary pot controls the level of the solo system when a Solo signal replaces the regular monitor signal. Check this before you depress "Solo" or the setting may be to high or low.

Control Room Group
This rotary pot and switch controls gain, and selects signal for the control room outputs on the back panel. The switch has three sources.

1. Set leftwards
   Selects the monitor eight-group gain and pan as source.



2. Set "Center"
   The signal source becomes the Cue 16 X 1 mixer.
3. Set rightwards the monitor signal source becomes the signal present at the "AUX IN" Jacks on the rear of the master module they appear in line" with the buss-out/aux-out jacks.



Outputs for the control room group signal are marked "CONTROL ROOM". Output level is –2.2dB (0.8V). Max output before clipping +18dB (7.9V).

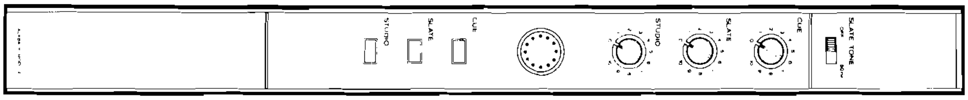
Auxiliary Line In Jacks
A stereo pair provided to allow the monitoring of a 2-track recorder or other subsidiary without the necessity of disrupting assignments or patches elsewhere on the M-35. These Jacks also feed the adjoining Studio Monitor Pot.

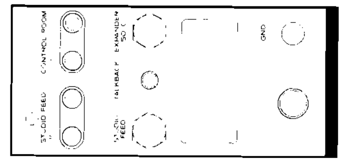
"Studio" Control Group
This group is identical to the control room group in its sources of signal and logic. It provides a duplicate identical in all ways except one. Solo, when depressed, will not appear on this output, it will still carry the "mix" you have selected.

Output level is –2.2dB (0.78V)
Max output before clipping is 27dB (22.4V)

Cue Out Jack
Shown earlier, this Jack carries the 16 X 1 (mono) output of the Cue system.

Effects Send Jack
Also discussed, this Jack carries the 8 X 1 (Mono) output of the effects-send Pots on the input modules.





Cue in Expander Jack
This jack, allows you to add signals to the 16 X 1 cue mix. Level control will be necessary from your accessory.

Effects in Expander Jack
This jack allows you to add signals to the 8 X 1 effects output. Again, since there is no separate control on this input, level control "outboard" will be required.

TALKBACK MODULE CONTROLS AND RELATED OUTPUTS

Talkback Mic
A built-in condenser microphone will allow you to "talk" to 3-circuits on the M-35.

The Cue Buss
A level control is provided, and the top push switch will enable the circuit.

The Studio Output Group
The lowest pot sets the mic level, the lowest push switch enables the circuit.

"Slate"
This push switch is in the center and its related volume control will apply talkback signal to all four output masters simultaneously, allowing you to verbally identify a tape (what take, what tune, etc.)

Slate Tone
If this switch is set rightwards, a 60 Hz tone will be added to the slate push switch and pot. When re-winding at high speed, slate tones can be heard as medium pitch "chirps". If you count them, you will have a guide as to where you are without having to slow down and "play" the tape.

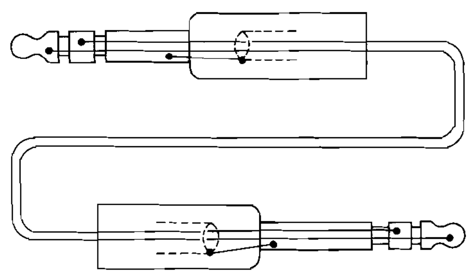
Talkback
This 2-wire jack is provided to give access to the talkback mic signal if you wish to run a separate amp and single speaker to your studio, and use the studio group for some other purpose.

Front Panel Headphone Jack
This standard ring-sleeve-tip (3) wire connector for stereo headphones can be connected here. This jack is driven by a headphone amp.
Max power into 8 ohms is 100mW.
What signal appears here is controlled by.....

Headphone Select Switch
Selects either the control room group (set left) or the studio group (set right).

Headphone Level
For control of headphone volume.

Solo Expander Coupling Jack
This three-wire ring-sleeve-tip circuit couples the solo function between the M-35 and the M-35 Expander Frame. It will not couple two M-35 solos together. Be sure the connecting cable you use is wired this way.



"Solo" cable

The two remaining controls on the M-35 are on the meter panel.

Power Switch
Enables AC, 120V, 60Hz, Max. power consumption is 30 watts.

NOTE: See page 44 for voltage conversion and note for U.K. Customers.

Meter #5 Select Switch
This extra meter can be switched to show the output level of either the cue mix or the effects mix. Set CUE for cue mix, set EFFECTS for effects send.

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 the standard setup. Line level is line level, whatever the source, and many line-level inputs to the M-35 offer a route to a mix that will be used for a function other than the one that is labelled on the top panel. 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 unorthodox ways.

Most people tend to look for a "permanent" set of connections when they set up a mixing system and it is true that the logic of control function just on the top of the mixer takes some time to become familier with, but multichannel recording has many mixing requirements. A permanent patch will severely restrict flexibility. Don't be affraid to re-plug. There is nothing wrong with the concept. If you can examine the system needs of each mode of operation and re-patch the M-35 to suit, you can get better results.

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 all all the connectors. You will need all the options you can get.

RECOMMENDED 8-TRACK SETUP

The basic function discussed in this manual assumes that you will need to playback what you have recorded many times before final mixes are made. Since it is unlikely that you will be recording all 8 tracks at one time, the fact that the M-35 has only 4 buss master modules is not a serious limitation. Here we show each buss master connected to more than one track. Tracks 1 and 5 are on the same buss, #1 and so forth up to Buss 4 Output, connected to tracks 4 and 8. When you are ready to mix to stereo, you will have to change your patch to feed the 2-track. Designed for quick playbacks, the Monitor system eliminates the need to distrupt the Input Module settings you are working with. Since "LINE IN" on the Input Module is not used for playback of your recorder in this patch, these Jacks are available for any other unit or units you may have. Moving to "LINE IN" will, of course, force you to re-set the input controls if you have been using a Microphone as an Input.

Only one Echo system is shown. Since most recording is done "dry" or without Echo, one chamber should be sufficient.

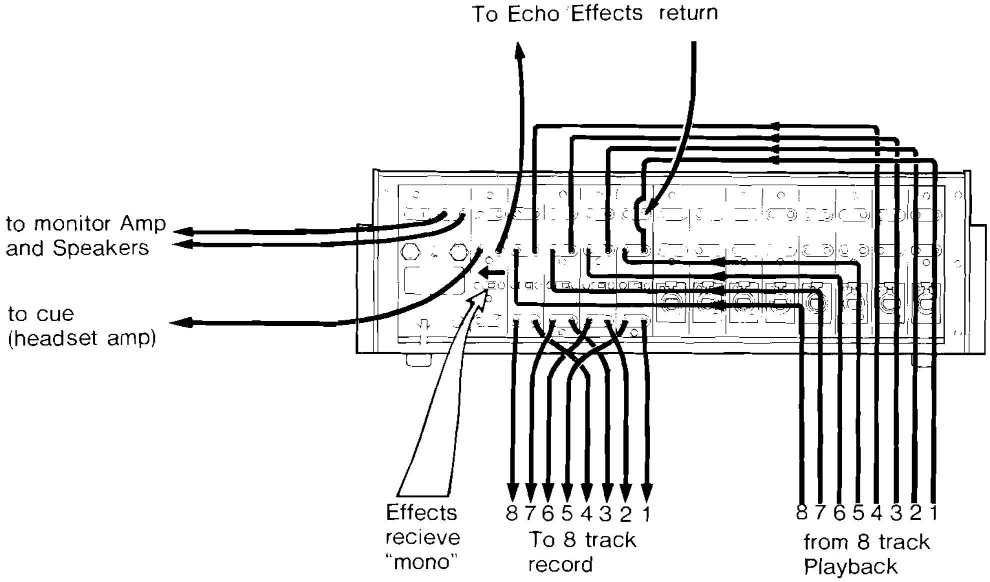
In all patching and connecting of 2-wire single ended, circuits two basic rules are worth remembering:

1. Keep your cable runs SHORT! – as short as possible. Installing a patchbay behind the engineer will require at least 20-foot runs and is not recommended. To the left or right side will allow much shorter runs, and wisest of all is to use our PB-64 mounted on top of the Meter Bridge itself. This location will permit the shortest lengths of cable run, and will improve your sound. Incidentally, short runs cost less, also a benefit. TEAC low-loss cable is available and its low capacitance per foot and superior insulation has been designed with systems like this in mind. It is well worth its extra cost. The use of 3-conductor professional cable such as Belden 8451 should be avoided. Even though it is of excellent quality, it is not the right idea for 2-wire systems. If you are going to make up your own cables we would suggest our 500 ft. bulk rolls or cable such as Belden 8218. Solid core insulator, low capacity wire is what you need. Foam-filled 2-conductor is not recommended, as the center conductor will cut through most foam with time, the capacitance will go up, and eventually the cable will short circuit. Don't use it.

2. Multiple output connections 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 may cost you quality, unless you are sure that a multiple connection is well within safe limits. Use the section on impedance matching in this manual, abide by the limitations it covers, and you will get better results.

   Using a Y-cable to "sum" or join two outputs to one connector 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 mixer as well as on to the next device. Summing, or adding two signals together, requires that they be isolated, and simple joining of the hot leads will not work properly.

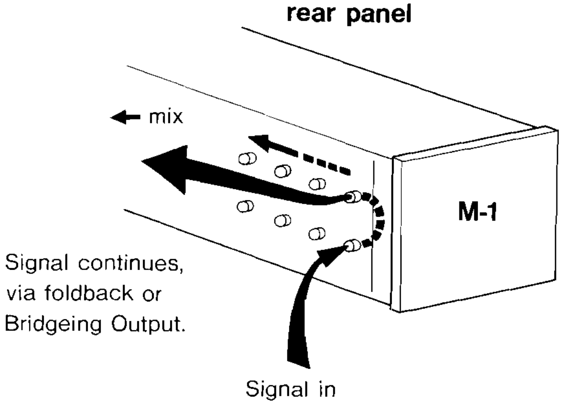
Recommended 8-Track Setup



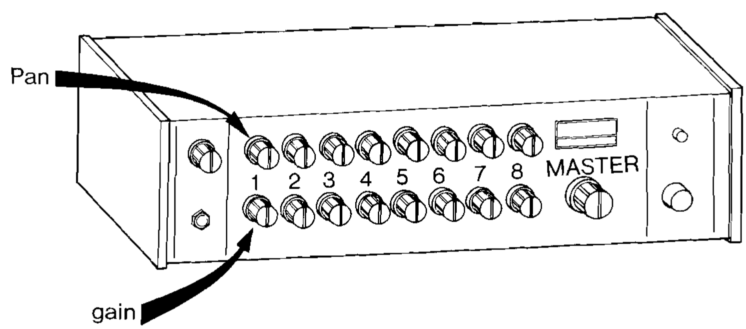
The M-35 has two basic subsystems for cue and effects but they are both "mono". If you need either another accessory system of a stereo version of pre- or post-fader signals, we offer an accessory line level device called the Model 1.

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 —

rear panel



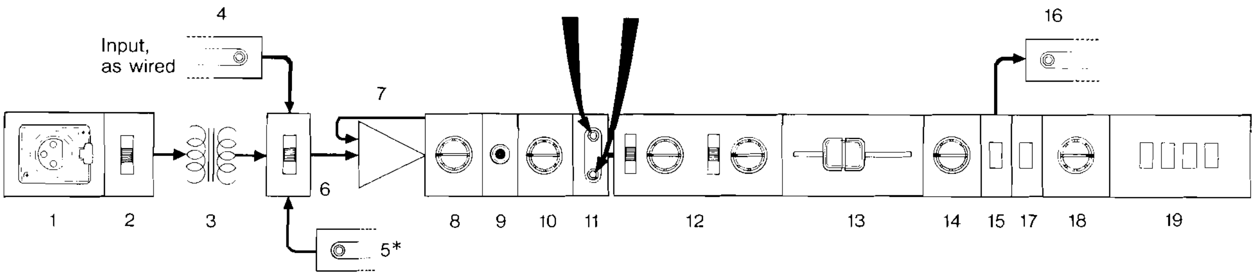
M-1 front panel



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,

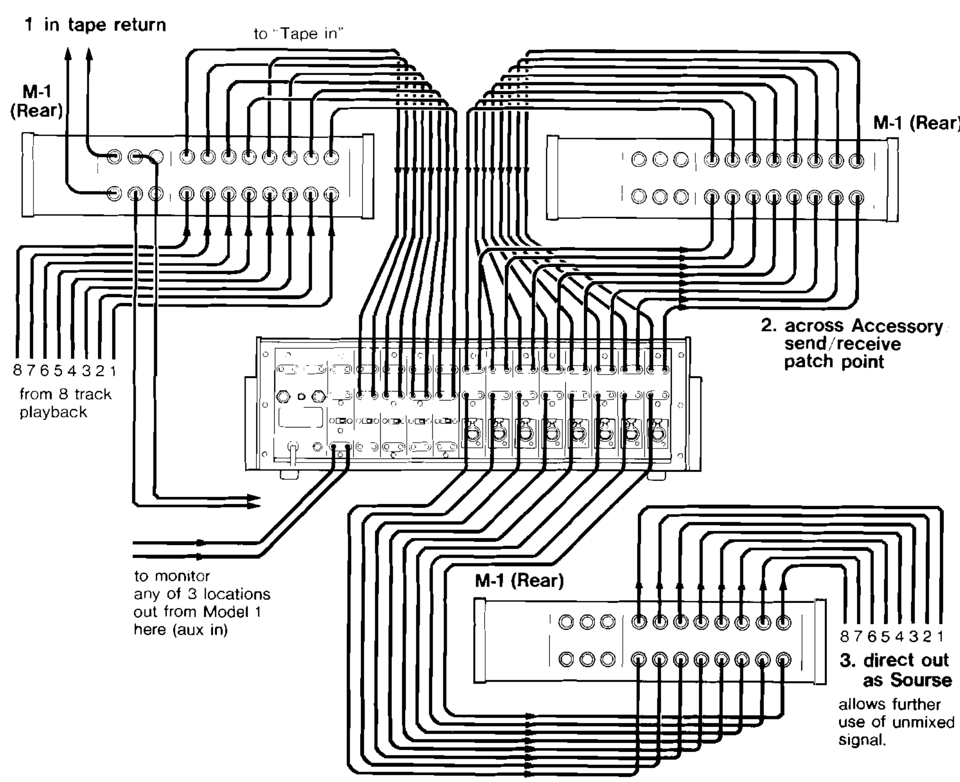
1. A "pass through" from an 8 track recorder, this "mix" can be used as a stereo cue, or an effects-send without EQ Since it is "pre" everything, it will stay on, and not be affected by any console control.

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 "mono" 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. Now your "mix" can be "muted" or "assigned" by using the direct-out button on the module and will also follow fader action. Two units may be cascaded as in the previous example if necessary.

3 Recommended Locations for Model 1

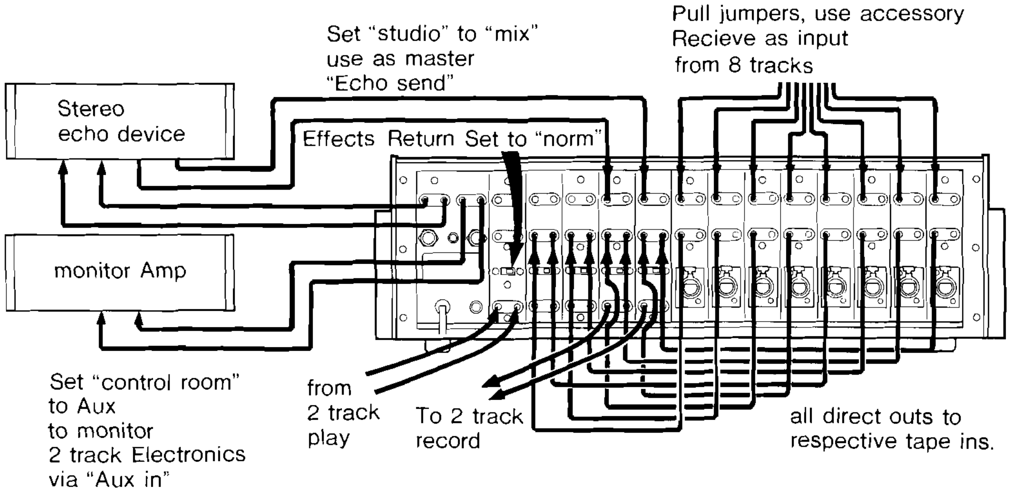


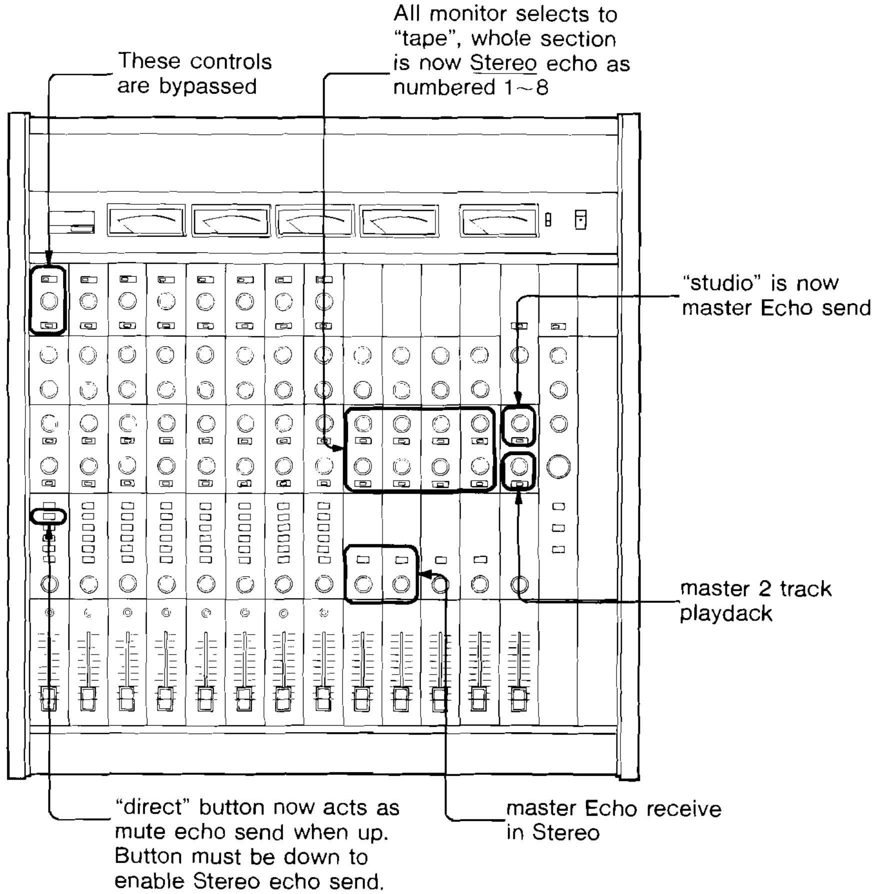
8-Track Mixdown

For our last patching example, we'll expalin a radical re-routing that can be used when your 8-track tape has been finished and you want to squeeze that last drop of performance out of your M-35 in final mixdown to stereo. First, we move the 8 track inputs from their standard location on the submaster "tape ins" to accessory receive. Pulling the jumpers will disable the mic preamp, the trim pot and the overload "light", but you don't need the mics and the circuits here will accept +10dB (3.2 volts) before overload, so it is safe.

Next, reroute all direct-outs to the tape-ins on the submasters. This will now allow you to use the entire monitor section for a stereo echo-send. For a master stereo-send, we'll use the "studio" monitor control group and select "mix" on the 3-way switch there. Echo return will be in the normal way – effects-receive 1 and 2, where you will assign the "dry" signal from the inputs. We now have only one problem left – we can't "hear" what we are doing and we need a monitor, so we use the electronics in the 2-track recorder. Route the playback signal to "aux in" and select "aux in" on the "control room" group. Now, switch from "tape" to source on your 2-track and your whole system is ready to go.

AN UNORTHODOX PATCH FOR REMIX WITH STEREO ECHO CAPABILITY



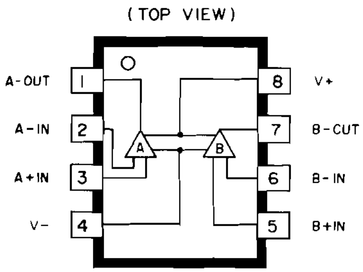


When unorthodox patches are used and the console top panel labels are no longer correct, we strongly recommend that you take the time to re-label each control to correspond to the new function that your re-patch has provided. Drafting tape applied to each group will prevent accidents from happening because you have tried to operate the M-35 "normally". It is also wise to lable 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.

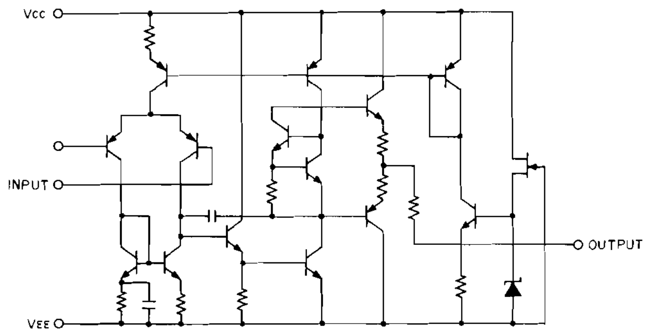
6. IC OPERATION
(PIN DISTRIBUTION AND LOGIC DIAGRAM)

NJH4559D Dual comparator

CONNECTION

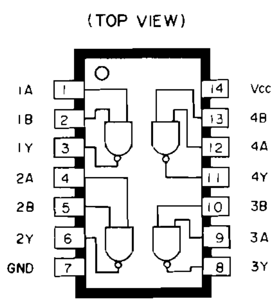


EQUIVALENT CIRCUIT (BASIC CIRCUIT)

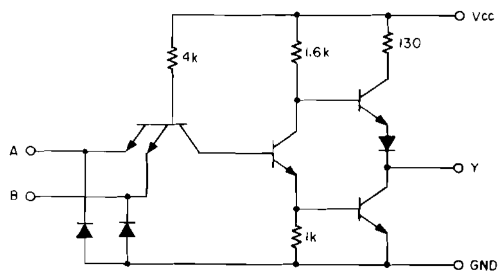


HD7400 Quad comparator

CONNECTION



EQUIVALENT CIRCUIT (BASIC CIRCUIT)



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