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INTRODUCTION TO THE MANUAL, AND HOW TO USE IT.

The Model 16 mixing console has been designed to satisfy the requirements of the modern multichannel recording process. The necessary auxiliary mixing systems are "built in" and can be assigned and re-routed to do more than one task. Complete and convenient multitrack operation with 8 or 16 track recorders 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 16 recognizes the fact that your signal processing needs may require a unique arrangement of subsystems. In order to cope with these changing needs, numerous patch points are provided throughout all signal pathways of the M-16.

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 beginner 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-16 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-16 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 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-16 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. In the following sections of this manual, we will do our best to describe the individual functions and controls of the M-16 and how they can be arranged in more than one sequence, but 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-16 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".

To begin, 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-16 to other equipment. Many aspects of scientific terminology will be discussed in the most basic terms we can use. 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.

Whenever possible, the scientific terms we use 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 nonlinear progressions to satisfy the ear’s 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 to 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 a low 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, 1.470 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 at 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 telephone 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 telephone 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 formulate 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:

\[ 10 \log_{10} \frac{P2}{P2} = N \text{ (dB)} \]

For voltage, current or pressure calculations:

\[ 20 \log_{10} \frac{V2}{V1} = N \text{ (dB)} \]

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 regretfully, 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 till 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 AERO 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-16 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 (bold 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 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".

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.

SOURCE (output) ——— plugs into ——— LOAD (input)

The classic procedure for measuring output impedance is to reduce the load's impedance until the output voltage drops 6dB (half the original power) and note what the load value is. In theory, you now have a load impedance 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 way:

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 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 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.

<table>
<thead>
<tr>
<th>Connection</th>
<th>Impedance</th>
</tr>
</thead>
<tbody>
<tr>
<td>PGM Buss Output/AUX Outpur</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>Stereo Master A Output/</td>
<td></td>
</tr>
<tr>
<td>Stereo Master B Output</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>AUX A Output/AUX B Output</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>Monitor A Output/</td>
<td></td>
</tr>
<tr>
<td>Monitor B Output</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>EQ Out/Direct Output</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>AUX 3, 4 Access Send Output</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>Access Send Out</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>(Input/Buss Master)</td>
<td>3.5 kohms</td>
</tr>
<tr>
<td>Talkback Output</td>
<td>3.5 kohms</td>
</tr>
</tbody>
</table>

**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-16:

<table>
<thead>
<tr>
<th>Connection</th>
<th>Impedance</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic Input</td>
<td>600 ohms, (balanced, XLR Type)</td>
</tr>
<tr>
<td>Line Input</td>
<td>50 kohms</td>
</tr>
<tr>
<td>AUX 3, 4 Access Receive Input</td>
<td>50 kohms</td>
</tr>
<tr>
<td>Effects Receive Input</td>
<td>20 kohms</td>
</tr>
<tr>
<td>Monitor Buss Input A/</td>
<td></td>
</tr>
<tr>
<td>Monitor Buss Input B</td>
<td>68 kohms</td>
</tr>
<tr>
<td>Tape Input</td>
<td>20 kohms</td>
</tr>
<tr>
<td>Buss IN Input</td>
<td>20 kohms</td>
</tr>
<tr>
<td>Access Receive Input</td>
<td></td>
</tr>
<tr>
<td>(Input/Buss Master)</td>
<td>68 ohms</td>
</tr>
<tr>
<td>Line A Input/Line B Input</td>
<td>100 kohms</td>
</tr>
</tbody>
</table>

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 then 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 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 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-16, 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 −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-16 — 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 6 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 this chart 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 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 + 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-16 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 gives -60 dB (1mV) 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-16. 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 cm² 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 (0.3 volt) so, for 94 dB spl, a mic with a reference output of -60 dB will need 50 dB of amplification from your M-16 or recorder in order to see "OVU" (-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-16 to make the needles on your recorder read "OVU".

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

THE BLOCK DIAGRAM AND GAIN BLOCK DIAGRAM

Before you begin reading the next section of this manual, flip out the extra fold on page 60, 61. 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-
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 24 modules are identical, and can be interchanged without modification, but the spaces in the chassis 17-24 do not have a “REMIX” line provided on the internal wiring. There is no “TAPE IN” on these last eight positions in the chassis.

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. Obviously, there is no real need to duplicate functions that can use the same parts at a later time 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 TAPE (remix) 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-16 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 stick to the wiring sequence, and describe the “options” as we come to them on a “once through” basis, and leave the practical decisions about a specific “working method” to you, the operator. Route “confusion” is an unavoidable side effect of flexibility, but a few weeks of operation will make you more familiar with the M-16 and it is not necessary to absorb ALL the possible routings in the first ten minutes of operation. Take your time, this is a Large and extremely flexible console.

MIC IN XLR

PHASE Reverse Switch
Since the MIC IN circuit is balanced at this point, it is possible to invert the “phase” or polarity of the incoming signal. Set NORM, the phase is unaltered, set REV, the phase is reversed. This function is not provided on the LINE or TAPE inputs.

MIC ATT Switch
Two positions are provided. Set 0, there is no effect, set 20, a “loss” of 20 dB is inserted to protect the transformer from excessive input signal.

INPUT Transformer
Maximum signal allowable without the use of the MIC ATT is 0 dB (1V). With the MIC ATT, maximum input rises to +20 dB (10V). This XLR connector, phase reverse, “pad” and transformer are the only “three wire” circuits in the M-16.

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, buzz, crackles and switching noise when motors start up (do you have an air conditioner on your A.C. 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 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, and you leave the “junk” behind and get only the mic signal.

**INPUT Select Switch**

This switch has three positions. The diagram shows a dotted line connecting three areas of the drawing and three representations of the switch. This imaginary “split” (there is only one actual control) is needed to include the LINE INPUT TRIM control, which is the lower section of the first rotary dual pot at the top of the module.

The switch positions are as follows:

- Set Leftwards (MIC)
- Set Center (LINE)
- Set Rightwards (REMX)

Selects the MIC IN.
Selects the LINE IN connector on the back of the module.
Selects the TAPE IN connector on the MASTER Module.

Each of the input modules selects one of the TAPE IN signals, the one that corresponds to the input module number. Since there are two TAPE INS on each MASTER Module, here is a chart showing which “Jack” leads to which Input module.
Right here we have our first major problem in comprehension. The TAPE IN connection and its circuit are plainly drawn on the block diagram, but what does it provide in terms of function? 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 or 16 track system “in use” and to explain, we’ll have to show the system in its entirety, even though we have not covered the first complete path to the recorder.

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

1. Simple playback to judge a 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 playback signals with “new” mic signals so musicians may hear a balance of both when “overdubbing”.
3. 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 with one tape playback cable, so, to avoid resetting all the controls on the input module, and “loosing” the EQ and record level settings that have taken much time to get “just right” every time you change from record to play, you use the TAPE INs on the MASTER Module instead of the LINE INs on the INPUT Modules. From this single connection, the playback signal can now be made available to the necessary subsystems.

A perspective drawing may help you to visualize the routing. We show one master module only so the wiring can be seen clearly.

All three “routes” can be active at the same time. The block diagram shows all three systems. Since the size of the page in this manual forces us to reduce the block diagram a lot, we suggest that you use this “block” as an “index” in order to find the area we are discussing on the large “Insert” version. Its large size will be easier to read.
Now that we have the signal routed to the right places, we'll keep going and use some more drawings to show you the rest of the signal pathway for each "job".

1. SIMPLE RECORD CHECK
Move the three position MONITOR SELECT switch to the rightmost position marked "TAPE". The monitor mix will now be derived from TAPE IN instead of BUSS OUT on each master module so switched. To finish this "route"; select either the STEREO MASTER A or B module, and depress the MON A, MON B push switches on the signal select group and the straight line fader at the base of the selected module will now give you final control of your "mix".

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 eight, let's just listen to track eight, and we won't have to remember under pressure how we "got there". With this monitor method, ANY line level source that carries a signal can be considered as a "feed" to a track, even if it has no monitor capability on the way "out" of the console, because 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 "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 playback for the musicians, because you are ALREADY listening to "playback". 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 when you rewind and play the tape. We recommend this monitor method highly, it saves much time and eliminates confusion.
The TAPE CUE gain and pan controls AUX 3 (X 8) and AUX 4 (X 8) on the master modules can be combined with the AUX 1 or AUX 2 gain and pan controls (X 24) on the INPUT modules to form a mixing system that will allow individual control of any signal needed for an overdub. Because there are separate controls for mics and tracks, even when you are working on a tape that has 15 tracks recorded, you can still “cue mix” the mics individually. On the M-16, “tape playback” does not progressively “eat up” your cue system flexibility. To get this “mix” out of the console to the cue system power amp, some additional control decisions will be required.
Layout for 16 Track

1. Depress the AUX 3 and AUX 4 push switches on the AUX A or AUX B master module, whichever one you have decided to use as a cue system master. This action adds the tape cue signals to the input module signals on the AUX master you select. There is also a "balance control" on each AUX master that allows adjustment between "all tape signal" (rotate fully rightwards) and "all input module signal" (rotate fully leftwards). When this "balance" control is centered, both "sides" of your cue system feed the AUX Master equally.

2. Advance the two "over-under" slide faders on the selected module, and the level of the resulting "mix" will be displayed on the meters (9-10 for AUX A, 11-12 for AUX B).

3. If you are using true stereo and want single fader control of your "mix", the lower fader (marked "R") can be converted to "Stereo" by depressing the push switch immediately above it. Moving the upper fader (marked "L") will now have no effect. To return to individual fader control of each "side" press the switch again.

4. To audition the sound of the "mix" in the control room, depress the appropriate signal select push switch (AUX A, AUX B) on the STEREO MASTER module that is being used as the control room master. In setting the level for the players, caution is advised, the control room volume set by the stereo master will not relate directly to the headphone volume separately set by the AUX master, so take care.

5. We suggest the use of AUX A if a single cue system is what you plan on, because the AUX A can receive any or all of the 8 EFFECTS RCV signals. Use AUX B as an effects send master and you will avoid the possibility of circular assignment (feedback) as it has no EFFECTS RCV capability built in. To add such a signal to AUX B, you must use an outboard patch.

6. Another limitation. ONE 16 track tape playback stereo "mix" requires that both the AUX 3 (top row) and AUX 4 (bottom row) controls be assigned to an AUX MASTER section. You CAN assign the two tape playback groups to both AUX sections at the same time, and thus get two separate stereo mixes from the input side, but the one 16 track "mix" from tape will have to do for both AUX systems. When using an 8 track recorder, the fact that you can assign and monitor each "row" separately ("8 track" is why there are two push switches required to enable for each function) allows extra "mixes" for cue and monitor and 8 track split AUX system and monitor operation with separate levels will be possible. For a complete 8 track working patch, see page 54.
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 1-16. The last 8 modules (17-24) accept line level signals only through their LINE IN connectors, and the large block diagram has a separate section for the modules 17-24 just to show this change. There are no other differences in module function, and this one alteration is caused by wiring differences in the MAINFRAME, not the module itself. If you move a module to a 1-16 mainframe location, it WILL accept the appropriate TAPE IN signal. The “logic” follows the chassis, not the module.

Selecting REMIX on any or all input modules will not disable the functions of CUE and MONITOR that we have discussed in the two prior sections. TAPE IN signal will be available on all three circuits at the same time. There are several advantages that this multiple feed offers that are added to the necessary CUE and MONITOR functions. In remix, the CUE function is not needed, so that Stereo “mix” can be used instead as an extra effects send. If a “machine electronics” monitor method is used on the 2-track mixdown machine, the MONITOR circuits can be used as yet another “effects” system, for a total of four accessory 16 input “spare stereo groups”.

During the course of normal multitrack pro-
duction, a good "Take" may be acceptable in every way except one; some doubt may arise to the “mixability” of one track. Since the "remix" function may be selected one input module at a time, a single track may be routed through the EQ section and a correction tried out to make sure that re-recording is not required. This checkout will only require the readjustment of one module. Many other consoles force you to switch the whole system to "remix" just to check one channel. The maximum permissible input level to the TAPE IN jacks is +18 dB (7.9 V).

A caution.

Although the REMIX function of this jack can be "trimmed" with the LINE IN trim control, there is no available "trim" for the monitor or cue functions, so observe the maximums.

TAPE IN input impedance = 56k ohms. LINE IN (on the input module) is 20k ohms.

LINE IN Jack
This RCA jack is the one located on the rear of each input module. In “normal” operation, this input can be considered as a “Spare”. It is a TRIM control that is shared with the REMIX function.

LINE IN TRIM (lower section)
Maximum input level with the LINE trim rotated fully leftwards is +28 dB (25 V). Maximum input level with the trim rotated fully rightwards is −22 dB (0.08 V).

MIC TRIM (upper section)
This control will alter the gain of the first amplifier in the console. It will only affect the level of mic signal. Maximum input signal to this first gain stage is 0 dB (1 V) at minimum gain (full left setting). Minimum input signal is −70 dB (0.0003 V) at maximum gain (full right setting). Remember that these overload figures refer to the amp input and the "Pad" (for MIC) and the prior LINE IN trim (for line and remix) will increase the maximum permissible input if they are used. On MIC signal, with pad and full trim the max is +20 dB (10 V).

This TRIM controls the gain of the amplifier by adjusting the amount of output signal returned to a secondary input "Port" (FEED-BACK CONTROL).

Because of this feedback control, we consider it unwise to adjust this 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 faders ONLY and adjust this special TRIM during the rehearsals.

Overload LED
When signals high enough to make the ACCESS SEND jack output exceed +15 dB (5.6 V) are applied to an input module, this LED will light. Adjust the appropriate gain reducing control (TRIM, or MIC pad) until this LED remains out when signal is present. When working with extremely percussive transient material, it may require full negative TRIM and PAD to prevent this LED from flashing 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 our first “patch point” in this useful location. The level at this “send” is −10 dB (0.3 V), nominal. The minimum load impedance is 5k ohms. A limiter connected to this point in the M-16 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-16. 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 switching and the overload indicator. This unorthodox “patch” is suggested for final “remix” when 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 +18 dB (7.9 V). The input impedance of the RCV jack is 100 k ohms. A complete “layout” for this type of remix patch can be found on page 55.

Four 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 four graphs showing the control ranges of each of the four 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".
Filters
The M-16 EQ section also provides a high-pass filter with a choice of two settings and slopes and a low-pass filter at 15 kHz. This chart shows the effect of the three possible selections.

**EQ IN-OUT switch**
This switch is provided to enable the entire EQ section with a single control. A setting can be compared to “flat” by alternately switching in and out, or, the setting can be bypassed until needed and “dropped in” with a single action. The filter function is not included in the action of this control. UP; the equalizer is bypassed. DOWN; the equalizer is engaged. 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 and then the cut or boost you use will be more effective. You get the result needed with less rotation of the control, and 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 mean 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 Buss, 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 (13 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 lower the TRIM to avoid overloading subsequent sections of the console when using the extreme boost settings of the EQ.

After the signal has passed this EQ section, the block diagram shows FIVE possible pathways for our signal that may be used individually, or all at once. Here there are many options so we’ll take some time to explain why each route has been included.

1. EQ OUT jack
   This output is provided for those who need to expand the “spare mix” aspect of the M-16, route this output to a separately controllable mixer, and you have another independently controllable mix for cue or effects that is post equalizer, pre-fader. When using a limiter, many engineers like to EQ the low end before limiting, to help avoid excessive “pumping” of the signal. If this is what you want to do, and you have another module free, do this; take the EQ out from the first module, go to the limiter (use module 1 for your pre-EQ), from the limiter out, go to the second module ACC RCV jack. Now set the limiter input level with #1 module TRIM (not the fader, you have bypassed it) run your final signal with the #2 module fader, and you will have EQ both before and after the limiter with the minimum of electronic stages. This “Patch” without the limiter will also give you “double EQ” with the smallest possible electronic package for those stubborn processing jobs that only brute force will fix.

2. PFL (pre-fader listen)
   Electronically, our first SOLO function. In Radio and P.A., there are many instances when it is desirable to check a signal BEFORE opening the fader and committing the signal to broadcast or a “house feed”. Is the mic working? Do you wish to talk to an announcer before going “on the air” or do you need to ask (and hear the answer) a vocal group about the cue balance while doing a background voat? Use this PFL function. When this push switch is depressed, the pre-fader signal goes directly to the SOLO circuits, replacing the monitor signal on the stereo master modules that have their SOLO/PFL push switch and LED active. The signal will be heard “center mono” and more than one PFL may be depressed at a time. Push to enable, push again to release, the switches latch to make a “mix”.

3. INPUT FADER
   The main mixing control for individual signals on the M-16. 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 “TRIM” or actual gain adjustment. In the M-16, only two of the many amplifiers used actually has “trim”, the first stage in the input module, and the buss combining amp (summing amp) on each master module. When you combining amp (summing amp) on each master module. When you advance any straight line fader on the M-16 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.

4. AUX 1 W-Pot PRE-Fader function
5. AUX 2 W-Pot PRE-Fader function
   The M-16 has two accessory mixing systems, each with its own master and meters. Either system may be used for CUE, EFFECTS, or some other secondary function, but there are some differences that favor one AUX system over the other when a specific task is considered. Most of these differences relate to special signal access capabilities further up the chain, and we'll deal with these “extras” when we discuss the AUX 1 and 2 master modules, or when we encounter an “option” or signal pickup point that expands the AUX function capability.

The first of these signal selecting “options” is the choice of PRE or POST fader on the input module (×24 for each AUX). The pre-fader “choice” is made by rotating the top AUX control LEFTWARDS (counterclockwise).
This will RAISE the level sent to the AUX system, this upper section of each AUX control is called a “W” pot and it works like this;

AUX 1 and 2 POST Fader Function
Rotated RIGHTWARDS; selects and adjusts level of the signal AFTER the INPUT FADER. This source is used when your AUX mix must “track” what you are doing with the main controls on the module. Echo send is a good example of such use, and this AUX source will now “fade out” the echo send as you “fade out” the main signal.

Selecting an AUX BUSS
The two accessory systems are identical in every way except one. AUX A has the capability of receiving the 8 effects return signals from the effects inputs on the master modules, AUX B does not receive these inputs. If you usually work without echo or effects signals in your cue mix, this limitation will not prevent you from having two CUES. Of course, the use of both AUX systems for CUE mixes will force you to work without effects unless you “Bridge” a device across a single accessory SEND-RCV point, or add another line level mixer such as our Model 1 to a group of discrete outputs in order to make up another “mix”. In 16 track operation the monitor section is completely “used up”, but if 8 track work is what you are doing, part of the monitor may be diverted to provide a separately controllable “mix” for effects. See page 46 for one solution to this type of problem.

Before we discuss the rest of the functions of the AUX systems, lets return to the main
signal path and cover the main line to the SUBMASTERS.

'UTE Push Switch

MUTE LED
Depressing this latching push switch will light the LED on the module. This MUTE does not kill the signal to the DIRECT OUT, and those signals sent to the POST of AUX 1 and AUX 2 (on the ASSIGN SWITCH GROUP, that selects submaster output) are cut off. This will allow you to “silence” a signal WITHOUT disturbing a critical fader setting. Since this MUTE does not cut off PRE AUX send, using mute in rehearsal will not disturb the talent. In multi-track REMIX this mute will not kill all the effects sends, just the POST’s. For full MUTE of the INPUT MODULE for all functions including AUX, switch the INPUT SELECT to an unused “LINE IN”. Even though it is not a console function, we should point out that all of our multitrack recorders can provide a “mute” if the record select switches are used, and most other brands offer a similar function. Check the owners manual for your multitrack recorder, and you may find yet another method of obtaining a “full program mute”.

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 purposes. This jack has no assign switch, it is “hot” even when the MUTE is depressed.

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 bypass unneeded summing networks and amplifier
stages. Going "direct" to the recorder will result in a cleaner signal. The use of a PB-64 patch bay will allow quick selection of a BUSS or a DIRECT assignment, and you then "monitor" by using the "input monitor logic" described on page 22. Since the meters on the M-16 can be switched to read a subsidiary group of signals, you can make the DIRECT you are using appear on the meter bridge even if it is not assigned to a BUSS. In the illustration below we show the PB-64, M-16, 85-16 system with one DIRECT only so you can see the "route" clearly. The actual patch bay layout can be varied to suit your specific requirements, but remember to keep your cable "runs" as short as possible. Less cable means less loss, less problems with hum and buzz and less money to "wire up". Since the amplifier that feeds the DIRECT out jack also feeds the BUSS ASSIGN switches and the post fader AUX lines it is wise to calculate the total load carefully. The output impedance of this jack is 100 ohms. The pre-wired circuits restrict the connection to a single circuit at a time, unless you are certain that a "multiple" is within safe limits. Use the formulas for impedance calculations found on page 13 to be sure.

The AUX 1 and 2 PAN Pots
This LOWER section on each AUX system dual concentric control is the final point on the input module for AUX adjustment. Each "pan" is permanently assigned to a stereo pair of internal busses that lead to the master module that will "sum" the signals and allow final control of the AUX mix.

AUX 1 pan: BUSS 9-10, AUX A.
AUX 1 pan: BUSS 11-12, AUX B.
Note: The BUSS numbers correspond to those used in the block diagram.

A caution; other controls on the master for each AUX must be set before a mix will pass out of the module. In addition to the regular AUX controls, there is a setting of the SOLO system logic that will mute the AUX system and cause confusion if it is not understood so check page 38 (solo) and page 48 (aux A, aux B masters) before you attempt an AUX assignment.

Channel Assign and Main PAN Pot
The channel assign switches and the pan control together make up the last section of the input module control. At this point you have selected a signal, trimmed or padded it to avoid overloading the input stage, checked it with
the PFL function, passed it through the EQ and filter section, it has appeared at the EQ and direct out jacks, AUX system levels and stereo routes have been covered for effects, cue, the mute button and the input fader have been reviewed. What is left? The stereo solo and this channel assign/pan pot group. Since the stereo solo is affected by the main pan we'll discuss signal assignment first. What SUBMASTER MODULE do you want your signal to go to? The four button rack of latching push switches will assign to either submasters 1-4 or 5-8 depending on whether the lowest switch is up (1-4 will be enabled and the green LED will light) or latched down (5-8 will be enabled and the red LED will light). A common practice in console design uses assignment switches wired in “pairs”, and then controls “single buss” assignment with a pan pot. While this seems convenient it does not allow you to bypass the pan when it is not needed, and crosstalk in the companion buss may be severely increased. TASCAM has used this matrix assign switch in every four buss and larger console we have ever built. It is MORE expensive, but it will allow single assignment to bypass the pan, and you can also separate your stereo pairs (1-4 or 1-3 will work) and “odd number” assignments (2-3) will not force you to “think backwards” because our pan-pot logic always works LOWER NUMBER (left rotation), HIGHER NUMBER (right rotation). Depressing more than one assign switch automatically engages the pan pot.

This matrix also permits 3 or 4 buttons to be down at the same time, the logic of these multiple arrays is explained simply with some drawings. If the shaded buttons are assumed to be down the logic is:

The limitations of this “BUSS OVER” style of matrix pan operation are that you must confine your panning to busses within a group. You can’t pan between 1 and 7 or 8, 4 and 5 because these numbers are not all in the same matrix.

The pan pot affects the SOLO system on the M-16 even if a single buss is all you have selected. Here’s the reason.

Stereo SOLO System — INPUT Module Source
The M-16 is provided with an extremely large and comprehensive SOLO system. If the PFL function is counted as well as the solo switches (it will “solo” even when “mute” is engaged, and you lose the stereo solo because signal no longer gets to the pan) there are 24 PFL, (mono solo) and 24 Input module solos. To get a source and a control for stereo, these signals are drawn off after the pan pot and they “track” the pan pot position whether or not it is being used to feed a multiple buss assign.

There are also 16 more stereo solo switches in the MONITOR section and 8 effects return stereo solo switches, for a total of 72 possible points that can be selected for a “I must check this RIGHT NOW” instantaneous isolation in the monitor. All solo switches are the “push to latch, push to release” type, so a mix can be made by depressing more than one, but this flexibility leads to the need for several cautions. This system can get you into trouble if you don’t observe the rules. Once again, we’ll have to jump ahead and get out of wiring sequence to show the SOLO system laid out properly.

SOLO System Operation — Additional Controls
1. Since SOLO is a production aid, not a recordable output, its action must be applied to the studio monitor control. On the M-16 either of the two STEREO MASTER modules can be considered for this purpose, so there is a “solo enable” push switch and LED on each module. Depress the solo enable switch on the module that you have selected to run your monitor chain. The LED will light.

2. Since SOLO and PFL levels may differ radically from the level of your monitor mix, a separate SOLO MASTER is provided on the STEREO MASTER A module. You must have some level set on this SOLO master or depressing a SOLO or PFL will only result in silence. Closing the stereo master
fader “silences” the SOLO/PFL levels.

3. ANY SOLO THAT IS DEPRESSED WILL MUTE THE OTHER PARTS OF THE MONITOR CHAIN EVEN IF THE SOLO OR PFL ENGAGED IS ON AN INACTIVE CIRCUIT!

   This can really cause confusion. If you have the M-16 obviously in a running condition, but you can’t get anything to “work” in the monitor, check the LARGE red SOLO indicator that is directly over the SOLO master volume. If any SOLO or PFL push switch is latched down, this light will be on. There is another LARGE red warning light that relates to the operation of the SOLO system. This second indicator is “in line” with the “solo” indicator, on the adjacent module over a rotary switch marked NORM-REMIX.

4. WHEN THE NORM-REMIX SWITCH IS SET TO REMIX, THE RED LIGHT OVER IT COMES ON. NOW, DEPRESSING A SOLO SWITCH WILL DISCONNECT ALL THE AUX 1 AND 2 FUNCTIONS OF GAIN AND PAN ON ANY INPUT MODULE NOT SOLOED! YOU WILL LOSE UP TO 42 SIGNALS! Obviously, we did this for good reason. We have no wish to destroy your session, and this action will not happen. If you set the NORM-REMIX switch to NORM, the warning light will then go out, and when you depress a solo, the cue mixes that you have made on AUX 1 and 2 will not be affected.

   If you set the NORM-REMIX switch to NORM, the warning light will then go out, and when you depress a solo, the cue mixes that you have made on AUX 1 and 2 will not be affected.

   **Why Is This REMIX/SOLO Function Provided?**

   Coupling the two words together is almost enough to indicate the purpose intended. Since the M-16 has only two AUX mixing positions on each module, they must serve all the functions necessary to the multitrack process. Two of the uses of an AUX mix have conflicting SOLO requirements, so, instead of putting in separate systems, we altered the AUX system response to the SOLO function.

   A. When you are working with talent in the studio on partially completed tapes, the SOLO should not interfere with the mixes made for cue or effects. USE “NORM”.

   B. When your tape is complete and you are using the SOLO function to help determine subtle things such as echo return balances, it would be a big help to have ONLY THE SIGNAL THAT YOU HAVE “SOLOED” activating the effects send mix. The switchable logic that can be seen on the block diagram attached to the lines leading to the AUX PAN functions performs the job of disabling all sends THAT ARE NOT “SOLOED”. Now, when you add a SOLO EFFECTS RETURN from one of the 8 effects solos on the master modules to the SOLO from your selected INPUT Modules, you will be able to hear a single signal or a selected group WITH EFFECTS RETURN without having ALL of the unwanted signals driving your effects device and making subtle decisions difficult. To get this logic logic for SOLO, switch the special control from NORM to REMIX.

   This completes the functions found on or related to the INPUT Module and we can now proceed to the submaster module.
SUBMASTER MODULE

BUSS IN (submaster in) Jack
This jack on the submaster module could be described as an input module with no controls at all. Electrically, it resembles the final summing point found on each input module, and is provided as a way to add another signal, or group of signals that have their own control functions to the submaster on the M-16. To "stack" two mixers, use this input and the first mixer will then have final master control over the second. Can't find this jack on our block? Run your eye down the buss lines to the bottom of the page and you will find the eight RCA symbols at the end of the lines, after they turn to the left. The input impedance of this jack is 22 k ohms. The "OVU" setting of your outboard device should be adjusted to -10 dB (0.3 V). Maximum input before clipping is +18 dB (7.9 V).

Echo Receive to PGM BUSS
Now that we have your eye directed to the lower part of our block diagram, we'll discuss part of the effects receive function. One of the many possible assignments is to the summing network, so effects can be recorded. There is one effects control group for every "submaster," and several signal routings are provided. When depressed, this top push switch causes signal to enter the submaster along with all the other signals from the modules and the level is then affected by the settings of the trim pot and straight line faders on the master. Since the other functions of this effects receive "group" enter the monitor at other points in the chain, the sound you hear will be different for each point. Selecting more than one effects receive point at a time is possible, but we caution you to make sure that you are not making a "quantity" judgement that does not relate to what you are actually recording. Several combinations of this assign group will add the "effect" to the monitor TWICE, making you think you have more "effect" than is in fact the case. Take care.

EFFECTS RCV Jack
EFFECTS RCV Pot
This RCA jack directs signal to the effects control group, and is provided with a "foldback" companion RCA jack. This jack is used to route a signal to more than one input in "daisy chain" fashion, so that the same effect n be assigned to more than one submaster at a time.

A CAUTION — TWO INPUTS MAY NOT BE MIXED INTO THIS CIRCUIT BY USING BOTH JACKS!
There are not many studios that have 8 echo chambers, so this input on each module may also be considered as a "spare" line in to the buss, the monitor groups or the AUX A buss for whatever purpose you may have other than the one we have described here. This input is provided with its own "solo" button (center feed mono, similar to PFL on the input module) and a pan pot that feeds stereo to the AUX A system. Nominal input here is -10 dB (0.3 V). Max input before clipping is +18 dB (7.9 V). The input impedance of ONE effects receive is 20 k ohms. If you use the "foldback" the total "load" on your effects device will be determined by the length of your "daisy chain". Be sure that the effects unit that you use can feed all of the connections without troubles.

Summing Amp, BUSS Overload LED and TRIM
This amplifier makes up the losses caused by the summing networks that combine all the input module signals. Since any submaster may have anywhere from one to twenty four inputs assigned, the range of signal voltages can be very wide. A trim pot is provided to adjust the input sensitivity of this stage with a single control, so when your input fader settings cause the LED to light, you can "cool off" your mix with a single control instead of being forced to re-set all the input faders. The LED is set to light at +25 dB.
BUSS ACCESS SEND-RCV Jack
This “patch point” is provided to add an accessory to an entire “group” of signals before the final level is set by the submaster fader. A limiter patched in here will not have its action affected by the submaster fader setting. This patch point must have its jumpers or a device “in line” or no signal will flow. There is no internal “normal”. The output impedance of the send jack is 100 ohms. Nominal output level is −10 dB (0.3 V). Max output before clipping is +18 dB (7.9 V).
The input impedance of the RCV jack is 10k ohms. Nominal inputs is −10dB, 0.3volts. Max input is +18 dB (7.9 V).

SUBMASTER Fader
Provides final control of your submaster mix. As you can see on our block diagram, there is not much left on the line that leads to the PGM BUSS dual RCA back panel connectors.
The parts remaining are:
1. A buffer amp right after the fader.
2. The PGM BUSS (SUBMASTER) signal feed line to the monitor. We have discussed some of the MONITOR functions in the course of the INPUT module description, but there are many details remaining. Let’s pass this point for now, and return to it when we finish the main line out.
3. Another buffer after the monitor feed.
4. The meter feed line.

5. The meter select switch.
This switch is mounted on the left hand side of the meter bridge. It allows you to meter any signal that you route to the meter bridge RCA patch point on the back of the bridge. The four push switches each control four meters.
PUSH SWITCH UP: Meter reads internal wiring from console.
PUSH SWITCH DOWN: Meter reads back panel (EXT IN) patch point.
Each meter bridge patch point is provided with a “foldback” or “pass through” type jack so that you can route a signal somewhere else after metering. Input impedance of this jack is 50k ohms. The level that will result in a “0VU” meter indication is internally adjustable (see page 81, service section). Initial factory settings are:
Meters 1 ~ 16, −10 dB (0.3 V) = 0VU
6. Overload LED Setting is 10 dB over the respective meter “0VU” point.

7. PGM BUSS DUAL RCA OUTPUT JACK (LINE OUT-AUX OUT).

These jacks are located on the back panel of each master module. Since each pair of jacks is connected to the same “amp”, any device (load) connected to one jack will affect the output performance of the other jack. To determine the true value for the load on the mixers final stage, the input impedance of BOTH devices must be included in your calculation, even if only ONE of them is in use. Some recorders and accessory electronics will present a “load” to the M-16 EVEN IF THEY ARE NOT SWITCHED ON. To remove the drain on the M-16, you must actually disconnect the signal cable (not the power cord). For this reason we suggest that you “unplug signal” to any device that you are not actually using when you make your most critical mixes. The output impedance of this circuit is 100 ohms. The nominal level is −10 dB (0.3 V). A switch provided inside of the module permits selection of −2.2 dB (0.775 V). Max output before clipping is +18 dB (7.9 V).

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 our discussion of the M-16 we have covered a complete signal pathway from the input module to the line outs. Patch points, optional sections like the EQ and the AUX systems, all are now either “in” or “out” and we are ready to record. We are done. The only problem remaining is — How do we hear what we are doing? The record ready signals must appear in the monitor. We have shown how this can happen “in brief” when we were working through the input module, but there are still many details and specific controls that have not been covered, so — we will return to the monitor system. In many ways, it can be considered as an entirely independent console in its own right and many features have been provided to offer extra speed and convenience to the working engineer. To make all the possibilities clear, we'll start on the monitor the same way we begin on the M-16 as a “whole” — with the input options. How many different signals can be supplied to this “console in miniature?”

The dotted lines surrounding the section marked SUB MASTER Modules 1-8 contain all the controls and switches on just ONE module. There are EIGHT DUPLICATES of this module, so you can see that we have a lot to go over before we get to the final output of the AUX MASTERS A and B, the STEREO MASTERS A and B and the final output logic of the SOLO system. To make the function
more understandable we’ll show the entire MONITOR top panel as it appears “on the outside”. Several aspects of Control use the entire section as a unit, and the whole top panel will help.

Signal Select Switch (BUSS-OFF-TAPE)
BUSS AS SOURCE FOR MONITOR
On each module there is an upper and a lower section of identical controls that begin electrically with these two switches, and in this leftmost position they seem to do the same thing. Why are there 16 Buss positions?
Even though the M-16 has only eight SUBMASTER modules, it has a 16 channel monitor system for TAPE playback. When you reset to TAPE (rightmost position) you will need 16 sections. When used as “buss master controls” they become redundant, but the two sections are not identical in tape mode. The top row (MON A) is connected to TAPE IN 1-8 and the bottom row (MON B) is connected to TAPE IN 9-16.
To assign both “rows” to a STEREO MASTER OUTPUT so you can run a pair of loudspeakers you must depress both MON A and MON B push switches on the STEREO MASTER Module you select. Why two push switches? Consider the requirements of 8 track operation. If two switches are provided, the “extra” monitor can be isolated from the STEREO MASTER Module you are using for the control room function (don’t depress the assign, leave it up) and used for another task. In 8 track operation, the use of the “foldback” on each of the TAPE IN jacks will allow you to have the same 8 signals available on both the upper and lower sections of the monitor. A possible use of this “extra” 8 input mix would be its use as a “send” to an echo for return to the monitor. A separate echo send from Tape but this “split” operation will allow one in 8 track operation.
When the buss function is what you are using, we remind you that the top effects push switch (PGM) will add its signal to the BUSS output at an earlier point not shown on the small part of the block printed here (PGM goes direct to BUSS, just like an input signal). You WILL be able to hear it in the monitor without depressing either of the next two, EFFECTS TO MON A, MON B. In fact, when the EFFECTS TO PGM switch is down, make sure the other two are up (out of circuit) or you will “hear” more effects than you are recording because you have added the signal to the monitor “twice” and to the BUSS LINE OUT, only once. This double sum will also happen if you monitor tape tracks instead of buss outs. The “monitor only” effects assigns are provided to experiment with, or, as extra “lines in” to the monitor. After you make your decision to record an effect, be sure you reduce to the single assignment (PGM) or you may be surprised when you actually “play back”, and the effect is reduced in strength.

TAPE AS SOURCE
If our basic 16 track setup on page 53 is used, you will have a different track on each section of the MONITOR. On module 1 you will have track 1 on the upper row, and track 9 on the lower row when the switch is set to the RIGHT. When the switch is CENTERED, only the EFFECTS TO MON signal can pass to the faders. Once again we call your attention to the fact that there is another complete system beginning to appear on our diagram, the AUX 3 and 5 groups. The primary use of these two 8 X 2 systems as CUE feeds makes it necessary to supply both the BUSS and TAPE signals to each AUX line as well. True independent use of MONITOR and AUX requires a separate signal select method for each task, so the BUSS-OFF-TAPE switch does NOT select signal for the AUX 3, 4 controls, which have their own “signal selecting” W pot. You may have “track logic” on the control room output, and “buss logic” on an AUX section if that’s what you need.

Monitor Fader
This straight line fader controls the level of the signal previously selected.

Monitor PAN
The Monitor Pan affects only the monitor mix. It will not alter the level or pan position on buss outputs that you have adjusted with the pan pots on the input modules. But if both busses (your stereo pair in fact) have their monitor pans set to the same side in your stereo monitor mix, you won’t hear any stereo effect when you rotate the INPUT pan. When more than one BUSS is selected on the INPUT assign, INPUT pan will always affect what you record, but to hear the effect that INPUT pan is producing, the two BUSSES that you have selected as a stereo pair must be
MONITOR “panned” to opposite sides in your monitor setup.

**JLO, Monitor Sections**
Each section of the monitor section on the M-16 has its own route into the SOLO system. The MONITOR PAN position will affect this stereo solo, so you will be able to adjust the stereo placement when listening to one signal. This solo is intended primarily to allow quick isolation of one or more TAPE signals when you are making a rough playback, and you don’t want to disrupt a module setting that will be needed when the session continues. MONITOR SOLO can also be used to positively confirm your BUSS ASSIGNMENTS quickly without the need to examine each module assign rack. Just switch to BUSS AS SOURCE, and then SOLO. In this way a 5 mic stereo submix can be clearly heard without disrupting the levels of other monitor sections, or switching the unwanted sections to the center-off mute, a tedious method.
The use of this MONITOR SOLO will also show the actual level of effects, if the effects return section has been assigned to the PGM BUSS. The effects SOLO need not be depressed to hear effects in this mode, in fact, reading so will constitute a double assignment and you will be misled. One last point in regards the SOLO. Remember that a STEREO MASTER must be selected and the SOLO MASTER pot advanced before you will hear anything. Take care.

**MON A and MON B Spare BUSS INs**
An extra STEREO RETURN WITH SEPARATE LEVEL CONTROL is provided for each row of the monitor. The RCA jacks and the pots appear on the adjoining AUX A and AUX B master modules. These “spares” are useful when you have a complex “patch” all worked out and one more input somehow must be found to the monitor, or you must have separate level control of a signal so you can retain a critical setting of a control that would normally allow adjustment. If the “normal” routing has been diverted to another function, this accessory circuit may solve the problem of control. The input impedance of these spare inputs to the monitor is 20 k ohms. Nominal level is -10 dB (0.3 V).
When the pot is rotated fully rightwards, there is a gain of 8 dB added to the level at the A jack.

**MON A L OUTPUT RCA.**
**MON A R OUTPUT RCA.**
**MON B L OUTPUT RCA.**
**MON B R OUTPUT RCA.**

These two stereo pairs serve as the complement to the inputs described above. When you wish to divert the monitor section to some non standard function, these outputs will give you access to the two stereo monitor groups BEFORE they go to the STEREO MASTER module switch assign rack. The output impedance of these circuits is 100 ohms. Nominal level is -10 dB (0.3 V). Max output before clipping is +18 dB (7.9 V). These outputs have no “master fader” and no meter.
AUX 3 and AUX 4 Circuits
Returning to the diagram, we have a top and bottom row of controls to allow either 8 or 16 track operation. This group is combined to form a 16 track TAPE CUE, along with the AUX 1 and 2 circuits on the INPUT module. When used with an 8 track recorder, the same “BUSS REDUNDANT” operation found in the MONITOR double rack becomes a virtue. If you can separate the two circuits, you can get two TAPE-BUSS cue mixes. The AUX A and AUX B MASTER modules have provision for selecting the AUX 3 and AUX 4 signal independently.

AUX Dual Concentric Control

The two sections of this control are shown on the block diagram as separate controls, but they are physically “stacked” one on top of the other. The LOWER section of this dual control is a PAN pot. The UPPER section has two functions.
1. Sets level
2. Selects which or the two signal sources the circuit will receive,
   - BUSS (rotate leftwards to raise the level of BUSS signal)
   - TAPE PLAYBACK (rotate rightwards to raise the level of TAPE PLAYBACK signal).

The use of a “W” pot eliminates a switch, but unless you look closely, you may assume that the switch provided for signal selection in the MONITOR will also determine the source of signal for the AUX 3, 4 sections as well, and that’s not so. The “W” pot rotation draws signal to the AUX independently of the MONITOR signal selection.

AUX 3 and AUX 4 ACCESS SEND-RCV PATCH POINT
Both stereo AUX 3 and 4 are provided with a non-normalled (jumers must be in place for signal to flow) patch point on their outputs. As an example of the use of this convenience “patch” we show the addition of 8 more signals to AUX 3 with the use of a Model 1. This extension will give you 16 signal capability for CUE, and if you duplicate this addition of a M-1 to AUX 4, you will have TWO FULL 16 track independent CUEs from tape. Other uses of this “patch point” will, no doubt occur to you as the need arises. The output impedance of the send is 100 ohms. The nominal level is −10 dB (0.3V). The max output before clipping is +18 dB (7.9V). The input impedance of the receive is 100k ohms. The nominal (expected) level is −10 dB (0.3V). Max input before clipping is +18 dB (7.9V). (Take care, there is no pad, or pot).
Full Double 16 Track CUE, with No EFFECTS

If each “group” of eight signals is FIRST passed through a model 1, the AUX patch point will now cause the LOWER (9-16) signals to appear in the upper (1-8) AUX mix, and the AUX 3 will now have all 16 signals instead of just 8... If this logic is inverted on AUX 4 (upper to lower) and another M-1 is used, each of the AUX systems will have ALL 16 signals, but each system will have separate control.

EFFECTS RCV Controls
This group of 8 inputs (one section per SUB-MASTER) routes signal to several different ints simultaneously, each route being controlled by a latching push switch, the level sent to ALL sections is set with the UPPER section of the dual concentric control. The lower section is a PAN POT that affects the action of the STEREO AUX A line only. It has no effect on any other STEREO circuit that the effects receive may feed.

1. EFFECTS SOLD (To Solo on STEREO MASTERS)
2. MON A.
3. MON B.
4. PGM.
5. AUX A.
On the back panel input you will find that each section has two RCA jacks, one marked EFFECTS RCV, the other marked FLB. ANY JACK MARKED “FLB” ON THE M-16 IS NOT A DUAL INPUT. IT IS A “BUILT IN” Y CONNECTION, AND IS PROVIDED TO CONTINUE A SIGNAL CHAIN TO ANOTHER INPUT. MIXING TWO SIGNALS INTO ONE EFFECTS RCV OR ONE TAPE IN BY USING BOTH JACKS WILL NOT BE POSSIBLE.

If difficulties arise when this style of input transfer is used, why provide one? As we have said on page 40, most studios don’t have 8 separate ECHO devices and when you want to assign echo or effects to a BUSS, you will have to re-patch. Here is an illustration of a “Chain” routing that connects one device to 4 inputs via “FLB” or foldback. If you do this you will have the ability to mix in the return without having to re-patch every time you change to another module, but take care, this multiple connection represents a more severe load on the OUTPUT stage of the accessory device than a single connection without FLB, each EFFECTS IN has an input impedance of 20k ohms. Nominal level is −10 dB (0.3 V). Max input level before clipping is +18 dB (7.9 V). Use the section on page 12 IMPEDANCE, to help determine the “length” of the chain that will be “safe” for the device you wish to use here. This concept of signal load calculation is so important that we have repeated the statement here as well in addition to the caution on page 40.
AUX A AND B MASTER MODULES

Since these two stereo modules are almost identical, and draw signal from the same sources, we'll save space by describing them together, even though they may be used for different tasks. The differences in function are small, even though the jobs the modules may be assigned to do will differ. It is possible to use either module as a CUE or EFFECTS master control, but AUX A has the ability to receive EFFECTS RETURN and thus is preferred as a CUE. AUX B has no EFFECTS RETURN, and its preferred use as an EFFECTS SEND will not run the risk of circular assignment (feedback).

INPUT Signal Sources to AUX A, B
The AUX A STEREO buss can receive signal from;
1. AUX 1 Input Module 24 Stereo Group
   This group of signals always appears on left rotation of the BALANCE CONTROL.
2. AUX 3 TAPE PLAYBACK 8 Stereo Group
   To have this signal on the right rotation of the BALANCE control, you must depress the AUX 3 push switch.
3. AUX 4 TAPE PLAYBACK 8 Stereo Group
   To have this signal on the right rotation of the BALANCE control, you must depress the AUX 4 push switch.
4. EFFECTS RCV to AUX A PAN POT 8 Signal Group
   First, depress the push switch on the submaster EFFECTS RCV section or sections that you want to add to the AUX A BUSS.
   Now the AUX A pan rotation will affect the send to either AUX A L or R. (The PAN is the bottom section of the dual concentric control, TOP section is EFFECTS RCV LEVEL.)
   Now adjust the level of the effects receive group as a whole with the EFFECTS to AUX A MASTER on the AUX A module.
5. Talkback to AUX A/B Momentary Push Switch
   The Talkback mic has a volume control for these circuits marked AUX A, B. The switch and pot are on the module with the mic. When you depress this momentary push switch, the signal from this mic enters BOTH circuits simultaneously AFTER the faders, center mono, so take care when you speak, and set this talkback control carefully. Talkback level in these circuits will not be affected by the faders.

The AUX B STEREO buss can receive signal from;
1. AUX 2 input module 24 stereo group
2. AUX 3 TAPE PLAYBACK 8 stereo group
3. AUX 4 TAPE PLAYBACK 8 stereo group
4. Talkback to AUX A/B momentary push switch

The BALANCE CONTROL on this module works in the same way as the one on the AUX A, but the options are;
AUX 1 (leftwards rotation), AUX 3, 4 (push switch first, then rightwards rotation)
This type of control action allows the re-balancing of a cue mix without the need to re-set each control individually. The singer says that the mix of the track is perfect but not loud enough? Just rotate the balance control a little to the right and raise the master. No need to reset all the AUX 3 and 4 controls and run the risk of losing that "perfect" balance.

AUX A and AUX B Master Faders
Both of these stereo circuits have their master faders arranged "Left OVER Right" to keep them on the same module, and thus lower the complexity of the wiring. If independent control is not required, depress the PUSH SWITCH above the MASTER L on either module and you will have control of BOTH faders switched to the lower fader for one fader action.
Now the MASTER R will work as a STEREO master, and the MASTER L will be “out of circuit” (no effect).
SOLO Logic Selector, NORM/REMX
This important switch has two positions;
1. NORM (LED OFF)
When set leftwards, and the LED is off, the solo system does not affect the action of the AUX 1 and AUX 2 circuits, when you depress a SOLO or a PFL button. Anything assigned to the AUX A or B MASTERS will remain "as set" so CUES or EFFECTS signals will not be interrupted.

2. REMIX (LED ON)
When set rightwards, and the LED lights up, depressing a solo will interrupt ALL AUX SENDS EXCEPT THE ONES ON SOLO'ED INPUT MODULES. This action will allow you to set EFFECTS sends and hear the results of such settings without interference from sends that you are not working on at the moment, a valuable function in the critical remix process. For more on this feature, see also page 21, Input Module section.

SOLO Master Volume Control
SOLO LED
Since there will be a large difference in level between one signal and an entire mix, a MASTER is provided to adjust the SOLO level in the monitor you select. Remember that this control and the STEREO MASTER will be the only level for PFL signals, as they are drawn off before the INPUT FADER.
The SOLO MASTER signal is added to the STEREO MASTER selected BEFORE the MASTER FADER. In order to hear SOLO, both the SOLO MASTER and the STEREO MASTER must have some setting or you will hear nothing.
When the SOLO LED is lit, it indicates that one or more solos are active. If you cannot get the monitor to work, check here first. There may be a solo engaged on a circuit that has no signal present.

AUX A and AUX B OUTPUT RCA Jacks
Output impedance of these circuits is 100 ohms. Nominal output level is –2.2dB(0.775V). Max output before clipping is +18dB (7.9V). These signals appear on the following meters.

- AUX A, L: 9
- AUX A, R: 10
- AUX B, L: 11
- AUX B, R: 12

Prior to the final output, but after all controls, a group of lines on the block diagram indicates that these AUX signals are routed to the switch racks on the STEREO MASTER modules for monitoring along with the outputs from the other subsystems on the M-16.
STEREO MASTER MODULES A AND B

These two modules are identical in every way, and can be used interchangeably for whatever purpose you have that requires a full function master control. All signal groups can be "SUM-MED" simultaneously in these modules, and any signal that is not wanted may be excluded. Here is the complete list of possible signal sources.

1. SOLO
   To receive, depress this push switch.

2. TALKBACK
   To "speak" into this circuit, depress. To "clear", press again, and the LED will go out to show that TALKBACK is not active on this module. The TALKBACK signal enters the modules AFTER the fader, and its volume in both masters is controlled with the small rotary control on the mic module. Take care in setting.

3. AUX A STEREO BUSS
   Top push switch in the "RACK".

4. AUX B STEREO BUSS
   Second push switch.

5. MON A STEREO
   Third push switch.

6. MON B STEREO
   Fourth push switch.

7. LINE A, L/R
   Fifth push switch.

8. LINE B, L/R
   Sixth push switch.

These last two stereo pairs are provided to route a stereo source into the modules for quick comparison. As an example a TWO TRACK mixdown machine may be connected to one set of these "spares" so that no complex switching will be required to hear playback of a mix, and to provide a "one button" source/tape comparison while mixing. FOUR RCA jacks appears on the back of the STEREO MASTER A module, and serves BOTH switch groups. The input impedance of these spare circuits is 100k ohms. Nominal input level is -10dB (0.3V). Max input before clipping is +18 dB (7.9V). There is no pot or pad, so take care.

ALL SWITCHES AND ENABLE BUTTONS MAY BE ENGAGED SIMULTANEOUSLY ON BOTH MODULES IF NECESSARY. Any combination that you require is OK.

STEREO MASTER Faders
All other modules except the AUX A, B (when stereo is selected) have straight line faders that are mono controls which adjust a single signal at a time. Each straight line fader on these master modules handles TWO signals in true STEREO operation, so resist the urge to grab both master faders when adjusting. Only one is needed, and if you are not "listening" to the other, but it is doing some other task, you will disrupt the "extra function" provided by the adjacent stereo master control.

Meters
Each section has its own stereo pair; STEREO MASTER A (Meters 13-14), STEREO MASTER B (Meters 15-16).

STEREO MASTER RCA Jack Outputs
Provide final output to studio power amp, or other full function purpose. Output impedance is 100 ohms. Nominal output level is -2.2 dB (0.775V). Max output before clipping is +18 dB (7.9V).
TALKBACK MODULE

This last module contains the talkback mic, the test tone oscillator and the switches and pots that enable these devices to reach the other circuits in the M-16.

TEST TONE Oscillator
Provides a source of signal for setup, continuity testing and tape recorder alignment. Six selectable frequencies are provided, 40Hz, 400Hz, 1kHz, 4kHz, 10kHz and 15kHz. In addition to the fixed -20dB (0.3V) outputs on the rear of the module, this oscillator can be assigned to all eight busses simultaneously by depressing the push switch labelled SLATE. The frequency set on the oscillator will then be applied to all busses along with the mic signal, the pot that adjusts the mic level does NOT adjust the slate level, so take care. When it is desirable to use this tone to “mark” tapes for fast search, use the SLATE position of the test tone switch. It provides a test tone of frequency more suited to this purpose.

Oscillator Control Switch
Set LEFT ........ Slate Tone
Set CENTER ...... OFF
Set RIGHT ......... Test Tone

MASTER A/B, Switch and Rotary Control
The mic signal can be assigned to the MASTER A/B module by depressing the push switch labelled MASTER A/B, and the level can be controlled by the MASTER A/B rotary control above.

AUX A/B, Switch and Rotary Control
The mic signal can be assigned to the AUX A/B module by depressing the push switch labelled AUX A/B, and the level can be controlled by the AUX A/B rotary control above.

HEADPHONES Select Switch
This switch determines the source of signal for the front panel headphone jack. The left position selects the output of STEREO MASTER A Module, the center position is off, and the right position selects the output of the STEREO MASTER B Module. The rotary control immediately below the switch provides a master for headphone jack.
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-16 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 sometime to become familiar with, but multichannel recording has many mixing requirements. A permanent patch will severely restrict flexibility. Don’t be afraid 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-16 to suit, you can get better results.

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

Recommended 16 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 tracks at one time, the fact that the M-35 has only buss master modules is not a serious limitation. Here we show each buss master connected to more than one track. Tracks 1 and 8 are on the same buss, #1 and so forth up to buss output, connected to tracks 9 and 16. 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 disrupt 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 500ft. 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.
In this 8 track setup, we show the tape playback lines 1-8 returning first to the meter bridge spare input. Depressing the leftmost pair of switches on the meter bridge will then switch the meters 1-8 from "read the M-16" to "read the tracks on the recorder, 1-8". In addition to the obvious BUSS-TAPE comparison, this read the recorder type logic will allow you to use a single output such as a direct from a module, and still have the ability to see a meter. The logic is "what track am I recording" instead of "what mix buss am I using".

After the meters, the signal is routed to the monitor, TAPE IN A. This 8 control system can now be used as the monitor and the AUX can feed the cue system, as described in 16 track operation. Since the TAPE IN B (9-16) is now not needed, you may ignore it or .... We show the 1-8 folded down to the TAPE IN B.

Now, the extra functions can be used as separate 8 signal mixes. The remainder of the lines on our drawing show the 9-16 group (now an extra 1-8) being used as a stereo mixer feeding an effect device, and return being controlled by the spare MON IN pot. This gives you an echo system that will work when simple playback before remix is the job at hand. The AUX 4 is not used.
In this unorthodox patch we show the tape playback lines returning from the 16 track directly to the input module patch point. This connection bypasses the first gain stage of the input module, and you lose the overload LED and the trim control. We assume that when your master tape is complete, all tape signals will be at a “safe” level, and you won’t need the trim or overload warning lights.

After the signals pass through the module and are assigned to a pair of busses to allow a stereo “mix” (we show 5-6, but any pair you find convenient will do). The individual signals are taken from the input module DIRECT OUTs, and routed to the MONITOR TAPE INs.

This re-route gives you another post-fader, post EQ 16 track mix to go along with the two aux’es you have on each input module giving you a total of 3 complete 16 track “sends” that are all separately controllable. The receivers? We indicated that you should ignore the regular “effects to PGM” method, and use the INPUT Modules 17-24 instead. Be careful not to use the aux circuits on these modules, just assign your effects returns with the buss assign switches. AND you will be “in business”.

Two 8 x 2 systems (AUX 3 and 4) are still uncommitted, but cannot be combined without another accessory mixer, but ... if you wish to use a fourth effect on a pair of signals in the same group (1-8, or 9-16) you can use PART of the system to send, and receive with the two remaining INPUT modules.

The monitor? Simple, listen to the 2 track on one of the spare “line ins” to the stereo master module. You know about the pre-post option on the AUX systems. If a prefader signal is desired on your third independent system, use the EQ OUT on the signal instead of the DIRECT OUT.
SPECIFICATIONS

1. 16(24)-Input/8-Output:

2. Input Selector:
   1-16 channel  MIC/LINE/(RMX)
   (17-24 channel) MIC/LINE/OFF

3. Mic Input:
   Mic Impedance  200 to 600 ohms
   Input Impedance  600 ohms, balanced, XLR type
   Nominal Input Level  -60dBV (1mV)
   Minimum Input Level  -70dBV (0.3mV)
   Maximum Input Level  +20dBV (10V)
   TRIM at max. and ATT to 0dB

4. Line Input:
   Input Impedance  50kohms
   Nominal Input Level  -10dBV (0.3V)
   Minimum Input Level  -22dBV (5.6mV)
   Maximum Input Level  +28dBV (25V)
   TRIM at min.

5. PGM Buss Output/AUX Output:
   Minimum Load Impedance  3.5kohms
   Nominal Load Impedance  10kohms
   Nominal Output Level  -10dBV (0.3V) / -2.2dBV(0.8V) switchable
   Maximum Output Level  +18dBV (8V)

6. Stereo Master A Output/Stereo Master B Output:
   Minimum Load Impedance  3.5kohms
   Nominal Load Impedance  10kohms
   Nominal Output Level  -10dBV (0.3V)
   Maximum Output Level  +18dBV (8V)

7. AUX A Output/AUX B Output:
   Minimum Load Impedance  3.5kohms
   Nominal Load Impedance  10kohms
   Nominal Output Level  -2.2dBV(0.8V)
   Maximum Output Level  +18dBV (8V)

8. Monitor A Output/Monitor B Output:
   Minimum Load Impedance  3.5kohms
   Nominal Load Impedance  10kohms
   Nominal Output Level  -2.2dBV (0.8V)
   Maximum Output Level  +18dBV (8V)

9. EQ Out/Direct Output:
   Minimum Load Impedance  3.5kohms
   Nominal Load Impedance  10kohms
   Nominal Output Level  -10dBV (0.3V)
   Maximum Output Level  +18dBV (8V)

10. AUX 3, 4 Access Send Output:
    Minimum Load Impedance  3.5kohms
    Nominal Load Impedance  10kohms
    Nominal Output Level  -10dBV (0.3V)
    Maximum Output Level  +18dBV (8V)

11. AUX 3, 4 Access Receive Input:
    Input Impedance  50kohms
    Nominal Input Level  -10dBV (0.3V)
    Maximum Input Level  +18dBV (8V)

12. Effects Receive Input:
    Input Impedance  20kohms
    Nominal Input Level  -10dBV (0.3V)
    Maximum Input Level  +18dBV (8V)

13. Monitor Buss Input A/Monitor Buss Input B:
    Input Impedance  68ohms
    Nominal Input Level  -10dBV (0.3V)
    Minimum Input Level  -22dBV (5.6mV)
    AUX BUSS IN at max.

14. Tape Input:
    Input Impedance  20kohms
    Nominal Input Level  -10dBV (0.3V)
    Maximum Input Level  +18dBV (8V)

15. Tape Foldback:
    Connected to tape in terminals in parallel.
    Equivalent to load impedance of tape recorder out put connected to tape in.

16. BUSS IN Input:
    Input Impedance  20kohms
    Nominal Input Level  -10dBV (0.3V)
    Maximum Input Level  +18dBV (8V)

17. Access Send Out (Input/Buss Master):
    Minimum Load Impedance  3.5kohms
    Nominal Load Impedance  10kohms
    Nominal Output Level  -10dBV (0.3V)
    Maximum Output Level  +18dBV (8V)
18. Access Receive Input (Input/Buss Master):
- Input Impedance: 68ohms
- Nominal Input Level: -10dBV (0.3V)
- Maximum Input Level
  - (Input): +18dBV (8V)
  - (Buss Master): +30dBV (30V)

19. Talkback Output:
- Minimum Load Impedance: 3.5kohms
- Nominal Load Impedance: 10kohms
- Nominal Output Level: -2.2dBV (0.8V)
- Maximum Output Level: +18dBV (8V)

20. Test Tone Output:
- Nominal Output Level: -10dBV (0.3V)

21. Headphones Output:
- Load Impedance: 8ohms
- Maximum Output Power: Greater than 100mW Monitor VR at Max.

22. Line A Input/Line B Input:
- Input Impedance: 100kohms
- Nominal Input Level: -10dBV (0.3V)
- Maximum Input Level: +18dBV (8V)

23. Frequency Response:
- PGM Buss Output: 20 to 20,000 Hz, ±1 dB
- AUX Buss Output: 20 to 20,000 Hz, ±1 dB
- Mon. Buss Output: 20 to 20,000 Hz, ±1 dB
- Headphones Output: 50 to 15,000 Hz, ±2 dB

24. Equalizer:
- Type: Peak/Dip Parametric, 4 Band
  - Level: ±15dB
- Frequency: 30 to 300 Hz, 200 to 2,000 Hz, 1,000 to 6,000 Hz, 5,000 to 12,000 Hz

25. Filter:
- Low Pass: 15,000 Hz, 12dB/oct.
- High Pass: 200 Hz, 3dB/oct., 60 Hz, 12dB/oct.

26. Signal to Noise Ratio (A weighted/unweighted):
- Equivalent:
  - 126dB WTD
  - 123dB WTD
  - (20 to 20,000 Hz)
- Mic:
  - 1 channel: Better than 67/65dB
  - 16 channel: Better than 55/53dB

27. Cross Talk:
- Better than 65 dB
  - (1 kHz, Nominal Input Level)

28. Total Harmonic Distortion:
- Less than 0.03%
  - (1 kHz, Nominal Input Level)

29. Fader Attenuation:
- 75dB or more

30. Overload Indicator Level:
- 15dB above Nominal Input Level

31. Peak Indicator Level:
- 10dB above Nominal Output Level

32. Power Requirement:
- 100/120/220/240V AC, 50/60 Hz, 130W
  - (General Export Model)
- 120V AC, 60 Hz, 130W (U.S.A./Canada Model)
- 220V AC, 50 Hz, 130W (Europe Model)
- 240V AC, 50 Hz, 130W (U.K./Australia Model)

33. Dimension (WxHxD):
- 1,560 x 320 x 850 mm
  - 61.4” x 12.6” x 33.5”

34. Weight:
- 132kg, 291 lbs

35. Power Supply Unit (Accessories):
- Dimension (WxHxD):
  - 420 x 185 x 250 mm
- Weight:
  - 10.5kg, 23 lbs

In these specifications, 0 dBV is referenced to 1.0 volt. Actual voltage levels also are given in parenthesis. To calculate the 0 dB/0.775 Volt 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 obligation.
1. LEVEL SETTING AND OPERATION CHECK

1-1 LINE IN → ACCESS SEND

1. Apply a −10 dB (0.3 V) signal to the LINE INPUT jack on back panel of the Model 501 input Module.
2. Pull out the jumper between ACCESS SEND and RECEIVE jacks for the channel in which the above signal is applied, and plug in an AC voltmeter to the ACCESS SEND jack. The AC voltmeter recommended here should have an input impedance of 50 kohms or more, input capacitance less than 20 pF, and maximum sensitivity of −60 dB with a sensitivity selector switch.
3. Set controls on panel of the Input Module as follows:
   - INPUT select switch: LINE
   - TRIM (LINE): At about position 6.5

4. For a normal condition of the circuit, the ACCESS SEND output should be −10 dB against a LINE IN signal of −10 dB.
5. If the output is not −10 dB, adjust with TRIM. After this adjustment, the TRIM knob should be between position 6 to 7 under normal condition.
6. If you notice any malfunction, refer to the preamplifier (Q1, Q2, U1) circuit schematic and check all transistor voltages and signal levels.
7. Check all remaining Input Modules in the same way.

1-2 LINE IN → DIRECT OUT

1. Apply a −10 dB (0.3 V) signal to the LINE INPUT jack.
2. Plug in an AC voltmeter to the DIRECT OUT jack for the channel in which the above signal is applied.
3. Set controls on panel of the INPUT MODULE as follows:
   - INPUT select switch: LINE
   - TRIM: To about position 6.5
   - H.P.F.: Set to out.
   - L.P.F.: Set to out.
   - EQUALIZER: Set to out.
   - INPUT FADER: "0" position

4. The circuit is in normal condition if the DIRECT OUT level is −10 dB against the LINE IN level of −10 dB.
5. If it is slightly off −10 dB, adjust by the TRIM knob. For a normal condition circuit, the TRIM knob position should be between position 6 to 7.
6. If you cannot obtain the proper reading, check the circuit. If the above Item 1-1 check shows normal but the DIRECT OUT level is incorrect, trouble could be in the INPUT FADER, and IC's (U1, U2, and U6). The gain of U1-1 and U2 is 1, and that of U5-1 is 10 dB.
   The attenuation of the INPUT FADER set at infinity should be:
   More than 75 dB at 20 Hz ~ 20 kHz