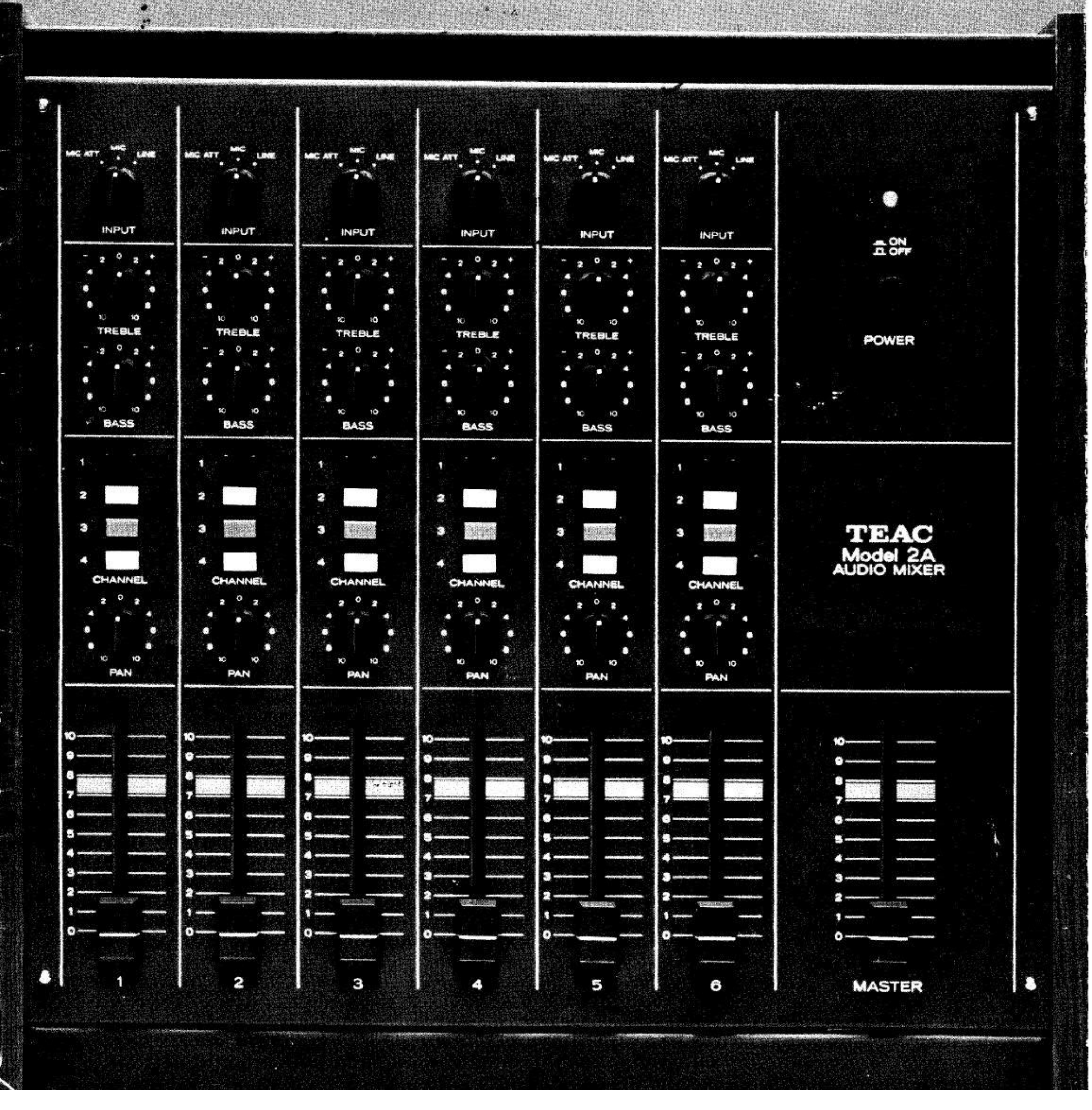
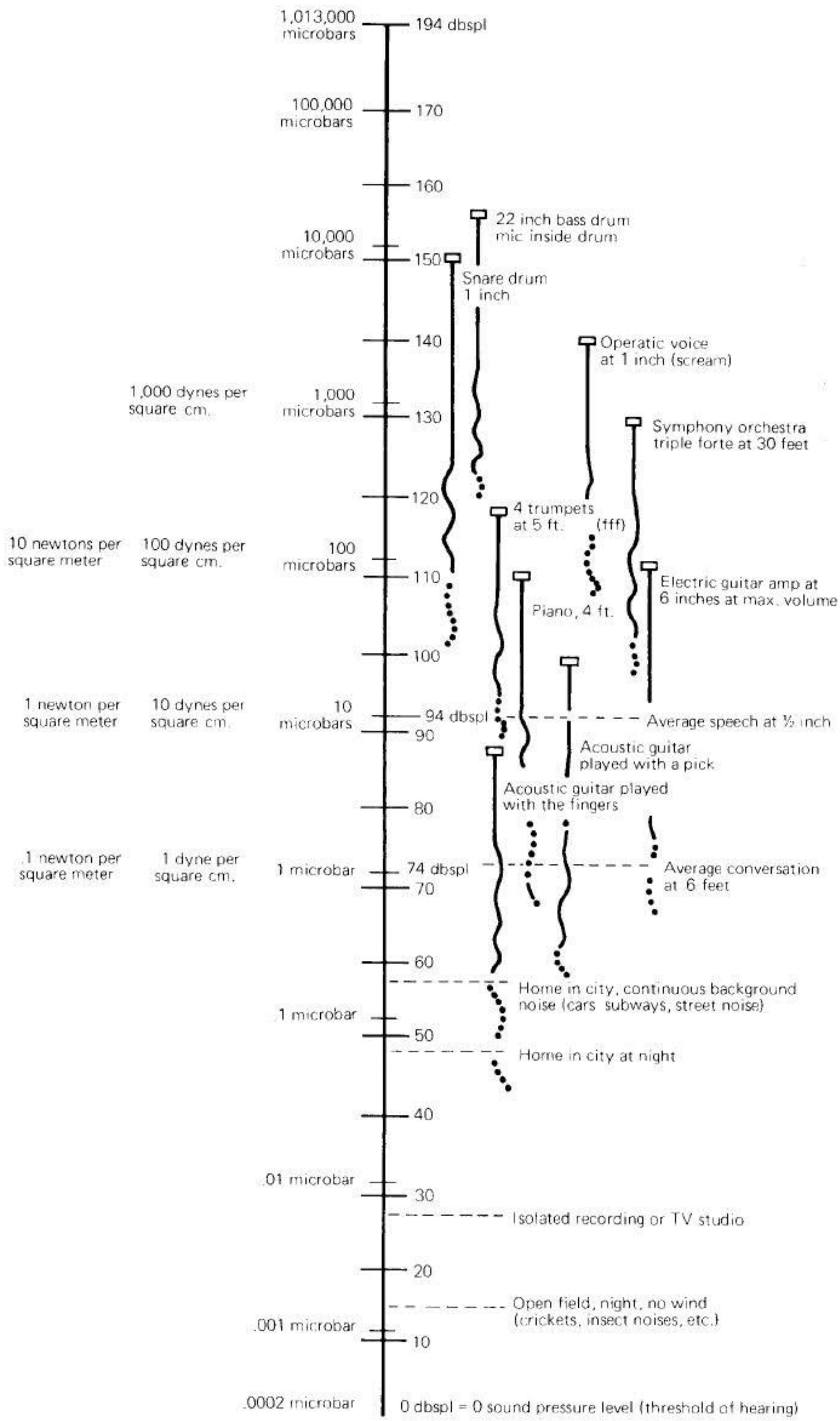
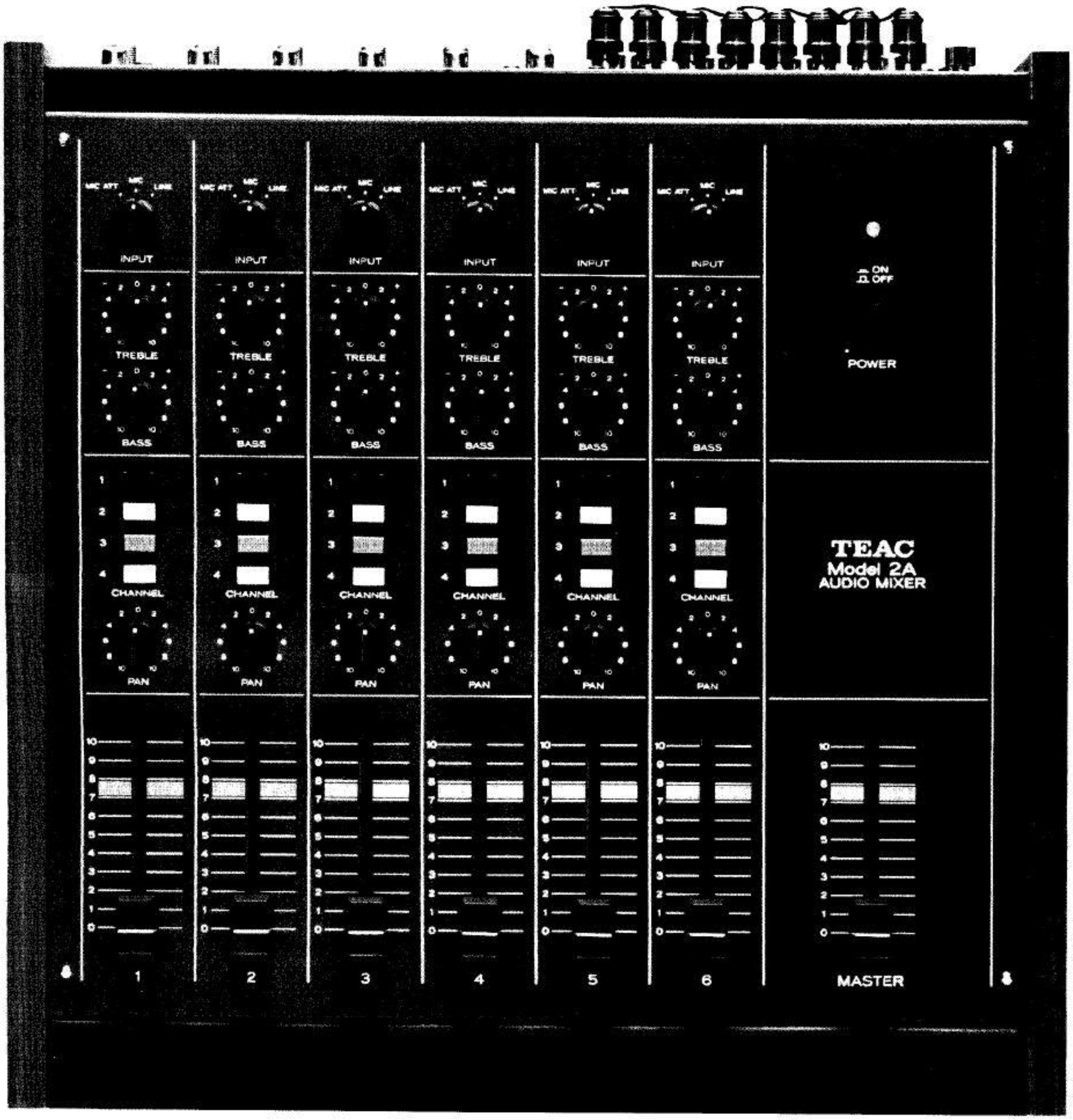
Model-2A

OPERATION MANUAL



SOUND AND MUSIC REFERENCE




TEAC
Model 2A
AUDIO MIXER
MIC ATT
MIC
LINE
INPUT
TREBLE
BASS
CHANNEL
PAN
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1
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MASTER

CONTENTS

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

1. INTRODUCTION

Even though there is a vast amount of information available to the beginning sound mixer, much of it is not basic enough to be easily understood, or, it assumes that the reader has an engineering or scientific background and will be interested in "**the Math**". Practical "rules of thumb" for the novice are not generally available. Something between a picture of the outside of the unit and a complete mathematical analysis of the circuits inside is needed. You don't have to build a mixer from scratch, you just need to know how to operate one.

However, some numbers are unavoidable. The 2A mixer does nothing useful without being connected to quite a lot of sophisticated gear. Mics, tape recorders, power amps, loudspeakers-all play a part in the process of mixing/recording and each piece of gear has its own requirements and problems. We have tried to make this manual as simple as technology will allow. Each section or topic will give you some basic instruction in the terminology used in the process of mixing as well as a list of what plug goes into which jack.

Whenever possible, the scientific terms have been related to understandable common references. Understanding what is going on inside your equipment will help you improve your sound. Think of this manual as a reference book. You won't need all of what is here to begin, and it certainly is not necessary to memorize it, but do try to find time to read it carefully at least once. That way you will be familiar with its contents, if you need the numbers they will be there waiting. Good luck with your sound.

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 men 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 travelling on wires as electrical energy, sound on tape as magnetic energy, sound in air, any place 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 of two things. No matter what system of units of other 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 the zero, or "no difference" point is placed somewhere. Then a comparison becomes 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 db 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 no linear progressions to satisfy the ears' demand.

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

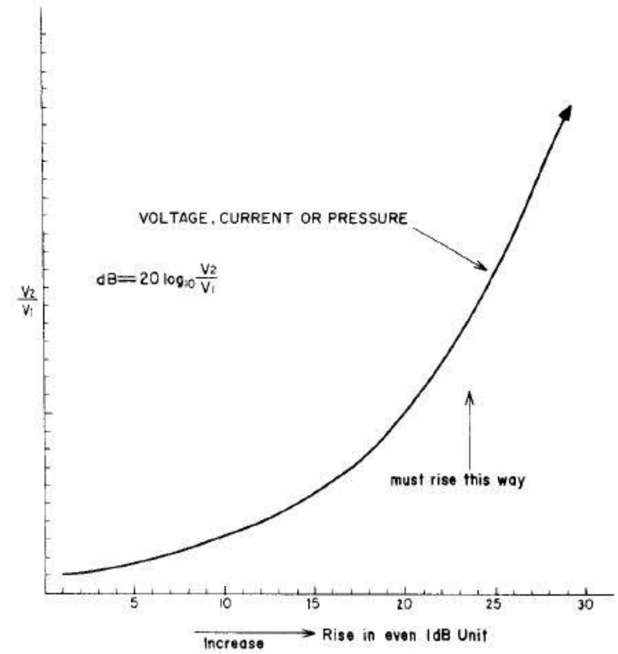
Here are the fomulas for unit increment, but they are necessary only for designers and unless you build your own gear, you won't have to deal with them. For power (watts) increase or loss, calculate by the following equation:

$$10 \log_{10} \frac{P_2}{P_1} = N \text{ (dB)}$$

For voltage, current or pressure cálculations:

$$20 \log_{10} \frac{V_2}{V_1} = N \text{ (dB)}$$

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 began with.

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 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.

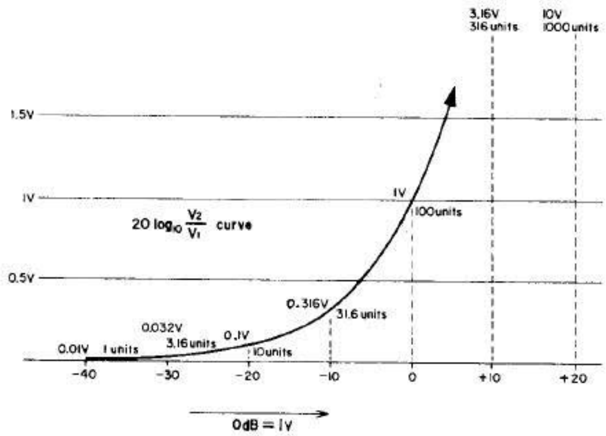
They now have the world's biggest audio mixing system and even when they started out electricity was not free. They set their electrical power signal reference as low as was practical at the time, and it has lowered over the years as electronic equipment got better. In 1939, the telephone company, radio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

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.
Because of this general use of the 600 ohms standard, 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 ohms, 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 the Model 2A 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 db = 1 Volt

This is now the preferred reference for all electronic work except for the telephone company and some parts of the radio and television business. Long distance electronic transmission still is in need of the 600 ohm standard.
If you have a voltmeter, or other test gear that reads in dbm, or "telephone standard", 0 db will be .775 volts and you will have to add a correction factor of +2.2 db to your readings.

If your test gear has provision for inserting a 600 ohm load, be sure the load is not used when working on the Model 2A.
Now that we have given a reference for our "0 db" 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.



If you are not familiar with the term "impedance" and have no knowledge of what is meant by "load" and "ohms", relax. It is the next section in our manual and will tell you what you need to know in order to use the Model 2A properly.

IMPEDANCE MATCHING – OR; WHAT PLUGS INTO WHAT

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

$$\text{OUTPUTS} \xrightarrow{\text{plug into}} \text{INPUTS}$$

OUTPUTS

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

1. The practical power supply is not infinitely large. At some point even if the circuit is capable of supplying more energy, you will run out of "juice".
2. Long before this happens, you may burn out other parts of the circuit. The output impedance may be close to the theoretically ideal "0 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 may seriously affect the quality of the audio. Distortion will rise, frequency response will suffer, and you will get poor results.

INPUTS

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

The classic measurement for Output impedance is to load a circuit until the voltage drops 6 db (to half the original power) and note what the load value is. In theory, you now have a load impedance that is the same as the output impedance. If you reduce the load gradually, the db reading will return slowly to its original value. How much drop is acceptable? What load will be left when an acceptable drop is read on the meter?

When the load value 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 TEAC must say we do not agree. We think that a seven to one ratio of input (7) to output (1) is not a high enough ratio and here's why.

1. The measurement is usually made at a midrange frequency and does not show true loss at the frequency extremes. What about drop at 20 hz?
2. All outputs are not measured at the same time. Most people don't have twenty meters, we do. Remember, everybody plays together when you record and the circuit demands in practice are simultaneous. All draw power at the same time.

Because of the widely misunderstood rule of thumb, the seven to one ratio, we will give you the values for outputs in complete form.

Even though the true output impedance is –

Cue Out	100 ohms
Acc Send	100 ohms
Line Out	100 ohms

For the practical reasons explained previously, we feel that 7 : 1 ratio is not sufficient. To use this rule of thumb, you must use this higher value. We'll call this value the "output load impedance".

Cue Out	50 K ohms
Acc Send	5 K ohms
Line Out	5 K ohms

This is a number that will give good results with the 7 : 1 method. To go one step further, here are the actual minimum ohmic values we feel are wise. Connect no **total input impedance load** higher than –

Cue Out	50 K ohms
Acc Send	10 K ohms
Line Out/Aux Out	10 K ohms

Input impedance is more straight forward and requires only one number. Load is load, and here are the values for the 2A.

Mic In	50 K ohms
Mic Att	50 K ohms
Line In	20 K ohms
Acc Receive	15 K ohms
Buss In	15 K ohms
Cue Out when Used as Input	10 K ohms

If one output is to be "Y" connected to two inputs, the total impedance of the two inputs must not exceed the load impedance, as mentioned before, and if it becomes necessary to increase the number of inputs with slight exceeding of the load spec, you must check for drop in level, 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 this when you have two identical inputs to connect to one output.

Take the **lower** value of the two input impedances and divide it in half. If the number you have is still 7 times the output impedance, you can connect both at the same time.

When you have more than two loads (inputs), just dividing the lowest impedance by the number of inputs will not be accurate unless they are all the same size but if you still get a safe load (higher than 7 : 1 ratio) by this method, you can connect without worry.

If you must have exact values, here are the formulae:

For more than 2 loads or inputs –

$$Rx = \frac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3} \sim \frac{1}{R_n}}$$

Rx = value of total load

For 2 loads or inputs –

$$Rx = \frac{R_1 \times R_2}{R_1 + R_2}$$

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 = minimum ohms, so read carefully.

When the minimum/maximum statement is made you can safely assume that the manufacturer has already done the "seven times is best" ratio calculation. And the number given in ohms does not have to be multiplied. You can **match** the ohmic 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 ohms 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 lowest **load** impedance, it may be higher or lower ratio than 7 times – and will be whatever the specific circuit in question requires. We have done this in a previous section on Input Matching for the Model 2A.

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 that it takes to blink. In order to give you the numbers for gain, headroom and noise in the 2A 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 2A – 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 1 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 -60 db for a sound pressure of 94 db, so you should have no problems getting up to "Ovu" on your recorder.

If you are going to record very loud sounds you may produce more electrical power from the mic than the 2A can handle as an input. How can you estimate this in advance? Well, the Spl chart and the mic sensitivity are tied together on a one-to-one basis. If 94 db spl in gives -60 db (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 2A. 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!

Cautions:
The 2A will not mix the output of power amps. Any output line rated in "watts" **should never** be connected to ANY input of the 2A without a properly designed large resistive pad. The parts in the 2A will not accept any more than 1/4 watt of power without literally burning out – Don't do it. Conversely, no output on the 2A is powerful enough to operate a loudspeaker or headphones directly. You **will** need a power amp in your system to hear the sound.

REFERENCE LEVELS

We should talk about one more reference, a practical one. Most mic manufacturers give the output of their mics as a minus-so-many-db number, but they don't give the loudness of the test sound in db, it's stated as a pressure reference (usually 10 microbars of pressure). This reference can be found on our sound chart. It is 94 db spl, 10 microbars, 10 dynes per cm^2 or 1 Newton per square meter for mics, the reference "0" is 1 volt (db). So, if the sound is 94 db spl the electrical output of the mic is given as – xx 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 or slightly higher for the better grade of condenser mics available today. TEAC Recorders and mixers work at a level of -10 db referenced to 1 volt (.316 volt) so, for 94 db spl a mic with a reference output of -60 db will need 50 db of amplification from your 2A 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 2A to make the needles on your recorder read "Ovu". The amounts of gain or loss available in each stage of the 2A are given in the section **Mic in to Buss Out** beginning on page 9.

THE BLOCK DIAGRAM AND GAIN BLOCK DIAGRAM

Before you begin reading the next section of this manual, flip out the extra fold on the back cover. On this page we have printed the two block diagrams. The upper one shows the signal flow through the Model 2A and it represents in simple form the actual electronic arrangement of all the jacks, controls and gain stages from mic in to line out.

The lower diagram 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 2A 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 "cue out" 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.

Read the text with the diagrams visible and you will become more familiar with them.

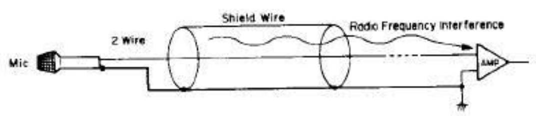
2. THE MODEL 2A FROM MIC IN TO BUSS OUT

MIC IN

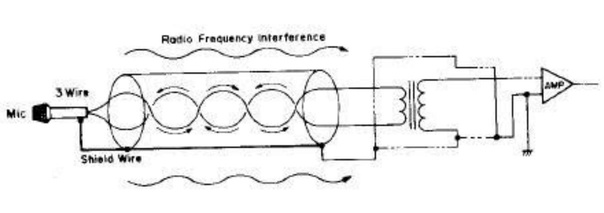
Accepts a standard 1/4 inch phone jack. This input is the first step in the chain of parts that make up the Model 2A. It is wired directly to the first gain stage. Low (50–250 ohm) or High (1k–10K ohm) impedance microphones can be connected here as long as their output does not exceed -30 db (.03 volt). This level of electrical energy can easily be reached by a microphone even if the sound pressure is only moderately loud, so a switchable "pad" or attenuator is provided. There are two positions, one marked **"Mic"** the other marked **"Mic Att"** on the input selector switch. In **"Mic"** the gain of the first stage is 40 db. In the **"Mic Att"** position the circuit components wired to the first gain stage are altered and the gain is lowered to 20 db. When the amount of amplification required is less, this circuit performs better, so if you have a choice between a low setting of the input fader with the selector switch in **"Mic"** or a rather high setting with the switch in **"Mic Att"**, **"Mic Att"** is the correct choice for maximum quality.

With the selector switch in the **"Mic Att"** position the maximum input becomes -10 db (.3 volt). Again, an improvement. Less gain required from the amplifier translates directly into more headroom, or reserve power. When Mic cable runs exceed 10 feet we strongly recommend the use of Mic input transformers and balanced (3 wire) microphones. Radio, television and the rapidly increasing use of Citizens Band transmitters make radio frequency interference a real problem in 2 wire mic lines.

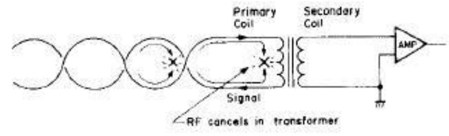
The only sure way to deal with this problem is the 3 wire system. Here's how it works –



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 primary coil will generate current in secondary coil. Signals with common polarity will cancel out in the primary coil and will not pass to the secondary. No signal in secondary coil means no signal in the amplifier.

The selector switch has one more function on it, marked **LINE IN**. In this last of its 3 positions it bypasses the first gain stage altogether and signal now enters the mixer from the Line in jack. Since this RCA jack is expected to carry an already amplified signal (at -10 db, .3 volt), we don't need to boost it up any further, we just need to get it to the controls that appear further on in the mixing chain. Because we have bypassed the first gain stage the line level input is capable of handling very powerful (by comparison) signals. The maximum peak input here is +20 db (10 volts). Right here we must remind you that the maximum undistorted output you can get from the Model 2A is still only +10 db (3 volts). Even though you can control a powerful signal by connecting it to the **LINE IN** jack, you will not get all 10 volts of it back. +10 db is tops.

Let's get back to our normal reference for line level of -10 db, .3 volt and continue –on the block diagram there is a wiggly section on the line leading into the selector switch. This indicates that there is a resistive pad in the line. This part reduces the line reference level from -10 db to -20 db so it will be the same as the output from the first gain stage, or mic amplifier.

We have now described the "mic in" jack, the "line in" jack, the first gain stage and all three positions of the input selector switch. This group of parts, both electronic and mechanical is usually called the "front end" of the mixer. After this stage, all controls deal with switching, processing or routing signal, and right at this point, we have done enough to the signal in the way of gain to allow us to call it "line level". It's not reference level yet, but it is powerful enough to be useful, so right here we put our first output.

THE CUE OUT

At this output you will get whatever signal is on its way to the input fader. The reference level will be -20 db from either a mic or a line source signal. What do you need an output here for anyway? It doesn't seem too useful, no fader yet to control level, or tone controls and such. Think carefully about the process of mixing/recording. One combination of sounds, one mix, will not always give you what you need. It's true that when you are finished with a piece, one mix is what you will listen to, but while you are making the recording, having a source of signals for a totally separate mix can be just what the doctor ordered.

Examples –

1. **ECHO SEND** – You may not want echo on everything.
2. **MUSICIANS MIX**

In many cases, what is being sent as a mix to the recorder is not quite right for a cue mix. The musician wants more or less of, say, the bass part than you think is right for the actual recording.

Well, how about the main line outs? There are 4 of them – why not use one for cue? No good, the signal on the main line will respond to the position of the input fader. You really need more than one "input fader" to have full control of another arrangement of the sounds you are working with.

With this cue out jack we've given you access to the signal **before** you set levels to the Master Line Outs. It's up to you to provide the rest of the secondary mixing system. Here are the rules. What you can connect here depends on what you are doing with the "front end" of the mixer.

If you are using "Mic" position on the input selector switch, the device you connect to cue out should have an input impedance of at least 10 K ohms. Lower values of input impedance may affect the quality of the signal going on through the mixer. If you don't know the exact value of the input impedance for the device you want to connect there are still a couple of ways to test it.

1. Listen to the main signal output (line out or Aux out) as you remove and re-connect the device to the cue out. If you hear a drop in level or a change in the tonal balance of the sound on the main line out, the device is likely to be drawing too much current and is unsuitable.
2. If you can get a steady tone to work with, you can read your recorders' VU meter and make this measurement. Patch the steady tone through the mixer from input to line out. Set the faders so the tone reads "0" VU on the recorder. Now, connect the device you wish to check to the "cue out" on the input module you are using for the steady tone. If the drop in level is more than 0.5 db, the device has too low a value of input impedance. It will not damage the mixer to use it, but it will affect the quality of the signal on the main line.

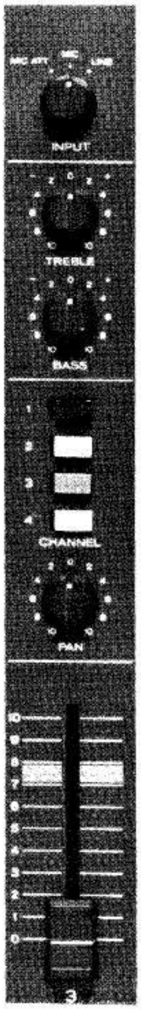
If making the connection to the cue out causes the main line signal to drop 6 db, the input impedance of the device is the same as the output impedance of the cue out.

You should also know that to make the "disconnect, reconnect the load" test in section 1 above, you must actually pull out the connection. Many devices will affect a circuit when the signal cable is connected, even if they are not plugged in the power line, or their power (AC) switch is in the off position.

If you are in "Line" position on the input select –
The effect on the main line signal of the "cue out" connection will depend on the line level output impedance of your source. In other words, what device have you got plugged into line in? It's output impedance will count, what are the numbers for your source of signal? You will have to figure them in, but the "disconnect, reconnect" test will work here as well.

Using a Y cable to reach two devices with one signal may work. But the total input impedance of both connections must be figured. Two leaks in a bucket of water are alwavs worse than one. For impedance calculations, the higher the number in ohms, the smaller the "leaks" in signal. The input impedance calculations and "rules of thumb" can be found on Page 6 in the section on impedance matching.

Using a Y cable to "sum" or join two cue 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. What happens? Let's assume that you have "Y" connected all six "cue out" jacks to one common line. Now, signal appearing on any input will flow across all six inputs by way of this common line, and raising any one of the six input faders will raise all six sounds on the main lines and you have no control of the main mix.



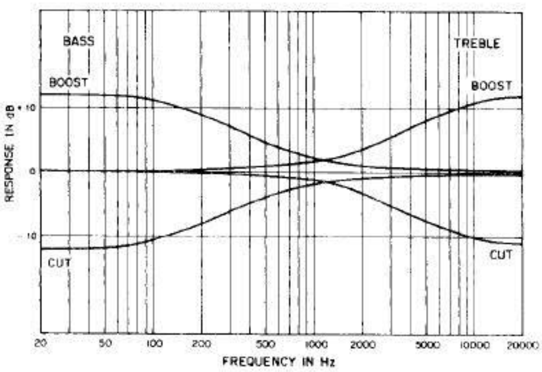
THE INPUT FADER

Controls the amount of signal. Faders, often called volume controls, pots 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. When you advance the setting of the input or master fader you are not "turning the gain stage up" it is already up all the way. You are just reducing the loss of the fader. The entire signal flows to the next stage only if the fader is "wide open" (as close to the channel assign switches as possible).

THE TONE CONTROLS

After passing through the input fader the signal arrives at the tone control, or equalizer section. The channel assigns? not yet. They come last on the input module. Remember, we are describing the actual wiring sequence, not the exterior control layout. Where we put the knobs and buttons is a matter of physical convenience, not electronic necessity.

There are two controls. The lower knob effects the bass frequencies and the upper knob the treble. Turn the knobs clockwise (rotate to the right) to boost, and counter clockwise (rotate to the left) to cut. How much can you get? Look at this graph. The maximum boost and/or cut for any frequency can be found here.



12 db at 100 Hz, boost or cut
12 db at 10KHz, boost or cut

With a continuously adjustable control the range of tonal effects is very great.

Before using the tone controls on Mic signals, it is better to get as close to the sound you want by moving the mic. Even a small change in mic location can make a big difference in the sound quality. Listen to the sound from the actual mic position. Place your head where the mic is, and listen carefully. What do you really hear? Is it what you want? Doing this check may help solve many problems. Too much bass? Not enough? Well, perhaps there is a better location for the mic in order to get the balance you need. When you have gone as far as you can in this fashion, the tone controls will get you the rest of the way.

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

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

On the block diagram and gain chart you can see that the tone control stage has a moderate gain (18 db) and a very large excess gain capability or "headroom" (25 db). This gain chart is made with the assumption that the tone controls are set to the "flat" or "no boost or cut" position. The reserve in the circuit is necessary to maintain a 20 db value of "headroom" when the tone controls are set for maximum effect. Without this extra margin, you would have to lower the setting of the input fader when you used the extreme boost or cut settings of the tone controls.

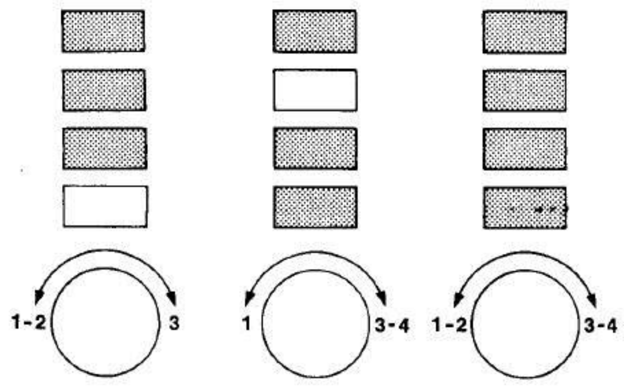
Add it up — if you start with a reserve of only 25 db and you boost 12 db (the maximum) at 10 Khz, your margin of safety is reduced to 13 db, For a steady tone from an organ or a violin, this might be just enough to avoid clipping, or serious distortion but it is definitely not enough to cope with any percussive peaks from things like piano, guitar or drums.

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

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

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

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



Typical examples of multichannel panning

Leaving the Pan/channel assign, the signal is passed through a **"summing resistor"** and suffers a big drop in level before it is allowed to pass down the wire to the **summing amplifier**. This loss is necessary to prevent the signal from one input going back into another instead of going down the line to the master fader. You can think of the summing resistor as the "traffic cop" that turns the line into the "one way street" you need here. The mixed signal is pretty low now, so we need a gain stage (summing amp.) to get our level back up to -10 db. The gain here is a fixed 16 db.

Accesory Send – Receive

At this point in the Model 2A we have a completed mix with the level at reference point. This output is used to insert a signal processing device such as a graphic equalizer or a limiter. Route your mix out here, pass through the accessory you want to use, and back into the accessory receive plug for final master gain control of the whole works. No signal will pass to the master fader unless the accessory send-receive section is connected in some way. Either a device or the jumpers provided must be in place. The accessory send has an output impedance of 100 ohms. The accessory receive has an input impedance of 15K ohms. Remember, impedance matching always counts. Reread the section on page 6 if you're in doubt about what connects to what.

If you decide you can handle your mix without benefit of a Master fader, the accessory send is the output with the best quality audio signal. We recommend this output for stereo mix down if you want to squeeze that last drop of performance out of your 2A. Connect the accessory receive to the line inputs of your two channel recorder, and monitor the signal from the recorder line outs.

Buss in Jack

The primary purpose of this final input on the block diagram is to "stack" or run a pair of mixers with one overall master control. The input impedance and signal point are identical to the accessory receive jack. Any electronic device that has a compatible output impedance may be connected here and its contribution to your mix will then be controlled by the master fader on the 2A.

Master Fader

Provides final level control of all four output sections of the mixer. This single control operates all four faders at once. Any signal added to the system from the various jacks on the 2A will be affected by the setting of this control, if your mixed signal is taken from the last output pair on the block diagram.

Line Out – Aux Out

One last gain stage appears after the master fader and just before this double ouput jack. A small amount of gain is necessary to make up the losses caused by wiring up the master fader, and to give a solid source of signal with a stable and relatively low output impedance to whatever you are mixing to. The final reference level is -10 db, .316 volt. The maximum level before clipping is +10 db, 3 volts.

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

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

3. PATCHING SYSTEM CONNECTION AND WORTHWHILE ACCESSORIES

Most people tend to look for a "permanent" set of connections when they set up a mixing system and it is true that the logic of control function just on the top of the mixer takes some time to become 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 2A to suit, you can get better results.
For this reason we suggest that you plan on access to the back panel of the mixer. Don't set up the system in such a way that you "hide all that mess". Leave yourself room to get at all the connectors. You will need all the options you can get.

Use good cable. The best quality interconnecting cable is not much more money than the least expensive and will improve the sound of your system out of all proportion to its cost. We realize that a "dollar more" is more than just one dollar (there are 38 jacks on the 2A alone) but low loss cable is worth it.

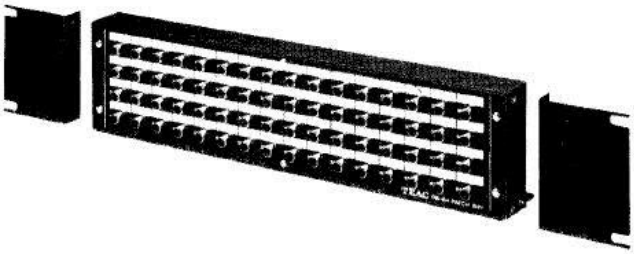
ACCESSORIES

TEAC Low Loss Cable

We strongly recommend the use of TEAC Low Loss Cable. It comes in prewired lengths from six inches to 20 feet and is the best quality cable for 2 wire systems that we could find. Low Loss, double braid shield, steel connectors and color coded ends make this cable a real value. Use it.
It's a good idea to label both ends of every cable in the mixing system. When you have 40 or more cables to sort out, a label will save you time and prevent confusion.

The PB-64 Patch Bay

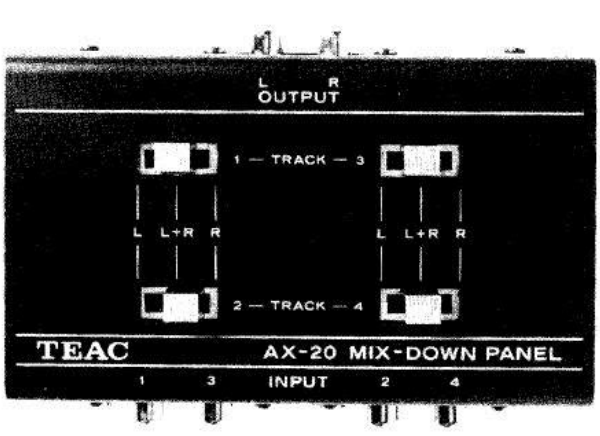
When your system begins to expand beyond the basic, sorting out where things go can take much time away from the recording process. This accessory will allow you to speed things up and get back to what you really want to do. Sixty-four RCA pins on a panel. So you can bring all those jacks to where you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



PB-64 Patch Bay

The AX-20 Mixdown Unit

For additional patching possibilities to expand the flexibility of your system, the AX-20 is an inexpensive and convenient 4-in / 2-out line-level mixdown panel that can provide extra signal-routing channels for an unlimited variety of applications. With a switching matrix for each input, signals can be rerouted, combined or mixed down in any number of different combinations for cue mixes, studio, control room or stage monitoring, special effects, signal processing, etc. The AX-20 is a small, low-cost optional accessory that can add a wide dimension of versatility to your system for a nominal investment.



AX-20 Mixdown Unit

Since the AX-20 is "passive" (it has no source of power) its insertion loss is high. We don't recommend using it in the actual signal path to a recorder input.

The TO-122A

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

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

The TO-122A Test Tone Oscillator

4. SETUPS AND PATCHING

The actual use of each line out or line in on the 2A will be determined by what you have in the way of extra equipment. It is not, in itself a complete system. You will need loudspeakers or headphones and some kind of power amplifier to monitor the sound, tape decks to record the sound and microphones to start the whole process by picking up the sound. Where you connect most of these things will be obvious.

Mics will go in "mic ins", tape decks will patch to "line ins" and so forth, limited only by the number of input jacks on the 2A. Since there is no headphone output on the 2A, you will have to find one elsewhere. The tape deck, or a receiver might have one, or you might consider the purchase of our MB-20 meter bridge. It has provision for expanding the functions of the 2A as well as providing 4 meters. It has a 4 X 2 matrix for feeding a stereo amplifier and a stereo headphone amplifier with a separate volume control. One of its extra functions allows you to playback 4 tape tracks without disturbing any of the six input fader settings. This allows you to make a playback while you are working and keep your level and EQ settings for mics. A valuable extra function indeed.

MB-20 Meter Bridge

5. UNORTHODOX PACHES

1. CUE OUT as an INPUT

If certain rules are followed the cue output can be used as an input to the Model 2A.

Rule 1: The selector switch on the input you wish to use MUST BE IN "LINE". This switch position disconnects the first gainstage (mic amplifier) from the circuits that follow afterwards.

Rule 2: You must disconnect any cable plugged into "Line In". The maximum signal level you can use as an input is reduced to 10 volts (+20db).

The Benefit –

At first this seems more than a little inconvenient. Why bother? What do you gain by operating this way? You will lose the cue outputs normal function.

The answer is –Quality–. This "input" is the simplest in terms of electronic parts and can be profitably used in doing your final mix to the 2 track. Since there are less parts you have less noise. If you shift the signal back and forth between "line in" and "cue out" by moving the connector you will discover that "cue" is louder. You will not need to advance the input fader as far to get level out. How much louder? 10 db, that much. Improvement in signal-to-noise is worth some inconvenience.

2. Using Two Modules to Get More Flexibility.

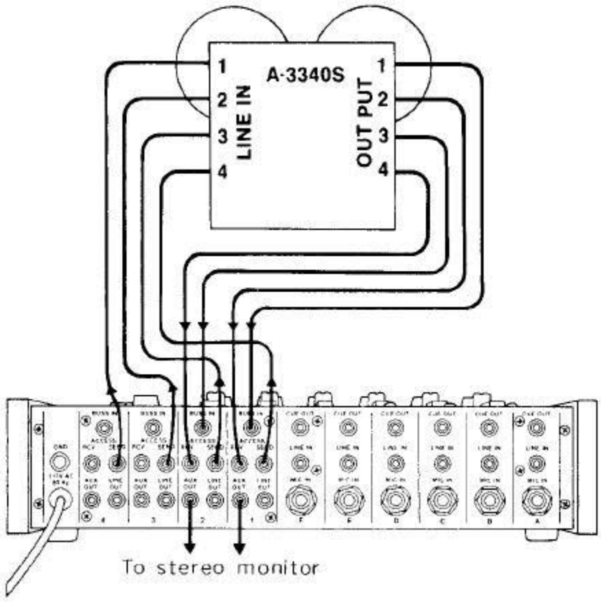
If you need to process a mic signal without affecting the entire contents of a buss, you can patch this way (any 2 inputs may be used, we'll describe A and B). Start the signal chain with input module "A", plug the mic into it. Take the "cue output" of A and connect it to the phasor/fuzz tone/equalizer/limiter/echo or what-have-you. Take the output from the processor and plug it into the Line in on input module "B". Now module A is being used to amplify the signal and send it to the what-have-you but the processed signal is controlled and assigned from module "B".

Here is a list of maximum gain factors for the various input plugs of the Model 2A through to the outputs.

If you start here with a level of–	And you measure the output here–	The gain will be –
MIC IN -60 dB	CUE OUT	40 dB
	ACCESSORY SEND	48 dB
	AUX OUT/LINE OUT	56 dB
MIC ATT -40 dB	CUE OUT	20 dB
	ACCESSORY SEND	28 dB
	AUX OUT/LINE OUT	36 dB
LINE IN -10 dB	ACCESSORY SEND	8 dB
	AUX OUT/LINE OUT	16 dB
CUE OUT as input -20 dB	ACCESSORY SEND	18 dB
	AUX OUT/LINE OUT	26 dB
ACCESSORY RCV -10 dB	AUX OUT/LINE OUT	8 dB
BUSS IN -10 dB	AUX OUT/LINE OUT	8 dB

3. Four Channel Monitor Functions

Since there is no separate stereo output on the 2A, you will have to arrange for some kind of 4 to 2 mix of the outputs in order to hear all four "line outs" at the same time. You can use an MB-20 or an AX-20, but if you can live without a master fader on your mixer there is a way to go 4 to 2 without any extra gear. Since the accessory send and receive jacks are not internally connected, you can separate the mixer into two independent parts by pulling out the jumpers and "bridging" this point with your 4 track recorder. You patch the accessory send to the recorder Line inputs and recorder outputs as shown in the diagram. Patch one pair of outputs to the "accessory receive" and the other pair of outputs to "buss in", now the master fader will work the output of the 4 channels of the recorder instead of the output of the 2A. Individual playback control of each monitor channel will now be possible by using the controls on the recorder. With this set up, your ability to monitor panning is restricted a bit. The pan between channels will still work, but you won't hear it if you pan between a pair that is connected to the same buss in-accessory receive pair. They share a speaker and become mono in the monitor system only.



Output from 2A taken from Acc send 3340 line outs then go to –

1. Buss in channel 1
2. Aux in channel 1
3. Buss in channel 2
4. Buss in channel 2

Master fader becomes master monitor – does not affect recording, becomes playback or source volume from 3340 line output controls.

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

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

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

If you want to set levels on your recorder more carefully, any source of continuous sound will work better.

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

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

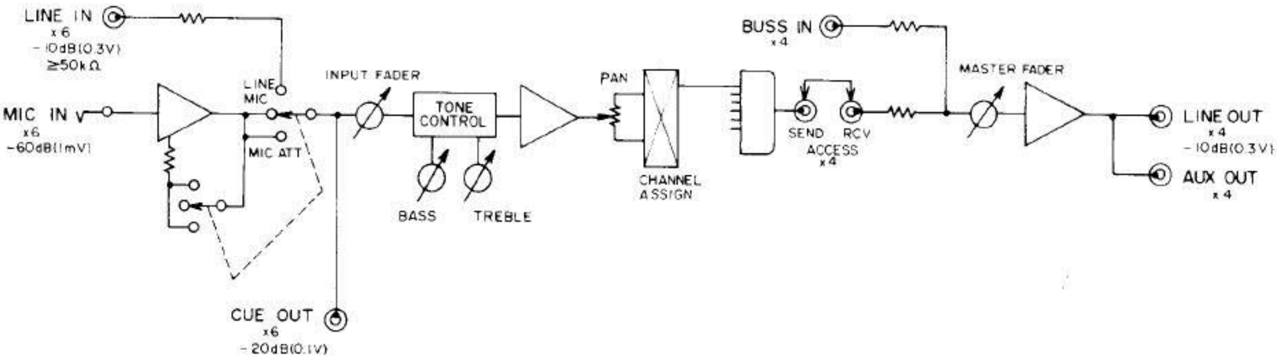
6. MORE INFORMATION IS AVAILABLE

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

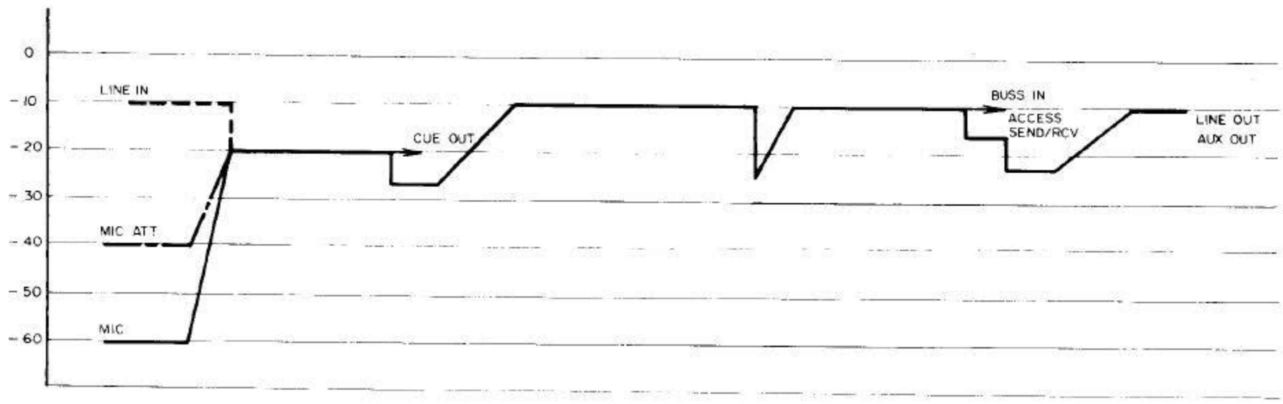
1. Home Made with TEAC. An LP we produced on a 3340, a 3300 and a Model 2. It talks about how multitrack recording has influenced music, and how you can record at home. It's filled with tips on mic placement, echo, reverb, acoustics and the like. Available through your TEAC retailer.
2. The White Paper. An informative 24 page booklet about the technology of tape recorders. Helps you understand bias and equalization, metering systems, and the important inter-relationships of the critical performance parameters. Free from your TEAC retailer, or from us directly.
3. Modern Recording Techniques. An excellent introduction to the equipment, controls and techniques used in modern studio recording. Written by Robert Runstein and published by Howard W. Sams & Co., Inc. Used as a text in seminars offered by and available thru The Recording Institute of America, Inc., 15 Columbus Circle, New York, NY 10023.
4. TAB Books, Blue Ridge Summit, PA 17214. Many fine publications covering subject material from general electronics theory to specific recording topics, including a thorough explanation of Setting Up and Using a Multi-Channel Recording Studio, by F. Alton Everest (#781; publication: fall, 1975). Write to publisher for catalogs and information.
5. Additionally, there are two magazines serving the recording industry that merit your attention:
   dB – Sagamore Publishing Co., Inc.
   1120 Old Country Road
   Plainview, L.I., NY 11803
   Recording engineer/producer – P.O. Box 2449
   Hollywood, CA 90028

We think you'll find these publications informative and useful – particularly with respect to supplementing the many fine articles that regularly appear in publications you're probably already familiar with: Stereo Review, Audio, High Fidelity and the like.

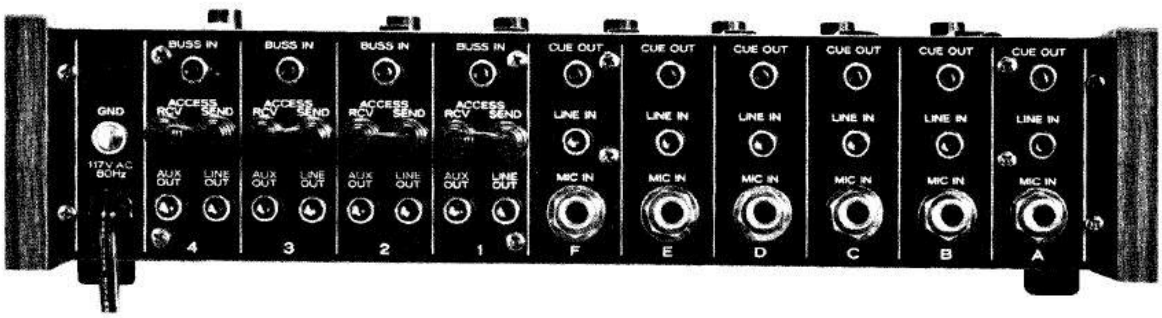
7. BLOCK DIAGRAM



GAIN BLOCK DIAGRAM



REAR PANEL



8. MODEL 2A SPECIFICATIONS

1. 6 Input/4 Output	
2. Input Selector	MIC ATT/MIC/LINE
3. Mic Input	
1) Mic impedance	150Ω ~ 10KΩ
2) Input impedance	50KΩ or higher
3) Input level	-60 dB, Hi-Z input
4. Line Input	
1) Line impedance	20KΩ or higher
2) Input level	-10 dB (0.3V)
5. Line Output/Aux. Output	
1) Load impedance	10KΩ or higher, unbalanced
2) Output level	-10 dB (0.3V)
6. Signal to Noise Ratio	
(Overall — input to output)	
1) One input	Greater than 62dB, WTD
2) 6 inputs	Greater than 55dB, WTD
7. Frequency Response	30 Hz ~ 20 KHz, ±2dB
8. Tone Control	
1) Treble control	Variable ±12 dB
2) Bass control	Variable ±12 dB
9. Crosstalk	Greater than 50 dB
10. Overall Distortion (Input to output)	0.1% THD maximum, at nominal -10 dB output
11. Fader Attenuation	Greater than 60 dB
12. Cue Output	
1) Load impedance	Greater than 50KΩ
2) Level	-20 dB (0.1V) with nominal input
13. Buss Input	
1) Input impedance	15 KΩ
2) Nominal level	-10 dB (0.3V)
14. Accessory Send/Receive	
1) Send load impedance	Greater than 10 KΩ
2) Nominal send level	-10 dB (0.3V)
3) Receive impedance	15 KΩ
4) Nominal receive level	-10 dB (0.3V)
15. Power Requirements	117V A.C., 60 Hz, 5 W
16. Dimensions	13-7/16"(W) X 3-17/32"(H) X 14-9/16"(D)
17. Weight	13.2 lbs.

*Specifications subject to change without notice.

TEAC

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