TAPCO®

TECHNICAL AUDIO PRODUCTS CORPORATION

# MODEL 6200A STEREO RECORDING/SOUND REINFORCEMENT MIXER

OWNER'S MANUAL

# Table Of Contents




**Technical Audio Products Corporation 3810 148th Avenue N.E. Redmond, WA 98052 Phone (206) 883-3510**



P/N 700005

M00123 0378/1M/JPI

# 1. Introduction

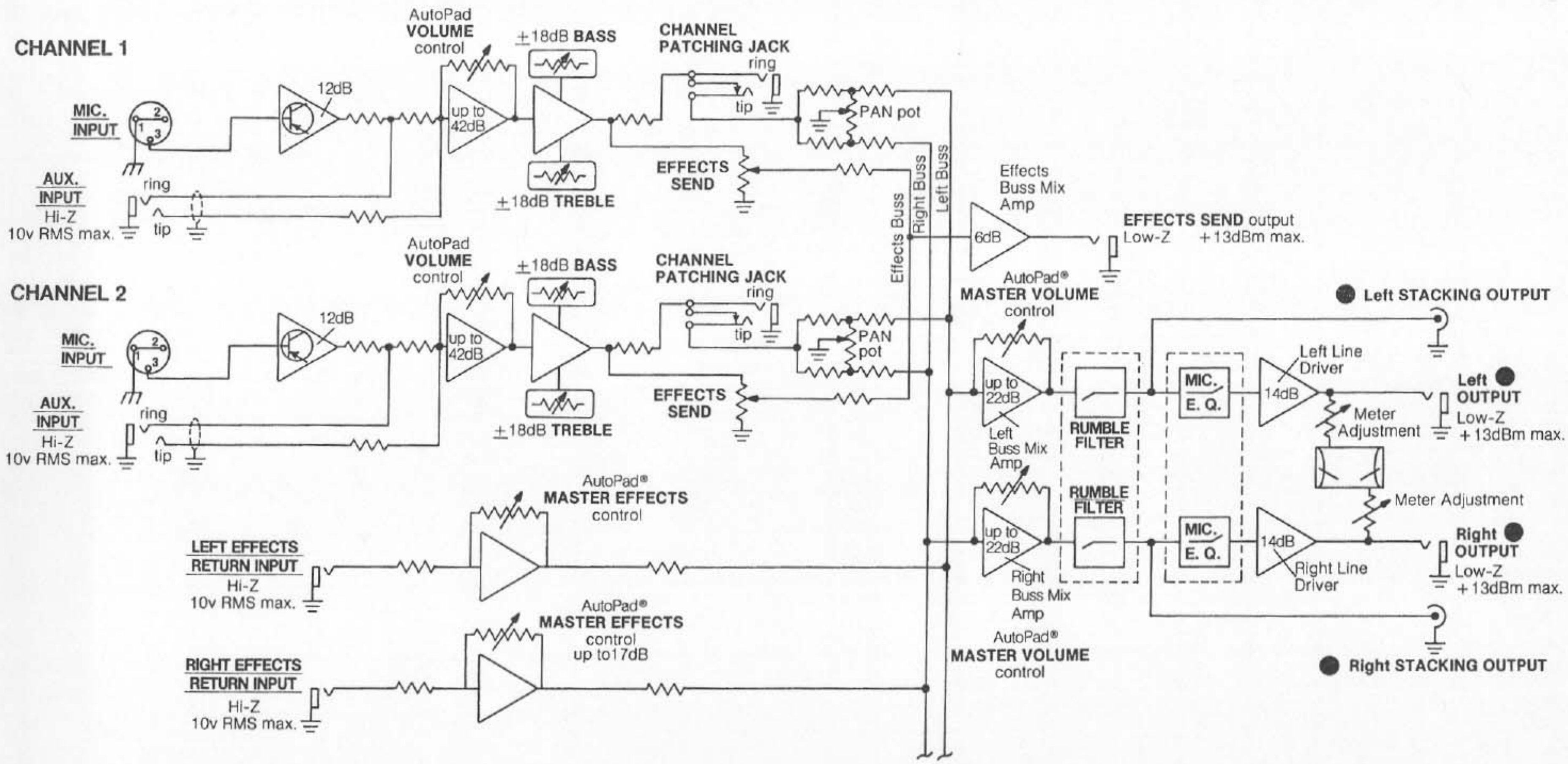
The 6200A stereo mixer is the natural result of exhaustive laboratory research and actual field testing. This mixer is designed to be easy to use, and to provide optimum performance, under even the most demanding conditions. Tapco builds professional audio equipment. Rugged construction for on the road musicians, premium performance on the stage and in the studio — that's what it's all about. The 6200A simplifies many of the problems encountered in both recording and sound reinforcement work. The panel layout makes every setting easy to see at a glance. Low-Z inputs match the 6200A to all professional microphones. AutoPad volume controls eliminate the need for external pads. And the circuit designs used in the 6200A make it compatible with all common peripheral equipment.

In addition to the Low-Z mike input, each channel has an aux. input for line level signals, and a channel patching jack that allows external processing devices to be inserted into the circuit of the channel (and also provides individual outputs for multi-track recording).

Every input channel has an AutoPad volume control, bass and treble controls, an effects control and a pan pot. Both output channels have master volume controls, switchable microphone equalization and rumble filters, and effects return level controls. Output levels are constantly monitored by a pair of meters specifically designed for stereo level matching. The net result of all these features is precise audio control — and that's what the 6200A is all about.

More and more popular music is becoming a truly symbiotic combination of artistry and technology. New techniques have made records sound almost better than the real thing, and high quality playback systems are found in almost every living room. Electronic musical instruments have so expanded the range of modern music that entire symphonic performances are now given with just one instrument, the synthesizer. Musicians have come to depend upon electronic systems, through sound reinforcement, recording and broadcasting, to get their music to the public. And of course, the public has come to depend upon those same systems. The quality of sound experienced by most people today is so good that it's taken for granted. It's just part of our everyday lives. As a result, poor quality sound becomes glaringly apparent, and this is (unfortunately) taken as an indication of the quality of the music itself. Music is no longer judged simply by the quality of the performance because people will no longer accept poor quality sound.

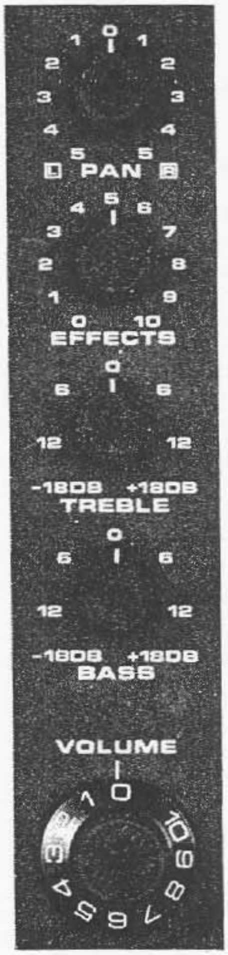
You don't expect high performance from outdated equipment. We can only assume that's why you now own a Tapco 6200A. But it's not always the equipment that's at fault when the sound is bad. The world's best equipment can't cure your sound problems if it's not used properly. To that end we urge you to read your Owner's Manual carefully. It provides the informational tools you need to learn to play the 6200A just as you would play a musical instrument. The control functions on the 6200A are meant to be dynamic — they are meant to be used. The settings on your mixer must flow just as smoothly as the music itself in order to produce a perfectly blended composite product. In the final analysis, the quality of the music will be only as good as the quality of the mixing.




## 2. 6200A Block Diagram

The block diagram shows all of the 6200A's inputs, outputs and internal circuitry in abbreviated form. As you read through the functional descriptions of the controls and connectors on the 6200A, use the block diagram to see where the signals go and how they're processed once they enter the mixer.

# 3. Input Control Functions




**PAN** pots are used to create the stereo image. The pan pot functions much like the the balance control on your stereo system, except that here it controls only its respective channel, not the whole system. Pan is short for panorama, and the audio panorama becomes apparent as some of the pan pots are moved to the left, and others to the right. You can place the sound wherever you want it — middle, extreme left, extreme right, or anywhere in between. Study the effect of the pan pots carefully. Knowing how to make good use of the pan pots will allow you to attain the highest levels of definition and spatial clarity in your recordings.

You will notice that panning a channel to the left or right causes an increase in the output level, as indicated on the VU meters. The pan pot circuitry is designed to produce this increase. When a channel is placed to the extreme left or right, the sound comes from only one speaker, instead of two. So, the loudness of that sound must be increased to compensate for the loss of the other speaker. The panning circuitry in the 6200A is designed to automatically compensate for loudness changes produced by the placement of a sound anywhere in the panorama, allowing you to create perfectly balanced stereo images.

**EFFECTS** provides a completely separate mono mix from the preamp output. As you can see in the block diagram, the signal fed to the effects buss will be equalized the same as the signal fed to the main buss. The effects control is usually used to provide an independent mix for external echo or reverb, headphone or stage monitoring, simultaneous p.a. (while recording), etc. (See "How to use the 6200A" for more ways to use the effects control.) The mix created by the individual effects controls comes out of the mixer at the Effects Send jack on the rear panel. If this signal is processed through a reverb system, for example, the output of the reverb could be fed back into the mixer through the effects return jacks.

**BASS** and **TREBLE** controls on the 6200A allow up to 18dB of boost or cut for the low and high frequencies. The actual performance of these circuits is shown in the graph on the specifications page. The 3dB points, and the overall shape of the curves produced, have been carefully tuned (electronically) for the best sounding results in all circumstances. Actually, 18dB of boost or cut is probably more than you'll ever need. But if you ever do need that much EQ, there's just no substitute — that's why we made so much control available.

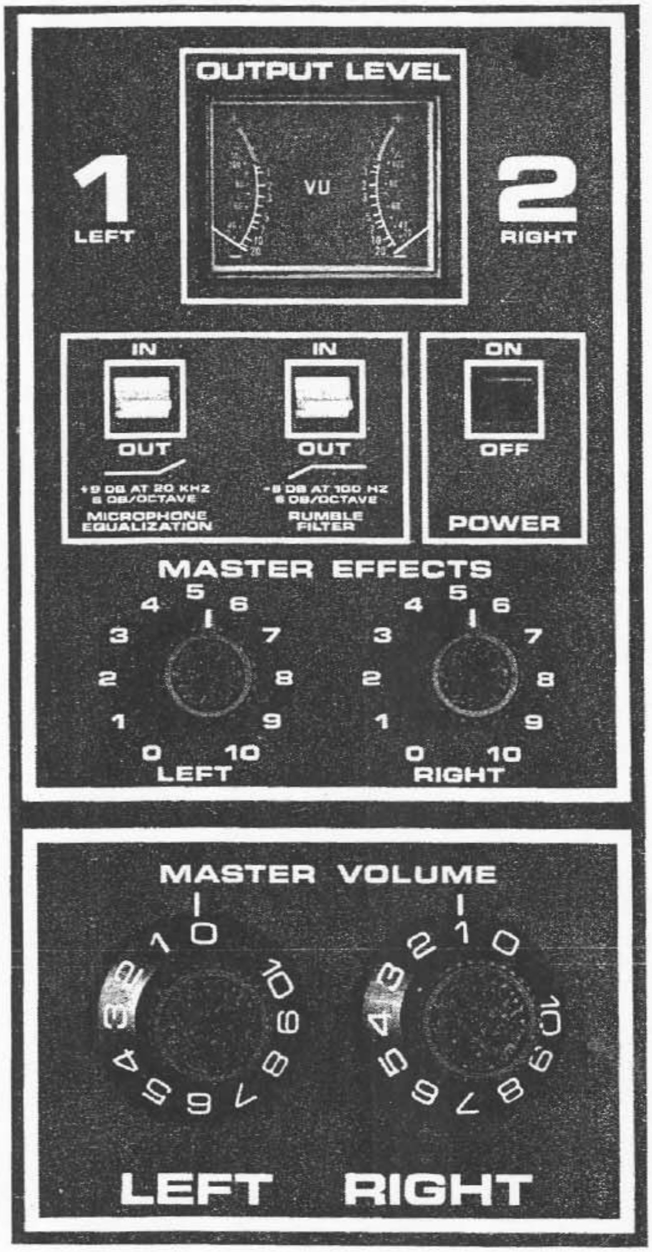
**VOLUME** sets the loudness of the input channel. But, unlike conventional mixers, Tapco's AutoPad volume controls vary the gain (sensitivity) of the preamp itself. When the volume is turned way up, the preamp is extremely sensitive to even the quietest sounds picked up by the microphone. With the volume set around "2" the preamp can handle the very large signals produced by a mike in extremely high sound pressure level situations — like when a vocalist "eats" the mike, for instance. We call this type of volume control AutoPad because it eliminates the need for the switchable pads that are used in conventional mixers to protect the input preamp from overload distortion. Most mixers use faders to regulate loudness. As the term implies, a fader simply reduces the output of the preamp as it is fed to the mix amp. Unfortunately, faders come *after* the input preamp where they can have absolutely no effect on overload distortion caused by hot input levels. With AutoPad volume controls, input overload distortion can be eliminated by just turning down the volume a little.

When you mix with a conventional mixer, the faders are set high at first, then brought down to create the the proper balance. This causes a loss of usable headroom, and impairs overall signal to noise ratio. With a gain-controlled mixing system like AutoPad, the volume controls are set low at first, then fine tuned to to create the perfect mix. In this way you end up using as little gain as practical for any given situation. This gives you the greatest amount of headroom, and optimizes signal to noise ratio for any input level. More headroom = cleaner sound. Lower noise = more usable dynamic range.

Remember that the mix amp is always going to be a little noisier than the input preamp — it's the nature of the beast. This means you should always try to have the input volumes on your 6200A set a little higher than the master volumes. This will allow the super-quiet preamps to do the actual work of boosting the signal strength to a usable level, while the mix amps do no more than combine the signals from all the input preamps. When this is done, any noise generated in the mixing section of the 6200A will be minimized. In actual use situations, S/N ratios in excess of 80dB are easily attained with the 6200A.

FOR A COMPLETE SEMI-TECHNICAL DESCRIPTION OF AutoPad® VOLUME CONTROLS READ SECTION EIGHT OF YOUR OWNER'S MANUAL. IT IS MOST IMPORTANT THAT YOU UNDERSTAND AutoPad® VOLUME CONTROLS TO COME UP WITH THE HIGHEST QUALITY SOUND REPRODUCTION.

## 4. Output Control Functions




**MASTER VOLUME** controls set the overall volume for each output channel. These are AutoPad controls — if you hear distortion in one of the outputs, just turn down the volume a little. The master volume controls will ordinarily register between 3 and 5.

When mixers are stacked the master volumes become submasters, controlling only the six inputs they would normally govern if the mixers were not stacked. The signal from the stacked mixer is fed into the system *after* the master volume control, so only its associated master volume will change its level. You can see this more easily in the block diagram.

**OUTPUT LEVEL** meters give you a constant stereo readout of the output levels. The meters have calibrating trim pots inside the mixer, behind the protective insulator that covers the highly lethal power supply board. Of course, it's only dangerous if you mess around inside the mixer without first disconnecting the AC cord from the wall outlet. You wouldn't do that, would you? At the factory the meters are set for OVU = +4dBm. This is the standard reference level used in recording studios in the United States, so the 6200A will match all the usual peripheral equipment. *Do not attempt to change the meter calibration unless you are technically competent to do so!*

**POWER** turns the mixer on. Or off.

**MICROPHONE EQUALIZATION** is a special circuit that boosts the extreme high end frequency response. The actual amount of boost provided by the Mic. EQ switch is shown in the graph on the specifications page. This gentle high end boost is used to help correct deficiencies in microphones and speakers. It is a much different sounding EQ from the regular channel treble controls, and can be used in conjunction with those controls to get just the right sounding high frequency response. Try a few different combinations using both the regular treble controls and the Mic. EQ switch. The Mic. EQ allows you to get more high end without boosting the mid-range as much as the treble controls. You might find the Microphone Equalization useful for brightening up some of those old, dull, faded mikes you've got hanging around.

**RUMBLE FILTERS** are provided at the outputs to eliminate that non-musical mess that often accumulates in the lower end of the sound spectrum. The Rumble Filters roll off the extreme low end without disturbing the more noticeable mid-bass sounds. This allows you to retain a decent bass sound while reducing unwanted noise. A few examples of problems that can be remedied with the Rumble Filters are: "popping" caused by close-talking vocalists, floor rumble from drums, over powering bass, and those annoying thumps produced by thoughtless toe-tapping musicians. The regular bass controls and the rumble filters can be quite complimentary, and should be used together to tailor the low end response.

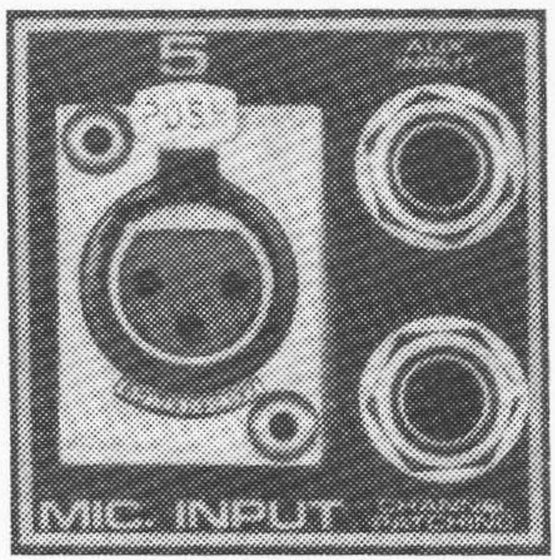
**MASTER EFFECTS** controls determine the loudness of any signal being fed into the Effects Return jacks. When the Effects Send is used to feed a reverb system, for example, the output(s) of the reverb would be fed back into the mixer through the Effects Return jacks. The overall amount of reverb in the mixer's outputs would then be set by the Master Effects controls. The same holds for any external processing device (phase shifter, tape echo, etc.).

When the Effects Return jacks are used for line inputs from another source, say a tape recorder, the loudness of that signal would likewise be determined by the Master Effects controls. Because the Master Effects controls utilize AutoPad circuitry, they will work with virtually any signal processing device.

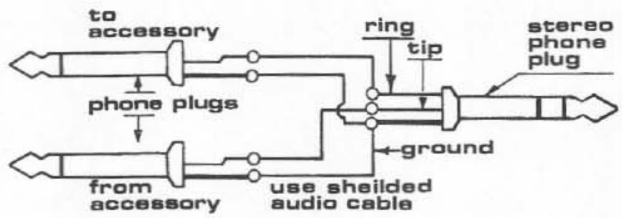
## 5. Input Channel Connectors

**AUX. INPUT** — A ¼" phone jack input for high level signals. Any signal that has already been run through a preamp should be fed into the mixer via the Aux. Input. This includes tape recorder, turntable preamp, and tuner outputs. The microphone input is automatically switched out of the circuit when a plug is inserted into the Aux. Input. This prevents any noise generated by the microphone, or the first amplifier stage, from degrading the signal being fed into the *Aux. Input.*

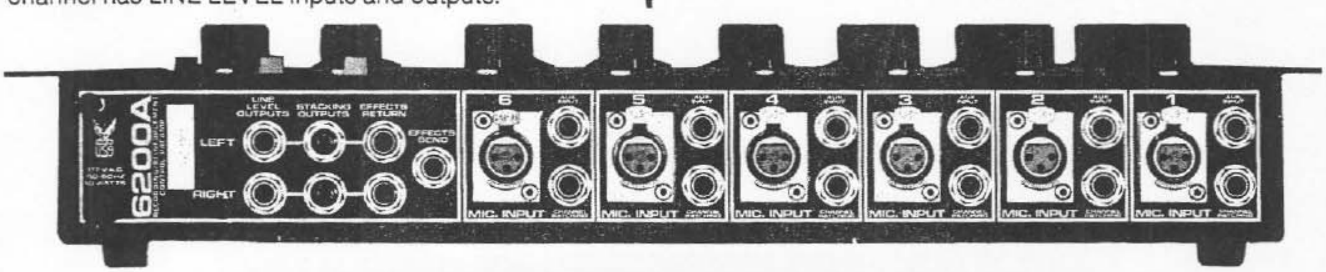
**MIC. INPUT** — A standard 3-pin XLR connector is provided for unbalanced Low-Z mike inputs. Because the 6200A is designed for professional use, no provision is made for the use of High-Z microphones. The mike input will accept input levels as high as —5dBm. The rated output level of most Low-Z mikes is between —50dBm and —70dBm. As you can see, this leaves the 6200A with at least 45dB of headroom in most cases. Microphones can be left connected even when the Aux. Input is in use.




**CHANNEL PATCHING** — This clever little, space saving (two-connectors-in-one) jack allows you to add more signal processing equipment to a single channel without disturbing any other channel. A single 3-conductor (stereo) phone plug is used for both the output from the preamp, and the return to the mixing stages from the external device. The diagram indicates how a channel patching cord should be wired — but if you're not into making your own, we have them available (check the Accessories section).



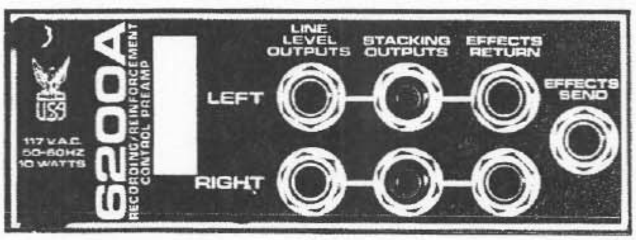

The channel patching jack can be used with equalizers, limiters, compressors, reverb systems and the like. Just be sure the device you want to patch into the channel has LINE LEVEL inputs and outputs.




SEE THE "HOW TO USE THE 6200A" SECTION FOR MORE WAYS TO USE THE INPUT SECTION CONNECTORS FOR OTHER INPUT/OUTPUT FUNCTIONS.

## 6. Output Section Connectors

**LINE LEVEL OUTPUTS** — These ¼" phone jacks are the mixer's main outputs. They are used to feed signals from the mixer to any high level input. This includes tape recorders, power amplifiers, reverbs, limiters, equalizers and the line inputs of other mixers. The line level outputs are capable of up to 10 volts RMS into impedances greater than 2K ohms, but the actual level of the output signals is determined by the master volume controls. Because the impedance of the line level outputs is low, the 6200A will automatically work with virtually any audio device in existance. The line drivers that supply signal to the main outputs are like little power amps. And like most power amps, they are short circuit proof. Don't be afraid to see if the 6200A's output will work with some other piece of equipment — the best way to find out is to plug it in and try it! But don't *ever* plug the output of the mixer into the output of anything else. Like a power amp, for instance. This would definitely put the 6200A out of commission for a while.



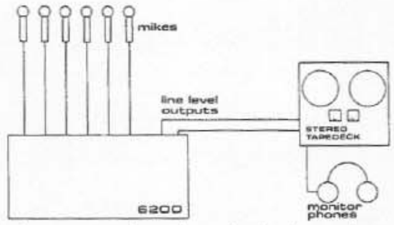

**STACKING OUTPUTS** — When more than 6 input channels are needed, two mixers can be paralleled by connecting these phono jacks together. Or, if you want to run the 6200A as a mono mixer with two independent outputs, connect the stacking outputs of one mixer. Notice on the block diagram that the stacking outputs connect the mixers after their respective master volume controls, but before the line drivers. This puts the signals from *both* mixers on *both* sets of outputs, so you need only connect one pair of outputs to the tape recorder or power amp.

**EFFECTS SEND** — The signal levels set by the individual channel effects controls are fed out of the mixer through the effects send jack. This output may be used to drive any external system you may want to use with the 6200A. This includes reverb systems, tape echo machines, phase shifters, power amps etc. If the effects send is used to feed an external effects system (like reverb), the output of that system should be fed back into the 6200A through the effects return jacks.

**EFFECTS RETURN** — These ¼" phone plug connectors are used to feed signals directly to the left and right output channels. The input impedance of the effects return circuits will match all the common effects boxes. This includes Echoplex, reverbs, phase shifters, digital delays, etc. If you want to feed the output of a mono device into both output channels, use a "Y" cord to make two outputs out of the one. Then feed both of those outputs to the effects return jacks.

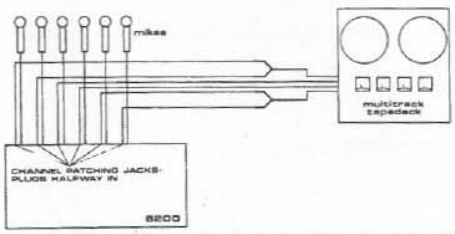
# 7. How To Use The 6200A

## STEREO RECORDING



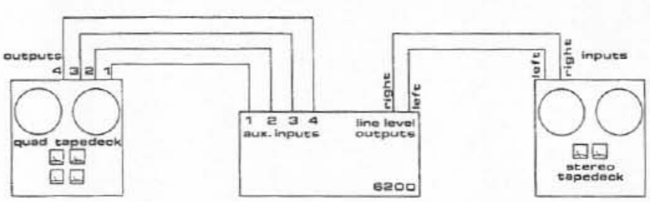

As illustrated below, the outputs of the 6200A are run directly to the line (or high level) inputs of the tape recorder. Set the controls on the recorder so "O" VU occurs at the same level on both the mixer and the tape machine. For playback you may connect the recorder's line outputs to 1) channel aux. inputs 2) effects return jacks 3) the stacking outputs. If you use the effects return jacks part of the recorded signal can be fed back into the mixer during recording for tape echo.

## MULTI—TRACK RECORDING



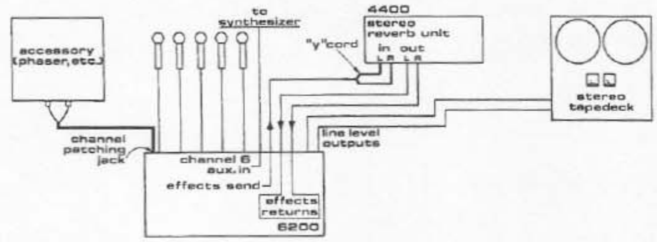

Use the channel patching jacks for individual channel outputs, as shown in the diagram. A regular two conductor (mono) phone plug is inserted *halfway* into the channel patching connector. This allows the signal to feed out of the mixer without interrupting the circuit. Of course, if the plug is inserted all the way into the connector, the circuit will be interrupted. The output impedance at this point is low and the signal level is high. This means you can connect this output to *any* high level input. The signal obtained at the patching connector is subject to the setting of the input channel controls, with the exception of Effect. The outputs from two channel patching jacks may be Y-corded together for recording on one track, with no loss of quality. However, this will render the pan pots on the connected channels inoperative.

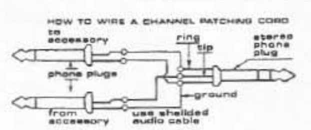
## MULTI-TRACK MIX DOWN

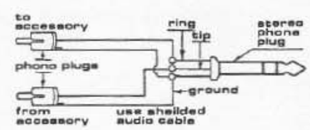



Once you have your tracks recorded, you gotta mix 'em down to stereo, right? Easily done. Just run the line outputs from the multi-track machine into the aux. inputs on the 6200A. The diagram assumes you want the final recording to be stereo. If you only need a mono recording, use one output from the mixer and pan all the channels you're using to that output. If you need to add reverb or echo or phasing (or anything), use the effects send/return loop. There's more detail on the use of this system in section "E". To process only one track from the tape machine you can simply insert the processing device in the line between the recorder and the mixer. When you're mixing down, set the output levels on the recorder so the mixer's volume controls can be run as high as 6 (if necessary) without distortion. Don't be afraid to try mixing down over and over. Careful mixing will produce professional results.

## CHANNEL PATCHING




**For Musical Instrument Equipment** **For Hi-Fi Equipment**

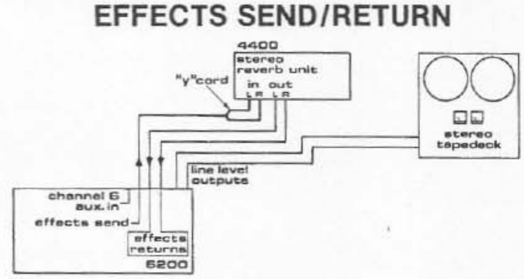
The individual channel patching jacks on the 6200A are one of the features that make this mixer really versatile. The diagram shows the "normal" use of these jacks — external signal processing. A special cord, wired as shown, performs both the output *and* input functions.

Notice on the block diagram that the patching jack incorporates a switch. When a plug is fully inserted into the jack this switch automatically interrupts the circuit. The signal flows out of the mixer (through the "ring" of the plug) instead of continuing on its usual path.

If nothing more were done with the signal this channel would not be heard at the output. But, when the signal is sent through another processing device, like in the drawing, it can be returned to the usual signal path through the "tip" of the plug on the patch cord. With this out/in link you can insert nearly any signal processor(s) in an input channel.

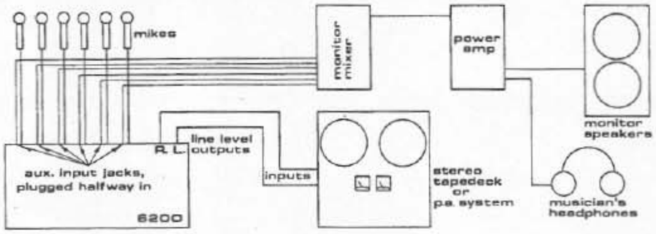
These specially wired patch cords are available from Tapco if you are unable to find them locally. Check the Accessories section in your Owner's Manual.

## EFFECTS SEND/RETURN



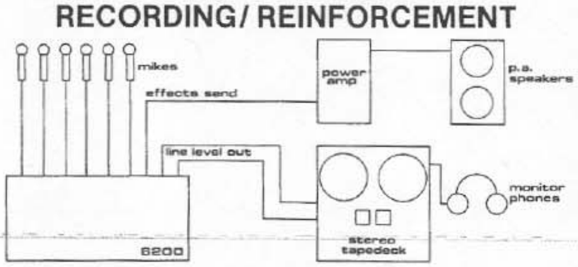

The Effects controls on each channel provide an altogether separate mix at the rear panel Effects Send jack. One of the ways this system might be used is to add reverb during recording. The output of the reverb system is fed into the mixer via the Effects Return jacks. The drawing shows how this might be accomplished using Tapco's 4400 Reverberation System. In actuality, the 4400 in the drawing could be almost any reverb, phase shifter, what have you.

## CUE OR MONITOR OUTPUTS



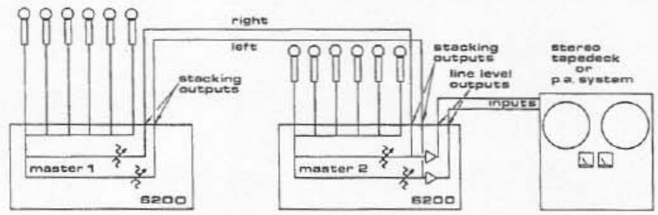

Many situations require individual channel outputs that are not affected by the regular volume, bass and treble controls. Such an output is available from the 6200A at the *Aux. Input* jacks! These inputs can be used as outputs by inserting a plug only halfway into the jack. Referring to the block diagram, notice that inserting a two conductor (mono) phone plug into the Aux. Input will short the first stage to ground, effectively shutting it off. But, if the plug is inserted only halfway, it will make contact with the first stage without shutting it off. This allows the signal from the first stage to be taken out of the mixer without disturbing the normal functions within the unit. The signal obtained from the Aux. Input jack in this manner will be low level (approx. —30dBm), and will require a *high impedance load.* This signal point must not be loaded with anything less than 10,000 ohms. Almost any High-Z mixer (like the Tapco 100K Keyboard Mixer, for instance) could be used for a control system for the cue/monitor outputs. Or, a single High-Z input preamp with a six position switch could provide selectable cue/monitor.

## SIMULTANEOUS RECORDING/ REINFORCEMENT




The 6200A is ideally suited to this application. The mixer and tape recorder are connected as usual. The reinforcement (p.a.) signal is derived from the Effects Send. The mix on the p.a. will then be determined by the individual channel Effects controls. This allows you to have independent mixes for both the recording and the p.a. If stereo p.a. is needed, the reinforcement power amps can be driven from the stacking outputs on the 6200A. The recording mix is monitored on headphones through the tape machine, while the p.a. mix is heard on the speakers.

## STACKING




All Tapco mixers have a special output that allows you to parallel units without tying up any input channels. The Stacking connectors are simply wired together with a standard male phono to male phono patch cord. Again referring to the block diagram, you can see that this will connect the mixers between their mix amps and line drivers. That means that signals from both mixers will appear on both sets of outputs, so you need connect only one set of outputs to the power amp or tape recorder. This hookup is illustrated above.

# 8. References

## AutoPad® Volume Controls

**(how they give you that clean quiet sound)**

We are assuming that if you've even begun to read this page you must be interested in finding out what really makes the 6200A tick. Every attempt has been made to keep this explanation brief. Of course, that means that we've jammed a tremendous amount of information into very few words, so you'll probably have to run through it a couple of times before it all sinks in. The basic concepts are actually very simple, even though the explanation is a bit thick. But rest assured, once you understand the operation of gain-controlled mixing systems, you'll know more about mixer design than some of the people who build them.

The two most basic requirements of professional quality microphone mixers, regardless of price, are that they be *quiet* and *distortion free*. For good signal to noise ratio the circuitry itself must be inherently quiet, and you must be able to match the gain to the signal level. And for truly distortion free reproduction, maximum headroom must be maintained at all times. This is why Tapco engineers designed the AutoPad volume control.

AutoPad volume controls work much differently from conventional volume controls. Conventional volume controls are usually called faders, because that's what they do. In the conventional mixing system the fader comes after the preamp, where it regulates loudness by reducing the signal level that goes to the mix amp. The conventional input preamp operates at a fixed gain no matter what the input level. When the signals are too loud, distortion occurs, and when they're too soft signal to noise ratio suffers. AutoPad controls, on the other hand, vary the gain (sensitivity) of the preamp itself — that is, they regulate loudness by changing *how much the circuit amplifies*. This means you can optimize the performance of the preamp for any signal level, eliminating distortion and noise problems. This is how it works:

All electronic circuits generate noise, and mix amps are, by their very nature, noisier than input preamps. Now, there are two general rules that apply: 1) The greater the difference between the amplifier's self generated noise, and the input level "seen" by the amplifier, the greater the resulting signal to noise ratio. 2) Any circuit that amplifies input signals must also amplify its own noise.

So, if this amplifier makes its own noise louder, it stands to reason we can minimize noise by keeping the gain as low as possible, right? For example, if the amplifier generates 1 unit of noise, and has a gain (multiplication factor) of 10, then the noise at the output of the amp will be 10 units. If the gain were only 5, the output noise would drop to 5 units.

As the term implies, signal to noise ratio is determined by the difference between the signal level *in* the amplifier, and the noise generated *by* the amplifier. If the amplifier's internal noise has a value of 1, and the maximum signal the amplifier can handle is 1000, the S/N ratio is 1000:1 (60dB). It becomes obvious then that if the amplifier could handle more signal, its signal to noise ratio would be better.

The amount of signal an amplifier can handle is predetermined by the *gain*, and the *maximum output capability*, of the amplifier. Unfortunately, the maximum output capability is governed by the power supply voltage, so it cannot be changed. However, the gain can be changed very easily. So, what happens when we change the gain? Suppose the maximum output capability of an amplifier is 10, and the gain is 10. The maximum input level must be 1 (Fig. 1A). When that input level goes beyond 1, the amplifier is overloaded — and we all know what that sounds like! (Fig. 1B). But what

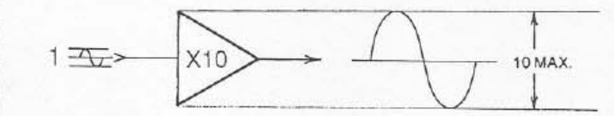



**Fig. 1A**
With a gain of 10, the maximum input level is 1.

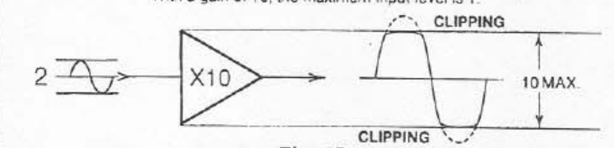



**Fig. 1B**
When the input level exceeds 1, the output "clips" because the amplifier cannot put out more than 10.

if the gain were lowered to 5? Then the amplifier could handle an input level as high as 2 before the output level tried to go beyond its maximum (Fig. 2A). And if the gain were lowered to 1 (unity), the maximum input level would be 10 (Fig. 2B).

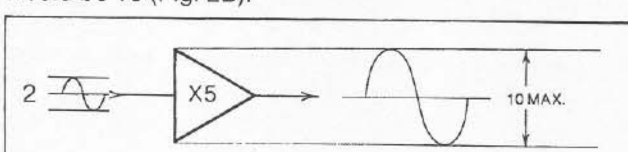



**Fig. 2A**
Lowering gain to 5 allows preamp to handle an input level of 2.

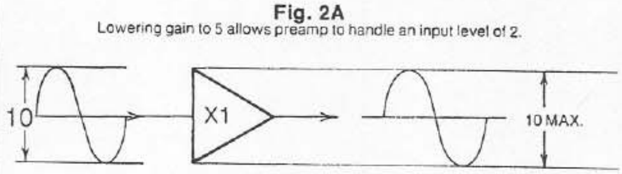



**Fig. 2B**
Further lowering gain to 1 allows preamp to handle an input in level of 10.

Now, if you understand that bit of electronic theorizing, you can see that if you can control the gain of an amplifier, you really *can* optimize the performance of that amplifier for many different situations. But how does this affect the operation of a mike mixer? A quick comparison of a gain-controlled mixer system and a conventional mixer will serve to illusrate the differences.

To be fair, we'll assume that all parameters of the preamps and mix amps shown in the drawings are exactly the same. Only the volume controls have been changed to protect the innocent. And we do mean protect, as you'll soon see.

The idea is to optimize S/N ratio and minimize distortion, right? OK. The maximum output capability of these circuits is 10, and the gain is set, for the moment, at 10 in both input circuits. Gain of the mix amps is 1.

We all know that volume controls are usually not run wide open. In fact, most manufacturers of conventional mixers recommend that the faders be operated at about 70% of maximum. This is because it is always necessary to have some control-ability above and below the "normal" volume setting, so the operator can accommodate the ordinary dynamic changes of the music. So, if the preamp gain is 10, and the maximum output is 10, we already know that the maximum input is 1. Look what happens to this output, though, when it leaves the conventional preamp. *The fader reduces its level by at least 30%!* (Fig. 3A) Remember now: Mix

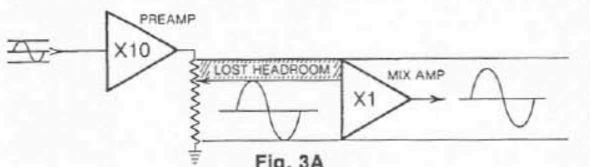



**Fig. 3A**
Conventional mixer reduces headroom, S/N ratio or both.

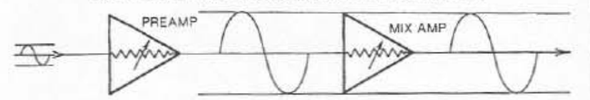



**Fig. 3B**
AutoPad's unobstructed output provides maximum headroom, S/N ratio for any input level.

amps are inherently much noisier than preamps. So, to come up with a quiet signal we have to cram as much real level into that mix amp as possible. It would somehow seem self defeating to reduce the signal strength before it can get to the mix amp, wouldn't it? There are two solutions. We can increase the gain of either the preamp, or the mix amp, by 30% to make up for the losses in the fader. But, if we *increase* the preamp gain 30%, we have to *reduce* the maximum input level by 30% to stay within the maximum output limit of the preamp. THAT MEANS WE'VE JUST SACRIFICED 30% OF OUR AVAILABLE HEADROOM TO THAT FADER. If we increase the gain of the relatively noisey mix amp we simultaneously increase the amount of noise at the output, because every amplifier amplifies its own noise as well as the signal. THAT MEANS THE OVERALL SIGNAL TO NOISE RATIO IS REDUCED 30%. With the AutoPad volume control, the full output of the preamp is available to the mix amp, with no losses. Right away you can see that the gain control provides maximum headroom, because the AutoPad preamp is still able to handle an input of 1, while the maximum level that the conventional preamp can handle has been reduced by 30%. S/N ratio is automatically optimized too, because the signal is amplified by the quieter preamp and merely passed along by the mixing stage.

All good and well for an input level of 1, but what happens when that actual input level drops to .001? Well, the conventional preamp does what it can, throws away 30%, and passes the result along to the mix amp. In this case it's likely that there's not enough signal left to disguise the noise generated by the mix amp, so the overall output is pretty noisey. In the same situation, the AutoPad preamp's gain could be increased to as much as 1000. The increased gain would allow the mix amp to "see" enough signal to perform its very necessary function quietly.

By the same token, what happens to these two preamp systems when the signal level goes up — to 2, for instance. With the conventional preamp, a pad must be used to protect the input from overload. A pad is a very simple resistive circuit that performs essentially the same function as a fader: it reduces the signal level. Of course, with the AutoPad preamp we'd simply turn down the gain a little, allowing the preamp to handle the increased signal level. So what's wrong with a pad, anyway? Nothing, really. Pads work just fine, for their intended purpose. We already know that the more signal you can get into the front end of that preamp, the better the S/N ratio will be. If you put in a 10dB pad, you've actually thrown away 10dB of S/N ratio. Now, the theory behind a pad is that your signal level should be 10dB too hot before you switch in that pad. Unfortunately, however, in real life signals don't come packaged in nice neat 10dB increments. That's why AutoPad volume controls were made infinitely variable — so you can precisely match the preamp gain to any signal level. The only other drawback of the pad is that they must be, by design, switchable. This means that pads are able to make only abrupt volume changes. Because AutoPad controls are continuously variable, they are able to make smooth volume changes, allowing you to create true dynamics without fear of overload distortion.

Tapco's AutoPad volume controls have solved the seemingly unsolvable problems of noise and distortion, without great expense. With AutoPad you get the best performance under all conditions.

## Where to get more information

Modern Recording Techniques by Robert Runstein, published by Howard W. Sams Co. *The* complete modern text book of studio recording techniques.

Modern Recording is a bi-monthly magazine from the Recording Institute of America. MR covers equipment techniques, actual studio sessions, etc., This magazine is highly recommended to anyone interested in recording.

Modern Recording
Recording Institute Publishing Inc.
15 Columbus Circle
New York, N.Y. 10023

Recording Engineer/Producer is another very good bi-monthly magazine dealing with the recording arts.

Recording Engineer/Producer
P.O. Box 2449
Hollywood, CA 90028

Basic Audio by Norman Crowhurst, is available from the John F. Rider Publishing Company. This book is just what the title implies.

Microphone Primer by Jim Long available from Electro-Voice, 600 Cecil Street, Buchanan, Mich. 49107. The basic guide to microphones.

The following are available from:

Sagamore Publishing Co. Inc.
1120 Old Country Road
Plainview, N.Y. 11803

DB Magazine — an authoritative, well known magazine, dealing with recording, sound reinforcement and general audio topics. Monthly.

Microphones: Design and Application by Lou Burroghs. The author was one of the two original founders of Electro-Voice, Inc. He is responsible for a great deal of today's accepted microphone theory and design. The book is a practical, non-theoretical reference manual for anyone in the audio industry.

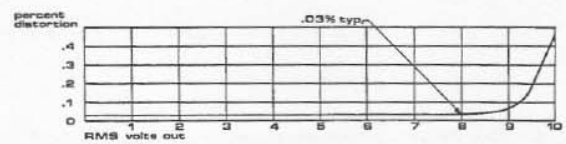
The Technique of the Sound Studio by Alec Nisbett. This is a handbook on radio and recording techniques, but the principles described are equally applicable to film and television sound. 264 pages; 60 diagrams; glossary, indexed.

Modern Sound Reproduction by Harry F. Olson. This basic text covers amplifiers, microphones, loudspeakers, earphones, tape systems, film sound, tv and sound reinforcement — the significant elements and systems of modern sound reproduction. Employs simple physical explanations which are easily understood without special engineering training. 328 pages.

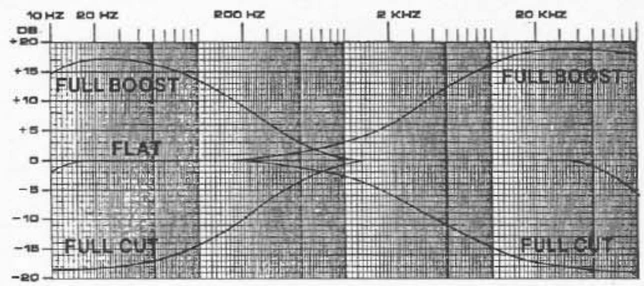
We might suggest that instead of investing your hard earned bucks in any of these publications on our word alone, go down to your library and look them over first. That much, at least, is free.

## 9. 6200A Specifications

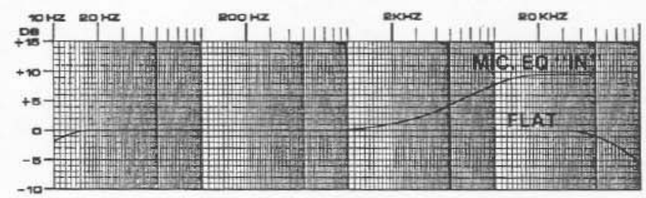
| | |
|---|---|
| GENERAL | |
| Frequency response | 15Hz to 40KHz ± 1dB |
| Harmonic distortion | < .1% |
| Equivalent input noise | —126dBV (20Hz-20KHz) |
| Signal to noise ratio (—50 dBM reference | 74dB |
| Output noise | 20 µV |
| Total available gain | 96 dB |
| Line output capability | 10 volts RMS at 2000 ohms + 13 dBm @600 ohms |
| Line Output impedance | 10 ohms |
| Output meters | adjustable |
| Microphone EQ | + 9 dB @20 KHz |
| Rumble filter | —6 dB @100 Hz 6 dB/octave |
| Power consumption | 10 watts |
| Weight | 12 lbs. |
| EFFECTS SYSTEM | |
| Effects send capability | 10 volts RMS + 13 dBm @ 600ohms |
| Effects output impedance | 10 ohms |
| Effects return capability | up to 10 volts RMS |
| Effects return impedance | 60 K ohms |
| INPUT CHANNELS | |
| Bass & treble action | ± 18 dB |
| Microphone impedance matching | 30-600 ohms |
| CHANNEL PATCHING | |
| Output capability | 10 volts RMS |
| Output impedance | 220 ohms |
| Return impedance | 12 K ohms |



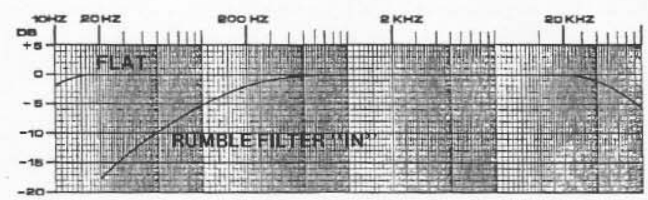
**Typical Distortion Curve**



**Bass and Treble Curves**



**Microphone Equalization**



**Rumble Filter**

## 10. Accessories

Channel patching cord — for
musical instrument equipment
Tapco part number **910002**

Channel patching cord — for
hi-fi type equipment
Tapco part number **910001**

The Box rack mounted carrying case (holds
TAPCO mixer, equalizer and reverb in one box)
Tapco part number **910100**