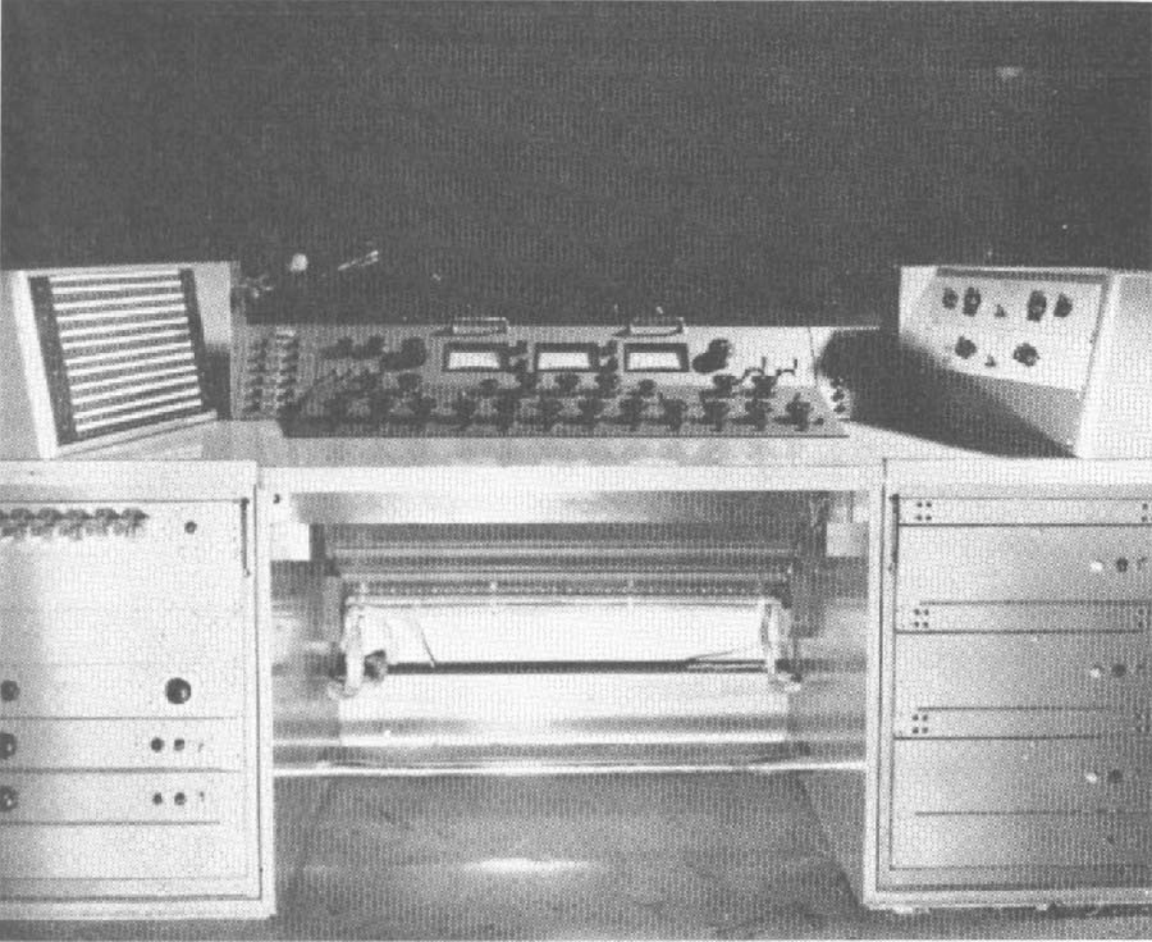


FIG. 4. View of control room.

usual shape of the room. Three speakers are mounted symmetrically as shown in Fig. 4. All three are used for three-track monitoring; the outside two speakers are used for two-track, and the center is used when recording monophonic only.

Considerable time was spent in the determination and correction of the over-all acoustic response of the monitoring system in these control rooms using both selected bandwidth of random noise and a special warble tape which provided a frequency variation of $\pm 10\%$. The warble tape provides discreet frequencies in a band from 30 to 15,000 cps.

Figure 5 shows the over-all acoustic response of the monitor system in the control room for the left, right, and middle speakers.

CONTROL CONSOLE

The requirements of a recording studio control console are considerably more complex by virtue of having to record three channel, two channel, and monophonically as well as the necessity for equalizing each channel individually and providing an independent control of reverberation on each channel. The control console described here is manufactured by Universal Audio, Inc. of Hollywood, and the block diagram is shown in Fig. 6 and pictured in Fig. 7.

Microphone Input Channels

There are twelve microphone input channels—normally without patching. The input impedance is 30-50 or 150-250 ohms, balanced. Each input channel may be switched to any of three mixer busses (three position key)—left, middle, and right—or it may be patched to a fourth program mixer buss. Each microphone input channel provides equalization facilities using the 100D preamplifier in conjunction with the 100D equalizer located six to the left and six to the right of the main control panel. Low frequency provides flat, low roll, +3, +6, or +9 db boost at 50 cps. High frequency provides HF roll –3 or –6 and +3, +6, +9 db boost which may be switched for either 5 or 10 kc. There are four high level input channels; channels 9, 10, 11, and 12 may be switched to 5 kohms bridging input line level.

Program Output Channels

There are four program output channels with provisions for two or three channel stereo. Left, middle, right, plus combined mixed for monophonic or fourth channel may be independent by patching from selected input channels. Output impedance of each buss is 150 or 600 ohms, balanced.

The feed to the mono network is provided after the submaster in each program channel by N4 (Fig. 6) which is a bridge isolation network providing 1 db loss in one direction and 60 to 70 db of isolation from the other two terminals. The variable resistance is used to balance and correct for variation in input impedance of the following program amplifier.

Echo Send Channels

There are three echo send channels. Two send channels (left, right) are normaled to booster amplifiers to feed approximate line level at 150 or 600 ohms, balanced. The third channel (middle) may be patched to a booster amplifier to feed the third echo driver buss. Each of the twelve microphone input channels is provided with an echo "send" key and individual "send" attenuators to accomplish independent control of the echo "send" from each position. The echo send buss is automatically selected as the left-right program key is thrown. Left and right echo "send" busses are provided with a vu meter.

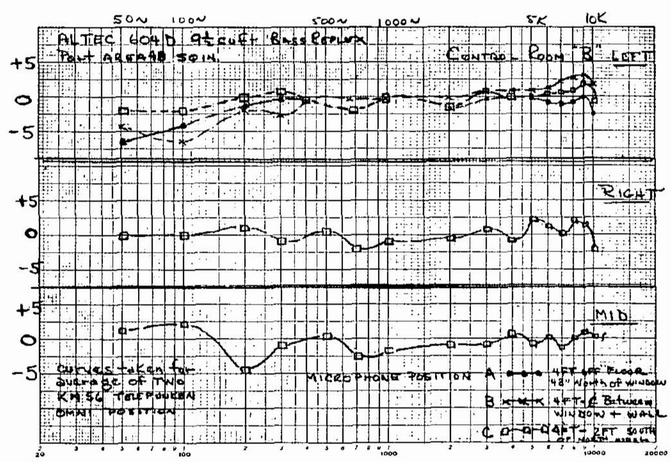


FIG. 5. Measured frequency response curves of the three monitor speakers in the control room of Studio B.

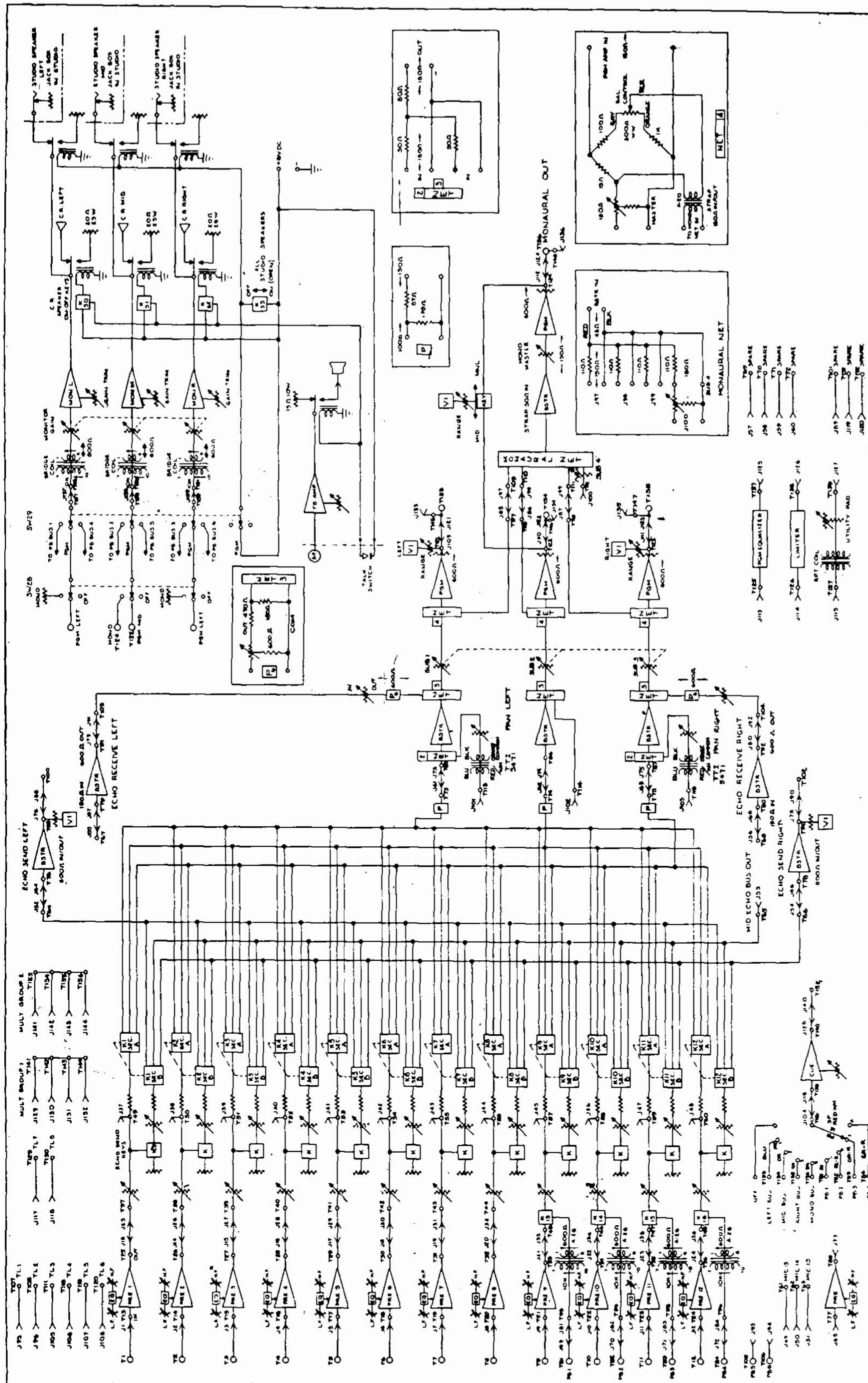


FIG. 6. Block diagram of the control console.

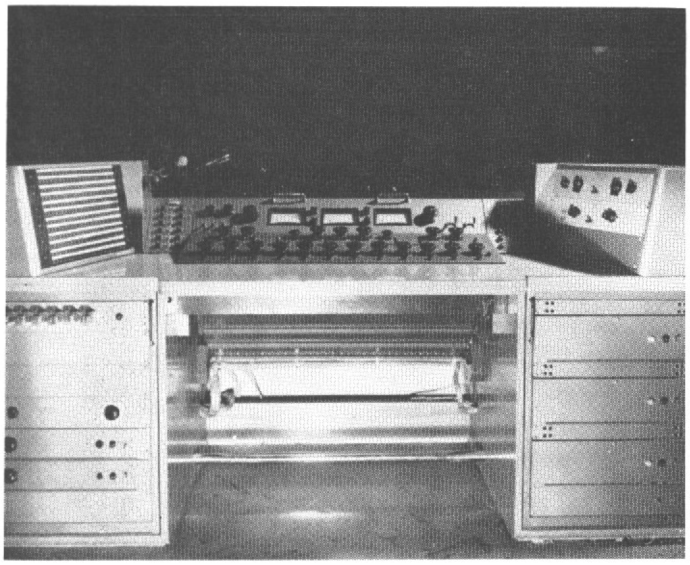


FIG. 7. Control console.

Echo Receive Channels

There are normally two echo receive channels. However, a third echo receive position is available by patching into the "pan" pot input middle channel from a third utility booster amplifier. The two echo receive channels are provided with booster amplifiers normaled to a mixer net in the left and right program busses. Input impedance to echo receive booster amplifier input is 50–150 or 600 ohms, balanced.

Special Mixing Facilities

Pan pots are provided to feed into each of the two stereo program busses (left and right). These pan pot inputs are 5 kohms bridging inputs so that any one of the 12 microphone input channels may be bridged and fed into the opposite program channels for producing special effects and spatial repositioning of each input channel. The monophonic program mixer network has a fourth high level input position for facilitating even greater flexibility of operation.

A unique feature is that the three submaster controls (left, middle, and right) may alternately be used as a ganged master to control the three channels simultaneously or be operated independently. This is accomplished by a ganging mechanism which is instantaneously engaged or disengaged with the turning of a small knob adjacent to these faders.

VU Metering Facilities

There are three program vu meters with +4, +8, and +12 range switch plus transfer switch for third meter to monophonic channel. The two echo drive vu meters indicate relative level to each echo driver.

Monitoring Facilities

Three 60-w amplifiers are provided for monitoring. Interlocked switching provides for convenient selection of monophonic monitoring or two or three channel stereo, without patching. In the monophonic position, the center monitor channel may be switched as follows: program, tape No. 1 playback or tape No. 2 playback. The left and right monitors are muted in this position. If the monitor selector is thrown in the stereo position, the monitor input selector switch provides for program, tape No. 1 playback or tape No. 2 playback on each channel. Keys are provided for disabling the studio speaker on each channel individually. A speaker relay panel actuated by the above controls provides the necessary muting for studio playback and control room muting for talkback.

Talkback Facilities

A 10-w talkback amplifier is provided to feed studio talkback speaker. A portion of the output of the talkback amplifier is bridged and fed to the program busses to provide "I.D." directly onto the tape. A separate gain control is provided to adjust the volume of the I.D. override circuit.

Monitor Cue Amplifier

An additional 10-w utility monitor is provided for use during dub-in's over musical backgrounds, etc., for feeding headphones or separate speakers to the studio.

Patch Panel Facilities

Twelve rows of double jacks (total 144 circuits) are arranged to provide access to every necessary equipment terminal or circuit. Each row has individual designation strips.

Power Supplies

Two identical power supplies are provided with instantaneous switching to either for uninterrupted service. Either power supply will operate the entire console at 70% of its normal rating at either 50 or 60 cps, 110-v input.

Over-All Electrical Specifications

Gain—microphone input to program line 94 ± 1 db. Frequency response: ±1 db, 20-20,000 cps (with equalizer in flat position). Distortion—less than 0.5% IM at +24 dbm output (40 cps and 7 kc 4:1). Noise—at least 65 db below +8 dbm output with an input level of –60 dbm.

One of the unique features of this console is that it is completely self-contained, requiring no interconnecting wiring to racks, etc. The complete installation of these consoles was accomplished in the control rooms in less than one day.

MICROPHONE PREAMPLIFIERS

Considerable emphasis was given to the preamplifier and equalizer design for the console in view of the more critical requirements of lower distortion, noise level, and higher output capabilities without overload. Figure 8 shows the schematic of the 100D preamplifier.

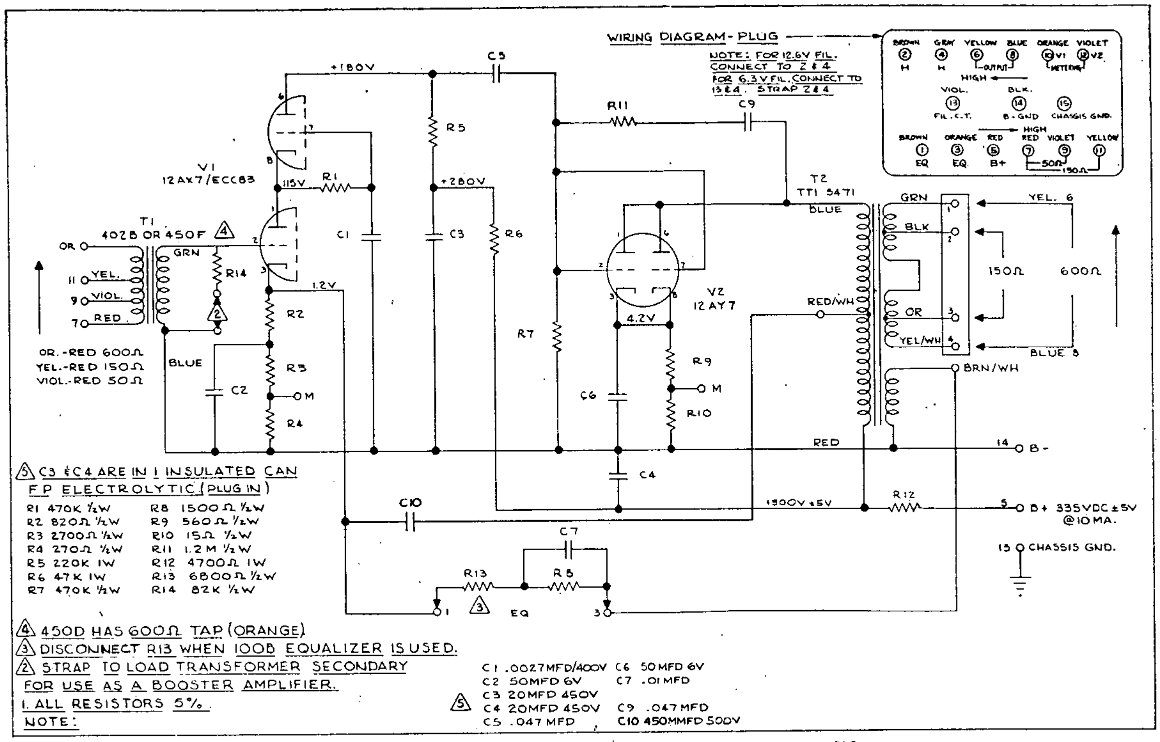


Fig. 8. Schematic of Model 100D microphone preamplifier.

Actually, very little change in the basic design of the studio microphone preamplifier has taken place in the last fifteen years with the exception of newer type tubes. With the new studies which have been presented showing the energy distribution characteristic of modern commercial recording, there is a definite need to improve the performance through new design concepts of such an important unit in the transmission chain.

The 100D preamplifier meets the following design objectives:

1. Power output capabilities of the preamplifier are at least +20 dbm at less than 0.5% IM.
2. The preamplifier is capable of handling a maximum input level of not less than −20 dbm.
3. Noise level is not greater than an equivalent input signal level of −125 dbm.
4. The design of the amplifier minimizes the necessity for critical selection of tubes.
5. The equivalent input impedance of the microphone transformers is high enough at the lowest operating frequency to prevent excessive loading of a ribbon microphone which would adversely affect the low-frequency response.
6. The amplifier incorporates variable equalization in a convenient flexible manner without loss of gain to provide boost or attenuation at both low and high frequencies.

The front end of the 100D is a cascode circuit using the ECC 83 and will provide a voltage gain of approximately that of a pentode stage. However, the equivalent input noise level has been found to be considerably lower than the pentode stage with random selection of tubes. The cascode amplifier was first described for audio use by Valley and Wallman.[1]

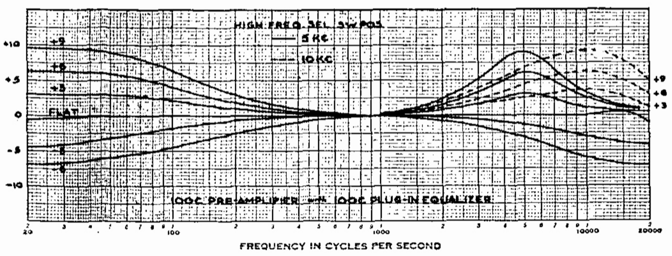
In many preamplifiers available today the input transformer is terminated in order to obtain flat frequency response when measured from a gain set. However, when the same preamplifier is driven by a reactive generator the frequency response may be entirely different. In view of the fact that we use a wide variety of microphones in which impedance varies widely as a function of frequency, it is necessary to design a preamplifier that will accommodate these microphones without affecting their performance.

In reviewing our design objective it was decided that we should incorporate variable equalization in the preamplifier circuit. The use of equalization in microphone channels is somewhat of a common practice in the recording studio, and when used with judicious care can be a useful tool to the recording engineer. Some of the reasons why equalization might be used in the microphone channels are as follows:

1. To complement the acoustics of a given studio.
2. To compensate for the deficiency of certain type microphones due to proximity or other location effects.
3. To attenuate certain frequency ranges which may be interfering with the desired sound source being picked up.

No attempt will be made here to argue the value of this technique; however, in view of the fact that it is being done,

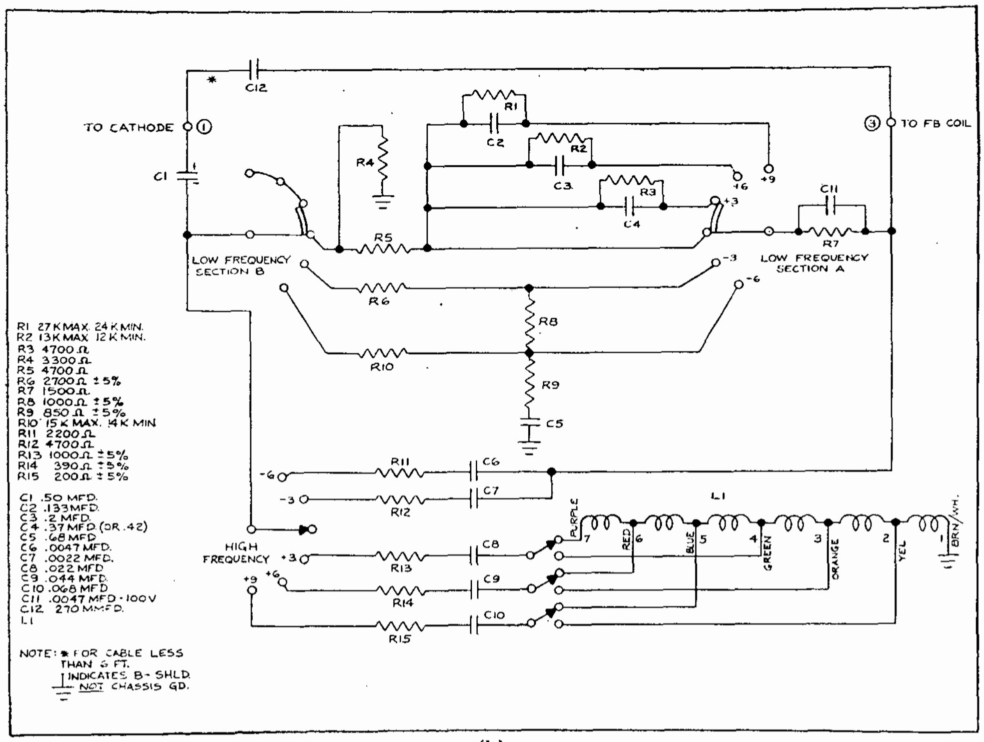
[1] G. E. Valley and H. Wallman, *Vacuum Tube Amplifiers* (McGraw-Hill Book Company, New York, 1948), p. 440.



(a)

FIG. 9(a). Frequency response curves of combination of preamplifier and equalizer.

we set about to design a more convenient, flexible, and economical means of accomplishing the desired results. In conventional practice the microphone channel consists of a preamplifier of 40 db gain followed by a passive equalization network with a minimum loss of approximately 12 db. This, then, means two things: (1) that the net gain of the channel with equalization is only 26 db, and (2) that the preamplifier is required to put out 12 db more power before the signal reaches the fader. The approach to this problem in the case of the 100D amplifier was that an environmental-



(b)

FIG. 9(b). Schematic of Model 108B equalizer.

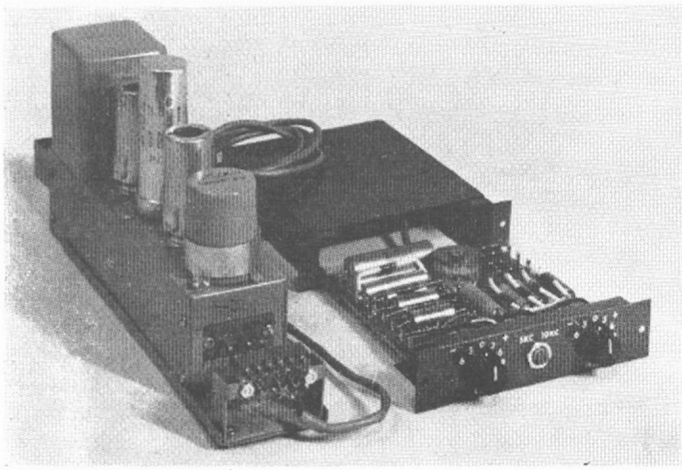


FIG. 10. Model 100D preamplifier and Model 100B plug-in equalizer package.

type equalizer was to be designed which would satisfy the following requirements:

A. The equalizer itself is a package unit, extremely small so that it could be panel mounted as part of either the control panel or immediately adjacent thereto.

B. That the preamplifier as a unit could be operated retaining all of its performance characteristics *with* or *without* this equalizer, by substituting one resistor for the equalizer package.

C. That the equalizer package itself should be designed so that it may be operated as far as 25 ft away from the preamplifier and that it could be connected by no more than three conductors. In this case a single pair of two conductor shield wire may be used.

D. Another important requirement being that the distortion characteristics of the preamplifier are such that in the boost or attenuate position the distortion would not be in

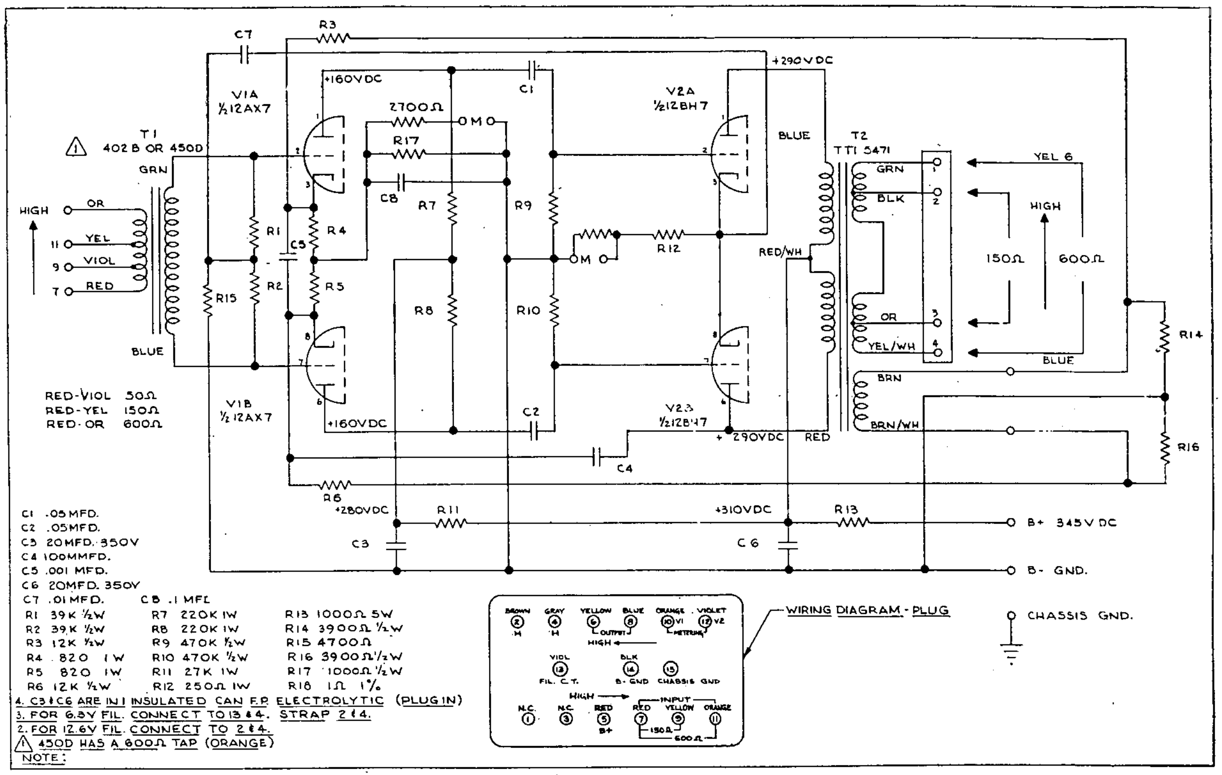


FIG. 11. Schematic of Model 101D program amplifier.

excess of ½ of 1% IM or 0.3 of 1% THD at an output level of +20 dbm.

Figure 9 shows the frequency response curves and the schematic of the Model 100B equalizer. Pictured in Fig. 10 is the preamplifier and plug-in equalizer package.

Aside from the fact that we have salvaged 12 db of gain, improved our power output characteristic, and reduced the package size of the equalizer, we have also accomplished another important feature; namely, that the cost of each equalized microphone channel is less than half that provided by the other systems.

It was determined that in order to incorporate variable equalization in the preamplifiers a minimum of 18 db of voltage feedback must be diverted to the *B* loop to facilitate the necessary equalizer performance. The *A* loop, which is the internal loop from the plate of the output stage, is primarily used to provide stability as well as keep the over-all distortion level extremely low, irrespective of the variable equalizer settings.

In view of the fact that it would be extremely difficult to incorporate variable equalizers in a push-pull amplifier and since the odd order harmonic distortion was reduced satisfactorily, it became necessary to use a parallel 12AY7 in the output stage in order to obtain the +20 dbm power output requirements.

The preamplifier may be used as a flat preamplifier or booster by connecting one resistor on the terminal board in place of the remote equalizer and will operate at the same performance characteristics. If the preamplifier is to be used as a booster a terminating resistor of proper value is connected across the primary which presents a solid terminal impedance to the incoming source. Specifications and performance curves may be seen from Fig. 9.

PROGRAM AMPLIFIER

Figure 11 shows the program amplifier 101D used in the console which incorporates an output stage consisting of a push-pull 12BH7. The circuit is straightforward incorporating two symmetrical feedback loops from the tertiary to the cathode of the input stage. An additional stabilizing network which corrects the distributed capacity unbalance in the transformers is represented by *C*4. In design it was found that in order to reduce the harmonic and intermodulation distortion to a satisfactory low level, some means of maintaining high-frequency dynamic balance with respect to the output stage was required. This is accomplished by sampling the unbalanced signal voltage across the unbypassed output cathode resistor and feeding it back in the proper magnitude to the center tap of the divider network across the secondary of the input transformer. In the case of a wide variation in the separate halves of the input 12AX7 and output 12BH7 tubes, a reduction of better than 5 to 1 in THD was obtained using this self-balancing loop.

The performance of the 101D program amplifier is as follows: Gain—42 db. Input impedance: designed to work from an input source of 30 to 60 ohms and 150 to 250 ohms, or 600 ohms (terminated input transformers). Output load impedance: 150 or 600 ohms, balanced. Due to the extremely low internal impedance, the change in output voltage from load to no load is negligible. Maximum output power: plus 30 dbm (1 w) at less than 1% IM (plus 28 dbm at less than 0.5%) using 40 cps and 7 kc 4:1. Total harmonic distortion at 1 w less than 0.5% 30 to 15,000 cps. Maximum input level: −12 dbm. Output noise level: better than 80 db below rated output. Frequency response: ±1.0 db 20 to 20,000 cps.

SUMMARY

There are many other phases of the recording studio operation comprising new equipment requirements with regards to mastering studios, re-recording consoles, echo chambers and auxiliary studio equipment which have not been covered.

It is hoped that the items discussed herein have been of particular interest to those involved in this field and to others who have allied interests.

THE AUTHOR

Milton T. Putnam graduated from Valparaiso Technical Institute in 1940 and in 1942 attended the Illinois Institute of Technology. Previous to World War II, he was employed in the engineering department of radio station WDAN, then he became chief engineer of WDWS.

In 1946, he founded the Universal Recording Corporation, Chicago, of which he was president until 1958. At the present time he is chairman of the board. He is also president of the United Recording Corporation of Hollywood and technical consultant for Discos Mexicanos Studios, Mexico City.

He has published numerous technical papers on the subject of audio and sound recording.

Mr. Putnam is a Fellow of the Audio Engineering Society and a past officer of the Chicago Acoustical and Audio Group.