THE COMPLETE TECHNICAL GUIDE TO WMFO

-- 1987 --

THE COMPLETE TECHNICAL GUIDE TO WMFO

INTRODUCTION

During the period 1984 - 1985 (or so), the studios of WMFO, Medford (Tufts University radio) were completely redesigned and revamped. The work was coordinated by Joe Paradiso and Leigh Shein. Details are specified in "press releases", crank letters from staff members, etc. contained in the appendix to this document. Various levels of assistance were donated by several staff members. The names of most of these are documented in the "Tech Crew Staff List" also included in the appendix. This is but one of many lists generated during the two year existance of the "Tech Crew", and represents the staff at nearly maximum strength. The small subset of this list who helped and supported us considerably in this effort (ie. George Homsy, Dan Ottenheimer, and Bill Rosenblatt) deserve special mention. Others who lent considerable aid and moral support (ie. Greg Butensky, Chris Rich, Sarah Hood, and Sylvia Giannitrapani) are specially thanked. Nearly 50% of the staff picked up a soldering iron at one time or other during the reconstruction (for better or for worse), so the list of folks to acknowlege is certainly too large to type. Before I abandon this trajectory, I'd like to also thank Grady Moates of Loud and Clean Engineering for his helpful advice and consultation.

The reconstruction effort turned out to involve considerably more time and toil than either Leigh or myself had anticipated (usually true for these things...). We often wondered why we continued in this endeavor, and frequently came close to calling it quits and leaving the studios in a complete shambles. In retrospect, the major force driving us onward was probably creative in nature (apart from the social glamour of radio). We had an "innovative" vision of what we wanted WMFO to become, and continued our efforts until we realized these ideas. The outcome of all of the head-scratching, solder burns, cheap coffee & donut dinners, and sleepless nignts are summarized in this epic volume. The first section was compiled by myself, and contains details on all of the custom circuitry designed for WMFO. This section is composed of schematics, brief technical writeups, and data sheets (where applicable). The schematics may not be entirely accurate, since they were derived from notes scribbled during circuit construction and may not always reflect modifications made afterward (the level of accuracy should be well above 99%; one may occasionally find a resistor of somewhat different value, but all circuits essentially follow these schematics). Most circuit descriptions contain a few brief suggestions on trouble-shooting. All writeups contain a description of the device function and operation, followed by a circuit description with discussion of trimming procedures, etc. Needless to say, I don't have much time to spend writing this junk, so pardon the stream-of-consciousness style and occasional bad spelling. Hopefully things will be essentially readable.

The second section (compiled by Leigh Shein) contains a few general notes on the technical layout of the studios, some procedural suggestions, and detailed descriptions & diagrams of the interstudio and patchfield wiring schemes.

The final section of this report is an "appendix", which contains a potpourri of helpful info and non-technical addendia which summarize the spirit, color, and bouts of irrationality encountered during the reconstruction period.

Hopefully this document is of some aid to interested readers. It should certainly prove invaluable to WMFO, and should be consulted whenever any studio modifications are planned or circuitry fails (as will certainly occur eventually). In addition, I'll try to set up the "operation" section of these writeups such that they may be read by the technically inept, and used as "owners manuals" to master the function of the various and sundry home-made modules scattered throughout the station.

Live long and prosper...... Joe Paradiso 2/3/87

SECTION ONE

Complete Details
on all
Custom-Designed
WMFO
Circuitry

WRITEUPS DESCRIBING OPERATION, FUNCTION, AND MAINTANANCE OF WMFO CUSTOM-DESIGNED CIRCUITRY

Preamble

This section of the WMFO technical report contains detailed descriptions of all custom designed audio and control circuitry. Much of this hardware is vital to the function of WMFO. While writing this text, I essentially "brain-dumped" everything I could remember about these devices and how they should be used and maintained. This sytle resulted in a wealth of information recorded in these pages that not only detail technical information, but also describe extensively the function of this equipment, and how to use it in a practical context. These writeups (particularly the "Function and Operation" sections) thus provide a "Functional guide for the advanced WMFO layman". If one reads these sections, one gains an understanding of how the station is put together at a minimum of technical jargon. Any individuals interested in exploiting the capability of the studios to their fullest should read through the introductions to some of the more relevant writeups in this report. Anyone who is technically inclined (or hopes to maintain the hardware) might want to read through these writeups cover-to-cover. I've tried to present the details as completely as possible.

Because of the wealth of "practical" information buried within individual writeups, I've assembled a quick cross-reference below that pairs information that may be needed to keep WMFO going on a day-to-day basis with the relevent writeups to consult. The pages of this report aren't numbered from start to finish; in order to locate a particular writeup, one can find its order in the Table of Contents and flip through the text until it is located.

Functional Cross-Reference:

Using the Intercom, exploiting the touch-tones, and communicating with Studio D; see The PA Mixer/Intercom, The Studio D PA/On-Air Unit

Patching and general studio structure; see The Distribution Amplifier

Connecting "home audio" and external equipment into the boards and patchbays; see The Stereo Interface, The Jacks Panels

How to use the Pitch Shifter; see The Pitch Shifter

Functional Cross-Reference (cont.):

Setting levels on tape decks, etc.; see The House Tone Oscillator, The Line Amplifier

Patching in Master Control, working with remote lines; see The Utility Amplifier

Communicating with people in a remote site (Curtis Lounge, Ballou Hall, & MacPhie Pub) via the WMFO Intercom; see: The PA Mixer/Intercom, The Intercom Phone Ringer

Making a DJ " Voice Skim" tape; see The Studio A Cassette Remote and Auto-Skim

Doing a "phone-in" show, or interfacing the telephones with the control boards and studios; see: The Studio A Kludge Cards, The Studio B Kludge Box, The Studio C Kludge Cards, The Music-On-Hold Driver and Intercom Bulb Relay

Switching studios into and out of the airchain; see The Studio Switcher

Monitoring and adjusting transmitter functions; see The Transmitter Audio and Power Monitor, see also the relevant short description in the next section of this report, ie. Misc. Procedures and Studio Wiring Conventions

__Schematics and Descriptions of WMFO Custom Designed Circuitry__

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1) Phone/Doorbell Unit
2) PA Mixer/Intercom Unit
3) Studio D PA/On-Air Box
4) Music-On-Hold Driver and Intercom Bulb Relay
5) Intercom KTU
6) Intercom Phone Ringer
7) Distribution Amplifier
8) Stereo Interface
9) Pitch Shifter
10) Line Amplifier
11) House Tone Oscillator
12) Utility Amplifier
13) Jacks Panels
14) External FM Buffer/Driver/Filter
15) Studio Switcher
16) ON-AIR Relay Boxes
17) Studio A Cassette Remote and Auto-Skim
18) Studio A Mono/Stereo Mic. Relays
19) Studio A Gas Light
20) Studio A Kludge Cards
21) Studio C Kludge Cards
22) Studio B Kludge Box
23) Transmitter Audio and Power Monitor

Principles and Operation

of the WMFO Phone-Ringer/Doorbell Unit

-- J. A. Paradiso

April, 1984

(Revised Jan. 1987)

Specifications and External-World Connections

OUTPUTS:

LED/Speaker outputs: 4 independent outputs capable of driving LED's, 8-ohm speakers, etc.....

Low-level audio output: Drives any mixer, amplifier, etc.

Pushbutton lights output: Drives light bulbs in doorbell pushbuttons when mike in air studio is live.

Doorbell gate output: A 5 volt gate which goes high when the inside or outside doorbell is sounding. This gate is used by the PA Mixer to engage a PA L-pad defeat.

INPUTS:

Outside door pushbutton.

Studio door pushbutton.

Common "ring line" from telephones.

"Normally ON" connection from mike circuit in air studio.

See Fig. 3 for I/O connection location.

PHONE/DOORBELL DESCRIPTION

I) Introduction

In this writeup, the phone/doorbell triggered oscillator system is described. Figures 1 and 2 are schematic diagrams of all circuitry, and Figs. 3 and 4 show chassis and card layout. In some of these schematics, resistors in parallel and other "irrational" things may appear. This is because these schematics reflect the actual hardware configuration as accurately as possible, and things like this were done after the unit had been completed, in order to trim original component values. The symbols used in the schematics are conventional, and include component values. A small "x" interrupting a line indicates a connection between the circuit cards and the chassis. This document is an update of the original published in 1977; all subsequent modifications are incorporated.

II) Functions

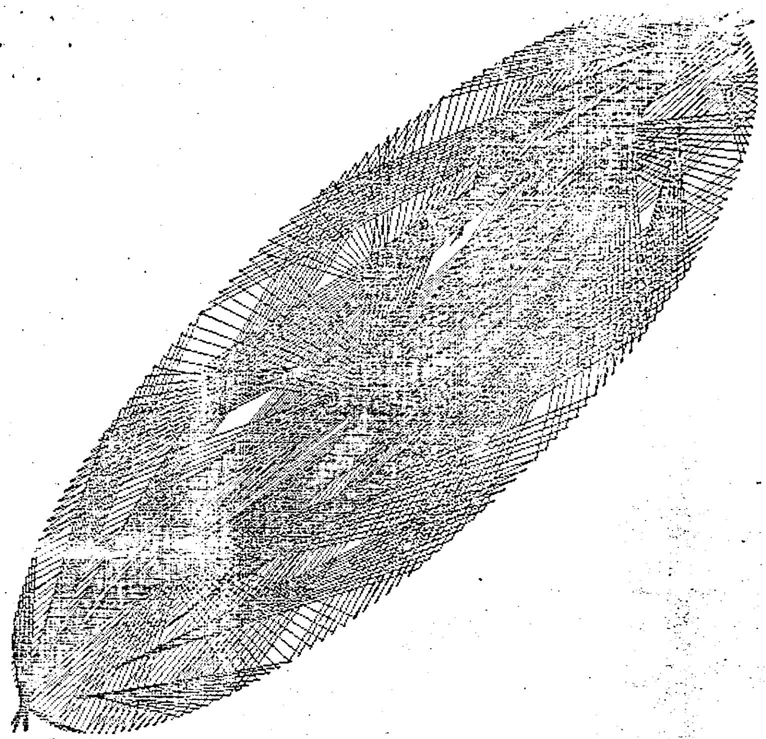
This device produces three distinguishable tones: a constant tone (signifying the studio door), a slow warbling tone (signifying the outside door), and a fast warbling tone (signifying the telephone). The frequency of these tones is adjustable over a considerable range (via P4, which is mounted on the front panel), and the phone-ringing sound can be disabled (via S1, which is also mounted on the front panel). This unit contains four semi-independent amplifiers, which can drive either LED's or 8-ohm speakers. The volume (intensity) controls are mounted on the front panel (beware, the pots are wired backwards...!), and the LED/speaker connections are accessable on the main terminal strip (as depicted in drawing 3). At present, these amplifiers will only be used to drive LED's (only 1 channel is needed), since a low-level audio output is tapped (also on the terminal strip) and routed to a master PA amplifier via the mixer/intercom unit, making the speaker option unnecessary. It exists if the need ever arises....

The duration of each tone (studio D., outside D., phones) is adjustable from roughly 10 millisecs. to 10 seconds via trimmers P1-P3, which are mounted on the power-supply board. Three LED's are mounted on the front panel, and depict the origin of each tone, ie. phone, studio door, outside door.

This device also provides facilities which connect to the air studio mike muting relays (a NRM. ON connection is used; this input will eventually come from the studio switcher), and illuminates light bulbs inside of the doorbell pushbuttons when the air studio mike is live (thus the potential enterer knows that the jock may not hear his plea).

Editorial Note:

SINCE THIS CIRCUITRY IS RELATIVELY UNCHANGED SINCE ITS INSTALLATION IN 1977 (I BUILT THE ORIGINAL IN 1975/76, BUT IT WAS DESTROYED IN THE FIRE; MOST BUGS WERE WORKED OUT DURING THAT TIME. WHEN THIS DEVICE WAS FIRST INSTALLED, WMFO WAS ESSENTIALLY THE ONLY LOCATION WITH SUCH "UNUSUAL" SOUNDING PHONE RINGS; NOW AT&T HAS INSTALLED DEVICES WITH SIMILIAR ACCOUSTICAL PROPERTIES AT EVERY BUSNIESS..... WMFO WAS FIRST!!!), I HAVE KEPT THE ORIGINAL DOCUMENTATION, AND INTERRUPTED IT IN A FEW PLACES TO ADD ANY RECENT UPDATES. I THINK IT'S LEGIBLE ENOUGH, AND PARDON ANY MISSPELLINKS....



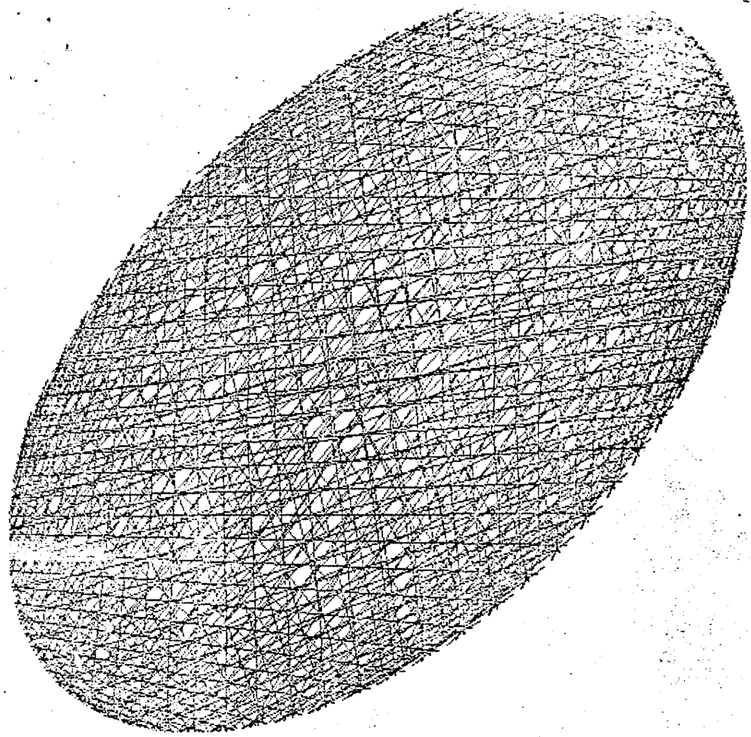
III) CIRCUIT DESCRIPTION:

This circuitry looks somewhat formidable upon first peek, but is actually quite simple in conception. The heart of the device is portrayed in drawing #1. The circuit is based entirely around the 555 timer IC, which is easily available anywhere. The audio oscillator is made of IC5, which is merely a 555 connected as an astable multivibrator, with free-running frequency controlled via P4 and C10. Another astable multivibrator drives the voltage-control input of IC5, namely IC4, which has a hard-wired rest frequency of about 10 hz, given by R10 and C9. IC4 thus produces a vibrato effect in our audio oscillator IC5, giving the "warbling" tone. IC5 and IC4 are normally held in re-set by R18 and R19 at pin #4, hence both the audio and vibrato oscillators are normally held low (OFF).

IC1, IC2, and IC3 are 555's wired as monostable multivibrators, ie the outputs are quiescently low, and once triggered, they go high for an amount of time set via the duration controls (P1-P3), and the timing capacitors (C1,C4,C7), and then drop low again. IC1 is triggered by S2, the outside door pushbutton. IC3 is triggered by S3, the studio door pushbutton. IC2 is triggered by a drop in resistance in the light-dependant resistor (LDR), caused by the neon bulb (NE2A), which flashes when the telephone rings. One must notice that the "common bell line" from the phone box is tapped to drive the neon bulb. This line is normally low, and only pulses when ANY phone-line rings (625-0800, 625-8566, or X444). It does not pulse when one dials, or hangs up any phone.

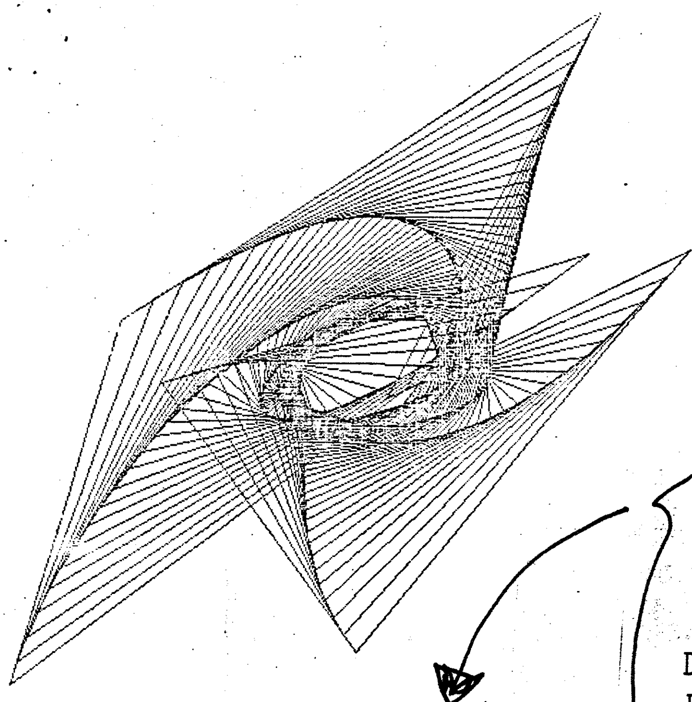
IC7 IS A CMOS SCHMIDT TRIGGER WHICH CONDITIONS THE DOORBELL LINES BEFORE FEEDING THEM TO THE TRIGGER INPUTS OF THE 555's. THIS REMEDIED THE RANDOM TRIGGERING OF THE DEVICE, CAUSED BY POWERING ON TAPE DECKS, THE BUS GOING BY, ETC...

The outputs of IC's 1-3 are connected to the re-set inputs of IC's 4 and 5 via some simple diode logic. Let's trace through each separately....



When the studio door pushbutton S3 is pressed, IC3 will go high for a few seconds. Current will then flow through diode D6, and hold the re-set input of audio oscillator IC5 high (pin #4). This enables the audio oscillator, and a tone is produced while IC3 is high. Since no current is fed to the re-set input of the vibrato oscillator IC4, it remains low, and no warbling is heard, thus the tone is constant.
When the phone rings, IC2 will go high (provided that the phone defeat switch S1 is off). This drives a current through diode D4, which enables the audio oscillator, as described above. However IC2 will also send current through diode D2, which will additionally enable IC4, the vibrato oscillator. Thus IC2 enables both audio and vibrato oscillators simultaneously, producing a warbling tone.
When the outside door pushbutton S2 is pressed, IC1 will go high. This will send currents through diodes D1 and D5, which enable the vibrato and audio oscillators respectively, as sketched above. Here, however, we also turn transistor T1 on through diode D3. This will pull the voltage-control input of the vibrato oscillator IC4 low, generating the net effect of slowing down the vibrato frequency. This results in the production of a slow, warbling tone.
Thus the basic tone production and logic are understood. LED's 1-3 are monitors connected to the outputs of monostables IC 1-3 (these LED's seem to frequently go bad, perhaps either better LED's, or higher limiting resistors, should be installed if someone feels ambitious and/or daring). C2,C3,C5,C8, and C8A are filter capacitors to prevent spikes induced on the wires running to the pushbuttons, etc. from falsely triggering the apparatus. The 5-volt square wave output from

Should now be OK -SAP-

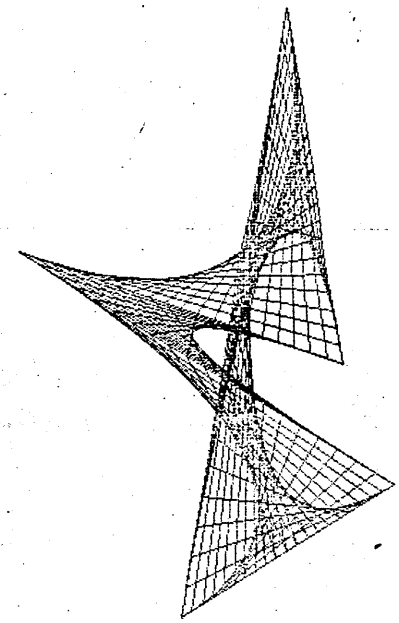


audio oscillator IC5 is divided by the network R20/R21, and provides a low-level audio output which is fed to the mixer/intercom unit (described later).
The output of the oscillator is also fed to the internal amplifier section, pictured in drawing #2. The oscillator output drives comparitor IC6 (an LM301 operational amplifier). The trip voltage of this comparitor is set at 3 volts by R23 and R24. Normally, the output of the audio oscillator remains low; thus the output of the comparitor remains low. However when it is enabled, the oscillator will swing 0-5 volts, hence the comparitor will swing back and forth between ~~it's~~ saturation points, ie. 0-10 volts or so.

D10 AND Q7 WERE ADDED TO INTERRUPT THE AMPLIFIER SIGNAL IN CORRESPONDANCE WITH THE VIBRATO OSCILLATOR OUTPUT. AN LED CONNECTED HERE WILL FLASH IN TIME WITH THE WARBLING TONE. SPEAKERS CAN STILL BE ALSO CONNECTED, BUT WILL NO LONGER SOUND AS PLEASANT.

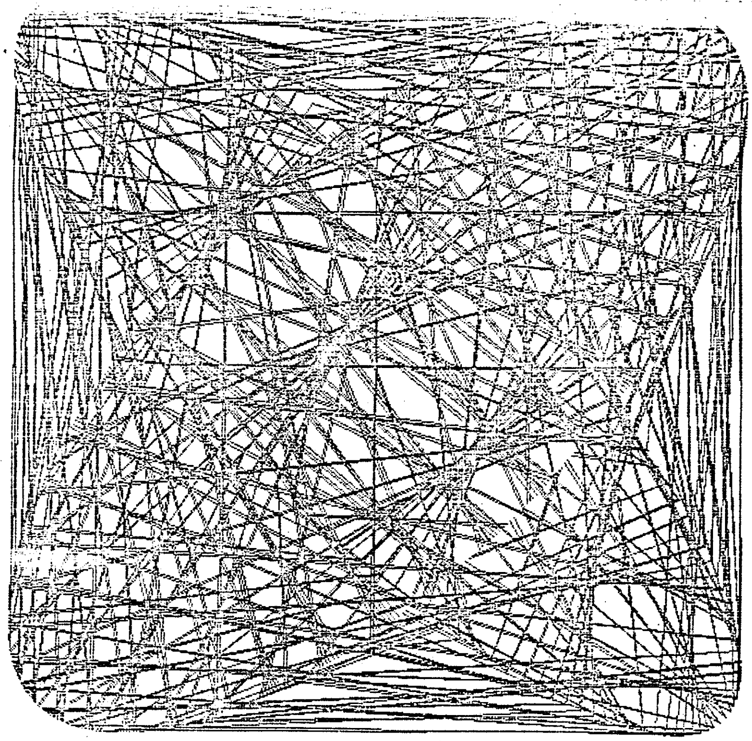
NOTE (1987): The input to the comparator (IC6) no longer comes from the audio oscillator output, but is tapped from its reset pin (which is driven high when it is allowed to sound). Thus there is no longer any audio at these outputs. This was done to avoid any crosstalk problems in the studios.

The output of the comparitor IC6 drives the 4 Darlington output buffers T2-T5. The base voltage of each Darlington is controlled via volume controls P5-P8. The emitter outputs of the Darlingtons are brought out to the terminal strip as speaker outputs A-D. (One note: these Darlington transistors are Radio Shack specials, and hence are very sensitive to short-circuits, etc. Watch out for them). This amplifier output section is not currently employed, since the low-level audio output direct from IC5 is used instead. [1 will be used for LED's]
The power supply is also depicted in drawing #2; it is quite standard, with the 6-volt supply for the 555's provided by zener series regulator arrangement T6 and D9. C11 and C12 are filter capacitors; the raw speaker amplifier supply for the Darlingtons is pulled directly off of C11, while the supply for the circuitry is further smoothed by C12. Transformer TR1 provides current for the electronics, while TR2 provides current for the on-air light bulbs inside the pushbuttons.
Drawing #3 shows the rear of the chassis assembly, and it indicates all



connections between the device and the outside world. All connections are made at two terminal strips, the long one at the upper left is for the electronics, while the short one at upper right ties connections to the on-air bulb facility. Drawing #4 gives a coarse layout of the card and power supply board.

BECAUSE ALL OF THE "NORMALLY OFF" CONNECTIONS IN THE STUDIO A MIKE RELAY SYSTEM WERE USED ELSEWHERE, THE SWITCH-BULB DRIVER CIRCUIT WAS MODIFIED TO RUN OFF OF "NORMALLY ON" CONTACTS BY ADDING TRANSISTOR T7 AS A SIMPLE INVERTER. THE PRESENT RELAY CONNECTION CAN NOW BE RUN FROM AN OPEN-COLLECTOR GATE; THIS IS WHAT WILL BE DONE AFTER THE STUDIO SWITCHER IS INSTALLED.



IV) TROUBLE-SHOOTING HINTS:

I'll try to sketch some possible difficulties below....

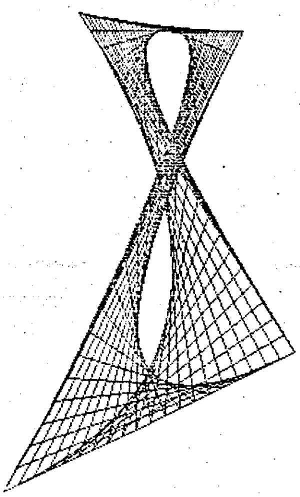
The 555 is a very rugged IC, and doesn't often blow out, provided you're not careless. Once a 555 DID blow on this unit, but that was because I had put the card in quickly, and accidently shorted some of the contacts. Regardless, 555's are cheap and plentiful, so there's nothing against replacing a 555 to see if that relieves the problem. However, I stress one obvious warning: DO NOT REPLACE THE 301 WITH A 555!!!

This brings to mind another "obvious" warning: DO NOT EVER PUT THE CARD IN BACKWARDS!!! THE IC'S FACE IN TOWARD THE CHASSIS, AND THE COMPONENTS FACE OUT AWAY FROM THE CHASSIS. CHECK THE ORIENTATION BEFORE YOU PULL THE CARD, AND MAKE SURE THAT YOU PUT IT BACK THE SAME WAY. This mistake could cause real problems.

The card is made out of a cheap Radio Shack plugboard, and the etch has ocassionally broken near the edge-connectors, where the card mates with its socket. The etch leading to these connections is quite thin and lifts and breaks easily. So handle the card carefully (we haven't had this problem in years..). You can determine the existance of this fault by checking the etch with an ohmmeter, and looking very carefully for cracks.

The hookup wire used in places is a bit thin, and may break, particularly the wiring to the card socket. Be careful of this area, and check for broken wires and/or shorts here.

The regulator transistor is a common 2N3055, and has never yet failed. If it does, it may very well blow out all the 555's, and any other active components in it's path. Check the 6-volt line, and make sure that it's not sitting pretty at either zero or 10 volts. If something's



wrong here, check the zener, the 2N3055, the bridge rectifier, the transformer, the fuse, etc. One recent problem occured where the outside and inside doorbells both made the same constant non-warbling sound. I remidied this by adding resistors R13-R14 in series with the collector circuit of T1 (evidently the conductance of T1 had increased somehow, and it pulled pin #5 of IC4 too low). This fix-up restored the familiar slow-warbling sound to the outside doorbell. If this effect happens again, check transistor T1, and make sure its collector resistance is sufficent.

Well, I've pretty much covered all of the problems I've encountered with the buzzer. Hopefully it won't present any additional difficulty. If it does, remember that reality is basically simple, even though it may appear fundamentally non-ideal at times.

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AGAIN, I STRESS, LOOK FIRST AND FOREMOST FOR BAD, LOOSE, BROKEN, AND/OR ROTTED CONNECTIONS AND COPPER STRIPS (SOLDER JOINTS!!! WORST CULPRIT) ON THE MAIN CARD. THIS IS WHAT NEARLY ALWAYS IS AT FAULT.

If you put the card in wrong, the zener (D9) may blow out. This happened the last time this unit was serviced. The original problem was a broken etch at the edge connector (as discussed above), but the card was replaced with the power on, and the zener blew, changing the power supply voltages appreciably.

(last serviced in Dec., 1985)

Phone Doorbell Updates (1/87)

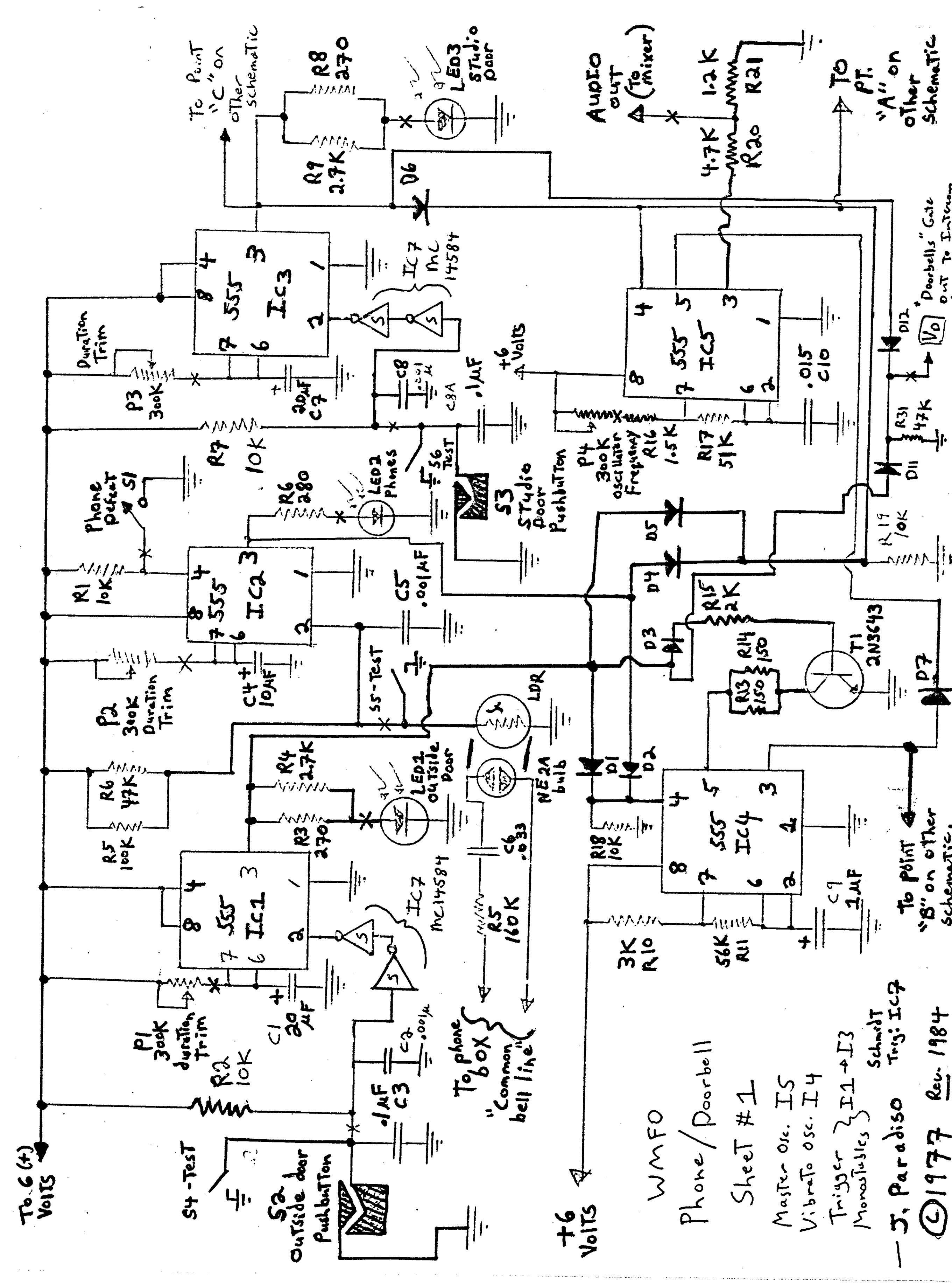
As all WMFO data is being collected, I'll give a quick rundown on the few modifications that were made after the previous text was edited and assembled in 1984. The device has survived relatively intact since its inception in 1977 (I never thought it would last that long). The only problem has been consistantly traced to broken etch lines near the edge connectors. With a little patience and understanding from future technical folks, I suspect this unit can last indefinately.....

The only changes that have been made have involved the LED drive circuitry and additional diode logic. The audio signal has been removed from the LED drive, as mentioned in the brief note a few pages back. This was done because the LEDs have been mounted on the mixing boards in all studios (causing the wiring to be run through the boards); in order to involve potential crosstalk from the square-wave tone output from the phone/doorbell unit, the LED driver signal was tapped from the oscillator enable line, giving a DC pulse at the LED output. The major consequence of this change is that speakers will no longer produce the phone/doorbell tone when connected to these outputs (this option was only really needed during 1977/1978 anyway, thus it rapidly became obsolete). One of the outputs is used to drive all LEDs, thus it should be full up (as marked on the front panel). The other outputs should also still work, but have no application for now...

The other change in this circuitry has been the addition of diodes D11 & D12 and R31 on Fig. 1. These diodes OR together the outputs of the two doorbell monostables IC1 and IC3, thus creating a composite "doorbell gate" that goes high whenever any doorbell has been pushed, and stays high through the duration of the audio tone. This gate is used by the PA mixer circuitry to engage an "L-Pad Defeat", which turns all speakers in the radio station up to full volume (except for a studio with live DJ mike). In this fashion, the doorbell will always sound throughout the station, increasing the chances of someone hearing it and answering the appropriate door.

One other related issue has been the addition of a standard AT&T bell to the telephone ring line in Master Control. This bell is located behind the grey monitor/PA rack (in which the Phone/Doorbell unit is stationed), and is mounted on the wall. It is normally switched off (the switch is mounted atop the bell's cover). If, for some reason, the telephones no longer ring through the phone/doorbell unit (first please check that the defeat switch S1 is off), this bell may be turned on to indicate incoming calls.

FIGURE 1



WMFO
Phone/Doorbell
Sheet #1
Master osc. I5
Vibrato osc. I4
Trigger Monostables I1→I3
Schmidt Trig: IC7
— J. Paradiso
©1977 Rev. 1984
Rev. 1986
To 6 (+) Volts
+6 Volts
S2 Outside door Pushbutton
S4-Test
S3 Studio Door Pushbutton
S5-Test
S6 Test
Phone defeat S1
To phone box "Common bell line"
To Point "C" on other schematic
To PT. "A" on other schematic
To Point "B" on other schematic
Audio out (To mixer)
"Doorbells" Gate out to Intercom Circuit
LED1 Outside door
LED2 Phones
LED3 Studio door
NE2A bulb
LDR
T1 2N3643
IC7 MC14584
P1 300K duration trim
P2 300K Duration Trim
P3 300K Duration Trim
P4 300K oscillator frequency

FIGURE 2

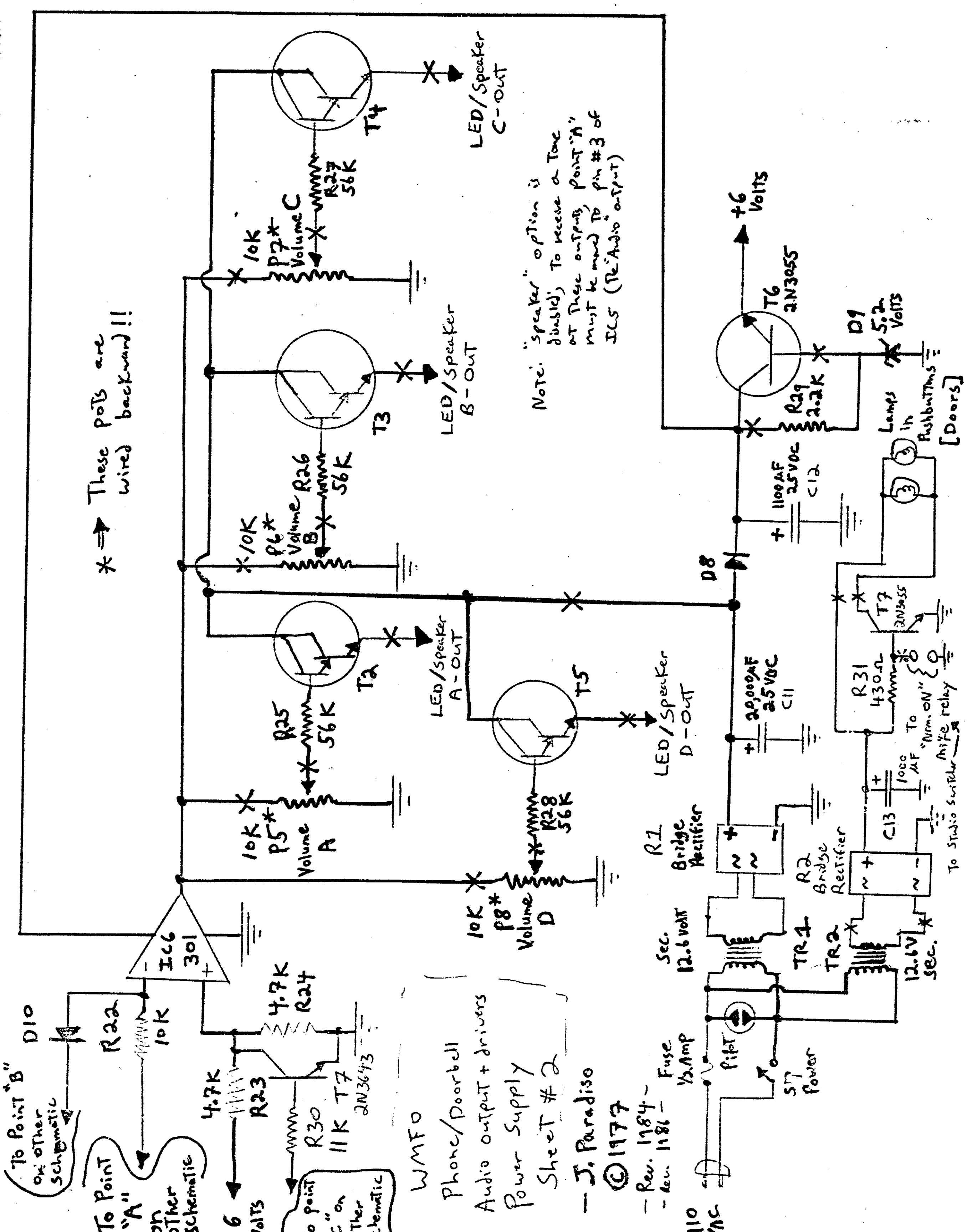

LED/Speaker C-OUT
T4
R27 56K
10K P7* Volume C
LED/Speaker B-OUT
T3
R26 56K
10K P6* Volume B
* ⟹ These pots are wired backward!!
Note: "speaker" option is disabled; To receive a Tone at These outputs point "A" must be moved To pin #3 of IC5 (The Audio output)
+6 Volts
T6 2N3055
D9 5.2 Volts
R29 2.2K
Lamps in Pushbuttons [Doors]
1100 μF 25 VDC C12
D8
LED/Speaker A-OUT
T2
R25 56K
10K P5* Volume A
T5
LED/Speaker D-OUT
R28 56K
10K P8* Volume D
2000 μF 25 VDC C11
T7 2N3055
R31 430Ω
To "Mike ON" Mike relay
To Studio Switches
1000 μF C13
R1 Bridge Rectifier
R2 Bridge Rectifier
Sec. 12.6 Volt
TR1
TR2
12.6V Sec.
IC6 301
D10
R22 10K
4.7K R24
4.7K R23
R30 11K
T7 2N3643
To Point "B" on other schematic
To Point "A" on other schematic
+6 Volts
To point "C" on other schematic
Fuse 1/2 Amp
Pilot
S1 Power
110 VAC
WMFO
Phone/Doorbell
Audio output + drivers
Power Supply
Sheet #2
— J. Paradiso
©1977
— Rev. 1984 —
— Rev. 1981 —

FIGURE 3

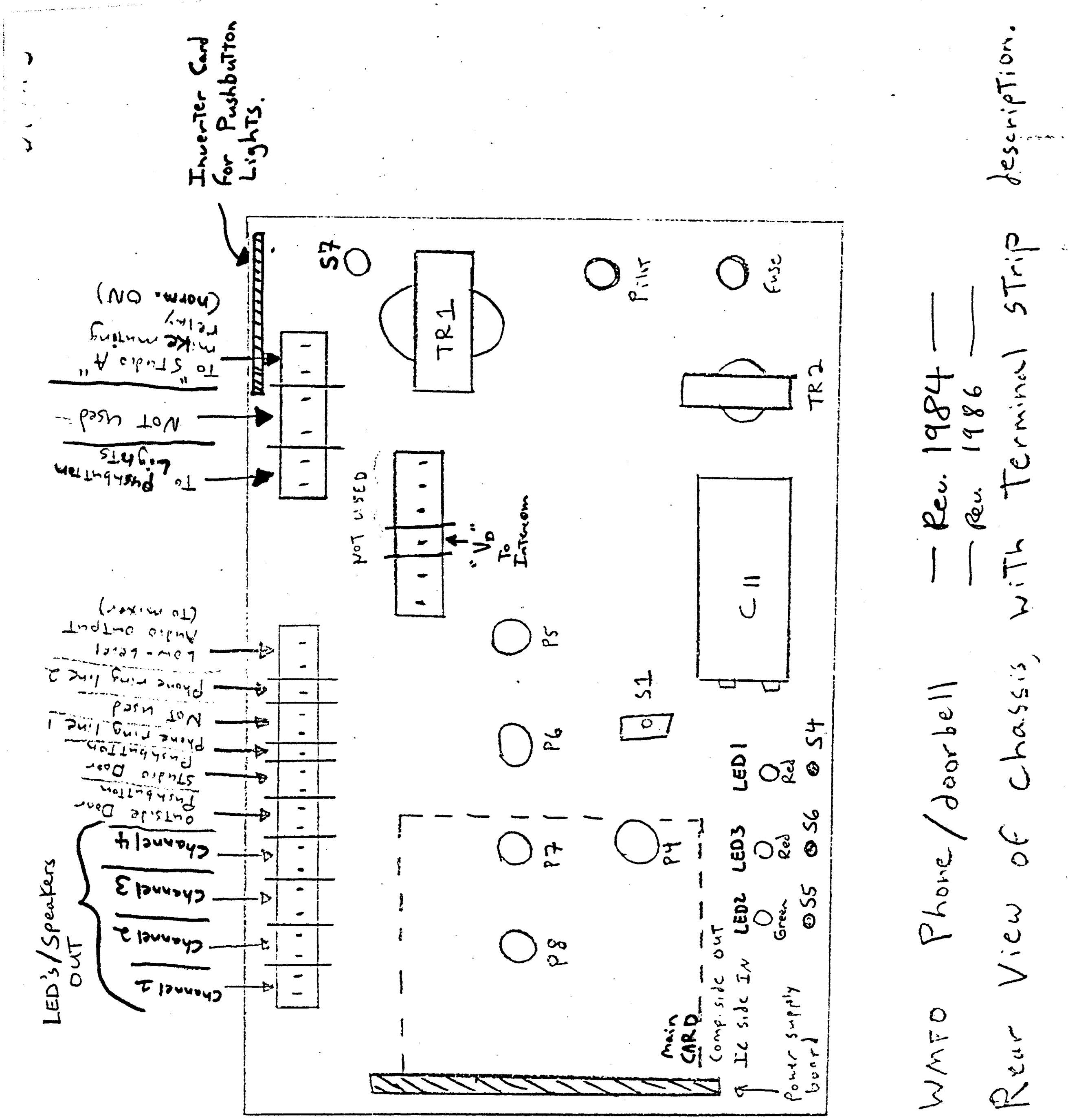
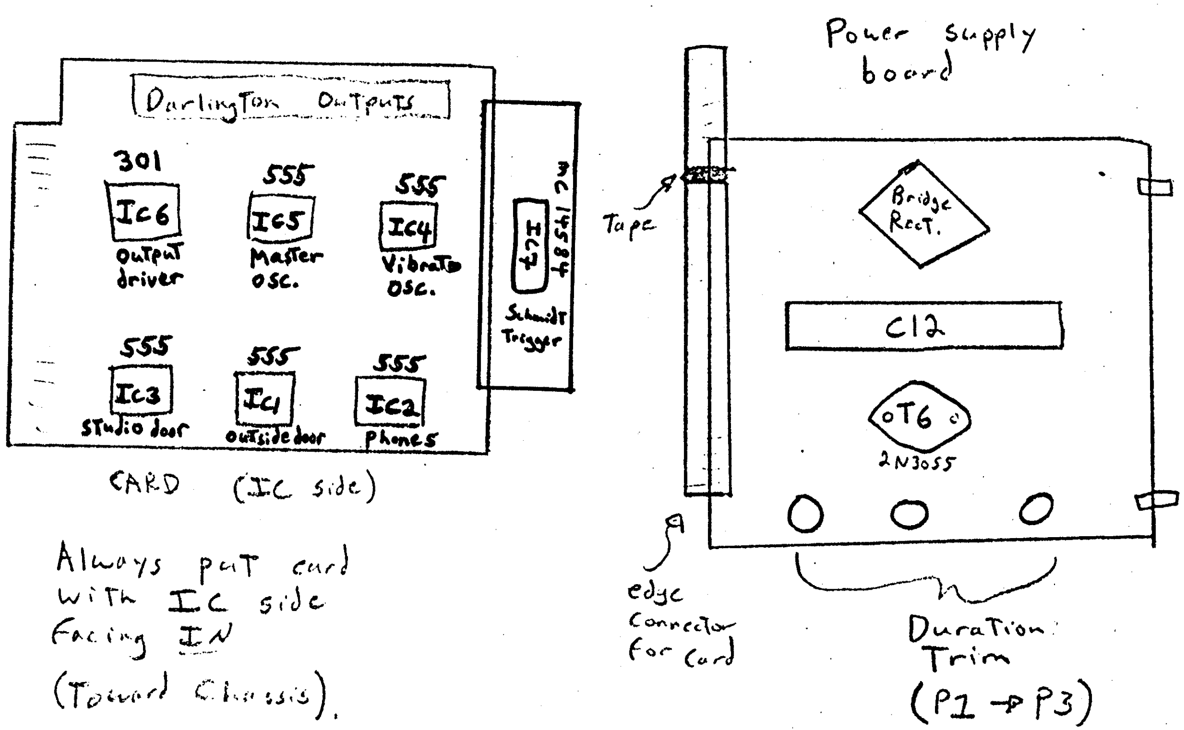

WMFO Phone/doorbell — Rev. 1984 —
— Rev. 1986 —
Rear View of chassis, with Terminal Strip description.
Inverter Card for Pushbutton Lights.
To "Studio A" mike muting relay (norm. ON)
Not used
To Pushbutton Lights
Not used
"V_D" To Intercom
Low-level Audio output (to mixer)
Phone ring line 2
Not used
Phone ring line 1
Pushbutton
Studio Door
Pushbutton
Outside Door
Channel 4
Channel 3
Channel 2
Channel 1
LED's/Speakers OUT
S7
TR1
TR2
Pilot
Fuse
C 11
P5
P6
P7
P8
P4
S1
LED1 Red
LED3 Red
LED2 Green
S4
S6
S5
Main CARD
Comp. side OUT
IC side IN
Power supply board

FIGURE 4



Always put card with IC side Facing IN

(Toward Chassis).

Do NOT Replace The 301 with a 555 IC !!

WMFO Phone/Doorbell

Main Card and Power Supply board Layouts

— Rev. 1984 —

Principles and Operation of the WMFO PA Mixer/Amplifier Unit

--- J. A. Paradiso

April, 1984

(Revised Jan., 1987)

The WMFO Mixer/Intercom PA Buffer Unit

Specifications and External-World Connections:

INPUTS

Mixer inputs:
- Phone/Doorbell audio in.
- FM monitor tuner (mono) in.
- Studio B program (mono) in.
- Studio C program (mono) in.
- Auxiliary audio in.
- EBS audio in

L-Pad defeat gate inputs..
- EBS receiver active in
- Doorbell gate in

----- All above inputs are referenced to a common local ground. -----

Spare phone line input (intercom line).

----- Above input has an independent floating ground. -----

OUTPUTS

Mixer outputs:
- Studio PA line out.
- Halls and Offices PA line out.

----- Above outputs are referenced to the common local ground, and feed each channel of the PA amplifier. -----

Stereo Headphone output (for mixing and diagnostics).

L-Pad defeat relay driver output.

----- The above line is referenced to the intercom ground (thus bears no relation to the common local ground). -----

THE WMFO MIXER/INTERCOM PUBLIC-ADDRESS BUFFER UNIT

I) Introduction:

This device serves two purposes:

1) It is a two-channel mixer, accepting 7 inputs [Phone/Doorbell audio, FM monitor tuner, Studio B/C program output, an arbitrary auxiliary audio source, EBS, the intercom audio output, and the Random Tune Generator (RTG) output], and routing them to the studio and Halls/Offices PA lines (the "studio" line serves Studios A, B, and C; the "Halls/Offices" line drives the speakers placed elsewhere). The weight of each input is independently adjustable for both channels; separate potentiometers allow one to set each input to an appropriate level for the studios and Halls/Offices. During normal use, the phone/doorbell, intercom, and RTG outputs are routed to the studios, while the former list plus the FM monitor tuner output are routed to the Halls/Offices. If interesting activity is occuring in Studio B or C, and one wishes to route this source to the monitor speakers without interrupting the normal on-air flow eminating from Studio A, one can pot down the FM tuner input, and pot up the Studio B/C program input (Studio B or C is switch selectable). An auxiliary line is also provided, with RCA input jacks convienently located on the front panel. One can thus plug independent external sources (tape deck, etc.) into these, and add them into the PA mix. The gain of each channel is adjustable from zero to approx. 5 via gain controls also mounted on the front panel. The mixer outputs feed a power amplifier which distributes the signal to the speakers scattered throughout the station; one must avoid turning the gain up too high, or the input stages of this power amplifier will start clipping.

The front panel of the unit is (was) well-labeled. The layout of the back side of the panel is shown in Fig. 7. The input weighting potentiometers are seen as P1-P5 (studios), and P7-P11 (halls/offices). The master gain controls are P6 (" ") and P12 (" "). The RTG level is controlled by trimmer potentiometers mounted on the Tune Generator card (see Fig. 8). The input and output connectors are RCA jacks mounted on a strip located between the gain and input level controls, in back of the panel (see Fig. 7). They are clearly labeled.

2) The second purpose of the unit is to handle the station intercom system. A spare phone line is patched into the unit via terminals also located on the above-mentioned strip. When one picks up a telephone connected to this line, one can speak over the monitor speakers (provided the "intercom" is potted up on the front panel; see above). In order to guarantee that this paging request is heard, two additional things happen when the intercom is engaged: 1) The FM monitor tuner and studio B/C inputs are attenuated (the attenuator level is set via trimmers P17 and P18; see Fig. 7), so the pager doesn't have to shout over any din, 2) The L-pad level controls controlling the volume at each PA speaker are disabled, so that the message is heard at full blast, regardless of any local setting (the news office speaker has a hidden switch to inhibit this feature; this switch should not be used except in extrodinary cases, and promptly returned to the "ON" position afterward). The yellow LED on the front panel (LED1; see Fig. 7) is illuminated when the intercom is active; the lights in the phones are not driven by this circuit.

The lower row on the touch-tone phones (*,0,#) is decoded in this device, and accomplishes some interesting functions. They are summarized on the following page. The Studio D "PA/on-air" box has the potential of decoding the intercom "9"; see the description of this circuit for additional details.

Key	Function
*	Mutes the intercom; the telephones are disconnected from the PA speakers, but remain active. One can still converse with other phones on the intercom line, but the conversation is not heard throughout the station. The FM monitor levels, local L-pad settings, etc. return to their quiescent settings after "*" is pressed.
#	Re-connects the intercom line to the PA speakers. This function is the inverse to "*".
0	Plays a bar or two of a random shlocky tune over the PA speakers. The monitor attenuation and L-Pad defeats are also engaged, so all within the station MUST hear the tune.

I'll list a few notes below pointing out possible applications of these features. A possible scenerio follows:

A few ambitious production folks need the facilities of both studio's B&C for an extrodinary creative effort. To coordinate activities (tape starts, effects, acting, etc.) between the studios during the extravaganza, they decide to use the intercom. Of course they don't want their dull conversation to disturb less adventurous folk elsewhere in the station, so they have wisely pressed the "*" to mute their conversation. However, some minutes later, the on-air jock has answered a telephone call for Mary Zeichmann, who isn't in the main studio at the time. Instead of getting up and looking for her (an important transition is imminent), he picks up the intercom for a page. Unfortunately (for him), the intercom is already being used by the production folks. However this jock knows his options, and wisely presses "#", re-connecting the intercom to the PA speakers. He then issues his page for Ms. Zeichmann (which is heard throughout the studio), and then presses "*", thus re-muting the intercom, and returning all PA channels to normal. He then apologizes to the production folks, hangs up, and makes the grand transition.

The zero feature has completely open application. One can hit the zero and play a tune before issuing a page; using it as a convinent PA break and attention-getter. One can also use the zero for the sake of pure joy and annoyance of the less enlightened. It is usually best to hit the "*" before playing with the "0" in order to mute the touch-tone keyboard and prevent it from interfering with the tune.

The device knows a total of 28 annoying melodies (see Appendix for a full list), and plays one of them at random when triggered. The tune may be also requested by depressing the pushbutton S3 on the front panel (see Fig. 7). The red LED2 is illuminated while the RTG is at work. Ocassionally (when one requests a tune too quickly after the preceeding tune has concluded), the tune processor can lock up into a metastable "active" state, although no tune is being output. This state is intercepted and cleared after a delay of about ten seconds, after which the PA and RTG behaive normally. If (for any reason), the tune generator should lock up and not clear itself, one can reset the unit by cycling on-and-off the AC power. At the moment, one can not pot the tune generator up over the air, but provisions will be made to bring the studio PA line up on the Studio C patch-panel. (NOTE 1/87: The PA line is now available at the patchfields in all studios, and may be directly potted up on the studio A board).

The L-pad defeat relay driver line originates from a pair of terminal posts in back of the front panel (see Fig. 7). The intercom ground (hence the L-pad defeat and phone lines) are completely isolated from the local and station grounds, and should not at any point be tied together.

The mixer output also drives a headphone jack; one can monitor the mix while balancing levels by plugging in a stereo headphone.

II) Circuit Description:

The circuitry is given in the schematics Fig. 1 through 5. I'll try and quickly go over them.....

Fig. 1 shows the power supply for both intercom and mixer sections. The mixer (common local) ground is given by the conventional symbol, whereas the intercom ground is always circled. Both grounds are independent (as sketched above), hence any voltages in the intercom system must be measured against the intercom ground (which does not come to the chassis; that is the mixer ground). I generally use the "-" line of the intercom power supply filter capacitor (C17) for a reference to the intercom circuitry. The mixer power supply is 18 volts regulated, derived via TR1/D1-D4/C16 & the 7818T. There is not much circuitry in the mixer, hence we build the supply lighter. The intercom supply is a bit more involved, with 2-amp TR2 feeding D5-D7/C17. Regulators 7809 and 7815 feed a smooth DC voltage to the intercom circuitry, with the former delivering +9 volts to the tone decoder circuit, and the latter supplying +15 volts to the rest. Both regulators are heat sunk and mounted on the tone decoder card.

Fig. 2 shows the entire mixer circuit. It is quite simple, with 741's IC1 and IC2 acting as inverting summers, and mixing all inputs. 741's aren't the best choice for audio applications, but for the moderate standard required in this application, they're certainly adequate. Because the op-amps are run from a unipolar supply, the non-inverting inputs are biased up by R9/R10, and all summing inputs and line outputs are capacitively decoupled. Emitter-followers T1 and T2 provide sufficent drive to power moderate-impedence stereo headphones. When the intercom is engaged, the coil of DPDT relay R2 is powered, which attenuates the Studio B/C and FM tuner inputs by P18 and P17 respectively, dropping the ambient noise level relative to the intercom audio signal. The phone/doorbell is not muted, and will still come through as set by P1 and P7. This circuitry dominates the "mixer card", see Fig. 7.

Fig. 3 shows the circuitry used in driving the intercom phone line, and decoding the touch-tones. When the intercom is picked up, the phone line is powered via R45 and R46 (thus LED1 is illuminated). Voice signals show up as variations in the telephone resistance, which give a proportionally varying current through R45/46 and the phone line. Thus the fluctuations in voltage across the phone line correspond to the voice signal, and this (provided relay R1 is OFF) is AC coupled into isolation transformer T3 (Diodes D20-D23 clip this voltage to attenuate the large spike created when the phone is picked up or hung up; capacitor C41 acts as a high-pass filter to eliminate a large "pop" which can be caused in this case). TR3 isolates the mixer ground from the intercom ground (point X feeds the summing nodes in the mixer).

When the intercom is picked up, comparator/Schmidt trigger IC8 goes high (due to the drop in voltage across the phone line), also provided R1 is OFF. This turns T5 on (D24 is merely to prevent the non-zero low-saturation voltage of IC8 from keeping T5 on when IC8 is low), and thus pulls current through the L-pad defeat relays tied here, bypassing all local volume settings at the PA speaker sites and turning them full blast (see Fig. 6). The tuner/studio attenuation relay R2 is also tied across here, hence it is also turned on via T5 in this case. When the intercom is hung up or muted (via R1), IC8 again goes low, and all relays powered by T5 are again turned off.

IC3 thru IC6 are LM567 tone decoders which are used to decode the touch-tones. Refer to any National Semiconductor or competitor manual for detailed information on their function. The inputs of these chips are all coupled directly into the phone line via C37; diodes D9-D11 protect against overvoltage spikes. The outputs of these chips (at pin# 8) clamps to ground when the pre-set tone is detected. Since the outputs are pulled-up by resistors R31-R34, one sees pin# 8 high when the tone is not present, and low when the tone is detected.

All conventional touch-tone signals are made up of two frequency components; one for the row, and one for the column. Since we decode the last row only here (*,0,#), one chip (IC3) detects the common row (hence fires whenever *,0,# are pressed), and IC4-IC6 discriminate the individual characters. Thus the outputs of each of IC4-IC6 are AND'ed with IC3 (via diode pairs D16/D17 (#), D14/D15 (0), and D12/D13 (*)) to determine the unique key pressed; the resultant goes LOW when the key is on. IC7 is a 555 timer here used as a Set-Reset latch driving intercom muting relay R1 (when IC7 is set, R1 is ON, hence the intercom is disconnected from TR3 and the mixer, conversely when IC7 is reset). As plainly seen, the decoded "*" signal sets IC7 (and mutes the intercom), while the decoded "#" resets IC7 via T4 (thus re-connects the intercom). When the intercom is hung-up (all intercom phones are down), the base of T3 goes high, hence pin# 4 of IC7 is pulled low, reseting the latch. Thus, when one hangs up the intercom, the muting action of "*" is also reversed.

The decoded "0" is used to trigger the random tune generator (see below). All tone decoder and muting circuitry is located on the Tone Decoder Card, the intercom decoupling and driving circuitry is located on the Mixer Card, and the L-pad defeat drivers, etc. are located on the Phone-Up Detect Card. This is diagrammed in Fig. 3, and card locations are depicted on the chassis in Fig. 7.

Hopefully the tone decoder center frequencies won't drift much in the future. They have drifted a bit, hence I installed 1% resistors for R23-R26, but they still seem to drift somewhat on occasion. This is noticed when it takes a longer time for a touch-tone button to achieve the desired effect. If the desired effect does not happen, and the touch-tone panel seems dead (this has not happened), the center frequencies must be adjusted. This is done by varying trimmers P13 thru P15 (see Fig. 8). First tune the row (P13/IC3), by playing one of the touch-tones, and tweaking P13 until pin# 8 of IC3 goes LOW. Then try the other tones and their corresponding pin# 8's and tuning pots, until they all respond appropriately. When looking at the voltage on pin# 8 (or any voltage here for that matter), make sure you reference your measurement to the intercom ground (see above)!!

Figure 4 shows the audio and control circuitry associated around the Random Tune Generator (RTG). The tunes themselves are produced by IC10. The basic driving circuit is based around Fig. 1 in the appendix writeup on the AY 3-1350. A tune is requested via the touch-tone "0" (Pt. Z), or S3, both of which bring Pin 2 of IC9 (a 555 also used as a RS latch) to ground, thus setting the output of IC9 high. This powers up IC10, causing it to scan its tune definition inputs and play the appropriate tune. Upon the tune's completion, IC10 asserts pin# 12, which resets IC9, dropping the power, and causing the circuit to wait until the next trigger. Pin 3 of IC9 also drives LED2 (the "tune active" indicator), turns on T8 (which is connected to the L-Pad defeator system), and triggers the timer in IC11. This timer is set for a duration of about 10 seconds, which is longer than the longest tune played. If the output of IC9 is still high after that time (signifying a "lock-up" condition if IC10), pin# 5 of IC11 goes low, and (via C50) resets IC9, clearing the "lock-up". IC11 is normally reset at the conclusion of the tune when pin# 3 of IC9 goes low.

The pitch of the tune voice is adjustable via P19, and the tempo of the tune can be tuned through P20 (see Fig. 8). The intercom ground is isolated via TR4, and P21 and P22 control the tune levels in the mixer. A 7806 regulator is driven from the +15 volt supply to provide a stable +6 volts for this system.

The circuit which randomizes the tune selected by the circuit is outlined in Fig. 5. A tune is uniquely defined in IC10 in one of 2 ways:

1) Grounding one of inputs A-E and routing output N to input 1-4 (or none); see Fig. 10. This generates one out of 25 possible tunes, as one can see in the list in the appendix.
2) Leaving all inputs A-E untouched (inputs 1-4 don't matter here), and grounding input F,G, both, or none. This generates one of the "chimes", as can be seen in the appendix.

The digital randomizer shown in Fig. 5 does this more-or-less at random. Decoder IC14 will ground only one of inputs A thru E according to a three-bit code present at pins 9-11. IC16 is a similar decoder which will route the output at N to inputs 1 thru 4 (or none) according to the three-bit code at pins 9-11. The 3-bit codes come out of 6 stages of shift register IC16. This shift register is fed by digital noise generator IC15, and it is clocked after the completion of every tune (via pin# 3 of IC9). Thus we have a chain of noise bits, advancing after every tune, and picking them at random.....

The chimes are selected with a probability of roughly 1/9 (since there are 3 chimes out of 25 tunes). When the output of astable IC11 goes low, inputs F and G can be grounded, and all inputs A thru E are disconnected via the inhibit on decoder IC14 (at pin# 6). In this case, a chime rather than a tune will be played. Since the duty cycle of IC11 is adjusted to be roughly 9/1, this will happen with roughly the desired probability of 1:9. Cute, eh??

All of this circuitry is located on the Tune Generator Card (see Fig. 8).

III) Trouble Shooting Hints

Since this circuitry is relatively new, there really aren't any. Watch out for bad connections. NEVER connect the intercom ground to the chassis ground. Beware for a monkey from Ma NYNEX yanking out our intercom connections in the phone box. Above all, good luck. You're on your own. (Note: see update comments on next page for additional hints).

Regards,

J. Paradiso, somewhere @ MIT 4/4/84

Updates on the PA Mixer/Intercom Unit 1/87

The PA mixer/intercom unit was initially built in 1977. This was a very simple device, which gradually degraded in performance and was never really properly repaired. The efforts made in 1984 incorporated major revisions, and the device was almost entirely re-built. Few modifications were made thereafter. These modifications were incorporated when this device was moved from Studio A into Master Control in 1985, and are outlined below.

Most new circuitry is outlined in Fig. 3A, which details the EBS and doorbell interrupt card. This circuit card was added (see Fig. 7 for its location) to enable the doorbell and Emergency Brodcast System to activate the L-Pad defeat line, thereby turning all speakers on at full volume whenever a doorbell has been pushed or an Emergency Broadcast System warning has been received. The EBS and doorbell gates are OR'ed together via the two diodes, thereby turning the transistor on whenever at least one gate is received. This activates the on-board relay, thereby shorting point "V" to intercom ground (see Fig. 3) and asserting the L-Pad defeat line. The EBS gate is derived from the relay in the EBS detector unit. The wiper of this relay is grounded, and the normally closed contact is applied via the 10K resistor and diode to the relay's transistor. The normally closed EBS relay contact is pulled up by the 5K resistor to create the gate when the EBS relay has fired (the normally open EBS relay contact is used by the studio switcher to flash the warning LED upon receipt of an EBS signal).

An EBS audio input has been added to the mixer circuitry (see Fig. 2). The designated EBS radio station (WROR @ present) appears at an output on the EBS receiver unit when an EBS warning signal has been detected. This audio is routed to the PA mixer, with level adjusted by a 100K trimmer potentiometer mounted on the EBS & doorbell interrupt card. When an EBS message is received, all PA speakers in the station (except for those in studios with live muting mics.) will blast out the signal from WROR (usually an ominous-sounding announcer gives the standard "this is a test..." spiel, although occasional weather warnings, and maybe someday the "big one" are supposed to come through as well). This is an unusual effect in the halls and offices, where the usual omnipresent air signal is abruptly replaced by WROR's audio. The only way to stop this barrage is to go into Master Control, and reset the EBS detector manually by pushing the appropriate button. This was done to encourage the jock to remember to log the receipt of EBS tests; one doesn't want him to become lazy... (when rebuilding Master Control, we found the speaker on the EBS unit taped up in an attempt to avoid answering its call; this is quite illegal, so we instituted this system to greatly lower the probability of missing EBS brodcasts in the future).

The EBS & doorbell gates and EBS audio are input on a terminal strip located on the EBS & doorbell interrupt card (see Fig. 3A). This card isn't mounted very well onto the chassis, so treat it gently.

Most circuitry has worked nearly flawlessly in this unit since 1984. Most problems are external (ie. in the intercom line, speakers, etc.). The external intercom connections have been added to drive intercom lines in MacPhie Lounge (a modular phone may be directly plugged in near the stage), Curtis Lounge (a modular phone may be plugged in at the WMFO patch box), and the transmitter site at Ballou Hall (a modular phone exists there at all times). At one occasion, the intercom lines to MacPhie shorted, causing the L-Pad defeat to fire, and keeping the intercom engaged. This was determined by disconnecting the external intercom lines one-by-one (they enter on the wall under the window in Master Control) until the offender was found, and complaining to Northern Telecom until our lines were fixed. If the intercom light bulbs remain lit on the telephones, but the intercom is not really engaged (ie. L-Pad defeat doesn't remain active), the error probably is in the circuitry to drive these bulbs (see schematics & writeups for the Intercom KTU and Intercom Bulb Relay), and not in this unit.

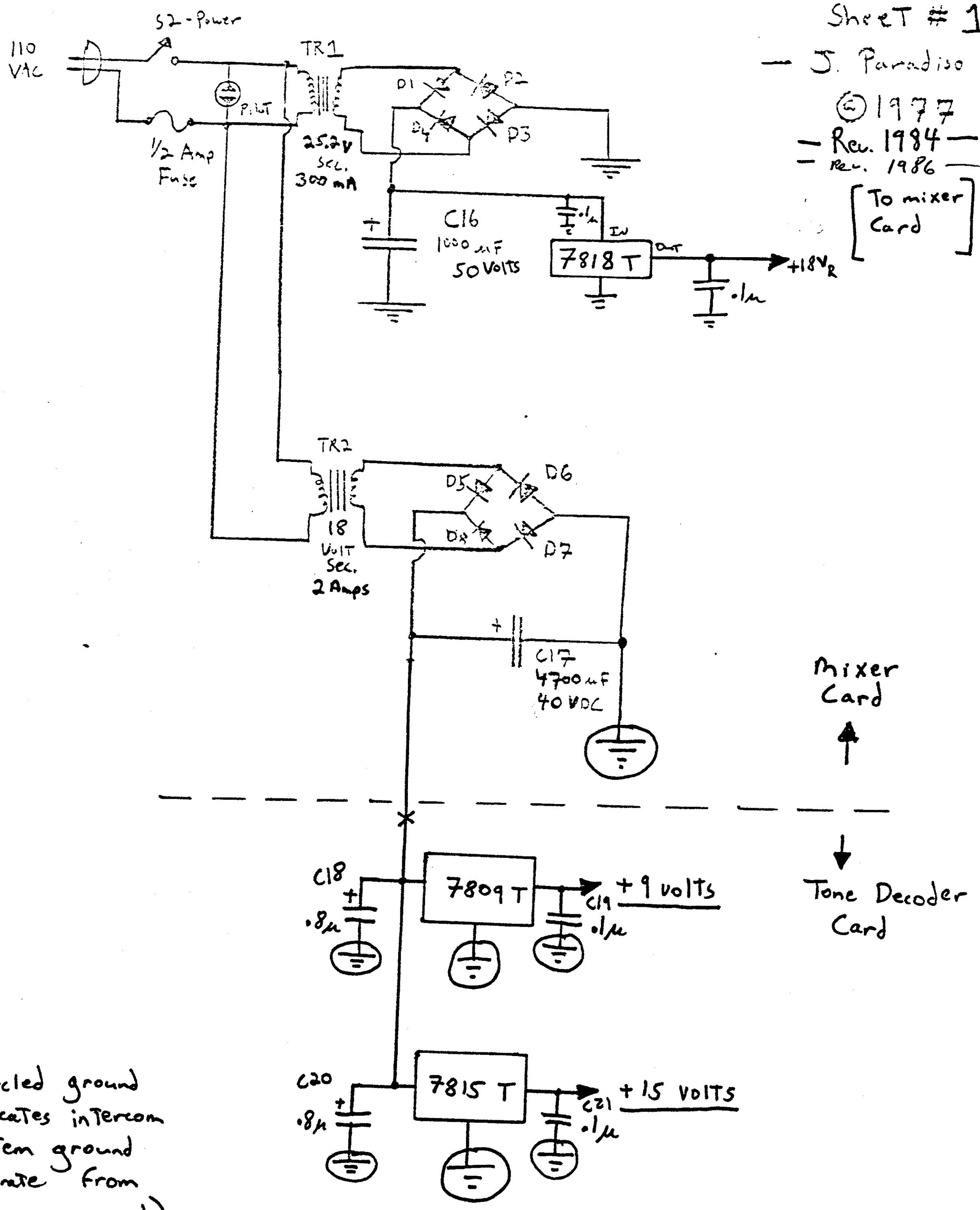
The touch tone detectors for the *,0,# still drift a bit occasionally; this is generally due to large temperature swings in the Master Control room, thus heating or cooling Master Control generally solves this problem (one shouldn't let this room get too hot anyway). If a persistant drift is observed, the tone detectors may be re-tuned as summarized earlier. I've used a simpler procedure in the past, which applies when a single digit (#,0, or *) is acting sluggishly. Under this method, one depresses the offending key, and tunes the appropriate potentiometer (P14 for *, P15 for 0, and P16 for #) until the function is activated. By repeating this operation to obtain maximum response speed, the unit may be quickly tuned. If all functions *,0,# are sluggish, they may be independently adjusted in this manner to obtain the best possible response, and then the composite "row" adjustment (P13) may be tweaked to get them all working promptly. This method of tuning is generally quite quickly accomplished, and doesn't involve poking around the circuit card with voltmeters, etc. One must note that the cheap modular telephones scattered throughout the station produce considerably louder touch tones than the Northern Telecom phones, thus generally evict much faster response on out-of-tune functions. When tuning, one generally should not use these modular phones in order to gain greater sensitivity in the adjustment. In fact, it is generally a good idea to take a few intercom phones off the hook to decrease the touch tone signal level further, and have someone send the desired tones using a Northern Telecom telephone. The tone detectors may be tuned with extreme precision by this method.

Modular telephone jacks have been installed in the Engineering closet (under the table and behind all the junk), Studio D (on the On-Air/PA unit), Master Control (on the grey monitor rack), and the news office/telephone room (on the wall near the phone box). Modular telephones were originally stationed in these areas as well, but some have begun to dissapear via rip-offs.... Any telephone plugged into one of these jacks will function as an intercom.

Figure 1

Power Supply

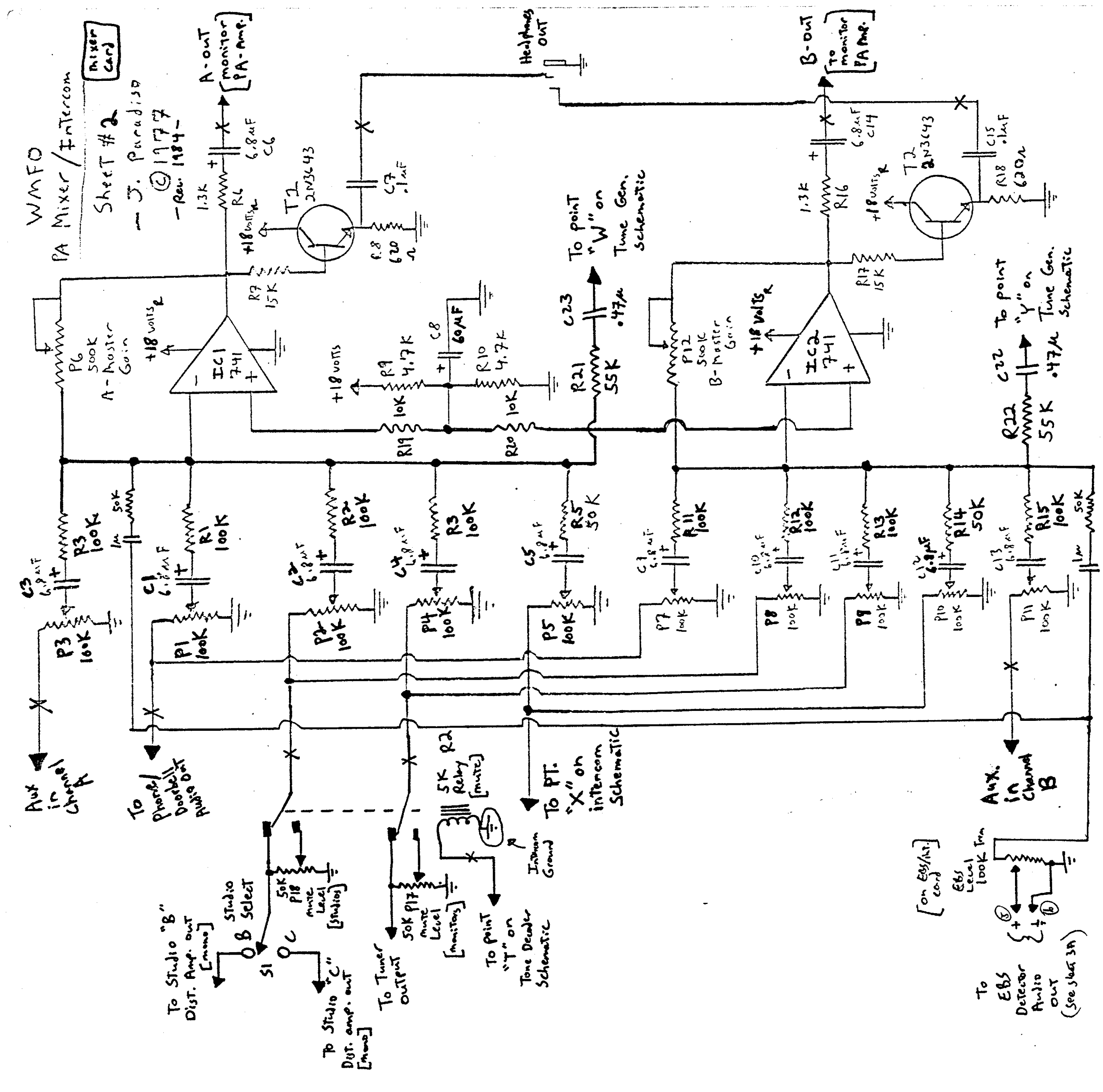
WMFO
PA Mixer/Intercom
Sheet # 1
— J. Paradiso
© 1977
— Rev. 1984 —
— Rev. 1986 —



Circled ground indicates intercom system ground (separate from common ground)

Above regulators are mounted with heat sinks

Figure 2



WMFO
PA Mixer / Intercom
Mixer card
Sheet #2
— J. Paradiso
© 1977
— Rev. 1984 —
A-OUT
[monitor PA-Amp.]
Headphones OUT
B-OUT
[To monitor PA Amp.]
To point "W" on Tone Gen. Schematic
To point "Y" on Tone Gen. Schematic
IC1 741
IC2 741
P6 500K A-Master Gain
P12 500K B-Master Gain
Aux in Channel A
To Phone/Doorbell Audio Out
To PT. "X" on intercom Schematic
Aux. in Channel B
5K Relay [mute]
Intercom Ground
To Studio "B" Dist. Amp. out [mono]
To Studio "C" Dist. amp. out [mono]
Studio Select
To Tuner output
To point "T" on Tone Decoder Schematic
[on EBS/Int. card]
EBS Level 100K Trim
To EBS Detector Audio out (see sheet 3A)

FIGURE 3

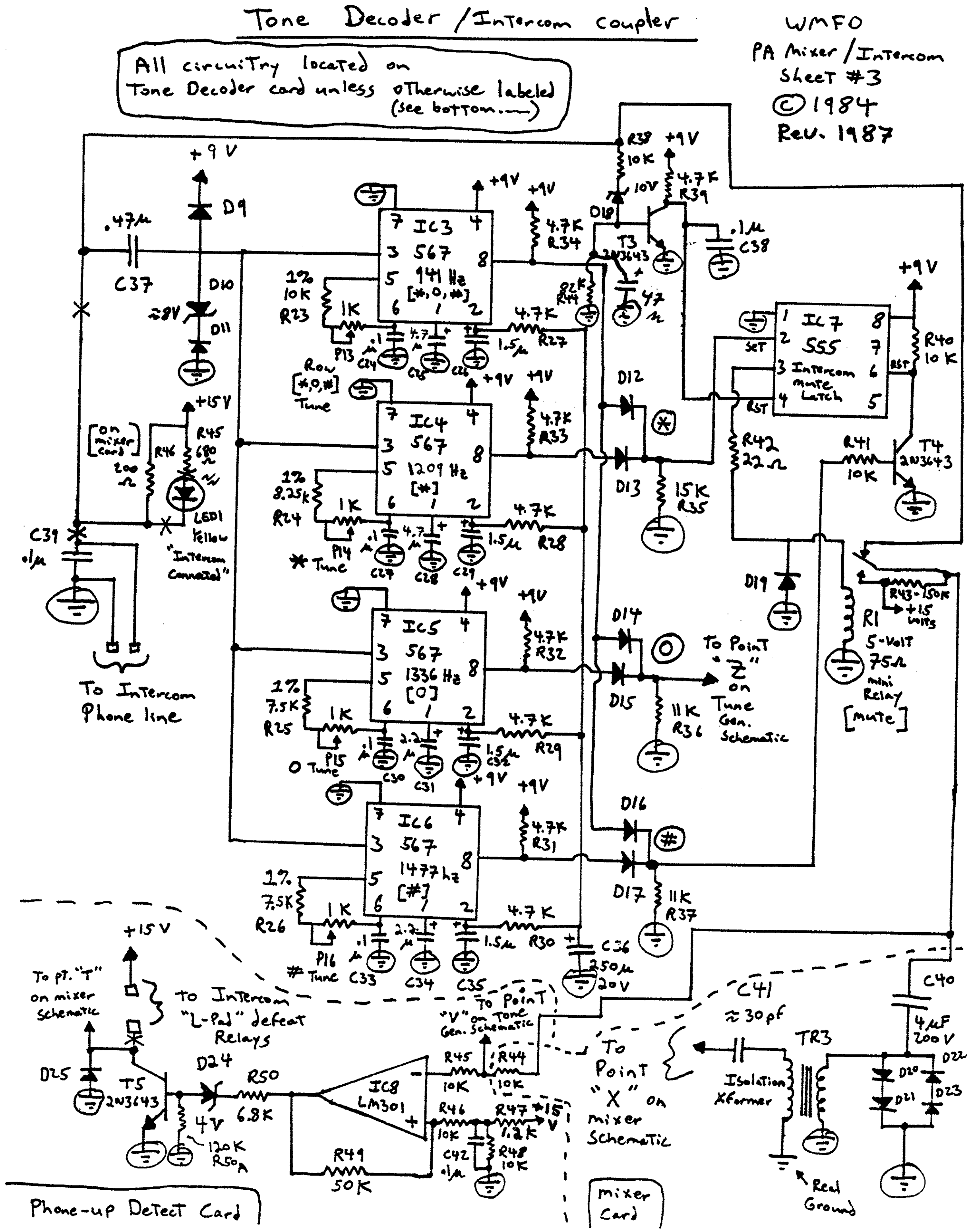
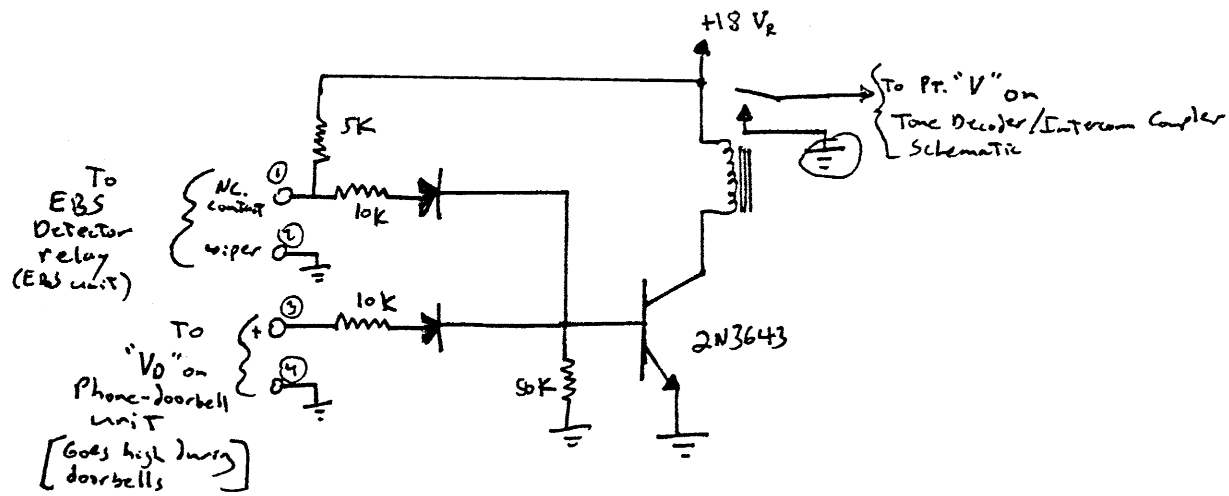

Tone Decoder / Intercom coupler
WMFO
PA Mixer / Intercom
Sheet #3
© 1984
Rev. 1987
All circuitry located on Tone Decoder card unless otherwise labeled (see bottom.....)
IC3 567 941 Hz [*,0,#]
IC4 567 1209 Hz [*]
IC5 567 1336 Hz [0]
IC6 567 1477 Hz [#]
IC7 555 Intercom mute Latch
IC8 LM301
Row [*,0,#] Tune
* Tune
0 Tune
Tune
To Intercom Phone line
"Intercom Connected"
LED1 Yellow
[on mixer card]
To Point "Z" on Tone Gen. Schematic
R1 5-Volt 75Ω mini Relay [mute]
To pt. "T" on mixer Schematic
To Intercom "L-Pad" defeat Relays
To Point "V" on Tone Gen. Schematic
To Point "X" on mixer Schematic
Isolation Xformer
Real Ground
Phone-up Detect Card
mixer Card

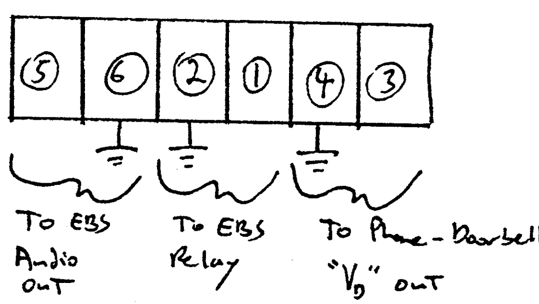
Figure 3A

WMFO
PA mixer/Intercom
Sheet #3A

EBS and Doorbell Interrupt Card

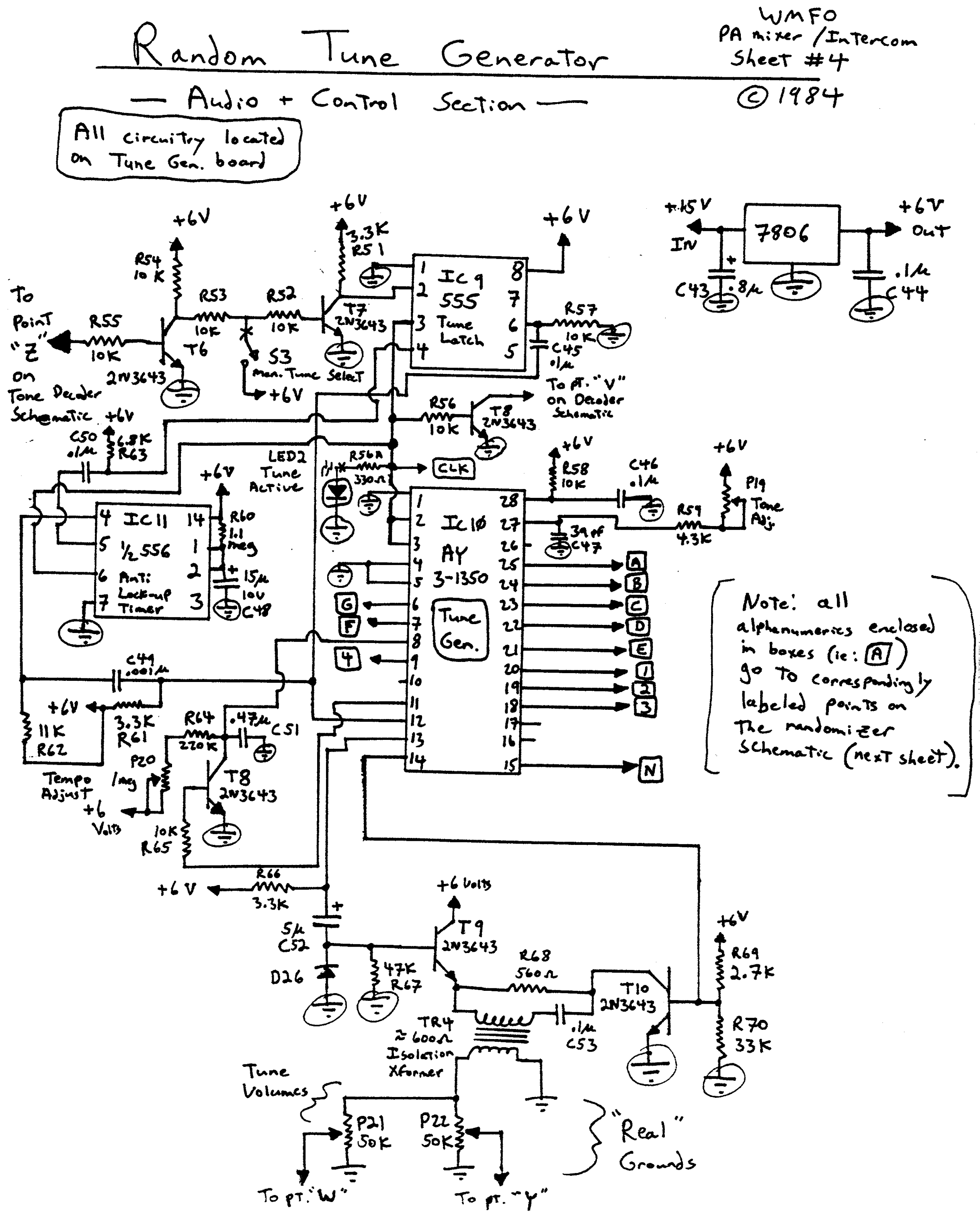


Terminal Block:



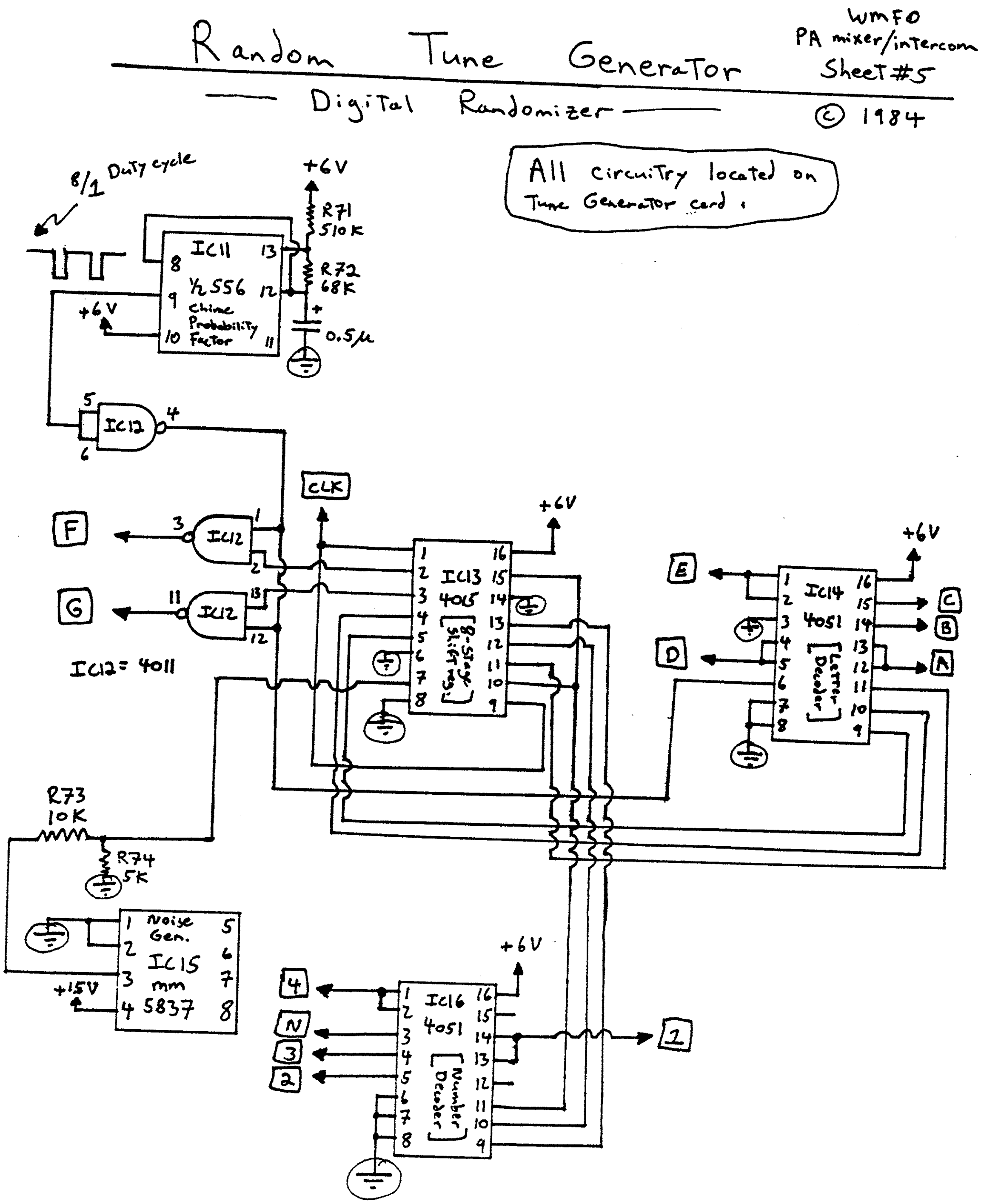
Note: EBS audio level Trimpot also on this card (see sheet 2).

FIGURE 4



WMFO
PA mixer / Intercom
Sheet #4
© 1984
Random Tune Generator
— Audio + Control Section —
All circuitry located on Tune Gen. board
7806
+15V In
+6V Out
IC9 555 Tune Latch
IC11 1/2 556 Anti-Lockup Timer
IC10 AY 3-1350 Tune Gen.
To Point "Z" on Tone Decoder Schematic
To pt. "V" on Decoder Schematic
S3 Man. Tune Select
LED2 Tune Active
Tone Adj.
Tempo Adjust
TR4 ≈ 600Ω Isolation Xformer
Tune Volumes
To pt. "W"
To pt. "Y"
"Real" Grounds
Note: all alphanumerics enclosed in boxes (ie: A) go to correspondingly labeled points on the randomizer schematic (next sheet).

FIGURE 5

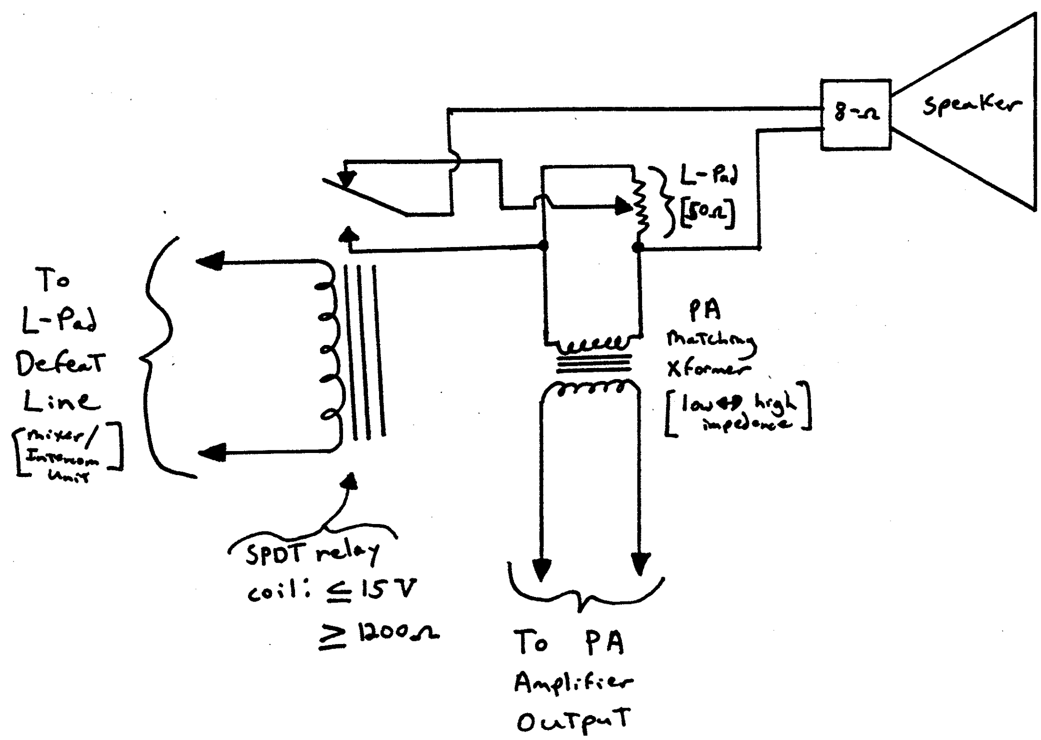


Random Tune Generator
— Digital Randomizer —
WMFO
PA mixer/intercom
Sheet #5
© 1984
All circuitry located on Tune Generator card.
8/1 Duty cycle
+6V
R71 510K
R72 68K
0.5µ
IC11
½ 556
Chime Probability Factor
IC12
IC12 = 4011
F
G
CLK
IC13
4015
[8-Stage Shift reg.]
IC14
4051
[Letter Decoder]
E
D
C
B
A
R73 10K
R74 5K
Noise Gen.
IC15
mm 5837
+15V
IC16
4051
[Number Decoder]
4
N
3
2
1

FIGURE 6

Mixer/Intercom: Speaker Connections

-1984-



Typical connections at each PA speaker....

Figure 7

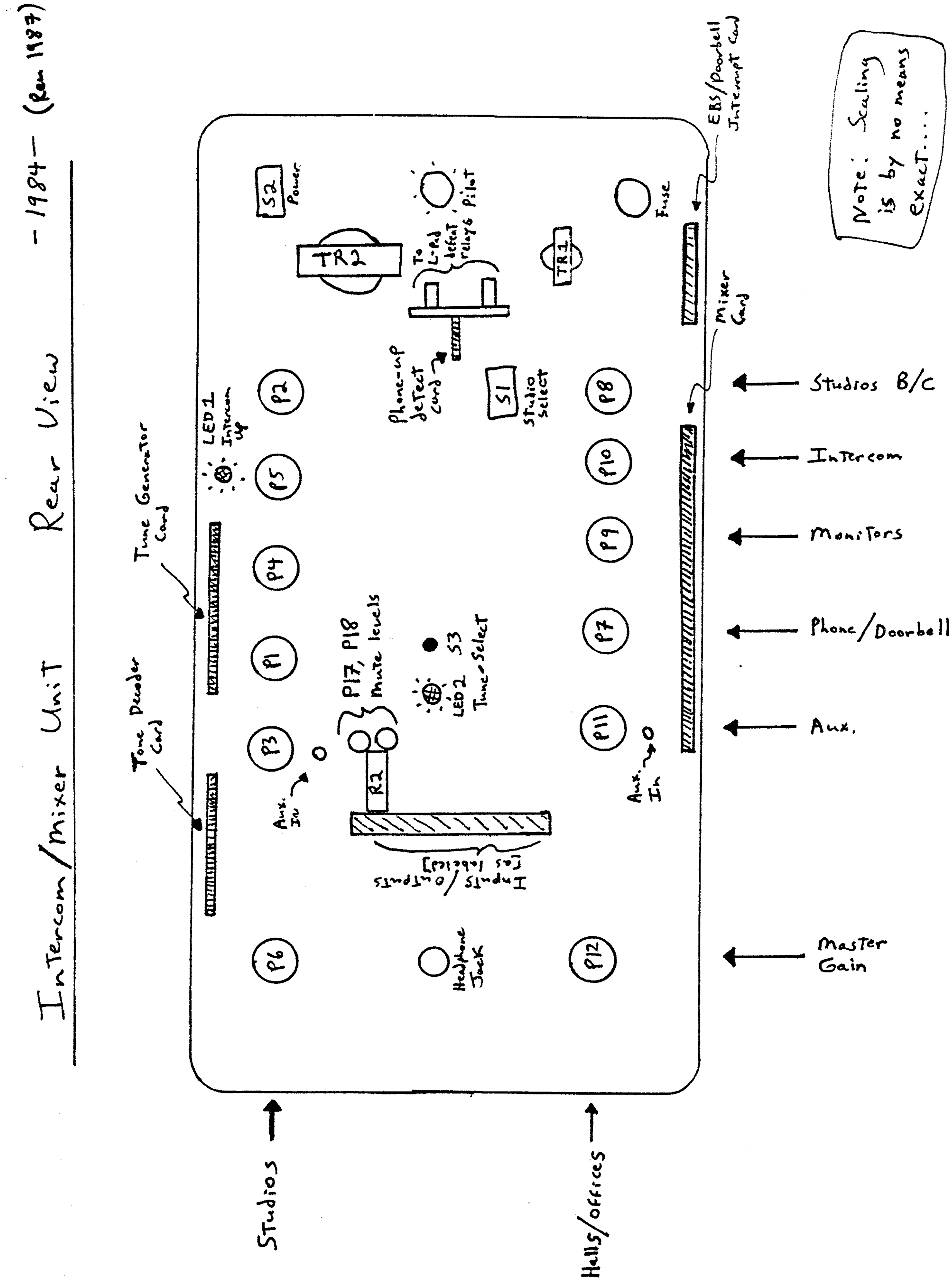
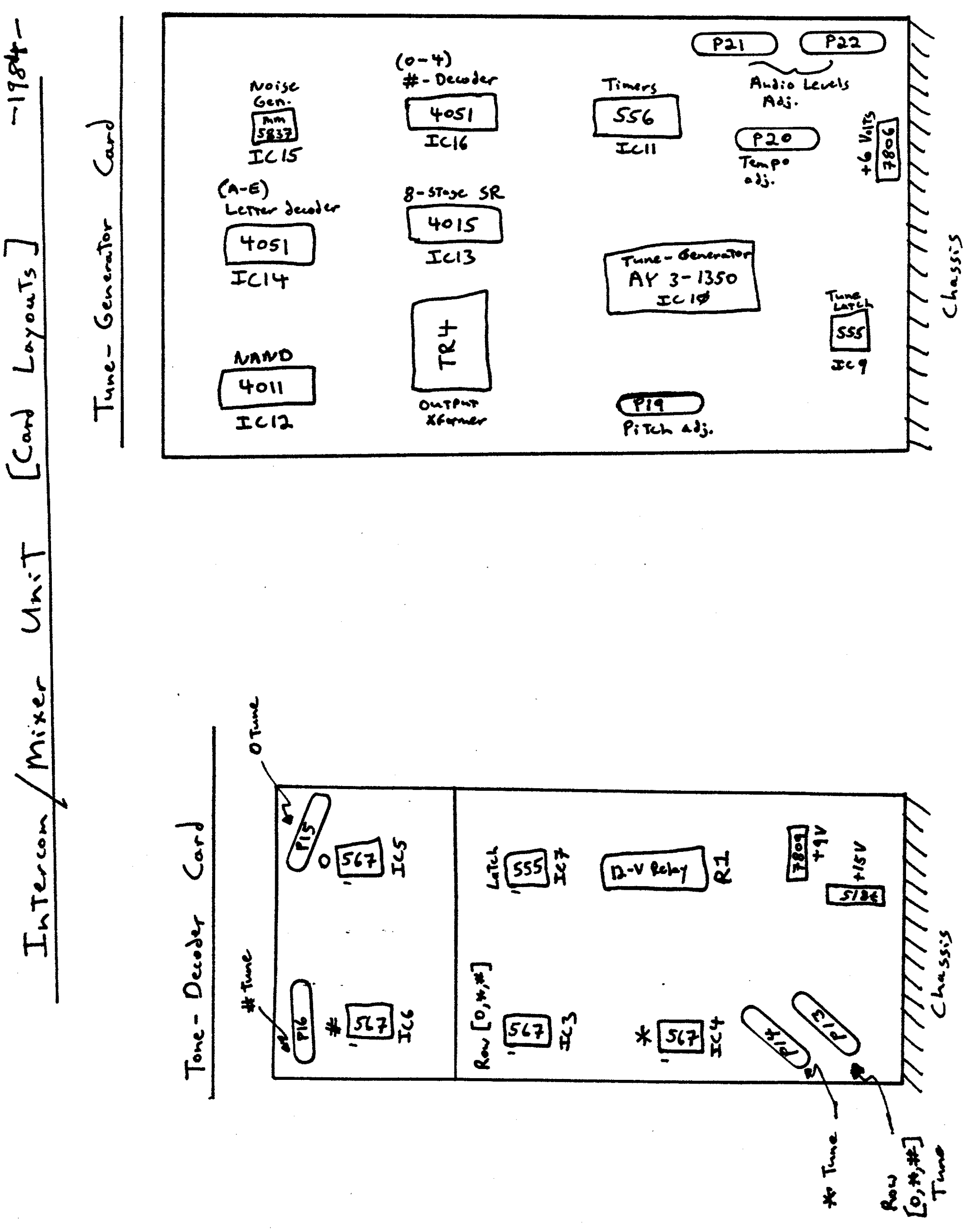

Intercom/Mixer Unit
Rear View
- 1984 - (Rev 1987)
Note: Scaling is by no means exact....
S2
Power
TR2
To L-pad defeat relays
Pilot
Fuse
TR1
EBS/Doorbell Interrupt Card
Mixer Card
Phone-up detect card
S1
Studio select
LED1
Intercom up
Tone Generator Card
Tone Decoder Card
P1
P2
P3
P4
P5
P6
P7
P8
P9
P10
P11
P12
P17, P18
Mute levels
S3
Tone Select
LED2
R2
Aux. In
Aux. In
Inputs/Outputs [as labeled]
Headphone Jack
Studios B/C
Intercom
Monitors
Phone/Doorbell
Aux.
Master Gain
Studios
Halls/offices

FIGURE 8



InTercom/Mixer UniT [Card LayouTs] -1984-
Tune-Generator Card
Noise Gen.
Am 5837
IC15
(0-4)
#-Decoder
4051
IC16
Timers
556
IC11
P21
P22
Audio Levels Adj.
P20
Tempo adj.
+6 Volts
7806
(A-E)
Letter decoder
4051
IC14
8-stage SR
4015
IC13
Tune-Generator
AY 3-1350
IC10
Tune Latch
555
IC9
NAND
4011
IC12
TR4
Output Xformer
P19
Pitch adj.
Chassis
Tone-Decoder Card
0 Tune
P15
0
567
IC5
Tune
P16
#
567
IC6
Latch
555
IC7
12-V Relay
R1
7809
+9V
+15V
5184
Row [0,*,#]
567
IC3
*
567
IC4
P14
P13
* Tune
Row [0,*,#] Tune
Chassis

APPENDIX (AY 3-1350 DATA AND L-PAD CONNECTIONS)

Catalog Number 276-1782

ARCHER®

TECHNICAL DATA

AN EXCLUSIVE RADIO SHACK SERVICE TO THE EXPERIMENTER

AY-3-1350 Melody Synthesizer IC

Features

- 25 Different Tunes Plus 3 Chimes
- Minimal External Components
- Automatic Switch-Off Signal at End of Tune for Power Savings
- Envelope Control to Give Organ or Piano Quality
- Sequential Tune Mode
- 4 Door Capability When Used as Doorchime
- Single Supply (+5V) Operation

Description

The AY-3-1350 is an N-Channel MOS microcomputer based synthesizer of pre-programmed tunes for applications in toys, musical boxes, and doorchimes. It has a set of 25 different popular and classical tunes plus 3 chimes for a total of 28 tunes).

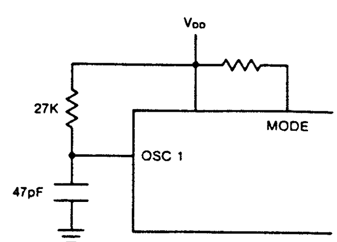
Tunes

A0	Toreador	A3	O Sole Mio
B0	William Tell	B3	Santa Lucia
C0	Hallelujah Chorus	C3	The End
D0	Star Spangled Banner	D3	Blue Danube
E0	Yankee Doodle	E3	Brahms' Lullaby
A1	John Brown's Body	A4	Hell's Bells
B1	Clementine	B4	Jingle Bells
C1	God Save the Queen	C4	La Vie en Rose
D1	Colonel Bogey	D4	Star Wars
E1	Marseillaise	E4	Beethoven's 9th
A2	America, America	Chime X	Westminster Chime
B2	Deutschland Leid	Chime Y	Simple Chime
C2	Wedding March	Chime Z	Descending Octave Chime
D2	Beethoven's 5th		
E2	Augustine		

Absolute Maximum Ratings

Storage Temperature . . . . −55°C to 150°C
Voltage on any pin with respect to ground (V_{SS}). . −0.3V to +10.0V

NOTES:
1. Total I_{OL} for all registers must be less than 150mA under any conditions.
2. Except following pins which have open drain outputs/inputs: 6, 7, 8, 12 and 13.
3. Test circuit:



Pin Configuration

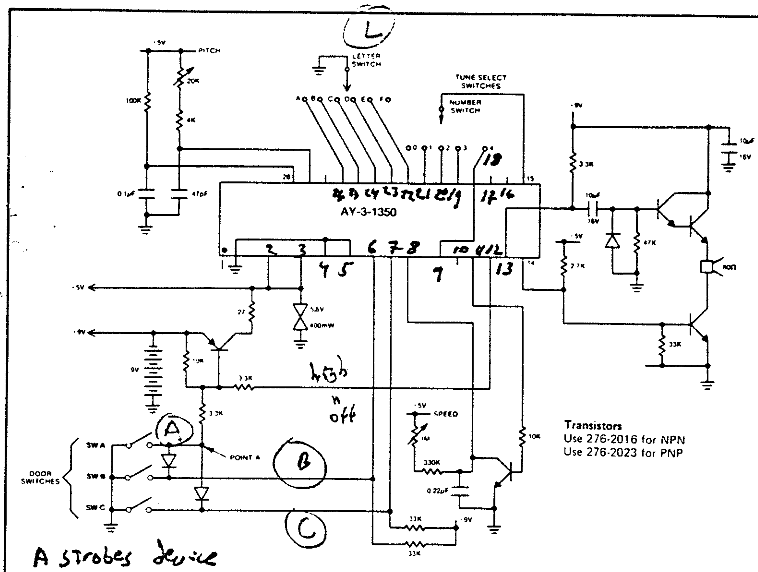
28 Lead Dual In Line

Top View

Pin	Name	Pin	Name
1	GND	28	RESET
2	Vcc	27	OSC
3	Vxx	26	CLKOUT
4	GND	25	Tune Select A
5	GND	24	Tune Select B
6	Door 3	23	Tune Select C
7	Backdoor	22	Tune Select D
8	Captest	21	Tune Select E
9	Tune Select 4	20	Tune Select 1
10	Next Tune	19	Tune Select 2
11	DISCRG	18	Tune Select 3
12	On/Off	17	RESTART
13	Envelope	16	Switch C Group Select
14	Tune Output	15	Tune Select Strobe

Recommended Operating Conditions

Characteristics	Sym	Min	Max	Units	Conditions
DC Characteristics					
Primary Supply Voltage	V_{DD}	4.5	7	V	
Output Buffer Supply Voltage	V_{XX}	4.5	9	V	
Primary Supply Current	I_{DO}	–	55	mA	No load
Output Buffer Supply Current	I_{XX}	–	5	mA	No load
Logic Input Low Voltage	V_{IL}	-0.2	0.8	V	
Logic Input High Voltage (Note 2) (Except RESET and OSC when driven externally)	$V_{IH}1$	2.4	V_{DD}	V	
Logic input High Voltage (RESET and OSC)	$V_{IH}2$	4	V_{DD}	V	
Logic Output High Voltage (Note 2)	V_{OH}	2.4		V	I_{OH}=100 µA
Logic Output Low Voltage	V_{OL}	–	0.45	V	I_{OL}=1.6µA, V_{XX}=4.5V
	–	–	0.90	V	I_{OL}=5mA, V_{XX}=9V
	–	–	0.50	V	I_{OL}=5mA, V_{XX}=9V
	–	–	0.90	V	I_{OL}=10mA, V_{XX}=9V (Note 1)
AC Characteristics					
Oscillator frequency variation for a fixed RC network	Δf	–20%	+20%		@CLK OUT 167KHz (Note 3)
CLK OUT Output Period	t_{CY}	4	20	µs	
High Pulse Width	t_{CLKH}	¼ t_{CY}			
Low Pulse Width	t_{CLKC}	¾ t_{CY}			



A strobes device
B ⏚ → octave
C ⏚ → simple   L unconnected → Westminster

Fig. 1 SYSTEM DIAGRAM

Operational Notes

There are many ways to connect the AY-3-1350 depending on the exact application. Figure 1 shows just one implementation of the device in a doorchime. This circuit gives access to all 25 tunes from switch A and one of 5 tunes from switch C as well as the descending active chime from switch B. The tune selected for switch B follows the tunes list according to the setting of the two tune select switches (A—E and 0—4). The tune selected from switch C in Figure 1 is one of the five tunes A0 through E0 depending on the setting of the letter switch. For example, with the letter switch set at E and the number switch set at 4, the tunes available will be:

Switch A: Beethoven's 9th (E4)
Switch C: Yankee Doodle (E0)
Switch B: Descending Octave Chime (Chime Z)

When the letter switch is in position F there will be chimes on all doors independent of the number switch setting as follows:

Switch A: Westminster Chime
Switch C: Simple Chime
Switch B: Descending Octave Chime

There is virtually no power consumption in the standby condition (external transistor leakages only). When any door switch is activated the circuit powers up, plays a tune, and then automatically powers down again to conserve the battery, even if the operator keeps his finger on the switch to the end of the tune. He must release it and re-press to play again with the circuit in Figure 1. Activating any of the door switches will pull point A to ground turning on the PNP transistor in the power supply line. This causes +5V to be applied to the AY-3-1350 and the first operation of the chip is to put ON/OFF (pin 12) to logic 0. This maintains the power through the PNP, even after the switch is released. The device can turn off its own power at the end of a tune by raising ON/OFF to logic 1.

Figure 1 shows only a typical one-chip implementation. Further options come from use of different switching and/or from use of the next tune facilities built into the chip. These will now be considered in turn.

Switching Options

In Figure 1 the Switch C Group Select pin (16) is not connected, and one of the five tunes (A0 through E0) will play if switch C is activated. Other number groups can be chosen by connecting the Switch C Group Select pin as follows:

RADIO SHACK, A DIVISION OF TANDY CORPORATION

U.S.A.: FORT WORTH, TEXAS 76102
CANADA: BARRIE, ONTARIO L4M 4W5

TANDY CORPORATION

AUSTRALIA	BELGIUM	U. K.
91 KURRAJONG ROAD MOUNT DRUITT, N.S.W. 2770	PARC INDUSTRIEL DE NANINNE 5140 NANINNE	BILSTON ROAD WEDNESBURY WEST MIDLANDS WS10 7JN

Table 2

Switch C Group Select pin (16) is connected to:	Switch C Tunes
no other pin	A0—E0
Tune Select 1 (pin 20)	A1—E1
Tune Select 2 (pin 19)	A2—E2
Tune Select 3 (pin 18)	A3—E3
Tune Select 4 (pin 9)	A4—E4

Which of the five possible switch C tunes will be played depends on the current setting of the LETTER SWITCH A—E.

Switch C selection can be made by hard-wire connection for a permanent selection or a third switch can be added for an additional group selection feature.

LED Direct Drive

V_{XX} drives the gate of the output buffer, allowing adjustment of drive capability:

V_{XX}	V_{OUT}	I_{SINK} (typ.)
5V	0.4V	2.5mA
5V	0.7V	4.2mA
10V	0.4V	5.8mA
10V	0.7V	10.0mA
10V	1.0V	14.1mA

Using the power-up circuit of Figure 1, the AY-3-1350 will have +5V applied and be latched within a few microseconds (dependent upon external components) from any bell-push closing. The device starts to operate when the RESET pin reaches logic 1 (about 10ms with components shown) but in fact the tune select switches are not interrogated until approximately 6ms later. The total is sufficient for most bell-pushes to complete any bounce period and for a firm selection of tunes to be made.

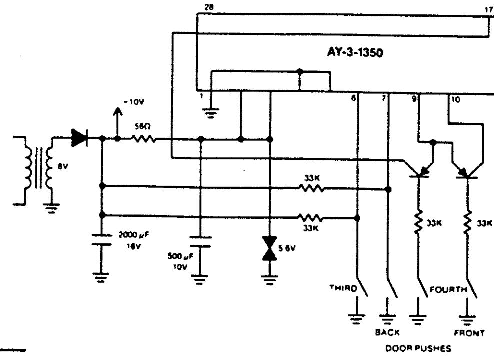
Next Tune Facilities

At the end of tune play the circuit of Figure 1 powers down because ON/OFF (pin 12) is raised to a logic 1. Before the power down there is a test for connection between NEXT TUNE (pin 10) then RESTART (pin 17) with TUNESELECT 4 (pin 9). At this time NEXT TUNE (pin 10) then RESTART (pin 17), which are normally at logic 1, output a logic 0. This is looked for at input TUNESELECT 4 (pin 9). If neither is found the power down system is reached as in Figure 1.

A NEXT TUNE (pin 10) — TUNE SELECT 4 (pin 9) connection at the moment of test causes the next tune in the list to be played after a short pause (equal to a musical breve—the actual time depends on the setting of the tune speed control). The order of the tunes is A0 to E4 as given in the listing of standard AY-3-1350 tunes. If the last tune (E4) was played then the circuit will go on to play the first tune A0 (and then successive ones). The chimes are not included in the cycling sequence.

A RESTART (pin 17) — TUNESELECT 4 (pin 9) connection at the moment of test at the end of a tune causes the same selected tune to be played again.

The connections referred to cannot be permanent because otherwise the circuit would never stop playing tunes. Figure 2 shows how transistors are used to make the connection in a practical application.

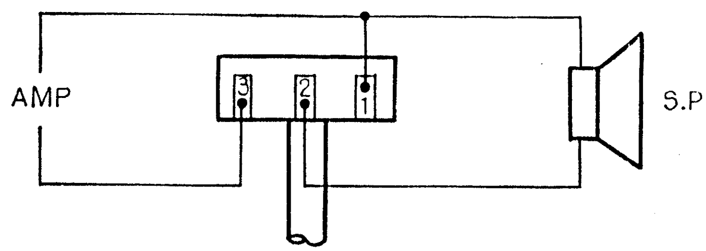


PIN	FUNCTION
6	DOOR 3
7	BACKDOOR
9	TUNESELECT 4
10	NEXT TUNE
17	RESTART

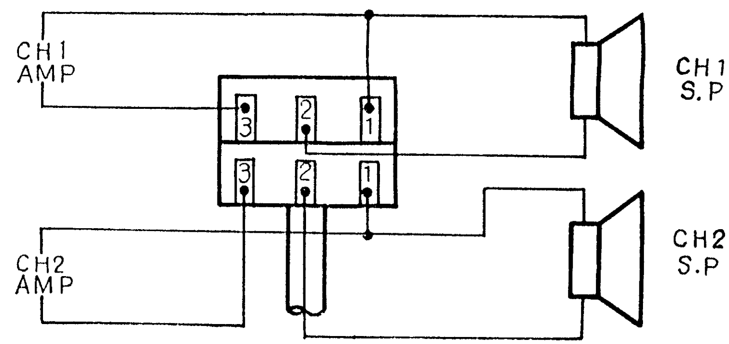
Fig. 2

L-PADS LEVEL CONTROL

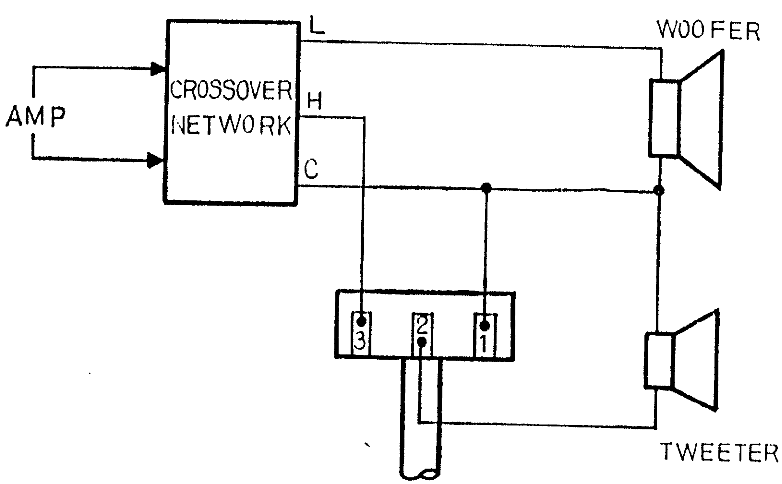
LEVEL CONTROL



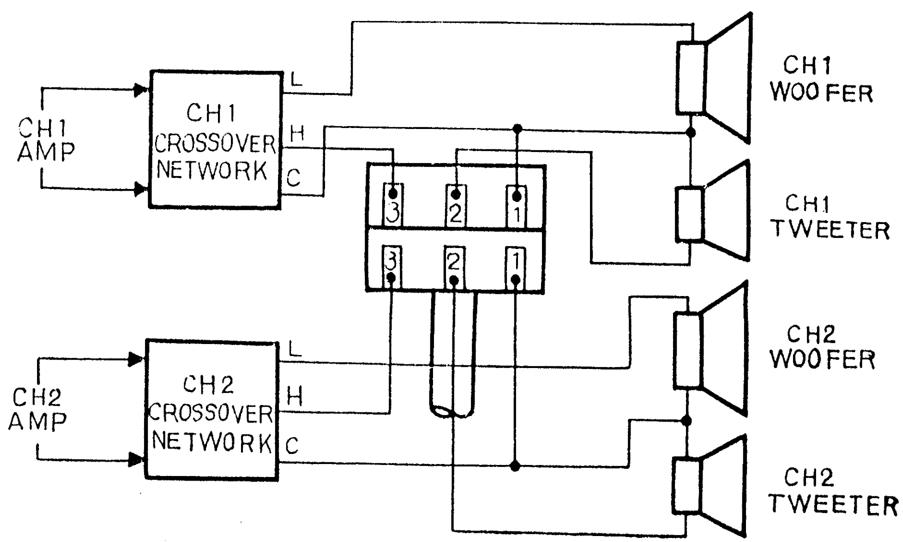
STEREO LEVEL CONTROL



2-WAY BALANCE CONTROL



STEREO 2-WAY BALANCE CONTROL



PRINTED IN JAPAN

THE WMFO STUDIO D PA/ON-AIR BOX

INPUTS:

* Intercom line
* Remote On-Air Switch

OUTPUTS:

* Modular Jack for Intercom Telephone
* 8-Ohm Speaker Output
* Remote On-Air LED
* 110-VAC Socket for ON-Air Light
* Intercom Speaker Mute
* Touch-Tone Testpoints TPR & TPC

THE STUDIO D PA/ON-AIR BOX

I) Functions and Operation:

This contraption is the aluminum box mounted on the wall of Studio D (opposite Studio B) with loads of cables running into it and a speaker hung alongside. The purpose of this device is to provide a talkback facility (from any intercom telephone) and to drive the ON-AIR warning light & speaker muting for the "talk" studio (ie. Studio D). Each portion of this device is described separately below:

This circuit is connected to the common WMFO intercom line, and listens for the "9" touchtone. Whenever a "9" is pressed on any intercom phone in the station, this circuit will activate an internal amplifier, and effectively connect the intercom audio to an attached speaker (with considerably loud volume, allowing one a chance to be heard above noisy rock bands). By talking into an intercom phone so connected, one is heard directly in Studio D. There is no muting of any sort on this speaker, so the connection takes place regardless of the air status of the studio. If one desires to speak to Studio D exclusively, and not be heard elsewhere in the station, one should first press the intercom "*", thereby muting the intercom from all of the PA speakers (as discussed in the "PA Mixer/Intercom writeup), and then press "9" for unique connection to Studio D. The speaker is turned off when all intercoms are again hung-up, or when the red "reset" button on the front panel is pressed. A green LED on the front panel is illuminated when the speaker is connected. If one wishes to speak with someone in Studio D over the intercom, the normal practice is to hit "*9" to connect exclusively to the Studio D speaker, and ask for the desired person to pick up the phone; as soon as he answers the Studio D intercom (if not before), this person should hit the "reset" button to turn the speaker off and prevent a blast of feedback from occuring when his intercom is picked up.

This device has exhibited some temperature sensitivity, and occasionally is sluggish in responding to touch tones. If the "9" function is sluggish or doesn't work, the tone decoders may need to be retuned, as described later in this report. Of course, at this point in time, the touch-tone drivers in most of the multi-line phones at WMFO have been disconnected as a matter of station policy, thus none of the touch-tone functions (including the "9") can be accessed from these devices.

The other function of this box is to manage the ON-AIR light switching and PA speaker muting for Studio D. The studio may be placed (or removed) from AIR status by toggling either the ON-AIR switch on the front panel of the box in Studio D, or the corresponding switch mounted on the wall of Studio B. These switches are wired in an exclusive OR, thus they function as dual light switches in conventional home wiring; ie. toggling any switch reverses the ON-AIR state of the studio, thus allowing one to take the studio on or off air from Studio B (where most recording is done) or Studio D itself. When Studio D is placed

ON-AIR, the warning light over the studio door is activated, and the PA speaker in the studio ceiling is muted (preventing even intercom pages from reaching Studio D; of course the "9" function described above still remains active).

II) Circuit Description:

The first page of the schematic diagram shows the power supply and circuitry for the "9" function. The intercom line is constantly monitored by two 567 tone decoders IC1 & IC2. IC1 is set to decode the touch-tone row, and IC2 is set to decode the touch-tone column. The outputs of these devices (at pin 8) clamp to ground when the appropriate tone is decoded (as set by trimmers T1 & T2 in concert with the 4.7 mfd capacitors). These outputs are diode OR'ed by D4 and D5 before being applied to pin #2 of the 555 latch IC3. When both tones are detected, pin #2 of IC3 goes low, which latches the output of IC3 (pin #3) high. IC3 is reset when pin #4 is brought to ground. This happens when either the reset switch (S2) is pushed, or transistor Q1 is turned on. Q1 monitors the intercom-off-hook status. Its base is driven high by the intercom line whenever the intercom voltage is above 11.5 volts (ie. when all intercoms are hung-up and the intercom line is un-engaged). This holds IC3 in reset, and prevents the latch from being driven high. When an intercom is off-the-hook, however, the intercom line voltage drops significantly, preventing any current flowing through D6 & D7, and turning Q1 off. This allows IC3 to be set by the appropriate touch tones.

The intercom line is amplified by the programmable power OP-AMP IC4 (which is an LM13080). Its gain is set by trimmer T3, and its output is buffered by power transistors Q3 & Q4 in order to allow an 8-ohm speaker to be driven at appreciable volume. In the quiescent case (when the output of IC3 is held low), zener D8 does not conduct, holding Q2 off, and inhibiting the amplifier from operating (its output is held at ground). When IC3 goes high (ie. after a "9" has been received), D8 conducts, Q2 is turned on, and the amplifier IC4 becomes functional, routing the intercom audio to the speaker output. LED #1 is driven by the output of IC3, thus provides a visual prompt for the "speaker connected" state. The intercom line is also routed to a modular jack mounted on the side panel of this device in order to enable a intercom phone to be directly attached.

The second page of the schematic diagram depicts the circuitry for the ON-AIR portion of this device. Components D9-D12 and Q5,Q6 form an exclusive OR gate, whose inputs are derived from the ON-AIR toggle switch mounted on the front panel and the remote ON-AIR toggle switch mounted in Studio B. By exclusive ORing these switches, one may reverse the ON-AIR state via throwing either switch (the remote switch is a clamp to ground at the tie-point "S" in the schematic). Q5 buffers the exclusive OR output, driving local ON-AIR LED #2 and a remote LED which attaches between tie-point "L" and ground. When the exclusive OR goes high, Q7 is also turned on, which activates the 14 Volt relay. One set of contacts on this relay apply 110 VAC to the ON-AIR socket mounted on the top of the unit (into which a line is plugged that runs to the ON-AIR light mounted atop the Studio D door). The other set of relay contacts are used to interrupt the PA line running to the ceiling intercom speaker, effectively "muting" it (the speaker driven by the "9" function remains unaffected).

III) Calibration Procedures:

The only portion of this circuit requiring calibration is the "9" detector, as portrayed on the first page of the schematic and described above. The frequencies set on the tone decoders IC1 & IC2 may drift, and may be re-adjusted by tweaking trimmers T1 and T2. These (along with tip jacks for test points TPR & TPL) are mounted on the right side of the device, and may be accessed without removing the unit from the wall. The row and column settings should be calibrated separately. First (starting with the row), monitor the voltage between TPR and ground with a voltmeter. Press a touch tone key on the "9" row other than "9" (ie. use 7 or 8). Adjust T1 while the touch tone sounds to get zero volts at TPR (it should have quiescently been at 7 volts or so). Repeat this procedure until the signal at TPR responds promptly to the touch-tones (to increase the sensitivity of this adjustment, several intercom telephones should be off the hook, thereby lowering the amplitude of the touch-tone signals). Once the Row has been calibrated, the process can be repeated for the column (ie. hit a column key other than "9" ["3" or "6"], monitor TPL, and adjust T2). After this process is completed for both row and column, the unit is calibrated, and should respond promptly to the "9". One important note; before trying to calibrate the tone decoders, make sure the unit has been powered on for several hours to avoid any transient effects.

Because the range of adjustment is limited (due to the series resistors together with the 1K range of the trimmer), the device doesn't always want to calibrate to the "9". There are two potential solutions to this problem. If possible, the unit may be able to be tuned to another number (ie. "8", "6", or "5"; the *,#, and "0" are already used by the intercom system). Otherwise, one could pull the box off the wall in order to lower the series resistors and increase the values of T1 and T2. I never bothered to attempt the latter modification myself, due to other pressing engagements. One could also replace IC1 and IC2 by a crystal-locked touch-tone decoder IC (which have only recently become available at Radio Shack, etc.). Just a few possibilities.....

The only other calibration is the amplifier gain set by trimmer T3. This trimpot controls the volume of the speaker. It is already set to be adequately loud, and probably will never need readjustment. It is located on the circuit card inside the box (as shown in the 3'rd page of the schematic).

IV) Practical Considerations and Troubleshooting Hints

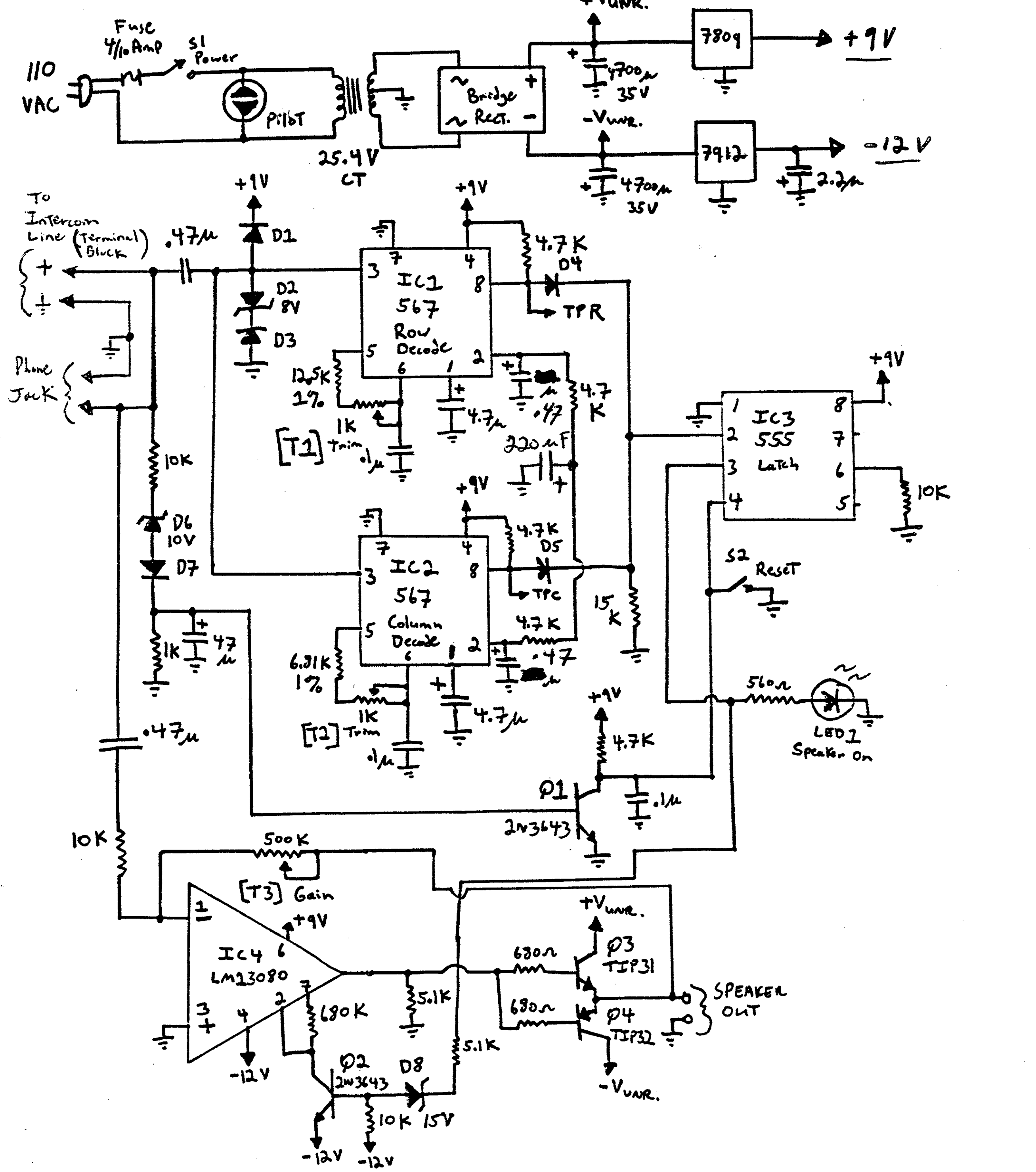
If one must take this box off the wall, watch out for unprotected 110 VAC contacts on the relay, power switch, fuse, etc.; it's really easy to get zapped if you're careless (the voice of experience). Before one decides to take any measures, first make sure that the unit is getting AC power; the pilot light is a bit dim, and people occasionally turn it off (which is strictly forbidden).

If no audio is coming out of the speaker after the "9" is engaged (and the green LED #1 goes on indicating command acceptance), first check that the speaker is connected properly; the connections at the speaker aren't extremely secure, and people occasionally mess with these things. If this is OK, and audio is still missing, the problem may be in the IC4 network. If Q2 & D8 are working OK and Q3 & Q4 are fine, the problem may indeed be a defective IC4. This has never happened before. If, however, IC4 has to be someday replaced, one must bear a few important pointers in mind. When building this circuit, I noticed that not all LM13080's disconnect their output (allowing the 5.1K resistor to bring it to ground and turn off Q3 & Q4) when disabled at pins 7 & 2 via Q2. I had to try several devices until I found some that worked this way. This is important; if there is a bias on the output of IC4, the transistors will quiescently load DC voltage into the 8-ohm speaker, causing things to become hot and unhappy. When checking different LM13080's for this effect, disconnect the speaker, thus unloading this output to prevent such undesired heating.

The maximum power supply voltage recommended for the LM13080 is 15 volts. Because of the +9 and -12 volt rails on the power supply in this circuit, the LM13080 is being driven approximately 6 volts above its rating (which may explain some of the anomolies noted above). This has not caused any apparent difficulty so far, and the device has been functioning flawlessly for over two years.

If the intercom line is connected backwards, Q1 will be unable to reset IC3 when all intercoms are hung-up, leaving the speaker connect and LED #1 illuminated until the reset button is pressed. It's connected properly now, but this should be monitored in case the intercom is monkeyed with.

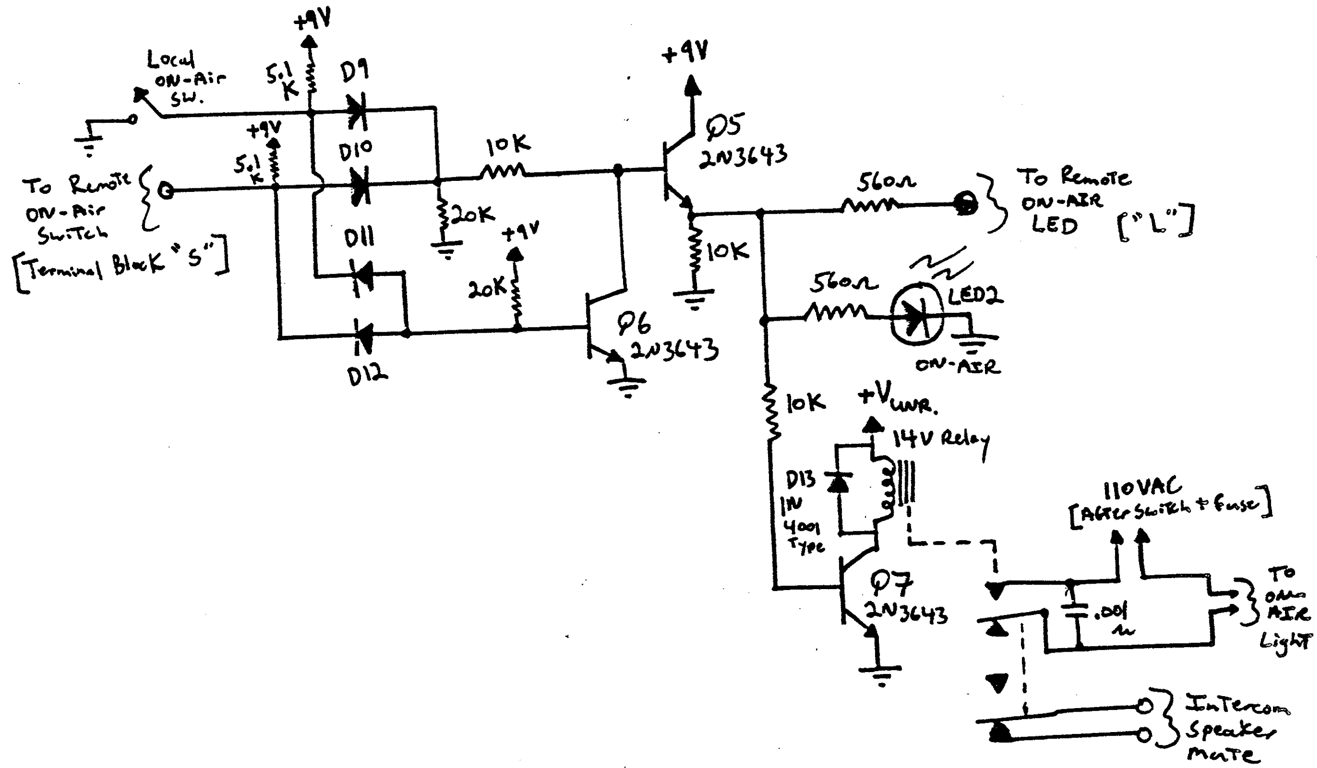
All other trouble-shooting should be fairly trivial; there aren't too many tricks applied in the other circuitry. The only problems which I've had in this circuit have always been traced to bad connections on the PC card (which is a bit of a mess). These have all been located and repaired years ago, and the device has been working fine since. Hopefully it continues this tradition (knock on aluminum)....



The WMFO "STudio D PA/ON-AIR" Box

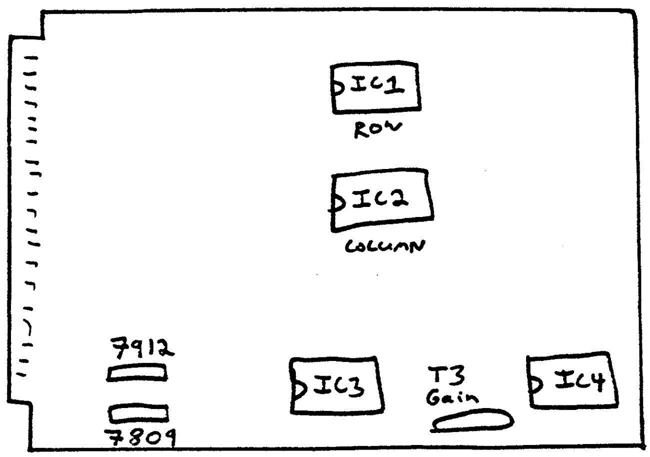
Sheet 1: PA Dial-up Amplifier + Power Supply

Sept., 1984 -JAP-



The WMFO "Studio D PA/ON-AIR" Box Sheet #2: ON-AIR Logic

Sept. 1984
—JAP—



TOP VIEW

The WMFO "STudio D PA/ON-AIR" Box : Card LayouT

THE WMFO MUSIC-ON-HOLD DRIVER AND INTERCOM BULB RELAY

INPUTS:

* 600 Ohm Balanced Audio Input (Left)
* 600 Ohm Balanced Audio Input (Right)
* Intercom Line
* Intercom L-Pad Defeat Line
* Voltages from Phone Box:
  -26 Volts, Ground, 10 Volts AC, BG

OUTPUTS:

* Music-On-Hold Audio (3 channels)
* "L" line for circuit #5 (intercom)
* Modular Jack for Intercom Phone

THE MUSIC-ON-HOLD DRIVER AND INTERCOM BULB RELAY

I) Function and Operation:

This device is located in the room with the main Northern Telecom Box (ie. it was once called the "News Office"). It is a small black plastic box with aluminum cover mounted on the wall next to the phone box. This box contains circuitry to drive the any (or all) of the three telephone lines with an audio signal when a caller has been put on hold. The audio lines which feed this device originate from Master Control, where they are normalled to the external monitor feed via the patchfield (one can conceivably override this patch to supply callers on hold with the audio of your choice). Three clearly labled toggle switches are mounted on the side of this plastic box which connect or disconnect the telephone lines from the music-on-hold feed. If any of these switches are OFF, the corresponding callers will not hear any audio while on hold. These switches should normally be left ON, so that the music-on-hold will be always working. They were installed so that this feature could be defeated when necessary (ie. when someone "important" with fragile moral stature calls during a potentially offensive on-air program). If any are turned off for any reason, remember to turn them on again afterward.

This circuit also contains a relay which connects to the phone box in order to illuminate the intercom bulbs (in multi-line phones) when the doorbell rings, the EBS system is activated, a paging "tune" is playing, or the intercom is engaged and connected to the speakers (this last task is performed a bit more thouroughly by the Intercom KTU mounted inside the master telephone box; see its writeup for more details). A modular jack is connected to the intercom line via this device, enabling an intercom telephone to be located in this room, thus allowing its occupants to communicate with the rest of the station.

II) Circuit Description:

This circuit is relatively straightforward, as seen in the one-page schematic diagram. The power supply is a regulated 18 Volts drawn from the -26 Volt main supply of the telephone box. One important thing to keep in mind here is the polarity reversal of voltages in this circuit relative to those in the Northern Telecom phone box. The -26 Volts is used as GROUND in this circuit, and the Telecom ground is used as an effective +26 Volts. This means that the ground of this box is at -26 Volts relative to the Telecom Box (and the rest of the radio station!!!). Thus, when hooking up test equipment, etc. DO NOT SHORT THE GROUND OF THIS DEVICE TO ANY REAL GROUND!!!! This is also true when hooking up the shielded audio inputs; DO NOT CONNECT THE GROUNDS OF THESE CABLES TO THE GROUND ON THE BOX!!! If the "grounds" are connected in either of these scenerios, the -26 Volt main supply for the Telecom circuitry will be shorted out, causing a fuse to blow in the Telecom Box, and knocking out all telephones in the station. So avoid this problem, and don't short out this "ground".

Separate left and right balanced audio pairs are run from Master Control and input to differential amplifiers IC1a and IC1b (granted, one pair would have been sufficent for a mono phone line, but we decided to leave room for future flexibility). These OP-AMPs convert the balanced audio inputs to single-ended signals. IC1 is a 1458 dual OP-AMP; not the best choice for quality, but certainly amply adequate for the telephone lines. Everything is AC isolated via C1-C4 since all OP-AMPS are referenced to a 9 Volt bias via R11 & R12. The telephone lines are driven via OP-AMPS IC2-IC4. These are 5534's, and are used here because of their high (100 mA or so) output current rating. The gain of each driver is adjustable via trimmers T1-T3. The output of each 5534 is AC isolated (because of the 9 Volt bias), and fed to line transformers TR1-TR3 via the defeat switches S1-S3 (which are mounted on the side of the box, as discussed above). These transformers isolate the telephone lines from the local ground. The 4 mfd capacitors (C13-C15) block the DC telephone voltage. High Voltage surge protectors (conventional Radio Shack VDRs) protect the circuitry from excessive transients and spikes (these lines do not have any ring signal on them).

The intercom relay is shown in the lower part of the schematic. It is driven by the station-wide intercom L-Pad defeat line (see intercom writeup). When the relay is closed, it connects the 10 Volt AC bulb supply to the "L" connection for circuit #5, thereby lighting the intercom bulbs in the multi-line telephones (An LED on the front cover of this device is also illuminated).

III) Calibration Procedures:

The only adjustment possible to make for this device is to set the audio level for each of the three telephone lines. This is done by tweaking trimmers T1-T3 (located on the circuit card inside the box). To get the best gain balance, call someone up outside of the university (the lines are "lossier" when calling outside), and adjust the gain for the corresponding line until the caller is satisfied that the music-on-hold level is optimal.

IV) Trouble-Shooting Hints:

This device is so simple that trouble shooting should be straightforward. Remember not to connect the local ground to any "real" ground, as emphasized earlier. The KTUs for the 3 telephone lines require a strap to be set in the proper position for music-on-hold to work. I've set these straps on the current KTUs, but they may need to be set again if our KTUs are replaced.

If the intercom bulbs seem to remain illuminated (even though the intercom is unengaged), the problem may be a stuck relay in this circuit, however it is more probably due to difficulty in the Intercom KTU circuit. Refer to the writeup on that device for details.

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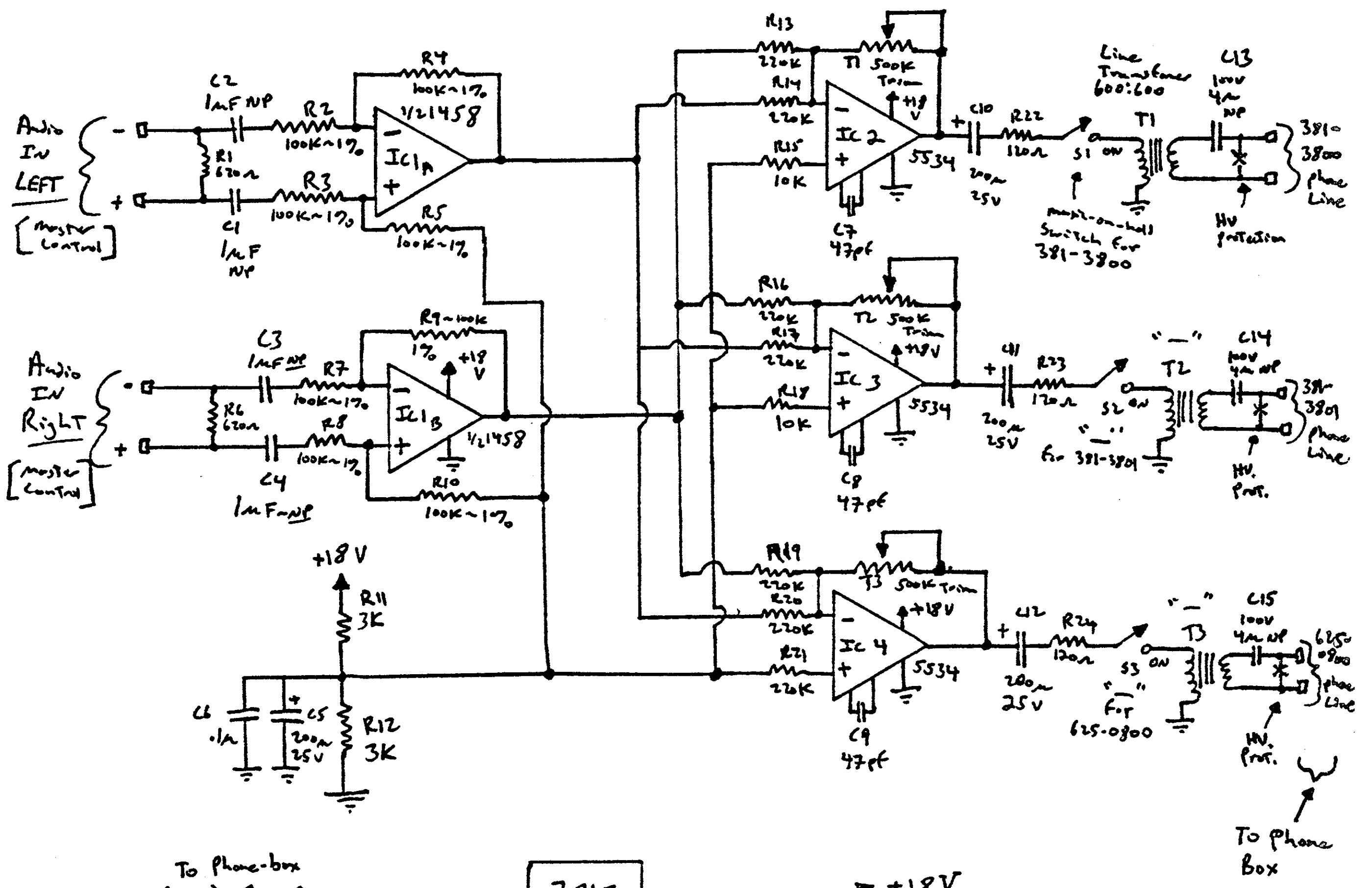
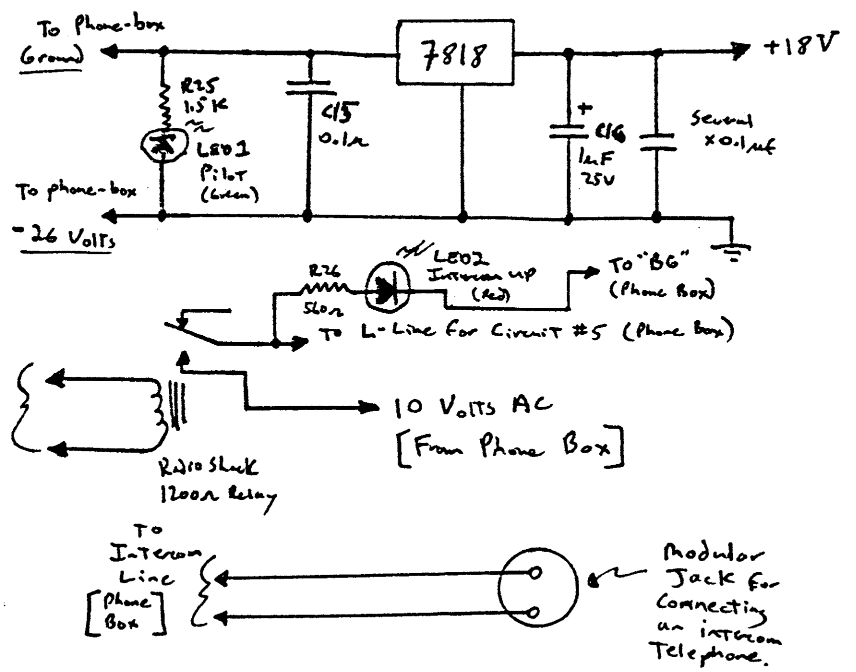
WMFO "MUSIC-ON-HOLD" DRIVER and Intercom Bulb Relay





Notes: All Circuitry on This page is located in a black Plastic box mounted on The phone room (ie. News office) wall.
Beware: The Telephone box uses a negative supply, hence our "ground" here is actually at a -26 Volt potential relative To The outside world (never connect up The shields of The audio input pairs To The box's ground!)

THE WMFO INTERCOM KTU CARD

INPUTS:

* Intercom Line

* Power Supply from Telecom Box:
  -26 Volts, 10 Volts AC, Ground

OUTPUTS:

* "L5" Bulb line for Intercom

THE INTERCOM KTU CARD

I) Function and Operation:

This circuitry is located on the KTU card sitting in slot #5 of the card cage in the Northern Telecom box located in the former News Office. It senses whenever any intercom telephones are off-hook, and illuminates the intercom bulbs accordingly in the multi-line telephones. This is its only purpose.

II) Circuit Description:

The basic unaltered KTU from Northern Telecom did not work, since it contained only a relay, which had too low an impedance for the intercom line and was not compatable with the lower intercom voltage (15 Volts quiescently, vs. the 50 Volts or so of a standard telephone line). As a result, the simple circuit portrayed in the schematic was hacked onto the Telecom KTU in order to drive this relay and activate the bulbs.

When all intercoms are hung-up, the 15 Volts present on the intercom line cause a current to flow through zener D1 and IR emitter LED1. The infra-red light produced by LED1 turns the infra-red phototransistor Q1 on, which drops the voltage at its collector, thereby turning Q2 off and leaving the KTU relay in its dormant position. When an intercom is picked up, however, the intercom line voltage drops, causing a reduction in IR output from LED1 (perhaps turning it off entirely if the intercom voltage becomes lower than the zener breakover of D1). This light reduction decreases the current flowing through Q1, correspondingly raising its collector voltage and turning Q2 on, thereby activating the KTU relay and illuminating the intercom bulbs. Zeners D2 and D3 block any quiescent collector voltage on Q1 from reaching the base of Q2 when all intercoms are hung-up. Beware of the polarity reversal; what I call +26 Volts here is really Northern Telecom ground, while what I term "ground" is actually the Telecom -26 Volt supply.

III) Trouble-Shooting Hints:

If the intercom bulbs someday appear to be kept illuminated (even though it has been proven that the intercom line is not somehow engaged), this circuit may be at fault (check also the intercom bulb relay in the music-on-hold box). An easy way of determining if this card is at fault is to remove it from the card cage. If the intercom bulbs then go out, you've found the culprit.

If the IR output of LED1 decreases or the efficency of Q1 somehow declines, the KTU relay may be held on perpetually. Check or replace LED1 and/or Q1. Perhaps diddle with the values of D2 or D3. Beware the ground reversal, as mentioned above. LED1 and Q1 must be butted closely together and isolated from outside light (a straw wrapped with black tape is now used).

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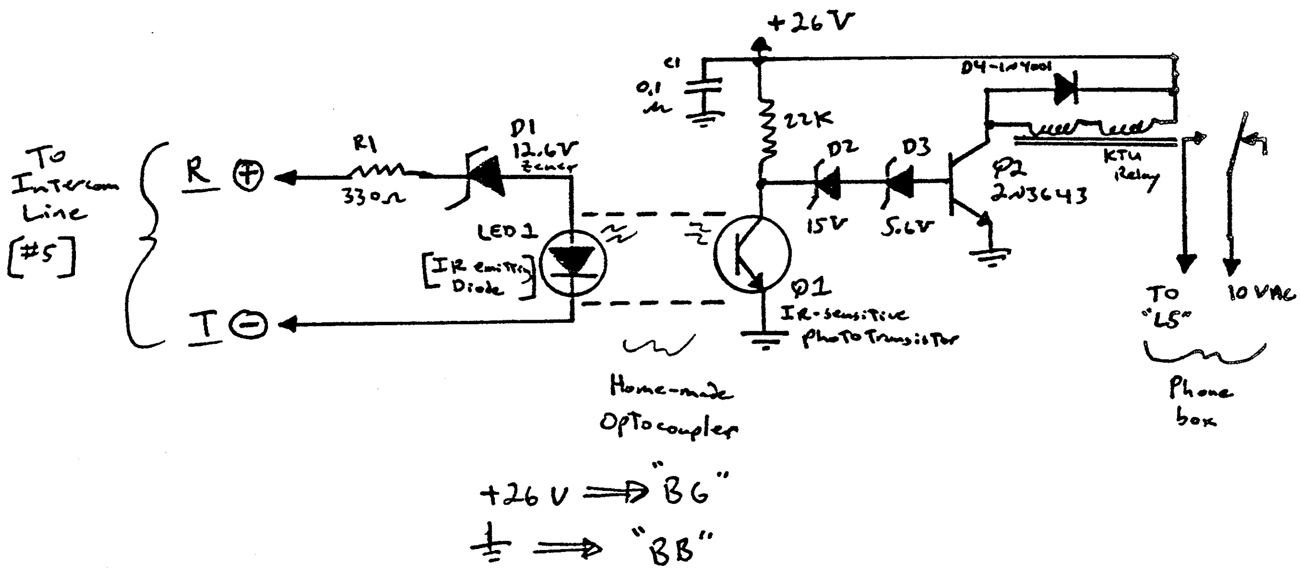
WMFO KTU To Drive Intercom Bulbs in Telephones



— This Circuit Resides on a "hacked-up" Circuit card sitting in Slot #5 of The Telephone KTU card crate in The main Telephone box.

— If The "Intercom Light" is stuck on (and it is determined That The intercom line is not engaged or shorted) This circuit may be responsible. Try pulling The KTU card, and see if The lights go out.

<u>THE WMFO EXTERNAL INTERCOM PHONE RINGER</u>

<u>INPUTS:</u>

* Intercom Line In

<u>OUTPUTS:</u>

* Intercom Line (with ring) Out

THE PHONE RINGER FOR EXTERNAL INTERCOM MODULAR PHONES

I) Functions and Operation:

This device is a small aluminum box (with a large transformer hanging below) mounted on the side of the grey "monitor" rack in the Master Control room. When one pushes the black button on the face of this box, all modular intercom phones connected at "external" intercom sites (ie. the Ballou Hall transmitter room, Curtis Lounge, and MacPhie Pub) will produce a "ring". This circuit is unable to ring standard old-style telephones with bells. It can, however, also illuminate a neon bulb. A switch on the side of this box selects the "ring rate". Most modular telephones will only ring when this switch is in the "LOW" position, which also yields an annoying clicking when the intercom is off the hook. The "HIGH" position will not ring most modular phones (it will work with some of them), but instead produces a piercing 700 Hz. squeal when an intercom is off the hook (providing a possibility of getting someone's attention in this case). The switch should normally be left in the "LOW" position. This device should not be activated for more than about 30 secs. at a time (otherwise the power transistor mounted on the bottom of the box begins to heat up); ie. don't lean excessively on the button!! Although there is some pickup on the PA lines, this device does not affect any intercom telephones inside of the station. It has no effect on the air signal, either.

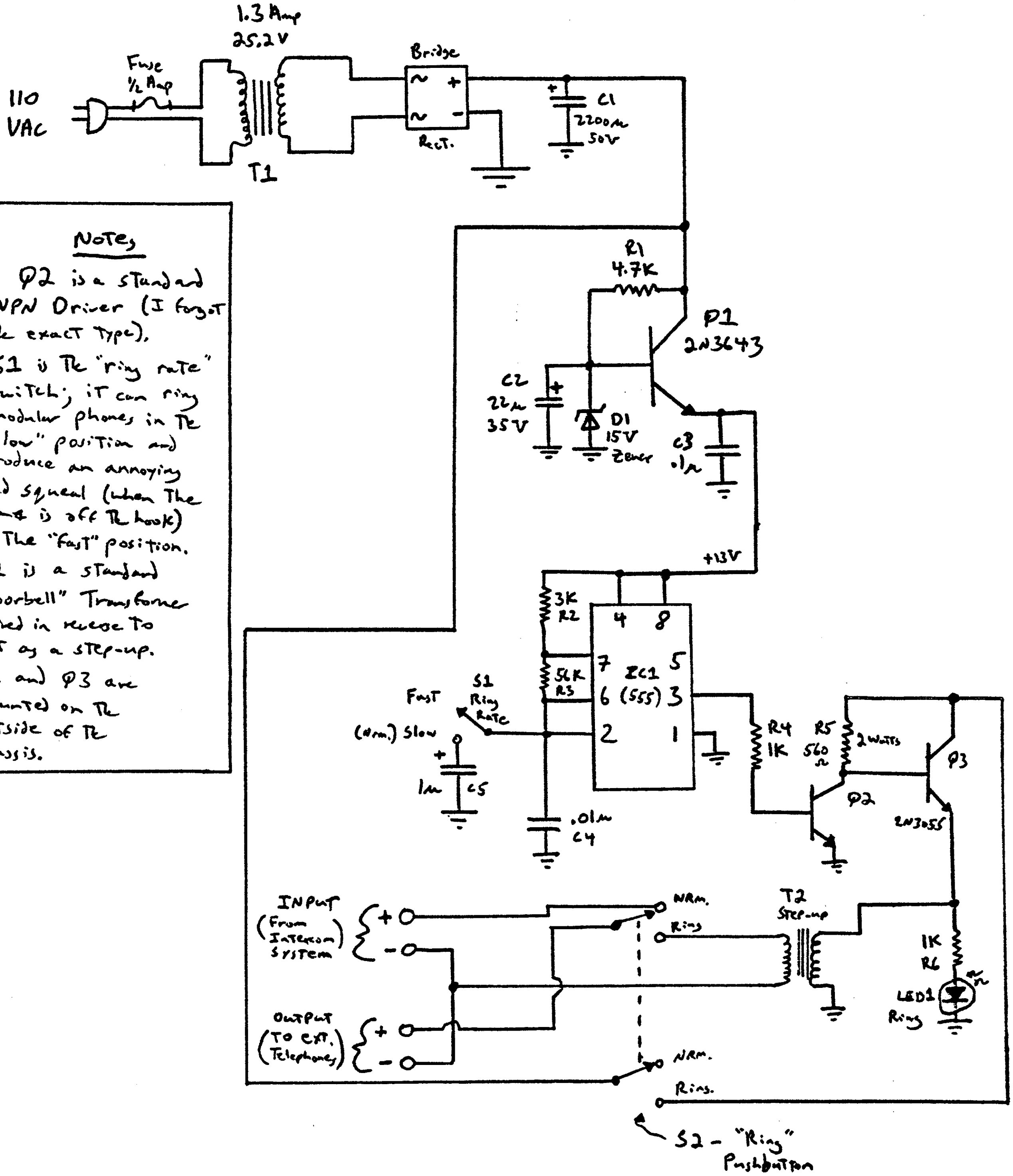
II) Circuit Description:

This circuit is quite simple, as seen in the one-page schematic. A 35 Volt (or so) DC supply is regulated to 15 Volts via Q1 and D1. This 15 Volt supply is used by a 555 astable clock (IC1). This clock runs at either approx. 7 Hz (when switch S1 is set slow) or 700 Hz (when S1 is set Fast). When the pushbutton S2 is pressed, the 15 Volt swing of the 555 is boosted to 35 Volts (ie. peaks @ the raw supply voltage) via Q2. This 35 Volt square wave at the collector of Q2 is buffered by emitter follower Q3 (which is a 2N3055 power transistor heat sunk to the chassis). Q3 drives the monitor LED1 and step-up transformer T2. The secondary of T2 is connected to the external intercom lines when S2 is pressed, thereby applying the high-voltage ring signal. When S2 is released, Q2 & Q3 are disconnected, and the input intercom line is routed directly to the external sites.

III) Trouble-Shooting Hints:

"Step-up" transformer T2 is actually a conventional 12 VAC "bell" transformer connected in reverse, hence its efficency isn't the best, and its lifetime may be limited. If problems crop up (and none ever have so far) T2 may need replacing (hopefully with something more efficent), or Q3 may be shot. Of course, first check the fuse and power supply transformer T1.

WMFO Intercom Phone Ringer



Notes

Q2 is a standard NPN Driver (I forgot the exact type).

S1 is the "ring rate" switch; it can ring modular phones in the "slow" position and produce an annoying loud squeal (when the phone is off the hook) in the "fast" position.

T2 is a standard "doorbell" transformer wired in reverse to act as a step-up.

T2 and Q3 are mounted on the outside of the chassis.

THE WMFO DISTRIBUTION AMPLIFIER

INPUTS:

* 600 Ohm Balanced Audio In
  (Left and Right Channels)

OUTPUTS:

* Eight Independent 600 Ohm
  Balanced Outputs per Channel

THE DISTRIBUTION AMPLIFIER

This circuit fans a single 600 Ohm balanced input into eight independent 600 Ohm balanced outputs, thereby allowing a single "professional" audio device (with output rated at a single 600 Ohm load) to drive up to eight devices with balanced 600 Ohm inputs. Each amplifier is built in pairs, enabling stereo signals to be handled. Each studio has a Distribution Amplifier wired between the mixing board output and all other device inputs (ie. all tape recorders and Master Control [ie. potential feed to other studios and eventually to the transmitter]). The in-studio distribution amplifiers are mounted on the Stereo Interface panels, and use their "raw" power supply voltages, thus turning the Interface power switch off will also remove power from the Distribution Amplifier and potentially prevent program output signals sent out of the mixing board from reaching the outside world (this is the reason behind the "ALWAYS LEAVE ON!" stickers taped to the front panels). Four Distribution Amplifier cards are located in Master Control (the lowermost unit in the blue inter-studio rack). Three of these "DA's" (to use the conventional epithet) fan signals from the studios (which come from their local DA's) to the other studios, the PA mixer, and the studio switcher. The fourth DA in this unit fans the external FM monitor signal out to each of the studios (incl. Studio D), the Music-on-Hold, etc. (the PA feed comes directly from the monitor unit). If this unit is ever powered down, signals generated in any given studio will not reach anywhere else, and most monitor feeds will be dead. All connection into and out of the DA's are routed through patchfields (in all studios and Master Control), thus they may be overridden via patching. In this fashion (for example), one may drive the distribution amplifier input with a source other than the studio's mixer board, and route it to whichever devices are desired (the tapedecks and Master Control feed are driven through the patchfield for the studio-based DA's). If a DA is down, one may patch directly from the DA input to the most important DA output to continue operations while the unit is being repaired (ie. for a studio, one might want to patch from the board's program output directly to Master Control). By using a passive 3-way split on the patchfield, one can feed two devices simultaneously from a single 600-ohm input, thereby allowing one to feed, for instance, both the Master Control line and a tape deck input.

There are no "controls" which allow operator interaction with the Distribution Amplifiers, except for the Interface power switch, which should be normally left on. The operation of the DA's are totally transparent to the casual WMFO disc jockey.

II) Circuit Description:

Since there are usually two of these devices in the airchain at any given moment, they are designed to have relatively little effect on the audio signal and are quite simple in nature, as seen in the single-page schematic depicting one channel of a stereo DA. The 5532 Dual OP-AMP provides two differential amplifiers which produce an inverted and noninverted audio signal (the + and - inputs are wired oppositely in each amplifier). The "gain" trim T1 adjusts the difference between + and - inputs, allowing the gain across the DA channel to be varied between zero and approx. 3. The common-mode balance is adjusted via trimmer T2. The OP-AMP outputs are buffered to very low impedance via power transistors Q1/Q2 & Q3/Q4 (these are conventional TO-220 TIP types; because of the relatively low loading no heat sinks are necessary). Diodes D1-D4 provide biasing to allievate crossover distortion. Because the emitter voltages are fed-back to their respective OP-AMPs, the output impedance is consistantly low, allowing several independent 600-Ohm loads to be tied to the + and - outputs via 300-Ohm resistors (each channel has been seen to easily drive 8 such loads). Integrating capacitors C1,C2,C5,C6 prevent the excessive speed of the 5532 from passing high-frequency "spikes" and causing instability.

The "raw" power supply from the Stereo Interface front panel (approx. +/-25 Volts or so...) is used by this circuitry; it is fed directly to the buffers Q1-Q4, and is dropped by a pair of regulators to form a +/- 18 Volt supply for the 5532's. The collector lines for Q1-Q4 are supplied by jumpers run on the underside of the circuit card. Although provisions for 78/7918's exist on both channels (the foil patterns are identical for each channel), the regulators are installed only on one channel, and the +/- 18 Volts are bussed directly to the 5532 on the other.

Printed Circuit laytouts are shown for a full stereo Distribution Amplifier card. As mentioned, there are no common connections between the channels, and all power supply voltages must be jumpered. The last page of this writeup shows the printed circuit pattern with component locations hastily penciled in (this was my assembly worksheet for the prototype). Refer to a completed unit for the actual component mounting, or puzzle it out yourself from the schematic. The location of trimmers T1 and T2 can be seen from the letters "gain" and "bal." written backwards on these sheets.

III) Calibration and Adjustment:

When a DA is first built, its gain and input balance must be trimmed upon installation. First the input balance is adjusted. Set the gain to (or near) maximum if it has not been adjusted. Apply the "-" end of an incoming signal (ie. an oscillator @ 1Khz) to the DA ground (ie. chassis or shield of patch cable). Apply the "+" end to the incoming signal to both + and - inputs of the DA under calibration. With a DA already tied into a patchfield, this can be done by patching one end of a cord into the DA input, and attaching clipleads from an oscillator to the other end (needless to say, only one channel of a DA should be calibrated at a time). Connect the DA output to a balanced 600-Ohm audio monitor (ie. patch it into the board, and listen to it on program or audition). Adjust T1 until the oscillator tone dissapears (or is minimized). The DA channel is now balanced.

All DA's at WMFO have been set to yield unity gain. This enables one to patch around a DA (when necessary) without suffering a change in volume. To adjust the DA gain, first patch a 600-Ohm audio source (ie. use the House Tone Oscillator on the patchfield) into a fader on the board, and adjust this fader until the meter level reads at 0 VU. Then patch the House Tone into the input of the DA under calibration, and patch one of its outputs into the same fader on the board. Adjust the gain trimmer T2 until the meter level on the board again reads at 0 VU. Assuming that the board impedance is a typical 600 Ohms, the DA channel is now set to unity gain, and its calibration is finished.

IV) Troubleshooting Hints:

These DA's have run beautifully for about 1.5 years. The only initial problems have been due to bad solder connections (remember, these units were assembled by staff volunteers). The major places where bad solder connections have been found were in the output transistor mountings. During assembly, these transistors were switched more often than required (due to reversals, swaps, etc.), causing the etch here to become a bit torn up. I have never found a bad transistor, diode, etc.; it's always been a bad solder connection at the output transistors. Check this first; otherwise the symptoms of a bad output transistor or biasing diode are fairly conventional. The 5532's should live for a while, although they are socketed and easy to replace (they normally run a bit hot).

If there is a problem, two spare and tested DA cards are currently living on a shelf in the Engineering Closet.

WMFO Distribution Amplifier

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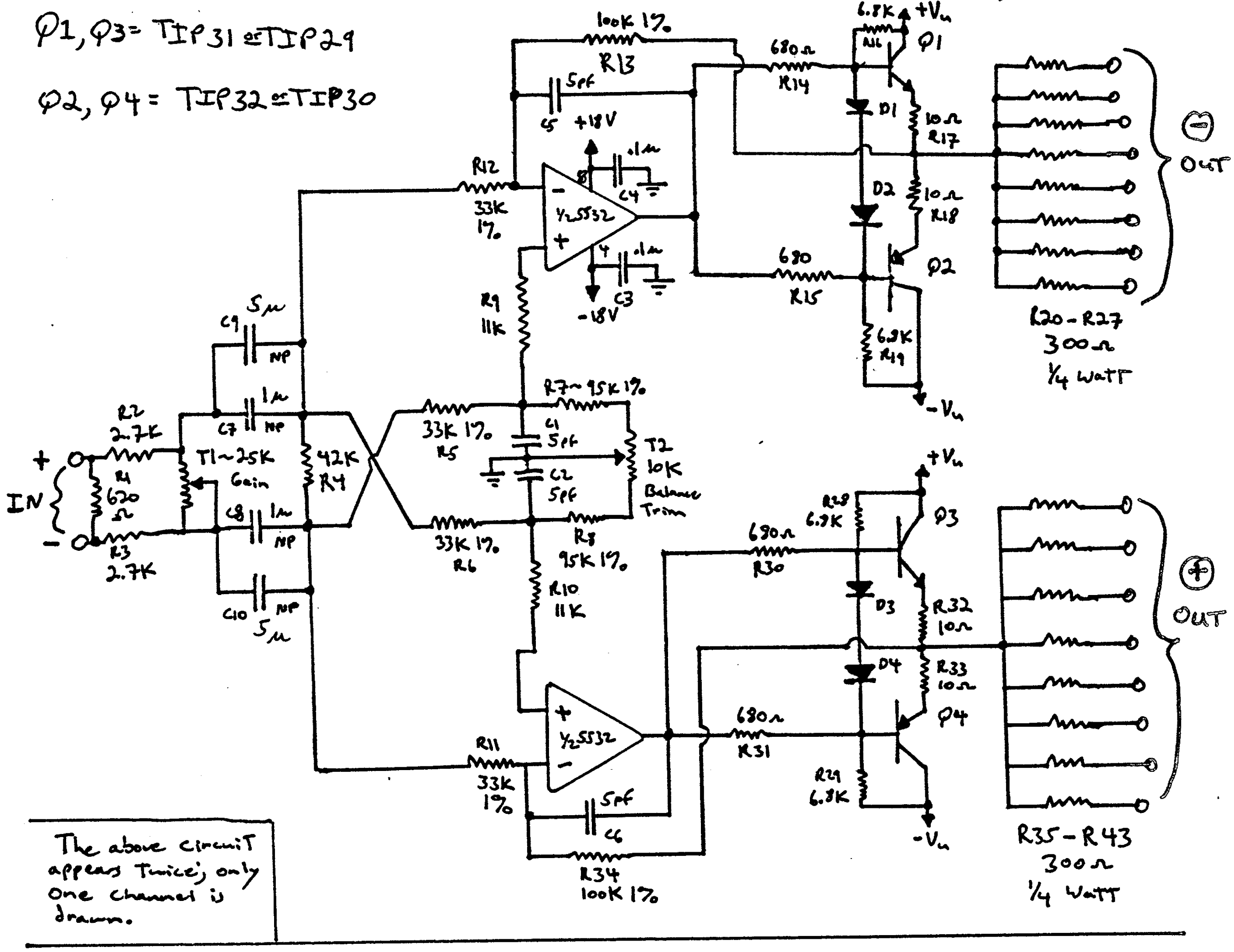
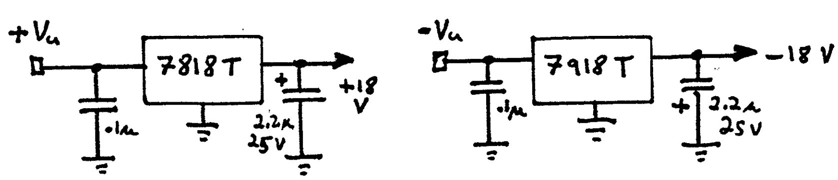
Q1, Q3 = TIP31 or TIP29

Q2, Q4 = TIP32 or TIP30



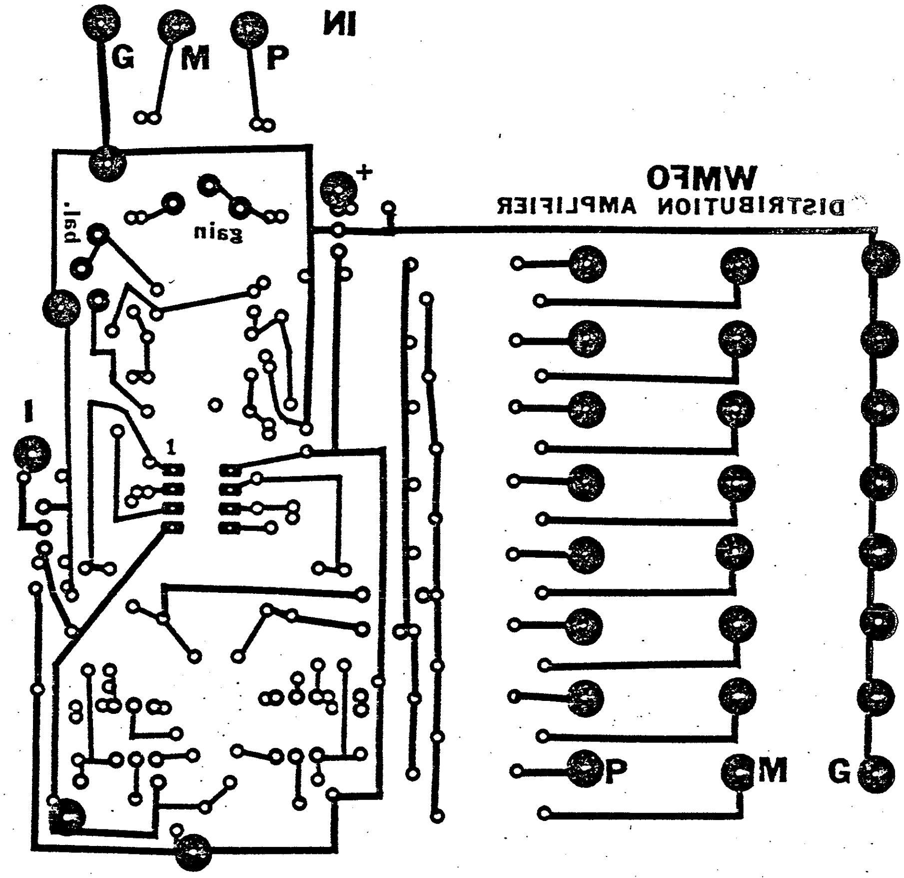
The above circuit appears Twice; only One channel is drawn.

Power Supply (Common to both DA channels)



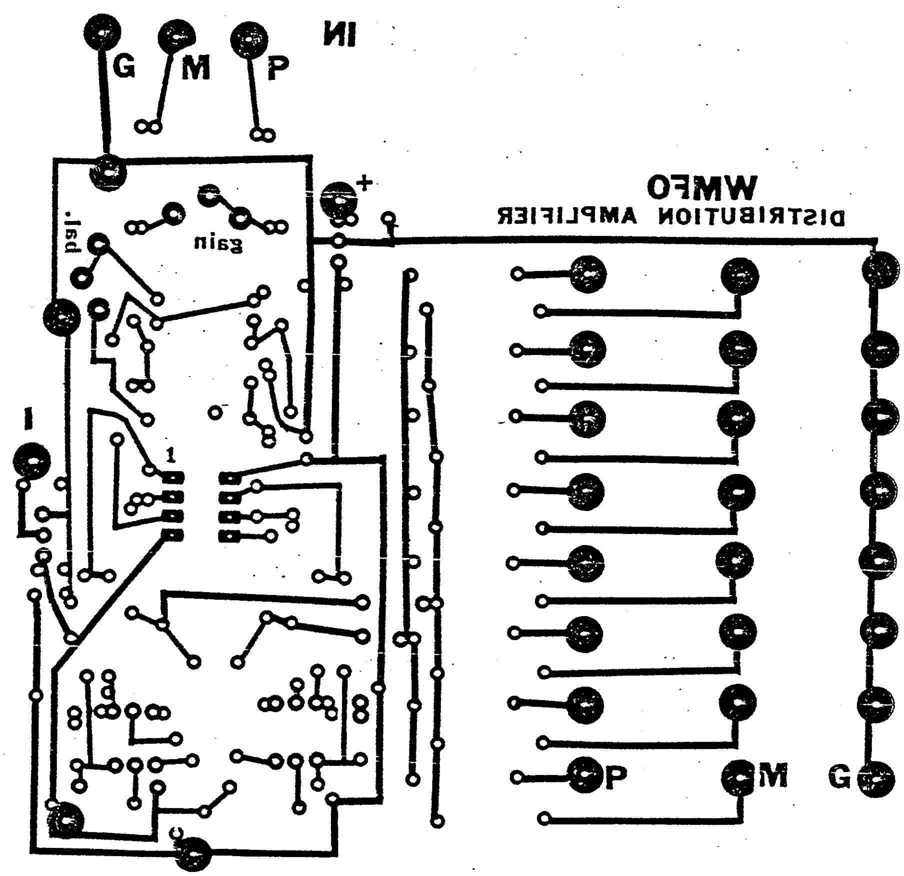
+/- Vu ≈ +/- 24 Volts unregulated

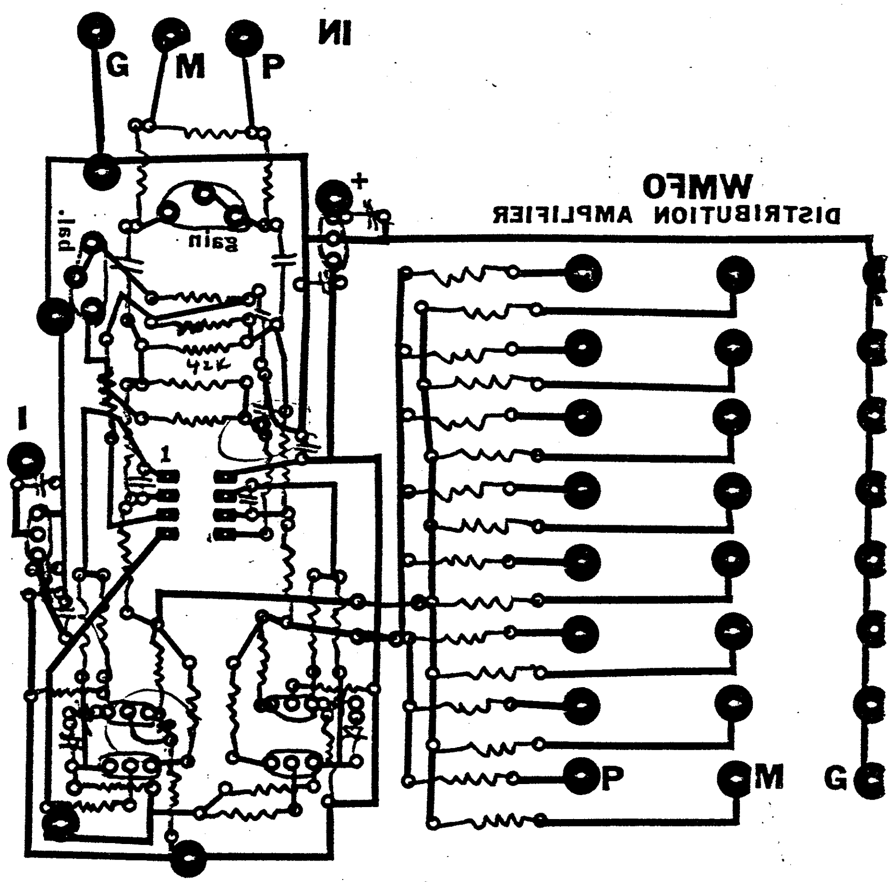
Notes: The Voltages +/- Vu are unregulated (≈25 VDC) supply voltages generally Tapped off The Filter capacitors [For The DA's in The studios, These are on The Interface Chassis] In some cases, 7815 + 7915 are used in place of the 18-Volt regulators.



Distribution Amplifier

Stereo P.C. Card Layout





Distribution Amplifier: Draft of Component Locations

[For detail, see a completed PC card]

THE WMFO STEREO INTERFACE

INPUTS:

* 600-Ohm Balanced Mix in (Left & Right)

* 7 Single-ended High-Impedance Stereo Mix Inputs (Pairs of various jacks on front panel).

* 600-Ohm Balanced Line in (Left & Right)

OUTPUTS:

* 600-Ohm Balanced Mix out (L & R)

* 5 Single-ended Stereo Line Outputs (Pairs of various jacks on front panel).

THE STEREO INTERFACE UNIT

I) Function and Operation:

A WMFO stereo interface unit is mounted in a rack in each studio. Its primary purpose is to enable stereo signals to be mixed, processed, and buffered to-and-from the 600-ohm balanced audio standard used throughout the studios. Most "home" tape decks and other audio components possess relatively high impedance unbalanced line outputs (ie. 1 KOhm or greater). One could somehow "kludge" these devices into the WMFO system by connecting the "-" input to ground, and the "+" input to hot (ie. going directly from the "jacks panels"). This has at least two difficulties; primarily the "unbalancing" of the audio line to a common ground (which can create hum and noise), and the excessive loading of the device output (which can create low audio levels, high-frequency drops, and distortion). The Stereo Interface solves this problem by providing a high-impedance input for "home audio components" and a balanced output to the 600-Ohm world. A series of jacks on the left side of the front panel (see third page of schematic) allow one to plug a variety of devices into the interface mix input. Provisions are made for two pairs of standard 1/4" phone jacks, two pairs of RCA jacks, one pair of 1/8" phone jacks, one pair of terminal post/bannana jacks, and one pair of pin jacks. These inputs are rated at 68 K-Ohms each, and can be used simultaneously without crosstalk or loading problems. An additional input is provided so that balanced signals from the patchfield may be included in the mix and processed via the Interface's controls (this input may be thought of as an "eigth" front panel input, although it is fully balanced [however with a 15K-Ohms input impedance to minimize loading!] and appears on the patchfield).

The Left and Right Channel gains of this mix may be independently adjusted. A switch is provided for quick cut-off of either the left or right channel. A switch and gain control is also provided for a special Mono signal. The left/right balance of the mono input may be adjusted via a "source" pot. The left/right destination of the mono signal may be adjusted via a "pan" pot. The mono feed may operate simultaneously with the stereo feed so that stereo dynamics and separation can be controlled. "Peak" LEDs are provided for each channel which illuminate when the corresponding signal level is excessively high.

A "Right-Channel Invert" switch is included in the mix circuitry. This allows one to invert the phase of the right input channel. This can be used to correct the phase of one-channel-inverted tapes and source material. It also can be used with the mono feed to cancel vocals, etc. (just turn the stereo feeds off, mono feed on, invert the phase, and adjust the "source" control until the desired track drops out). When this feature is unused, this switch should be left in the "Normal" or un-inverted position, otherwise the next person to use the interface may inadvertantly have his right channel inverted.

The rightmost section of the interface panel is concerned with a balanced to single-ended line output feed. Five pairs of single-ended jacks are provided for easy mating to most home equipment (ie. 1/4" phone jacks, 1/8" phone jacks, RCA jacks, bannana/terminal posts, and pin jacks are provided). These outputs may be used simultaneously, and have a 1K-Ohm impedance. Independent controls are provided for adjusting the left and right output levels. These line outputs are fed from the patchfield, and are normalled to a Distribution Amplifier output (which is normalled to the board's program output).

The distribution amplifier for each studio is also mounted behind the interface front panel, and taps the interface power supply. This means that the power switch for the interface also controls the distribution amplifier, thus the interface should never be powered down (except in the case of a total studio shutdown), otherwise the studio becomes effectively disabled.

A small clock/timer unit is also mounted on the interface front panel. This unit can count seconds (up to 15 min.), and gives the time, date, etc. See attached sheet for instructions. This clock also runs off the interface power supply, thus is reset when it is turned off.

II) Circuit Description:

The first page of the schematic diagram shows the "mix" circuitry. 5532 OP-AMPs are used throughout the audio chain to retain quality. The balanced mix input from the patchfield is converted to a single-ended signal via differential amplifiers IC1a and IC2a (note the high 15K input impedance). This signal is mixed with the inputs from the front-panel jacks in IC1b and IC2b. The phone-jacks (with integral switch contacts) clamp their inputs to ground when nothing is plugged in. The output of the right mix may be selectively inverted by throwing a 5534 inverter into the circuit via S1. Both left and right signals are weighted by the "mono source" pot P1 before being summed in IC5. The driver sections for each channel are composed of 5532s IC3 and IC4 (which are capable of driving the 600-Ohm outputs). These selectively mix the feed of each channel (weighted by P4/S3 & P5/S4) with the mono output. The mono gain is set by the inter-stage pot P2, and it is selected by switch S2. The "mono pan" pot P3 adjusts the imbalance in the mono feeds to the left and right channel drivers. A 1458 OP-AMP (IC6) is wired as a comparator to detect large signal swings on either channel. Whenever the audio output ("+" end used here) exceeds a preset level (set by trimmers T1 & T2), the corresponding "Peak" LED is illuminated.

The line-output and power supply sections of the interface circuitry are depicted in the second page of the schematic. IC8a and IC9a are differential amplifiers that convert balanced audio into single-ended signals (note the high 15K input impedance). These signals are scaled by P6 & P7 before being buffered by voltage followers IC8b and IC9b. The individual single-ended outputs are isolated by 1.3K resistors before being brought to the front panel.

The before-regulator power supply circuitry is all mounted on the front panel (this is also used by the distribution amplifier, as noted). On-board 7818 & 7918 regulators yield a stable supply for all OP-AMPs (some interface units may use the 7815 & 7915 for +/-15 Volts). Diodes D1-D3 & buffer Q1 drop the +18 Volt supply down to 1.5 volts in order to power the clock/timer mounted on the front panel.

The printed circuit pattern for the interface is given at the end of this writeup. A hastily-sketched component placement guide is also included; refer to a completed unit for details.

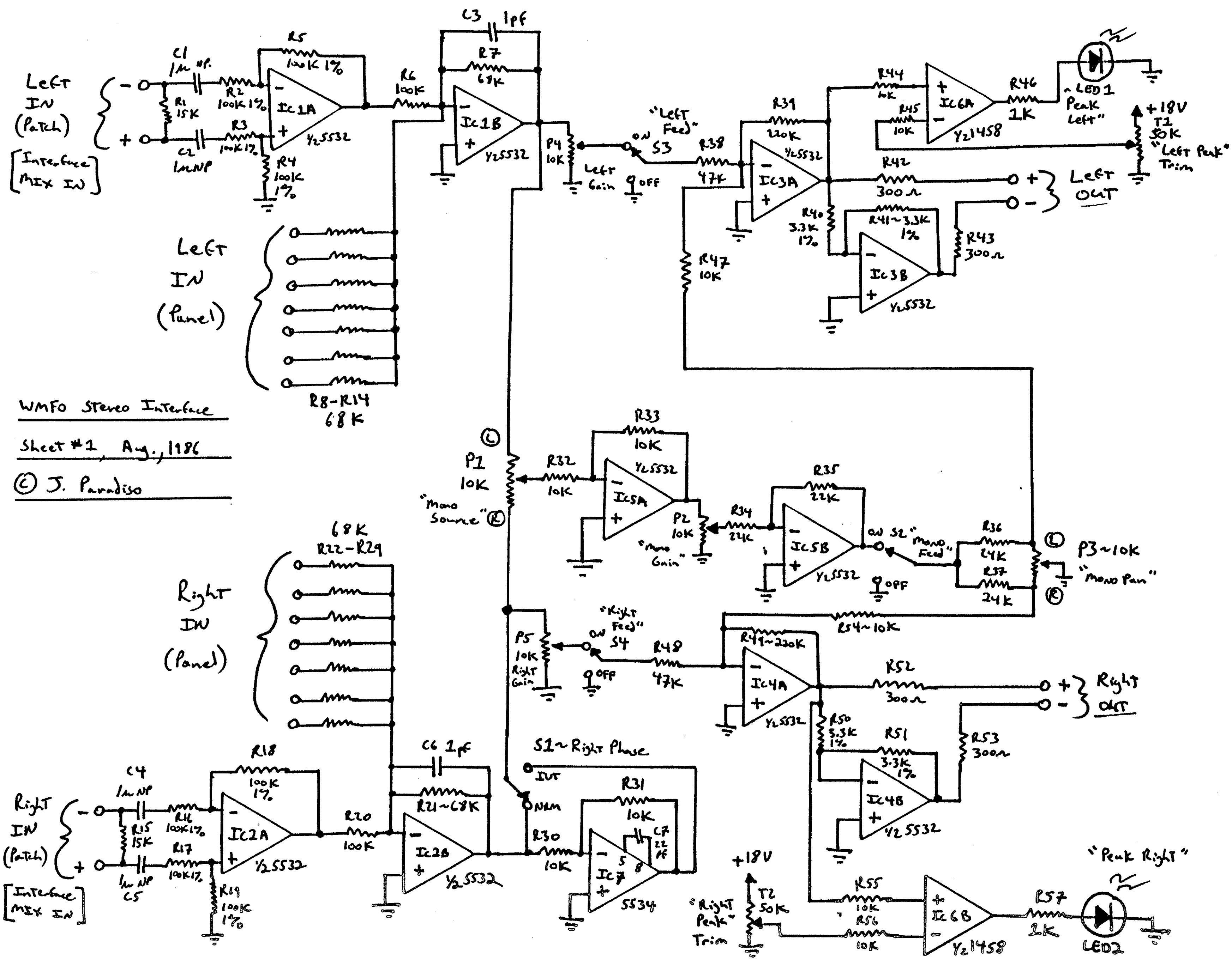
III) Calibration and Adjustment:

The only adjustments on the interface circuit cards are the thresholds for the "peak" LEDs. These have been set to agree with the corresponding LEDs in the LPB boards. Patch the house-tone oscillator into the interface mix input (via the patchfield). Turn the left & right feeds off, and mono feed on. Adjust the mono pan pot to center. Patch the interface mix output to an LPB board input, and adjust the corresponding fader to the 12:00 half-pot setting. Turn the "mono" volume pot on the interface up until the red LEDs on the LPB VU meters first illuminate. Adjust trimmers T1 & T2 on the interface card so the interface's "peak" LEDs just barely come on. The adjustment is done..... If one is calibrating the Studio B interface unit, patch a feed from one of the LPB boards in another studio into the interface input (the Studio B board is currently a Gates unit which has no LEDs).

If the clock on the front panel reads the wrong time, refer to the attached instruction sheet for re-setting directions.

IV) Troubleshooting Hints:

There have been no problems with these units in their 1.5 years of operation. Since most of their assembly has been performed by staff volunteers, watch out for bad solder joints (I found quite a few...) and defective wiring on the front panel.


Left IN (Patch)
[Interface MIX IN]
Left IN (Panel)
R8-R14 68K
Right IN (Panel)
68K R22-R29
Right IN (Patch)
[Interface MIX IN]
WMFO Stereo Interface
Sheet #1, Aug., 1986
© J. Paradiso
C1 1µ NP.
R1 15K
R2 100K 1%
C2 1µNP
R3 100K 1%
R4 100K 1%
R5 100K 1%
IC1A ½5532
R6 100K
C3 1pf
R7 68K
IC1B ½5532
P4 10K
Left Gain
"Left Feed" ON S3 OFF
R38 47K
R39 220K
IC3A ½5532
R40 3.3K 1%
R41~3.3K 1%
IC3B ½5532
R42 300Ω
R43 300Ω
Left OUT
R44 10K
R45 10K
IC6A ½1458
R46 1K
"LED1 Peak Left"
+18V
T1 50K
"Left Peak" Trim
R47 10K
P1 10K
"Mono Source"
R32 10K
R33 10K
IC5A ½5532
P2 10K
"Mono Gain"
R34 24K
R35 22K
IC5B ½5532
ON S2 "Mono Feed" OFF
R36 24K
R37 24K
P3~10K
"Mono Pan"
R54~10K
P5 10K Right Gain
"Right Feed" ON S4 OFF
R48 47K
R49~220K
IC4A ½5532
R52 300Ω
Right OUT
R50 3.3K 1%
R51 3.3K 1%
IC4B ½5532
R53 300Ω
C4 1µ NP
R15 15K
R16 100K 1%
R17 100K 1%
C5 1µ NP
R18 100K 1%
IC2A ½5532
R19 100K 1%
R20 100K
C6 1pf
R21~68K
IC2B ½5532
S1~Right Phase
INV
NRM
R30 10K
R31 10K
C7 22pf
IC7 5534
+18V
T2 50K
"Right Peak" Trim
R55 10K
R56 10K
IC6B ½1458
R57 2K
"Peak Right"
LED2

WMFO Stereo Interface

Sheet #2, Aug., 1986

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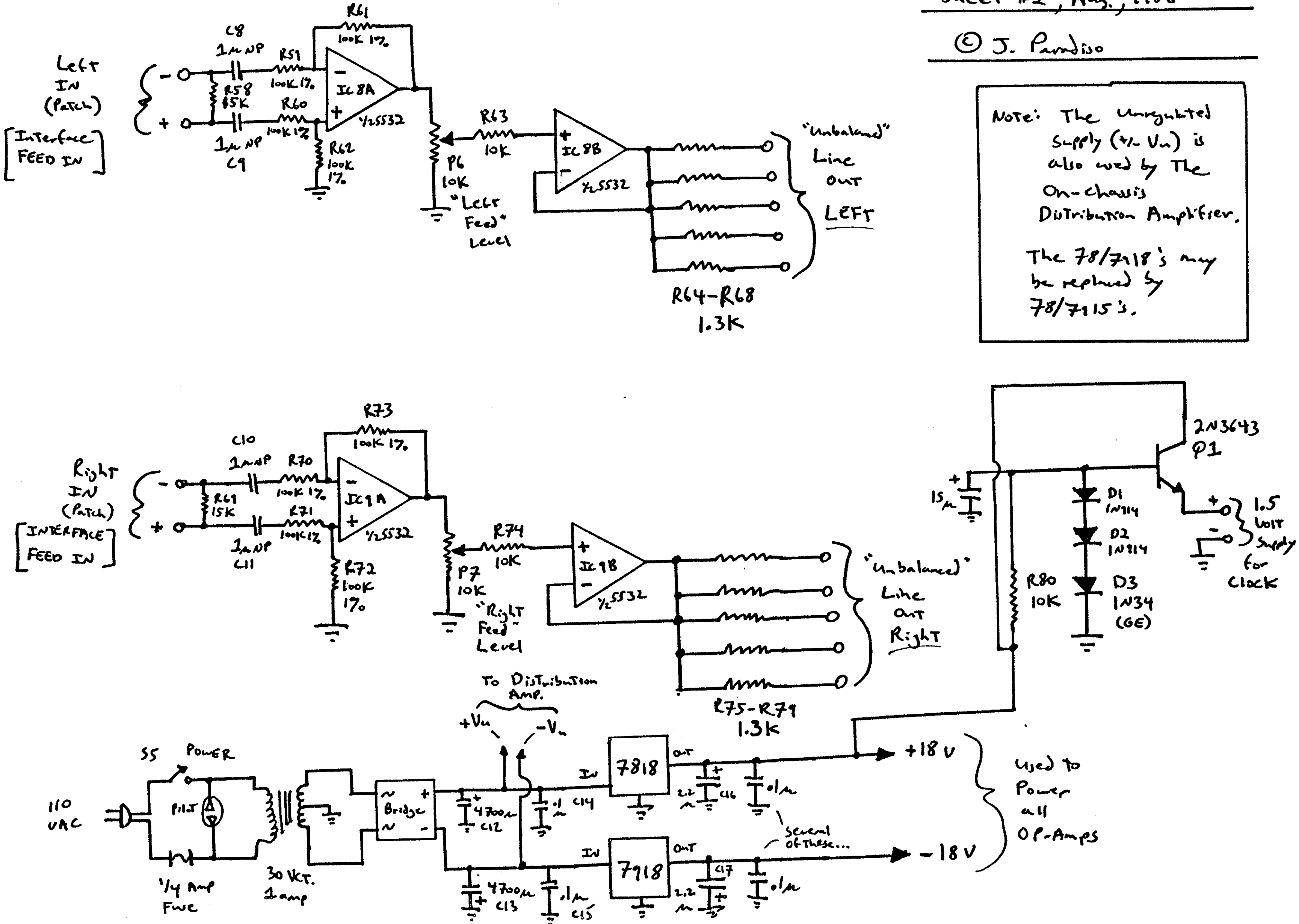
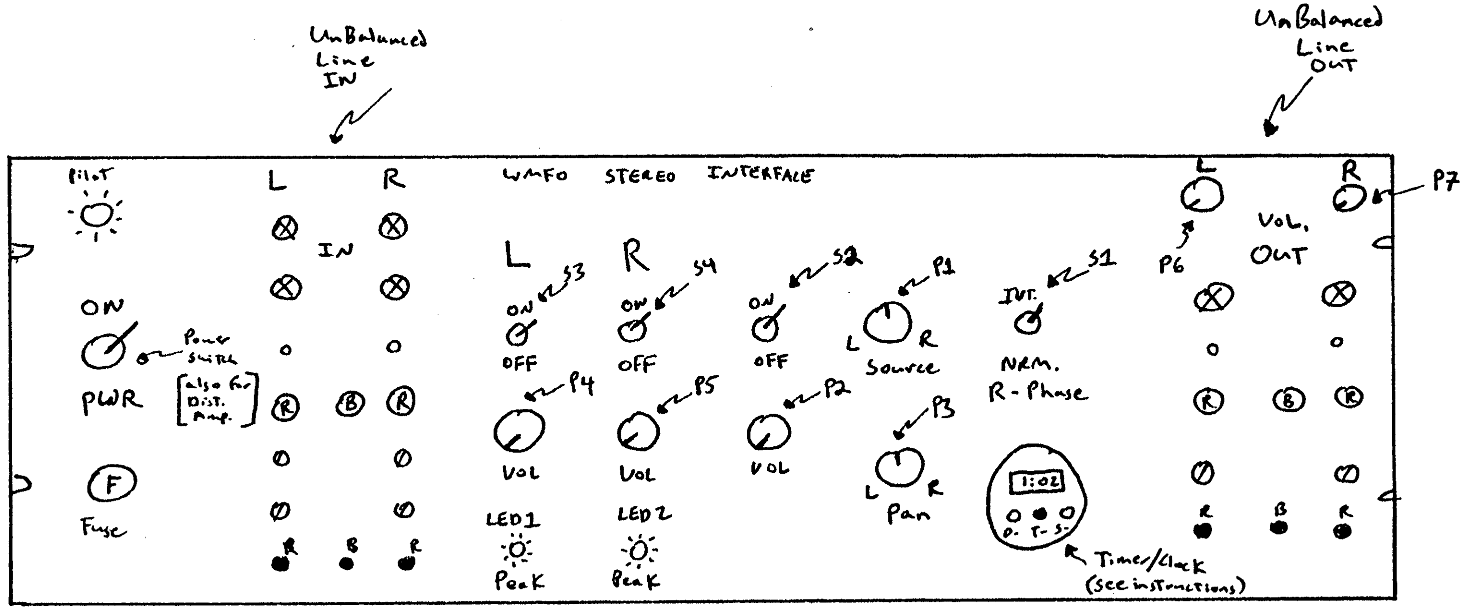
Note: The unregulated supply (+/- Vu) is also used by the On-chassis Distribution Amplifier.

The 78/7918's may be replaced by 78/7915's.



WMFO Stereo Interface — Sheet #3 — Jan, 1986

— J. Paradiso

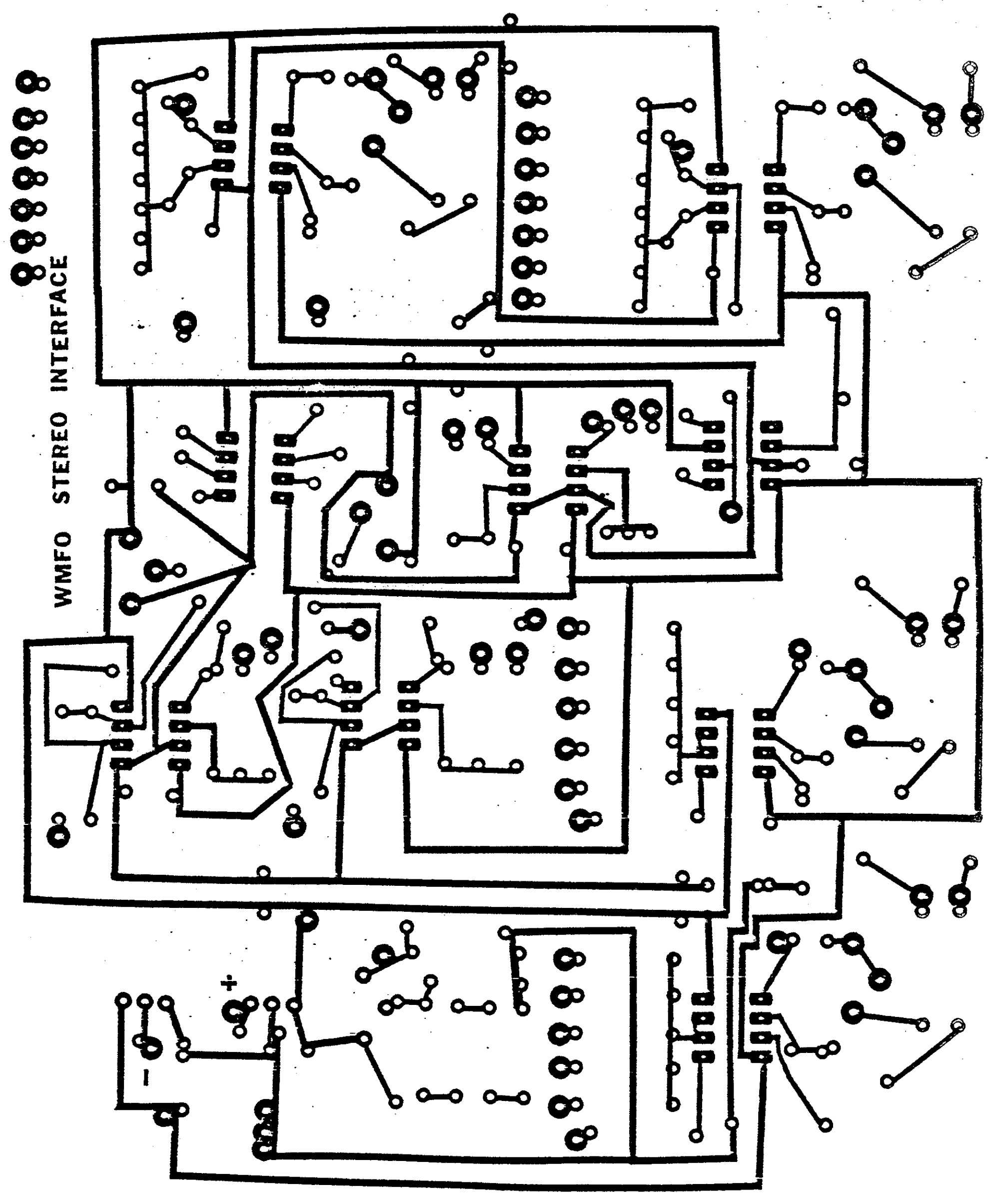


Front Panel Layout

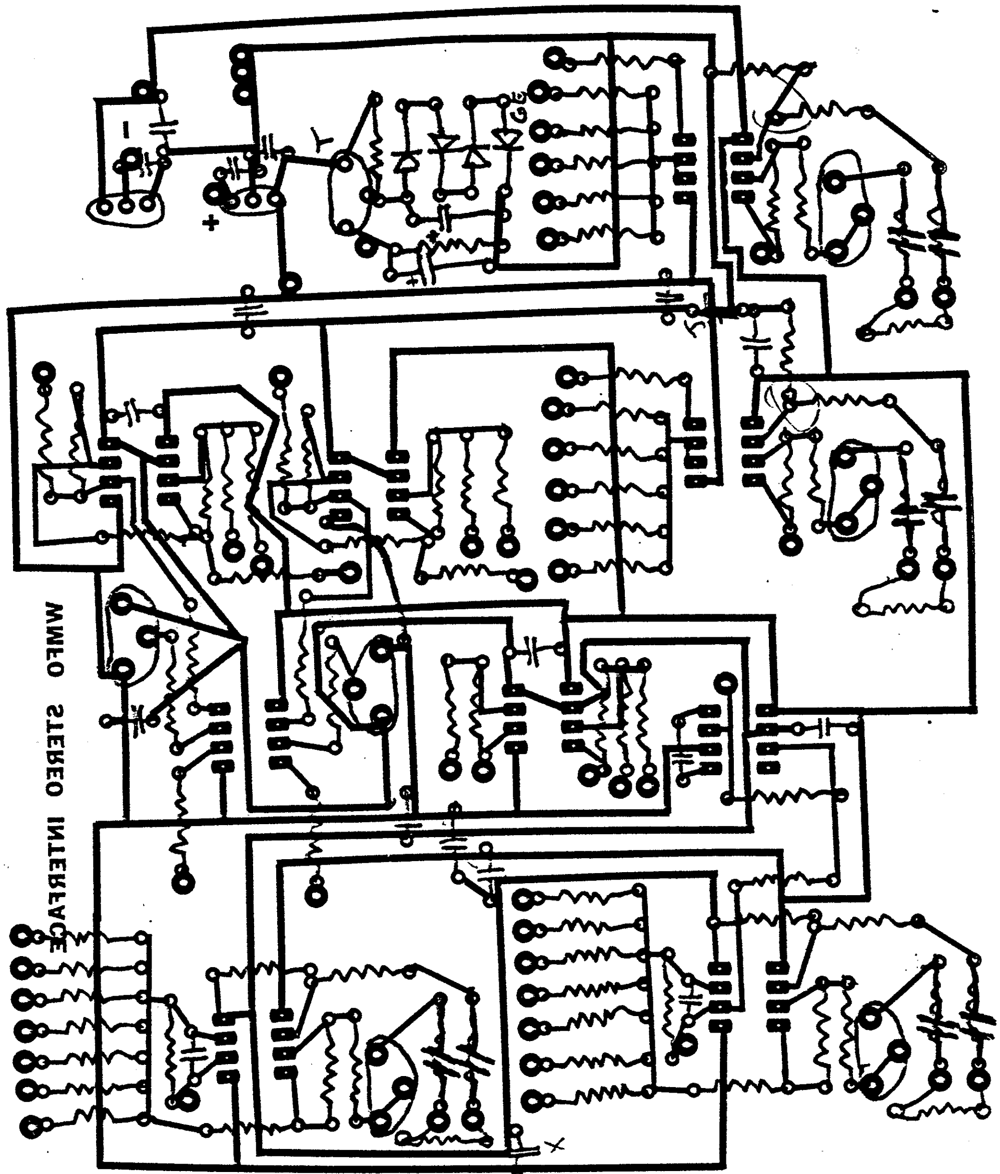
Jack Convention:

⊗ = 1/4" phone jack
o = 1/8" mini phone jack
Ⓡ, Ⓑ = Red, Black Terminal Post
⊘ = RCA Jack
•, • = Red, Black Pin Jack

Note: The Power Switch also cuts Voltage To The Studio's Distribution Amplifier; Turn This switch off, and The board's output signal will not propagate beyond The patchfield.



Layout for Interface PC. Card



Draft of Interface Component Locations

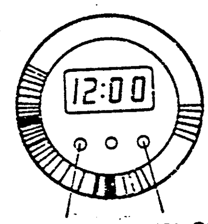
[For detail, see a
Completed Card.]

Instructions for Interface Timer/Clock

INSTRUCTIONS

TO READ TIME:

1. Normal display shows Hour:Minute.
2. Press D once to show Month & Date.
3. Press D again to show Seconds only.
4. To return to normal display, press D once more.



(D) DISPLAY (S) SET

TO SET TIME:

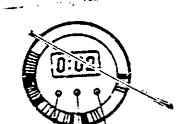
1. All setting is done by pressing S. The setting sequence is: Month, Date, Hour, Minute and Second.
2. To change any function, press S until function to be changed appears. Press D until correct figure appears. When all functions are correct, press S until Hour:Minute shows with colon flashing. If colon is not flashing, press D.

SET MODE FORMAT

Set Month	3
Set Date	1
Set Hour	12
Set Minute	20

TO USE TIMER:

1. Press T for timer mode. Display will show 0.00 (as shown at right).
2. Press T again to start timing.
3. To stop timing, press T once more. Display will show elapsed time.
4. Press D to return to normal display, or press T again to reset timer.
5. Timer counts to 15 minutes, then resets to zero and keeps counting in 15-minute increments.



(D) DISPLAY (S) SET

(T) TIMER

THE WMFO PITCH SHIFTER

INPUTS:

* 600-Ohm Balanced Audio In (L & R)

* Single-ended 10K-Ohm
  Microphone input (on panel)

* External Pitch modulation
  voltage input (on panel)

OUTPUTS:

* 600-Ohm Balanced Audio Out (Mono)

The Pitch Shifter

I) Function and Operation:

There is only one pitch shifter installed currently at WMFO. It is mounted in the effects rack in Studio B, and its input and output come up on the patchfield. The second page of the schematic shows a rendering of its front panel.

This device can shift the pitch of an input audio source down (by up to several octaves!!). It is also capable of generating vibrato and strange resonant feedback effects. It is based upon an analog delay (vs. newer high-end devices using all digital technology), hence its sound is a bit rough, noisy, and glitchy. For the price and simplicity involved, nobody's complaining. It sounds a bit rough on music, but is fine with voice. The pitch shifter is a mono device, and sums both left and right inputs into a single channel. Balanced inputs and outputs are provided on the back of the panel (which come up on the patchfield). A high-impedance, high-gain input is provided via a 1/4" phone jack on the front panel in order to allow a microphone to be plugged directly into the pitch shifter.

The third knob from left (labeled "pitch") controls the amount of pitch shift. Set at its counter-clockwise limit, the input audio source is passed essentially un-shifted and nearly intact. As this knob is rotated clockwise, the input is shifted progressively lower in frequency, until everything sounds like an ominous growl. The next knob to the left (labeled "Feedforward") controls the amount of original input audio signal that is mixed with the pitch-shifted output (ie. allowing one to speak in "chords"). The first knob at left (labeled "Feedback") adjusts the amount of pitch-shifted signal to be fed back to the pitch-shifter input. This can yield some very weird resonant effects, and is particulary effective when little or no pitch shift is selected. If too much feedback is used, the pitch shifter will screech in an absolutely horrible-sounding oscillation; back off on this pot, and it will cease. When the pitch shifter is first turned on, this pot should be turned full off (counter-clockwise) so that one may set the desired effects without encountering oscillation or other bizarre distortion.

The remaining controls on the pitch-shifter panel allow one to set an internal "vibrato" source. The vibrato option is enabled or disabled via the accordingly labeled switch (this switch should be OFF when one first turns on the pitch shifter, again to minimize confusion when initially trimming an effect). The vibrato rate is monitored by the panel-mounted red LED. The vibrato rate is set by the second knob from right (from well under 1 Hz up through 25 Hz or so). The intensity of vibrato is set via the rightmost knob. One must remember that this machine only shifts the pitch of the input signal DOWN. If the pitch knob is set to give very little shift and the vibrato intensity is turned up (with vibrato switch on), one will hear a very loud and annoying "pop" each time the vibrato waveform attempts to shift the pitch up. This pop can be removed by either turning the pitch knob to decrease the average pitch, or turning the vibrato intensity down (one generally adjusts both of these controls to achieve the desired effect).

A set of terminal posts/bannana jacks is provided to allow one to input an external pitch modulation voltage (ie. from an oscillator, envelope generator, etc.). A positive voltage at this input shifts pitch down (only a few volts will do; remember that this is not a "floating" input, and the black post is tied to chassis ground). This input is unaffected by the vibrato switch or any vibrato settings. If the disgusting "pop" is heard as discussed above, either drop the mean pitch further, or lower the amplitude of the external input signal.

II) Circuit Description:

The core of this circuit was derived from the Radio Shack VSC-1000 Variable Speech Control Cassette Recorder. It is based around a VSC-474 IC which controls the clocking and audio processing of a 512-stage bucket-brigade analog delay. Most of the circuitry around IC2, IC4, IC5, and IC6 is directly lifted from the VSC-1000. IC2 is an AGC which compresses the input audio somewhat in order to reduce noise from the analog delay. The other ICs are involved in the delay and audio processing. Rather than repeat the concepts involved, I've enclosed the actual documentation from the VSC-1000 service manual which beautifully elucidates this topic. It's really fascinating reading, and I urge all interested parties to give it a skim. I'll base most of this section instead around the customizing circuitry.

There are two circuit boards in this device, as seen in page 2 of the schematic. The large printed circuit board includes IC1-IC7, and a 7809 (+9 V) and 7912 (-12 V) regulator. The small hand-wired board includes IC8-IC10, and a 7812 (+12 V) regulator. The left and right balanced inputs are converted to single-ended signals via the 5532-based differential amplifiers (IC8). They are mixed with the feedback signal (scaled by P5) in IC1 (gain set by trimmer T1). A 10K input resistor from the summing node of IC1a also provides a high-gain microphone input. The resulting signal is applied to compressor IC2, and into the pitch shifter circuitry, as described in the VSC-1000 document. After the "raw" pitch-shifted output is buffered by Q3, it is doubled, filtered, and inverted by IC7 (and fed-back to IC1a via P5). The output of IC7 is boosted additionaly by IC9a. A 600-ohm differential drive is tapped from the 5532 outputs of IC9.

The 566 function generator (IC10) generates a triangle wave at pin 4, which is scaled by P2 and input (via S1) to the modulation summer IC3, which also adds the external modulation voltage via R47, and a voltage from the pitch control P3 via R53. The vibrato waveform is monitored by LED1, which is driven by the square-wave output from pin 3 of IC10 (buffered by Q2).

The compensation network based around Q1 was added to the VSC-1000 circuitry to cancel a small "thump" heard when the controlling ramp-wave was reset. It subtracts a filtered component of the clock waveform from the audio output of the bucket brigade delay before the audio waveform is smoothed and filtered by IC5. The weight of this correction is adjusted by trimmer T7. This circuitry was added later as a "quick fix", hence it does not appear on the circuit card, and is jumpered and "patched".

The printed circuit pattern is included with the schematic pages. A coarse component location diagram is also provided; refer to an existing unit for details.

The power supply is depicted on the second page of the schematic. All circuitry on the printed circuit card (IC1-IC7) is powered by the +9 Volts; the OP-AMPs on this card also use the -12 Volts. The circuitry on the hand-made card (IC8-IC10) use a +12 Volt supply in addition to the -12 Volts fed across from the printed circuit card.

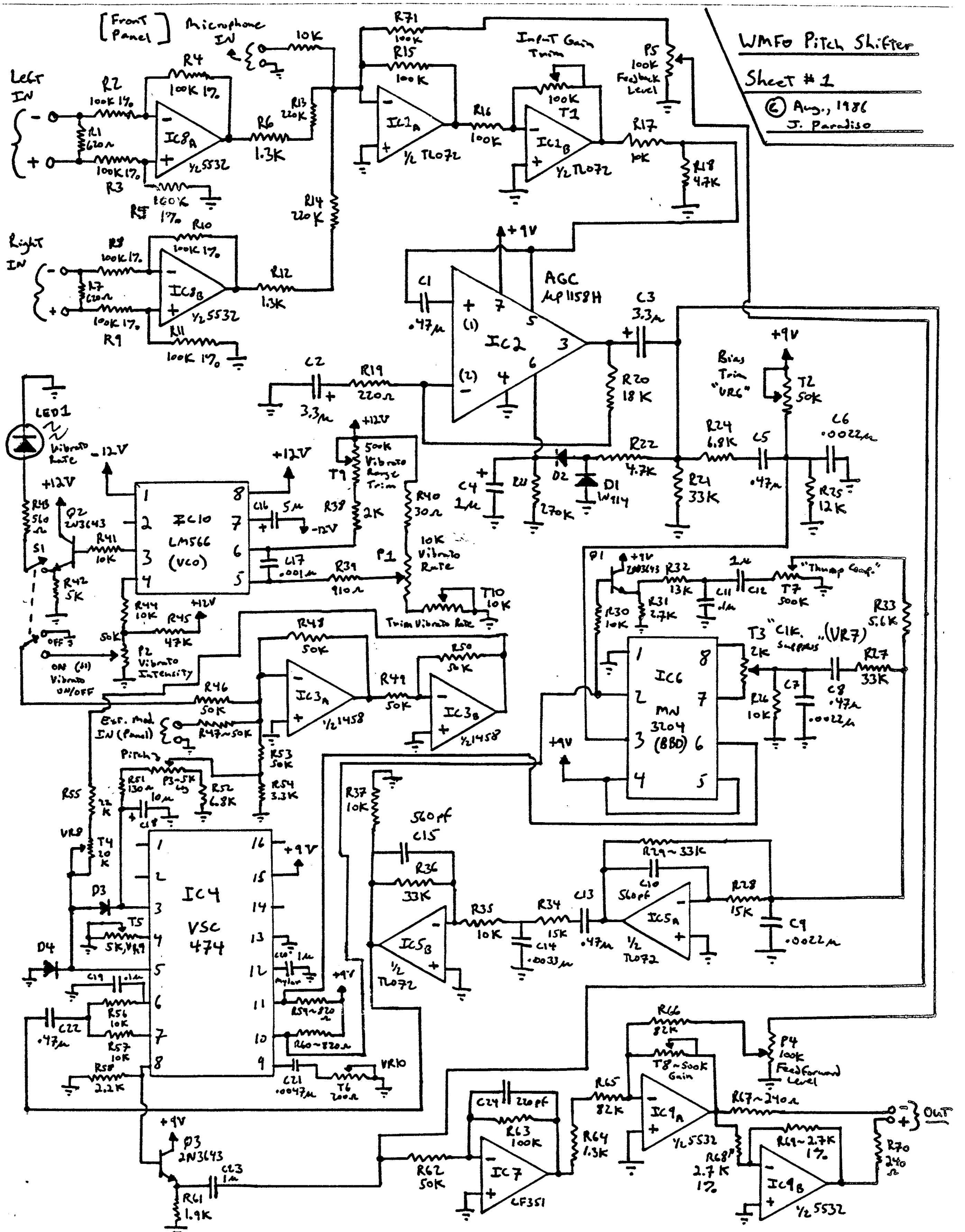
III) Calibration and Adjustment:

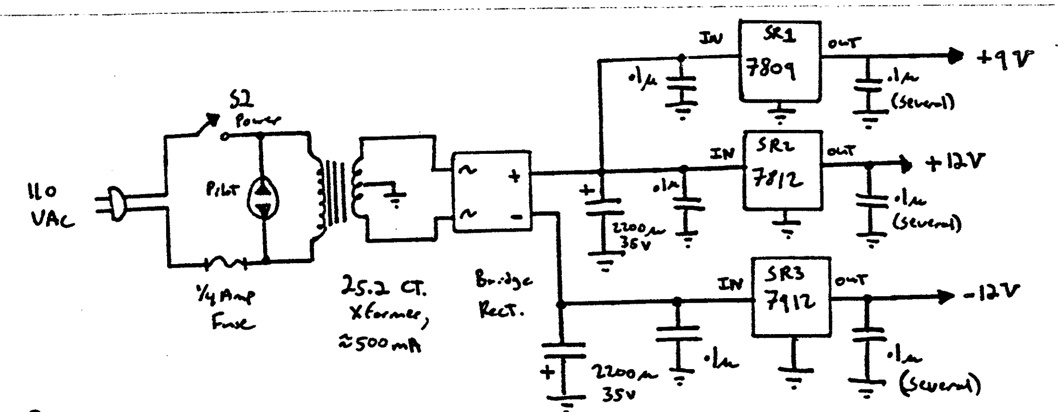
Some of the calibration of this device is straightforward, and some is a bit more arbitrary. Trimmer locations are shown in the second page of the schematic. Adjust the input gain control T1 so a nominal input appears well above any noise, and distortion is minimal (it may be best to adjust the analog delay trimmers first before bothering much with this). T8 is adjusted to yield an optimum output level (again, it may be best to adjust the analog delay first). Trimmer T9 adjusts the range of the vibrato oscillator, and T10 sets its base frequency. These can both be tuned as desired, using LED1 as a monitor. The calibration of the VSC-1000 circuitry is detailed in the enclosed VSC-1000 documentation. I've labeled the trimmer pots in the schematic also with their "VR" numbers, so one can refer directly to the VSC-1000 data. Trimmers VR6-VR10 must be set more-or-less as described (it's been so long since I built this device that I've forgotten most helpful hints). When setting the "VR" adjustments, turn the "thump compensation" T7 full off. After the "VR" pots are essentially set, adjust T7 to yield minimum clock noise and thump across the entire pitch-shifting range.

IV) Troubleshooting Hints:

This unit has not broken over its two-year current lifespan. In some respects, this is a miracle, considering how people abuse and "bang" on it in Studio B. If it ever does break, I've got no special hints. Since I built the unit myself, no volunteers were involved, and most of the solder joints (the bane of the community projects) should be OK. Good luck....

One final note: IC2, IC4, and IC6 may be somewhat difficult to locate commercially if they ever need to be replaced. The easiest means of buying these parts may be to order them from Radio Shack as replacement components. I've listed the Radio Shack Part # of these ICs (along with their conventional designation) on a handwritten page appended to this writeup. This Radio Shack Part # is needed in order to obtain any cooperation from the conventional Radio Shack staff.


WMFo Pitch Shifter
Sheet # 1
© Aug., 1986
J. Paradiso
[Front Panel]
Microphone IN
Left IN
Right IN
Input Gain Trim
P5 100K Feedback Level
AGC
μP1158H
IC2
Bias Trim "VRG"
T2 50K
LED1
Vibrato Rate
IC10
LM566
(VCO)
500K Vibrato Range Trim
P1 10K Vibrato Rate
Trim Vibrato Rate
"Thump Comp."
T7 500K
T3 "Clk. Sweep" (VR7)
IC6
MN 3204
(BBD)
P2 Vibrato Intensity
Vibrato ON/OFF
Ext. Mod. IN (Panel)
Pitch
IC4
VSC
474
IC5A
IC5B
1/2 TL072
IC7
LF351
T8~500K Gain
P4 100K Feedforward Level
IC9A
IC9B
1/2 5532
OUT



Power Supply

Notes: IC2, IC4, and IC6 are driven exclusively via the +9 V supply.

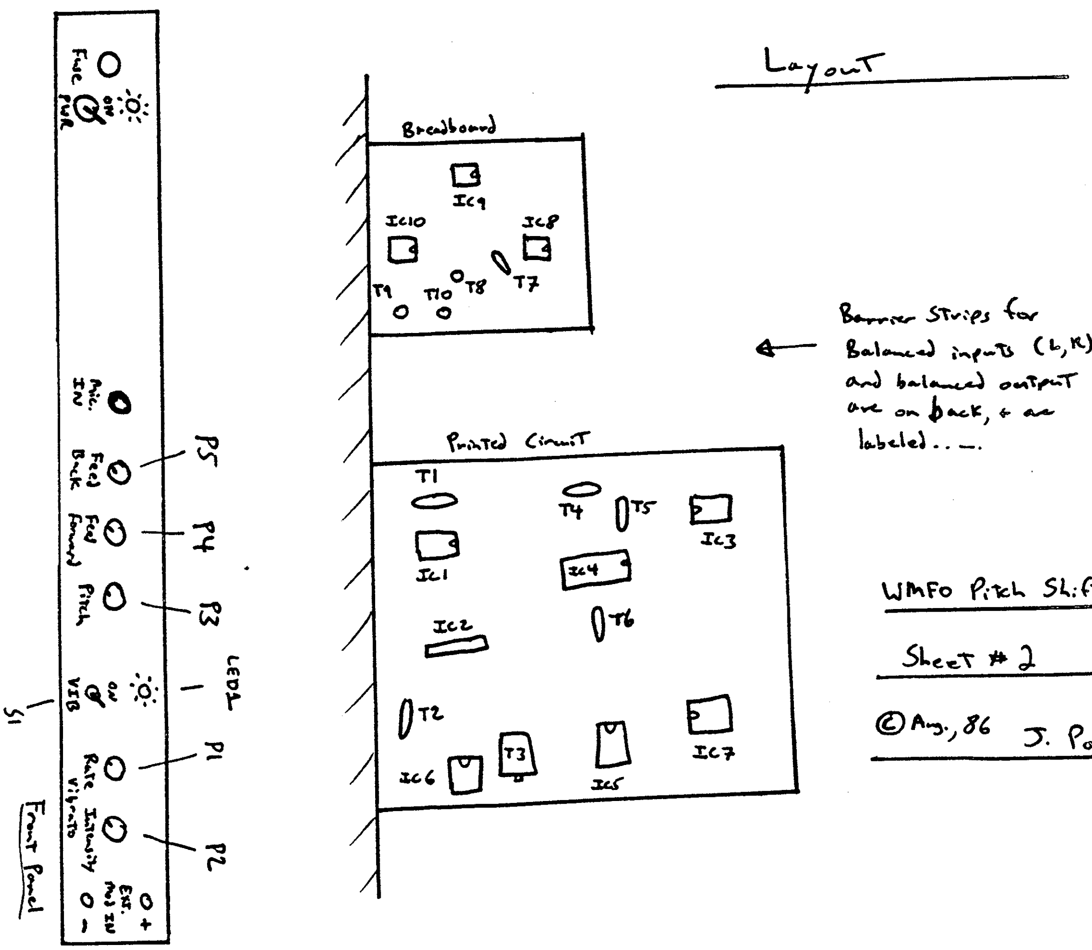
IC1, IC3, IC5, and IC7 use the +9 V and - 12 V supply.

IC8, IC9, and IC10 use the +12 V and - 12 V Supplies.

SR1, SR2, + IC1 → IC7 are located on the printed Circuit Card.

SR3 + IC8 → IC10 are located on the breadboard.

Layout



WMFO Pitch Shifter

Sheet # 2

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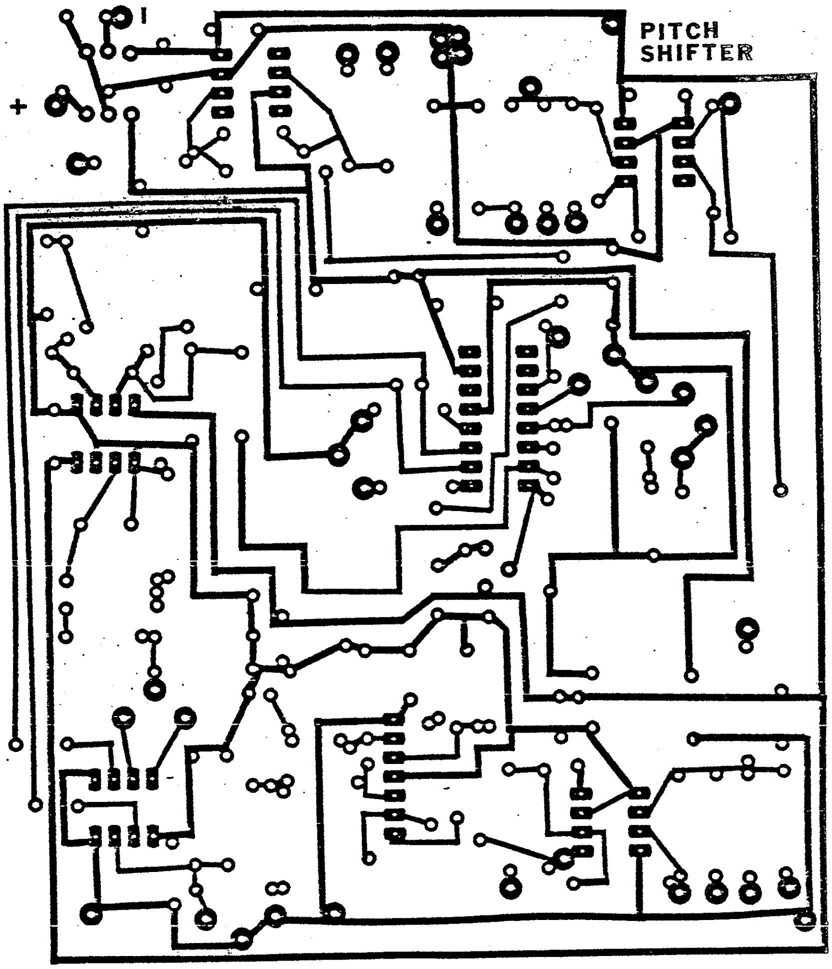
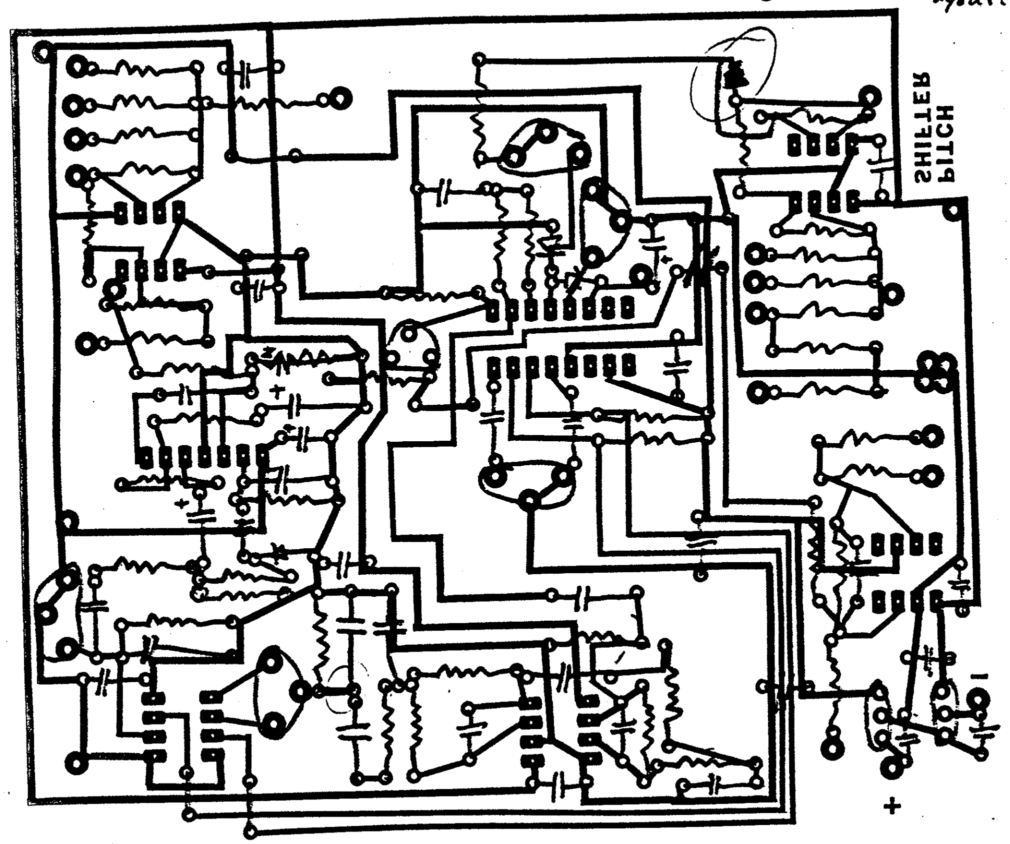


Layout for Pitch Shifter P.C. card

Warning: Glitch on board layout.

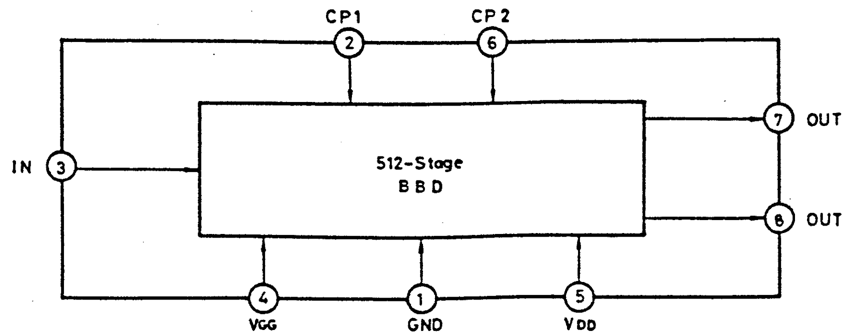


Component locations on Pitch Shifter P.C. Card

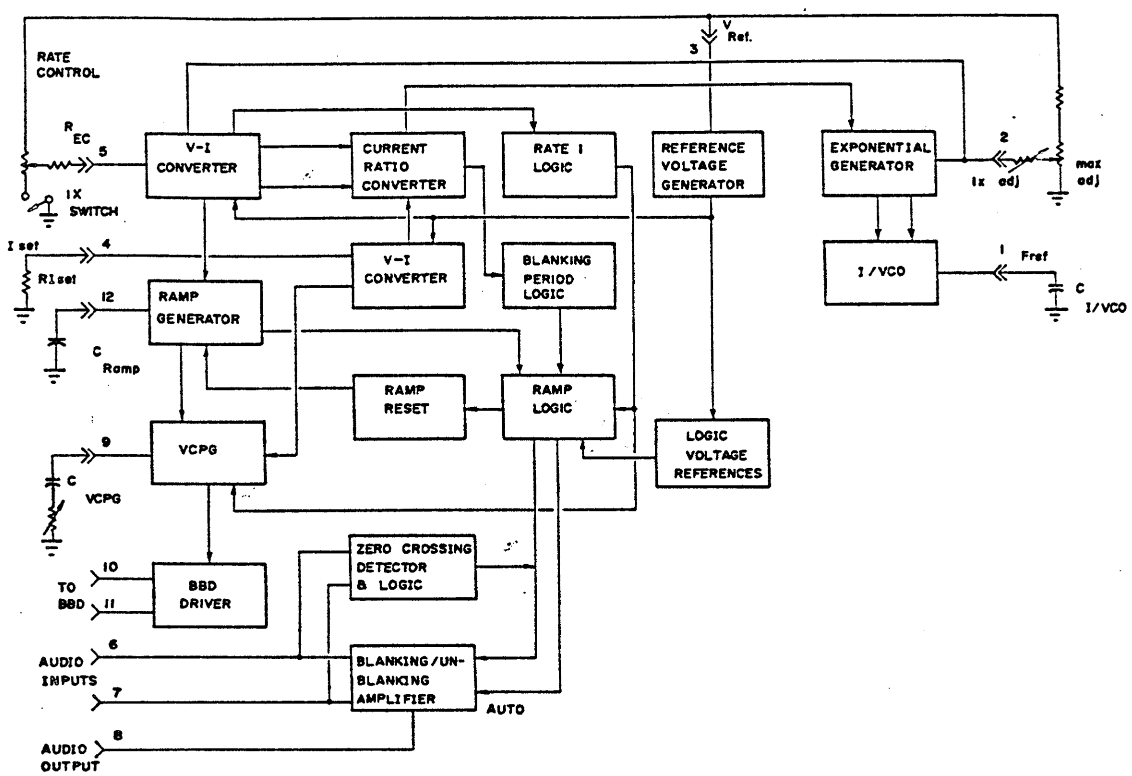
[For Detail, inspect a Completed Pitch Shifter Card]

IC INTERNAL CIRCUIT

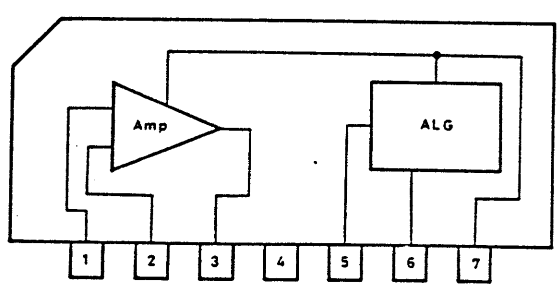
MN3204



VSC474



μPC1158H



MEMO

	Part #	Cross-Reference → Radio Shack
IC3	μPC1158H	MX-3542
		911-021-3-00-1
	MN3204	MX-4999
		911-021-5-00-3
	VSC474	MX-5000
		911-021-6-00-4

For catalog # 14-1002

VSC-1000

Cassette Recorder

REALISTIC®

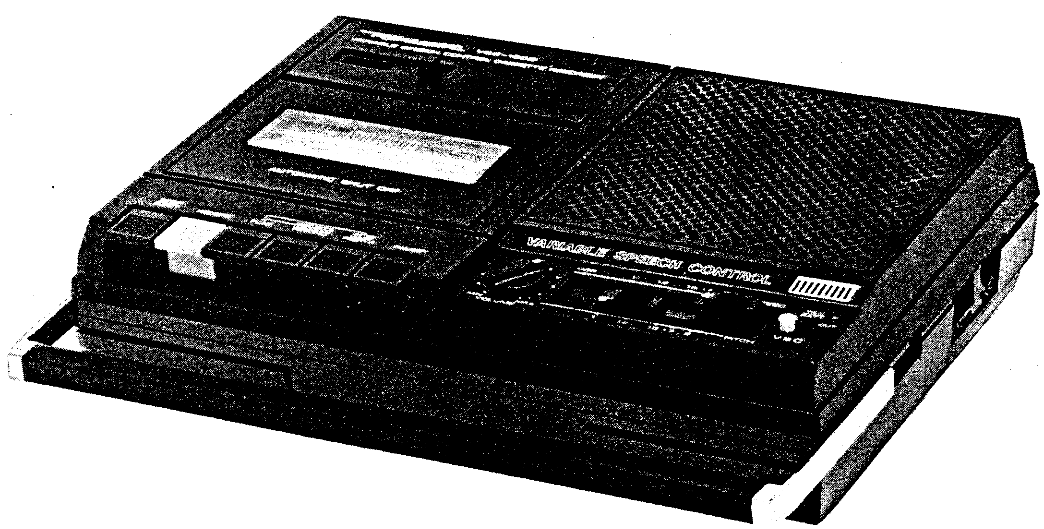
Service Manual

14-1002

VSC-1000

VARIABLE SPEECH CONTROL™ CASSETTE RECORDER

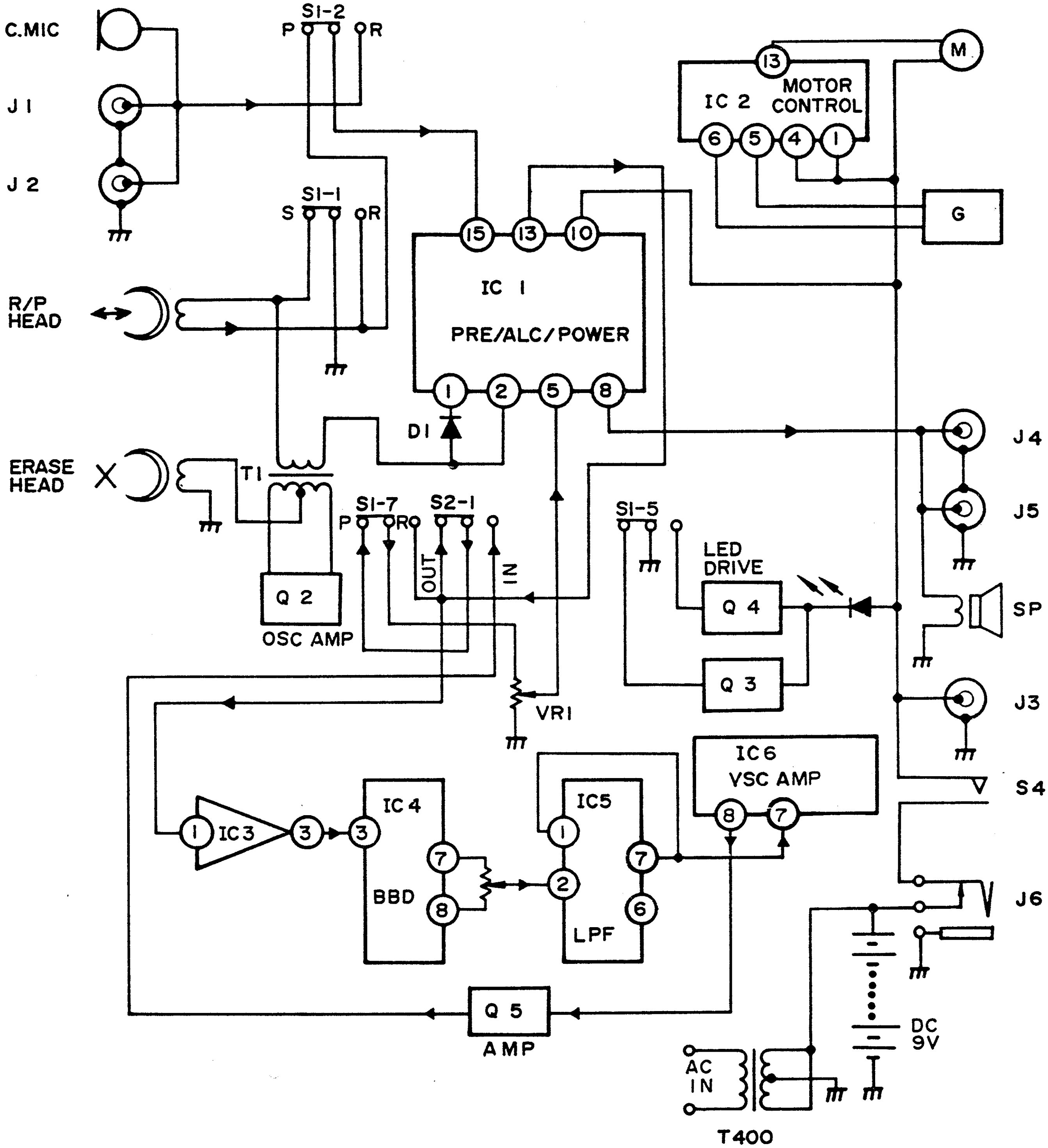
Catalog Number: 14-1002



CUSTOM MANUFACTURED FOR RADIO SHACK, A DIVISION OF TANDY CORPORATION

BLOCK DIAGRAM

of VSC-1000
Electronics



BBD DRIVE ADJUSTMENT (Figure 3 and Figure 4)

1. Connect a DC Regulator to TP1 and TP2 (ground).
2. Connect a Frequency Counter to the TP3 and TP2.
3. Adjust the Pitch Control VR (VR3) to 2.0 position.
4. Adjust DC Regulator output to 2.462V, and adjust VR9 on the PCB to Frequency Counter reading 12.00kHz.
5. Adjust DC Regulator output to 0.270V, and adjust VR10 on the PCB to Frequency Counter reading 300kHz.

RAMP PULSE ADJUSTMENT (Figure 3 and Figure 4)

1. Connect a Frequency Counter to TP1 and TP2.
2. Adjust the Pitch Control VR (VR3) to 2.0 position.
3. Adjust VR8 on the PCB so that one cycle wave is 24.9ms.

BBD CIRCUIT ADJUSTMENT (Figure 3 and Figure 4)

1. Connect a VTVM to TP4 and TP2.
2. Adjust Pitch Control VR (VR3) to 1.0 position.
3. Connect a SG. to TP5 and TP2, and adjust the SG. output signal to 36mV and 1kHz.
4. Adjust VR6 on the PCB to minimum distortion.
5. Adjust VR7 on the PCB to minimum noise.

Note: VR6 in very close to mechanism chassis.
Pay attention not to ground the circuit when adjusting VR6.

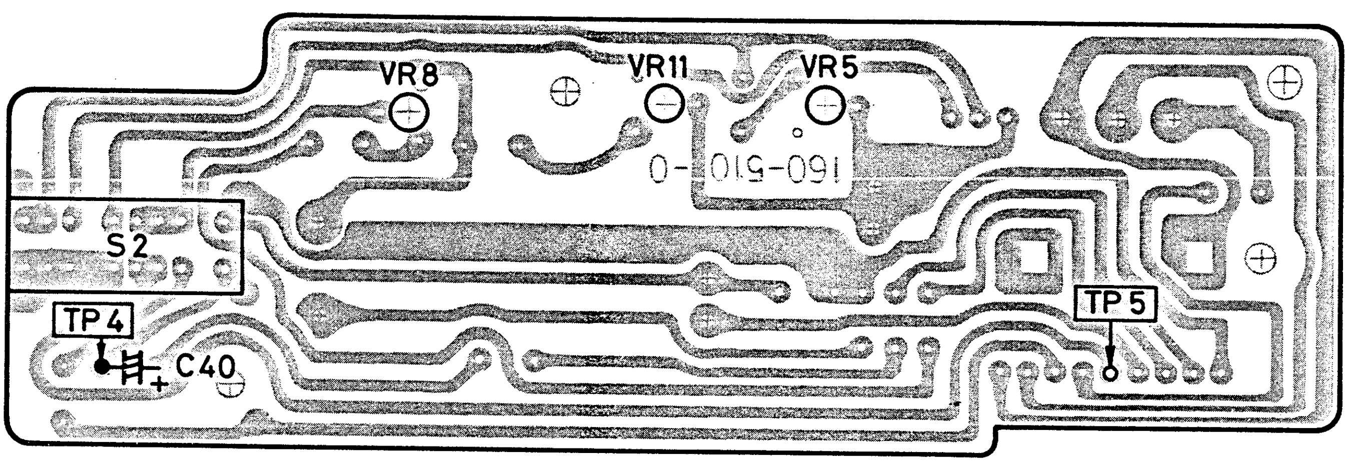


Figure 4

PRINCIPLE OF VSC OPERATION

For a variety of applications, one would like to play back recorded speech at speeds faster than the original recording speed and still have the correct pitch, for maximum intelligibility.
VSC (Variable Speech Control) provides a practical, low cost method of accomplishing this.
When recorded speech is played back at a speed different from that at which it was recorded, all the frequency components are changed proportionally. If the speech is speeded up, its waveform is "compressed" into a shorter time frame and all frequencies are increased. The result is a distorted, high-pitched ("chipmunk" sounding) voice. To correct this distortion, the waveform must be restored to its original frequencies as illustrated in Figure 5-1.

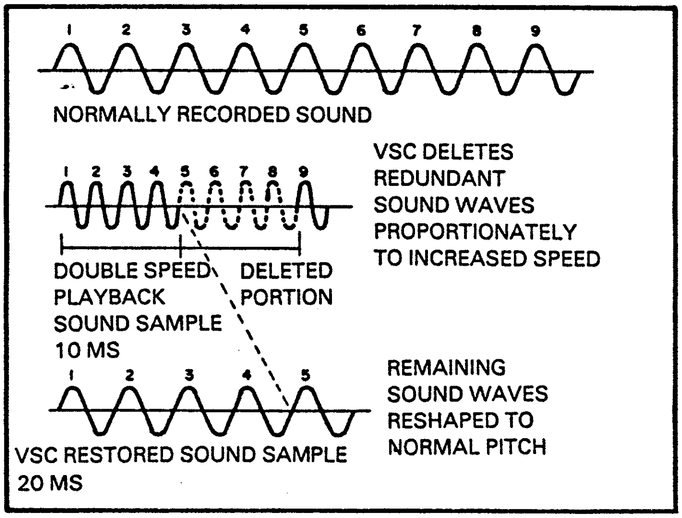


Figure 5-1 VSC Method of Pitch Restoration of Speeded Speech.

This restoration consists of "stretching" the waveform back to its original shape. To make room for the stretched pieces, alternate portions of the waveform must be deleted. Fortunately, this can be done without impairing intelligibility, since there is a great deal of redundancy in speech. By keeping the discarded portions short and splicing the stretched (pitch-corrected) segments only at zero crossings, the effect of discontinuities is minimized.
To determine the amount of pitch correction needed at any speeded up playback speed, we first define a term "C", the speech compression factor, equal to the factor by which the recording has been speed up: C = playback speed/record speed. Without pitch correction, the frequency of any signal played back in this way would be:

$f_{out} = Cf_{in}$ where: f_{out} = frequency of play back signal
f_{in} = frequency of original recorded signal

The pitch correction circuit must produce a pitch change factor, P, such that
P = I/C, for all C. If this is done,

$$f_{out} = PCf_{in} = \frac{1}{C} CF_{in} = f_{in}$$

and the frequency of the processed output signal will be the same as the original frequency of the recorded signal. Thus, pitch corrected, intelligible "compressed speech" is obtained.

While many approaches to speech compression are covered by the various VSC patents, (including two-channel, read/write "bi-frequency" techniques), the approach used in the new VSC-474 IC is considered to be the most cost-effective. It is a single-channel "variable delay line" technique. That is, pitch restoration of the speeded audio signal is accomplished by gradually increasing the delay of a variable delay line during each sample segment, while audio signal information is continually fed into the delay line. When the maximum delay is reached, the output is blanked for a short period while the delay cycle is reset (to the minimum delay) and processing of the next segment is begun. During this blanking time the required signal segment deletion is automatically accomplished. (see Figure 5-2)

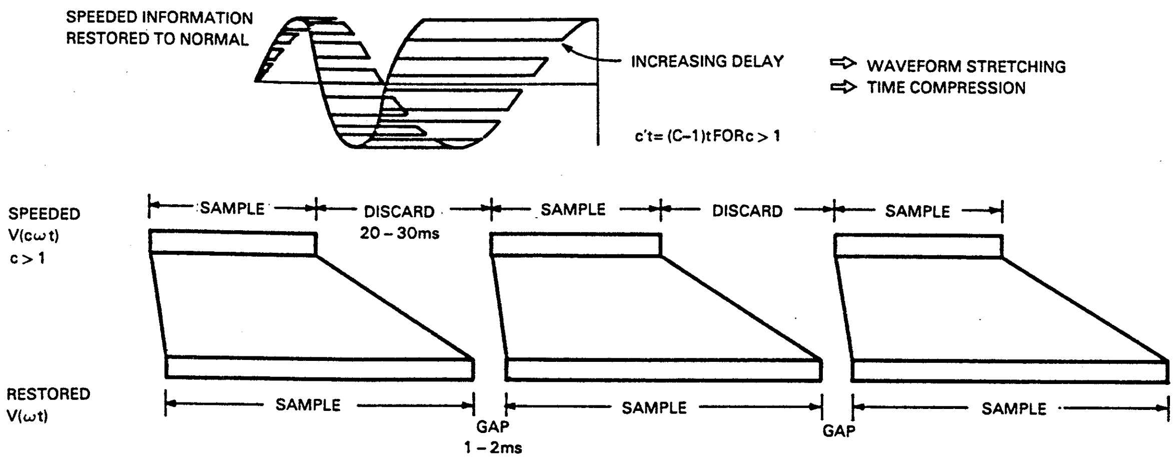


Figure 5-2 Signal Stretching by Increasing Delay

A Bucket Brigade Device (BBD), which is an analog shift register, can be used as the required variable delay line, since its delay time can be controlled by an external clock driver.
It can be shown that if the frequency (period) of this clock driver varies continuously and in such a way that each clock pulse period maintains a constant ratio to the previous clock pulse (i.e. the clock pulse periods form a geometrical progression), then frequency change by a factor (P) will occur to any signal passing through the BBD. The actual relationship is as follows:

$$\frac{1}{P} = R^n$$

Where P is the pitch factor

R is the geometric ratio, and

n is the number of stages in the BBD

And, as shown above, if $\frac{1}{P} = C$, for all C, the requirements for speech compression will be satisfied.

GENERAL DESCRIPTION OF VSC-474 INTEGRATED CIRCUIT

The VSC-474 is a low-voltage monolithic bi-polar integrated circuit in a 16 pin dip package. It provides all the circuitry for the BBD clock drivers, logic for segment splicing (including blanking), synchronized motor speed control circuit, etc. The addition of a BBD, filter circuits, and a motor drive circuit completes the system.

This VSC system is capable of producing cost-effective speech compression at up to 2.5 times original recording speed. Its low cost and low voltage operation make it ideal for use in portable, 6 volt consumer cassette players. It can also be effectively applied in dictating machines, telephone answering devices, and video players (to produce intelligible audio during speeded video playback) and other applications.

The VSC-474 IC contains:

a. A ramp generator and its reset I^2L logic circuitry.
b. A voltage controlled period generator (VCPG), which generates the proper series of transfer frequencies for the analog shift register, combined with a 2-phase clock driver which can be directly interfaced with available BBD's.
c. A zero crossing detector.
d. Blankable audio amplifier.
e. A phase lock loop (PLL) motor speed control circuit providing drive and brake capabilities.
f. An exponential generator (actually incorporated in the motor speed control circuit) which is required because (due to the geometry of the VCPG) the pitch correction factor varies exponentially with the control voltage. This circuit ensures that motor speed will vary in exactly the same way, for accurate tracking.

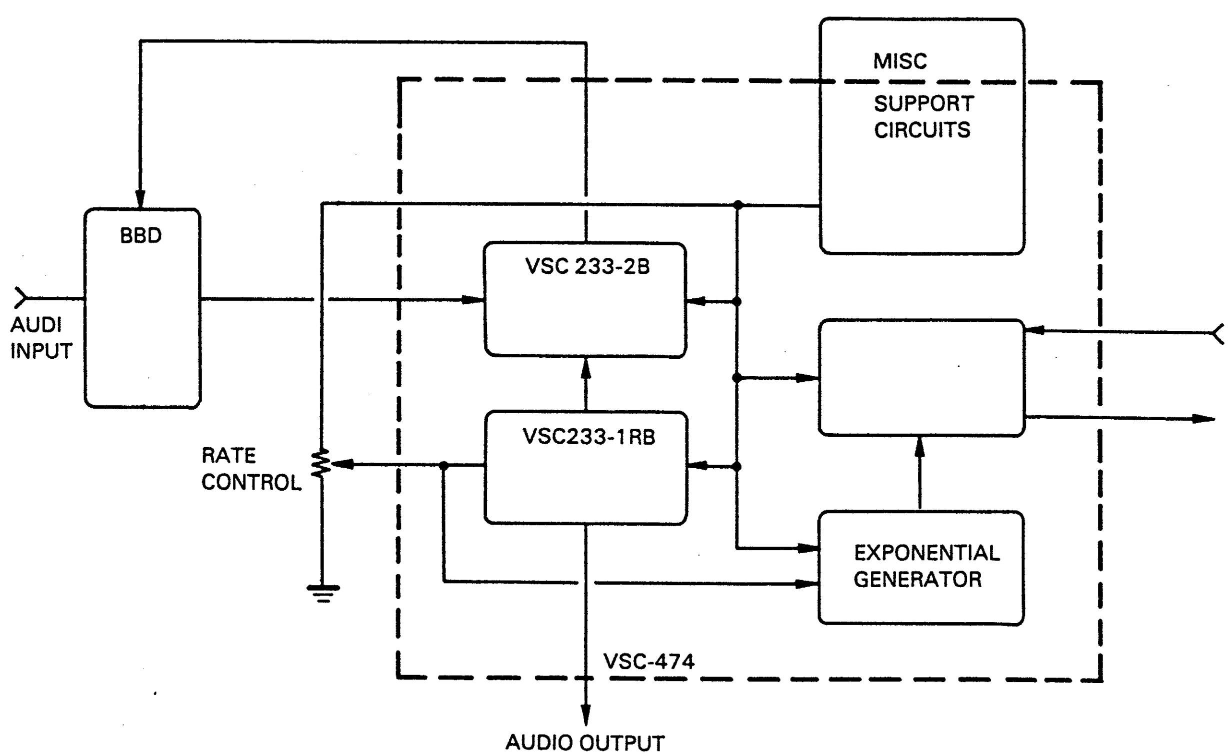


Figure 6-1 The VSC-474 IC includes all circuitry and ICs indicated within the dotted lines.

VSC474 SPEECH COMPRESSION IC BLOCK DIAGRAM

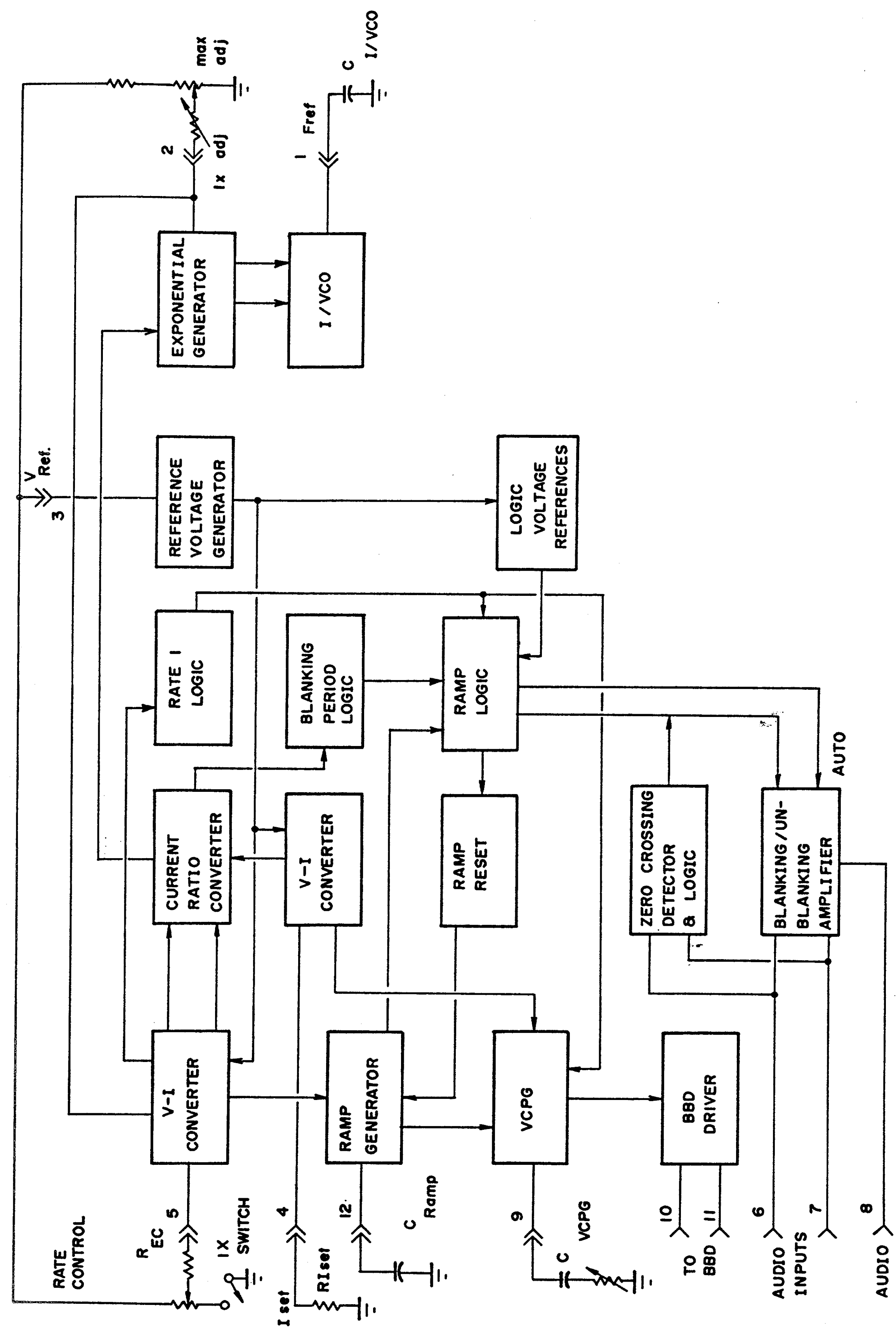


Fig. 4

FUNCTIONAL DESCRIPTION (refer to Figure 7-1)

Ramp Generator:

The "ever-increasing delay", which is one of the central features of the variable delay line type of VSC process, is based on a sawtooth waveform (series of ramps) generated at pin 12. The compression rate command voltage V_{EC} referenced to V_{ref} (pin 3) is developed across R_{EC} (pin 5) as the rate control pot goes toward ground (0 volt), causing a current to flow out of pin 5 and thru R_{EC}. This current is internally mirrored to pin 12 for charging of the ramp capacitor.

The ramp duration (sample period) for various VSC compression rates is as follows, assuming auto reset is triggered:

VSC Rate	Ramp Duration (ms)
1.1	220
1.2	115
1.3	80
1.5	52
1.75	38
2.0	30

The ramp waveform is buffered and fed to the input of 3 comparators. These comparators "look" at different ramp voltage points, each of which is a certain fraction of the referenced voltage at pin 3. See Figure 8-1 for action of the ramp voltage in relation to the 3 comparators (for logic point s A, C, and D.)

Ramp Reset (refer to Figures 7-1 and 8-1):

Ramp reset occurs under the following conditions. (a) when the "zero crossing enable" occurs (at 17% below the top of the ramp) the ramp will reset at the next zero crossing of the audi, signal at pins 6 and 7. (b) when no zero crossing occurs during this period (final 17% of ramp), the ramp will automatically reset at the end of this period (top of the ramp). When there is no audio signal, the "zero crossing" condition is automatically satisfied.

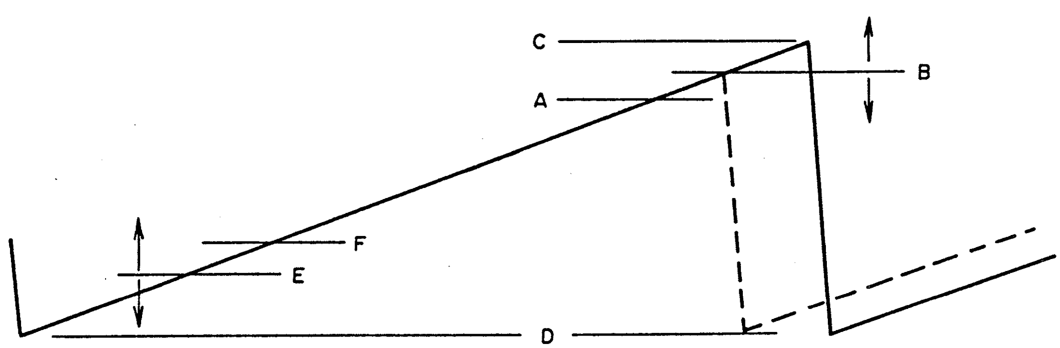


Figure 8-1

LOGIC A: to blank at next zero crossing (zero crossing enable)
B: Zero crossing present, blank audio and reset ramp
C: Auto reset, zero crossing not required
D: End of reset, start ramp again.
E: Unblank at the next zero crossing (unblank enable)
F: Zero crossing present, unblank audio.

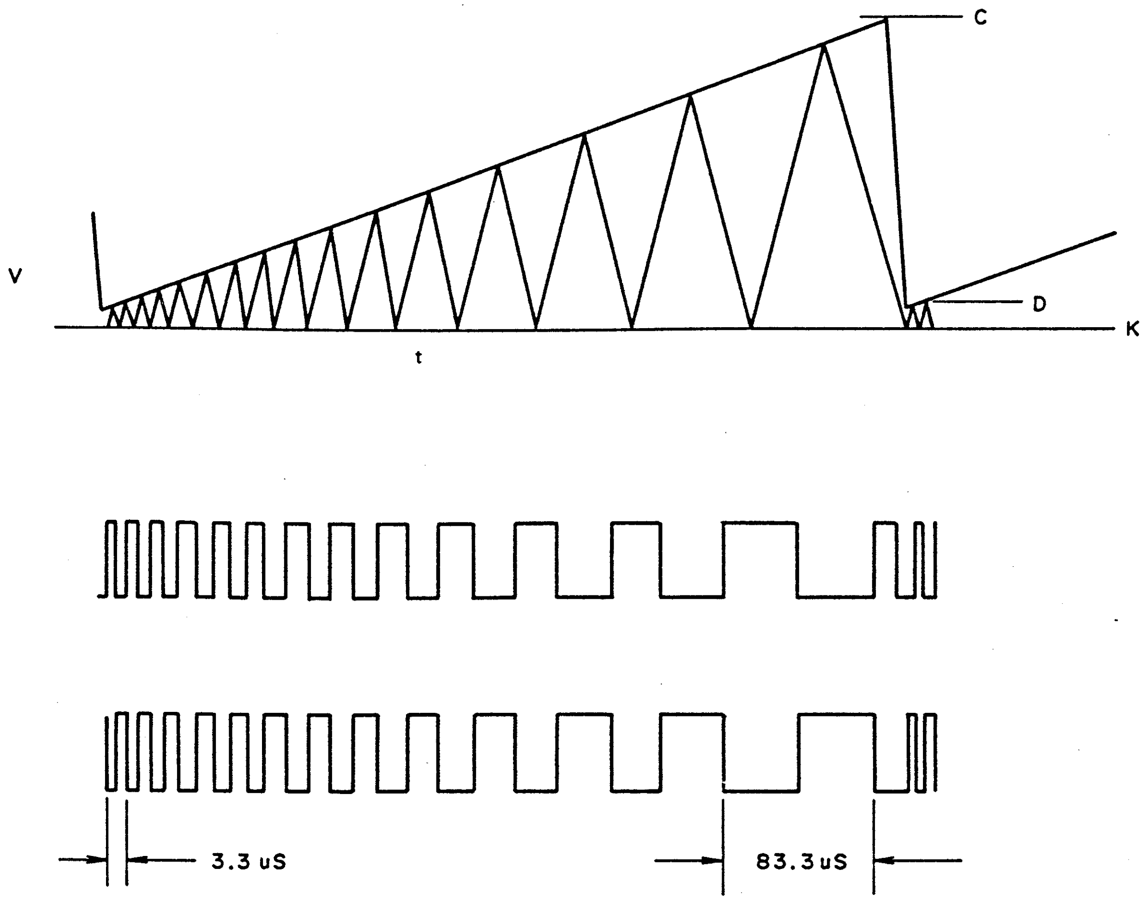


Fig. 8-2

Voltage Controlled Period Generator (VCPG) and BBD Driver (refer to Figures 7-1 and 8-2):
As explained in Section above, VSC requires that the BBD be driven by a (square wave) clock driver, such that the period of each clock pulse maintains a constant ratio to the previous pulse (i.e. the clock pulse periods form a geometric progression). This required waveform is generated in the VCPG in the following way. A charge-discharge circuit produces a symmetrical triangular wave which is bounded by the ramp on the top side and a fixed voltage level on the bottom side. This envolope causes the period of the triangular wave to change geometrically. The geometric ratio is a function of the slope of the ramp:

$$R = \frac{1 + K \tan \theta}{1 - K \tan \theta}$$

K = the slope of the VCPG waveform (Lø)
$\tan \theta$ = the slope of the ramp
R = geometric ratio (see previous)

This, in turn defines P, the pitch correction factor, since $\frac{1}{P} = R^N$ and N, the number of stages of the BBD, is fixed.
To actually drive the BBD, the BBD Clock Driver converts the output of the VCPG into two square wave clock driver signals of opposite phase (see Figure 8-2).
The shortest period of the swept VCPG is 3.3 μs due to the fact that it is impractical to clock most BBD's at frequencies higher than 300 kHz. On the other hand, the longest period of the swept VCPG should not exceed approximately 83 μs (12 kHz clocking frequency), since a lower clock frequency would make it difficult to pass the higher audio frequencies in the audio range of interest. The result is a sweep range of about 25 to 1 and hence, in Figure 8-2, the voltage between logic points C and K must be 25 times the voltage between points D and K. These logic points on the ramp must therefore be generated by precise ratios from the reference voltage at pin 3.

Blanking (Discard) Logic (refer to Figures 7-1 and 8-3):

When the ramp is reset, there is a discontinuity in the geometric progression. Just prior to the ramp reset, audio information is being loaded into (and transferred through) the BBD at a transfer frequency approaching 12 kHz. After the ramp resets (assuming an instantaneous reset with zero reset time), the loading and unloading are now suddenly occuring at a rate of approximately 300 kHz. The audio information in the BBD, which was loaded in a 12 kHz rate is being unloaded at a 300 kHz rate. This causes the audio information being unloaded just after the ramp reset to be shifted <u>up</u> in frequency by a factor of approximately 25X instead of being properly pitch-corrected.

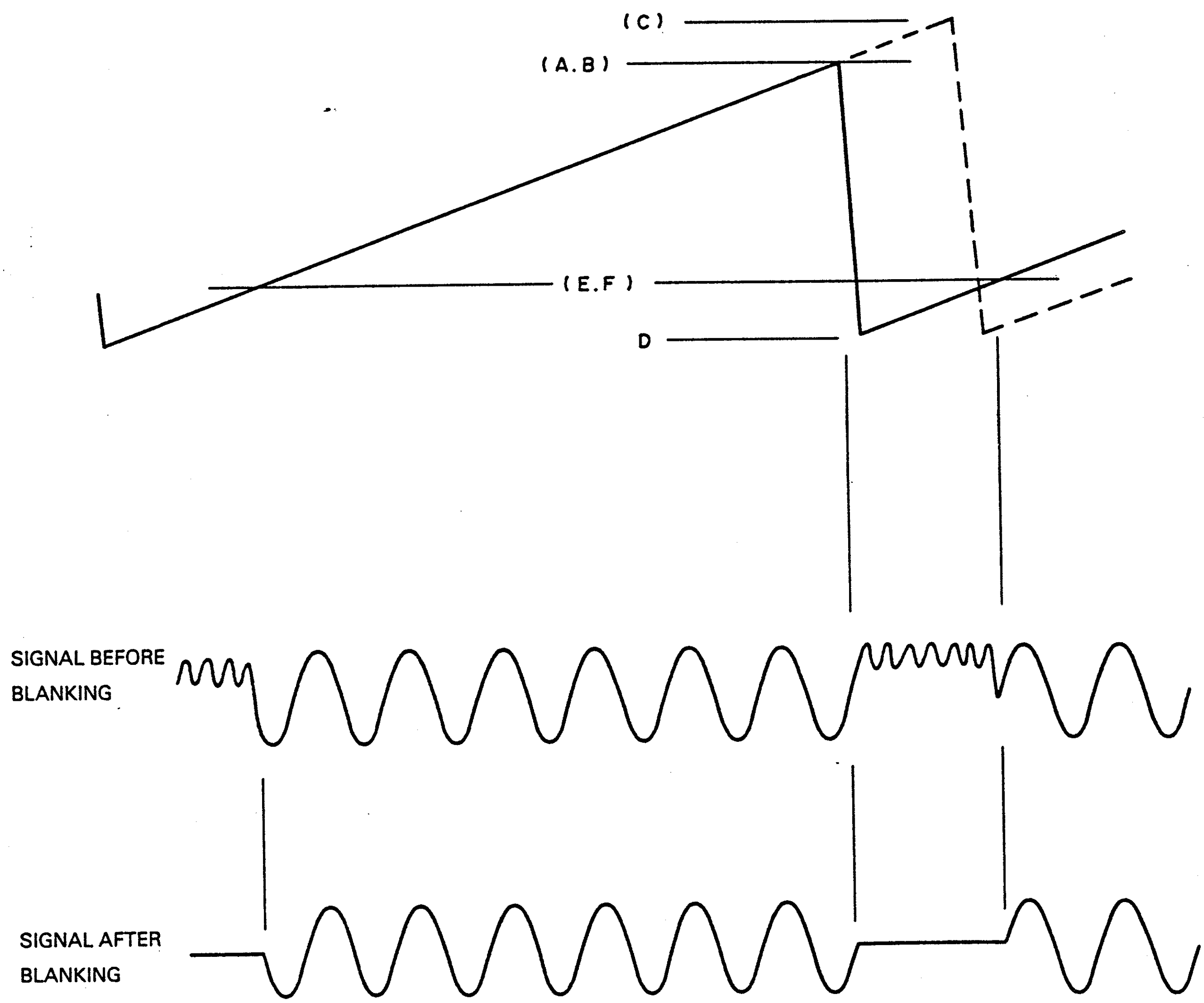


Figure 8-3

The blanking period is required until all information which had been loaded into the BBD at a low frequency is unloaded at the high frequency i.e., while improperly pitch-corrected information is "dumped." Once information which was clocked into the BBD after ramp reset, begins to appear at the output of the BBD, the signal will be properly pitch-corrected and the output may be unblanked.

It can be shown that the length of the blanking period should be long enough for the 512 cell BBD (256 stages) to be clocked 256 times.

The time required for the 256 count is:

$$T = \sum_{i=1}^{256} P_i$$ where Pi is the period of the i^{th} clock pulse after reset

$$\text{or} = \sum_{i-1}^{256} R^{i-1} P1$$ where R, the geometric ratio, is $C^{\frac{1}{256}}$ and P_1, the first clock pulse period, is $3.33\mu s$

$$= P_1 \frac{(R^n-1)}{R-1}$$

$$= P_1 \frac{(C-1)}{R-1}$$

Typically varies from approximately 1.05 ms for C= 1.5X to 1.228 ms for C= 2X. The VSC-474 IC generates a fixed blanking period of approximately 2 ms. This period is generated by a mirrored I_{EC} through a resistor to create a threshold voltage for the blanking comparator/logic.

The I_{EC}, the angle o (ramp) and the logic E point (see Figure 8-1) track together to generate a nearly constant blanking period. It is very important that the ramp be linear to maintain a constant blanking period (and, of course, accurate pitch correction for the whole duration of each sample period). To insure good linearity of the ramp, a high-grade capacitor with low series resistance such as a polyester (mylar) capacitor, must be used.

Audio Amplifier Blanking and Unblanking Logic (refer to Figures 7-1, 8-1 and 5-3):

The logic causes the audio amplifier to blank during the signal discard interval and unblank when the discard has been completed.

The logic function is:

Blank = (A·B) + C

Unblank = (E·F) (refer to Figure 8-1)

Blank and unblank become inputs to an R/S flip-flop which then gates the audio amplifier.

Zero Crossing Detector and Audio Amplifier (refer to Figure 4-1):

The zero crossing detector inputs are connected in parallel with the differential inputs of the Audio Amplifier. The detector is actually a window comparator ±5mV wide, centered at the blank level output of the Audio Amplifier. The detector output, which is independent of the Audio Amplifier, is used in the Blank and Unblank control logic function described in previously.

The Audio Amplifier, is a differential input unity-gain device capable of driving a 1K ohm load. It can be blanked to a zero-gain state by the Blank Logic without switching transients. When this blank action is always initiated at a signal zero crossing, minimum disturbance is introduced in the reconstructed output signal. Input offset adjustment to center or match the zero baseline from blanking to unblanking is not necessary, due to the excellent internal matching of devices.

Exponential Generator (refer to Figure 7-1):

In most speech compression applications, it is required that a single control be used. That is, the V_{EC} produced by the rate control potentiometer R_{EC} must control both the motor speed and the pitch correction factor P. It can be shown from the geometry of the ramp and triangular wave that R, the geometric ratio of the clock driver pules, is equal to $\left(\frac{1+K_1 \tan\theta}{1-K_1 \tan\theta}\right)$, where $\tan\theta$ is the slope of the ramp, then,

$$C = \frac{1}{P} = R^N = \left(\frac{1+K_1 \tan\theta}{1-K_1 \tan\theta}\right)^N$$

Since $\tan\theta = \frac{I_{EC}}{C_{ramp}}$,

$$C = \left(\frac{1+K_2 I_{EC}}{1-K_2 I_{EC}}\right)^N \qquad \text{Where} \quad K_2 = \frac{K_1}{C_{ramp}} \qquad \text{Or,}$$

$$C = \left(\frac{1+X}{1-X}\right)^N \qquad \text{Where} \quad X = K_2 I_{EC}$$

(I_{EC} being the "rate control current" from pin 5)

Taking the ln of both sides of the equation, we have the log series.

$$\ln C = N \ln \frac{(1+X)}{(1-X)} = 2N\left(X + \frac{X^3}{3} + \frac{X^5}{5} + \cdots\right) \cong 2NX$$

since X is very small for N very large and C in the range of interest.

Therefore, $C \cong e^{2NX} = e^{2N(K_2 I_{EC})}$

Thus, the compression ratio, C, varies exponentially with I_{EC}. For single lever control, motor speed must also vary exponentially with I_{EC}, and hence the need for an exponential generator. The output of the exponential generator drives an I/VCO.

I/VCO (refer to Figure 4-1):
The I/VCO provides a triangular wave at pin 1, the frequency of which varies exponentially in relation to I_{EC}, is given by the following:

$$\Delta f = 2\,\frac{\Delta I_{EC} + A}{(B - \Delta V_o)\,C_{I/VO}}$$

Where $V_o = D(\quad I_{EC})$
and A, B and D are constants.

It can be shown that this indicates a relationship between f and I_{EC} which closely approximates an exponential function. Thus, the result is a system which provides a linear relationship to the amount of pitch correction, expressed as C, for the range of the IC (i.e. C=1 to C=2). This reference frequency is converted to a voltage and used to drive a motor. This makes possible a single lever system in which the amount of pitch correction will automatically "track" (be appropriate for) the tape speed, over a wide speed range.
In accordance with the signal flow inside the VSC circuit, the VSC works on the 14-1002 in the following way.
The signals amplified by the IC-1, Audio Amp, are input to the IC-3, the ALC Amp, which keeps the level of signal constant (1.2 V p-p). This is to maintain the signal level not to exceed the linear operation range of IC-4, Analog Shift Register.
The signals are then applied to IC-4, and, in accordance with the clock signal frequency generated by pins 10 & 11 of IC6, are output from pins 7 & 8 of IC-4. VR6 adjusts for the minimum distortion on these signals. VR7 adjusts to minimize clock signal. However, the clock signal cannot totally be eliminated by VR7, so the Low Pass Filter (IC-5) is incorporated to remove the clock signal. The output from IC-5 pin 7 is then applied to VSC circuit (IC-6).
The control voltage, set by the Pitch Control VR3, is processed to the Ramp Signal generating circuit, and determines the slope and direction of the Ramp signal. The amplitude of this signal is controlled to be fixed by certain value, and the repetition frequency is changed by the value of slope. The repetition frequency can be checked at pin 12. The value at VR3 maximum setting (x2) is determined by VR8 to be 24.9 ms.
Then the Ramp signal is added to the circuit which generates the variable frequency clock signal. The clock signal is output at pins 10 & 11, which drives IC-4, the Analog Shift Register. The clock signal changes in the range of 300 kHz to 12 kHz. 300 kHz signal is set by the VR10, 12 kHz signal is set by VR9.
The rest of VSC circuit operation is as outlined in the previous section.
The motor speed control is done by the IC-2 and the SPEED control VR (VR2) and not directly related to VSC circuit.

THE WMFO LINE AMPLIFIER

INPUTS:

* 600-Ohm Balanced Audio In (L & R)
* Unbalanced Audio In (L & R)

OUTPUTS:

* Unbalanced Audio Out (L & R)
* 600-Ohm Balanced Audio Out (L & R)

THE LINE AMPLIFIER

I) Function and Operation:

The line amplifier circuitry is used to convert 600-Ohm balanced audio signals (which is the "professional" convention used at WMFO) into single-ended signals (as used by most "home" audio equipment). The opposite function (conversion from unbalanced to balanced format) is also accomplished. All line amplifier units functioning at WMFO provide a set of female RCA jacks for single-ended input and output. Unbalanced audio lines are connected via barrier strips, which are mounted on the panel or box housing the circuitry and are generally hand-labeled. Two triple line-amplifier units (offering three channels of stereo balanced-to-single-ended conversion and vice-versa) have been installed at WMFO. One is in the Studio C rack, where it buffers the cassette deck, the Revox Reel-to-reel, and the Equalizer. The other is in the Studio A rack, where it buffers the two cassette decks (there is a spare line amplifier card in this unit). A single-channel line amplifier is installed in Studio D (mounted in a box on the wall opposite the windows), where it interfaces the Studio D stereo system to Master Control and the rest of the station. These units are marked "Always Leave ON!", since removal of power will cause the correspondingly buffered devices to be effectively isolated from the outside world (ie. they will be unable to receive input and send output). One-half of a line amplifier (the single-ended to balanced portion) has been installed in the grey Monitor rack in Master Control to buffer the "Spare Tuner" outputs.

There are no controls on the front panels of the line amplifier units aside from a power switch (which should generally be left on except in the case of a studio shutdown), thus their operation is completely transparent to the typical disc jockey. It is possible, however, to adjust the gains across the balanced/single-ended barrier by setting trimpots in the line amplifier circuitry as discussed below, thus allowing one to "customize" the interfaces to particular devices.

II) Circuit Description:

The line amplifier circuitry is exceedingly simple, as seen in the single-page schematic depicting a balanced to single-ended converter and its opposite partner. 5532 Operational Amplifiers are used exclusively for their low distortion, high bandwidth, and ability to directly drive 600-Ohm loads at moderate amplitudes. The capacitors C1 and C2 in schematic A may degrade the audio quality slightly (although there aren't many people who would notice this); they are present in all line amplifier circuits except for those installed in Studio C. One 5532 is required for the two differential amplifiers used to convert stereo balanced lines to single-ended format, while two 5532's are required to convert a pair of single-ended lines to a stereo balanced set (ie. the schematic shows only one stereo channel).

III) Calibration and Adjustment:

There are two trimmers T1 to attenuate the single-ended feeds from the balanced pair, and two corresponding trimmers T2 to allow adjustment of gain when converting single-ended to balanced. In order to calibrate the single-ended feed of circuit A (which, for instance, supplies input to a cassete deck), patch a source of average amplitude (the House Tone is perfect) into the balanced input, and set the input level on the single-ended device (ie. the cassete deck) to a moderate value. Adjust the two T1 trimpots (one for left & one for right channels) until both channels of the device under test exhibit sufficent and identical response (ie. both VU meters of the cassete deck read at zero dB).

In order to calibrate the single-ended to balanced circuit (Schematic B), output a steady mono sound of moderate volume from the device (ie. play a tape on the cassete deck of the House Tone recorded at zero VU), and patch the balanced line amplifier output to a fader on the local board set at half-pot. Adjust the two trimmers T2 until both meters on the board read at zero VU. The line amplifier circuit is now calibrated and ready for operation.

Unfortunately, I have no general drawing depicting the location of trimmers T1 and T2 on the line amplifier circuit cards (since these cards were all hand-assembled, there is some small difference between individuals). For trimmers T1, look for two trimpots clustered around a single OP-AMP chip. The two T2 trimmers are each dedicated to an individual 5532, so these may be positioned farther apart. That's the only advice I can offer; otherwise it's trial and error.

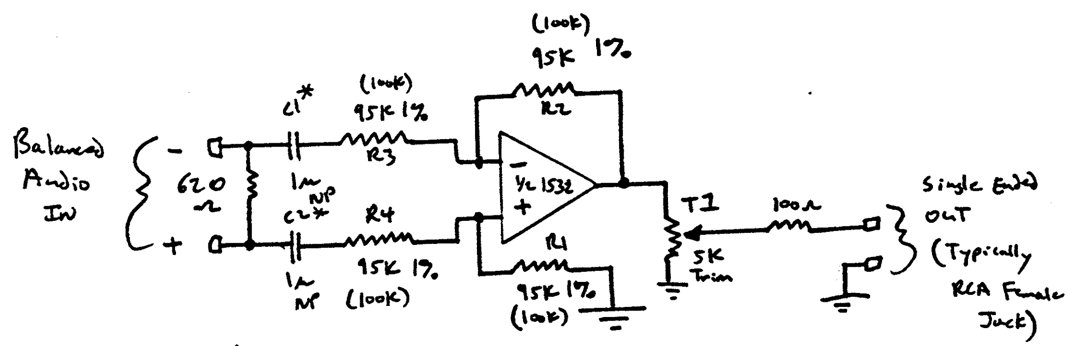
III) Troubleshooting Hints:

These units are so simple that none have failed so far. If there seem to be problems, first check the RCA cables or single-ended device under question, as these will probably be at fault. Otherwise first check the power supply voltages, and then perhaps try swapping or replacing 5532's (they're all socketed). Before loosing hope, search for bad solder joints; there may be a few of them still buried in one of these cards. Also look for broken wires leading to-and-from the barrier strips mounted behind the front panel.

"Standard" WMFO Line Amplifier Circuit

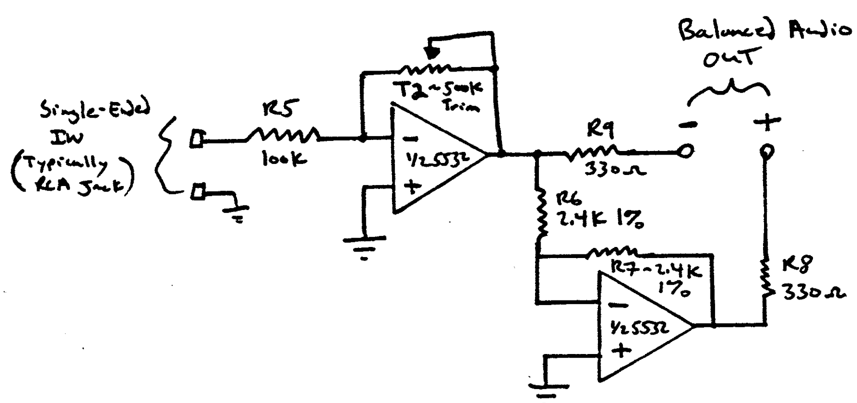
11-Nov-86

-J. Paradiso



*- In some units (Particularly in Studio C) C1 and C2 are not present.

A) Balanced To Single-Ended Converter (This circuit is repeated for stereo, using one 5532)



B) Single-Ended To Balanced Converter (This circuit is repeated for stereo, using two 5532's)

Notes: These circuits are used throughout WMFO as a buffer between high-impedance, single-ended audio equipment (ie. cassette decks, Equalizers, etc.) and standard 600Ω balanced audio which is used conventionally in the station. Several minor variants of these circuits exist.

Banks of three stereo line amplifiers are used in studios A and C to buffer cassette decks, etc. Line amplifiers are also used in studio D (to buffer the stereo console), and a line driver (B) is used in Master Control to buffer the spare Tuner output.

They run from conventional bi-polar supplies ranging from +/-12 → +/-18 Volts. Up to three are run together from a single supply ~ there is no crosstalk.

THE WMFO HOUSE TONE OSCILLATOR

INPUTS:

* Linear Sweep In (front panel)

OUTPUTS:

* Studio A Balanced House Tone Out
* Studio B Balanced House Tone Out
* Studio C Balanced House Tone Out
* Sine, Triangle, & Square Out (panel)

THE HOUSE-TONE OSCILLATOR UNIT

I) Function and Operation:

This unit is mounted in the blue inter-studio rack in Master Control. It is a sweepable function generator which produces the "House Tone" signal fed to all studios. Sine, triangle, and square wave sources can be selected independently for each studio. The frequency of the oscillator can be adjusted from (these are approximate guesses...) 0.1 - 200 Hz (range switch on "low") or 80 Hz - 20 Khz (range switch on "high"). An external frequency modulation voltage can be applied via a set of front panel terminals.

The house tone serves several purposes at WMFO. First of all, it is the common signal source to which all studios are calibrated (ie. all boards are calibrated such that the house tone signal will produce a 0 dB meter deflection at half-pot). All line amplifiers and tape-deck inputs are also adjusted so a 0 dB meter swing is produced with the house tone input at a moderate record level. Thus the house tone amplitude is the standard to which everything at WMFO is ultimately referenced. The house tone frequency should be quiescently left at 1 Khz (the appropriate settings on the range switch and frequency pot are marked with a pen). The house tone amplitude is one of the most important parameters governing WMFO, and wouldn't you know that I've forgotten what we actually set it to (I'm fairly certain that we're putting out a volt or two P-P into a 300 Ohm load (ie. two stereo channels bridged to mono). Sorry, but I can't quote you an exact figure in dBm. For details in adjusting this amplitude, see the "calibration" setting below.

The house tone can also be used "qualitatively" to set record levels on tape recorders, check out patches, etc. It also can be used as an audio effect; the frequency may be adjusted and swept by hand via pot P1 on the front panel (see diagram on second page of schematic). Interesting effects can also be derived by inputing a modulation signal (ie. another oscillator or envelope generator) at the "Sweep In" terminals on the front panel. These terminals are high impedance inputs (100K Ohms) referenced to ground (beware; the black terminal is actually chassis ground). In order to suppress spurious noise in the signal, these terminals are shorted by a guard strap when not in use. When inputting an external modulation signal, remove this guard strap (it is affixed to the black terminal, and may be swung around when both terminals are loosened). Make sure that this strap is returned after finishing with the modulation input.

The waveform is selected for each studio via rotary switches S2 - S4. These should normally be set on the sine position, however for diagnostic tests and/or special effects, triangle and square waves can also be selected. These switches are multi-position rotaries, thus a "blank" setting results in no audio being transmitted to the corresponding studio (ie. blank settings are OFF).

Single-ended outputs for sine, triangle, and square waves are available from terminal posts mounted on the right side of the front panel. These may be used for monitoring or testing in Master Control. The balanced house tone outputs are routed through the Master Control patchfield before being sent to the various studios, thus one may interrupt this connection and patch a house tone output to any desired destination. This is an extremely useful technique to employ when testing connections, devices, and complex patch schemes.

Since the house tone should always be made available to all studios, this unit should generally be left on, except in the case of a station shutdown. Remember to return the frequency adjustment (P1), range switch (S1), waveform selection switches (S2 - S4), and sweep input ground strap to their default positions after changing them for any purpose.

II) Circuit Description:

The first page of the schematic shows all circuitry involved. The 8038 function generator (IC1) is used to make all waveforms. IC5 is an inverting summer which combines voltage from the "frequency" pot P1 with signals at the "Linear Sweep" input. The output of IC5 feeds the voltage control input of the 8038 (at pin 8). Zener D1 protects the 8038 modulation input from severe voltage swings. The frequency of the 8038 is determined by the output of IC5, together with the values of timing resistors R1 & R2 and timing capacitors C4 & C5. Trimmers T3-T5 set the amplitudes for sine, triangle, and square waveforms. Trimmer T2 adjusts the shape (hence "purity") of the sine wave. Each output is AC coupled and buffered by voltage followers IC2-IC4. The outputs of these voltage followers are routed directly to the terminal posts on the front panel. They are also bussed to the "waveform select" rotary switches, which route a specified waveform to 5532-based differential drivers IC6-IC8, which boost the input by a factor 1.25 before sending the signal to the respective studios. Note that these are mono outputs; the stereo taps on the patchfields are bussed together for the house tone signal, resulting in a net 300-Ohm load. The 5532's have no difficulty in driving these loads at the current house tone signal levels.

The power supply is depicted at the bottom of the schematic. There are two bipolar supplies used in the house tone circuitry; +/-12 Volts to drive the 8038, and +/-15 Volts to drive the OP-AMPs and related networks. This was done in order to derive maximum range from the 8038, whose modulation input saturates at about 400 milliVolts above its positive supply rail. Thus the 8038 was run at a lower voltage to enable the OP-AMPs to reach this maximum. This lower voltage on the 8038 also causes it to run a bit cooler; the chip does generate some heat due to passive dissapation in the sine shaping network. To allieviate any problem this may cause, I've stuck a heat sink onto it with a bit of silicon mounting paste. If one replaces or fools with this chip, please remember to replace this heat sink.

III) Calibration and Adjustment:

Trimmer locations are depicted in a circuit card diagram drawn on the second page of the schematic. Before setting any adjustments, make sure the device has remained under power for at least a half hour or so to allow thermal transients to settle. During calibration, an oscilloscope can monitor the waveform outputs via the front-panel terminal posts.

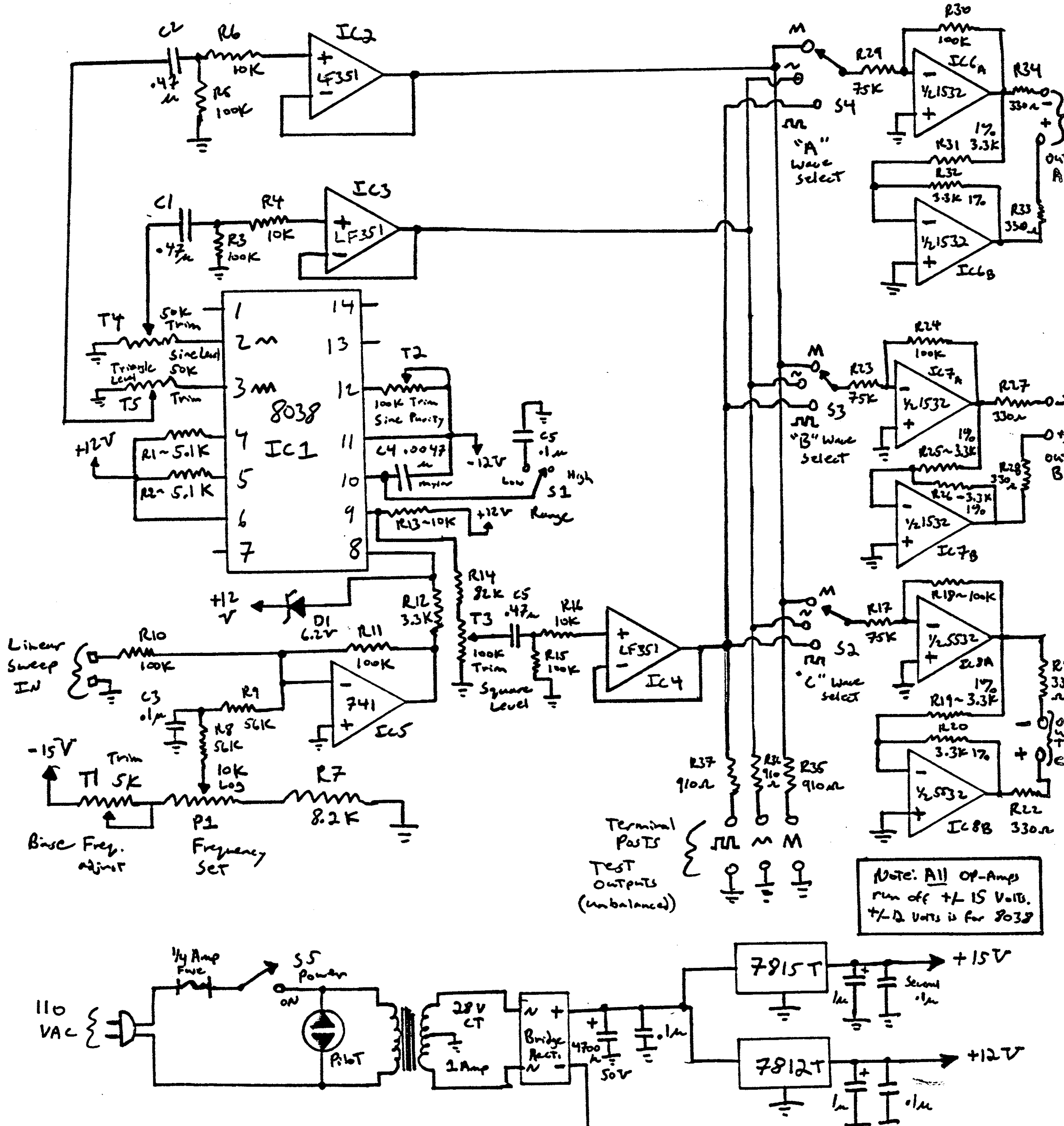
Trimmer T1 adjusts the base frequency of the oscillator. To set this, put switch S1 on "high", and set P1 at minimum. Adjust T1 so that the 8038 yields a stable waveform at the lowest possible frequency (the best waveform to monitor while making this measurement may be the triangle; the symmetry may be significantly skewed at extremely low frequencies). If T1 is way out of alignment, the range of P1 may be quite limited, and the oscillator may shut down when P1 is set to its lowest frequency extreme.

Trimmer T2 sets the shape of the sine wave. Monitor the sine wave with an oscilloscope, and set the oscillator frequency to around 1 kHz. Adjust T2 until the best-looking and most symmetric sinusoid is obtained.

Trimmers T3-T5 set the output levels for each oscillator waveform. Adjust one of these (perhaps T4) to yield the desired signal (perhaps throw a differential meter across a balanced output when it is driving a full 300-Ohm load, and set T4 appropriately). After T4 is set, patch the house tone into the board in one of the studios, and set its input pot to yield 0 dB on the sine wave. Now set the appropriate selector switch to send the triangle wave down, and adjust T5 until the meter in the board again reads at zero dB (this may be a two-person operation; use the intercom for convienent communication). Now select the square wave, and set T3 to once more yield zero dB on the board. All waveforms will now output at the same RMS level, and the house tone oscillator is now calibrated.

IV) Troubleshooting Hints:

This unit has behaved reasonably well during its 1.5 years of use. Once a regulator blew and dropped a supply rail (I think this also took out the 8038). This may have been a singular phenomenon. Watch for bad solder joints, and check the supply rails before coming to any conclusions. 8038 chips were once plentiful (even Radio Shack carried them when this unit was built). They shouldn't be too hard to locate if replacement is necessary (all chips are socketed). If one swaps the 8038, remember to re-mount the heat sink.



WMFO House Tone Oscillator

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WMFO House Tone Oscillator

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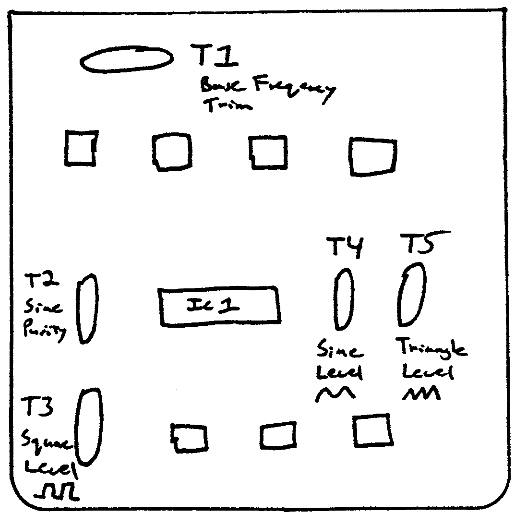
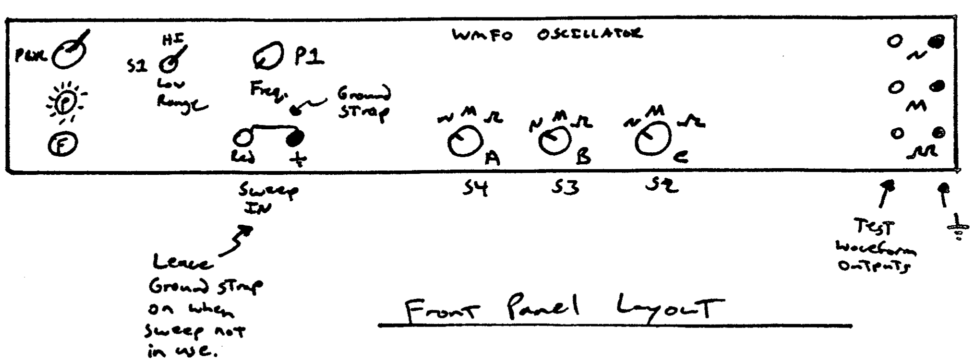


Circuit Layout



Front Panel Layout

THE WMFO UTILITY AMPLIFIER

INPUTS:

* Balanced Audio Input (L & R)
  (selectable 600 or 16K ohm)

* PA line (Halls & Offices)

* PA line (Studios)

OUTPUTS:

* 8-Ohm Speaker Output

* Stereo Headphone Output

* Left/Right/Mono single-ended
  outputs (on front panel)

THE UTILITY AMPLIFIER

I) Function and Operation:

The utility amplifier is mounted at the top of the blue inter-studio rack in Master Control. This unit allows one to monitor a balanced audio source from the Master Control patchfields. The source may be monitored in stereo via a headphone jack on the front panel. Either the left or right channels (or a mixed mono feed) may also be selected to drive the monitor speaker mounted in the ceiling of Master Control. The PA lines (as in other studios and offices) may also be routed to this speaker. Single-ended signals (for left, right, and mono sources) are available at the front panel (via 1/4", RCA, and terminal/bananna jacks). These feeds may drive conventional single-ended audio devices, or serve as taps for monitoring and testing. They may be used simultaneously.

The front panel is portrayed on the second page of the schematic. The input to the utility amplifier comes up on the Master Control patchfield. The single-ended line outputs are available at the left of the panel (the black terminal post is tied to chassis ground). The audio level present at the Left and Right outputs is unaffected by the volume (P1) or selector switch (S2 & S3) settings. The Mono outputs are affected by P1 (but still not S2 & S3).

Switch S1 allows one to set the impedance of the balanced inputs. In the down position, the inputs are loaded at 600 Ohms (which is the standard for WMFO patchfields). In the up position, the inputs are loaded at a mere 16K Ohms; this often yields a louder audio level, and can be used to monitor lines which are (for some reason...) at higher impedance.

A set of stereo headphones (any impedance) may be plugged in where indicated. The headphone volume is set by P1. P1 also sets the master volume of the speaker feed. In order to send the amplifier audio to the overhead speaker, "Audio Level" pot P2 should be set at maximum and selector S3 should be set to "Amp". The source (Left, Mono, Right) actually sent to the speaker is selected via S2. Adjust the speaker volume with P1. Do not adjust the speaker volume with P2 (leave it at maximum), since this may cause you to set P1 too high and overdrive the amplifier (causing much distortion). If one desires to turn the speaker off (ie. when monitoring with headphones), turn P2 entirely down, and set S3 to "PA Studios".

The speaker feed may also be assigned to the "Halls & Offices" PA line, or "Studios" PA line via S3. This assignment does not employ any active circuitry in the utility amplifier, thus the device may be powered off when monitoring PA lines. The L-Pad P2 will then exclusively control the speaker volume. When not in use, this unit is generally powered off (an exception for the devices outlined in this document), and the speaker select is set to one of the PA lines.

The ability to monitor any balanced audio feed (and selectively audition each channel) is extremely useful when setting up patches, remote broadcasts, or making tests in Master Control. This unit played an extremely vital role when the interstudio wiring was installed and debugged.

II) Circuit Description:

Most circuitry is depicted in the first page of the schematic diagram. The balanced audio inputs are converted to single-ended signals through differential amplifiers IC1. Switch S1 is a DPST which shunts 620-Ohm resistors across the inputs when low impedance is selected. The single-ended panel outputs are derived from the outputs of IC1, and are isolated via resistors R12-R17. Stereo pot P1 controls the amplitude of the signal fed to the remainder of the circuit. IC3 and IC4 are 5534 OP-AMPs, which are well suited as headphone amplifiers because of their sizable drive capability (I originally used LM13080's here, but they rapidly disintegrated due to the 30-volt supply rails). IC2 sums both left and right audio signals to produce a mono source (beware; the phase of this mono source is inverted with respect to the original left and right signals). This mono signal is also routed to a set of front panel jacks, and is isolated via R21-R23. The Left, Right, or Mono sources are selected by switch S2, which feeds the input of an integrated 5 Watt amplifier IC5. This is a 2002 IC in a heat-sunk TO-220 package. It runs off the +15 Volt supply, thus has internal biases that are isolated by C8, C9, & C11. C7 and C10 are important for stability.

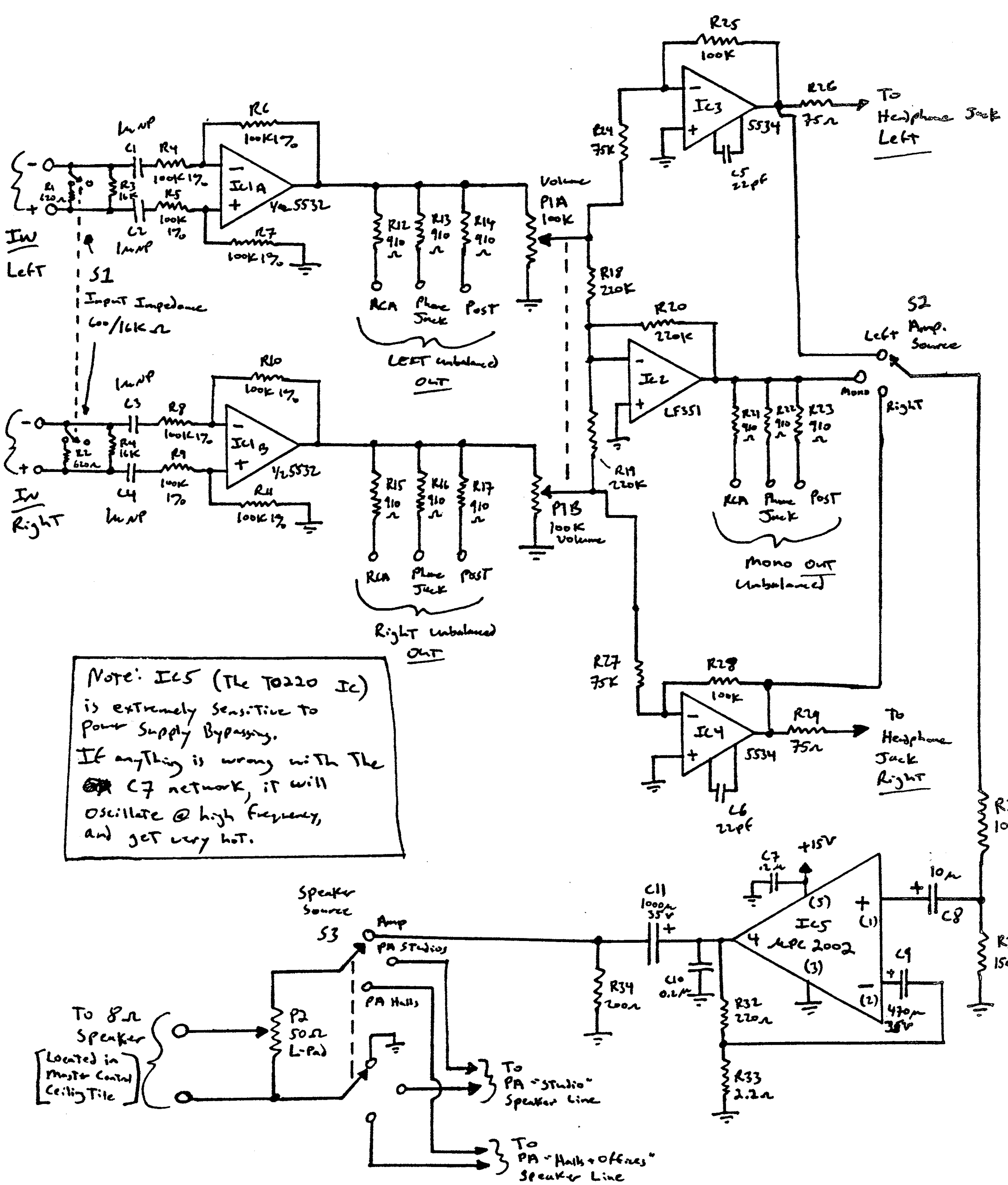
S3 selects the line which drives the ceiling-mounted 8-Ohm speaker. In the Amp position, the 2002 output drives the speaker directly; otherwise a PA line can be chosen (which does not involve any internal utility amplifier circuitry). P2 is a standard 50-Ohm Radio Shack L-Pad which attenuates the voltage applied to the speaker.

The power supply is depicted on the second page of the schematic. 7815 and 7915 regulators provide a stable +/- 15 Volt supply for all active circuitry. The +15 supply must also drive the 2002 audio amplifier, thus it is built for higher current. The 7815 regulator is a TO3 package heat sunk directly to the chassis. The 7915 is a TO 220 package mounted on the circuit card.

III) Troubleshooting Hints:

There's not much to go wrong with this circuit except (perhaps) for the 2002. In case of trouble, first check for bad solder joints, and make sure the power supply is working OK. The power supply bypass capacitor on the 2002 is CRITICAL!! This bypass must occur very close to the 2002's +15 and ground connections, and must be at least 0.2 mfd. If there's a problem here, the 2002 will oscillate across its supply rails at approximately 1 mHz, and become very hot. I've placed several capacitors across the 2002's power line to protect against this contingency.

WMFO UTILITY AMPLIFIER © Jun., 1987 – J. Paradiso



Note: IC5 (The TO220 IC) is extremely sensitive to power supply bypassing. If anything is wrong with the ~~C8~~ C7 network, it will oscillate @ high frequency, and get very hot.

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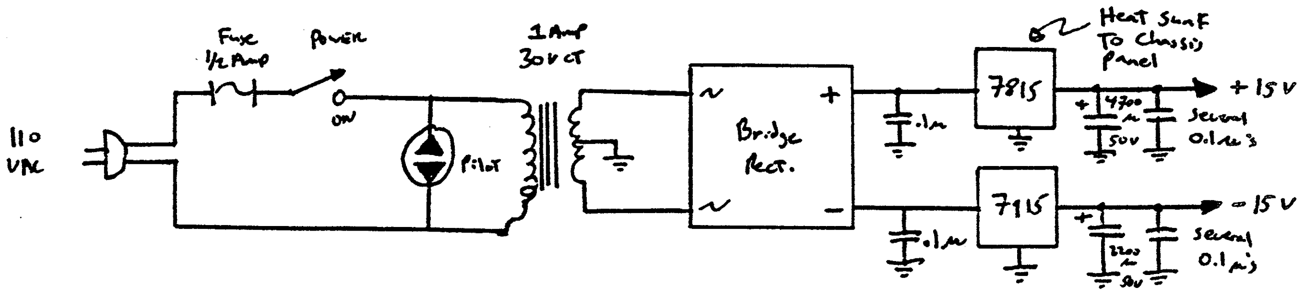
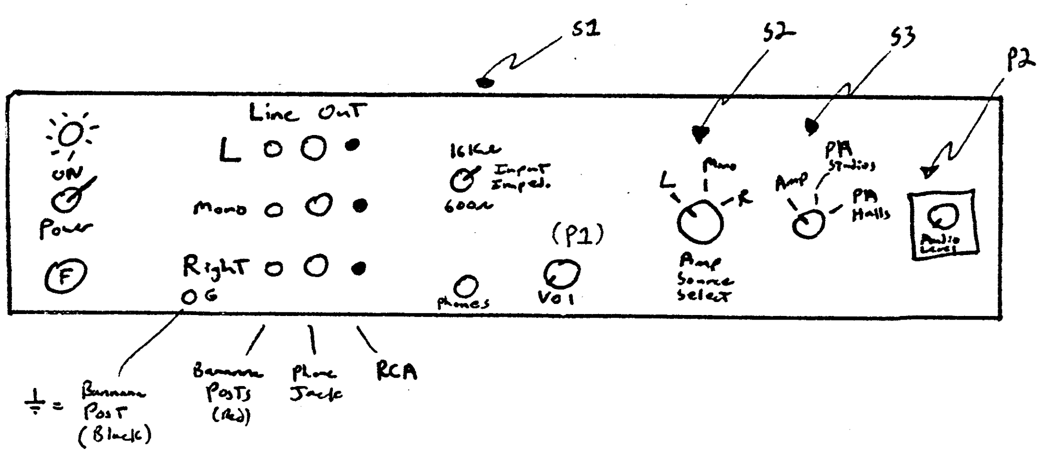
WMFO Utility Amplifier (Sheet #2 of 2)



Power Supply



Front Panel Layout

THE WMFO JACKS PANELS

INPUTS and/or OUTPUTS:

* 2 Pairs of 1/4" stereo phone jacks for balanced inputs/outputs.

* 2 Pairs of RCA jacks for single-ended inputs/outputs.

* 2 Stereo patch points in patchfield

THE JACKS PANELS

I) Function and Operation:

The Jacks Panels provide a means of connecting two stereo sources (balanced or single-ended) into the relevant patchfields. A jacks panel is located in each studio (mounted on the side of the cabinets). A jacks panel is also mounted in the blue inter-studio rack in Master Control. Each jacks panel contains two pairs of 1/4" stereo jacks in order to allow external balanced signals to be routed directly to and from the patchfield (and patched anywhere). The tip of each jack is the "+" balanced connection, and the ring is the "-". The shields are all connected to the common studio (or Master Control) ground.

Single-ended signals may be introduced or extracted from the RCA jacks mounted below the corresponding 1/4" jacks. These single-ended jacks may only be used when nothing is plugged into their respective 1/4" jack, since an engaged 1/4" jack will potentially float the "-" balanced connection from ground. One must remember that most of the devices which come up on the patchfield expect all outputs and inputs to be 600-Ohm balanced. Patching a single-ended signal through the jacks panel (which grounds the "-" connection) may potentially degrade the audio quality and introduce interference & crosstalk. Single-ended signals should normally be input and output via the Interface circuit (see associated writeup), which properly buffers and isolates them before they reach the patchfield. If this isn't logistically possible (ie. cables too short), any single-ended signals to be introduced to a 600-Ohm device (ie. the board) should be first routed through the Interface Mix input on the patchfield, since it has an input impedance of 16K-Ohms, which should not provide an excessive load for most equipment. Of course, different devices vary in their input/output capacities, thus certain devices may work beautifully when patched into the 600-Ohm lines, while others may not. Experiment a bit....

The Two stereo Jacks lines tied to each panel are labeled "J1" and "J2". Each has a Left and Right jack. Studio B has four additional jacks on its panel, grouped into two pairs "E1" and "E2" (each has L & R components). These are for the Biamp mixer "channel patches", and are also stereo jacks (the tip is "send" and ring is "receive" according to the Biamp manual). Corresponding "send" and "receive" points are available on the patchfield; this enables processing and effects to be added independently for each mixer channel. These jacks can also be used as auxiliary connectors for additional access to the patchfield (remember that these jacks are not balanced, but carry two single-ended signals!), and with a bit of improvisation, they certainly can also be used with any other mixer systems.

II) Circuit Description:

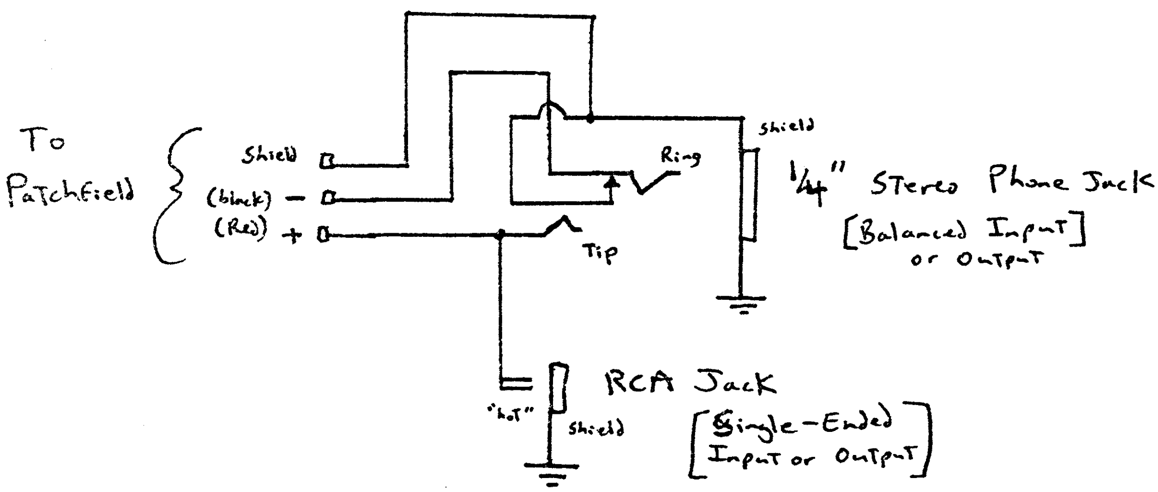
There is no circuitry associated with the jacks panels. The single-page schematic denotes how the switch contacts on the 1/4" jacks are used to ground the "-" input when the jack is not engaged (so the single-ended RCA jacks can be used). A diagram of the layout of the standard Radio Shack 1/4" jack used in the jacks panels is given at the bottom of the schematic.

III) Troubleshooting Hints:

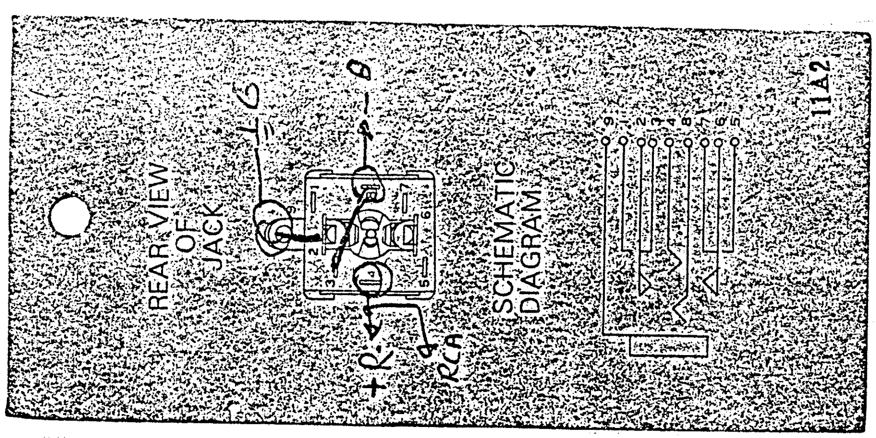
If anything is flaky, first check that you are not using bad patchcords. Otherwise, check the connections between the jacks panel cables and the patchfield Xmas Trees; in some of the studios, these cables were soldered by monkeys.

WMFO Jacks Panels

— Feb., 1987
— J. Paradiso



Schematic for Single Channel of Jacks connection



Phone Jack Layout and Connections

THE WMFO EXTERNAL FM BUFFER/DRIVER/FILTER

INPUTS:

* External Monitor In
  (Single-ended L & R)

OUTPUTS:

* External Monitor Out
  (Balanced L & R)

THE EXTERNAL FM BUFFER/DRIVER/FILTER

I) Function and Operation:

This circuit is located in a black plastic box mounted in the grey monitor rack in Master Control. It takes the left and right single-ended outputs from the Belar Stereo Monitor, and converts them to balanced outputs (with optional filtering), which are nominally input to the External Monitor Distribution Amplifier via the Master Control Patchfield. An 8 KHz notch filter is introduced into the signal path by flipping a toggle switch mounted on the box. This filter can suppress some static & noise, and all-in-all make the external monitors sound a bit "sweeter" and less harsh. It was built when the noise content in the stereo Belar signal was fairly high, before the attic Yagi antenna and FM cavity filter (also mounted in the grey monitor rack) was installed. The noise content is now substaintally reduced, thus the action of this filter is a bit subjective, and the associated toggle switch may be set arbitrarily (ie. whichever setting sounds better). Note that a spare tuner is also mounted in the monitor rack; it also has a "black box" associated with it (which is only a line amplifier without filter). The spare tuner outputs terminate at the Master Control patchfield, and must be patched manually to any desired destination. The mono output of the Belar FM monitor supplies the PA system (this is connected directly, and is not routed through any filters, line amplifiers, patchfields, etc.). The left and right outputs of the Belar Stereo monitor (through the Buffer/Driver/Filter) drive the External monitor DA, and hence supply the monitor signal to all studios, etc.

II) Circuit Description:

All circuitry is listed in the single page schematic. The single-ended input signal is first routed through band-pass/notch filters built around IC1. These filters are designed to boost or cut the frequency components around 8 KHz. Trimmers T1 and T3 select the amount of boost or cut (one extreme is full boost, center is flat, and the opposite extreme is full cut). Balanced Line drivers are built around IC2 and IC3. The inputs to these line drivers is selected by S1 to be either the filtered outputs of IC1 or the "Raw" input (S1 is mounted on the plastic box for easy external access). Trimmers T2 and T4 allow the gain of each channel to be independently adjusted.

III) Calibration and Adjustment:

The filter settings are a bit arbitrary. To eliminate noise, remove the program material feeding the transmitter, and leave a blank carrier. Adjust T1 & T3 to attenuate residual static and noise (make sure S1 is set to include the filters). Don't adjust these trimmers to attenuate the signal too far (or audio will sound too "dull") and don't set them to give a boost; usually a small bit of notch response is optimal. Put the program signal back on the air, and make sure the sound isn't too flat; if it is, back off slightly on the notch ajdustment. Try and set both left and right channels to yield an identical response.

The gains of each channel are adjusted by trimmers T2 and T4. Send a mono house tone to the transmitter, reading at zero dB on the on-air board. Adjust T2 and T4 to yield identical levels (it may be a good idea to put the studio switcher in "mono" while making this adjustment). Set these gains so that the external monitor amplitude roughly matches the actual house-tone level (ie. it should drive the meters on any board to zero dB at half-pot).

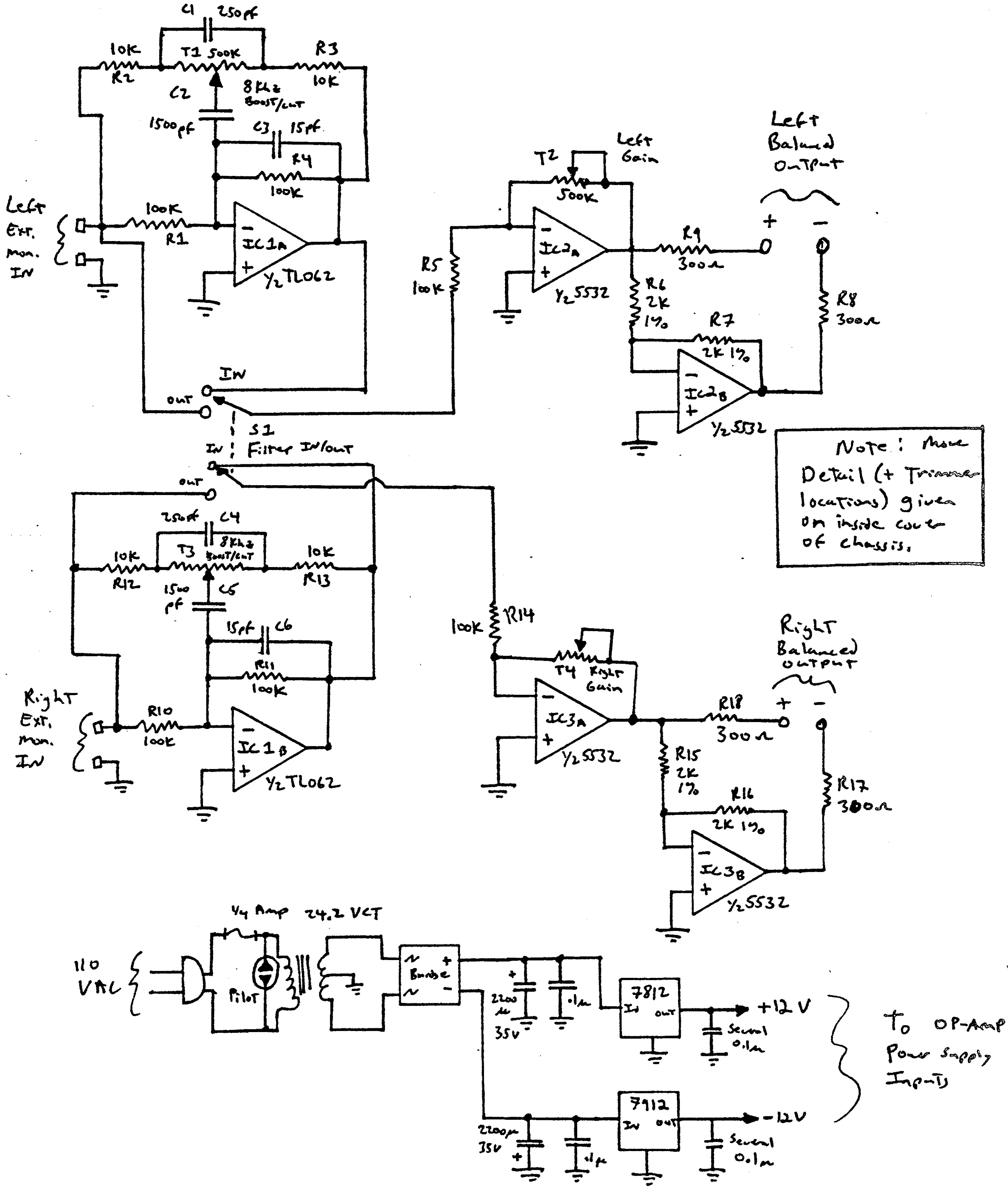
The locations of all trimpots and ICs are listed inside the top cover of the circuit housing. Be sure not to adjust trimmers T1 or T3 in lieu of T2 or T4.

IV) Troubleshooting Hints:

This circuit is very simple. It has never failed. If it ever does, it should be very easy to fix. Check power supply voltages first. All ICs are socketed.

-J. Paradiso

WMFO External FM Buffer/Driver/Filter - Jan., 1987



Note: More Detail (+ Trimmer locations) given on inside cover of chassis.

THE WMFO STUDIO SWITCHER

INPUTS:

* Studio A 600-Ohm Balanced audio in (L & R)

* Studio B 600-Ohm Balanced audio in (L & R)

* Studio C 600-Ohm Balanced audio in (L & R)

* EBS Tone 600-Ohm Balanced audio in (mono)

* EBS Alarm Nrm. ON contacts

* Over Modulation warning signal in

* Studio A,B, & C Mic. Live Nrm. ON relay contacts

* Studio A,B, & C armed/air pushbutton contacts

OUTPUTS:

* Main Balanced audio mix out (L & R) (50-Ohms)

* Aux. Balanced audio mix out (L & R) (600-Ohms)

* Pull-Down output for activation of doorbell bulbs

* EBS Warning LEDs (Studios A,B, & C)

* Over-Mod LEDs (Studios A,B, & C)

* Mic. Live LEDs (Studios A,B, & C)

* Armed/Air LEDs (Studios A,B, & C)

THE WMFO STUDIO SWITCHER

I) Function & Operation:

The WMFO Studio Switcher is mounted in the blue interstudio rack located in Master Control. It performs several services related to studio airchain coordination. It's primary function is to route the selected studio (or studios) to the airchain. It has been designed to allow studios to be placed on or off-air quickly and simply (however a basic security procedure has been instituted to prevent studios from being put on-air accidently without authorization from a studio already having air status). The Studio Switcher drives all LED indicators in the small remote panels mounted atop the console in each studio. A sketch of the remote panel layout is given in page 8 of the schematic. There are three bi-color LEDs (labeled "Armed/Air") mounted below the row of pushbuttons. These reflect the airchain status of each studio. If a LED is glowing red, the corresponding studio is connected to the airchain, and any program output will be routed to the transmitter (unless patches have been thrown to override defaults inside the studio or Master Control). If a LED is glowing green, it is in "armed" status. In this case, the studio's output is not routed to the airchain, although one may bring the studio's status to "Air" (ie. red LED) by depressing the studio's button (only in the studio that is armed). If the LED is unilluminated, the studio is unarmed and off-air; the studio must generally be armed by another studio already having air status in order to be connected to the airchain. Any combination of studios may possess "air" status simultaneously. More detail on the control procedure is given below.

The remote panels contain a row of three yellow "mic. live" LEDs mounted below the "armed/air" LEDs. These LEDs are illuminated whenever a microphone in the corresponding studio goes live (and speakers are muted). They may be consulted to determine microphone and muting status in other studios while coordinating inter-studio events, attempting to page someone working in another studio, etc.

Two large "superbrite" LEDs are mounted on the front of the remote panels. The green LED flashes at a high, annoying rate in all studios whenever an Emergency Broadcast System (EBS) message is received. This is accompanied by a loud audio feed from the designated EBS station (currently WROR) blasting over every PA speaker in the station (see PA Mixer/Intercom writeup for details). Since the PA feed in the studio is muted when the studio mic. is live, these flashing green lights were added to attract the jock's attention.

The yellow "superbrite" LED flashes (only in studios having "air" status) when the Belar FM monitor detects an overmodulation condition, thereby alerting on-air jocks. It remains unilluminated in studios not connected to the airchain, where it would be merely irrelevant and annoying.

The Studio Switcher also provides a feed to the phone/doorbell unit which causes the bulbs inside the doorbell pushbuttons to be illuminated whenever a microphone goes live in a studio having air status. This is to alert people waiting at the doors that the jock (who may be the only person in the station) is talking on-air, and can't answer the door.

The row of buttons mounted on the studio switcher remote panels enable control and modification of the airchain status. Each pushbutton corresponds to a particular studio. The cover of the button pertaining to the location of the remote panel is colored red; the other two are black (ie. the "A" button in Studio A is red, the "B" button in Studio B is red etc.).

These buttons must be used in accordance with an elementary security protocol in order to place studios on and off air. Their function may be most easily explained via a set of examples. Suppose the only studio with air status is Studio A (ie. the "armed/air" LED for Studio A is red; all others are dark). In order to allow Studio B to join the airchain, someone in a studio with air status (presently only Studio A) must press the button on his/her remote panel that corresponds to Studio B (the middle button). The Studio B button will now glow green, meaning that Studio B is "armed". If the person in Studio A decides that he doesn't actually want Studio B to eventually join the airchain afterall, he can press Studio B's button on his panel again, which will turn the Studio B LED off, and remove it from armed status. The Studio B armed status can thus be "flipped" from Studio A by repeatedly depressing Studio B's button. Assuming, however, that the folks in Studio A actually want Studio B to go on-air, they will only press Studio B's button once in order to drive it to armed status. If someone in Studio B then presses their button (which is colored red), they will promptly attain air status (ie. their LED will glow red, and Studio B will be connected into the airchain).

There are now two red LEDs (hence two studios on the air chain) corresponding to Studios A & B. Studio C's LED is still dark; whenever a studio has no air/armed status (ie. the LED is dark), one may press any pushbuttons on the studio's remote panel without attaining an effect. Studios may not change their status (or the status of any other studio) without first being armed from a studio with "air" status. Studio C (ir this scenerio) may, of course, now be armed by either Studio A or Studio B; if this is done (and the person in Studio C depresses their button after being armed), all three studios may be placed into the airchain simultaneously.

There is one exception to the security function, however. If all studios have no status (ie. all three LEDs are dark), the first studio to push its own button will be able to arm itself and subsequently go to air. This feature was implimented to allow rapid recovery from power-failure and lightning glitches which may knock all studios out of the airchain.

So far, I have not discussed the means of removing a studio from air status. This is relatively straightforward. Any studio having air status (ie. red LED) may take any other studio (including itself) off air by pressing the button corresponding to the studio desired to be removed. There is a 2-second delay placed into this operation, however, to prevent studios from accidently leaving the airchain due to unintentional hits on the remote panel. Whenever an "air" studio depresses the button of another "air" studio, that studio's LED will start flashing. One must hold this button down until the LED stops flashing (approx. 2 secs.) to take the studio off-air.

When taking any other studio remotely off air, the studio's status will drop to "armed"; one more depress of the button will remove the "armed" status and disable the studio. When taking your own studio off air, the studio status will be entirely dropped; ie. the studio goes from air (red LED) directly to "dead" (dark LED). This will not allow you to put your studio back into the airchain without authorization from another on-air studio, so before putting yourself off-air, make sure that it's really what you want to do (or make sure you have friends in another studio with air status).

With regard to our above scenerio, the people in Studio A can remove their studio from the airchain (hopefully after placing Studio B on the air) by pressing their own button and holding it down for a few seconds.

The above discussion summarized the function of the studio switcher's remote panels; the airchain configuration can be dynamically and easily modified by following these procedures. Applications and methods of studio switching may certainly be practiced and developed for maintaining "smooth" on-air continuity. The techinque of switching studios in the midst of a cut (as opposed to a segue or break in programming) deserves to be outlined for posterity; it has proved quite useful, particularly when there is only one person in the studios who must manage the switch in the middle of his show.

If one decides to switch from Studio A to Studio C while a cut is playing (for example), one first puts Studio C into the airchain with Studio A (per the procedure discussed above; both LEDs should be red). Studio A's program feed can be routed to a fader in Studio C via the Studio C patchfield. One then slowly brings the Studio A feed up on program in Studio C by gradually incrementing the corresponding fader; bring it up until it reads at a good median level. Studio A is now feeding the airchain both directly and through Studio C. Studio A may now be put off air from Studio C (hold down the Studio A button on the remote panel). The small "glitch" in volume created when Studio A goes off is compensated fairly well by the airchain compressor; in most cases, one has no chance of noticing the switch. When the record in the now unattended Studio A is finished, one can pot it down in Studio C, and continue with the show from the new location.

The front panel of the main Studio Switcher unit in Master Control is depicted in page 6 of the schematic. A row of three "mode" switches (S1 - S3) may override and "hardwire" the airchain status. When one of these switches is in the "up" position, the corresponding studio is held in "air" status, and can not be taken out of the airchain by the remote panel switches. When one of these switches is flipped down, the studio is removed from air (ie. the LED goes dead), and can not be placed into the airchain via the remote switches. When this switch is in the center (auto) position, the airchain switching is entirely governed by the remote panels, as discussed at length above. Needless to say, these switches should stay in the center position (allowing full remote functioning) unless some special circumstances warrant airchain control directly from Master Control.

The switcher front panel also contains a red "reset" pushbutton (S5). This is a "panic" button which removes all studios from the airchain when it is pressed. Duplicate airchain status and mic. live LEDs are also mounted on the switcher panel. LEDs are also present to indicate EBS and Over Modulation states. A panel LED is illuminated whenever the doorbell bulbs are commanded on.

A switch (S1) is also provided on the switcher panel to allow the airchain feed to be bridged into mono. The normal position of this switch is down (ie. stereo). If it is flipped up, a warning LED will be illuminated, and the station airchain output will be monaural. No change in audio level occurs when this switch is thrown. It has been installed in the studio switcher in order to facilitate calibration of the airchain feeds, Optimod unit @ the transmitter site, and Belar Stereo monitor. It may also be used to put the station into mono in the event of phone line trouble on a single channel (the stereo pilot may also be dropped via the transmitter remote control unit in the Studio A rack). Needless to say, this switch should stay in the "stereo" position during normal operation.

Two LEDs labeled "peak" are also present on the switcher front panel. These LEDs are illuminated by large audio swings (they don't really correspond to saturation of the Studio Switcher [there should be lots more headroom], but serve as an indication of generally "hot" levels). These LEDs are driven by a timer, and stay illuminated for a second or so after being triggered by loud levels.

The studio switcher inputs all come through the patchfield. They may be dynamically re-defined by throwing the appropriate patches (ie. a studio may be replaced in the airchain by a remote line, etc.). The EBS tone signal is also routed through the patchfield; this is a special multi-tone sound which should be transmitted during designated EBS tests (which aren't done at the moment), and as otherwise authorized by the EBS pooh-bahs. This feed into the studio switcher and airchain is always active; it is not switched in or out (it is also mono, in case it is overridden by another patch). There is no signal on this line unless the EBS tones are triggered by the EBS generator. There is a remote keyswitch to do this in the Studio A board; by inserting the key and turning, EBS tones will immediately be transmitted, regardless of the air status of any studio.

Of course, the studio switcher output is only routed to the airchain if it is normalled or patched appropriately in Master Control. The default normals currently installed route the main switcher output through the airchain compressor (a Compellor unit), and from there directly to the transmitter phone lines. The normal from Compressor output to telephone lines is currently a bit flaky, so a patch cord remains in semi-perminant residence across this connection. The studio switcher also provides a "spare" 600-Ohm output which comes up on the Master Control patchfield. This output may be routed or monitored as desired, yielding a simultaneous tap on the composite airchain signal.

II) Circuit Description:

This circuit is perhaps the most complicated ever built at WMFO, as seen in the 9 pages associated with the schematic diagram. I got the idea for this circuitry nearly three years ago (the basic functional design was doodled on a napkin while I was waiting in a restaurant for a pizza). I was quite enthusiastic about constructing it; in fact the idea of eventually installing my hi-tech studio switcher is one of the major impulses that inspired me to continue pushing to complete the WMFO rennovation. Ah, we're all occasionally slaves to abstract concepts, aren't we? Perhaps someday I finally learn my lesson, and stop getting such wild ideas.

The circuitry is broken into two sections, one "audio" card (which contains all analog and audio electronics), and one "control" card (which is primarily composed of CMOS digital electronics). Both cards are mounted on the rear of the rack panel; the audio card is mounted vertically, while the control card is horizontal. The only connections between the two cards are sent on a 7-conductor cable which contains the three AIR signals (one for each studio, high [+5 Volts] on air status) and their complements, along with a +5 Volt power supply line. Both circuit cards use a common unfiltered power supply mounted behind the rack panel.

All circuitry on the audio card is depicted in the first page of the schematic. At left, one sees the 7 differential line receivers that convert the balanced 600-Ohm input signals to single-ended format. Three dual 5532 OP-AMPs (IC1-IC3) buffer the three studio inputs (one IC dedicated to each studio), and the 5534 (IC4) buffers the EBS tone input. Superior audio quality may be obtained if the coupling capacitors (C1-C14) are removed (they aren't really needed). These are 1 mfd. mylar non-polar capacitors, which aren't the best choice, and may introduce a bit of "capacitor distortion" due to hysteresis effects, etc. The quality of the studio switcher audio chain is probably good enough, but if one is upgrading, these capacitors could either be bypassed with a more suitable grade of capacitor, or removed.

The three studio lines are switched by a bank of three LF13202 JFET analog switches (I've included data sheets on these devices at the conclusion of this report). One LF13202 chip is dedicated to each studio (each chip switches both left and right channels). Each LF13202 package is wired as a DPDT switch, which routes the inputs of summers IC8 either to the studio output (when the studio has "air" status, and the appropriate AIR line is high and its compliment "AIR-bar" is low), or to ground (when the studio does not have "air" status, and the states of the AIR lines are reversed). The clamp to ground was incorporated to avoid crosstalk problems due to summing nodes left floating in the vicinity of live audio signals.

The two summing amplifiers (IC8) equally mix the switched outputs from each studio into two channels. The EBS tone signal is also mixed (unswitched) in IC8; the amplitude of the EBS tone in the mix is adjustable via trimmer T1. Trimmers T2 & T3 allow the left and right master mix gains to be specified. The outputs of IC8 are mixed in IC9 to form a mono feed (the input resistors are calibrated to give a unity mono gain). Switch S1 (mounted on the front panel) allows one to select either the original stereo mix (directly out of IC8) or this mono feed from IC9 to be routed to the output buffer circuitry. IC10 and IC11 create a differential signal from each channel to form balanced audio lines (half of IC10 and IC11 act as unity gain inverters, the other half are non-inverting voltage followers). IC10 and IC11 drive four complimentary transistor buffers (Q1-Q8), which allow balanced lines of lower impedance to be driven. Diodes D1-D8 bias these transistors to avoid crossover distortion. Two outputs are tapped from the emitters of the output buffers; A set of outputs isolated by 25 Ohm resistors provides a low impedance drive (main output), while another set of outputs isolated by 300 Ohm resistors yields conventional 600-Ohm balanced pairs (the auxilary output).

The "+" end of each channel is also input to comparators in IC12. These comparators detect signal levels crossing thresholds adjusted via trimmers T4 & T5. When a threshold is crossed, the appropriate monostable timer in IC13 is activated, and the corresponding LED lights for a second or two to indicate the "peak" condition. Zeners D9 & D10 clamp the +/-15 Volt swing of IC12 to a 0-5 Volt excursion that is within the supply limits of IC13.

The next 4 pages of the schematic depict circuitry living on the control card. All ICs are labeled with their location on the control card (letter column and numeric row; see layout on page #7). Pin numbers are listed for all IC connections. A cross-reference which converts alphanumeric IC designation into an actual device type is given on page #9. All logic circuitry is driven by a 12 Volt regulated supply, except where explicitly noted.

Page #2 shows an assemblage of miscelaneous logic functions which are provided by the switcher. The dual clock (C1) creates clocks used to flash the EBS warning LEDs and blink the armed/air LEDs before a studio is removed from air status (ie. "warning flash clock" here). The EBS alert signal (buffered in A2) is gated into the EBS clock such that the EBS LEDs flash only when an EBS alert has been received. The EBS flash rate is adjustable (over a small margin) by trimmer T2. The Off-Air flash rate (and total "safety provision" Off-Air deadtime) is adjustable via trimmer T1. The "Over Modulation" flasher voltage from the Belar FM monitor is buffered in A2 and gated by the AIR signals from each of the studios before being applied to their remote "OV MOD" LEDs (note that the subscript "L" on the AIR lines refers to a "local" set of signals which go to logic 1 (12 Volts here) when a studio has air status).

The "Mic. Live" lines from each studio are buffered by A3 and routed to the remote panels in each studio to illuminate the "mic. live" yellow LEDs. The "mic. live" status from each studio is AND'ed with the AIR status of the studio in A4; these outputs are all OR'ed in D2 before being routed to the phone-doorbell unit (via Q9) in order to activate the doorbell bulbs whenever at least one studio with air status has a live mic.

The three AIR status lines are OR'ed together to form a composite signal which goes high whenever at least one studio is in the airchain; the "All Air" signal is used in the switching logic, as discussed below.

The next three pages of the schematic depict the switching logic used for each studio. The circuitry is identical in each schematic; the only differences are the ICs and pins dedicated to the particular studio, as marked on the diagrams. This discussion will analyze the Studio A circutry for simplicity; all other studios follow identical concepts.

The pushbutton contacts dealing with Studio A from all remote panels are input at left, and conditioned/buffered by Schmidt triggers (A1/A2) and filter capacitors (C1-C3). The Studio A pushbuttons located in the other two studios are first gated with their AIR signals, in order that they can have no effect unless their studio is connected to the airchain. These signals are all OR'ed in D2 (along with the A pushbutton in Studio A gated with the "All Air" signal; this allows the studio to arm itself if no studios have air status). The output of D2 clocks a T-type latch (D6) which reverses the "armed" status of the studios. The output of the "arm" latch D2 gates the studio's pushbutton in D3; this allows the next push of the studio's pushbutton (after the studio has been armed) to clock latch D6, which brings the "air" status of the studio high (the output of the air latch [D6 here] provides the local studio AIR line referred to in the logic diagrams). The air latch output also resets the Arm latch (D6) through C3. If the studio has air status (ie. "Q" of D6 is high), depressing the "A" pushbutton in any studio having air status (including Studio A; these are OR'ed in C3 and gated with A's AIR line in D4) will allow counter D5 to be clocked. The divide-by-2 output of this counter causes the Armed/Air LED of Studio A to flash (via gate C5); this provides the off-air warning signal. When the counter has cycled through 64 clock input pulses (32 LED flashes), pin 4 of D5 will go high, which (via OR gates in C2 & C3) will reset the AIR latch D6, thereby removing Studio A from air status. Counter D5 (which is clocked from the "Warning Flash Clock" derived from C1 as discussed earlier) thus provides the "deadtime" in taking a studio off-air; if one releases the pushbutton while the LED is flashing, counter D5 will reset, and subsequent attempts (occuring when the button is again held down) will still require 64 clock pulses before the air latch is reset.

The armed and air latches can also be instantly reset by the Studio A mode switch on the main panel (S2) or the Master Reset Button (also mounted on the main panel). Switch S2 (in its up position) can also directly set the air latch, thereby putting the studio immediately into the air chain. The "AIR" and "AIR-bar" signals are derived from the Q and Q-bar outputs of Air latch D6; buffer F1 acts as a level-shifter to convert the 12-Volt output of D6 to a 5-Volt swing that is digestable by the LF13202's on the audio card.

The armed/air LEDs for each studio are driven by LM13080 power OP-AMPs which take the difference in voltage between the outputs of armed and air latches. When the armed latch is high, the air latch is low, and the LM13080 is driven negative, which illuminates all LEDs pertaining to the corresponding studio in green. When the air latch goes high, the armed latch is reset, and the LM13080 swings positive, illuminating all LEDs for this studio in red. If the studio is "dead" (ie. neither armed or air), both inputs to the LM13080 are at ground, and the output stays at zero, thereby keeping all LEDs associated with the studio darkened.

The power supply is depicted on the sixth page of the schematic. +/- 5 Volts (for the audio card and LM13080's), +/- 15 Volts (for the audio card), and +12 Volts (for the logic on the control card) are provided by 3-terminal 78/79 type regulators. The +/-15 Volt regulators are mounted on the audio card; all others (I think...) are located on the control card.

The remote panels consist of 8 LEDs and 3 switches which have one end tied to ground, and the opposite end tied to a cable which feeds to the studio switcher in Master Control. See the diagram on page 8 of the schematic.

III) Calibration and Adjustment:

Even though the circuitry is fairly complex, there's not too much to tune in the studio switcher. Locations of all trimmers are alluded to in the card layouts listed on page 7 of the schematic.

First, I'll detail calibration of the audio card. Place only one of the studios in the airchain (let's say Studio A). Pot up house-tone @ zero dB in Studio A. Patch the output of Studio A's distribution amplifier in Master Control (which should be normalled to the studio switcher input) into a metering device (the EQ in Master control may suffice; otherwise use a board in another studio). Adjust the gain on the EQ (or board) to yield a zero dB meter deflection. Now patch the studio switcher main output directly to the calibrated meter (ie. board or EQ; pull the patch from the Studio A DA output, and put it into the switcher output). Adjust the gain trimpots (T2 & T3 on the audio card) to again yield zero dB meter deflection. The studio switcher gain is now adjusted to be unity (the convention we've established @ WMFO; the only airchain component giving gain is the Compellor). If needed, the switcher is capable of yielding gains of zero through 5 or so; we're presently using the unity gain standard.

Next, remove Studio A from the airchain (ie. flip front panel toggle S2 down), activate the EBS tones (make sure they aren't transmitted for legality's sake), and adjust trimmer T1 on the audio card to again yield a zero dB swing on the calibrated meter (board or EQ).

Finally, the peak LEDs should be adjusted. Put Studio A back into the airchain (via front panel switch S2). Pot up the house tone in Studio A until the LEDs in the board begin to illuminate. Adjust threshold trimmers T4 & T5 on the audio card until the peak LEDs on the studio switcher panel barely begin to light (the one second-or-so minumum flash on these LEDs makes this adjustment a bit hairy). The audio card is now calibrated.

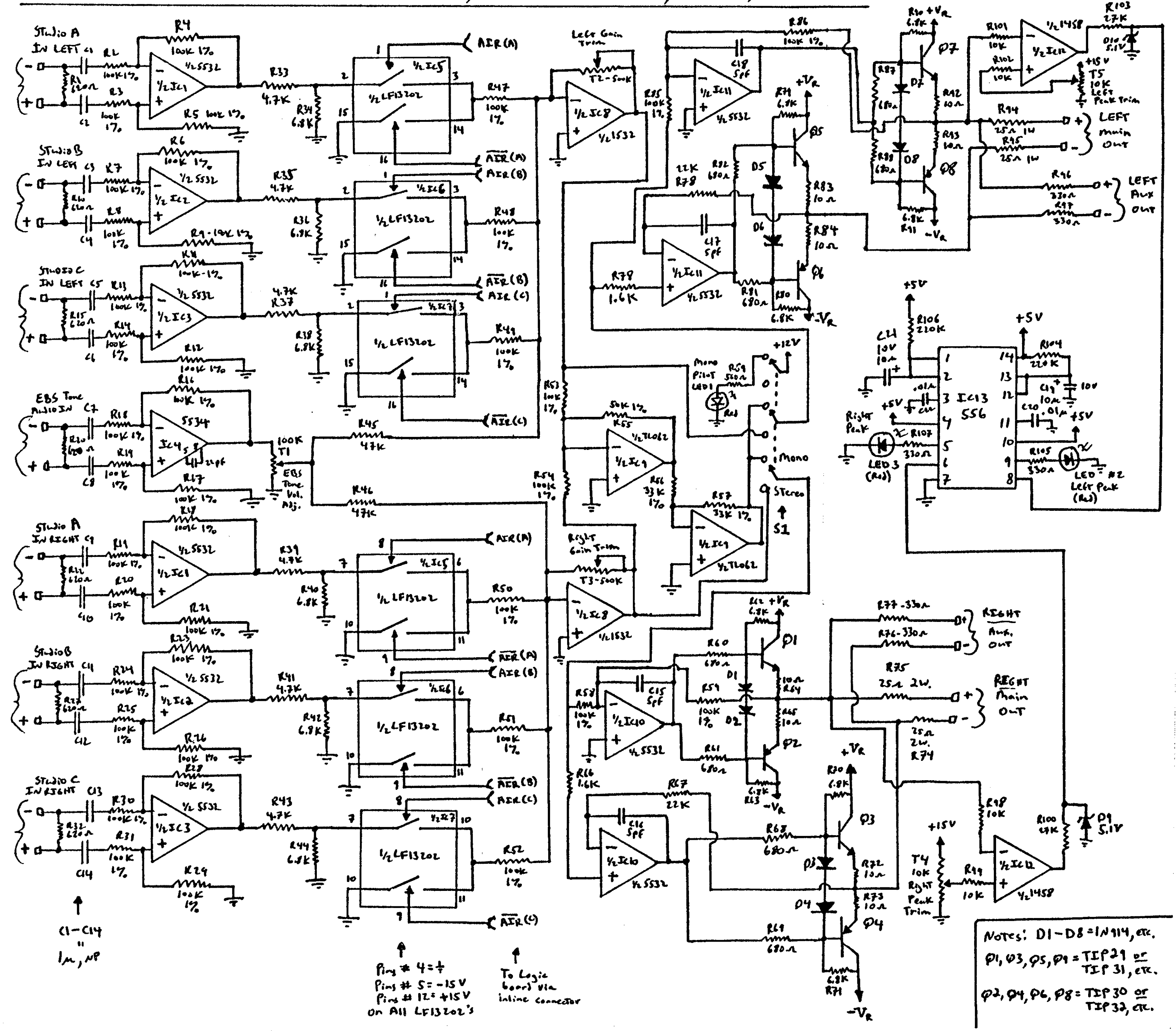
There are only two adjustments to make on the control card. First, trip off the EBS alarm, and adjust trimmer T2 on the control card to yield a nicely irritating flashing rate on the EBS LEDs (if I can recall, there's not too much leeway in this adjustment; decrease R4 and increase C4 if more range is desired). Next, reset the EBS (to avoid all the noise), and set the "Off-Air warning flash rate/duration" by having someone hold down a button to put a studio off-air. While the LED is flashing, adjust trimmer T1 on the control card to yield an acceptable flash rate and reasonable deadtime (I found a 2 second deadtime (or so) to be optimal). The studio switcher is now calibrated; remove all patches, and let her rip.....

IV) Troubleshooting Hints:

The Studio Switcher, in spite of its complexity, has not failed during its first year of use. If it ever does (and the audio circuitry stops working), it may be easily patched around in order to keep the station working (after all, this flexibility is the purpose of the station patchfields).

The audio circuitry may be traced fairly easily. The digital circuitry is a bit of a "rats nest", however all IC pins and connections are given in the schematics, which should make troubleshooting and tracing a bit simpler, and certainly do-able. Before tearing into the circuitry, check the audio and control signals coming into the switcher to make sure that they are all proper (the problem may indeed be a connection, relay, or patchfield in a studio). Check also the power supply voltages. Above all, don't panic, and good luck.....

WMFO Studio Switcher: Sheet #1, Audio Card Circuitry - © Jan, 1987 - J. Paradiso



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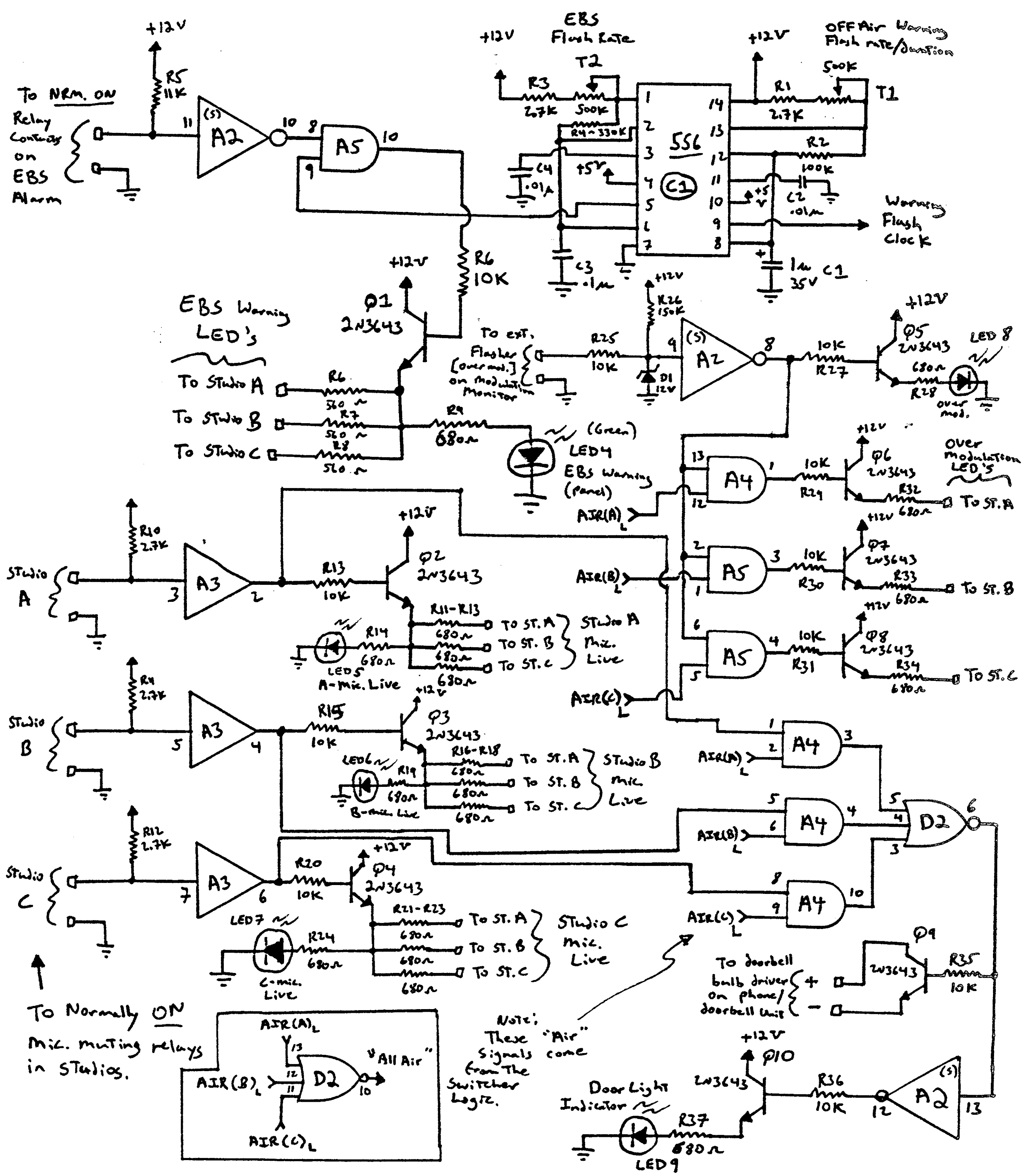
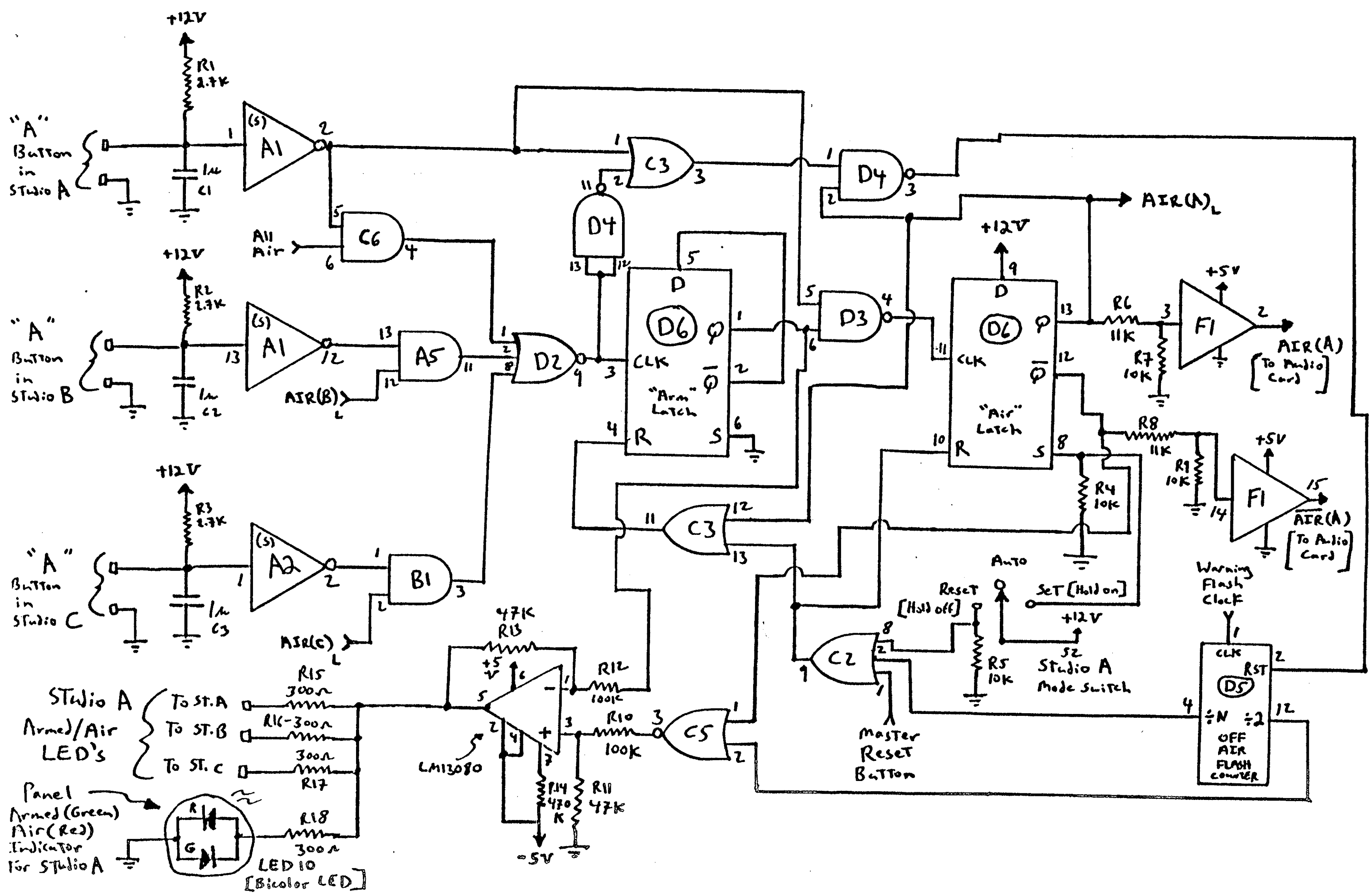
WMFO Studio Switcher : Sheet #2 : Misc. Logic



WMFO Studio Switcher: Switching Logic for Studio A   Sheet #3



WMFO Studio Switcher: Switching Logic for Studio B

Sheet # 4

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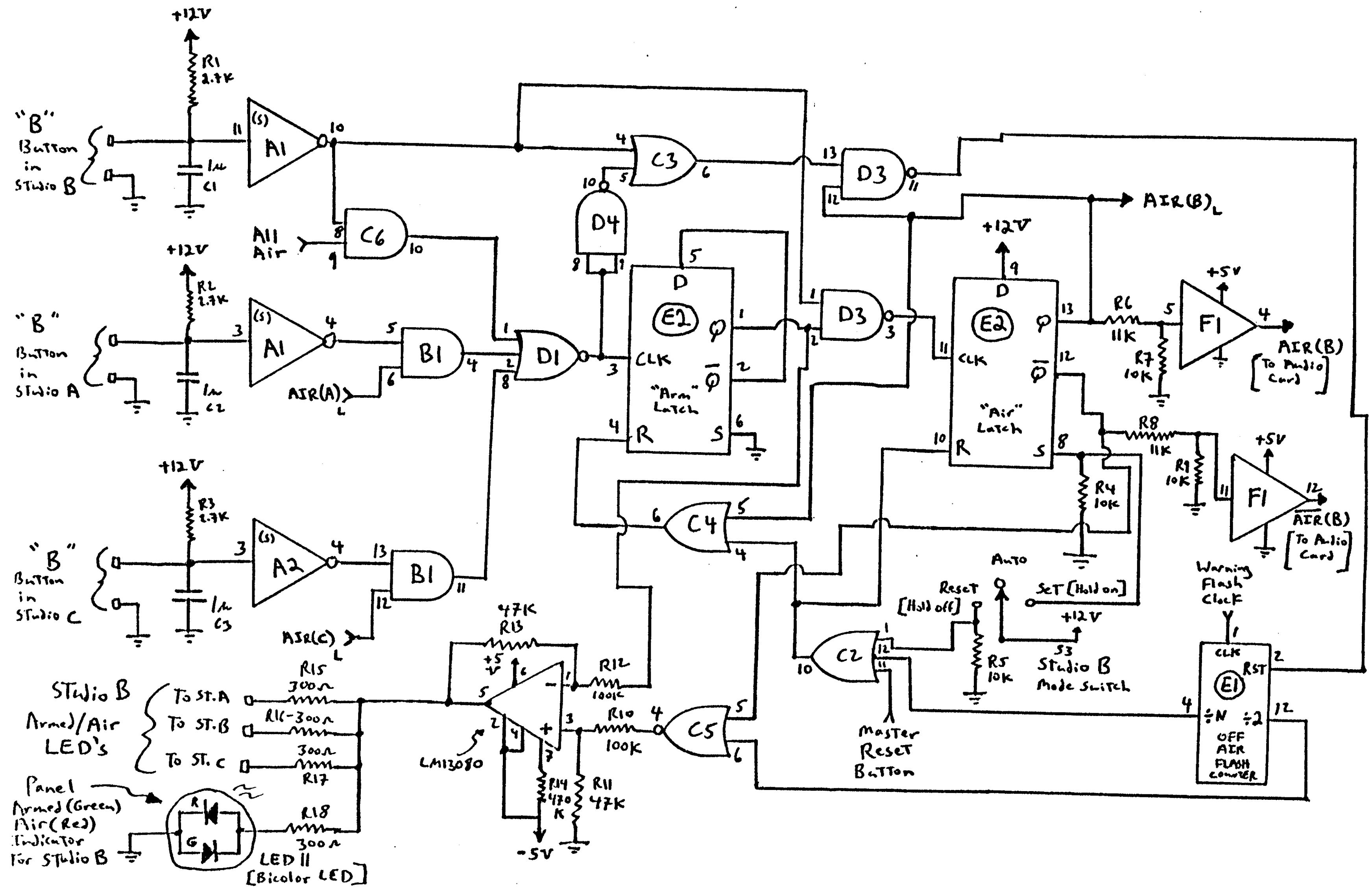
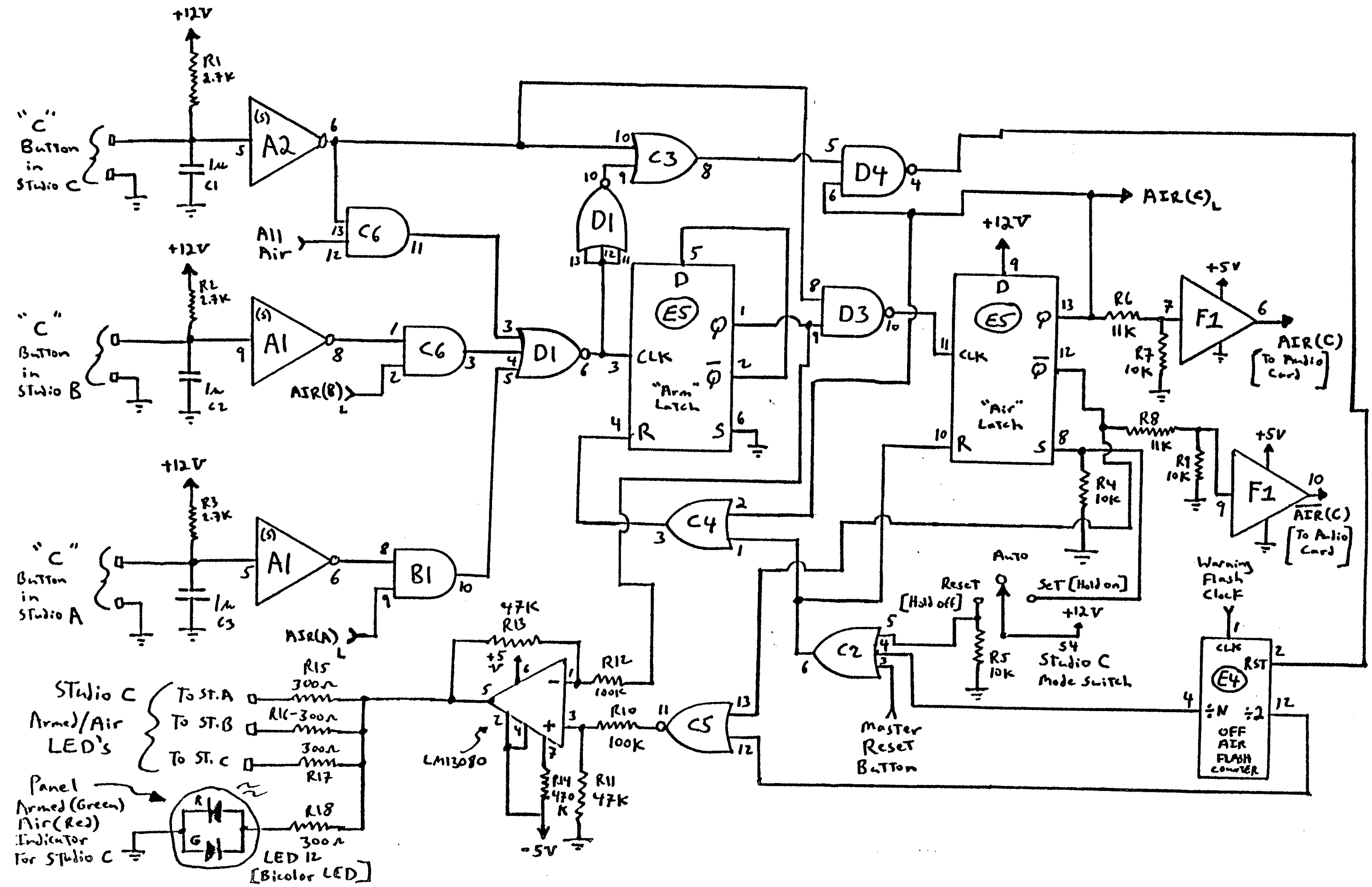


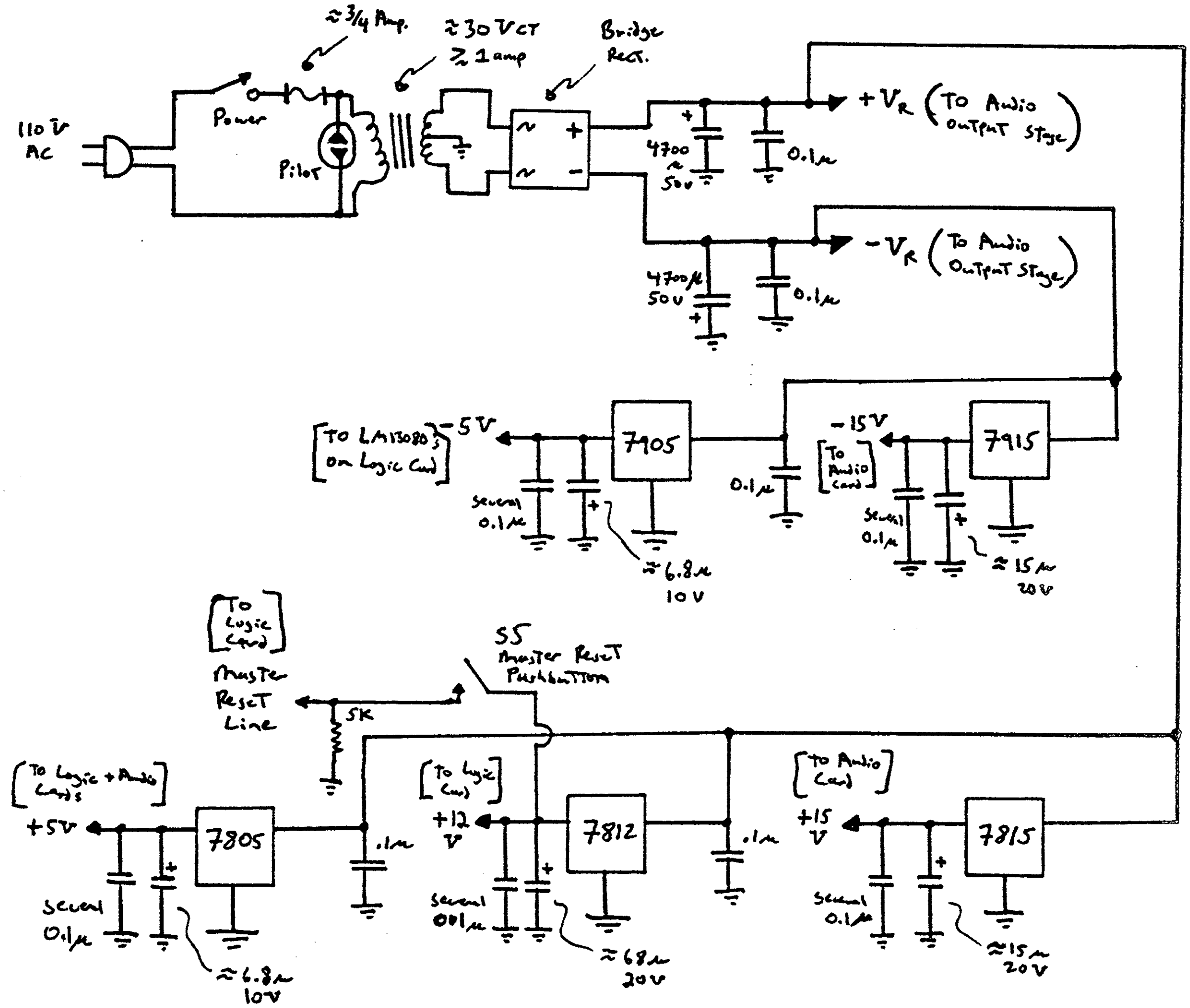
WMFO Studio Switcher: Switching Logic for Studio C

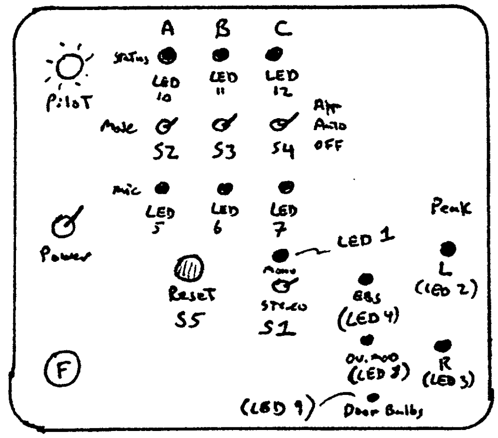
Sheet # 5



WMFO Studio Switcher: Power Supply + Panel Layout

Sheet #6 - © Juni, 1986 - J. Paradiso

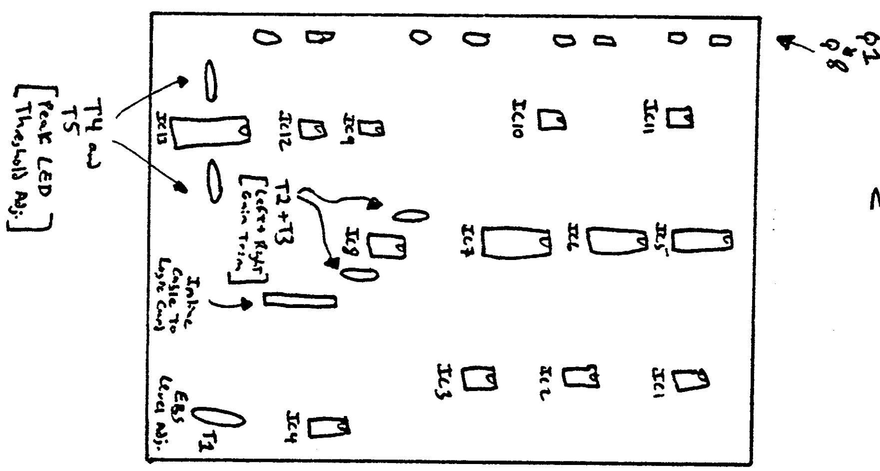




Panel Layout

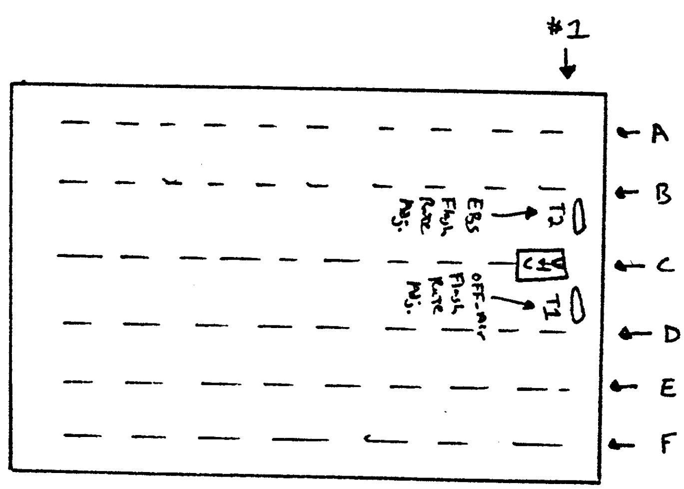
WMFO Studio Switcher: Card Layouts

Sheet # 7 - © Jan., 1986 - J. Paradiso



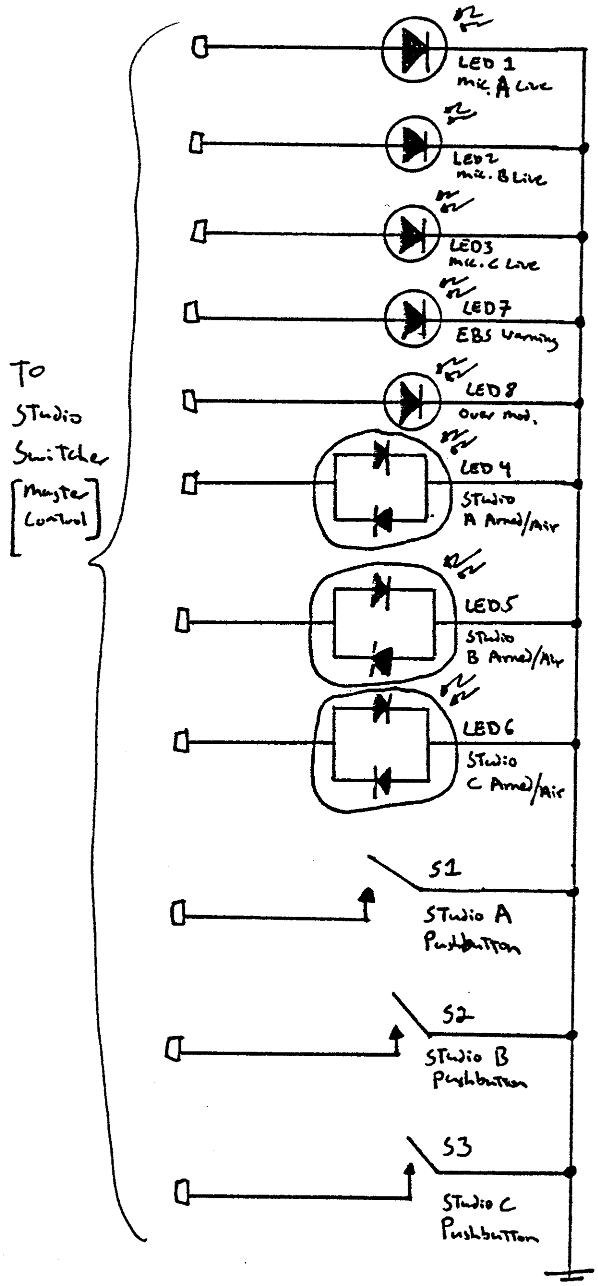
a) Audio Card Layout

Note: Left/Right Channel Assignments and IC's assigned to Studio's A/B/C are uncertain in This drawing [best guesses were Taken for The latter assignments]



b) Logic Card Flash Rate

WMFO Studio Switcher — Remote Control Wiring

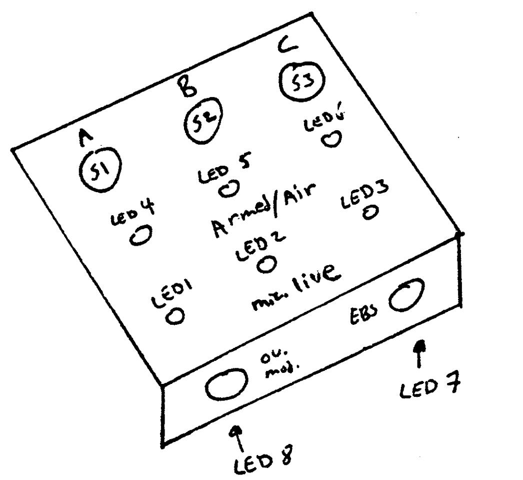


Notes: LED's 1, 2, 3 are yellow

LED's 4, 5, 6 are bi-colored (Red/Green)

LED 7 is Green, LED 8 is yellow (both are "Superbright" LEDs)

The cover of the switch (S1→S3) that corresponds to the particular studio where this circuit is mounted is colored Red. All other switch covers are black.



Remote Panel Layout

WMFO Studio Switcher

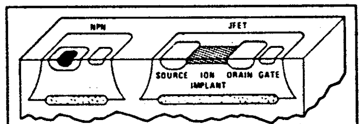
Parts List for Logic Card

A1, A2 = Hex Inverting Schmidt Trigger = 4584
A3, F1 = Hex Buffer/Driver = 4050
A4, A5, B1, C6 = Quad 2-in. AND = 4081
C1 = Dual Timer = 556
C2 = Triple 3-in. OR = 4075
C3, C4 = Quad 2-in. OR = 4071
C5 = Quad 2-in. NOR = 4001
D1, D2 = Triple 3-in. NOR = 4025
D3, D4 = Quad 2-in. NAND = 4011
D5, E1, E4 = 7-stage binary Counter = 4024
D6, E2, E5 = Dual D-Flip-Flop = 4013
[D7, E3, E6] = LM13080 (Power op-Amp)

See Schematics for other parts, and parts on Audio Card.

National Semiconductor

Analog Switches



BI-FET Technology

LF11331, LF11332, LF11333, LF11201, LF11202 Series

6

quad SPST JFET analog switches

LF11331/LF12331/LF13331	4 normally open switches with disable
LF11332/LF12332/LF13332	4 normally closed switches with disable
LF11333/LF12333/LF13333	2 normally closed switches and 2 normally open switches with disable
LF11201/LF12201/LF13201	4 normally closed switches
LF11202/LF12202/LF13202	4 normally open switches

general description

These devices are a monolithic combination of bipolar and JFET technology producing the industry's first one chip quad JFET switch. A unique circuit technique is employed to maintain a constant resistance over the analog voltage range of ±10V. The input is designed to operate from minimum TTL levels, and switch operation also ensures a break-before-make action.

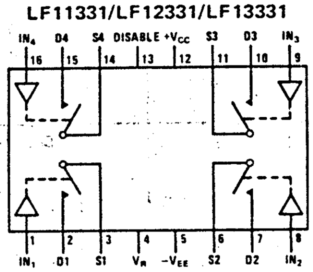
features

- Analog signals are not loaded
- Constant "ON" resistance for signals up to ±10V and 100 kHz
- Pin compatible with CMOS switches with the advantage of blow out free handling
- Small signal analog signals to 50 MHz
- Break-before-make action $t_{OFF} < t_{ON}$
- High open switch isolation at 1.0 MHz −50 dB
- Low leakage in "OFF" state <1.0 nA
- TTL, DTL, RTL compatibility
- Single disable pin opens all switches in package on LF11331, LF11332, LF11333
- LF11201 is pin compatible with DG201

These devices operate from ±15V supplies and swing a ±10V analog signal. The JFET switches are designed for applications where a dc to medium frequency analog signal needs to be controlled.

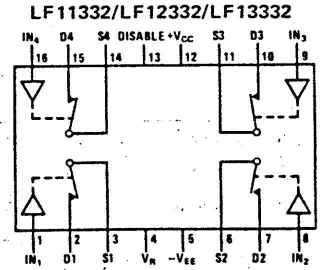
connection diagrams (Dual-In-Line Packages) (All Switches Shown are For Logical "0")

LF11331/LF12331/LF13331



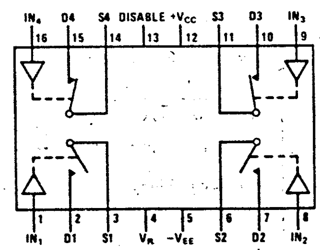
TOP VIEW

LF11332/LF12332/LF13332



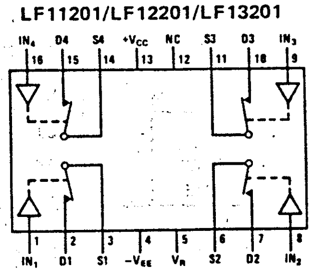
TOP VIEW

LF11333/LF12333/LF13333



TOP VIEW

LF11201/LF12201/LF13201

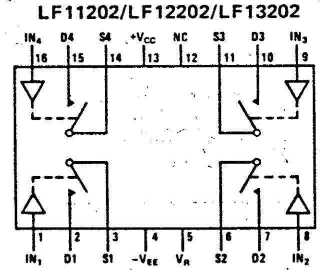


TOP VIEW

Order Number LF11201D, LF12201D, LF13201D, LF11202D, LF12202D, LF13202D, LF11331D, LF12331D, LF13331D, LF11332D, LF12332D, LF13332D, LF11333D, LF12333D or LF13333D
See NS Package D16C

Order Number LF12201N, LF13201N, LF12202N, LF13202N, LF12331N, LF13331N, LF12332N, LF13332N, LF12333N or LF13333N
See NS Package N16A

LF11202/LF12202/LF13202



TOP VIEW

This is The IC used in The Studio switcher

test circuit and schematic diagram

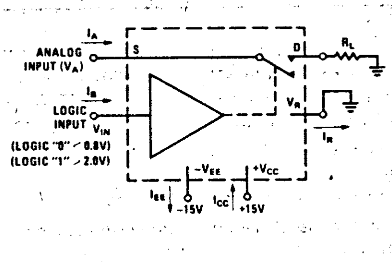


FIGURE 1. Typical Circuit for One Switch

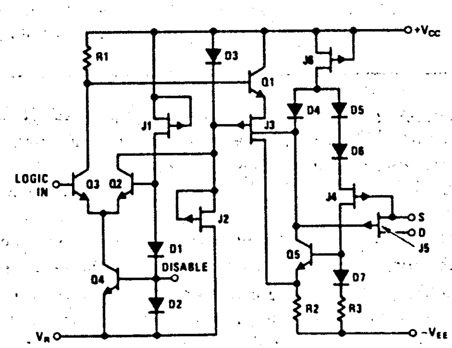


FIGURE 2. Schematic Diagram (Normally Open)

absolute maximum ratings

Positive Supply – Negative Supply (V_{CC}–V_{EE})	36V
Reference Voltage	$V_{EE} \leq V_R \leq V_{CC}$
Logic Input Voltage	V_R –4.0V $\leq V_{IN} \leq V_R$ +6.0V
Analog Voltage	$V_{EE} \leq V_A \leq V_{CC}$ +6V; $V_A \leq V_{EE}$ +36V
Analog Current	$\lvert I_A \rvert$< 20 mA
Power Dissipation (Note 1)	
Molded DIP (N Suffix)	500 mW
Cavity DIP (D Suffix)	900 mW
Operating Temperature Range	
LF11201, 2 and LF11331, 2, 3	–55°C to +125°C
LF12201, 2 and LF12331, 2, 3	–25°C to +85°C
LF13201, 2 and LF13331, 2, 3	0°C to +70°C
Storage Temperature	–65°C to +150°C
Lead Temperature (Soldering, 10 seconds)	300°C

electrical characteristics (Notes 2, 7)

SYMBOL	PARAMETER	CONDITIONS		LF11331/2/3 LF11201/2			LF12331/2/3 LF12201/2 LF13331/2/3 LF13201/2			UNITS*
				MIN	TYP	MAX	MIN	TYP	MAX	
R_{ON}	"ON" Resistance	V_A = 0, I_D = 1 mA	T_A = 25°C		150	200		150	250	Ω
					200	300		200	350	Ω
R_{ON} Match	"ON" Resistance Matching		T_A = 25°C		5	20		10	50	Ω
V_A	Analog Range			±10	±11		±10	±11		V
$I_{S(ON)}$ +	Leakage Current in "ON" Condition	Switch "ON," V_S = V_D = ±10V	T_A = 25°C		0.3	5		0.3	10	nA
$I_{D(ON)}$					3	100		3	30	nA
$I_{S(OFF)}$	Source Current in "OFF" Condition	Switch "OFF," V_S = +10V,	T_A = 25°C		0.4	5		0.4	10	nA
		V_D = –10V			3	100		3	30	nA
$I_{D(OFF)}$	Drain Current in "OFF" Condition	Switch "OFF," V_S = +10V,	T_A = 25°C		0.1	5		0.1	10	nA
		V_D = –10V			3	100		3	30	nA
V_{INH}	Logical "1" Input Voltage			2.0			2.0			V
V_{INL}	Logical "0" Input Voltage					0.8			0.8	V
I_{INH}	Logical "1" Input Current	V_{IN} = 5V	T_A = 25°C		3.6	10		3.6	40	μA
						25			100	μA
I_{INL}	Logical "0" Input Current	V_{IN} = 0.8	T_A = 25°C			0.1			0.1	μA
						1			1	μA
t_{ON}	Delay Time "ON"	V_S = ±10V, *(Figure 3)*	T_A = 25°C		500			500		ns
t_{OFF}	Delay Time "OFF"	V_S = ±10V, *(Figure 3)*	T_A = 25°C		90			90		ns
t_{ON} – t_{OFF}	Break-Before-Make	V_S = ±10V, *(Figure 3)*	T_A = 25°C		80			80		ns
$C_{S(OFF)}$	Source Capacitance	Switch "OFF," V_S = ±10V	T_A = 25°C		4.0			4.0		pF
$C_{D(OFF)}$	Drain Capacitance	Switch "OFF," V_D = ±10V	T_A = 25°C		3.0			3.0		pF
$C_{S(ON)}$ + $C_{D(ON)}$	Active Source and Drain Capacitance	Switch "ON," V_S = V_D = 0V	T_A = 25°C		5.0			5.0		pF
$I_{SO(OFF)}$	"OFF" Isolation	*(Figure 4)*, (Note 3)	T_A = 25°C		–50			–50		dB
CT	Crosstalk	*(Figure 4)*, (Note 3)	T_A = 25°C		–65			–65		dB
SR	Analog Slew Rate	(Note 4)	T_A = 25°C		50			50		V/μs
I_{DIS}	Disable Current	*(Figure 5)*, (Note 5)	T_A = 25°C		0.4	1.0		0.6	1.5	mA
					0.6	1.5		0.9	2.3	mA
I_{EE}	Negative Supply Current	All Switches "OFF," V_S = ±10V	T_A = 25°C		3.0	5.0		4.3	7.0	mA
					4.2	7.5		6.0	10.5	mA
I_R	Reference Supply Current	All Switches "OFF," V_S = ±10V	T_A = 25°C		2.0	4.0		2.7	5.0	mA
					2.8	6.0		3.8	7.5	mA
I_{CC}	Positive Supply Current	All Switches "OFF," V_S = ±10V	T_A = 25°C		4.5	6.0		7.0	9.0	mA
					6.3	9.0		9.8	13.5	mA

Note 1: For operating at high temperature the molded DIP products must be derated based on a +100°C maximum junction temperature and a thermal resistance of +150°C/W, devices in the cavity DIP are based on a +150°C maximum junction temperature and are derated at +100°C/W.

Note 2: Unless otherwise specified, V_{CC} = +15V, V_{EE} = –15V, V_R = 0V, and limits apply for –55°C ≤ T_A ≤ +125°C for the LF11331,2,3 and the LF11201,2, –25°C ≤ T_A ≤ +85°C for the LF12331,2,3 and the LF12201,2, and 0°C ≤ T_A ≤ +70°C for the LF13331,2,3 and the LF13201, 2.

Note 3: These parameters are limited by the pin to pin capacitance of the package.

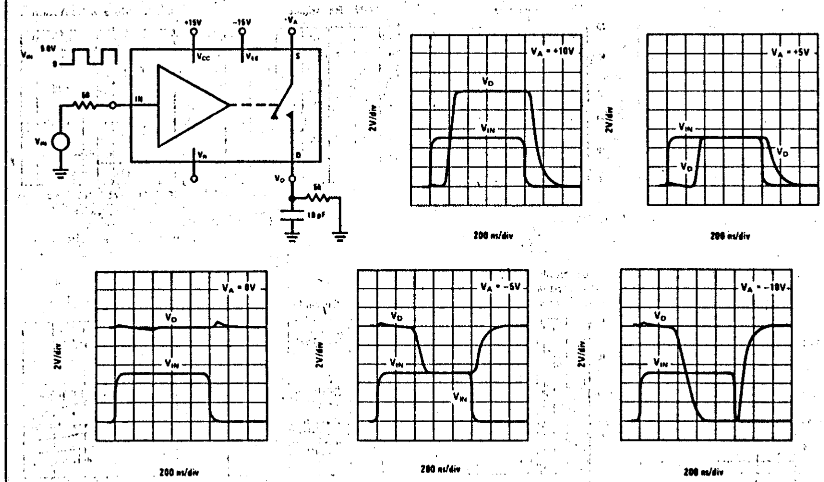
Note 4: This is the analog signal slew rate above which the signal is distorted as a result of finite internal slew rates.

Note 5: All switches in the device are turned "OFF" by saturating a transistor at the disable node as shown in *Figure 5*. The delay times will be approximately equal to the t_{ON} or t_{OFF} plus the delay introduced by the external transistor.

Note 6: This graph indicates the analog current at which 1% of the analog current is lost when the drain is positive with respect to the source.

test circuit and typical performance curves

Delay Time, Rise Time, Settling Time, and Switching Transients



additional test circuits

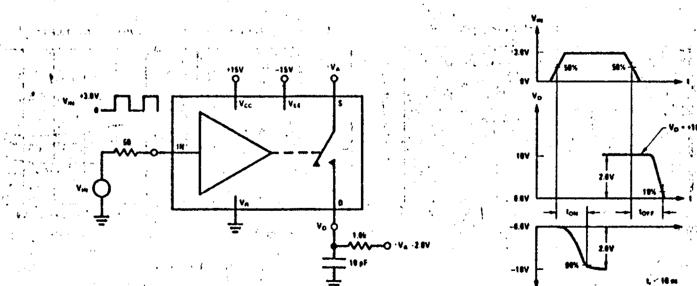


FIGURE 3. t_{ON}, t_{OFF} Test Circuit and Waveforms for a Normally Open Switch

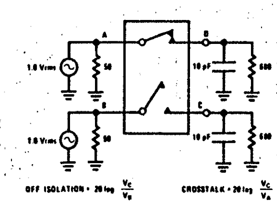
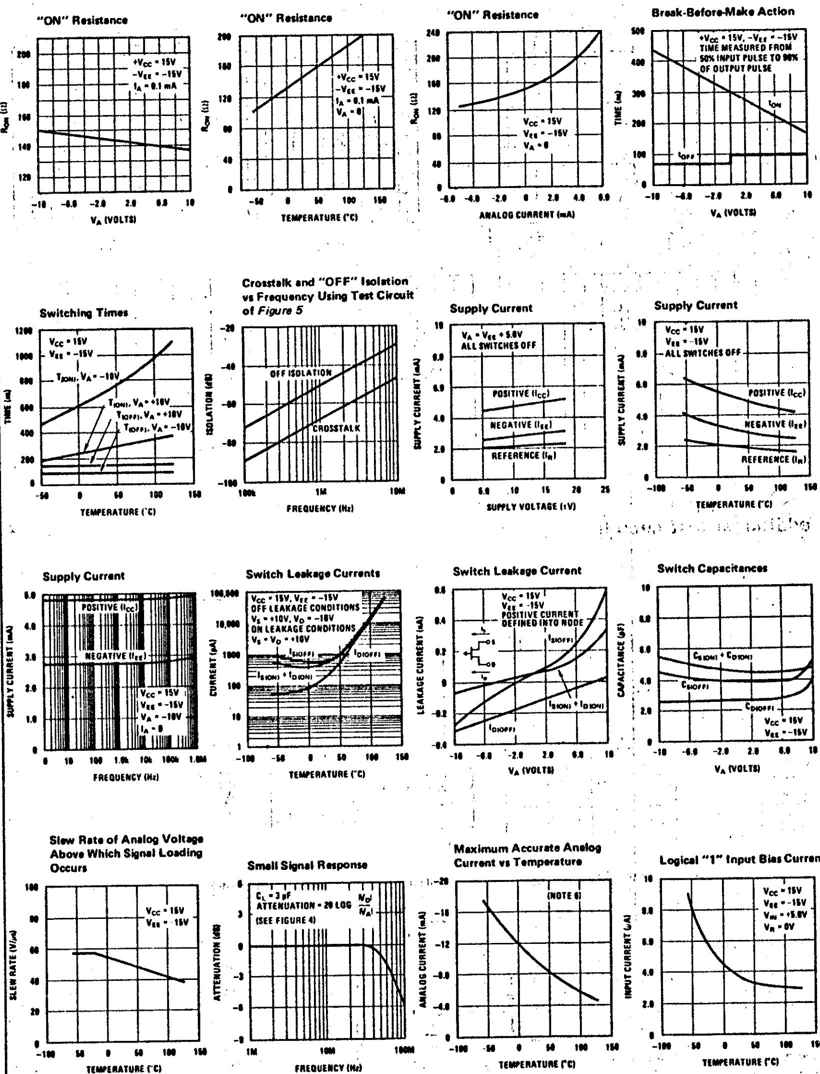


FIGURE 4. "OFF" Isolation, Crosstalk, Small Signal Response

typical performance characteristics



application hints

GENERAL INFORMATION

These devices are monolithic quad JFET analog switches with "ON" resistances which are essentially independent of analog voltage or analog current. The leakage currents are typically less than 1 nA at 25°C in both the "OFF" and "ON" switch states and introduce negligible errors in most applications. Each switch is controlled by minimum TTL logic levels at its input and is designed to turn "OFF" faster than it will turn "ON." This prevents two analog sources from being transiently connected together during switching. The switches were designed for applications which require break-before-make action, no analog current loss, medium speed switching times and moderate analog currents.

Because these analog switches are JFET rather than CMOS, they do not require special handling.

LOGIC INPUTS

The logic input (IN), of each switch, is referenced to two forward diode drops (1.4V at 25°C) from the reference supply (V_R) which makes it compatible with DTL, RTL, and TTL logic families. For normal operation, the logic "0" voltage can range from 0.8V to −4.0V with respect to V_R and the logic "1" voltage can range from 2.0V to 6.0V with respect to V_R, provided V_{IN} is not greater than (V_{CC} − 2.5V). If the input voltage is greater than (V_{CC} − 2.5V), the input current will increase. If the input voltage exceeds 6.0V or −4.0V with respect to V_R, a resistor in series with the input should be used to limit the input current to less than 100μA.

ANALOG VOLTAGE AND CURRENT

Analog Voltage

Each switch has a constant "ON" resistance (R_{ON}) for analog voltages from (V_{EE} + 5V) to (V_{CC} − 5V). For analog voltages greater than (V_{CC} − 5V), the switch will remain ON independent of the logic input voltage. For analog voltages less than (V_{EE} + 5V), the ON resistance of the switch will increase. Although the switch will not operate normally when the analog voltage is out of the previously mentioned range, the source voltage can go to either (V_{EE} + 36V) or (V_{CC} + 6V), whichever is more positive, and can go as negative as V_{EE} without destruction. The drain (D) voltage can also go to either (V_{EE} + 36V) or (V_{CC} + 6V), whichever is more positive, and can go as negative as (V_{CC} − 36V) without destruction.

Analog Current

With the source (S) positive with respect to the drain (D), the R_{ON} is constant for low analog currents, but will increase at higher currents (>5 mA) when the FET enters the saturation region. However, if the drain is positive with respect to the source and a small analog current loss at high analog currents (Note 6) is tolerable, a low R_{ON} can be maintained for analog currents greater than 5 mA at 25°C.

LEAKAGE CURRENTS

The drain and source leakage currents, in both the ON and the OFF states of each switch, are typically less than 1 nA at 25°C and less than 100 nA at 125°C. As shown in the typical curves, these leakage currents are dependent on power supply voltages, analog voltage, analog current and the source to drain voltage.

DELAY TIMES

The delay time OFF (t_{OFF}) is essentially independent of both the analog voltage and temperature. The delay time ON (t_{ON}) will decrease as either ($V_{CC} - V_A$) decreases or the temperature decreases.

POWER SUPPLIES

The voltage between the positive supply (V_{CC}) and either the negative supply (V_{EE}) or the reference supply (V_R) can be as much as 36V. To accommodate variations in input logic reference voltages, V_R can range from V_{EE} to (V_{CC} − 4.5V). Care should be taken to ensure that the power supply leads for the device never become reversed in polarity or that the device is never inadvertantly installed backwards in a test socket. If one of these conditions occurs, the supplies would zener an interal diode to an unlimited current; and result in a destroyed device.

SWITCHING TRANSIENTS

When a switch is turned OFF or ON, transients will appear at the load due to the internal transient voltage at the gate of the switch JFET being coupled to the drain and source by the junction capacitances of the JFET. The magnitude of these transients is dependent on the load. A lower value R_L produces a lower transient voltage. A negative transient occurs during the delay time ON, while a positive transient occurs during the delay time OFF. These transients are relatively small when compared to faster switch families.

DISABLE NODE

This node can be used, as shown in *Figure 5*, to turn all the switches in the unit off independent of logic inputs. Normally, the node floats freely at an internal diode drop (≈ 0.7V) above V_R. When the external transistor in *Figure 5* is saturated, the node is pulled very close to V_R and the unit is disabled. Typically, the current from the node will be less than 1 mA. This feature is not available on the LF11201 or LF11202 series.

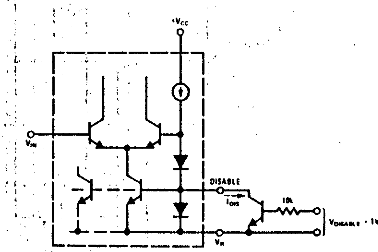


FIGURE 5. Disable Function

THE WMFO ON-AIR RELAY BOXES

INPUTS:

* Normally-off contact from control board muting relay or microphone switch

OUTPUTS:

* Several sets of Normally-off and Normally-on contacts

THE WMFO ON-AIR RELAY BOXES

I) Functions and Operation:

There is an on-air relay box located in each studio. In Studios A & B, the relay boxes are mounted on the side of the right Turntable Cabinet, beneath the board. In Studio C, the relay box is mounted on the side of the Turntable Cabinet, inside the door below TT1.

The purpose of these devices is to create several readily available relay contacts (which switch when a studio mic. is live) from a single normally-off contact inside of the board. Since these relays are located outside of the board chassis, one may switch any type of signal through them without risking crosstalk into the audio lines (only a low-current DC signal is brought into the control board to trigger the relay box). The relay box contacts are normally used to mute the studio PA speakers, activate the "ON-AIR" warning lights atop the studio doors, and illuminate the "Mic Live" LEDs inside the studio switcher panels (as well as illuminating the doorbell pushbuttons when the transmitting studio goes live). The relay box contacts in Studio A are also used for the "Cassete Skim" option (see "Studio A Remote Control/Auto-Skim" writeup). Spare contacts are available in most studios for future expansion. Additional relays may also be added fairly easily if extention of capability is required.

II) Circuit Description:

The basic relay driver circuit is very straightforward, as seen in the single-page schematic. A 35-Volt (unfused; be careful!) power supply is input to emitter-follower Q1. When the air mike goes live in the control board, the base of Q1 is driven high through R2, causing it to pass collector current and switch any relay(s) wired between its emitter and ground (D1 protects against back-bias transients from the relay coils). The actual implimentation of this circuit varies in each studio. In Studio C, all electronics except the relay are housed inside a small aluminum box; the relay (which has 4 poles) is mounted externally. In Studio B, everything is mounted in a single aluminum chassis; here Q1 drives several socketed multi-pole relays. The arrangement is a bit different in Studio A, where all electronics and a single relay are mounted in a black box. The relay has two contacts; one is available for general use, while the other switches 110 VAC which is used to drive another set of socketed relays mounted in a separate aluminum chassis. Sorry for the lack of standardization; these boxes were all built out of available junk, and assembled at different times.

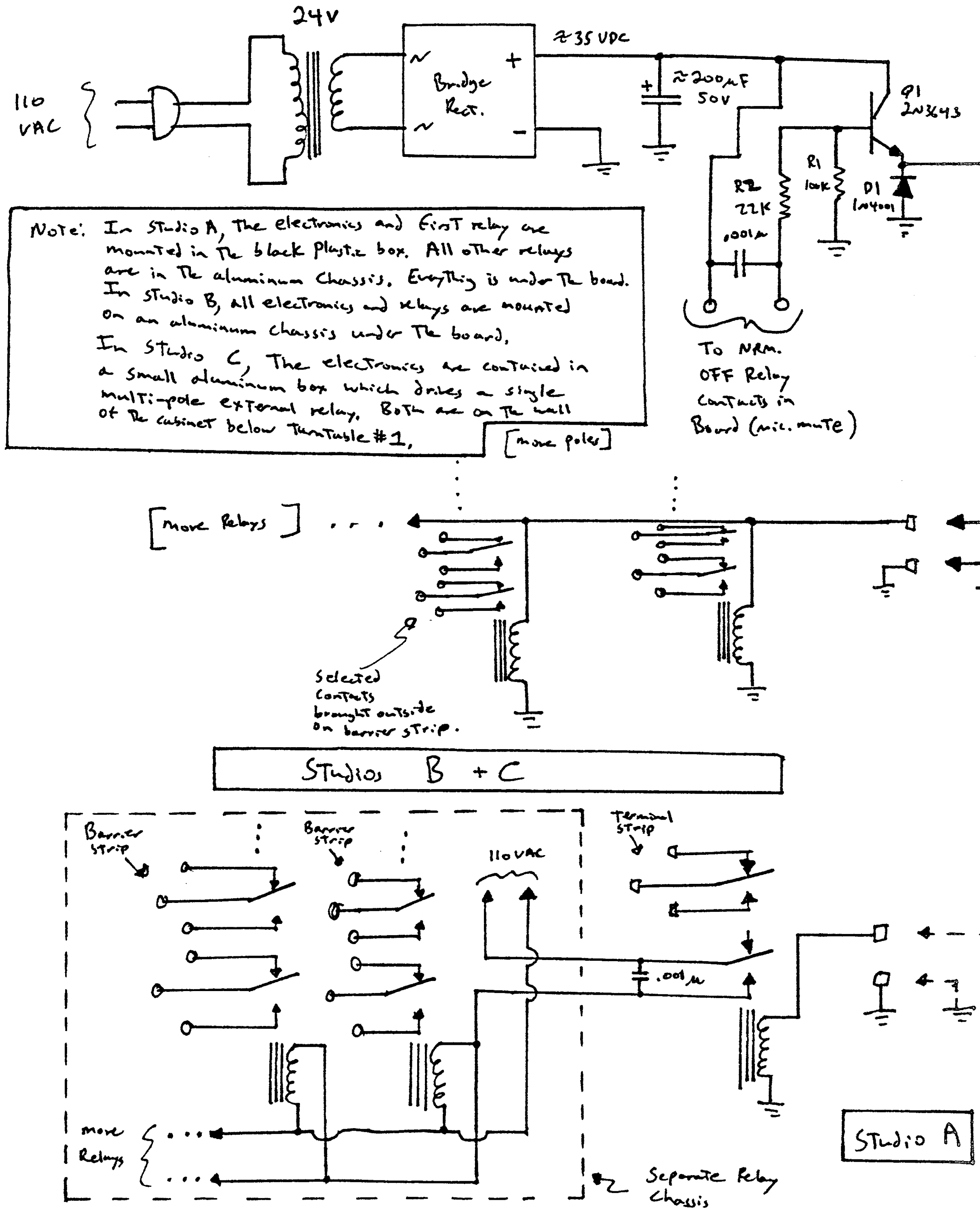
III) Troubleshooting Hints:

The symptoms to watch for are PA speakers that don't mute (or are always off), ON-AIR lights that are always on or off, and "Mic. Live" LEDs in the Studio Switcher control boxes that remain lit or dead. This circuit is readily checked with a volt-ohmmeter. Make sure that the relay box is plugged in, and receiving AC power. Make sure that the input from the board is switching OK. Make sure that the problem is not a single bad relay contact (ie. if only one device driven by the relay box is malfunctioning, this might be the problem). If the difficulty seems inside the relay box, first check the power supply voltage, and then check (or replace) Q1. The only problem with one of these happened in Studio C recently, where Q1 blew out (ie. the beta went down to zero). This is probably because I originally omitted D1 from the design. I installed it while making repairs (the relay boxes in the other studios may not have D1 installed; if they blow out due to a bad Q1, install D1). Their longevity seems quite good; the device in Studio A has been running already for 10 years without a hitch. The box in Studio B has been operating for 3 years without problems.

Note that the PA mute and "Mic Live" LEDs use Normally ON contacts. The ON-AIR lights employ Normally OFF contacts. The 110 VAC switching for the ON-AIR lights actually occurs in a circuit mounted behind a panel sitting in the grey monitor rack in Master Control. This circuit uses a low trigger voltage to switch the line current via solid-state relays (and was built by Jeff Goldsmith, E74). The light bulbs inside the ON-AIR light fixtures will occasionally need replacing; they use conventional "night-light" 7 Watt (or so) bulbs.

The connections on the relay socket block in Studio C might be a bit flaky; if this relay seems to be problematic, check these connections first.

WMFO On-Air Relay Boxes   June, 1987 — J. Paradiso



THE WMFO CASSETTE REMOTE CONTROL/AUTO-SKIM

INPUTS:

* Wiper, Normally ON, Normally OFF contacts from a "live mic." relay

* +26 Volts from LPB control board

OUTPUTS:

* Pause and Play pull-to-ground outputs to control cassette deck

THE CASSETTE DECK REMOTE CONTROL/AUTO-SKIM

I) Function and Operation:

This device is located in a small chassis mounted atop the Studio A control board. It provides remote-control capability for a rack-mounted cassette deck. Only the "pause" & "play" buttons are remoted in this device. Needless to say, beware of removing a cassette under "pause"; the "stop" button on the deck should always be hit before the cassette is ejected.

This circuit also enables "skim" tapes to be made (ie. recordings are made only when the studio microphones are live, causing the DJs voice and/or live on-air material to be exclusively captured). To set up "skim" operation, first place the cassette deck into record, and activate "pause". Turn the switch on the remote control panel to "skim". Whenever the studio mics. are activated, the green light on the remote control panel will flash, and the cassette deck will record. When the mics. are off, the green light is extinguished, and the cassette deck goes back into pause.

Under normal operation, leave the remote control in the "NRM" position, otherwise the cassette deck will always try and put itself into pause when the mikes are off, and put itself into play when the mics are on. If a cassette tape winds to its conclusion under "skim" operation, the remote will still keep trying to put the deck into play (even though the cassette deck continues to stop itself shortly thereafter). This will cause an annoying periodic "clunking" as the cassette deck cycles between play and stop. To avoid this, flip the switch on the remote to "Nrm." shortly after the skim tape winds to its end.

This remote unit was designed for a modified JVC-220 cassette deck. The pause and play pushbutton contacts on this deck were brought out externally via a 3-terminal connector mounted on the rear of the chassis. The deck that was modified, unfortunately, died after a year or so (presumably not due to the action of this remote). The deck is now sitting on a shelf in the tech room waiting for eventual repair. If this deck is ever repaired properly, it might be returned to its place in the Studio A rack, and re-connected to the remote unit. Otherwise, this remote unit may easily be modified to work with other cassette decks.

II) Circuit Description:

The circuitry in this remote control box is quite simple, as drawn in the single-page schematic. The JVC-220 pushbuttons operate by clamping their contacts to ground. This is also accomplished in the remote control whenever pushbuttons S2 or S3 are pressed, or whenever transistors Q2 or Q3 are turned on. If switch S1 is in the "Nrm." Position, these transistors remain off, and control must be accomplished solely via S2 and S3. When S1 is in the "skim" position, the output of the 555 astable IC1 will drive either Q2 or Q3, depending on the state of the "mic-on" relay to which the remote is connected. When the mics. are off, IC1 will pulse Q2, which keeps the deck in pause. When the mics. go live, Q3 is pulsed instead, which puts the deck into play. The "mic.-on" relay which is used here is located in the "On-Air Relay Box", located under the counter (see the associated writeup for details).

IC1 produces square waves at approx. a 2 Hz. rate. It was introduced due to transients in the response of the cassette deck. Because the internal processor in the deck scans the control buttons periodically, a rapidly changing state (such as a switching relay) may be missed, causing unpredictable operation. By pulsing the control buttons repeatedly via IC1, the cassette deck's processor eventually responds to a changing input (usually before the third pulse arrives).

Transistor Q1 and zener D1 drop the 26 Volt supply from the LPB board down to 5 Volts or so, which happily and safely drives the 555.

This circuit will directly control any cassette deck which operates its control lines on a "pull-to-ground" basis. By adding pull-up resistors onto the outputs, it will also work for any opto-isolated system. Additional relays or isolation circuits may be installed to drive systems which don't work on a pull-to-ground basis. By using a bit of ingenuity, one should easily be able to adapt this circuit to control any cassette deck. Now that the original deck has been removed from the Studio A rack, the remote cable is dangling there, terminated in a 3-prong connector.

III) Troubleshooting Hints:

First, make sure the deck is not at fault. This is easily accomplished by disconnecting the remote cable, and testing the deck independently. Check the relay connection in the on-air relay box to make sure the remote-skim is being switched correctly. Check that power supply voltages are proper. Check the 555.

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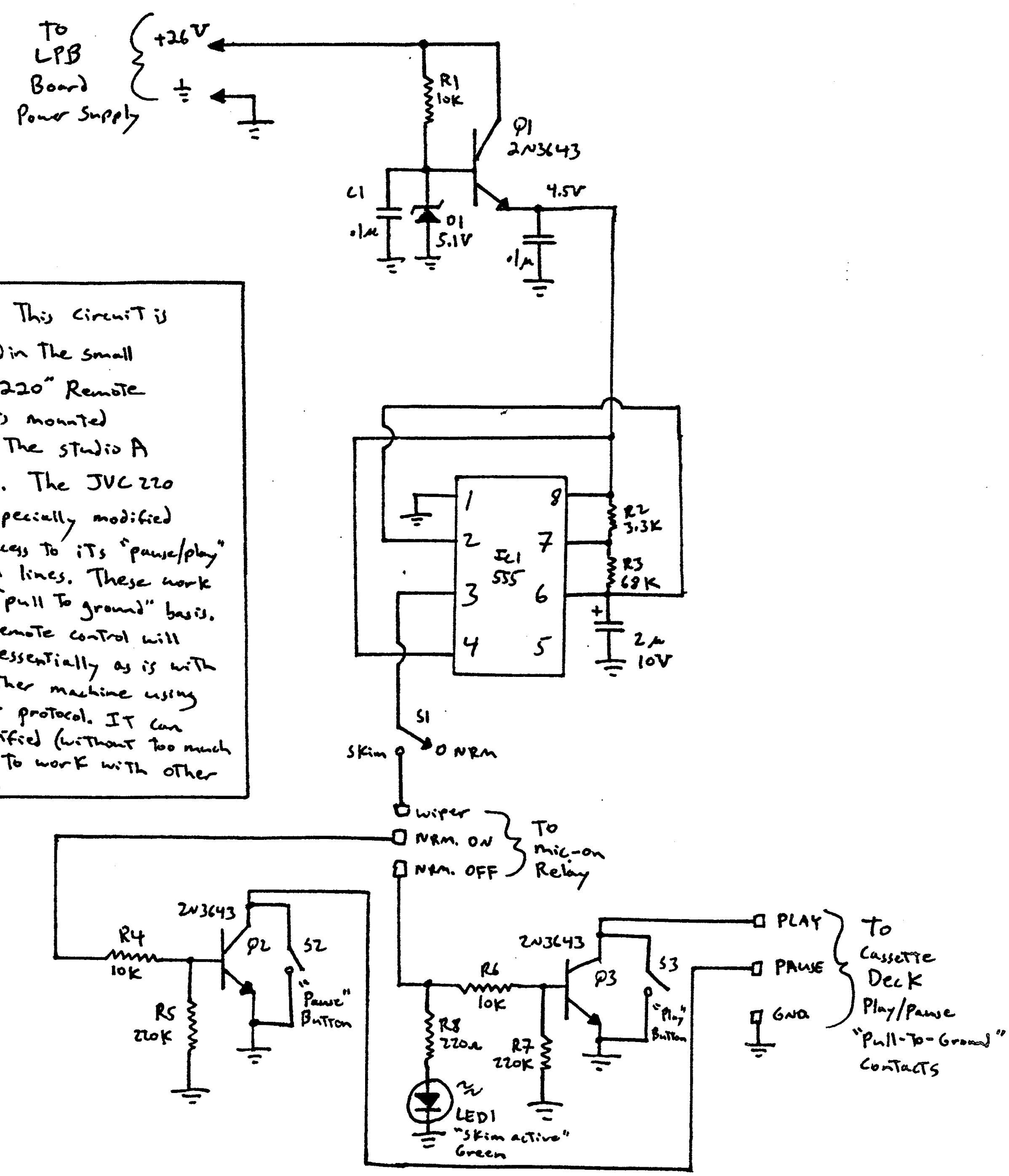
WMFO Remote Control/Auto-Skim for Cassette Deck



Notes: This circuit is located in the small "JVC 220" Remote chassis mounted atop the studio A board. The JVC 220 was specially modified for access to its "pause/play" control lines. These work on a "pull to ground" basis. This remote control will work essentially as is with any other machine using similar protocol. It can be modified (without too much effort) to work with other systems.

THE WMFO STUDIO A MONO/STEREO MIC. RELAYS

INPUTS:

* 2 Pairs of Left & Right Mic. Inputs (from Pot #1B and Pot #2B)

* 2 DC activation lines (from selector switches for Pots #1 & #2)

OUTPUTS:

* 2 Mono mic. outputs (with both left and right inputs mixed). These are routed to Pots #1A & #2A.

THE STUDIO A MONO/STEREO MIC. RELAYS

I) Function and Operation:

This circuit is located on a breadboard mounted on the rear wall of the Studio A control board (inside the console). It engages special relays whenever the studio mics. (ie. the mics. on pots #1 and #2) are on-channel and the respective selector switches are set in position B ("Stereo Mics."). Each pot (#1 or #2) has two mics. associated with it. Pot #1 has the DJ mic. (which is assigned to the right channel), and one guest mic. (on left). Pot #2 has two interview mics. (one each for left and right). XLR jacks for Pot #2 (R & L) and Pot #1 L are mounted on the outside face of the cabinet supporting the control board. The XLR jack for the DJ mic. is mounted on the top counter of this cabinet, adjacent to the control console and DJ mic. boom. When the selector switch for Pot #1 or Pot #2 is in the "A" (ie. "mono") position, the two respective mics. are padded together, and both left & right channels are supplied with a mono signal (ie. both mics appear in the "center"). When the selector switch is in the "stereo" position (ie. "B"), this mixing pad is broken, and each mic. will appear separately in the assigned stereo channel.

The creative applications of this feature are considerable; one can have the DJ assigned to center, and guests on either side, or assign DJ and guests to the sides and place a phone caller in the center (phone calls will always be in mono unless somebody does some creative patching). Have fun......

II) Circuit Details:

There is no active circuitry here, as seen in the single page schematic. The LPB board in Studio A has 4 mic. preamps; one each for left and right channels of both pots #1 and #2. The balanced microphone lines coming into the board are tied to the input terminals for pots #1B and #2B. These terminals are also routed to the relays located on the handwired card mounted on the rear wall of the console. When the relays are off (ie. mike selectors are in the "A" position), the Left and Right mic inputs are mixed by 75-Ohm balanced pads (ie. the 27-Ohm resistors). The outputs of these pads are routed to both left and right input terminals for Pots #1A and #2A (ie. the mono feeds). If a mic. selector is placed in the "Stereo" (ie. B) position, the associated relay on the handwired card will be activated, causing the corresponding mixing pad to be broken and placing each microphone discretely in left and right channels. The 680-Ohm resistor drops the 26 Volt LPB supply to provide 12 Volts needed to activate the DIP relay (these resistors must be rated at least 2 Watts). Diode D1 is mounted on the selector switch, and isolates the control of the DIP relay from the master on-air muting line.

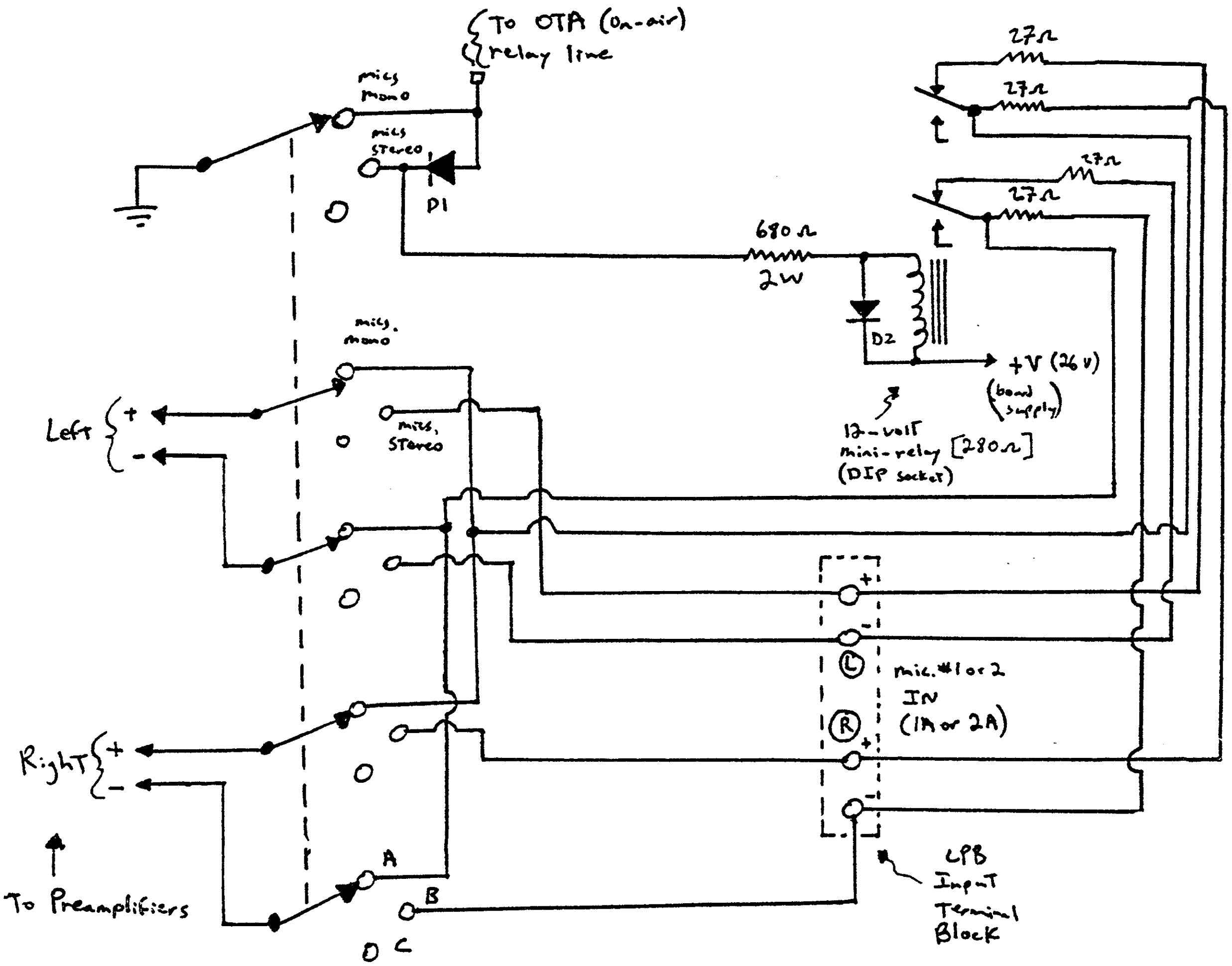
II) Troubleshooting Hints:

Check the microphone input connections. Check the mic. preamps. Check first the wiring in the board (after re-building the board, this wiring was a bit flakey due to repeated stress. Most of the bad wiring and poor connections have already been isolated and repaired; the board has been running fine for over a year now). Check the selector and program/audition switches; these contacts occasionally get corroded, and need cleaning. The DIP relays are socketed for easy replacement.

Jan., 1987 – J. Paradiso

WMFO Studio A mono/stereo mic. Relays

D1, D2 = 1N4001 or sim.



Selector switch for mic. pot (#1 and #2)

[This is the standard LPB A/B/C Knife Switch]

The selector switch for Pot #1 has 6 switch poles (one extra to defeat the cue).

Notes: There are two of these circuits; one for mic. #1 and one for mic #2.

The 680Ω and 27Ω resistors, D2, and the 12-V mini relays are mounted on a kludge card hung on the back cover of the board.

THE WMFO STUDIO A GAS LIGHT

INPUTS:

* Balanced Mono Audio In
  (appears on Studio A Patchfield)

OUTPUTS:

* Modulated 110 VAC
  (standard outlet which normally
  is connected to GAS Light socket)

THE WMFO STUDIO A GAS LIGHT

I) Function and Operation:

The Studio A GAS Light is mounted at the top of the Studio A rack. The GAS Light is a small red indicator light which illuminates to display the word "GAS" whenever an input audio level crosses a pre-set threshold. The GAS light input is routed through the Studio A patchfield, and normalled to an output of the Studio A Distribution Amplifier (which is normalled to the program output of the board).

The GAS light is mainly for aesthetic purposes (ie. it's a hack). It flashes amusingly with the source material on program in Studio A. It may also be patched to another source, and thereby serve a potentially useful purpose as an indicator of audio level.

II) Circuit Description:

The GAS light circuitry is quite straightforward, as shown in the single-page schematic. The circuitry is actually located in two places, as noted in the schematic. The mixer/driver OP-AMP (IC1) is mounted on a piece of perfboard piggybacked on the Studio A distribution amplifier card (from which it draws its +/- 18 Volt supply). If one turns the distribution amplifier (ie. interface) off, the gas light certainly won't work (although many more severe things will happen as well). The triac and associated circuitry which modulate the 110 Volt line current for the GAS Light bulb are located in a small aluminum box mounted behind the chassis at the top of the Studio A rack. The OP-AMP driver illuminates LED #1; this light changes the resistance of the CDS Photoresistor PR1, which can turn the Triac TR1 on, thereby illuminating the GAS Light. The photoresistor is shunted via potentiometer P1; this adjustment allows one to tweak the GAS light sensitivity and response.

III) Calibration and Adjustment:

First, remove all audio from the GAS light input (ie. cut the Studio A program output, or insert a dummy patch into the GAS light input). Adjust P1 (mounted on the aluminum box located behind the GAS Light panel) so that the GAS light just begins to illuminate. Back off a bit so that the GAS light remains completely dark.

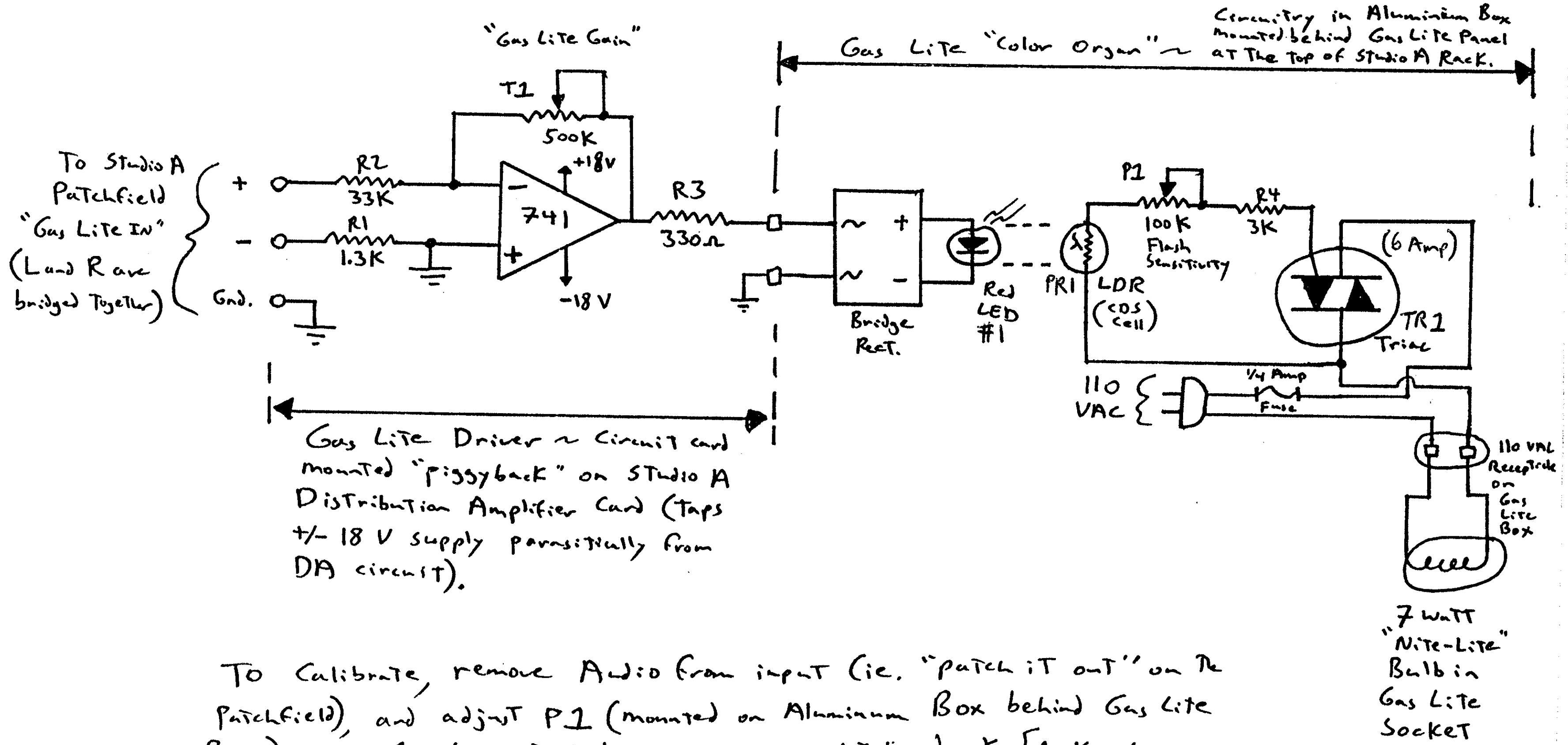
Now introduce nominal audio into the GAS light input (ie. remove the dummy patch, and put something on the Studio A board at average level). Adjust trimmer T1 (located on the perfboard piggybacked on the Studio A distribution amplifier [behind the interface panel]) so that the GAS Light pulses nicely with the music. The GAS Light is now calibrated.

IV) Troubleshooting Hints:

Every year or two, the light bulb in the GAS Light fixture blows out and will need replacing. THIS IS A VERY IMPORTANT TASK; THE GAS LIGHT MUST REMAIN FUNCTIONAL!!! It was a favorite hack of the tech crew, and we'd like to see it maintained. This is very easy to do. The GAS Light takes conventional 7 Watt (or so) "nightlight" bulbs, easily available anywhere (the ON-AIR lights mounted above the studio doors take the same bulbs).

If the GAS light remains on for some reason, check the adjustment of P1, as mentioned above (check also the audio input). If the GAS light still remains on, the triac TR1 may need replacing (this should be available via Radio Shack). If the GAS light remains off, check the fuse. Check also the input audio feed, and the adjustment of trimmer T1. Check LED1. Before you do anything rash, make sure the GAS light bulb is OK, and plugged into the driver box. Make sure the driver box is plugged into the 110 Volt AC line. Don't get a shock.

The WMFO Studio A "GAS LITE" © 1987 – J. Paradiso and L. Shein

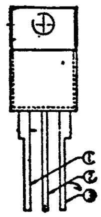


To Calibrate, remove Audio from input (ie. "patch it out" on the Patchfield), and adjust P1 (mounted on Aluminum Box behind Gas Lite Panel) until Gas Lite just barely goes completely dark [make sure Gas Lite Bulb is good, + 1/4 Amp fuse is OK if problems...].
Apply audio @ nominal level, + adjust T1 (mounted on small card "piggybacked" on Studio A's DA card) for optimal flash rate.
If problems, check Audio path and LED in optocoupler.

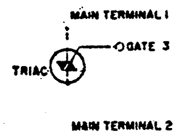
Note: Since this was hastily assembled, the drawing of "Color Organ" circuitry @ right may not be entirely accurate.... Watch for mods....

6 AMP TRIAC

276-1000 400 PIV



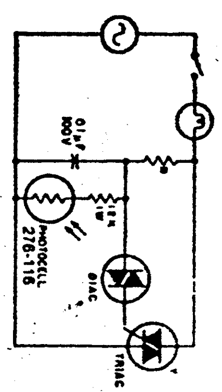
PACKAGE OUTLINE (TO - 220)



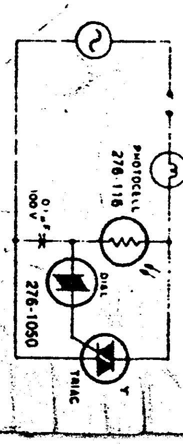
MAXIMUM RATINGS

CATALOG	PART #	PIV	FORWARD CURRENT	GATE CURRENT	GATE VOLTAGE
276-1003	-1003	50 VOLTS	6 AMPS	25 MA	3 VOLTS
276-1002	-1002	100 VOLTS	6 AMPS	25 MA	3 VOLTS
276-1000	1000	400 VOLTS	6 AMPS	25 MA	3 VOLTS

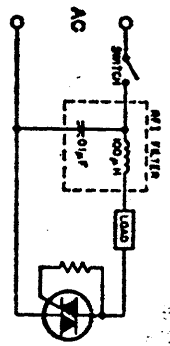
TRIAC APPLICATIONS



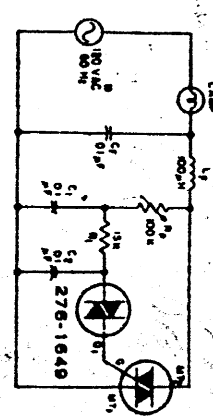
Light Controlled Turn-Off Circuit.



Light Controlled Turn-On Circuit.



RFI-suppression filter.



A typical light-dimmer circuit.

THE WMFO STUDIO A KLUDGE CARDS

INPUTS:

* Balanced External Monitor Feeds (L & R)
* Stereo Headphone Feeds (from headphone level pot and source selector switch)
* 4 Balanced microphone inputs
* Program Feeds (L & R)
* Center-Tap from Telephone fader (L & R)
* Cue Bus Feed
* Audition Feed
* +26 Volts from LPB board

OUTPUTS:

* Single-Ended External Monitor (L/R)
* Stereo Headphone Outputs (L/R)
* Telephone audio feed (mono)

THE WMFO STUDIO A KLUDGE CARDS

I) Function and Operation:

The phrase "Studio A Kludge Cards" refers to two handwired perfboards mounted inside of the Studio A mixing console. The purpose of these cards is to augment some of the functions of the LPB board. The card mounted at left (facing the board from front) contains a stereo headphone amplifier and stereo external monitor line receiver.

The headphone amplifier allows several sets of monitor headphones (of any impedance) to be plugged into the console. Two headphone jacks are provided below the console countertop (in front), two headphone jacks are provided at the side of the console cabinet (next to the interview mic. jacks), and one headphone jack is provided at the rear of the board. All of these jacks are driven from the headphone amplifier mounted on the left kludge card. The headphone source (ext. mon., program, cue) is selected by the appropriate knife switch on the console, and the headphone volume is adjusted via the console-mounted control.

The external monitor line receiver converts the balanced external monitor signal supplied to the studio into a single-ended line used by the LPB board. Connecting the balanced signal directly to the single-ended inputs caused considerable noise and crosstalk, hence the line receivers were installed.

The kludge card mounted at the right side of the console creates the audio feed supplied to the studio telephone. When working with the telephones, the "handset" switch mounted on the top of the telephone should be off, inhibiting audio feeds to and from the telephone handset. The handset should be removed from the switchhook, and the desired line depressed on the telephone selector buttons. Audio from the telephone line appears on pot #6; by potting this up, callers may be placed on the air. Pot #6 may also be placed in cue so callers can be auditioned before going live (eliminating the necessity of ever using the handset). The handset can be left off the hook throughout the duration of the show; to disconnect the caller, merely punch the button on the telephone corresponding to the unused line (presently line #4). Make sure the telephones are potted down on the board before doing this in order to avoid a blast of noise and potential feedback.

The Studio A setup allows one of three signals to be fed down the telephone line to the caller. The telephone source is determined by the setting of the "Feed to Telephone" knife switch on the console. If this switch is in the "mics. only" position, an audio feed from all 4 microphones in the studio (2 on pot #1 [including DJ] and 2 on pot #2) is continuously fed down the selected telephone line (irrespective of whether any mics are on program or potted up). This is the simplest way of doing a telephone interview, and perhaps the most commonly applied @ WMFO. Using the "mics. only" feed, one can easily preview callers in cue without ever having to use the handset, as mentioned above.

If the telephone selector switch is set to the "Aud." position, the console's audition mix is fed down the telephone line. This allows one to pot up music, voice, etc. on audition, and send it to the caller. Anything appearing only on program is not sent. By also potting up the audition line on program, the audition audio is directed both to the caller and on-air audience.

If the telephone selector switch is set to the "Pgm." position, the console's program output is sent down the telephone line to the caller. Since the signal from pot #6 (which is defaulted to the telephone) is subtracted from the caller's feed, the caller may be potted up on program without feedback while all other program material is still sent down the telephone line. This is perhaps the most convienent means of using telephones in a music/talk context (ie. "Name That Tune"); whatever is being played over the air in program (except the caller's voice) is sent to the caller. This option has two minor hitches, however. Because of inherent difficulties in subtracting out the caller's voice from the telephone feed, feedback and "howling" can result when the caller is potted up excessively high. This is generally not a problem, since enough feedback-free range exists for most telephone connections. The problem may become more evident when the caller is at a very low level (ie. poor telephone connection), causing more gain to be needed. In these cases, one should resort to the "feedback-free" mics. only or audition feeds. Another quirk of this "Pgm." feed is that anything on cue will also be injected into the telephone line (except for cue on pot #6). Thus, if you're cueing up a record while talking to a caller under the "Pgm." feed, the caller (and potentially the listening audience) will hear the record being cued. Again, use another type of telephone feed if this becomes a problem.

Because of the way in which pot #6 is subtracted from the telephone feed under the "Pgm." setting, the telephone output can not be patched to another fader when this feature is being used (otherwise terrible feedback will immediately result). The telephone output may be re-routed to any fader when either the Audition or Mics. Only feeds are used.

The outputs of the telephone feeds from each studio are routed to the Master Control patchfield, where they are normalled to the audio inputs of the Symmetrix telephone interface units. These audio feeds may be re-directed in Master Control for other purposes; for instance, the "mics. only" feed can provide a direct line-level microphone output from each studio which is independent of the mixing consoles. This feed has proven to be ideal for espionage, vocal effects processing (ie. real-time pitch shifting), etc.

Symmetrix TI-101 telephone interfaces are used to connect to the telephone lines. There are three of these units mounted in the blue interstudio rack in Master Control. Each of these units buffers the telephone line selected by the telephone in each studio (ie. one unit is dedicated to Studio A, another is dedicated to Studio B, and a third is dedicated to Studio C).

The telephone line selected is the line punched up on the studio telephone. Several lines may be punched up simultaneously on these telephones (they've been specially modified; try punching two buttons at once); this enables the lines to be "conferenced". The WMFO intercom line may also be used with the telephone interface in the same fashion. One can as well conference the intercom to an outside line (this is useful if you are located in a remote site [ie. transmitter room], and need to make an outside call on the intercom).

The Symmetrix TI-101 buffers the audio feed to the telephone line, and also creates an audio output from the telephone to the studio. This audio output supposidly "subtracts" the input audio feed (ie. the DJ's voice) from the output feed in true "hybrid" fashion. This does not work too well, however, hence our Symmetrix TI-101s have been modified to work also as "phone switches", where the amplitude of the DJ's voice attenuates the telephone output. This can create potential conflict when both parties try to speak simultaneously, however this problem has not surfaced under operation, and the "switching" action of the telephone interface enables a very crisp, clean, and professional-sounding telephone show to be conducted.

II) Circuit Description:

The circuitry on the left-hand kludge card is depicted in the first page of the schematic. The headphone amplifier (A) is based around two LM13080 power OP-AMPs (IC1 & IC2). The 7815 regulator drops the 26 Volt LPB supply down to 15 Volts, which is digestable by the LM13080's. Since they operate off a single-ended supply, the OP-AMPs must be biased up (R9,R10) and DC isolated from the outside world (C1,C2,C6,C7). The output of the headphone amplifier essentially comes right off of the OP-AMPs; 75-Ohm buffering resistors (R3-R7 and R16-R20) are inserted in series with the headphone outputs at each headphone jack (these resistors are not mounted on the card). Because these resistors effectively isolate each headphone port, headphones of any impedance may be plugged in without disturbing the other sets already connected.

The lower portion of this diagram (B) shows the external monitor line receiver circuitry. This is a standard pair of differential amplifiers built around a 5532 dual OP-AMP. Because of the single-ended supply, all OP-AMPs are biased up via R21/R22, and outputs are DC isolated (C9 & C10). The output of left and right channels may be independently trimmed via T1 & T2.

The telephone interface circuitry (located on the right kludge card) is depicted in the second page of the schematic. Four differential amplifiers based around 5532 dual OP-AMPs (IC2 & IC3) pre-amplify the signals from all 4 studio microphones (3 interview and 1 DJ). These amplifiers are fairly high impedance (22 KOhm to ground on inverting inputs), thus they are parallel jumpered with the console mic lines (at the input terminals for faders 1B and 2B) with negligable effect on the audio quality of the console microphone feeds. The outputs of IC2 & IC3 are summed by IC4, with additional gain set by trimmer T4. The DC bias on the output of IC4 may be adjusted to 13 Volts (half of the LPB 26-Volt supply). via trimmer T5. When the selector switch (S1) is set to "Mics. Only", the output of IC4 is additionaly amplified and converted to a differential balanced signal by driver IC5. When S1 is set to "Aud.", the single-ended audition output from the console is input directly to the balanced driver (IC5) to send the audition channel down the telephone line.

The center-tap of the telephone fader (Pot #6) is subtracted from the program output in IC1b (these signals are inverted with respect to one another, thus they are simply summed). The left/right program balance is adjusted via T2, and the program level is set via T3 such that the signal from pot #6 is completely nulled. Because the input audio is also routed to the fader center-tap when it is put into cue, the cue bus output must as well be subtracted from the fader signal to enable fader #6 to be used in cue. IC1a inverts the cue bus for this subtraction, and T1 adjusts the cue amplitude such that the cue feed of pot #6 can be completely canceled.

Because of the single-ended supply voltage (all circuitry in the telephone driver operates off the LPB 26 Volt supply), all OP-AMPs in this circuit are biased up at 13-Volts (via R10 & R11), and all inputs and outputs to this circuit must be DC isolated via coupling capacitors.

III) Calibration and Adjustment:

First, I'll outline the calibration of the left-hand card (first page of schematic). The Headphone amplifier needs no calibration. The external monitor levels may be set by adjusting T1 & T2. With a good, robust audio source sounding over the air, place the board into "Ext." monitoring, set the monitor level at mid-pot, and tweak T1 & T2 until a comfortably loud, left/right balanced audio eminates from the monitor speakers. The External monitor line driver and distribution amplifier should already be calibrated before attempting this adjustment; see their associated writeups for details.

I'll now outline the calibration of the right-hand card (second page of schematic). Several adjustments must be made to set up the telephone driver circuitry. First place the selector switch (S1) into the "mics. only" position. Measure the output of IC4 with a voltmeter, and adjust trimmer T5 to yield 13 Volts. Next, speak normally into a microphone, and adjust trimmer T4 until a nominal, understandable audio level is recieved over the telephone (this adjustment may depend somewhat upon the settings of the Symmetrix TI-101 telephone interfaces; consult their manual for details). After T4 is adjusted, T5 may require additional trim to maintain the 13 Volt quiescent output bias.

Next, the program nulling circuitry must be balanced. First, pot up the house tone oscillator over program (set it to read zero dB on the meters; don't use fader #6). Adjust T2 to yield an equal left/right balance at its output. This can be done in several ways; one is to invert one channel of the program source (ie. plug the house tone through the Interface before it is input to the board, and hit "right channel invert". Use only the mono output, and make sure it is perfectly balanced [patch it through a mono bridge to insure this]). Monitor the output of the telephone feed on audition (ie. patch the Studio A telephone input feed from Master Control into a fader on the Studio A board (other than fader #6), and put it into audition). Set T3 to maximum, and adjust T2 until the house tone is nulled (ie. dissapears from the audition audio). Next put both house-tone channels back into phase (ie. flip the "R-Ivt." switch on the interface back to "Nrm.", or disconnect the interface entirely). Patch the house-tone (or interface output) into fader #6, and put it into program at half-pot. Adjust trimmer T3 such that the house-tone component in the telephone feed (monitored over audition) is at a minimum. The house-tone feed is entirely removed when T3 is full off, however this extreme is undesired; another intermediate setting of T3 will produce the null, and this is what is called for.

Next, place pot #6 into cue, and adjust trimmer T1 such that the house-tone audio (which should re-appear over audition), is again nulled from the telephone feed.

The Studio A kludge cards are now calibrated. The telephone program null is affected by the setting of the master program gain control mounted inside the board. If this gain control is adjusted for any reason, the telephone program null must be entirely re-calibrated as discussed above. Always trim these nulls after playing with the master program gain!!

The location of all kludge card trim pots are labeled inside the top cover of the LPB board. Refer to these diagrams for assistance.

IV) Troubleshooting Hints:

The functions of the kludge card are discretely broken up; ie. headphone driver, external monitor interface, and telephone feed (the telephone feed circuitry is further subdivided into mics., audition, and nulled program sections). The symptoms of defects (ie. which functions are inhibited) should point fairly directly to the vicinity of the problem. The kludge cards tie quite heavily into the innards of the console; if anyone monkeys with the console, these connections may be disturbed, causing an apparent malfunction.

If the microphone feed seems distorted or inhibited, check the bias adjustment T5; it may drift. If the nulled program feed seems to be feeding back and howling excessively, T1-T3 on the telephone card may need to be re-calibrated as discussed above.

The Studio A kludge cards have operated without a hitch for 1.5 years. Hopefully there's lots more longevity in store. If another console is installed in Studio A, one should perform similar custom modifications to maintain the functions of these kludge cards. Perhaps the original kludge cards in the LPB board can even be transplanted.....

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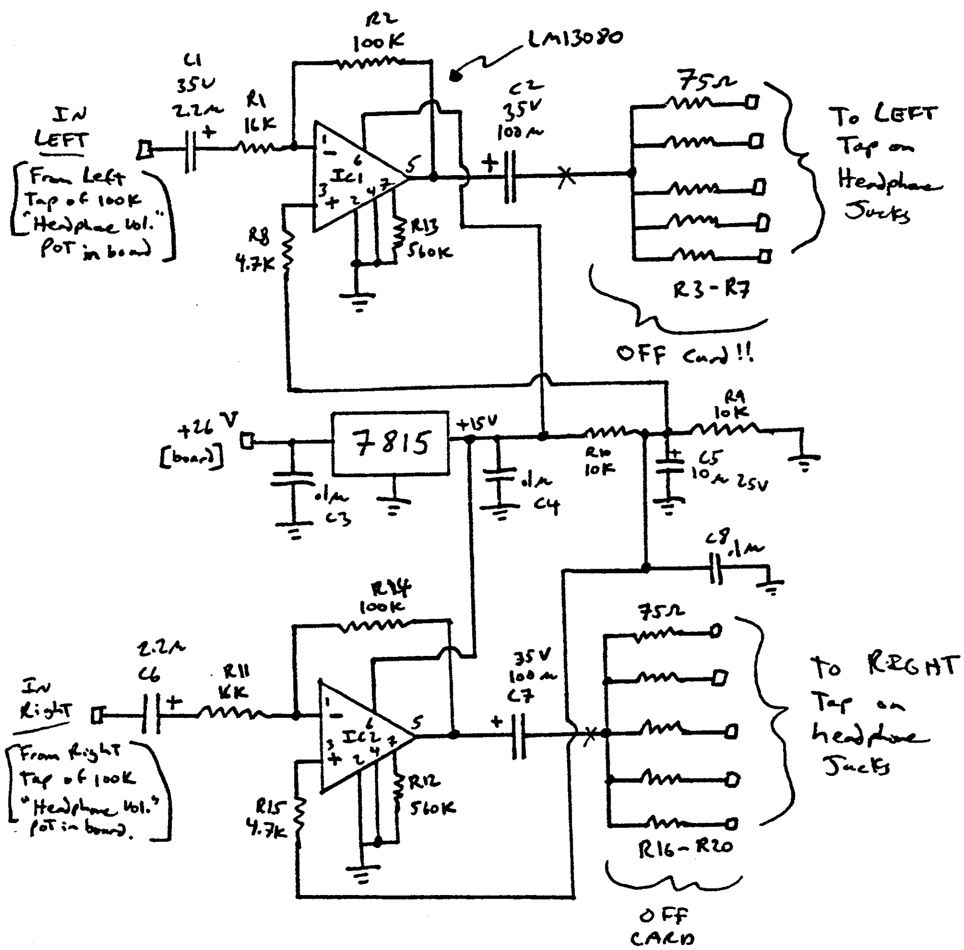
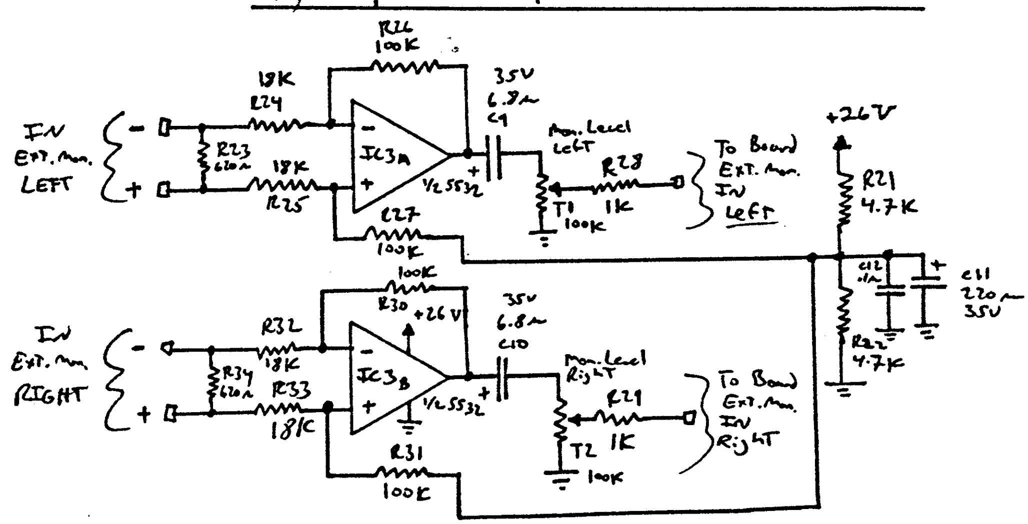
WMFO Studio A Kludge Cards

sheet #1 of 2



A) Headphone Amplifier

Note: Circuits A and B are located on LEFT hand Kludge card. More detail on inside cover of board.



B) External Monitor Line Receiver

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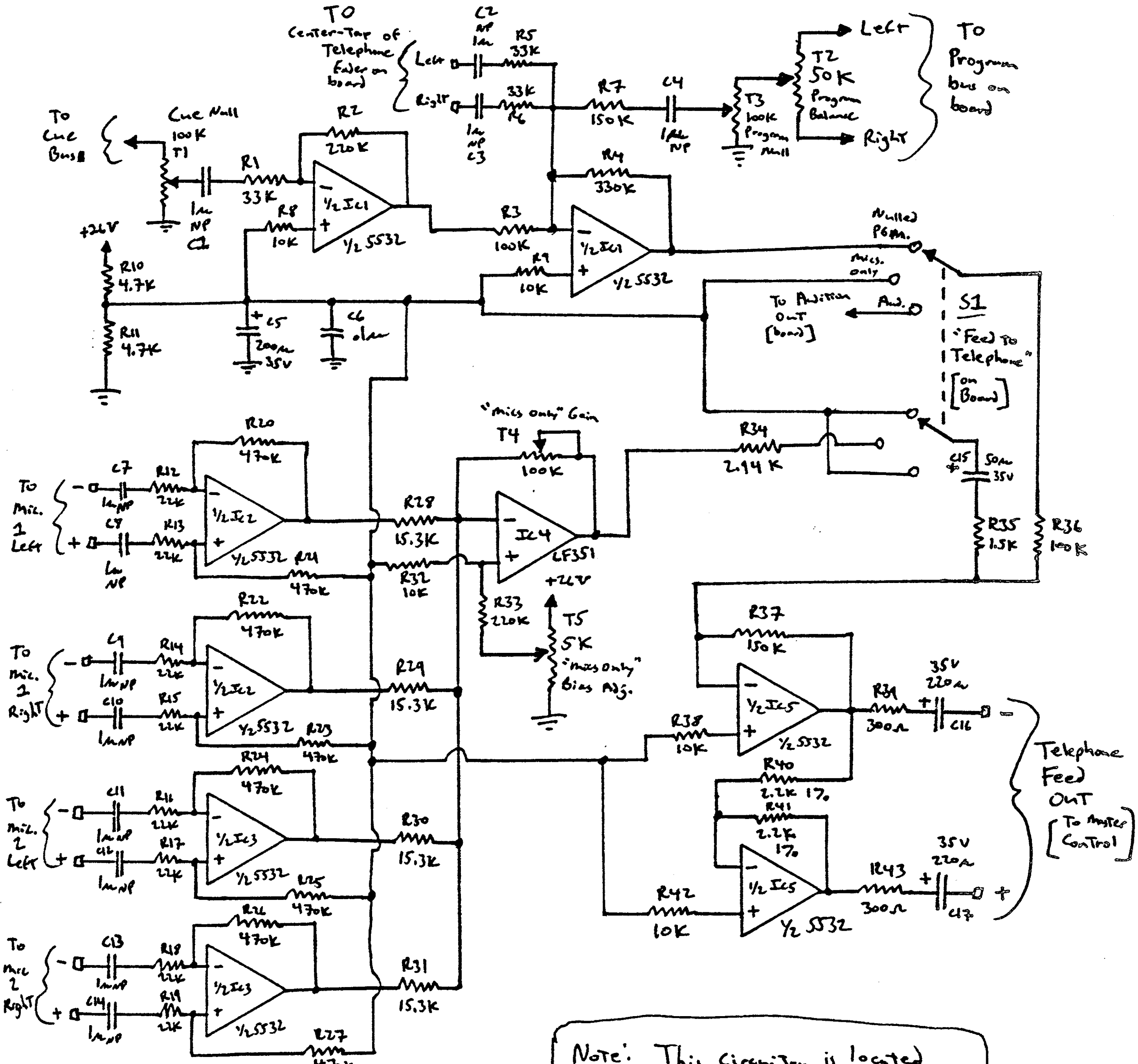
WMFO Studio A Kludge Cards [Sheet #2 of 2]



c) Studio A Telephone Feed Generator/Driver

Note: This circuitry is located on the right hand Kludge Card. The location (+ nomenclature) of all adjustments are given on the inside cover of the board. All OP-Amps draw power from the +26 Volt board (LPB) supply and ground.

THE WMFO STUDIO C KLUDGE CARDS

INPUTS:

* Balanced External Monitor Feeds (L & R)
* Stereo Headphone Feeds (from headphone level pot and source selector switch)
* 2 Balanced microphone inputs
* Program Feeds (L & R)
* Center-Tap from Telephone fader (L & R)
* Cue Bus Feed
* +26 Volts from LPB board

OUTPUTS:

* Single-Ended External Monitor (L/R)
* Stereo Headphone Outputs (L/R)
* Telephone audio feed (mono)

THE WMFO STUDIO C KLUDGE CARDS

I) Function and Operation:

The phrase "Studio C Kludge Cards" refers to two handwired perfboards mounted inside of the Studio C mixing console. The purpose of these cards is to augment some of the functions of the LPB board. The card mounted at right (facing the board from front) contains a stereo headphone amplifier and stereo external monitor line receiver.

The headphone amplifier allows several sets of monitor headphones (of any impedance) to be plugged into the console. Two headphone jacks are provided below the console countertop (in front), and two headphone jacks are provided at the side of the turntable cabinet (next to the interview mic. jack). All of these jacks are driven from the headphone amplifier mounted on the right kludge card. The headphone source (ext. mon., program, cue) is selected by the appropriate knife switch on the console, and the headphone volume is adjusted via the console-mounted control.

The external monitor line receiver converts the balanced external monitor signal supplied to the studio into a single-ended line used by the LPB board. Connecting the balanced signal directly to the single-ended inputs caused considerable noise and crosstalk, hence the line receivers were installed.

The kludge card mounted at the left side of the console creates the audio feed supplied to the studio telephone. When working with the telephones, the "handset" switch mounted on the top of the telephone should be off, inhibiting audio feeds to and from the telephone handset. The handset should be removed from the switchhook, and the desired line depressed on the telephone selector buttons. Audio from the telephone line appears on pot #4; by potting this up, callers may be placed on the air. Pot #4 may also be placed in cue so callers can be auditioned before going live (eliminating the necessity of ever using the handset). The handset can be left off the hook throughout the duration of the show; to disconnect the caller, merely punch the button on the telephone corresponding to the unused line (presently line #4). Make sure the telephones are potted down on the board before doing this in order to avoid a blast of noise and potential feedback.

The Studio C setup allows one of two signals to be fed down the telephone line to the caller. The telephone source is determined by the setting of the "Feed to Telephone" toggle switch on the console. If this switch is in the "mics. only" (up) position, an audio feed from all 3 microphones in the studio (1 on pot #1 [ie. DJ] and 2 on pot #2) is continuously fed down the selected telephone line (irrespective of whether any mics are on program or potted up). This is the simplest way of doing a telephone interview, and perhaps the most commonly applied @ WMFO. Using the "mics. only" feed, one can easily preview callers in cue without ever having to use the handset, as mentioned above.

If the telephone selector switch is set to the "off" (center) position, no audio is fed down the telephone line (this allows one to use the telephone and handset in a conventional fashion.

If the telephone selector switch is set to the "Pgm." (down) position, the console's program output is sent down the telephone line to the caller. Since the signal from pot #4 (which is defaulted to the telephone) is subtracted from the caller's feed, the caller may be potted up on program without feedback while all other program material is still sent down the telephone line. This is perhaps the most convienent means of using telephones in a music/talk context (ie. "Name That Tune"); whatever is being played over the air in program (except the caller's voice) is sent to the caller. This option has two minor hitches, however. Because of inherent difficulties in subtracting out the caller's voice from the telephone feed, feedback and "howling" can result when the caller is potted up excessively high. This is generally not a problem, since enough feedback-free range exists for most telephone connections. The problem may become more evident when the caller is at a very low level (ie. poor telephone connection), causing more gain to be needed. In these cases, one should resort to the "feedback-free" mics. only feed. Another quirk of this "Pgm." feed is that anything on cue will also be injected into the telephone line (except for cue on pot #4). Thus, if you're cueing up a record while talking to a caller under the "Pgm." feed, the caller (and potentially the listening audience) will hear the record being cued. Again, use another type of telephone feed if this becomes a problem.

Because of the way in which pot #4 is subtracted from the telephone feed under the "Pgm." setting, the telephone output can not be patched to another fader when this feature is being used (otherwise terrible feedback will immediately result). The telephone output may be re-routed to any fader when the Mics. Only feed is used.

The outputs of the telephone feeds from each studio are routed to the Master Control patchfield, where they are normalled to the audio inputs of the Symmetrix telephone interface units. These audio feeds may be re-directed in Master Control for other purposes; for instance, the "mics. only" feed can provide a direct line-level microphone output from each studio which is independent of the mixing consoles. This feed has proven to be ideal for espionage, vocal effects processing (ie. real-time pitch shifting), etc.

Symmetrix TI-101 telephone interfaces are used to connect to the telephone lines. There are three of these units mounted in the blue interstudio rack in Master Control. Each of these buffers the telephone line selected by the telephone in each studio (ie. one unit is dedicated to Studio A, another is dedicated to Studio B, and a third is dedicated to Studio C). The telephone line selected is the line punched up on the studio telephone. Several lines may be punched up simultaneously on these telephones (they've been specially modified; try punching two buttons at once); this enables the lines to be "conferenced".

The WMFO intercom line may also be used with the telephone interface in the same fashion. One can as well conference the intercom to an outside line (this is useful if you are located at a remote site [ie. transmitter room], and need to make an outside call on the intercom).

The Symmetrix TI-101 buffers the audio feed to the telephone line, and also creates an audio output from the telephone to the studio. This audio output supposidly "subtracts" the input audio feed (ie. the DJ's voice) from the output feed in true "hybrid" fashion. This does not work too well, however, hence our Symmetrix TI-101s have been modified to work also as "phone switches", where the amplitude of the DJ's voice attenuates the telephone output. This can create potential conflict when both parties try to speak simultaneously, however this problem has not surfaced under operation, and the "switching" action of the telephone interface enables a very crisp, clean, and professional-sounding telephone show to be conducted.

II) Circuit Description:

The circuitry on the right-hand kludge card is depicted in the first page of the schematic. The headphone amplifier (A) is based around two LM13080 power OP-AMPs (IC1 & IC2). The 7815 regulator drops the 26 Volt LPB supply down to 15 Volts, which is digestable by the LM13080's. Since they operate off a single-ended supply, the OP-AMPs must be biased up (R1,R2) and DC isolated from the outside world (C3,C4,C5,C6). The output of the headphone amplifier essentially comes right off of the OP-AMPs; 75-Ohm buffering resistors are inserted in series with the headphone outputs at each headphone jack (these resistors are not mounted on the card). Because these resistors effectively isolate each headphone port, headphones of any impedance may be plugged in without disturbing the other sets already connected.

The lower portion of this diagram (B) shows the external monitor line receiver circuitry. This is a standard pair of differential amplifiers built around a 5532 dual OP-AMP. Because of the single-ended supply, all OP-AMPs are biased up via R11/R12, and outputs are DC isolated (C9 & C10). The output of left and right channels may be independently trimmed via T1 & T2.

The telephone interface circuitry (located on the left kludge card) is depicted in the second page of the schematic. Two differential amplifiers based around a 5532 dual OP-AMP (IC1) pre-amplify the signals from both sets of studio microphones (2 interview [which are padded together] and 1 DJ). These amplifiers are fairly high impedance (22 KOhm to ground on inverting inputs), thus they are parallel jumpered with the console mic lines (at the input terminals for faders 1 and 2) with negligable effect on the audio quality of the console microphone feeds. The outputs of IC1 are passively summed via R11 & R12. When the selector switch (S1) is set to "Mics. Only", this sum is additionaly amplified and converted to a differential balanced signal by driver IC3. When S1 is set to "OFF", the input to IC3 is held at the common bias voltage, and no audio is passed.

The center-tap of the telephone fader (Pot #4) is subtracted from the program output in IC2b (these signals are inverted with respect to one another, thus they are simply summed). The left/right program balance is adjusted via T3, and the program level is set via T2 such that the signal from pot #4 is completely nulled. Because the input audio is also routed to the fader center-tap when it is put into cue, the cue bus output must as well be subtracted from the fader signal to enable fader #4 to be used in cue. IC2a inverts the cue bus for this subtraction, and T1 adjusts the cue amplitude such that the cue feed of pot #4 can be completely canceled.

Because of the single-ended supply voltage (all circuitry in the telephone driver operates off the LPB 26 Volt supply), all OP-AMPs in this circuit are biased up at 13-Volts (via R1 & R2), and all inputs and outputs to this circuit must be DC isolated via coupling capacitors.

III) Calibration and Adjustment:

First, I'll outline the calibration of the right-hand card (first page of schematic). The Headphone amplifier needs no calibration. The external monitor levels may be set by adjusting T1 & T2. With a good, robust audio source sounding over the air, place the board into "Ext." monitoring, set the monitor level at mid-pot, and tweak T1 & T2 until a comfortably loud, left/right balanced audio eminates from the monitor speakers. The External monitor line driver and distribution amplifier should already be calibrated before attempting this adjustment; see their associated writeups for details.

I'll now outline the calibration of the left-hand card (second page of schematic). Several adjustments must be made to set up the telephone driver circuitry. The amplitude of the telephone feed is determined by the corresponding settings on the Studio C Symmetrix TI-101 telephone interface in Master Control; this should be adjusted properly, essentially as discussed in the Symmetrix literature (remember that we have had these devices modified to act as phone switches; perhaps Symmetrix should be consulted for details if it becomes necessary to tweak these units [they should be OK as presently set-up]).

The discussion below describes how to calibrate the nulled program feed. First, pot up the house tone oscillator over program (set it to read zero dB on the meters; don't use fader #4). Adjust T3 to yield an equal left/right balance at its output. This can be done in several ways; one is to invert one channel of the program source (ie. plug the house tone through the Interface before it is input to the board, and hit "right channel invert". Use only the mono output, and make sure it is perfectly balanced [patch it through a mono bridge to insure this]). Monitor the output of the telephone feed on another device (ie. patch the Studio C telephone input feed from Master Control into a tape deck, and listen to it with headphones). Set T2 to maximum, and adjust T3 until the house tone is nulled (ie. dissapears from the headphone audio). Next put both house-tone channels back into phase (ie. flip the "R-Ivt." switch on the interface back to "Nrm.", or disconnect the interface entirely). Patch the house-tone (or interface output) into fader #4, and put it into program at half-pot. Adjust trimmer T2 such that the house-tone component in the telephone feed (monitored over the headphones) is at a minimum. The house-tone feed is entirely removed when T2 is full off, however this extreme is undesired; another intermediate setting of T2 will produce the null, and this is what is called for.

Next, place pot #4 into cue, and adjust trimmer T1 such that the house-tone audio (which should re-appear over your headphones) is again nulled from the telephone feed.

The Studio C kludge cards are now calibrated. The telephone program null is affected by the setting of the master program gain control mounted inside the board. If this gain control is adjusted for any reason, the telephone program null must be entirely re-calibrated as discussed above. Always trim these nulls after playing with the master program gain!!

The location of all kludge card trim pots are labeled inside the top cover of the LPB board. Refer to these diagrams for assistance.

IV) Troubleshooting Hints:

The functions of the kludge card are discretely broken up; ie. headphone driver, external monitor interface, and telephone feed (the telephone feed circuitry is further subdivided into mics. and nulled program sections). The symptoms of defects (ie. which functions are inhibited) should point fairly directly to the vicinity of the problem. The kludge cards tie quite heavily into the innards of the console; if anyone monkeys with the console, these connections may be disturbed, causing an apparent malfunction.

If the nulled program feed seems to be feeding back and howling excessively, T1-T3 on the telephone card may need to be re-calibrated as discussed above.

The Studio C kludge cards have operated without a hitch for over 2 years. Hopefully there's lots more longevity in store. If another console is installed in Studio C, one should perform similar custom modifications to maintain the functions of these kludge cards. Perhaps the original kludge cards in the LPB board can even be transplanted.....

WMFO Studio C Kludge Cards

– 11-Nov-86
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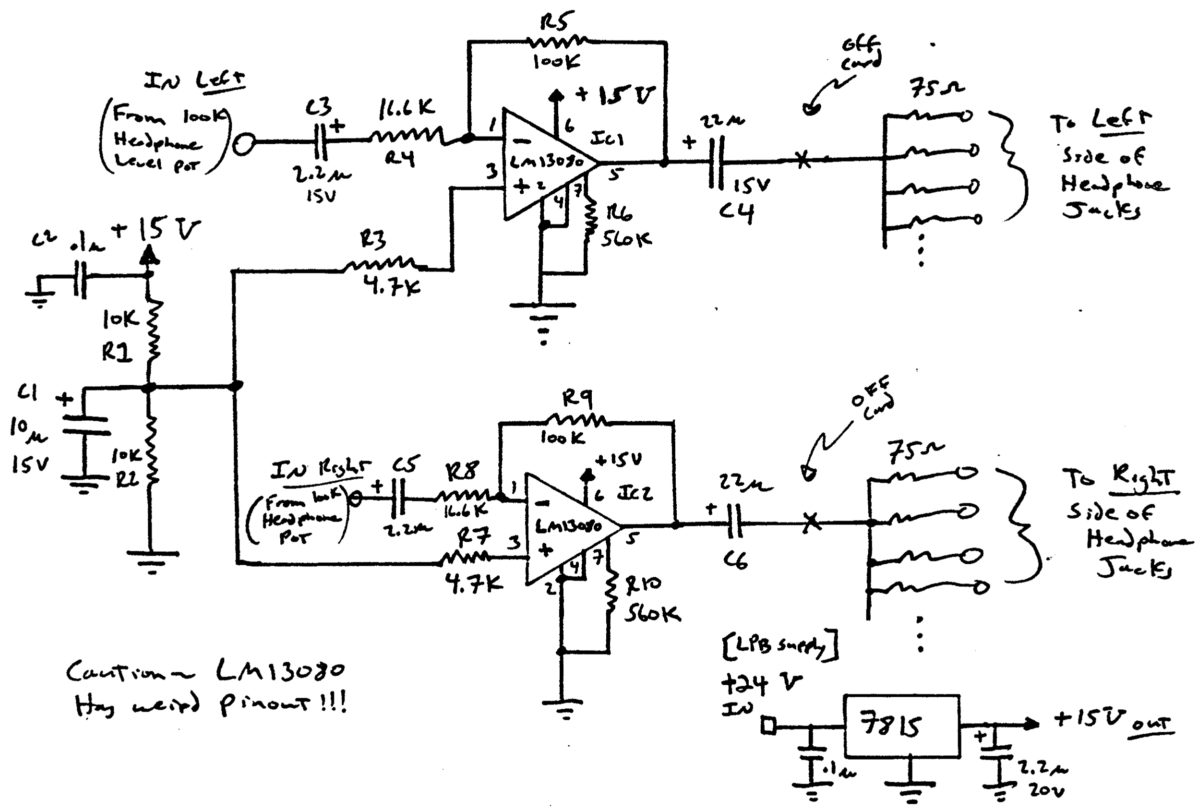
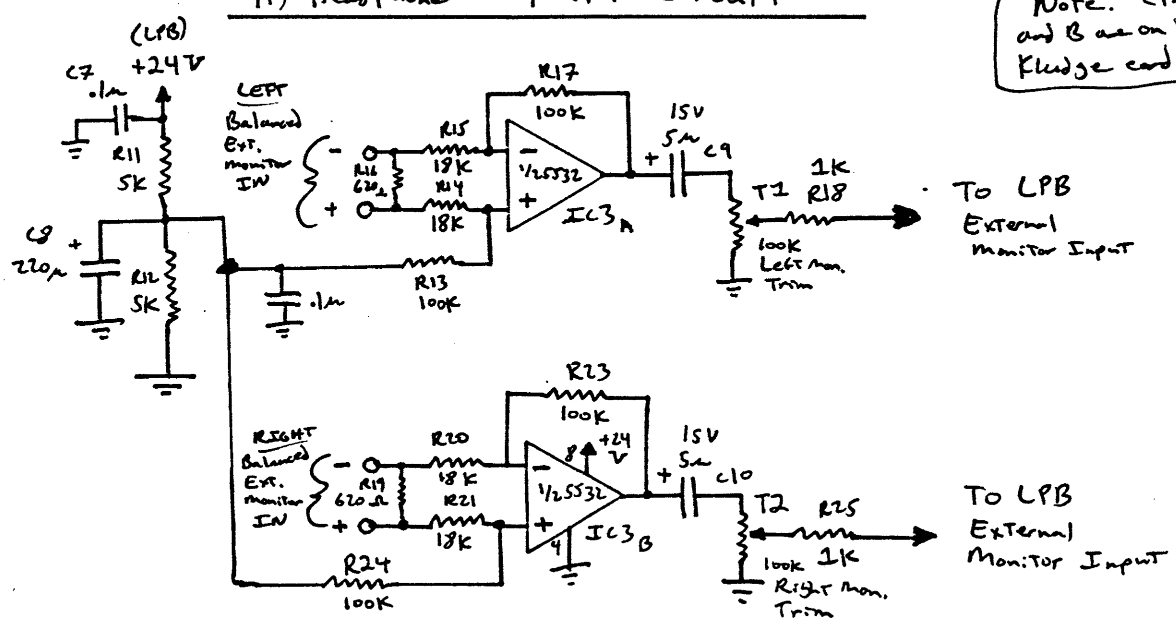
– Sheet #1 of 2



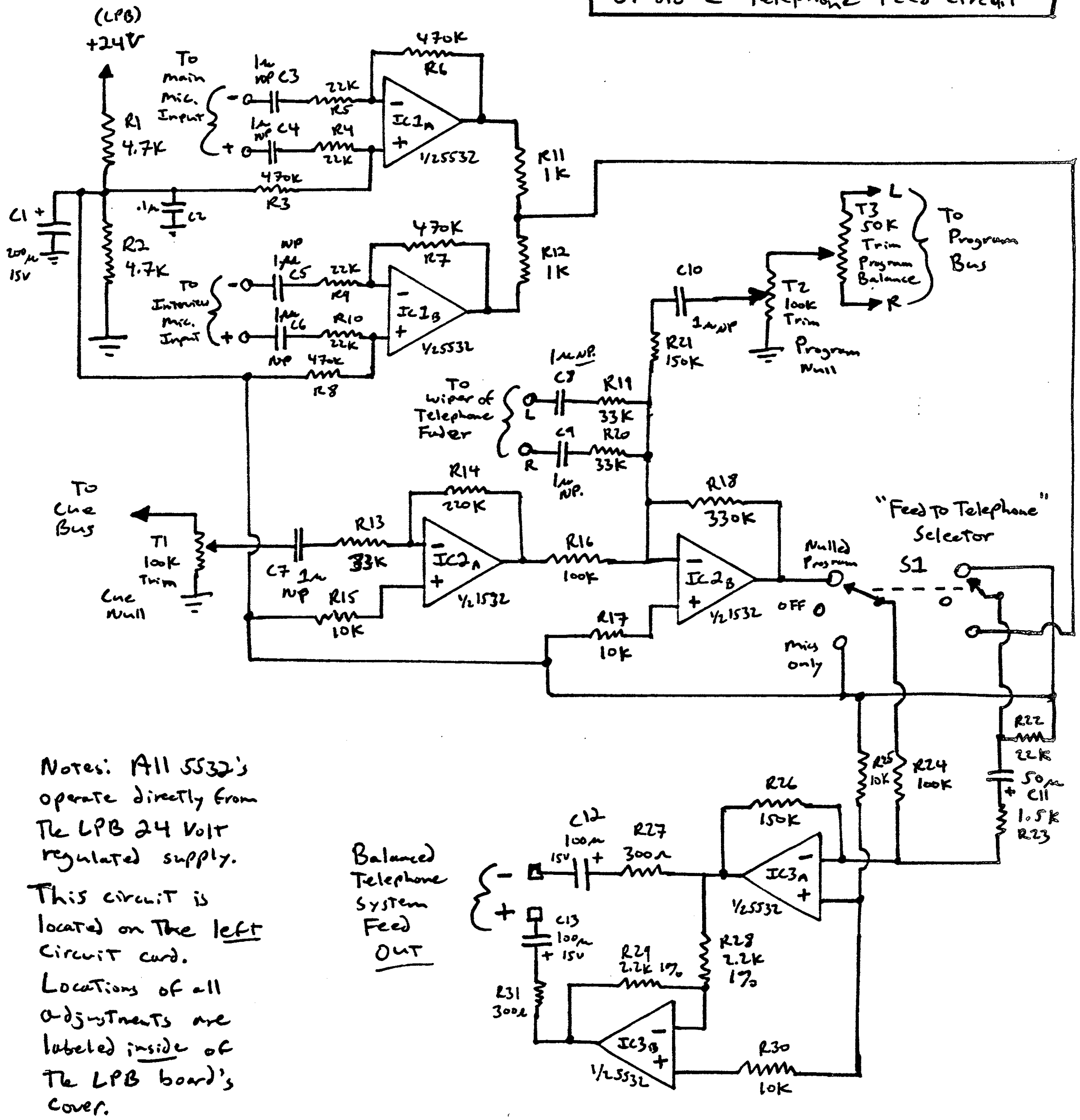
A) Headphone Amplifier Circuit

Note: Circuits A and B are on the Right Kludge card.



B) External Monitor Line Receiver

Studio C Telephone Feed Circuit



Notes: All 5532's operate directly from The LPB 24 Volt regulated supply.
This circuit is located on The left circuit card.
Locations of all adjustments are labeled inside of The LPB board's cover.

WMFO Studio C Kludge Cards ~ Sheet #2 of 2

— J. Paradiso
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THE WMFO STUDIO B KLUDGE BOX

INPUTS:

* Balanced External Monitor Feeds
  (L & R)

* Stereo Headphone Feeds
  (from headphone level pot and
  source selector switch)

* 2 Microphone preamplifier inputs

* Program Feeds (L & R)

* Center-Tap from Telephone fader
  (L & R)

* Audition Feed

* Single-Ended Cassette Deck Inputs
  (L & R)

OUTPUTS:

* Single-Ended External Monitor (L/R)

* Stereo Headphone Outputs (L/R)

* Balanced Cassette Deck Outputs (L/R)

* Telephone audio feed (mono)

THE WMFO STUDIO B KLUDGE BOX

I) Function and Operation:

The phrase "Studio B Kludge Box" refers to a black plastic box (with green pilot light) mounted beneath the counter supporting the Studio B console. The purpose of this box is to augment some of the functions of the Gates board. The card mounted inside this box contains a stereo headphone amplifier, stereo external monitor line receiver, cassette deck buffer/driver, and telephone feed generator/driver.

The headphone amplifier allows several sets of monitor headphones (of any impedance) to be plugged into the console. Two headphone jacks are provided on the front panel of the console, and two headphone jacks are provided at the side of the tape-deck cabinet (next to the interview mic. jack). All of these jacks are driven from the kludge box's headphone amplifier. The headphone source (ext. mon., program, cue) is selected by the appropriate knife switch on the console, and the headphone volume is adjusted via the console-mounted control.

The external monitor line receiver converts the balanced external monitor signal supplied to the studio into a single-ended line used by the Gates board. Connecting the balanced signal directly to the single-ended inputs caused considerable noise and crosstalk, hence the line receivers were installed.

The Cassette deck buffer/driver converts the high-impedance single-ended outputs of the cassette deck into balanced 600-Ohm lines which are compatible with the "professional" standard used at WMFO.

The kludge box also creates the audio feed supplied to the studio telephone. When working with the telephones, the "handset" switch mounted on the top of the telephone should be off, inhibiting audio feeds to and from the telephone handset. The handset should be removed from the switchhook, and the desired line depressed on the telephone selector buttons. Audio from the telephone line appears on pot #5; by potting this up, callers may be placed on the air. Pot #5 may also be placed in cue so callers can be auditioned before going live (eliminating the necessity of ever using the handset). The headset can be left off the hook throughout the duration of the show; to disconnect the caller, merely punch the button on the telephone corresponding to the unused line (presently line #4). Make sure the telephones are potted down on the board before doing this in order to avoid a blast of noise and feedback potential.

The Studio B setup allows one of three signals to be fed down the telephone line to the caller. The telephone source is determined by the setting of the "Feed to Telephone" rotary switch on the console. If this switch is in the "mics. only" position, an audio feed from both microphones in the studio (DJ and interview, both on pot #1) is continuously fed down the selected telephone line (irrespective of whether any mics are on program or potted up). This is the simplest way of doing a telephone interview, and perhaps the most commonly applied @ WMFO. Using the "mics. only" feed, one can easily preview callers in cue without ever having to use the handset, as mentioned above.

If the telephone selector switch is set to the "Aud." position, the console's audition mix is fed down the telephone line. This allows one to pot up music, voice, etc. on audition, and send it to the caller. Anything appearing only on program is not sent. By also potting up the audition line on program, the audition audio is directed both to the caller and on-air audience.

If the telephone selector switch is set to the "Pgm." position, the console's program output is sent down the telephone line to the caller. Since the signal from pot #5 (which is defaulted to the telephone) is subtracted from the callers feed, the caller may be potted up on program without feedback while all other program material is still sent down the telephone line. This is perhaps tne most convienent means of using telephones in a music/talk context (ie. "Name That Tune"); whatever is being played over the air in program (except the caller's voice) is sent to the caller. This option has one minor hitch, however. Because of inherent difficulties in subtracting out the caller's voice from the telephone feed, feedback and "howling" can result when the caller is potted up excessively high. This is generally not a problem, since enough feedback-free range exists for most telephone connections. The problem may become more evident when the caller is at a very low level (ie. poor telephone connection), causing more gain to be needed. In these cases, one should resort to the "feedback-free" mics. only or audition feeds.

Because of the way in which pot #5 is subtracted from the telephone feed under the "Pgm." setting, the telephone feed can not be patched to another fader when this feature is being used (otherwise terrible feedback will immediately result). The telephone feed may be re-routed to any fader when either the Audition or Mics. Only feeds are used.

The outputs of the telephone feeds from each studio are routed to the Master Control patchfield, where they are normalled to the audio inputs of the Symmetrix telephone interface units. These audio feeds may be re-directed in Master Control for other purposes; for instance, the "mics. only" feed can provide a direct line-level microphone output from each studio which is independent of the mixing consoles. This feed has proven to be ideal for espionage, vocal effects processing (ie. real-time pitch shifting), etc.

Symmetrix TI-101 telephone interfaces are used to connect to the telephone lines. There are three of these units mounted in the blue interstudio rack in Master Control. Each of these units buffers the telephone line selected by the telephone in each studio (ie. one unit is dedicated to Studio A, another is dedicated to Studio B, and a third is dedicated to Studio C). The telephone line selected is the line punched up on the studio telephone. Several lines may be punched up simultaneously on these telephones (they've been specially modified; try punching two buttons at once); this enables the lines to be "conferenced". The WMFO intercom line may also be used with the telephone interface in the same fashion. One can as well conference the intercom to an outside line (this is useful if you are located at a remote site [ie. transmitter room], and need to make an outside call on the intercom).

The Symmetrix TI-101 buffers the audio feed to the telephone line, and also creates an audio output from the telephone to the studio. This audio output supposidly "subtracts" the input audio feed (ie. the DJ's voice) from the output feed in true "hybrid" fashion. This does not work too well, however, hence our Symmetrix TI-101s have been modified to work also as "phone switches", where the amplitude of the DJ's voice attenuates the telephone output. This can create potential conflict when both parties try to speak simultaneously, however this problem has not surfaced under operation, and the "switching" action of the telephone interface enables a very crisp, clean, and professional-sounding telephone show to be conducted.

II) Circuit Description:

The circuitry for the headphone amplifier and external monitor line driver is depicted on the first page of the schematic. The headphone amplifier (1) is based around two LM13080 power OP-AMPs (IC1 & IC2). The kludge box power supply delivers +/-12 Volts; this is a bit higher than the 15 Volt maximum at which the LM13080 is rated, however these ICs have run without problem for over two years, and the overvoltage hasn't proven catastrophic. The output of the headphone amplifier essentially comes right off of the OP-AMPs; 75-Ohm buffering resistors (R4-R7 and R11-R14) are inserted in series with the headphone outputs at each headphone jack (these resistors are not mounted on the card). Because these resistors effectively isolate each headphone port, headphones of any impedance may be plugged in without disturbing the other sets already connected.

The lower portion of this diagram (2) shows the external monitor line receiver circuitry. This is a standard pair of differential amplifiers built around a 5532 dual OP-AMP. The output of left and right channels may be independently trimmed via T1 & T2.

The Cassette Driver (3) circuitry is depicted at the top of the second page of the schematic. Due to the limited space available on the kludge box's circuit card, these are not standard differential line drivers, but single ended buffers (with adjustable gain) which drive external isolation transformers used to balance the cassette outputs. These transformers are housed in a metal box which is mounted below the console countertop. This metal box also contains 2 additional transformers to buffer the cassette inputs; these transformers are inserted directly into the cassette input lines (between patchfield and cassette deck) without additional intervening circuitry.

The telephone interface circuitry (4) is depicted at the bottom of the second page of the schematic. Since the microphones are input directly to two microphone preamplifiers in the Gates board (without first being routed through the channel selection switches, as in the LPB boards), the microphone preamplifier outputs may be tapped directly, and used as the input to the "Mics. Only" telephone feed. When the feed selector switch (S1) is set to "Mics. Only", the output of these microphone preamplifiers are mixed and converted to a differential balanced signal by driver IC5. When S1 is set to "Aud.", the single-ended audition output from the console is routed to the balanced driver (IC5) via IC6, thereby sending the audition channel down the telephone line.

The center-tap of the telephone fader (Pot #5) is subtracted from the program output in IC5a (the program signals are inverted by IC6 so they can be subtracted). The left/right program balance is adjusted via T6, and the program level is set via T5 such that the signal from pot #5 is completely nulled.

III) Calibration and Adjustment:

First, I'll outline the calibration of the circuitry on the first page of the schematic. The Headphone amplifier needs no calibration. The external monitor levels may be set by adjusting T1 & T2. With a good, robust audio source sounding over the air, place the board into "Ext." monitoring, set the monitor level at mid-pot, and tweak T1 & T2 until a comfortably loud, left/right balanced audio eminates from the monitor speakers. The External monitor line driver and distribution amplifier should already be calibrated before attempting this adjustment; see their associated writeups for details.

I'll now outline the calibration of the circuitry on the second page of the schematic. In order to set the cassette driver level, play a tape recorded at an ideal level (ie. house tone recorded at zero dB), adjust the output level of the cassette deck to its midpoint (ie. "5"), and pot the cassette up on program at half-pot. Adjust T3 & T4 until both channels meter at zero dB on the console.

Adjustments to set up the telephone driver circuitry (lower portion of schematic) are detailed in the following text. The amplitude of the telephone feed is determined by various settings on the Studio B Symmetrix TI-101 telephone interface located in Master Control. This unit should be calibrated as discussed in its associated literature. Remember that these units have been modified to act as phone switches (this mod. was made by Symmetrix themselves), thus the front panel controls may not all act as expected. They are calibrated properly at present, and should not need further adjustment unless something drastic occurs in the telephone lines or telephone feed.

The only kludge box calibration necessary for the telephone feed to operate properly is the adjustment of the program nulling circuitry. First, pot up the house tone oscillator over program (set it to read zero dB on the meters; don't use fader #5). Adjust T6 to yield an equal left/right balance at its output. This can be done in several ways; one is to invert one channel of the program source (ie. plug the house tone through the Interface before it is input to the board, and hit "right channel invert". Use only the mono output, and make sure it is perfectly balanced [patch it through a mono bridge to insure this]). Monitor the output of the telephone feed on audition (ie. patch the Studio B telephone input feed from Master Control into a fader on the Studio B board (other than fader #5), and put it into audition). Set T5 to maximum, and adjust T6 until the house tone is nulled (ie. dissapears from the audition audio). Next put both house-tone channels back into phase (ie. flip the "R-Ivt." switch on the interface back to "Nrm.", or disconnect the interface entirely). Patch the house-tone (or interface output) into fader #5, and put it into program at half-pot. Adjust trimmer T5 such that the house-tone component in the telephone feed (monitored over audition) is at a minimum. The house-tone feed is entirely removed when T5 is full off, however this extreme is undesired; another intermediate setting of T5 will produce the null, and this is what is called for. The Studio B kludge box is now calibrated.

The location of all trim pots are labeled inside the top cover of the kludge box. The location of all ICs and trimpots are also depicted in the third page of the schematic diagram. Refer to these figures for assistance.

IV) Troubleshooting Hints:

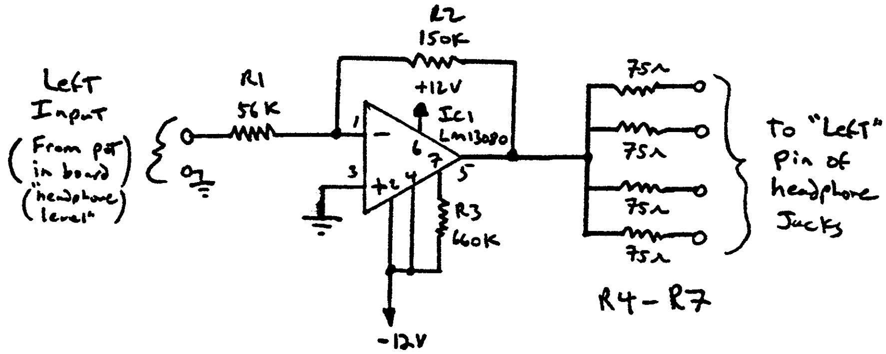
The Kludge Box performs many important services in Studio B. It provides headphone outputs, buffers the external monitor inputs, buffers the cassette deck output, and generates the telephone feed. If none of these functions are working, a global problem exists within the kludge box. Check the green pilot light mounted at the front of the unit (under the counter in Studio B). If this light is dead, the device is not receiving 110 Volt line current. Make sure it is plugged in properly. Check the fuse. If some extravagant loading problem inside the kludge box persists in blowing repeated fuses (don't raise the fuse rating!), check the kludge box power supply, and look for shorted OP-AMPs. Remember that we are running the LM13080's nearly 10 Volts over their recommended maximum rating; these have given no problem over the years, but are a prime suspect if problems evolve. If these are at fault (and you don't want to continue using LM13080's) one might consider using another type of power OP-AMP here; just remember that the LM13080 pin-out is not conventional, and the socket may need to be re-wired if a substitution is attempted.

The functions of the kludge box are discretely broken up; ie. headphone driver, external monitor interface, cassette driver and telephone feed (the telephone feed circuitry is further subdivided into mics., audition, and nulled program sections). The symptoms of defects (ie. which functions are inhibited) should point fairly directly to the vicinity of any problem. The kludge box ties quite heavily into the innards of the console; if anyone monkeys with the console, these connections may be disturbed, causing an apparent malfunction.

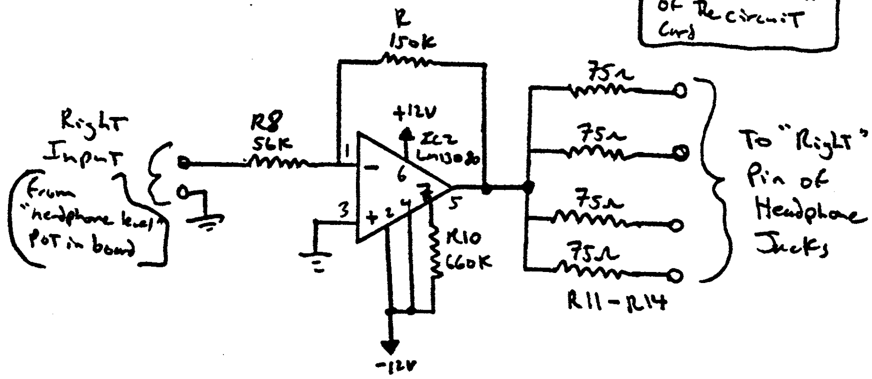
If the telephone program feed seems to be feeding back and howling excessively, T5 & T6 may need to be re-calibrated as discussed above.

The Studio B kludge box has operated essentially without a hitch for over 2 years. The only problem has been a manually dislodged fuse (perhaps by a roudy trainee), which killed all kludge box functions entirely (once the fuse was replaced, the unit again worked fine). Hopefully there's lots more longevity in store. If another console is installed in Studio B, the kludge box could still be used if some simple custom console modifications were performed.

Studio B "Kludge Box" ~ Sheet #1 Dec, 1986 – J. Paradiso

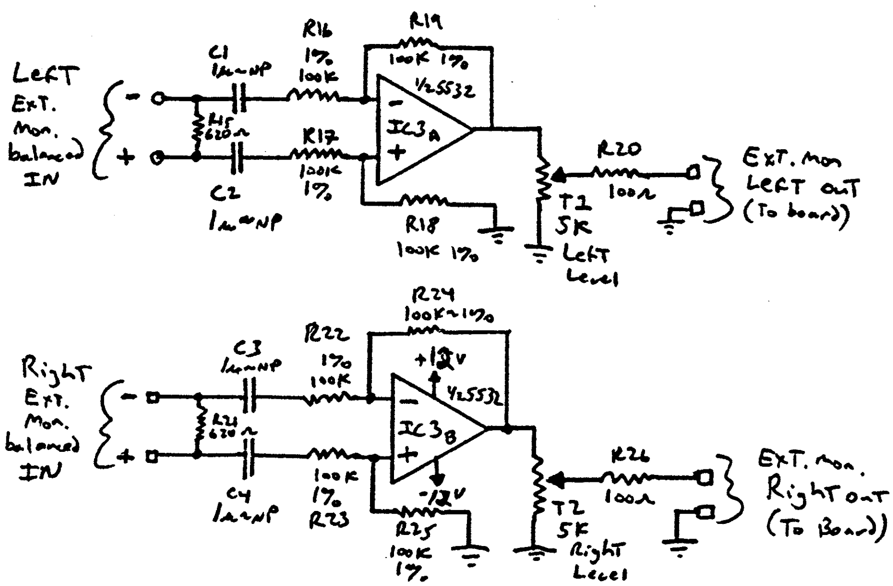


Note: R4–R7 and R11–R17 are located off of the circuit card



Caution! The LM13080 is rated only at a 15 volt supply voltage (Here they're run at 24 volts). When replacing these chips, certain individual devices may not operate properly @ the 24 volt potential. See text for details....

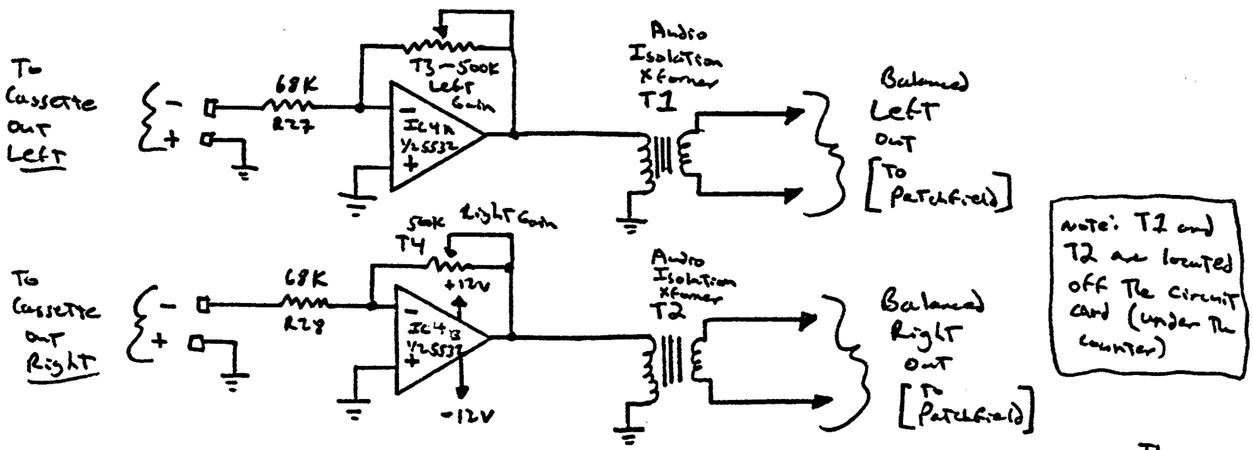
1) Headphone Driver



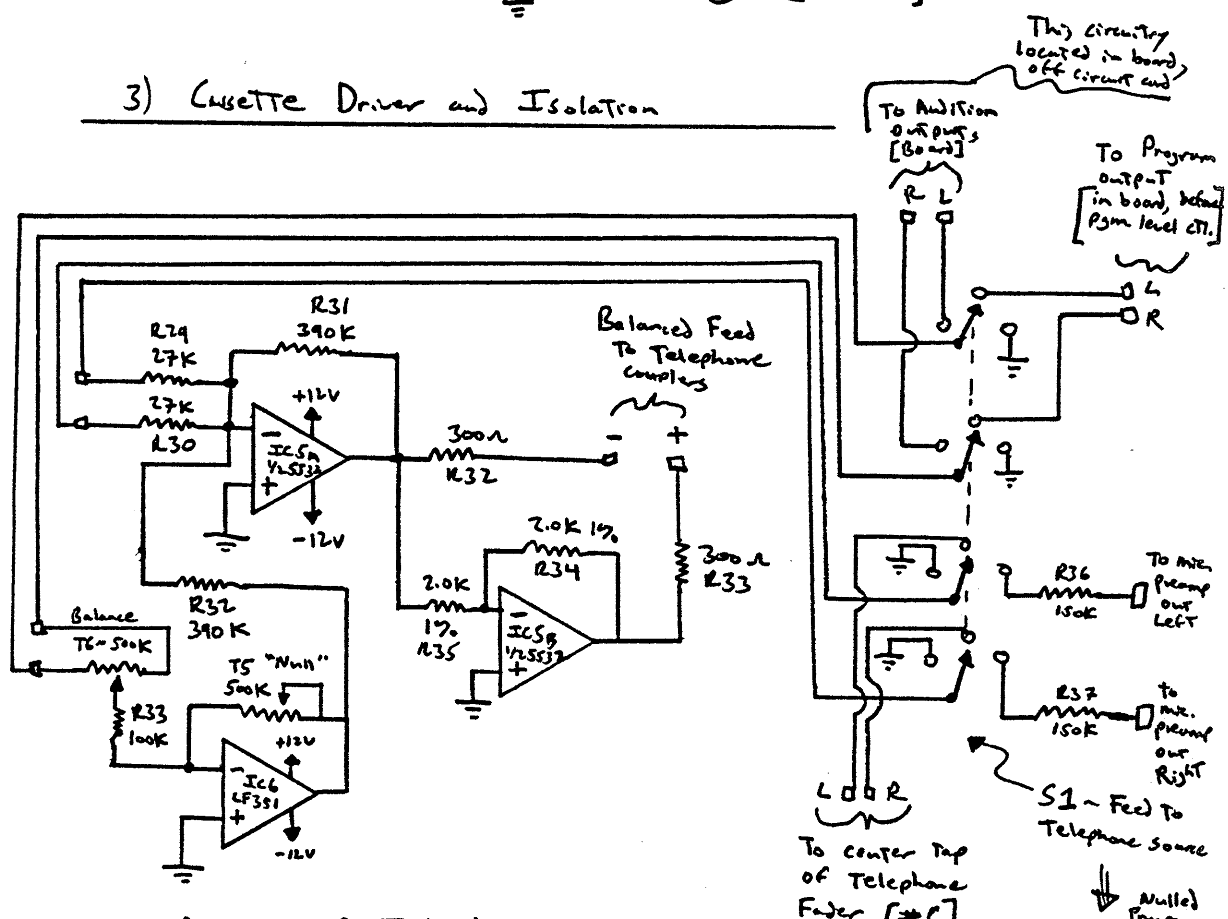
2) Ext. Monitor Line Receiver

WMFO Studio B "Kludge Box"

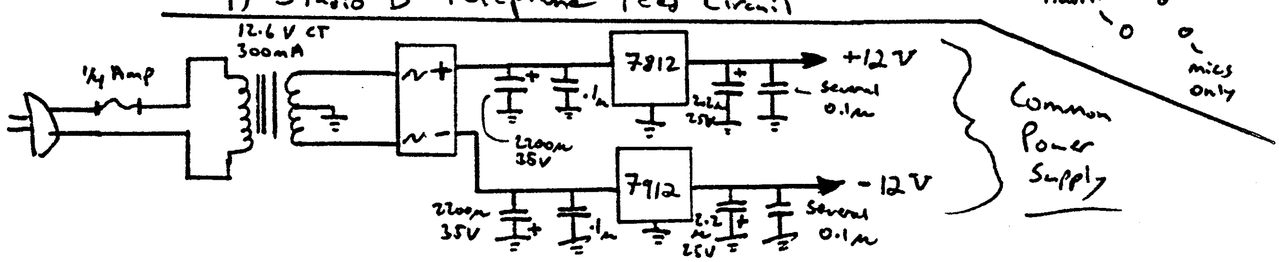
Sheet #2
Dec., 1986 – J. Paradiso



3) Cassette Driver and Isolation

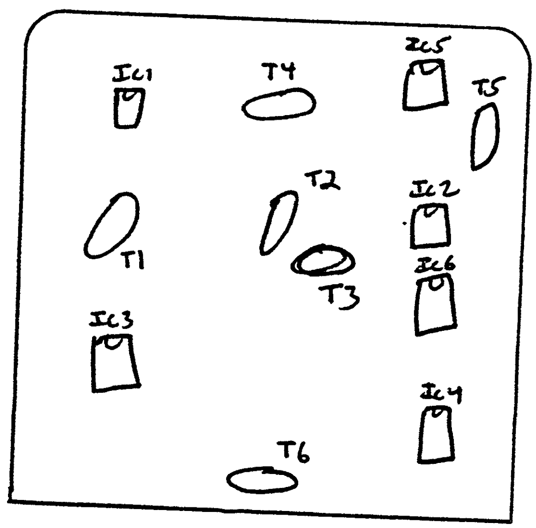


4) Studio B Telephone Feed Circuit



WMFO Studio B "Kludge Box"

Sheet #3
Dec., 1986



Circuit Card layout.

[See inside cover of box (located under counter in Studio B) for more detail...]

THE WMFO TRANSMITTER AUDIO AND POWER MONITOR

INPUTS:

* WMFO Intercom line input

* Antenna input for FM tuner

* Ext. audio line inputs (L & R)
  (panel RCA jacks)

* Transmitter current tap input
  (0.3 Volt differential at 50 Volt common mode)

* Transmitter voltage sense input
  (standard tap off transmitter)

OUTPUTS:

* Speaker Outputs (L & R)

* Stereo headphone output
  (panel jack)

* Audio line outputs (L & R)
  (panel RCA jacks)

* Transmitter current sense voltage out (barrier strip on rear)

* Transmitter power monitor voltage out (barrier strip on rear)

* Transmitter power monitor voltage out (test terminals on front panel)

* Drive voltage output for remote Optimod stereo/mono switching

THE WMFO TRANSMITTER AUDIO AND POWER MONITOR

I) Fundamentals and Operation:

The Transmitter Audio and Power Monitor is a device mounted in the WMFO rack located at the transmitter site on the fourth floor of Ballou Hall. This device serves several purposes. It contains a FM/AM stereo tuner and amplifier connected to two speakers. This allows one to monitor WMFO in reasonable quality stereo; an important and necessary feature when making transmitter adjustments and tweaking the airchain components. A panel-mounted headphone jack is also provided for precise stereo monitoring. Two RCA jacks provide the capability of introducing an external single-ended line input into the amplifier, and two other RCA jacks provide single ended line output taps. These line outputs allow one to monitor the transmitted audio with an oscilloscope or other piece of test equipment. The auxilary inputs allow one to use the amplifier & speakers to monitor an alternative audio source (a switch on the front panel cuts off the tuner feed). The tuner can, of course, be tuned to any other station when necessary; this can provide much needed entertainment and noise to break the monotony of working on the transmitter in Ballou Hall during the evening.

The front panel of this unit contains two volume controls (one each for left and right channels) and two toggle switches (one to mute the speakers, and another which cuts the tuner feed). The FM/AM tuner is also mounted on the front panel; its operation is fairly conventional.

When the intercom line is "rung" to get the attention of the Ballou crew (the "Intercom Phone Ringer" which does this is mounted on the grey rack in Master Control; see its associated writeup for details), an annoying "tone" will eminate from the speakers (provided that they are switched on). This is to insure that folks in the studio can gain the attention of the Ballou crew, even when the tuner audio is potted up at appreciable volume.

Another major function of this device (one which has very little interface with the front panel) is to monitor the transmitter output power. A specially-installed tap into the QEI transmitter measures the transmitter's output stage current (the current tap provided by QEI was drifting, thus this alternative source was added), and another tap (QEI-installed) is used to measure the output stage voltage. By multiplying these two signals together, this device derives a voltage which is proportional to the power output from the transmitter (ideally....). This power monitor voltage is routed to the remote metering system, and currently may be read on channel #7. It also appears on terminal posts mounted on the front panel to aid in calibration and diagnostics. Another monitor voltage (which is proportional to the transmitter's output stage current) is also generated by this unit. This signal may be monitored by the remote metering system as an alternative to the current tap provided by QEI in the event of problems (this is not presently used; the QEI tap is employed, and appears on Channel #2 of the remote meter).

Because of problems with the QEI current consumption, the linearity of these current and power monitor voltages is somewhat questionable. In order to develop a more reliable feed, the directional coupler in the QEI transmitter was also tapped to feed the remote meter. This feed directly measures foward transmitter power, and appears on remote channel #3. It is calibrated to yield "1000" on the meter at the maximum allowed operating power of WMFO (72 Watts presently [1987]). The "calculated" transmitter output power appearing on remote channel #7 (produced by this unit as mentioned above) is also calibrated to yield "1000" at the FCC maximum of 72 Watts. The latter signal corresponds to the total power put out by the transmitter (and is accurate to better than 10% or so...). The former signal is the measured foward power feeding the antenna; it is used as the primary power measurement (it is regularly logged, and the transmitter is continually adjusted to keep it at 1000). The "raise" and "lower" buttons on the remote control console will raise and lower transmitter power on channels #2 and #3. The calculated power appearing on remote channel #7 is used as an additional transmitter check and backup reading.

Another tap from the transmitter's directional coupler measures reflected power. This is also read by the remote meter (appearing on channel #4), and is calibrated in Watts x 100. If it reads above 200 or so, the reflected power is running a bit high. Since it is a function of transmitter and antenna cabling, weather conditions, etc. it is impossible to adjust without performing repairs. It should also be logged, since it affects the air signal quality considerably. The "forward/reverse" switch on the QEI transmitter should be left in the indicated position to maintain the calibration of the forward and reflected power as monitored on the remote meter.

This unit also provides a voltage which is used to enable remote switching of the Optimod stereo/mono functions. The Optimod stereo pilot is monitored on remote channels #5 and #6. It is calibrated to read 1000 when transmitting in stereo, and nearly (or exactly) zero when transmitting in mono. The "raise" and "lower" controls on remote channels #5 and #6 will put the station into stereo (raise) or mono (lower). Lowering channel #5 puts the Optimod into "mono left" mode, and lowering channel #6 puts the Optimod into "mono right" mode (beware; my memory may be backwards....). These options might be needed if one of the telephone lines feeding audio to Ballou Hall goes bad (the station output may be bridged to mono via a front panel switch on the Studio Switcher).

Since this device provides these voltages for monitoring transmitter operation and switching the Optimod, it should normally be left on. The tuner feed should be switched off when the crew leaves the Ballou area (otherwise the omnipresent audio will certainly annoy the staff that works there during the day).

II) Circuit Details:

All circuitry is depicted in the 3-page schematic diagram. The first page shows the audio circuitry used to amplify the tuner signal and phone ring. IC1 is a telephone ringer circuit, which produces an anoying "ping/blast" at its output when a ring voltage is applied on the intercom line input. Data sheets on IC1 are appended to this writeup. The general quality of this sound may be adjusted by trimmer T1. Transformer T1 isolates the intercom ground from the local Ballou ground. Diode D1 aids in reducing RF pickup (the AM induced signal in Ballou Hall is immense!!! If your grounds aren't clean, AM stations show up over everything!). This "ring" audio is inverted and filtered via IC2, and fed to audio mixers IC3, which also mix outputs from the Aiwa TU-01 tuner and auxilary input jacks. The level of the "ring" signal which is applied to IC3 is trimmed via the "Dynamic Ring Trim" T4. Since the outputs of IC3 are scaled by the volume controls (P1/P2), this "Dynamic" ring level is the amount of ring audio which scales with the selected master volume. IC3 also provides the line outputs that feed the front panel RCA jacks. IC4 mixes audio from IC3 (scaled by P1/P2) with the "static" ring signal (which is a bias level at zero volume) as weighted by T5. The outputs of IC4 feed the integrated 5 Watt audio amplifiers IC5 & IC6. These are 2002's (data sheets are contained in the appendix to this document). Their outputs feed the speakers (via muting switch S2) and headphone jack (which will accept any impedance of headphone). Since the 2002's are very sensitive to power supply bypassing (they will oscillate at extremely high frequency and get quite hot if the bypassing isn't adequate), redundant supply shunting-capacitors are installed across the circuit card in this region.

The power supply for the Aiwa TU-01 tuner pack (which came from a former Walkman of mine) produces a stable 6 Volts by dropping the 15 Volt supply rail with a 7806. Switch S1 allows one to mute the tuner feed to the amplifier (thereby allowing exclusive monitoring of the external inputs).

The circuitry used in calculating the transmitter output power is given in page #2 of the schematic. The current monitor input is actually the voltage across a very small resistor placed in series with the output stage collector circuit (this resistor wasn't put in by me; it's standard in this QEI transmitter). The drop across this resistor is generally under 0.3 Volts at a 50 Volt common mode potential. QEI solved the problem of removing the common mode by employing a chopper stabilized amplifier. The resulting current monitor often drifts quite a bit, so the resistor leads were brought out of the transmitter (with 4.7K 1% series resistors buffering each side) to enable common mode isolation to be performed directly with the circuitry depicted in this schematic.

The input voltages are buffered by voltage followers IC9 & IC10 (both LM308 precision OP-AMPs). The 50-Volt common mode is dropped to within the 15 Volt input range of the OP-AMPs via dividers R1/R2 and R4/R5. IC12 (which is a 725 precision OP-AMP) is used as a differential amplifier to subtract the outputs of IC9 & IC10, thereby removing the common mode and amplifying the residual signal (which should be proportional to transmitter current). The differential balance in IC12's input is adjusted by trimmer T1. Trimmer T7 injects a small bias into IC12 which compensates for non-linearity in the QEI transmitter's output stage current-vs.-power characteristic (this nonlinearity seems to be significant!). Switch S3 (mounted behind the rack panel) allows this compensation voltage to be turned off for calibration.

The output of IC12 is additionally amplified by IC11 (again a LM308); this voltage appears on the rear barrier strip as an alternative "current sense" output that can be monitored by the remote meter (this tap may be useful if the chopper amplifier in the QEI transmitter starts to drift again).

The "voltage sense" output of the QEI transmitter is amplified in IC14. The output of IC14 (transmitter voltage) is multiplied with the output of IC11 (transmitter current) in the precision multiplier IC13 (this is an Analog Devices AD532H), yielding a voltage proportional to transmitter output power. T6 trims the quiescent multiplier offset to zero. The outputs of IC13 are routed to the front panel terminal posts (where the calculated power may be monitored for calibration) and the rear barrier strip (which connects to the remote meter). Data sheets on the AD532H are appended to this writeup.

All critical resistors in this portion of the circuit are 1% metal film. All OP-AMPs are selected for excellent DC characteristics. Because of the lousy QEI current tap, however, I don't trust these current and power outputs to extreme precision; I think they're better than 10%, but have never really tried to measure their stability. This power output has been somewhat obsoleted by the QEI directional coupler tap on channel #3. It provides a valuable backup and verification capability, however, and should be maintained. Remember, this calculated power signal measures power actually burnt in the transmitter output, while the directional coupler tap detects only power that flows to the antenna.

The third page of the schematic shows the power supply, which is mounted behind the rack panel. The +/-15 Volts is used to power all OP-AMPs. Since the 2002 audio amplifiers also use the +15 Volt supply, the 7815 regulator is a TO3 case heat sunk to the rack panel (where it never grows hot). The 7915 is a TO220 mounted on the circuit card. The circuit layout is diagrammed in the lower portion of the page; all ICs and trimpots are specified.

III) Calibration and Adjustment:

All trimpot locations are depicted on page #3 of the schematic. The first adjustments to be made set the intercom ring levels and sound quality. Have someone in Master Control hold the button down on the phone ringer (don't ring for over approx. 30 sec. without waiting a few seconds for the phone ringer to recover; it wasn't built for continuous operation). Adjust Trimmer T1 to yield the most acceptable ring sound (make sure T5 and T4 are potted up and speakers are switched on). The phone ring seems to be flaking out these days; it might sound more like a "pop" rather than a "ring". Adjust T1 to your preference, although you might not have much range. With the volume controls on the front panel (P1 & P2) all the way down, set T5 to yield an acceptable ring level. Next, turn the tuner feed on, pot up the volume controls (P1 & P2) to yield a comfortable listening level, and adjust trimmer T4 to obtain a ring level that can easily be heard over the music.

The remaining calibration tunes the transmitter output power circuitry. Turn switch S3 (mounted behind the panel) off. Ground the + and - inputs to the current monitoring circuitry. These inputs appear on the barrier strip mounted behind the rack panel. They are clearly labeled. Disconnect the taps from the transmitter before grounding these inputs. Measure the output voltage of IC11 with a voltmeter (this output is brought to the rear barrier strip as "I out"). Adjust IC12's offset trim T2 to yield zero volts at this output. Next, remove the + and - inputs to the current monitor from ground, and connect them both up to one of the taps from the QEI transmitter (which should be sitting at 50 Volts with the transmitter on). Adjust the common mode trim T1 to again yield zero volts at "I out" (ie. output of IC11 on barrier strip). Disconnect the transmitter taps, and once more put both current monitor inputs back at ground. Monitor the output of IC13 (which is available at the terminal posts on the front panel) with the voltmeter. Adjust output offset trimmer T6 to yield zero volts here.

Next, re-connect both transmitter current taps to the current monitor in the conventional configuration (if you get + and - reversed, the output voltage of IC11 will be negative!). With the transmitter powered up (it should have been powered up during this entire calibration), measure the output of IC13 at the panel-mounted terminal posts. Turn the FM exciter off (but leave the transmitter on!). The transmitter output power and current should drop to zero, as seen on the meters mounted on the transmitter. Turn switch S3 on and adjust trimmer T7 to again yield zero volts at the terminal posts. Leave S3 on; it should be in this position during normal operation. The Transmitter Audio and Power Monitor is now calibrated.

The power monitor voltage range will scale with the amplitude of the transmitter voltage sense output. This output is adjustable at the rear of the QEI transmitter (it is a screwdriver-tune pot). If the power monitor voltage is too small or too large (it should measure around 4-6 Volts), adjust this screwdriver pot to yield a better value. Remember, if this pot is moved, the remote meter channel #1 must be re-calibrated.

After adjusting the power monitor circuitry, channel #7 of the remote monitor must be calibrated to give a reading of 1000 at an output power of 72 Watts.

The transmitter output power may be measured directly via the panel-mounted inline Bird Wattmeter. This meter (with slug set to monitor forward power) should read at 72 (that's our limit in 1987; it may eventually change if WMFO ever gets a power increase). If it's too high or too low, the exciter output power may be tweaked to return it to 72. Make sure that the remote adjustment on transmitter power is at mid-range before attempting to change the exciter output.

IV) Troubleshooting Hints:

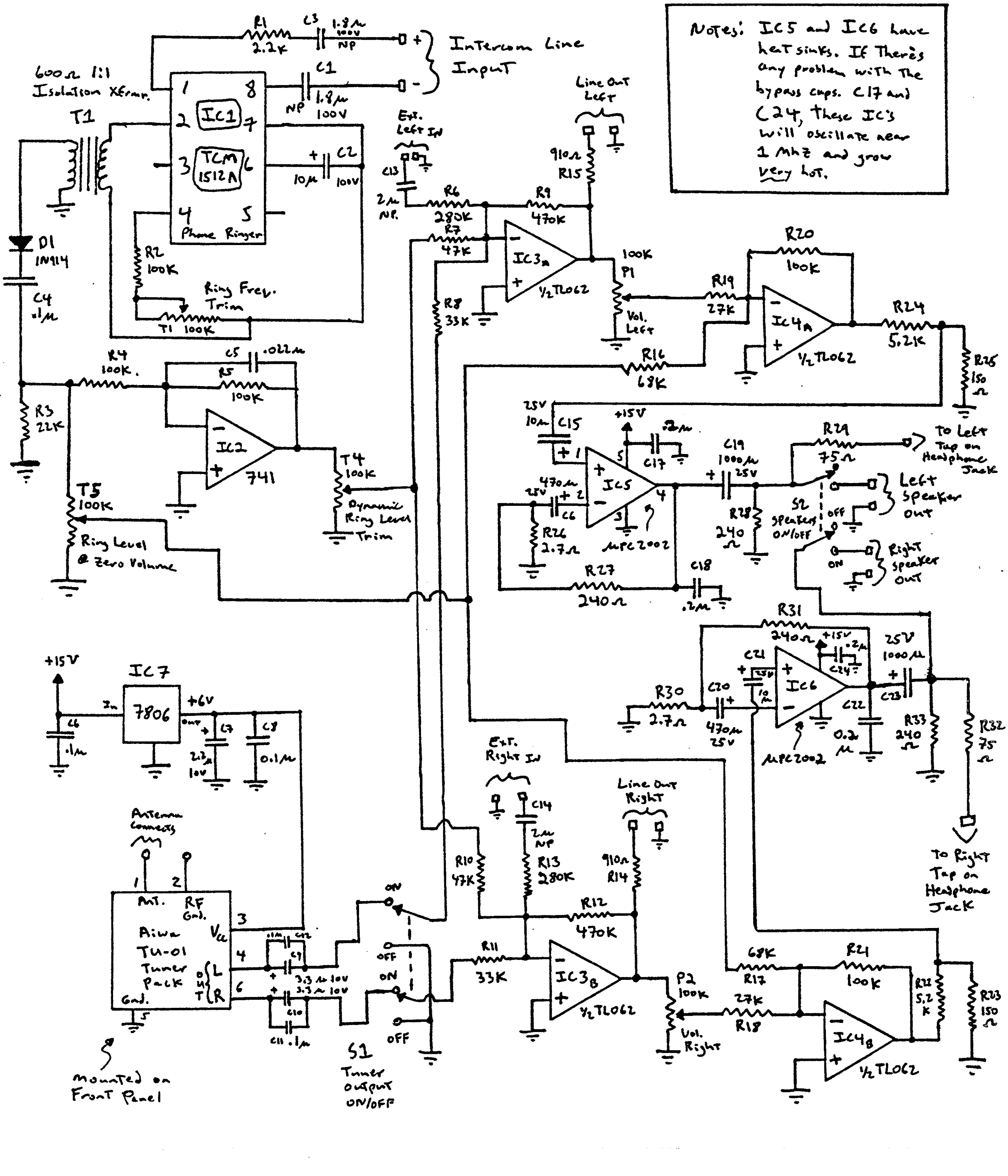
This unit has been operating nearly perfectly for approx. 9 months. The only problem was a bad 7815 regulator (which dropped the +15 Volt rail to +5 Volts). If problems crop up, they should be relatively easy to isolate. Check power supply voltages first. If the difficulty is in the transmitter monitoring circuitry, make sure the input feeds are all OK.

Jan., 1987

WMFO Transmitter Power and Audio Monitor ©J. Paradiso

Sheet #1: Audio Circuitry

Notes: IC5 and IC6 have heat sinks. If there's any problem with the bypass caps. C17 and C24, these IC's will oscillate near 1 MHz and grow very hot.



- Jan., 1987

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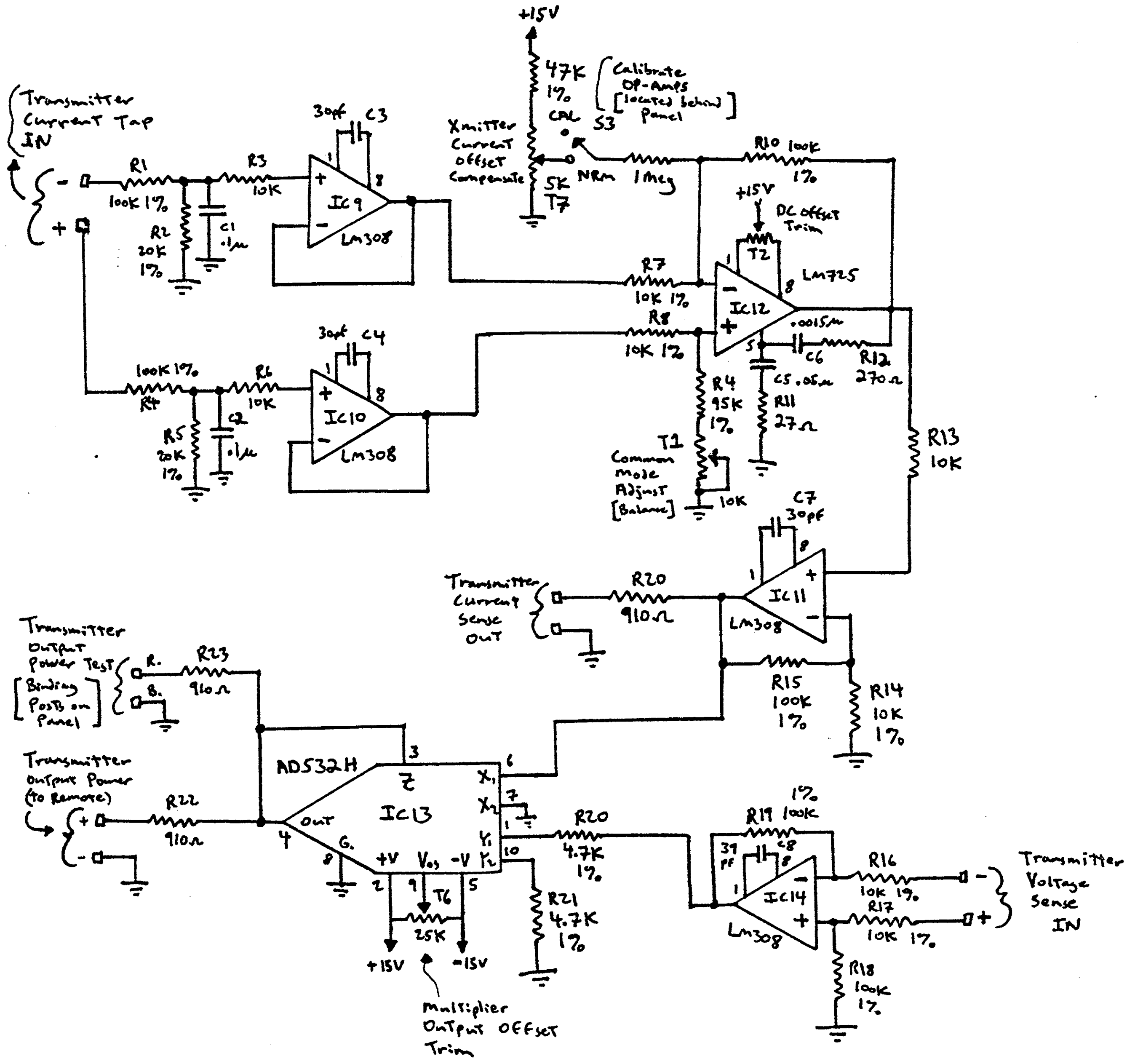

Sheet #2: Transmitter Output Power Multiplier

Sheet # 3
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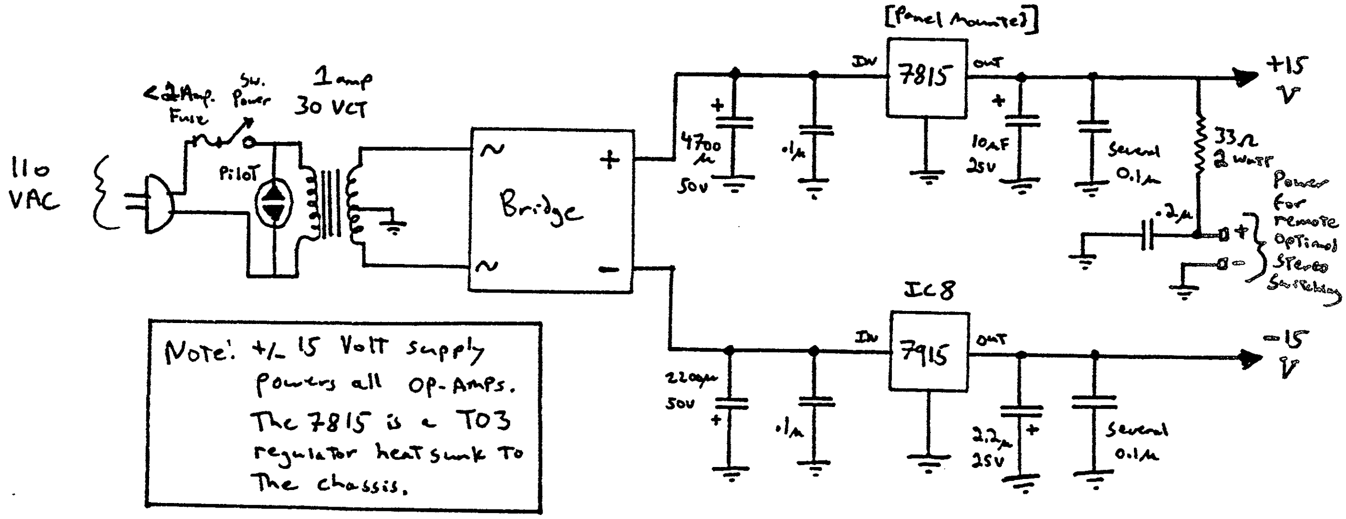
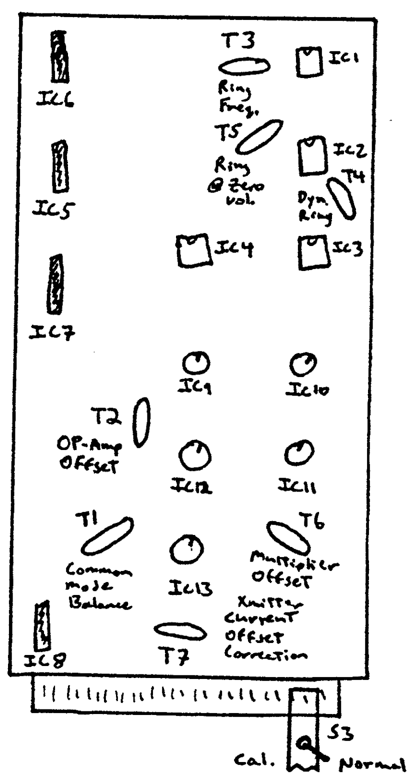


Power Supply



Note: IC14 is mounted on a small circuit card @ right angles to this card.

Circuit Card Layout and Trimmer Locations

Catalog Number 276-1302

ARCHER®

TECHNICAL DATA

AN EXCLUSIVE RADIO SHACK SERVICE TO THE EXPERIMENTER

TCM1512A Ring Detector/Driver IC

Description

The TCM1512A is designed for use as an alerting device in a line powered telephone. The IC (with a minimum of external components) is powered and activated by the telephone line's AC "ring" voltage to generate a signal suitable for driving a piezo buzzer element. In a typical telephone application this AC ring voltage can vary from 40 to 150 V_{RMS} over a frequency range of 14 to 68 Hz. The output signal is a square wave alternating between two frequencies in a ratio 1.14:1; and with the average of these two frequencies set at 1250 Hz, a "warble" rate or shift rate of approximately 10 Hz.

During standby (prior to activation) the ringer presents an impedance of 100K ohms or greater to prevent any interference with parallel "off-hook" telephones transmitting DTMF or voice frequencies. The IC is designed to handle lightning strikes on the line of 1500 V, 200 µsec duration. In addition, dial pulses from parallel phones are ignored so a false ringing of the bell (tapping) won't occur.

The primary application for the TCM1512A is to detect telephone ring voltages and to drive a piezo buzzer, replacing the "standard" electromechanical bell.

In the typical application, shown in Figure 1, the network formed by the 1.8 µF DC blocking capacitor, the 2200-ohm resistor and the full-wave diode bridge, supply the IC with power from the phone line. A differing ring voltage may require adjustment of the value of the blocking capacitor (as indicated in the graph shown in Figure 2).

Typical values of R_{OSC} will be in the range of 100-200K. Figure 3 is a curve showing the typical average output frequency vs R_{OSC}. Values may vary from IC to IC.

Typical Application

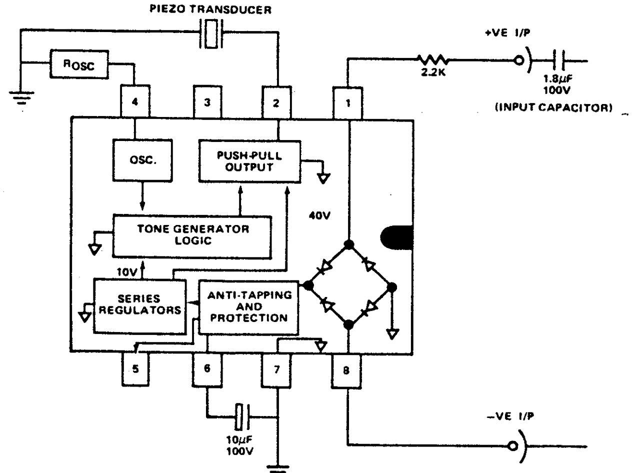
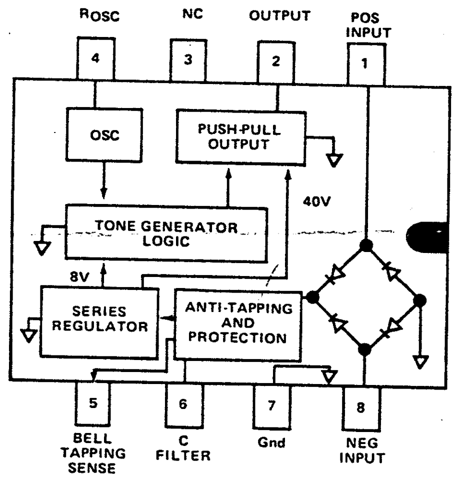


Figure 1: Typical Application



Pin Configuration

Features

- Low external component count
- Built-in static and lightning protection circuitry
- Built-in anti-"tapping" circuitry
- Built-in voltage regulators
- High standby impedance

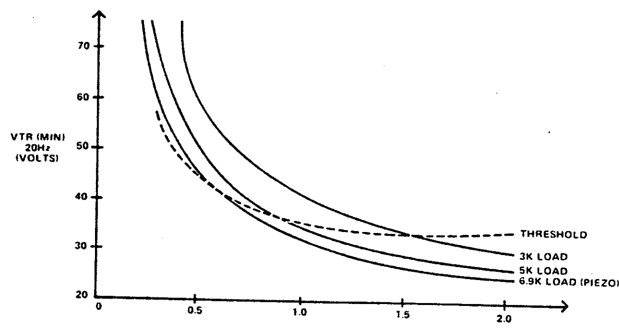


Figure 2: Minimum Operating Voltage vs Input Capacitance as a Function of the Load

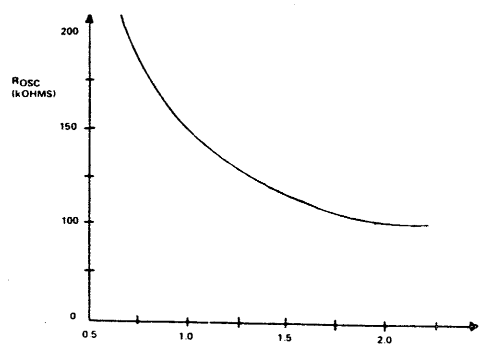


Figure 3: Typical Average Output Frequency F_{AVG} vs Tuning Resistor R_{OSC}

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Absolute Maximum Ratings

Input Voltage (between pins 1 and 8):	70 volts
Supply Voltage (between pins 6 and 7):	63 volts
Operating Temperature:	−20 to +70° C
Surge SCR on-state current:	0.9 amperes
Input Current:	0.5 amperes (RMS)
Maximum Power:	1 watt (25° C)

Recommended Operating Conditions

Parameter	Conditions	Min	Typ	Max	Units
Operating Voltage (noload after threshold reached		14		65	V
Operating Current (Pin 1)	R_L=open V_{IN}=40V V_{IN}=55V			1.1 4.0	mA mA
Standby Current (Pin 1)	R_L=open V_{IN}=5V			20	µA
Threshold Voltage	R_L=open, Pin 5 open, R_L=open, P5 activated	18 10		28 15	V V

Parameter	Conditions	Min	Typ	Max	Units
Pin 6 Voltage @ Threshold	(relative to P7) P5 activated	7		14	V
Threshold Current	R_L=open, P5 open		0.8	1.0	mA
SCR Trigger Current Trigger Voltage Holding Current		55 65 100		105 90	mA V µA
DC Pk Output Voltage	V_{IN}=65V Current Sink/ Source = 5 mA	37			V
Output Frequency Range-Vary R_{OSC} Maximum Minimum			2000 800		Hz Hz
Maximum Oscillator Drift over Full Operating Temp			1		%
Output Frequencies, when F_{AVG}=1250 Hz	High Tone Low Tone Warble (Shift Rate)		1339 1172 9.8		Hz Hz Hz

Additional Application

The TCM1512A contains all the active components necessary to build a security alarm system. A few external capacitors and resistors are required, along with a 16 V or 24 V transformer. The external 20 µF capacitor stores voltage during alternate half-cycle's that is added to the next half-cycle's voltage to generate 30 V (Pk) across Pins 6 and 7. The voltage doubler allows the IC to turn on with lower supply voltages than are normally required.

The IC is in standby (Off), when the "trip" switches are in their normally closed position (Figure 4). The 6.8K-ohm resistor doesn't allow the 10 µF capacitor (Pins 6 and 7) to charge up to its 11.5 V threshold voltage. The instant the trip switch is opened, the capacitor charges and the alarm sounds. Even if the trip switch is closed, the alarm will continue to sound. This is due to a latch circuit internal to the IC. The alarm is disarmed by opening the Reset/Set switch. When using a "Piezo Horn" for a transducer, sound pressure levels of 105 dB or greater can be achieved with as little as 10 mA supply current.

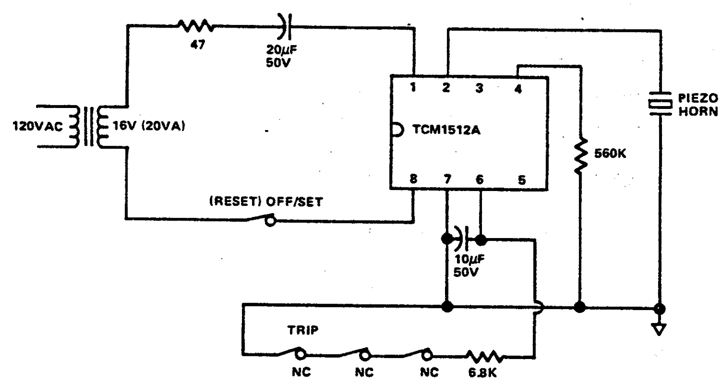


Figure 4: Alarm Circuit

RADIO SHACK, A DIVISION OF TANDY CORPORATION

U.S.A.: FORT WORTH, TEXAS 76102
CANADA: BARRIE, ONTARIO L4M 4W5

TANDY CORPORATION

AUSTRALIA	BELGIUM	U. K.
91 KURRAJONG ROAD MOUNT DRUITT, N.S.W. 2770	PARC INDUSTRIEL DE NANINNE 5140 NANINNE	BILSTON ROAD WEDNESBURY WEST MIDLANDS WS10 7JN

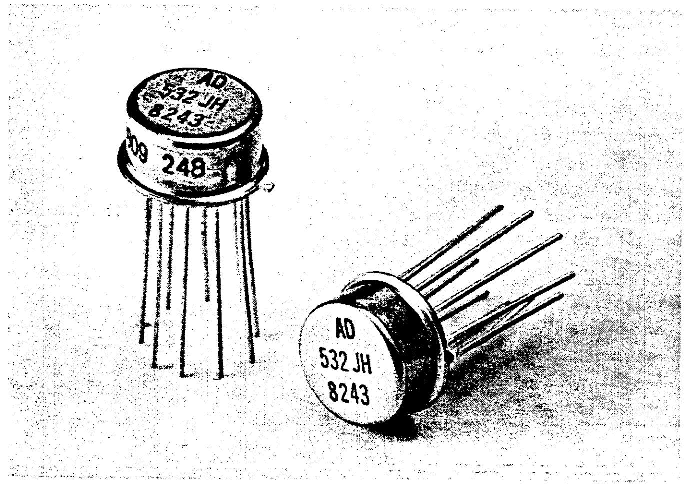
ANALOG DEVICES

Internally Trimmed Integrated Circuit Multiplier

AD532

FEATURES
Pretrimmed To ±1.0% (AD532K)
No External Components Required
Guaranteed ±1.0% max 4-Quadrant Error (AD532K)
Diff Inputs For $(X_1-X_2)(Y_1-Y_2)/10V$ Transfer Function
Monolithic Construction, Low Cost

APPLICATIONS
Multiplication, Division, Squaring, Square Rooting
Algebraic Computation
Power Measurements
Instrumentation Applications



PRODUCT DESCRIPTION
The AD532 is the first pretrimmed single chip monolithic multiplier/divider. It guarantees a maximum multiplying error of ±1.0% and a ±10V output voltage without the need for any external trimming resistors or output op amp. Because the AD532 is internally trimmed, its simplicity of use provides design engineers with an attractive alternative to modular multipliers, and its monolithic construction provides significant advantages in size, reliability and economy. Further, the AD532 can be used as a direct replacement for other IC multipliers that require external trim networks (such as the AD530).

FLEXIBILITY OF OPERATION
The AD532 multiplies in four quadrants with a transfer function of $(X_1-X_2)(Y_1-Y_2)/10V$, divides in two quadrants with a $10VZ/(X_1-X_2)$ transfer function, and square roots in one quadrant with a transfer function of $\pm\sqrt{10VZ}$. In addition to these basic functions, the differential X and Y inputs provide significant operating flexibility both for algebraic computation and transducer instrumentation applications. Transfer functions, such as $XY/10V$, $(X^2-Y^2)/10V$, $\pm X^2/10V$, and $10VZ/(X_1-X_2)$ are easily attained, and are extremely useful in many modulation and function generation applications, as well as in trigonometric calculations for airborne navigation and guidance applications, where the monolithic construction and small size of the AD532 offer considerable system advantages. In addition, the high CMRR (75dB) of the differential inputs makes the AD532 especially well qualified for instrumentation applications, as it can provide an output signal that is the product of two transducer-generated input signals.

GUARANTEED PERFORMANCE OVER TEMPERATURE
The AD532J and AD532K are specified for maximum multiplying errors of ±2% and ±1% of full scale, respectively at +25°C, and are rated for operation from 0 to +70°C. The AD532S has a maximum multiplying error of ±1% of full scale at +25°C; it is also 100% tested to guarantee a maximum error of ±4% at the extended operating temperature limits of -55°C and +125°C. All devices are available in either the hermetically-sealed TO-100 metal can or TO-116 ceramic DIP.

ADVANTAGES OF ON-THE-CHIP TRIMMING OF THE MONOLITHIC AD532

1. True ratiometric trim for improved power supply rejection.
2. Reduced power requirements since no networks across supplies are required.
3. More reliable since standard monolithic assembly techniques can be used rather than more complex hybrid approaches.
4. High impedance X and Y inputs with negligible circuit loading.
5. Differential X and Y inputs for noise rejection and additional computational flexibility.

P.O. Box 280; Norwood, Massachusetts 02062 U.S.A.
Tel: 617/329-4700 Twx: 710/394-6577
Telex: 924491 Cables: ANALOG NORWOODMASS

SPECIFICATIONS (@ +25°C, $V_S = \pm 15V$, $R \geq 2k\Omega$ V_{os} grounded)

Model	AD532J Min	AD532J Typ	AD532J Max	AD532K Min	AD532K Typ	AD532K Max	AD532S Min	AD532S Typ	AD532S Max	Units
MULTIPLIER PERFORMANCE										
Transfer Function		$\frac{(X_1-X_2)(Y_1-Y_2)}{10V}$			$\frac{(X_1-X_2)(Y_1-Y_2)}{10V}$			$\frac{(X_1-X_2)(Y_1-Y_2)}{10V}$		
Total Error ($-10V \leq X, Y \leq +10V$)		±1.5	**±2.0**		±0.7	**±1.0**		±0.5	**±1.0**	%
T_A = min to max		±2.5			±1.5				**±4.0**	%
Total Error vs Temperature		±0.04			±0.03			±0.01	**±0.04**	%/°C
Supply Rejection (±15V ±10)		±0.05			±0.05			±0.05		%/%
Nonlinearity, X (X = 20V pk-pk, Y = 10V)		±0.8			±0.5			±0.5		%
Nonlinearity, Y (Y = 20V pk-pk, X = 10V)		±0.3			±0.2			±0.2		%
Feedthrough, X (Y Nulled, X = 20V pk-pk 50Hz)		50	**200**		30	**100**		30	**100**	mV
Feedthrough, Y (X Nulled, Y = 20V pk-pk 50Hz)		30	**150**		25	**80**		25	**80**	mV
Feedthrough vs. Temp.		2.0			1.0			1.0		mV p-p/°C
Feedthrough vs. Power Supply		±0.25			±0.25			±0.25		mV/%
DYNAMICS										
Small Signal BW (V_{OUT} = 0.1 rms)		1			1			1		MHz
1% Amplitude Error		75			75			75		kHz
Slew Rate (V_{OUT} 20 pk-pk)		45			45			45		V/μs
Settling Time (to 2%, ΔV_{OUT} = 20V)		1			1			1		μs
NOISE										
Wideband Noise f = 5Hz to 10kHz		0.6			0.6			0.6		mV (rms)
f = 5Hz to 5MHz		3.0			3.0			3.0		mV (rms)
OUTPUT										
Output Voltage Swing	±10	±13		±10	±13		±10	±13		V
Output Impedance (f ≤ 1kHz)		1			1			1		Ω
Output Offset Voltage		±40				**±30**			**±30**	mV
Output Offset Voltage vs. Temp.		0.7			0.7				**2.0**	mV/°C
Output Offset Voltage vs. Supply		±2.5			±2.5			±2.5		mV/%
INPUT AMPLIFIERS (X, Y and Z)										
Signal Voltage Range (Diff. or CM Operating Diff)		±10			±10			±10		V
CMRR	**40**			**50**			**50**			dB
Input Bias Current										
X, Y Inputs		3			1.5	**4**		1.5	**4**	μA
X, Y Inputs T_{min} to T_{max}		10			8			8		μA
Z Input		±10			±5	±15		±5	±15	μA
Z Input T_{min} to T_{max}		±30			±25			±25		μA
Offset Current		±0.3			±0.1			±0.1		μA
Differential Resistance		10			10			10		MΩ
DIVIDER PERFORMANCE										
Transfer Function ($X_1 > X_2$)		$10V\,Z/(X_1-X_2)$			$10V\,Z/(X_1-X_2)$			$10V\,Z/(X_1-X_2)$		
Total Error										
($V_X = -10V$, $-10V \leq V_Z \leq +10V$)		±2			±1			±1		%
($V_X = -1V$, $-10V \leq V_Z \leq +10V$)		±4			±3			±3		%
SQUARE PERFORMANCE										
Transfer Function		$\frac{(X_1-X_2)^2}{10V}$			$\frac{(X_1-X_2)^2}{10V}$			$\frac{(X_1-X_2)^2}{10V}$		
Total Error		±0.8			±0.4			±0.4		%
SQUARE-ROOTER PERFORMANCE										
Transfer Function		$(X_1-X_2)^2/10V$			$(X_1-X_2)^2/10V$			$(X_1-X_2)^2/10V$		
Total Error ($0V \leq V_Z \leq 10V$)		±1.5			±1.0			±1.0		%
POWER SUPPLY SPECIFICATIONS										
Supply Voltage										
Rated Performance		±15			±15			±15		V
Operating	±10		±18	±10		±18	±10		±22	V
Suply Current										
Quiescent		4	6		4	6		4	6	mA

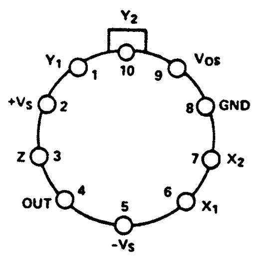
NOTE
Specifications subject to change without notice.

Specifications shown in boldface are tested on all production units at final electrical test. Results from those tests are used to calculate outgoing quality levels. All min and max specifications are guaranteed, although only those shown in boldface are tested on all production units.

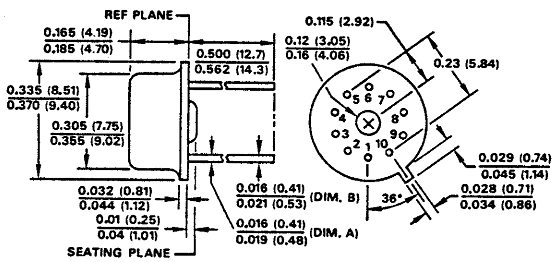
PIN CONFIGURATION & DIMENSIONS

Dimensions shown in inches and (mm).

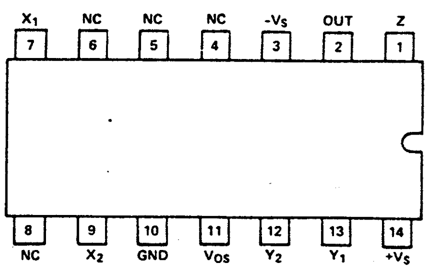
AD532H



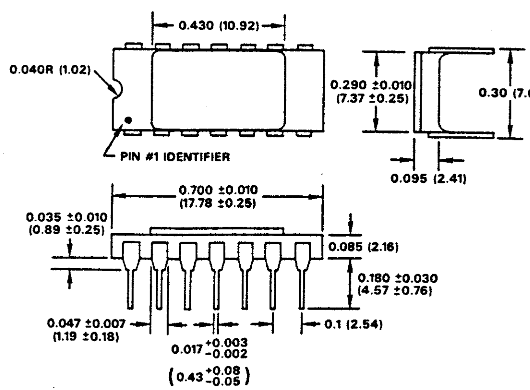
TO-100



AD532D



TO-116



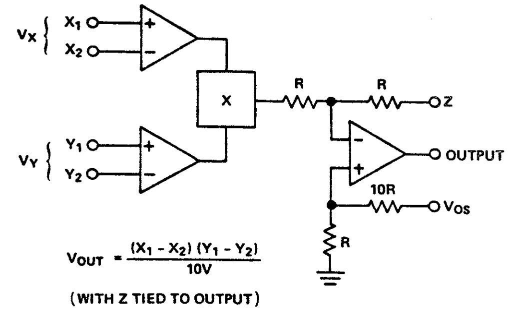


Figure 1. Functional Block Diagram

FUNCTIONAL DESCRIPTION

The functional block diagram for the AD532 is shown in Figure 1, and the complete schematic in Figure 2. In the multiplying and squaring modes, Z is connected to the output to close the feedback around the output op amp. (In the divide mode, it is used as an input terminal.)

The X and Y inputs are fed to high impedance differential amplifiers featuring low distortion and good common mode rejection. The amplifier voltage offsets are actively laser trimmed to zero during production. The product of the two inputs is resolved in the multiplier cell using Gilbert's linearized transconductance technique. The cell is laser trimmed to obtain $V_{out} = (X_1 - X_2)(Y_1 - Y_2)/10$ volts. The built-in op amp is used to obtain low output impedance and make possible self-contained operation. The residual output voltage offset can be zeroed at V_{os} in critical applications . . . . otherwise the V_{os} pin should be grounded.

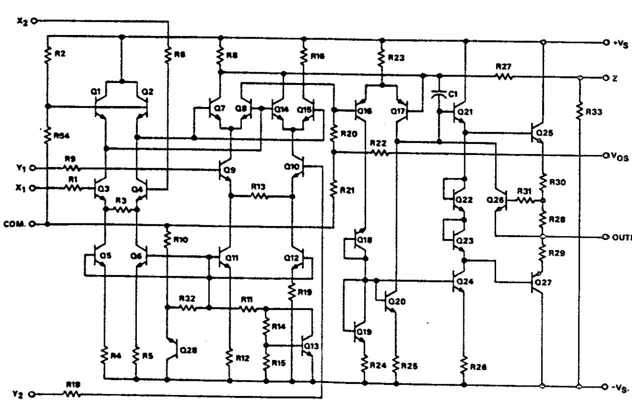


Figure 2. AD532 Schematic Diagram

ORDERING GUIDE

Model	Max Mult Error	Temperature Range	Model	Max Mult Error	Temperature Range
AD532JH	±2.0%	0 to +70°C	AD532SH	±1.0%	-55°C to +125°C
AD532JD	±2.0%	0 to +70°C	AD532SD	±1.0%	-55°C to +125°C
AD532KH	±1.0%	0 to +70°C	AD532SH/883B	±1.0%	-55°C to +125°C
AD532KD	±1.0%	0 to +70°C	AD532SD/883B	±1.0%	-55°C to +125°C

AD532 PERFORMANCE CHARACTERISTICS

Multiplication accuracy is defined in terms of total error at +25°C with the rated power supply. The value specified is in percent of full scale and includes X_{in} and Y_{in} nonlinearities, feedback and scale factor error. To this must be added such application-dependent error terms as power supply rejection, common mode rejection and temperature coefficients (although worst case error over temperature is specified for the AD532S). Total expected error is the rms sum of the individual components, since they are uncorrelated.

Accuracy in the divide mode is only a little more complex. To achieve division, the multiplier cell must be connected in the feedback of the output op amp as shown in Figure 13. In this configuration, the multiplier cell varies the closed loop gain of the op amp in an inverse relationship to the denominator voltage. Thus, as the denominator is reduced, output offset, bandwidth and other multiplier cell errors are adversely affected. The divide error and drift are then $\epsilon_m \cdot 10V/X_1-X_2)$ where ϵ_m represents multiplier full scale error and drift, and (X_1-X_2) is the absolute value of the denominator.

NONLINEARITY

Nonlinearity is easily measured in percent harmonic distortion. The curves of Figures 3 and 4 characterize output distortion as a function of input signal level and frequency respectively, with one input held at plus or minus 10V dc. In Figure 4 the sine wave amplitude is 20V(p-p).

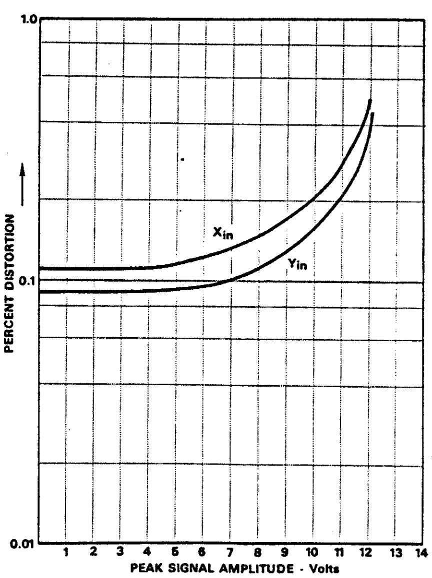


Figure 3. Percent Distortion vs. Input Signal

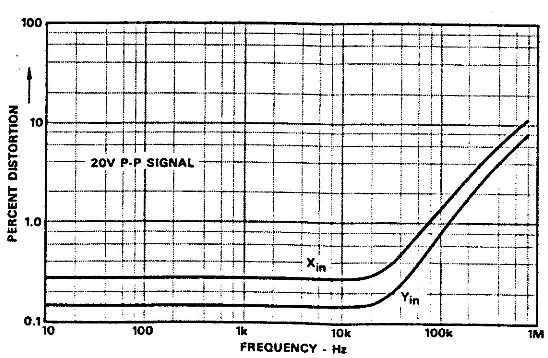


Figure 4. Percent Distortion vs. Frequency

AC FEEDTHROUGH

AC Feedthrough is a measure of the multiplier's zero suppression. With one input at zero, the multiplier output should be zero regardless of the signal applied to the other input. Feedthrough as a function of frequency for the AD532 is shown in Figure 5. It is measured for the condition $V_X = 0$, $V_Y = 20V(p\text{-}p)$ and $V_Y = 0$, $V_X = 20V(p\text{-}p)$ over the given frequency range. It consists primarily of the second harmonic and is measured in millivolts peak-to-peak.

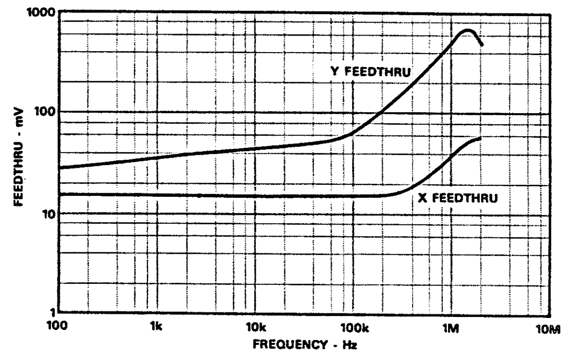


Figure 5. Feedthrough vs. Frequency

COMMON MODE REJECTION

The AD532 features differential X and Y inputs to enhance its flexibility as a computational multiplier/divider. Common mode rejection for both inputs as a function of frequency is shown in Figure 6. It is measured with $X_1 = X_2 = 20V(p\text{-}p)$, $(Y_1-Y_2) = \pm10V$ dc and $Y_1 = Y_2 = 20V(p\text{-}p)$, $(X_1-X_2) = \pm10V$ dc.

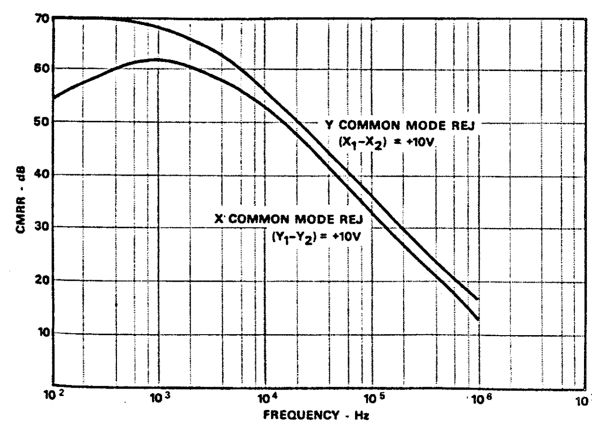


Figure 6. CMRR vs. Frequency

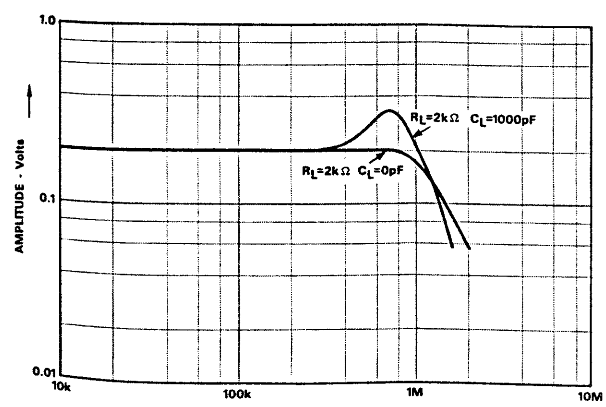


Figure 7. Frequency Response, Multiplying

DYNAMIC CHARACTERISTICS

The closed loop frequency response of the AD532 in the multiplier mode typically exhibits a 3dB bandwidth of 1MHz and rolls off at 6dB/octave thereafter. Response through all inputs is essentially the same as shown in Figure 7. In the divide mode, the closed loop frequency response is a function of the absolute value of the denominator voltage as shown in Figure 8.

Stable operation is maintained with capacitive loads to 1000pF in all modes, except the square root for which 50pF is a safe upper limit. Higher capacitive loads can be driven if a 100Ω resistor is connected in series with the output for isolation.

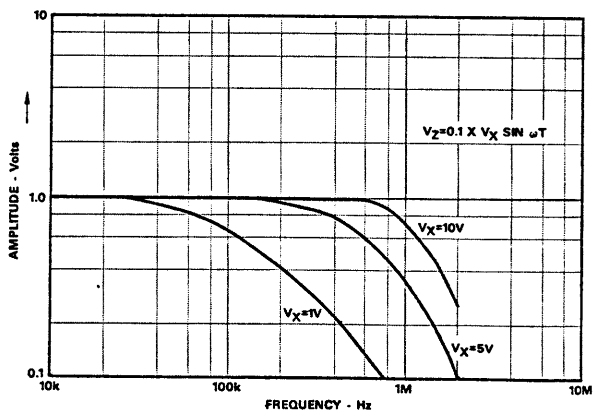


Figure 8. Frequency Response, Dividing

POWER SUPPLY CONSIDERATIONS

Although the AD532 is tested and specified with ±15V dc supplies, it may be operated at any supply voltage from ±10V to ±18V for the J and K versions and ±10V to ±22V for the S version. The input and output signals must be reduced proportionately to prevent saturation, however, with supply voltages below ±15V, as shown in Figure 9. Since power supply sensitivity is not dependent on external null networks as in the AD530 and other conventionally nulled multipliers, the power supply rejection ratios are improved from 3 to 40 times in the AD532.

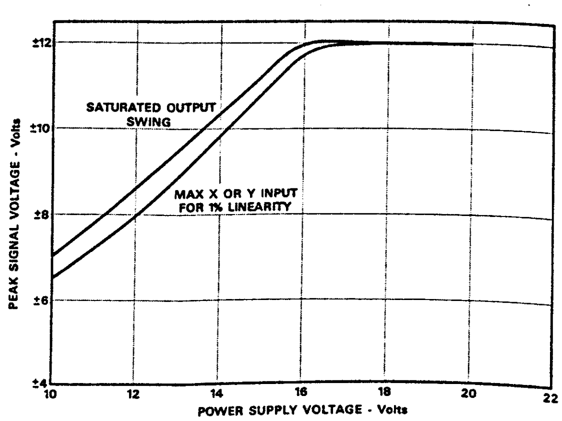


Figure 9. Signal Swing vs. Supply

NOISE CHARACTERISTICS

All AD532s are screened on a sampling basis to assure that output noise will have no appreciable effect on accuracy. Typical spot noise vs. frequency is shown in Figure 10.

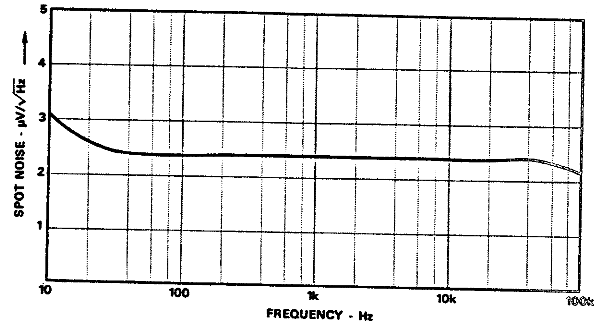


Figure 10. Spot Noise vs. Frequency

APPLICATIONS CONSIDERATIONS

The performance and ease of use of the AD532 is achieved through the laser trimming of thin film resistors deposited directly on the monolithic chip. This trimming-on-the-chip technique provides a number of significant advantages in terms of cost, reliability and flexibility over conventional in-package trimming of off-the-chip resistors mounted or deposited on a hybrid substrate.

First and foremost, trimming on the chip eliminates the need for a hybrid substrate and the additional bonding wires that are required between the resistors and the multiplier chip. By trimming more appropriate resistors on the AD532 chip itself, the second input terminals that were once committed to external trimming networks (e.g., AD530) have been freed to allow fully differential operation at both the X and Y inputs. Further, the requirement for an input attenuator to adjust the gain at the Y input has been eliminated, letting the user take full advantage of the high input impedance properties of the input differential amplifiers. Thus, the AD532 offers greater flexibility for both algebraic computation and transducer instrumentation applications.

Finally, provision for fine trimming the output voltage offset has been included. This connection is optional, however, as the AD532 has been factory-trimmed for total performance as described in the listed specifications.

REPLACING OTHER IC MULTIPLIERS

Existing designs using IC multipliers that require external trimming networks (such as the AD530) can be simplified using the pin-for-pin replaceability of the AD532 by merely grounding the X_2, Y_2 and V_{OS} terminals. (The V_{OS} terminal should always be grounded when unused.)

APPLICATIONS

MULTIPLICATION

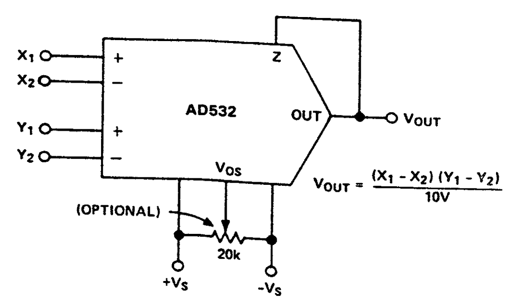


Figure 11. Multiplier Connection

For operation as a multiplier, the AD532 should be connected as shown in Figure 11. The inputs can be fed differentially to the X and Y inputs, or single-ended by simply grounding the unused input. Connect the inputs according to the desired polarity in the output. The Z terminal is tied to the output to close the feedback loop around the op amp (see Figure 1). The offset adjust V_{OS} is optional and is adjusted when both inputs are zero volts to obtain zero out, or to buck out other system offsets.

SQUARE

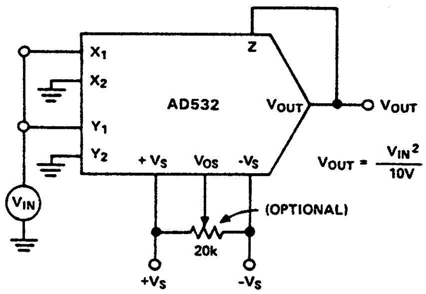


Figure 12. Squarer Connection

The squaring circuit in Figure 12 is a simple variation of the multiplier. The differential input capability of the AD532 can be used, however, to obtain a positive or negative output response to the input....a useful feature for control applications, as it might eliminate the need for an additional inverter somewhere else.

DIVISION

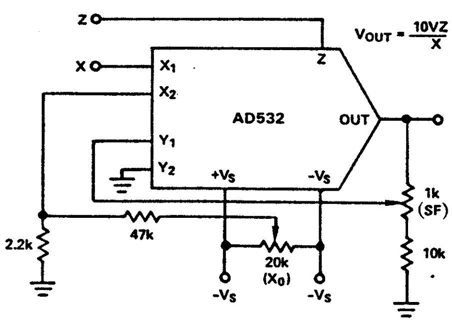


Figure 13. Divider Connection

The AD532 can be configured as a two-quadrant divider by connecting the multiplier cell in the feedback loop of the op amp and using the Z terminal as a signal input, as shown in Figure 13. It should be noted, however, that the output error is given approximately by $10V\epsilon_m/(X_1-X_2)$, where ϵ_m is the total error specification for the multiply mode; and bandwidth by $f_m \cdot (X_1-X_2)/10V$, where f_m is the bandwidth of the multiplier. Further, to avoid positive feedback, the X input is restricted to negative values. Thus for single-ended negative inputs (0V to −10V), connect the input to X and the offset null to X_2; for single-ended positive inputs (0V to +10V), connect the input to X_2 and the offset null to X_1. For optimum performance, gain (S.F.) and offset (X_0) adjustments are recommended as shown and explained in Table I.

For practical reasons, the useful range in denominator input is approximately $500mV \leqslant |(X_1-X_2)| \leqslant 10V$. The voltage offset adjust (V_{os}), if used, is trimmed with Z at zero and (X_1-X_2) at full scale.

SQUARE ROOT

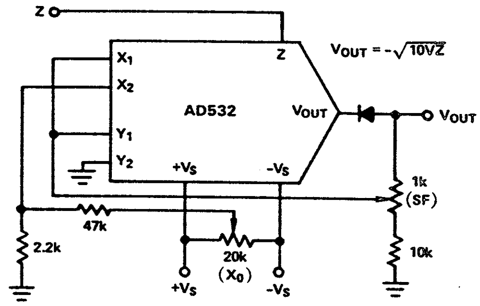


Figure 14. Square Rooter Connection

The connections for square root mode are shown in Figure 14. Similar to the divide mode, the multiplier cell is connected in the feedback of the op amp by connecting the output back to both the X and Y inputs. The diode D_1 is connected as shown to prevent latch-up as Z_{in} approaches 0 volts. In this case, the V_{OS} adjustment is made with Z_{in} = +0.1V dc, adjusting V_{OS} to obtain −1.0V dc in the output, $V_{out} = -\sqrt{10VZ}$. For optimum performance, gain (S.F.) and offset (X_0) adjustments are recommended as shown and explained in Table I.

DIFFERENCE OF SQUARES

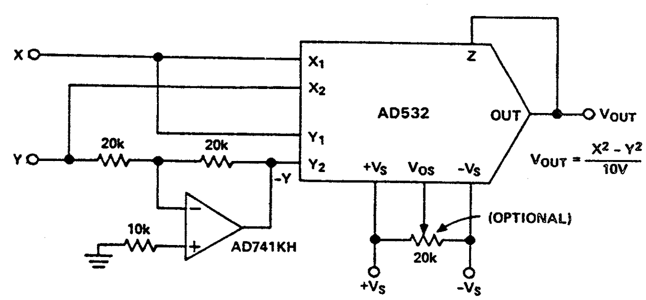


Figure 15. Differential of Squares Connection

The differential input capability of the AD532 allows for the algebraic solution of several interesting functions, such as the difference of squares, $X^2-Y^2/10V$. As shown in Figure 15, the AD532 is configured in the square mode, with a simple unity gain inverter connected between one of the signal inputs (Y) and one of the inverting input terminals ($-Y_{in}$) of the multiplier. The inverter should use precision (0.1%) resistors or be otherwise trimmed for unity gain for best accuracy.

TABLE I

ADJUST PROCEDURE (Divider or Square Rooter)

	DIVIDER			SQUARE ROOTER	
	With:		Adjust for:	With:	Adjust for:
Adjust	X	Z	V_{out}	Z	V_{out}
Scale Factor	−10V	+10V	±10V	+10V	−10V
X_0 (Offset)	−1V	+0.1V	±10V	+0.1V	−1V

Repeat if required.

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SECTION TWO

Misc.
Procedures
and
Studio Wiring
Conventions

A) Interstudio Wiring Details

WMFO GENERAL WIRING NOTES:

Internal master gain pots on all mixing consoles are adjusted to yield OdB meter deflection for a standard source (Housetone @ 1khz) at fader midpoints. All sources have been padded or adjusted to match standard source meter deflection at fader midpoint. All mixing consoles are padded at the output to yield unity gain at fader midpoint. All DA's and the Studio Switcher are adjusted to yield unity gain into 600 ohms. Thus, any element of the air chain may be bypassed or added, via patch bays, with minimal gain change. The only gain change (increase) is at final stage, the compressor (Compellor), feeding the phone line to the transmitter.

In Master Control (M.C.) and Studio A the blue "GEPCO" cables connecting the patch bays have ten pairs each. These are numbered in decades corresponding to the patch bay jack numbers they are tied to (Unless otherwise marked). To find actual pair number, add the inner jacket number to the outer jacket number. For example, the cable in Studio A labeled "I20+" contains pairs 21 to 30. Cable "I20+", pair "8" corresponds to patch bay I, jack number "28".

In Master Control cable "I90+", pair 100 is spare (hanging) from D.A. rack to wall.

Tie Lines linking patch bays in one studio to patch bays in another are listed as "A-to-B 1 L" and "A-to-B 1 R". All tie lines are normalled through the Master Control patch bays.

If telephone outputs in the studios are patched to another fader, the interface system will not function when feeding the "program" signal down the telephone line.

Channel#1 = Left, Channel#2 = Right

INTER-STUDIO AND MASTER CONTROL WIRING NOTES:

Cables from the studios to Master Control terminate at christmas trees on each end. In the studios they are located under the consoles, in M.C. they are on the wall at the Distribution Amp (blue) rack.

Inter-rack cables from the D.A. (blue) rack to the Monitor (gray) rack terminate at christmas trees on the wall.

For cable and pair number positions see diagrams.

For pair assignments see tables.

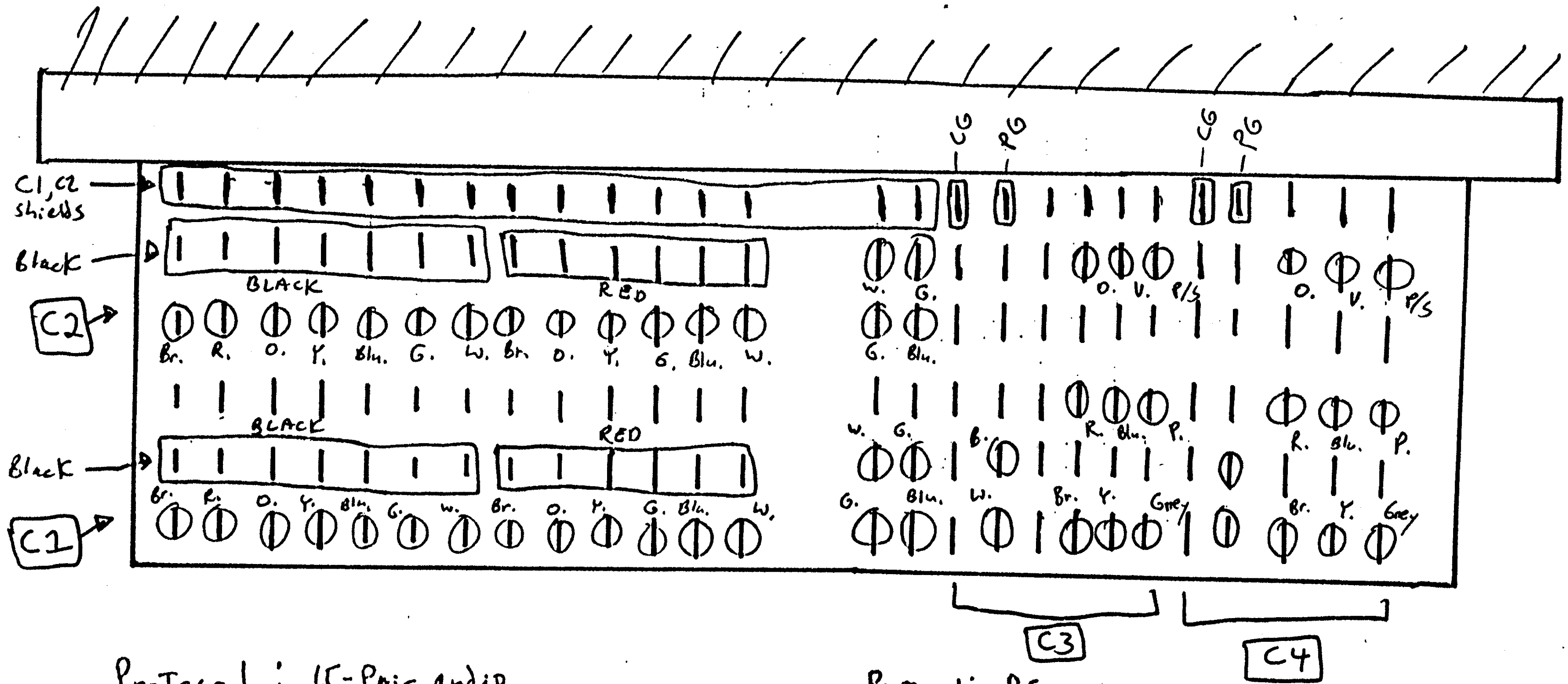
Pair numbers not listed are assumed unused.

Color codes listed in notes column are for linking cables. Colors are continued on all links except where noted.

STUDIO A CABLE ASSIGNMENTS

Pair Number	Signal	Notes
C1-1	Program (DA) feed to MC 1	
C1-2	Program (DA) feed to MC 2	
C1-3	Studio B program out 1	
C1-4	Studio B program out 2	
C1-5	Audition out (mono)	
C1-6	Studio D out 2	
C1-7	Ext. FM monitor feed 1	
C1-8	Ext. FM monitor feed 2	
C1-9	Telephone audio return 1	
C1-10	Telephone audio return 2	
C1-11	Curtis Lounge 1	
C1-12	Curtis Lounge 2	
C1-13	MacPhie Pub 1	
C1-14	MacPhie Pub 2	
C1-15	blu: EBS LED, grn: Overmod LED	wht from wall to Switcher
C2-1	Housetone	
C2-2	Telephone audio send	
C2-3	Studio C program out 1	
C2-4	Studio C program out 2	
C2-5	A-to-B 1 L	
C2-6	A-to-B 1 R	
C2-7	A-to-B 2 L	
C2-8	A-to-B 2 R	
C2-9	A-to-C 1 L	
C2-10	A-to-C 1 R	
C2-11	A-to-C 2 L	
C2-12	A-to-C 2 R	
C2-13	Intercom L-pad defeat	Bussed across all studios
C2-14	N.O. Mic relay	To ON AIR light control
C2-15	N.C. Mic relay	To Studio Switcher gry/brn
C3-brn	Studio Switcher switch A	
C3-red	Studio Switcher switch B	
C3-org	Studio Switcher switch C	
C3-yel	Studio Switcher status LED A	
C3-blu	Studio Switcher status LED B	
C3-vio	Studio Switcher status LED C	
C3-gry	Studio Switcher mic LED A	
C3-pnk	Studio Switcher mic LED B	
C3-stripe	Studio Switcher mic LED C	beige, wall to Switcher
C3-Pair	Intercom speaker feed	Bussed across all studios & padded to feed fader
C3-~~Pair~~ main Gnd.	Studio Switcher ground	blk from wall to Switcher
	Phone/Doorbell LED (~~hot~~ ground)	Bussed across all studios
C3-~~Main~~ Pair Gnd.	Phone/Doorbell LED (~~ground~~ hot)	Bussed across all studios
C4-brn/red	EBS gen switch	
C4-org/vio	Line feed from telephone set	org(+), #197 from wall to rack
C4-Pair	Studio D out 1	

XMAS TREE: C1/C2/C3/C4... side view



Protocol: 15-Pair audio

1 — brown/black
2 — red/black
3 — orange/black
4 — yellow/black
5 — blue/black
6 — green/black
7 — white/black
8 — brown/red
9 — orange/red
10 — yellow/red
11 — green/red
12 — blue/red
13 — white/red
14 — green/white
15 — blue/Green

Protocol: DC cable

9 wire + 1 pair + 2 shields

common ⏚
Pair ⏚
pair — white/Black
Brown
Red
orange
Yellow
blue
Violet
Grey
Pink
Pink/Striped

Studio A

A1, A2 = 15-pair Cable

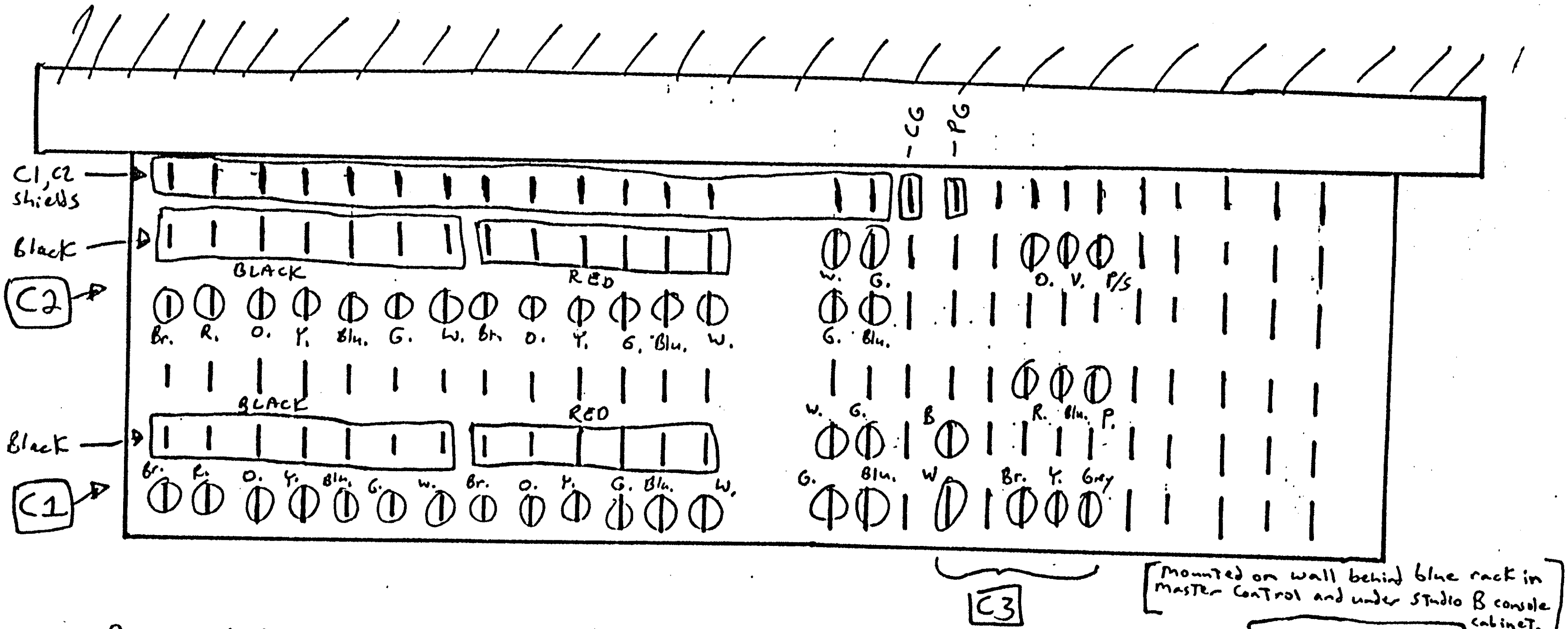
A3, A4 = DC cables

[Located behind blue rack in Master Control (on wall) and under Studio A console cabinet.]

STUDIO B CABLE ASSIGNMENTS

Pair Number	Signal	Notes
C1-1	Program (DA) feed to MC 1	
C1-2	Program (DA) feed to MC 2	
C1-3	Studio C program out 1	
C1-4	Studio C program out 2	
C1-5	Audition out 1	
C1-6	Audition out 2	
C1-7	Ext. FM monitor feed 1	
C1-8	Ext. FM monitor feed 2	
C1-9	Telephone audio return 1	
C1-10	Telephone audio return 2	
C1-15	blu: EBS LED, grn: Overmod LED	wht from wall to Switcher
C1-14	Line feed from telephone set	Pair I98 from wall to rack
C2-1	Housetone	
C2-2	Telephone audio send	
C2-3	Studio A program out 1	Phase (red/blk) reversed
C2-4	Studio A program out 2	Phase (red/blk) reversed
C2-5	A-to-B 1 L	
C2-6	A-to-B 1 R	
C2-7	A-to-B 2 L	
C2-8	A-to-B 2 R	
C2-9	B-to-C 1 L	
C2-10	B-to-C 1 R	
C2-11	B-to-C 2 L	
C2-12	B-to-C 2 R	
C2-13	Intercom L-pad defeat	Bussed across all studios
C2-14	N.O. Mic relay	To ON AIR light control
C2-15	N.C. Mic relay	To Studio Switcher pnk/red
C3-brn	Studio Switcher switch A	
C3-red	Studio Switcher switch B	
C3-org	Studio Switcher switch C	
C3-yel	Studio Switcher status LED A	
C3-blu	Studio Switcher status LED B	
C3-vio	Studio Switcher status LED C	
C3-gry	Studio Switcher mic LED A	
C3-pnk	Studio Switcher mic LED B	
C3-stripe	Studio Switcher mic LED C	beige, wall to Switcher
C3-Pair	Intercom speaker feed	Bussed across all studios & padded to feed patch bay
C3-~~Pair~~ Main Gnd.	Studio Switcher ground, Phone/Doorbell LED (~~hot~~ ground)	blk from wall to Switcher Bussed across all studios
C3-~~Main~~ Pair Gnd.	Phone/Doorbell LED (~~ground~~ hot)	Bussed across all studios

XMAS TREE: C1/C2/C3/C4... side view



STUDIO's B and C

Protocol: 15-Pair audio

1 — brown/black
2 — red/black
3 — orange/black
4 — yellow/black
5 — blue/black
6 — green/black
7 — white/black
8 — brown/red
9 — orange/red
10 — yellow/red
11 — green/red
12 — blue/red
13 — white/red
14 — green/white
15 — blue/Green

Protocol: DC cable

9 wire + 1 pair + 2 shields

common ⏚ → CG
Pair ⏚ → PG
pair — white/Black
Brown
Red
orange
Yellow
blue
Violet
Grey
Pink
Pink/Striped → P/S

(B) C1, (B) C2 = 15-pair cable

(B) C3 = DC cable

Notes: In MC, The

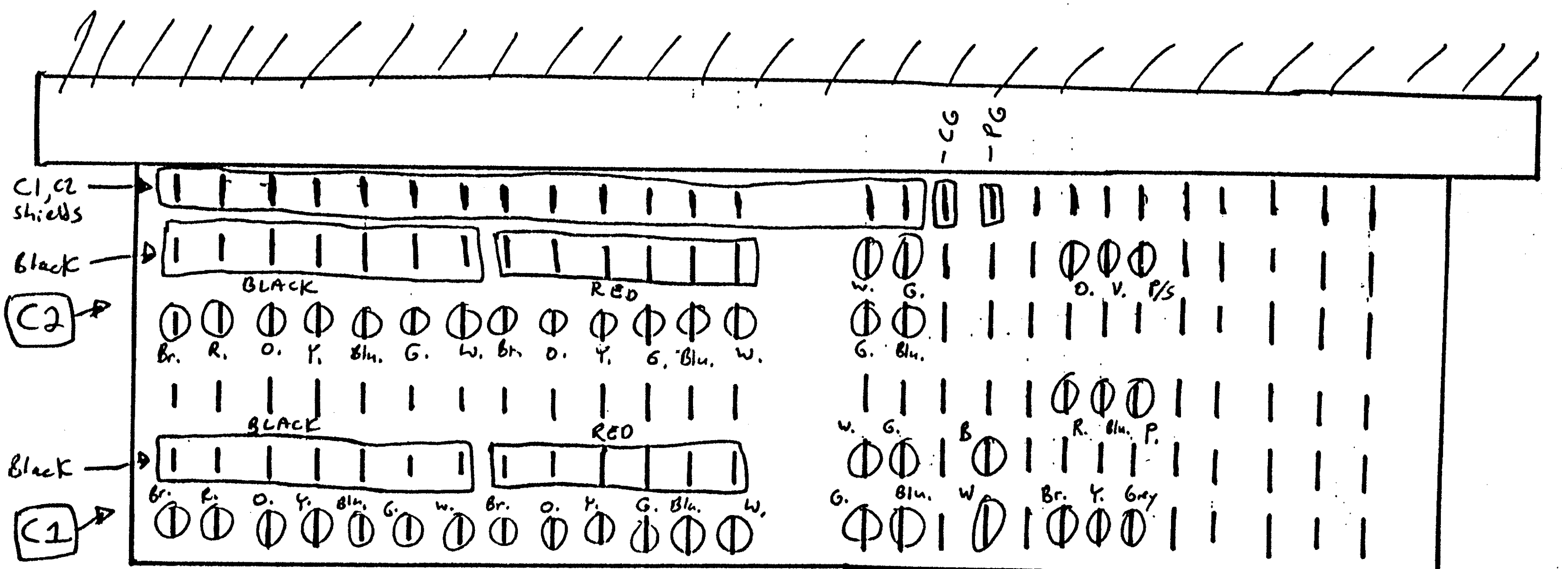
→ Red/Hot connections for cable C1, pairs 8→13 are reversed!!

→ In Studio B, The DC cable doesn't come To The Terminal block, but To a barrier strip.

STUDIO C CABLE ASSIGNMENTS

Pair Number	Signal	Notes
C1-1	Program (DA) feed to MC 1	
C1-2	Program (DA) feed to MC 2	
C1-3	Studio B program out 1	
C1-4	Studio B program out 2	
C1-7	Ext. FM monitor feed 1	
C1-8	Ext. FM monitor feed 2	
C1-9	Telephone audio return 1	
C1-10	Telephone audio return 2	
C1-15	blu: EBS LED, grn: over mod LED	wht from wall to Switcher
C1-14	Line feed from telephone set	#I99 from wall to rack
C2-1	Housetone	
C2-2	Telephone audio send	
C2-3	Studio A program out 1	
C2-4	Studio A program out 2	
C2-5	A-to-C 1 L	
C2-6	A-to-C 1 R	
C2-7	A-to-C 2 L	
C2-8	A-to-C 2 R	
C2-9	B-to-C 1 L	
C2-10	B-to-C 1 R	
C2-11	B-to-C 2 L	
C2-12	B-to-C 2 R	
C2-13	Intercom L-pad defeat	Bussed across all studios
C2-14	N.O. Mic relay	To ON AIR light control
C2-15	N.C. Mic relay	To Studio Switcher beige/org
C3-brn	Studio Switcher switch A	
C3-red	Studio Switcher switch B	
C3-org	Studio Switcher switch C	
C3-yel	Studio Switcher status LED A	
C3-blu	Studio Switcher status LED B	
C3-vio	Studio Switcher status LED C	
C3-gry	Studio Switcher mic LED A	
C3-pnk	Studio Switcher mic LED B	
C3-stripe	Studio Switcher mic LED C	beige, wall to Switcher
C3-Pair	Intercom speaker feed	Bussed across all studios & padded to feed patch bay
C3-~~Pair~~ Main Gnd.	Studio Switcher ground	blk from wall to Switcher
	Phone/Doorbell LED (~~hot~~ ground)	Bussed across all studios
C3-~~Main~~ Pair Gnd.	Phone/Doorbell LED (~~ground~~ hoT)	Bussed across all studios

XMAS TREE: C1/C2/C3/C4... side view



[Mounted on wall behind blue rack in Master Control and under Studio B console cabinet.]

STUDIO's B and C

Protocol: 15-Pair audio

1 — brown/black
2 — red/black
3 — orange/black
4 — yellow/black
5 — blue/black
6 — green/black
7 — white/black
8 — brown/red
9 — orange/red
10 — yellow/red
11 — green/red
12 — blue/red
13 — white/red
14 — green/white
15 — blue/Green

Protocol: DC cable

9 wire + 1 pair + 2 shields

common ⏚ → CG
Pair ⏚ → PG
pair — white/Black
Brown
Red
Orange
Yellow
blue
Violet
Grey
Pink
Pink/Striped → P/S

(B)C1, (B)C2 = 15-pair cable

(B) C3 = DC cable

Notes: In MC, The

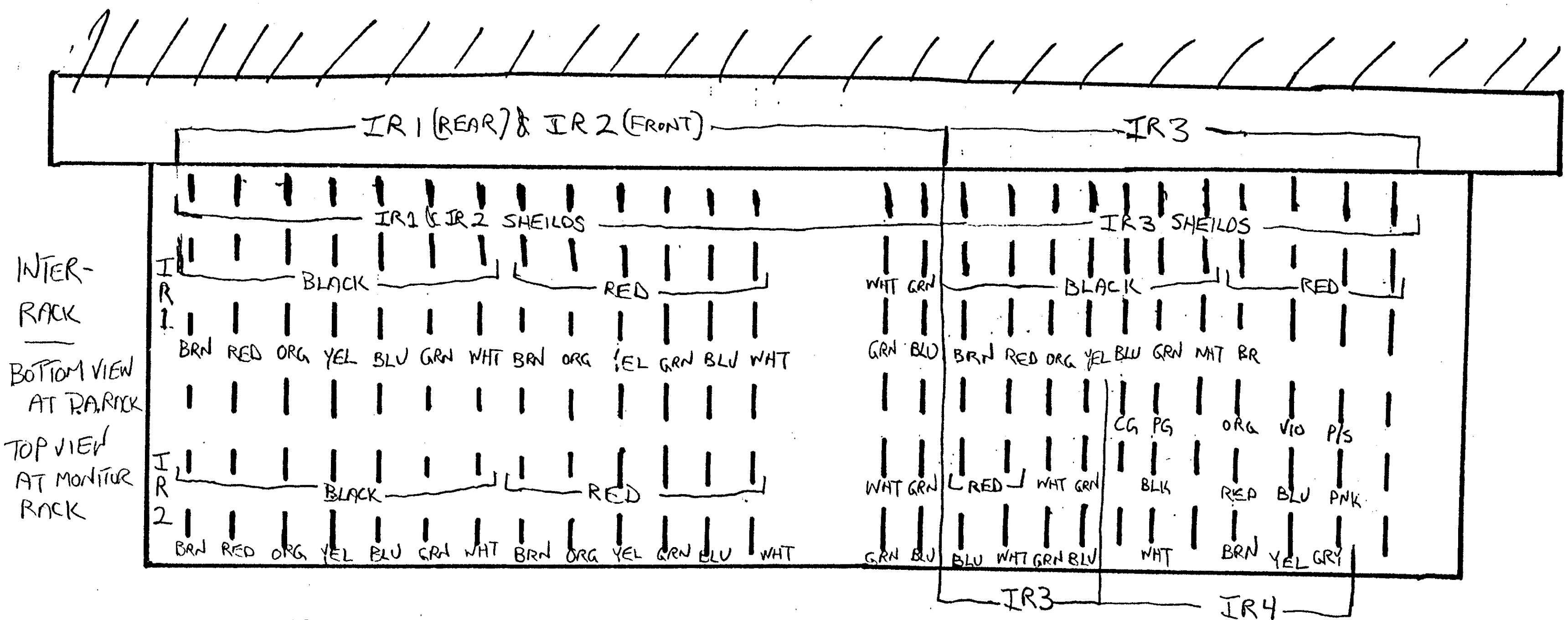
→ Red/Hot connections for cable C1, pairs 8→13 are reversed!!

→ In Studio B, The DC cable doesn't come To The Terminal block, but To a barrier strip.

INTER-RACK CABLE ASSIGNMENTS

Pair Number	Signal	Notes
C1-1	Mod. Mon. flasher signal	grn(+)/blk(- gnd)
C1-2	FM Monitor out L	
C1-3	FM Monitor out R	
C1-4	Spare Tuner out L	
C1-5	Spare Tuner out R	
C1-6	Studio B mono out	Padded to mono on wall at DA rack
C1-7	Studio C mono out	Padded to mono on wall at DA rack
C1-8	Intercom L-pad Defeat	brn/wht
C1-9	Phone/Doorbell LED feed	blk/red
C1-10	Doorbell lights	yel(+)/vio(-), jumped from terminal strip
C1-11	Spare Remote Line L	red(+)/grn(-)
C1-12	Spare Remote Line R	yel(+)/blk(-)
C2-1	Studio A N.O. mic relay	org/yel
C2-2	Studio B N.O. mic relay	blu/grn
C2-3	Studio C N.O. mic relay	brn/wht
C2-4	PA Speakers 1 Studios	org/yel
C2-5	PA Speakers 2 Halls	blu/grn
C2-6	EBS Gen. start switch	
C2-7	EBS Generator tone out	
C2-8	EBS Mon. Alert relay	WIPER to Gnd(blu), N.O. to HOT(wht)
C2-9	Spare to Monitor rack	
C2-10	Spare to Monitor rack	
C3-1	Upstairs doorbell switch	red/org from switch, blk/yel to Monitor rack
C3-2	Upstairs doorbell light	blk/yel to Monitor rack
C3-3	Intercom line	blu/wht(+) from Monitor rack, jumped from teminal strip
C3-4	Intercom line with ring	red/grn from Monitor rack, jumped from terminal strip
C3-5	MacPhie Pub line 1	
C3-6	MacPhie Pub line 2	
C3-7	R/C from Transmitter	padded to isolate
C3-8	R/C to Transmitter	padded to isolate
C3-9	P/L to Transmitter 1 L	
C3-10	P/L to Transmitter 2 R	

XMAS TREE: C1/C2/C3/C4... side view



IR1, IR2, IR3

Protocol: 15-Pair audio

1 — brown/black
2 — red/black
3 — orange/black
4 — yellow/black
5 — blue/black
6 — green/black
7 — white/black
8 — brown/red
9 — orange/red
10 — yellow/red
11 — green/red
12 — blue/red
13 — white/red
14 — green/white
15 — blue/Green

IR4

Protocol: DC cable

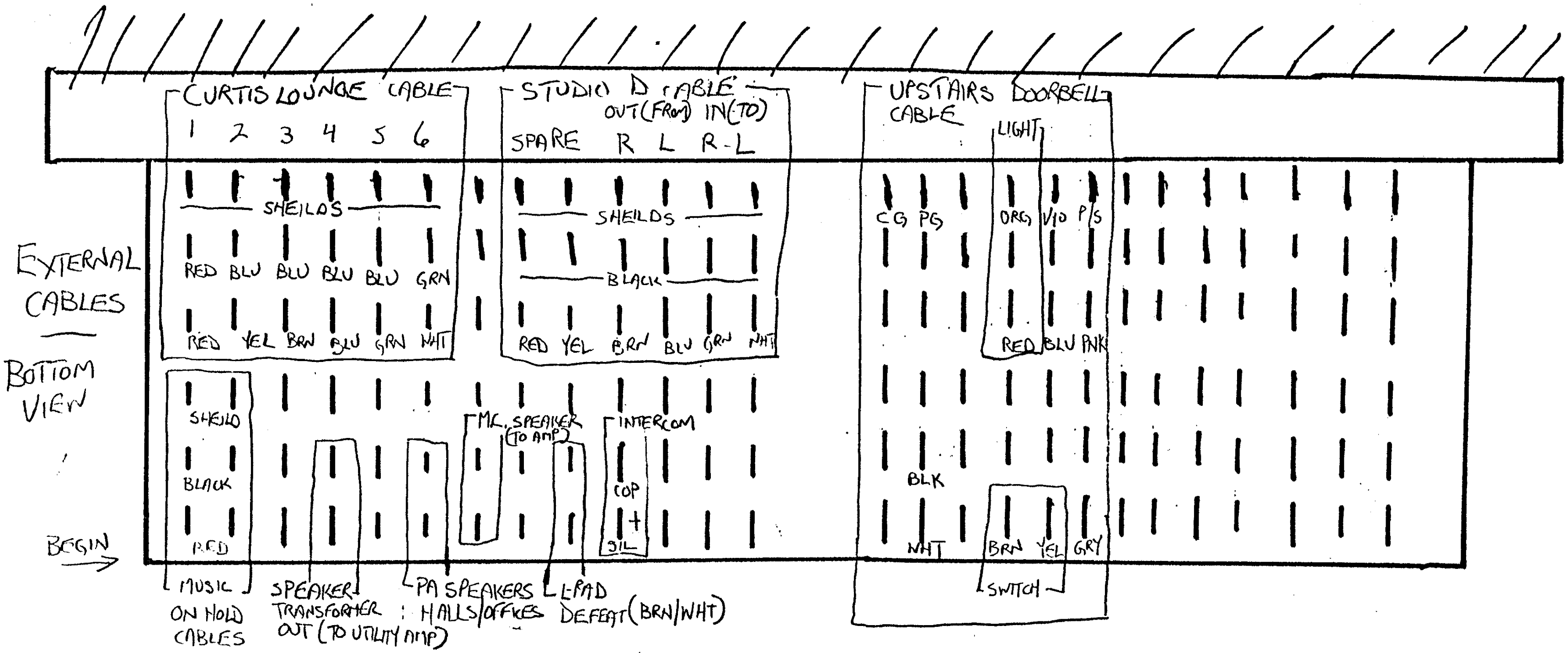
9 wire + 1 pair + 2 shields

Common ⏚ = CG
Pair ⏚ = PG
pair — white/Black
Brown
Red
Orange
Yellow
blue
Violet
Grey
Pink
Pink/Striped = P/s

Inter-Rack Tree

(Located behind Blue rack and behind grey rack in Master Control).

XMAS TREE: C1/C2/C3/C4... side view



Protocol: 15-Pair audio

1 — brown/black
2 — red/black
3 — orange/black
4 — yellow/black
5 — blue/black
6 — green/black
7 — white/black
8 — brown/red
9 — orange/red
10 — yellow/red
11 — green/red
12 — blue/red
13 — white/red
14 — green/white
15 — blue/Green

Protocol: DC cable

9 wire + 1 pair + 2 shields

common ⏚ = CG
Pair ⏚ = PG
pair — white/Black
Brown
Red
orange
Yellow
blue
Violet
Grey
Pink
Pink/Striped = P/S

External Cables Tree

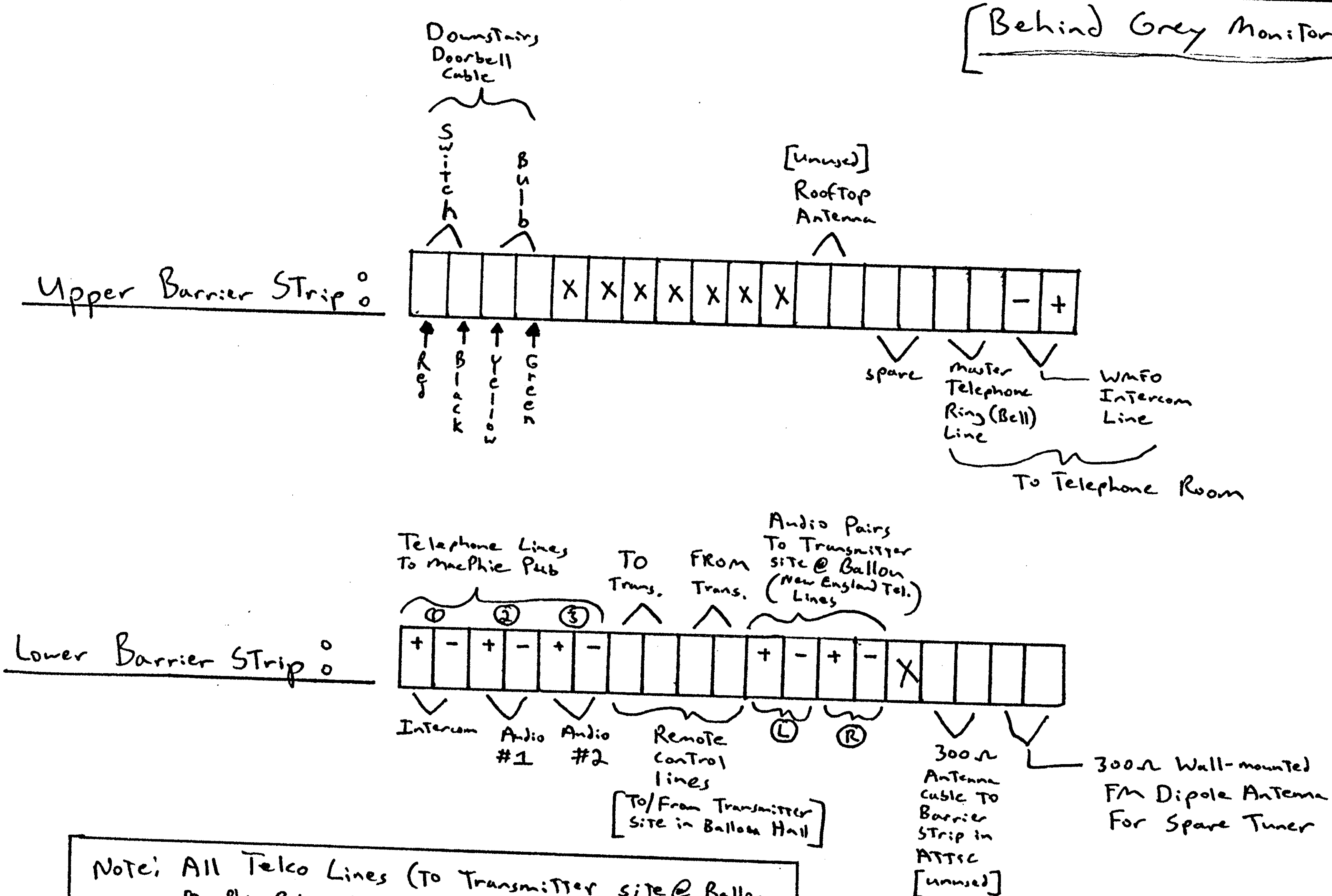
(Located on wall behind blue rack in Master Control)

MISCELLANEOUS TERMINAL STRIPS NEAR MONITOR (GRAY) RACK
(On TELCO pairs: solid = hot, stripe = ground)

Color Code	Signal	Notes
	TELCO Room (near Studio C) Lines (6 pair twisted TELCO)	
blu/stripe	Intercom line to TELCO box	blu/wht from rack
grn/stripe	Ring signal from TELCO box	pnk/vio to rack
	MacPhie Pub Remote Lines (6 pair twisted TELCO)	
blu/stripe	Intercom line	red/grn to inter-rack tree
org/stripe	Audio line 1	red/blk to inter-rack tree
grn/stripe	Audio line 2	grn/yel to inter-rack tree
	Downstairs Doorbell Line (4 condutor TELCO)	
	Switch contact	beige/gry to rack
	Light	brn/org from rack

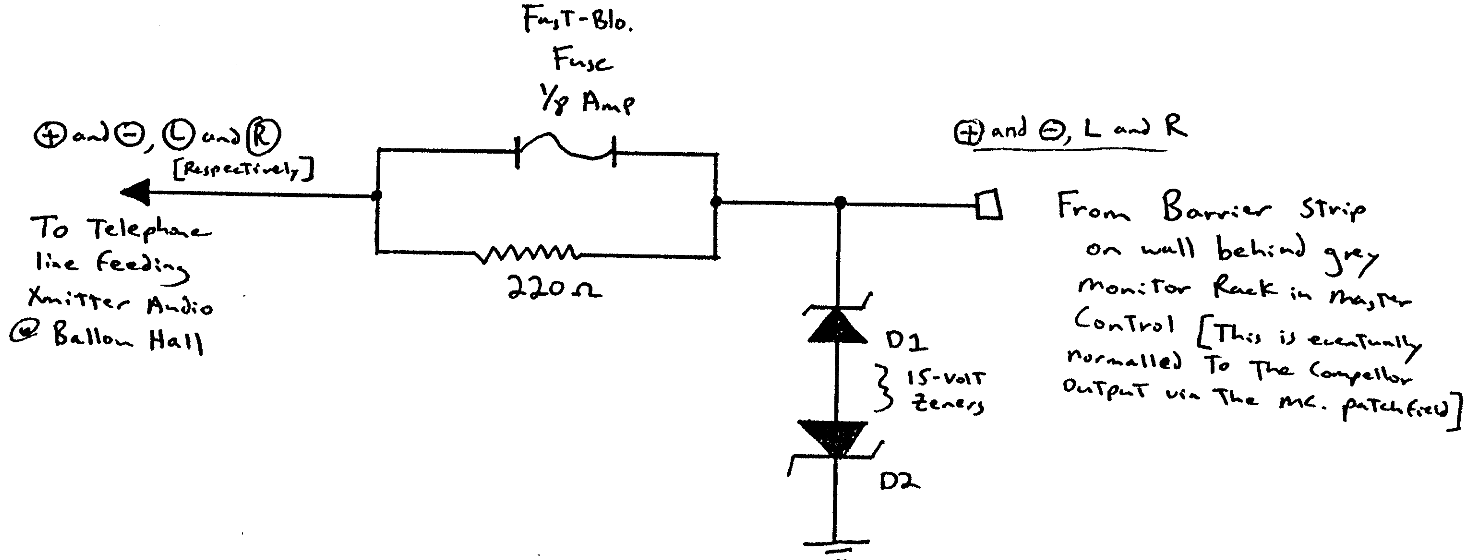
Layout + Purposes of Barrier Strips mounted on wall of Master Control

[Behind Grey Monitor Rack]



Note: All Telco Lines (To Transmitter site @ Ballou, MacPhie Pub, and telephone room) come to this Barrier Strip before being wired into The Studio cabling scheme.

Transmitter Telephone Line Protection Circuit



There are 4 copies of This circuit installed in series with each conductor of The Audio line To Ballon Hall. They are mounted on a piece of wood fixed To The Wall of Master Control somewhere above The Telephone punch-block behind The grey Monitor rack. In The event of a phone-line short, The 1/8 Amp fuse will blow, protecting The line from over heating. In The event of a large spike being introduced into The phone line (ie. Lightning or Telephone Ring signal), The surge into our audio equipment will be limited by D1 and D2 (if The surge is really bad, The 1/8 Amp fuse may also blow out), Thereby protecting our audio Equipment. If The audio feed To The Transmitter spontaneously degrades or drops in amplitude on at least one channel, The 1/8 Amp fuses and Diodes D1/D2 should be checked on This circuit!

B) Patchfield Layouts

- Studio A
- Studio B
- Studio C
- Master Control

WMFO Studio A
Patch Bay I (upper)

Fader 1c	{ 49 50	<->	1 2 }	(spare)
Fader 2c	{ 51 52	<->	3 4 }	PA Line out
Fader 3a	{ 53 54	<->	5 6 }	Turntable 1
Fader 3b	{ 55 56	<->	7 8 }	Jacks J1
Fader 3c	{ 57 58	<->	9 10 }	Curtis Lounge
Fader 4a	{ 59 60	<->	11 12 }	Turntable 2
Fader 4b	{ 61 62	<->	13 14 }	Interface Mix out
Fader 4c	{ 63 64	<->	15 16 }	McPhie Pub
Fader 5a	{ 65 66	<->	17 18 }	Turntable 3
Fader 5b	{ 67 68	<->	19 20 }	Cassette 2 out
Fader 5c	{ 69 70	<->	21 22 }	Housetone (mono)
Fader 6a	{ 71 72	<->	23 24 }	Cart 1
Fader 6b	{ 73 74	<->	25 26 }	Telephone out
Fader 6c	{ 75 76	<->	27 28 }	Studio B out
Fader 7a	{ 77 78	<->	29 30 }	Cart 2
Fader 7b	{ 79 80	<->	31 32 }	Cassette 1 out
Fader 7c	{ 81 82	<->	33 34 }	Studio C out
Fader 8a	{ 83 84	<->	35 36 }	Cart 3
Fader 8b	{ 85 86	<->	37 38 }	Reel out
Fader 8c	{ 87 88	<->	39 40 }	Studio D out
External Mon. in	{ 89 90	<->	41 42 }	FM Monitor out
Interface Mix in	{ 91 92		43 44 }	FM Monitor out (spare)
A-to-B 1	{ 93 94		45 46 }	A-to-C 1
A-to-B 2	{ 95 96		47 48 }	A-to-C 2

<-> = normal
Odd numbers = left channel
Even numbers = right channel
(revised 11/85)

WMFO Studio A
Patch Bay II (lower)

To M.C.	49 50	<->	1 2	DA out 1
Reel in	51 52	<->	3 4	DA out 2
Cassette 1 in	53 54	<->	5 6	DA out 3
Cassette 2 in	55 56	<->	7 8	DA out 4
Interface Send in	57 58	<->	9 10	DA out 5
DA in	59 60	<->	11 12	Program out
To M.C. (mono)	61 62	<->	13 14	Audition out
15 dB Pad	63 64		15 16	Jacks J2
15 dB Pad	65 66		17 18	Split
Mono Bridge	67 68		19 20	Split
Mono Bridge	69 70		21 22	Split (in/out)
Gas Light in	71 72	<->	23 24	DA out 6
	73 74		25 26	
	75 76		27 28	
	77 78		29 30	
	79 80		31 32	
	81 82		33 34	
	83 84		35 36	
	85 86		37 38	
	87 88		39 40	
	89 90		41 42	
	91 92		43 44	
	93 94		45 46	
	95 96		47 48	

<-> = normal
Odd numbers = left channel
Even numbers = right channel
(revised 11/85)

KEY -(Odd Numbers-Left Channel
Even Numbers-Right Channel

UPPER PATCH BAY

WMFO-FM
STUDIO B
11/84 revised

DA In	49 / 50	↔	1 / 2	Program Out
Interface Send In	51 / 52	↔	3 / 4	DA 1 Out
Interface Mix In	53 / 54		5 / 6	Interface Mix Out
Tape 1 (Reel) In	55 / 56	↔	7 / 8	DA 2 Out
Tape 2 (Reel) In	57 / 58	↔	9 / 10	DA 3 Out
Cart 1 In	59 / 60	↔	11 / 12	DA 4 Out
Cassette In	61 / 62	↔	13 / 14	DA 5 Out
Fader (Cart 2)	63 / 64	↔	15 / 16	Cart 2 Out
Fader (TT 1)	65 / 66	↔	17 / 18	Turntable 1 Out
Fader (Tape 1)	67 / 68	↔	19 / 20	Tape 1 (Reel) Out
Fader (TT 2)	69 / 70	↔	21 / 22	Turntable 2 Out
Fader (Tape 2)	71 / 72	↔	23 / 24	Tape 2 (Reel) Out
Fader (Cart 1)	73 / 74	↔	25 / 26	Cart 1 Out
Fader (Cassette)	75 / 76	↔	27 / 28	Cassette Out
Fader (Telephones)	77 / 78	↔	29 / 30	Telephones Out
Fader (Studio C)	79 / 80	↔	31 / 32	Studio C Out
	81 / 82		33 / 34	Studio A Out
Ext. Monitor In	83 / 84	↔	35 / 36	FM Monitor Out
EQ In	85 / 86		37 / 38	EQ Out
Space Station In	87 / 88		39 / 40	Space Station Out
DBX In	89 / 90		41 / 42	DBX Out
To M.C.	91 / 92	↔	43 / 44	Audition Out
BC 1	93 / 94		45 / 46	AB 1
BC 2	95 / 96		47 / 48	AB 2

KEY -(Odd Numbers-Left Channel
Even Numbers-Right Channel

LOWER PATCH BAY

WMFO-FM
STUDIO B
11/84 revised

	-(49 50	1 2)-	
	-(51 52	3 4)-	
	-(53 54	5 6)-	
	-(55 56	7 8)-	
	-(57 58	9 10)-	
	-(59 60	11 12)-	
	-(61 62	13 14)-	
	-(63 64	15 16)-	
	-(65 66	17 18)-	
	-(67 68	19 20)-	
	-(69 70	21 22)-	
	-(71 72	23 24)-	
	-(73 74	25 26)-	
15dB PAD	-(75 76	27 28)-	
15dB PAD	-(77 78	29 30)-	Split
PA Line Out	-(79 80	31 32)-	Split
House Tone Out	-(81 82	33 34)-	Split (IN/OUT)
Jacks E2 Receive	-(83 84	35 36)-	Jacks E2 Send
Jacks E1 Receive	-(85 86	37 38)-	Jacks E1 Send
Jacks J1	-(87 88	39 40)-	Jacks J2
Mono Bridge 2	-(89 90	41 42)-	Mono Bridge 1
Mono Bridge 2	-(91 92	43 44)-	Mono Bridge 1
Pitch Shifter In	-(93 94	45 46)-	Pitch Shifter Out
DA6 Out	-(95 96 ↔	47 48)-	To M.C.

KEY -(Odd Numbers-Left Channel
Even Numbers-Right Channel

PATCH BAY

WMFO-FM
STUDIO C
12/84 revised

DA In	49/50	↔	1/2	Program Out
To M.C.	51/52	↔	3/4	DA 1 Out
Tape (Reel) In	53/54	↔	5/6	DA 2 Out
Cassette In	55/56	↔	7/8	DA 3 Out
Cart 2 Input	57/58	↔	9/10	DA 4 Out
Interface Send In	59/60	↔	11/12	DA 5 Out
Interface Mix In	61/62		13/14	Interface Mix Out
Mono Bridge	63/64		15/16	Mono Bridge
Fader 1 (A)	65/66	↔	17/18	Cart 1 Out
Fader 2 (B)	67/68	↔	19/20	Cart 2 Out
Fader 3 (A)	69/70	↔	21/22	Turntable 1 Out
Fader 4 (A)	71/72	↔	23/24	Turntable 2 Out
Fader 4 (B)	73/74	↔	25/26	Telephones Out
Fader 5 (A)	75/76	↔	27/28	Tape (Reel) Out
Fader 5 (B)	77/78	↔	29/30	Cassette Out
Fader 3 (B)	79/80	↔	31/32	Studio B Out
Jacks J1	81/82		33/34	Studio A Out
Jacks J2	83/84		35/36	PA Line Out
Ext. Monitor In	85/86	↔	37/38	FM Monitor Out
EQ In	87/88		39/40	EQ Out
Split	89/90		41/42	House Tone Out
Split	91/92		43/44	Split (IN/OUT)
AC 1	93/94		45/46	BC 1
AC 2	95/96		47/48	BC 2

WMFO Master Control
Patch Bay I (upper)

P/L to Transmitter	49 / 50	<->	1 / 2	Compressor out
DA (A) in	51 / 52	<->	3 / 4	Studio A out
DA (B) in	53 / 54	<->	5 / 6	Studio B out
DA (C) in	55 / 56	<->	7 / 8	Studio C out
DA (Monitor) in	57 / 58	<->	9 / 10	FM Monitor out
To Studio Switcher (A)	59 / 60	<->	11 / 12	DA (A) out 1
To Studio B	61 / 62	<->	13 / 14	DA (A) out 2
To Studio C	63 / 64	<->	15 / 16	DA (A) out 3
To Studio Switcher (B)	65 / 66	<->	17 / 18	DA (B) out 1
To Studio A	67 / 68	<->	19 / 20	DA (B) out 2
To Studio C	69 / 70	<->	21 / 22	DA (B) out 3
To PA Mixer (B)	71 / 72	<->	23 / 24	DA (B) out 4
To Studio Switcher (C)	73 / 74	<->	25 / 26	DA (C) out 1
To Studio A	75 / 76	<->	27 / 28	DA (C) out 2
To Studio B	77 / 78	<->	29 / 30	DA (C) out 3
To PA Mixer (C)	79 / 80	<->	31 / 32	DA (C) out 4
To A FM Mon.	81 / 82	<->	33 / 34	DA (Mon.) out 1
To B FM Mon.	83 / 84	<->	35 / 36	DA (Mon.) out 2
To C FM Mon.	85 / 86	<->	37 / 38	DA (Mon.) out 3
To D FM Mon.	87 / 88	<->	39 / 40	DA (Mon.) out 4
To Music-on-Hold	89 / 90	<->	41 / 42	DA (Mon.) out 5
To Curtis (lines 3&4)	91 / 92	<->	43 / 44	DA (Mon.) out 6
(spare) REMOTE TERMINALS (on wall under window)	93 / 94	<->	45 / 46	DA (Mon.) out 7 (spare)
Transmitter R/C tap	95 / 96		47 / 48	DA (A) out 4 (spare)

<-> = normal
Odd numbers = left channel
Even numbers = right channel
(revised 11/85)

WMFO Master Control
Patch Bay II (middle)

To Studio A	49 50	<->	1 2	Studio D out
To Studio A	51 52	<->	3 4	Curtis (lines 1&2)
To Studio A	53 54	<->	5 6	MacPhie Pub
A-to-B 1 (to B)	55 56	<->	7 8	A-to-B 1 (to A)
A-to-B 2 (to B)	57 58	<->	9 10	A-to-B 2 (to A)
A-to-C 1 (to C)	59 60	<->	11 12	A-to-C 1 (to A)
A-to-C 2 (to C)	61 62	<->	13 14	A-to-C 2 (to A)
B-to-C 1 (to C)	63 64	<->	15 16	B-to-C 1 (to B)
B-to-C 2 (to C)	65 66	<->	17 18	B-to-C 2 (to B)
To Studio A (mono)	67 68	<->	19 20	Housetone out (A)
To Studio B (mono)	69 70	<->	21 22	Housetone out (B)
To Studio C (mono)	71 72	<->	23 24	Housetone out (C)
To Studio A	73 74	<->	25 26	Tel. Sys. Audio out (A)
To Studio B	75 76	<->	27 28	Tel. Sys. Audio out (B)
To Studio C	77 78	<->	29 30	Tel. Sys. Audio out (C)
Tel. Sys. Audio in (A)	79 80	<->	31 32	Tel. Audio Feed from A (mono)
Tel. Sys. Audio in (B)	81 82	<->	33 34	Tel. Audio Feed from B (mono)
Tel. Sys. Audio in (C)	83 84	<->	35 36	Tel. Audio Feed from C (mono)
Utility Amp in	85 86	<->	37 38	Spare Tuner out
Jacks J1	87 88		39 40	B Audition out
Jacks J2	89 90		41 42	A Audition out (mono)
To Studio Switcher (EBS)	91 92	<->	43 44	EBS Tone out (mono)
Intercom	93 94	<->	45 46	Curtis (lines 5&6)
EQ in	95 96		47 48	EQ out

<-> = normal
Odd numbers = left channel
Even numbers = right channel
(revised 11/85)

WMFO Master Control
Patch Bay III (lower)

Mono Bridge 2	49 50		1 2	Mono Bridge 1
Mono Bridge 2	51 52		3 4	Mono Bridge 1
Spilt 2	53 54		5 6	Split 1
Split 2	55 56		7 8	Split 1
Split 2 (in/out)	57 58		9 10	Split 1 (in/out)
15 dB Pad	59 60		11 12	15 dB Pad
Compressor in	61 62	<->	13 14	Studio Switcher out (main)
	63 64		15 16	Studio Switcher out (aux.)
	65 66		17 18	
	67 68		19 20	
	69 70		21 22	
	71 72		23 24	
	73 74		25 26	
	75 76		27 28	
	77 78		29 30	
	79 80		31 32	
	81 82		33 34	
	83 84		35 36	
	85 86		37 38	
	87 88		39 40	
	89 90		41 42	
	91 92		43 44	
	93 94		45 46	
	95 96		47 48	

<-> = normal
Odd numbers = left channel
Even numbers = right channel
(revised 11/85)

THE WMFO PATCHFIELD PADS

As noted on the patchfield diagrams, the patchfield arrays in all studios and Master Control have at least one "3-way split" and "mono bridge". These are passive resistor networks (wired physically onto the Xmas Trees associated with the corresponding patchfield), which allow one to combine two independent stereo outputs ("split"), or sum both stereo channels into mono ("bridge"). Diagrams of the two types of resistor pads are given on the following page, and a summary of operational modes is included below.

1) 3-Way Split

The 3-way split essentially follows the function of a stereo "Y" cord. It allows one signal to be split into two paths, or combines two independent signals into one. This "splitting" is accomplished in a balanced fashion; ie. all patches plugged into the split will sink or source somewhere around 600 Ohms (each leg of the "Y" is isolated by a resistor to prevent shorting). These pads are denoted simply by the phrase "split" on the patchfields; Studios A,B,&C each have one split on their fields, while Master Control has two.

Keeping the "Y-cord" analogy in mind, the "in/out" designation on the split patches denotes the base of the "Y", while the two other split locations correspond to the arms of the "Y". In practical terms, when combining two outputs via the split to form a sum, the two outputs to be combined are patched into the non-descript "split" locations (the arms of the "Y"), and the composite signal is available at the "in/out" port (the base of the "Y"). Conversely, when splitting one output into two, the original output is patched into the "in/out" port (the base of the "Y"), fanning it into two outputs available at the other "split" locations (the arms of the "Y"). In order to prevent crosstalk, funny levels, or excessive loading, this policy should always be followed.

2) Mono Bridge

The Mono Bridge patch allows one to bridge a stereo output into mono (ie. both channels are combined). Studios A&C each have one mono bridge wired into their fields, while Master Control and Studio B each have two.

The mono bridge is very simple to use; simply patch the stereo signal desired to be "mono'ed" into one port of the bridge (it doesn't matter which patchfield port is used), and the mono signal is available at the other patchfield port. Like the 3-way split, the mono bridge uses resistor networks to maintain somewhere around a 600-Ohm impedance and prevent direct shorting of Left and Right signals.

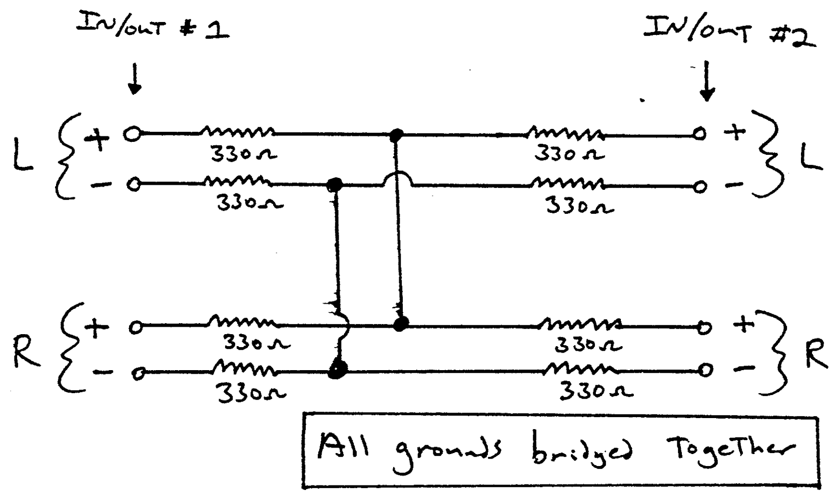
The mono bridge is perhaps the most direct means of combining both channels of an audio source into mono (the Interface circuit provides a more flexible means of doing this; the Studio B board and Studio Switcher also possess the capability of being switched into a "mono" mode). One may desire to use a patch of this sort if, for example, a tape that is recorded onto a single channel must be played over the air;

naturally, one desires audio on both channels, thus the tape deck output can be patched first into the mono bridged before being patched into the mixing board.

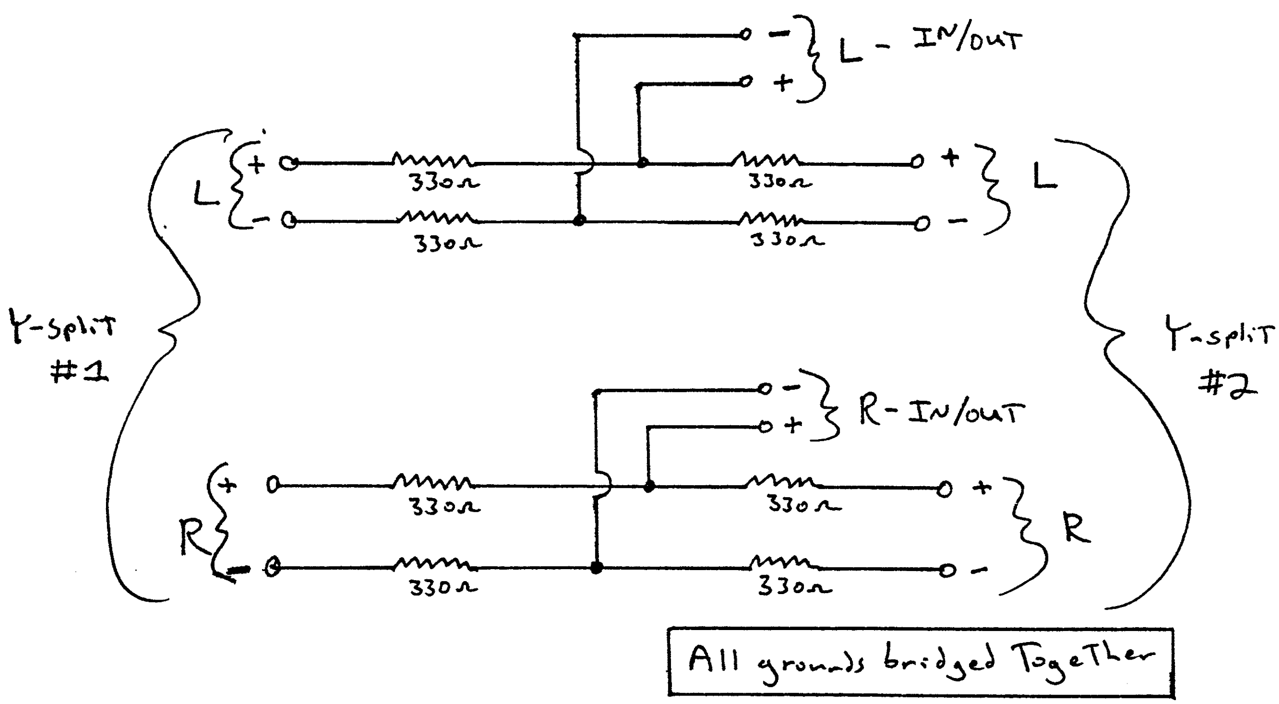
3) Other Patchfield Pads

Studio B also has a "15 dB Pad" wired into its patchfield; any audio source patched through is reduced in amplitude by 15 dB. This patch is used to reduce the level of "hot" audio sources (such as the Biamp Mixer) before introducing them into the Gates mixing board.

WMFO Patchfield Pads



1) Mono Bridge



2) 3-Way Split

Notes on Interstudio Patches

Each studio possesses two independent lines to the other two studios which may be accessed via the patchfields (ie. Studio A has 2 lines to Studio B labeled "A-to-B", and 2 lines to Studio C labeled "A-to-C"). A source patched into "A-to-C 1" in Studio C is available at the corresponding "A-to-C 1" port in Studio A, and so on for the other interstudio patch lines. These allow signals to be routed to other studios for processing, enable devices in one studio to be directly potted-up in another, etc. They provide an extremely flexible environment for interstudio coordination and production.

All interstudio lines are also normalled through the Master Control patchfields, thus they may be interrupted, comandeered, and re-routed elsewhere. This provides yet another level of flexibility; for instance, suppose one would like to use four stereo balanced lines between Studios B and C. The normal protocol gives only two "B-to-C" lines, however the "A-to-B" and "A-to-C" lines may be re-routed in Master Control so they really run between Studios B and C, thereby creating the two additional needed lines. When patching and re-assigning these interstudio defaults in Master Control, one must remember which end of the line one is working with. To eliminate any ambiguity, the Master Control patchfields are clearly labeled with the destination of the cable concerned; ie. the "A-to-C" line is normalled across "A-to_C (to A)" and "A-to-C (to C)" patches.

Several other audio lines run from the studios into Master Control (ie. program lines, studio outputs, telephone feeds, etc.). These may also be overridden via the Master Control patchfield to provide additional interstudio cables. Since the studio structure is defined by the Master Control patchfield, one must remember to remove all patches thrown there after work is completed, so signals will appear where they should, and the studio can run as it normally does.

C) Transmitter Power Monitoring

TRANSMITTER POWER CALCULATION

The table below was used to calculate Transmitter output power from voltage and current readings. This is now performed by an analog multiplier and the product is available on Remote Control channel #7 (see schematic for "Transmitter Audio and Power Monitor"). This reading is normalized such that 1000 = 100% = nominal operating power.

Since this signal posesses some drift, a forward power tap directly off the Transmitter's directional coupler is available on Remote channel #3. This is also normalized such that 1000 = 100% = nominal operating power. This channel should be logged to monitor Transmitter output power.

The power may now be increased or decreased via the "Raise" and "Lower" buttons on channel #3.

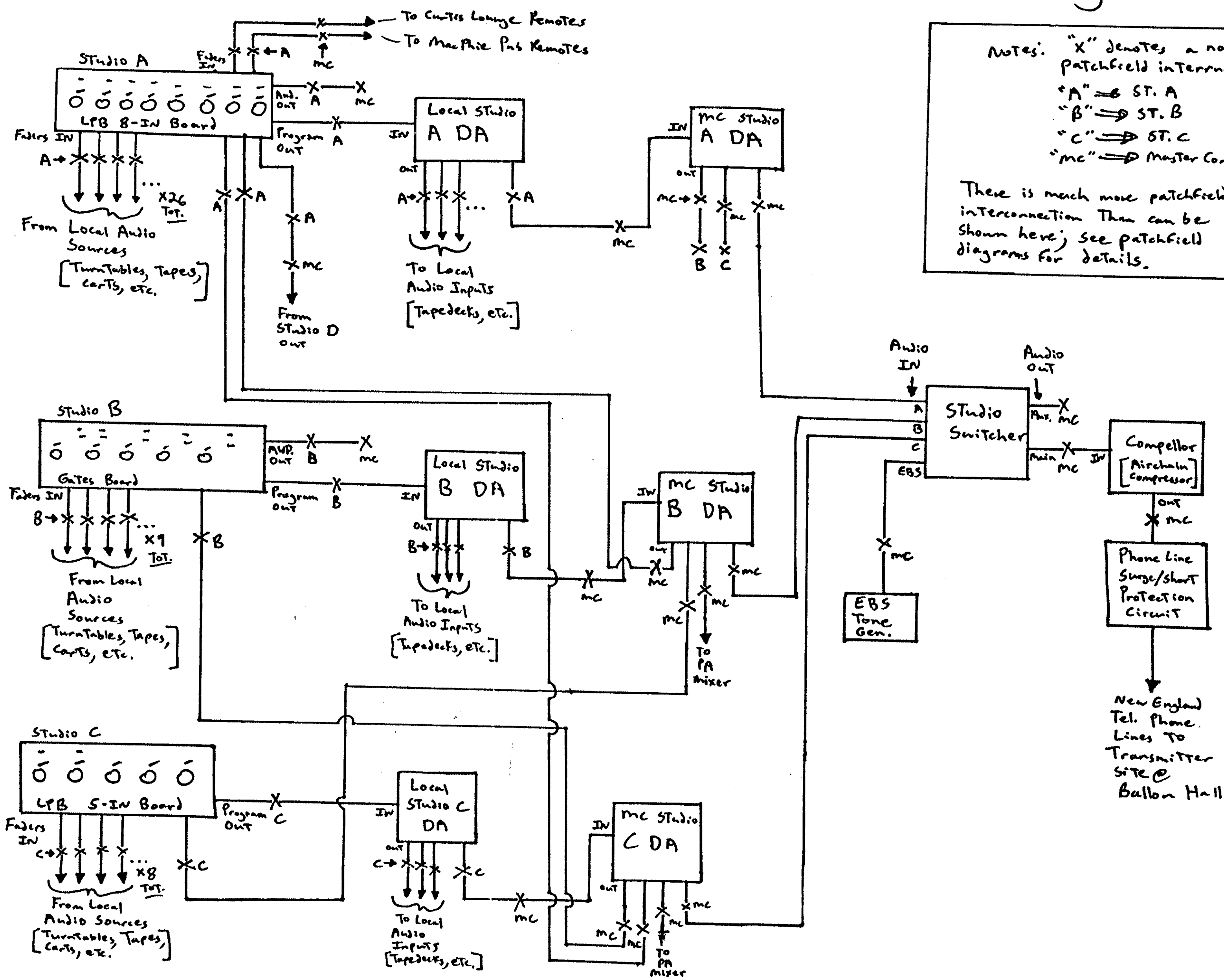
Old Power Calculation Table:

To calculate power match voltage across top with current along side. Lower current if power is under dotted line.

	47	48	49	50	51	52
1.8	55.84	57.34	58.21	59.40	60.59	61.78
1.9	58.94	60.19	61.45	62.70	63.95	65.21
2.0	62.04	63.36	64.68	66.00	67.32	68.64
2.1	65.14	66.53	67.91	69.30	70.69	72.07
2.2	68.24	69.70	71.15	72.60	74.05	75.50

D) General Studio Structure

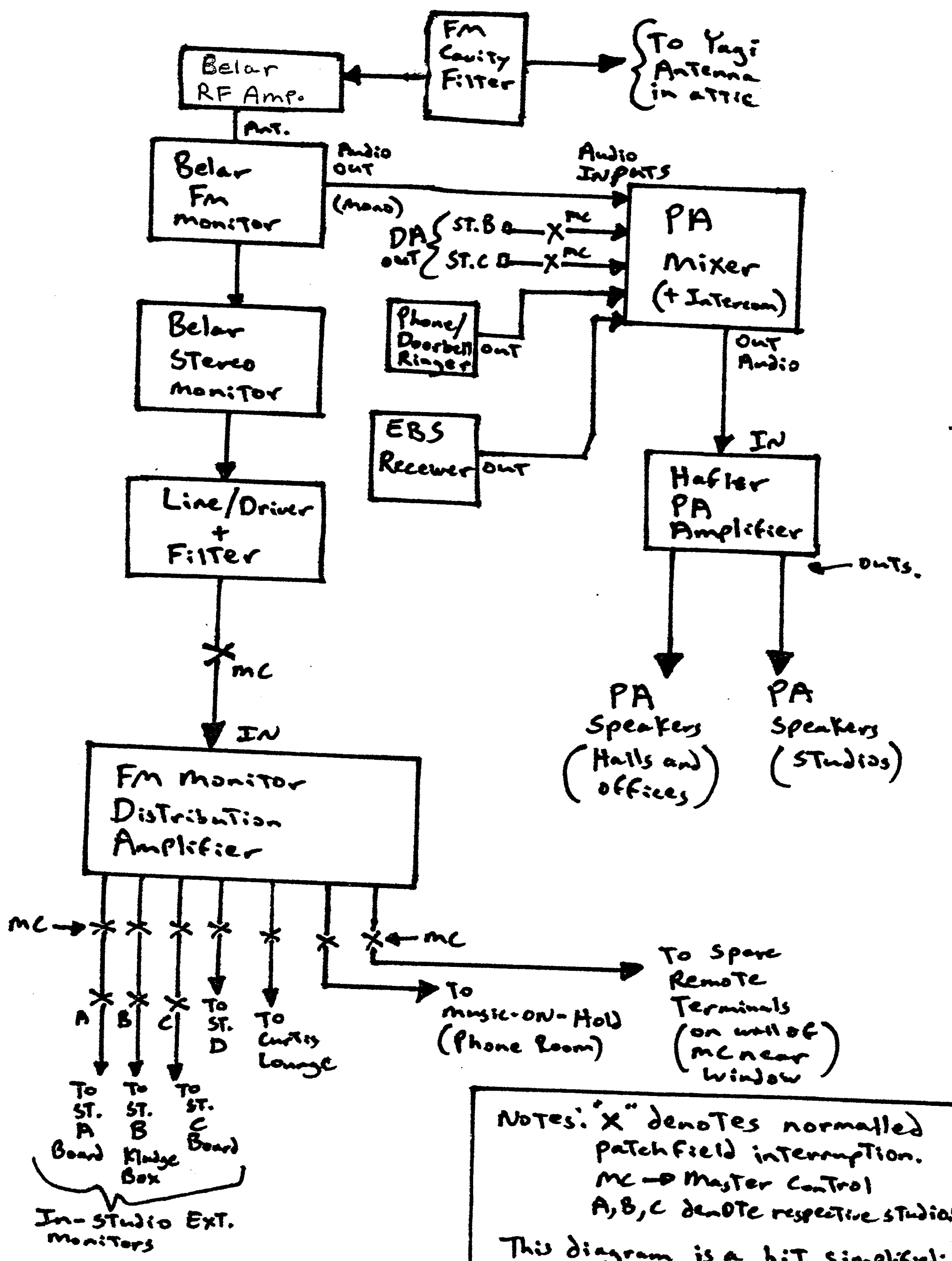
WMFO Studio/Airchain Audio Organization



Notes: "X" denotes a normalled patchfield interruption.
"A" ⟹ ST. A
"B" ⟹ ST. B
"C" ⟹ ST. C
"mc" ⟹ Master Control

There is much more patchfield interconnection than can be shown here; see patchfield diagrams for details.

WMFO Monitor Audio Chain



Notes: "X" denotes normalled patchfield interruption.
MC → Master Control
A, B, C denote respective studios

This diagram is a bit simplified; see text and patchfield layouts for more details.

SECTION THREE

APPENDIX

- Relevant data sheets
- Important Company Names, Addresses, and Contacts
- WMFO Insurance Estimates
- Press Releases, Crank Letters, and other Tech Crew Trivia

Chris, Fred, and whoever comes after.....

Just a quick note to specify some ideas and priorities to future upgrades with regards to the WMFO proposed budget. Since we recently finished upgrading the boards somewhat during the Tech Crew overhaul, they aren't in terrible shape. We also customized the boards considerably (ie. telephone patch, headphone amplifiers, external monitor feed, multi-microphone feeds, patchfield layout, etc.). Any new boards installed in WMFO should also be modified in this fashion so that they will fit into the station protocol (this may not be too simple a job). I don't think that a "cheap" board is by any means worthwhile. The board used at WMBR and WZBC is becoming a standard these days, and has an excellent reputation. This is a "Wheatstone A500" eight fader board, and costs around $14,000. If you want to replace the board, I'd encourage you to wait until you save such dough (ie. fund raisers, benefits, sponsorships, and good relations with the university may help). It's not worthwhile at this point to comprimise in haste (things are working fine with a bit of routine maintainance; this is required with any board). Wait until you have enough bucks for something good before you move on this issue. Also remember; you either need an experienced and willing tech. crew to install the board (not an easy job to do it properly), or get ready to shell out $1000 - $2000 to have somebody come in and do it.

The boards already installed can be modified fairly easily (and much less inexpensively) to upgrade their performance. Because of our airchain compressor, the discrete fader steps can become quite noticable when they're set at low volume or the source material is very soft. This can be remidied by replacing the critical faders (ie. the three turntable faders in A) with "continuous" rotary faders that have no discrete steps. These can be purchased from broadcasting outlets (perhaps ask LPB for advice; make sure you get units with integral cue switches), and probably cost around $100. apiece. They are easily installed. Make sure you continue to lubricate the remaining switch faders approx. every six months with the appropriate grease to minimize "scratchyness" and extend their lifespan. I've noticed occasional "glitching" in the board switches; whenever this happens, I've repaired it fairly easily with a hit of "tuner cleaner" spray. This tradition should also keep the switches operational for a while.

Another "obvious" improvement to the board audio quality is to replace all isolation transformers (both input and output) with high-quality devices (ie. Jensens). This is harder work, and is considerably more expensive. I don't think this is really needed at WMFO (the board quality is good enough for now), and when it comes time for this upgrade, it's probably better to get a new board.

The most important activity at WMFO should be to improve the transmitter and air signal. This can start now by having someone look at the tower and radiators (the reflected power [remote #4] is often up ferociously; it's running above 300 at the moment, and it should meter at 100 or so). I've heard mention that our tower guy wires are metallic. If this is so (and it nearly certainly is true), they can cause considerable multi-path and "fade-in/fade-out" effect in the signal. Get a tower service (ie. "Art Bump and the Beacon Tower Service" were recommended by Grady) to look at the situation, and get an estimate for replacement of our guy wires with a non-conducting composite, such as "Philly Strand". The antenna cable or its tie-point to the radiators may also be damaged (ie. the large reflected power). This should also be investigated.

Grady spoke of the possiblity of power boosts for all of the stations @ 91.5. This should be extrapolated and checked. Grady seems to be very difficult to locate these days, but the information actually comes from "Ed Perry", who is another engineer around here. This guy orgainizes power boosts, etc. for all college stations around, and has worked quite a bit with WZBC and WMLN. He should probably be contacted directly. Be careful, he may be a bit of a sleeze, but the oppertunity could be grand. If our power can go appreciably upward, we may need a new transmitter at some point, so this should be kept in mind for furture outlays....

Finally, misc. things around the station which were always needed can be entered into the budget. Stereo cart machines (incl. a triple decker in A) would be nice. A second operational cassette machine for A would also be good. The Otari for C has long been a needed luxury. This list can be extended almost forever.....

In closing, there is a lot which should be accomplished and investigated before coming near to the reality of buying a new board. The technical maintaince is almost non-existant. You guys should also budget money to bring someone competent in on a periodic basis (perhaps also on-call) to handle this. I've heard Ken Haverly mentioned, and he's a reasonable choice. The station rebuilding put things into a good state; with a bit of maintaince, it could last a while. As the station evolves, new innovation can be added wherever needed. Please make sure that things are done properly; a quick "kludge" will often cause more destruction than benefit, and I've seen plenty of the former at WMFO.

Good luck getting it all together,

Joe Paradiso 16-Feb.-87

TI-101
Telephone Interface

Data Sheet

The Symetrix TI-101 Telephone Interface is the most practical, yet economical device on the market today designed *specifically* for the connection of professional audio equipment to telephone lines in broadcast and production operations.
The TI-101 employs a carefully engineered electronic hybrid circuit which creates a maximum trans-hybrid loss, yielding effective isolation between your studio's send to the telephone line, and your caller return signal. Your net result is clear, intelligible telephone audio which helps you keep pace with today's rapid improvements in AM, FM, and TV audio.

The TI-101 provides you with not just some, but all of the features you need in a telephone interface. At the same time the unit is simple to connect, simple to adjust, and simple to operate. An output buss or mic pre-amp feed from your console connects to the TI-101 and sends your studio signal down the telephone line. The TI-101 allows you to optimize your send level for maximum intelligibility. At the same time you can adjust the unit's built-in send limiter to prevent overdrive of the telephone line. The return signal from the TI-101 to your console is just the caller's voice. The DJ or host voice has been effectively nulled out.

Impedance and level matching to and from the telephone line and your console are done correctly by the TI-101. In addition, you can bring the caller's voice up on your studio monitor speakers without feedback. There's no need to wear headphones if you don't want to.

One of the most important features of the TI-101 is that it is a true hybrid. This means that there is no objectionable "gating" as in "speakerphone" type phone boxes. You get natural, *two-way* conversation between host and caller.

Features

Caller equalization. A two band equalizer with 8dB of boost and cut at 400 Hz and 2.5k Hz brightens up the caller and enhances intelligibility.

Send limiter. The send signal passes through a limiter with user adjustable threshold level. When properly adjusted this limiter maximizes host level, prevents overdrive of the phone line, and helps improve the send and receive circuit isolation.

Receive compressor/expander. This circuit also has a user adjustable threshold level. Above threshold the compressor rides gain on the caller, helping to maximize level, and most importantly keeping things under control during caller-host "shouting matches." Below threshold a noise reducing expander takes over and is especially valuable for keeping long distance noise down during caller pauses.

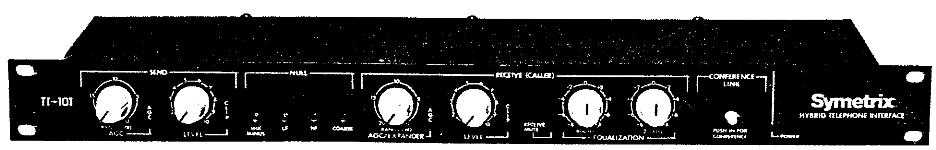
Caller mute. A user provided remote contact closure mutes the caller instantly without clicks or pops.

LED clip indicators are provided for simple optimization of send and receive levels. When setting levels there's no guesswork involved. The user simply increases the level controls until the LEDs flash on and then backs off on the setting slightly for the optimum operating point.

Conference linking. Two TI-101's may be selectively linked together for simultaneous conferencing between two incoming telephone lines and the host.

Level compatibility. Back-panel gain switches permit the TI-101 to operate with virtually any professional mixer or console.

Bandpass filtering. A sophisticated elliptical filter is provided on the TI-101's send section which prevents studio generated signals outside of the telephone passband from interfering with telephone company signaling frequencies. On the receive section, sharp "Chebychev" filters limit the passband from 300 to 3k Hz preventing spurious signals on the telephone line from interfering with your broadcast audio.



Symetrix

109 Bell Street
Seattle, Washington 98121, USA
Telephone (206) 624-5012
Telex 32-0281 GLOBECEN SEA

Professional Audio Products

BROADCAST SUPPLY WEST
7012 27TH ST. W.
TACOMA, WA 98466
1-800-426-8434

TI-101
Telephone Interface

Specifications

Input impedance		16.7K ohms (electronically balanced)
Output load impedance		> 600 ohms (transformer balanced)
Telephone port impedance		560 ohms (transformer isolated)
Nominal input and output level ranges		back panel switchable between −10dBm and +8dBm
Maximum output level		+20dBm
Maximum input level		+21dBm
Typical THD		.1%
Controls	send level, send limit, receive level, receive compress/expand, 400 Hz and 2.5k Hz equalization, conference	link, coarse, low frequency, and high frequency null adjust.
Visual indicators	LED's for indication of send clip, send limit, receive clip, receive compress/	expand, receive mute, and power on
Frequency response		(measured from telephone port to output port) 300 Hz to 3k Hz, ±3dB.
Typical transhybrid loss		20dB over the specified frequency bandwidth
Connectors	3 pin "XLR" type for input and output ports, dual banana posts for telephone tip and ring, ¼" phone jax for external	mute and conference interconnect cables.
Physical size		1¾" high, 19" wide, 6" deep (4.45 x 48.3 x 15.2 cm)
Shipping weight		7 lbs (15.4 kg)
Power requirements	60 Hz, 120 VAC, standard	50 Hz, 220 VAC, upon request
Construction	Aluminum front panel, plated steel chassis, all connectors pc mounted for	maximum reliability

In the interest of continuous product improvement and development, Symetrix, Inc., reserves the right to change or modify any of the above specifications or features, whenever, in our opinion, such a change produces an advantage mutual to our customers and ourselves.

The expander has been Disabled on the caller side of the TI-101 And Now The caller side has about 10db more gain.

The Send limit Control now acts as a Ducker providing 6 to 10db of ducking as Long as it is turned Counter-clock wise

Hope this mod works out for you. If you have any Questions feel free to call us at 282-2555

$250 + SHIPPING

Jeff Brown
Service Manager

SHIPPED OCT 0 8

STandard PinouT on a "TIP" Transistor

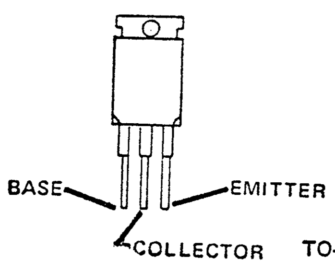
2020

Silicon	NPN
h_{FE}(min)	20
Max Ratings:	
V_{CBO}	100V
V_{CEO}	70V
V_{EBO}	7V
I_C	15A
f_T	3MHz
Dissipation	60W

Derated @ 150°C

Applications: Power amplifier and high-speed-switching.

Pin diagram viewed with heat sink down.

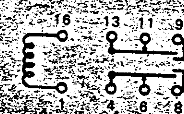


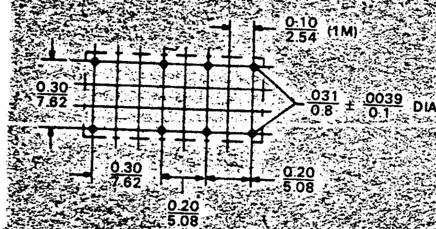
TO-220 CASE

STandard DIP Relay (STudio A Stereo Mics, and Mixer/InTercom)

275-213

12VDC
DPDT RELAY





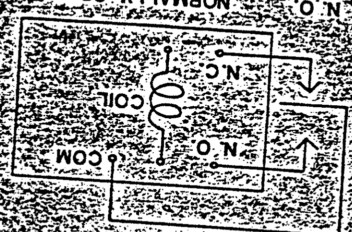
features
- Fits std. 16 pin IC socket
- Silver contacts for reliability
- Subminiature size — mounts in any position

absolute maximum ratings
Ambient Temperature: −25°C to +60°C
Continuous Coil Voltage: 15.5VDC at 20°C
Coil Dissipation: 1w at 20°C

electrical characteristics 2A5
Coil Resistance: 280 ohms ±10%
Pull In Voltage: 9VDC
Contact Rating: 1 amp at 125VAC
Pull In Time: < 6mS

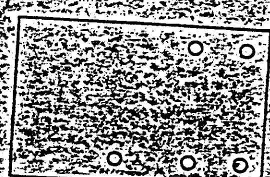
Standard "Intercom" Relay

SCHEMATIC DIAGRAM



N.O. NORMALLY OPEN
N.C. NORMALLY CLOSED
VOLTAGE 12VDC
RESISTANCE 1200Ω
CURRENT 10MA
CONTACT RATING 1AMP

PC TEMPLATE



TAIWAN

5A4

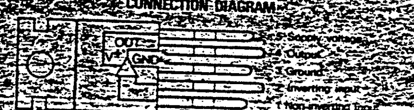
For UTILITY Amp

276-703A

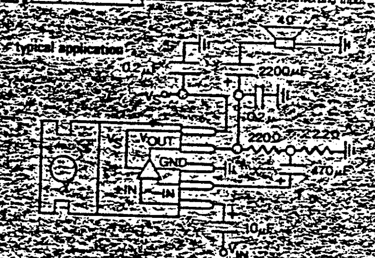
μPC2002 8-Watt Audio Amplifier IC

Direct replacement for TDA2002 and LM383

CONNECTION DIAGRAM



typical application



features

- Wide supply voltage range (5-18V)
- Short circuit/current limited/thermally protected
- Small size — minimum external parts
- Low distortion — High Z input (50kΩ typical)
- Selectable gain

absolute maximum ratings

Supply voltage — 20V
Peak output current — 3.5A
Junction temperature — 150°C

electrical characteristics

Power output — 5 watts RMS — 4Ω load
V_{cc} = (13.8V) — 8 watts RMS — 2Ω load

FOR FURTHER INFORMATION SEE RADIO SHACK'S SEMICONDUCTOR REFERENCE GUIDE

1A5

REFERENCES AND CONTACTS:

WMFO - Shipping Address
Tufts University, Curtis Hall, 490 Boston Av., Medford MA 02155

Loud & Clean Engineering (Consultant)
961-5007 Grady Moats

GEPCO Intl. (cable, connectors, & related parts), Chicago IL
312-644-5881 Gary Geppert

EAR Audio (Otari service), Mass. Av, Boston MA
262-0155 David Butler

Milam Audio, Peakin IL
309-346-3161 Mike

Lake Audio
617-244-6881 Rocky

Martin Audio/Video Corp., 423 W. 55 Street, NY NY 10019
212-541-5900 Don Cooper

ADC, St. Paul MN
612-893-3008 Jana Sturmer

Audio-Technica, 1221 Commerce Drive, Stow OH 44224
216-686-2600 Bob Herrold

M.I.L. Electronics (Technics parts), 27 Jones Rd., Waltham MA 02154
891-6730

LPB, 28 Bacton Hill Rd., Frazer PA 19355
215-644-1123 Charles Sheridan or Mary Kiger

Symmetrix, 109 Bell St., Seattle WA 98121
206-282-2555 Dane Butcher

Dr. J. Paradiso
C.S. Draper Lab.
Mail Station 4C
555 Technology Sq.
Cambridge, MA.
02139

Charles Woodard
Factory Mutual Engineering
Box 9102
Norwood, MA. 02062

Dear Mr. Woodard,

Enclosed is the estimate which we have compiled detailing the cost of replacing the equipment at the WMFO studios. The summary is broken down into several different areas (each studio is listed separately, and individual lists are given for the master control room, misc. other equipment around the station, and the Ballou transmitter site). These are rough estimates of what it may cost to replace the existing hardware with new equipment (We've compared notes with other college stations in the Boston area that have recently upgraded their studios with modern equipment). Much of the hardware at WMFO has been completely designed and built in-house or has been custom modified (making it difficult to assign an exact figure), but we have done our best to arrive at these estimates. Feel free to call me if you find any ambiguities.

Best Regards,

Joseph A. Paradiso

Factory Mutual System

Factory Mutual Engineering

1151 Boston-Providence Turnpike
P.O. Box 9102
Norwood, Massachusetts 02062
Telephone (617) 762-4300
Telex 92-4415

February 13, 1987

Dr. J. Paradiso
C.S. Draper Laboratory
Mail Station 4C
555 Technology Sq.
Cambridge, MA 02139

Dear Dr. Paradiso:

Thank you for the inventory and replacement cost estimates for WMFO, Tufts.

It is immensely helpful to me in preparing the overall valuation of university property.

Very truly yours,

Charles F. Woodard, Jr.

Charles F. Woodard, Jr.

CFW:wm

FINANCIAL ESTIMATE FOR FACILITIES OF WMFO, TUFTS UNIVERSITY

Compiled on 8-Jan.-87 by: J. Paradiso, L. Shein (Tech. Crew Directors)
D. Ottenheimer (General Manager, 1985/86)

I) Studio A (Air Studio)

1 Mixer board	$14,000.
3 Turntables (with tonearms,cartridges)	900.
3 Turntable preamplifiers	500.
3 Cart machines (or one triple machine)	5,000.
1 Otari reel-to-reel deck	2,000.
2 Cassete decks (prof. quality)	800.
2 Patchbays (prewired)	1,400.
1 Triple line amplifier	700.
1 Interface/Distribution amplifier unit	1,000.
1 "Gas Light" unit	500.
1 Transmitter Remote control (studio half)	2,000.
1 Monitor amplifier and speakers	800.
Misc. custom hardware, cabling, racks, etc	1,500.
TOTAL	31,100.

II) Studio B (Production Studio)

1 Mixer Board	$10,000.
2 Otari reel-to-reel decks	4,000.
2 Catridge machines (one with recording capability)	4,500.
2 Turntables (with tonearms, cartridges)	600.
2 Turntable preamplifiers	300.
1 Cassete deck (prof. quality)	500.
2 Patchbays (prewired)	1,400.
1 Stereo Equalizer	1,000.
1 Stereo digital reverberator (Space Station)	3,000.
1 Stereo DBX noise reduction system	1,500.
1 Pitch Shifter	1,000.
1 Interface/Distribution amplifier unit	1,000.
1 Monitor amplifier and speakers	800.
Misc. custom hardware, cabling, racks, etc	1,000.
TOTAL	30,600.

III) <u>Studio C (News & Training Studio)</u>

1 Mixer Board	$ 7,500.
2 Turntables (with tonearms, cartridges)	600.
2 Turntable Preamplifiers	300.
1 Otari reel-to-reel deck	2,000.
1 Cassette deck (Med. quality)	300.
1 Patchbay (prewired)	700.
2 Cartridge machines (one with recording capability)	4,500.
1 Triple line amplifier unit	700.
1 Stereo Equalizer	1,000.
1 Interface/Distribution Amplifier unit	1,000.
1 Monitor Amp. and Speakers	600.
Misc. custom hardware, cabling, racks, etc	750.
TOTAL	19,950.

IV) <u>Studio D (Live Recording and Audition Studio)</u>

1 Turntable (with tonearm, cartridge)	$ 300.
1 Integrated Amplifier	$ 300.
1 Dubbing Cassette Deck	$ 300.
2 Monitor Speakers	$ 500.
1 Line amplifier unit	$ 200.
1 On-Air driver/PA talkback device	$ 500.
TOTAL	2,100.

V) <u>Master Control Room</u>

1 Belar Frequency Monitor System (includes RF amplifier, FM monitor, Stereo monitor)	$ 6,500.
1 EBS system (incl. receiver, detector, tone gen.)	1,500.
1 Haffler stereo PA amplifier	300.
1 Intercom/PA mixer unit	2,500.
1 Telephone ringer/doorbell unit	1,000.
1 Spare FM tuner	200.
1 On-Air light driver	400.
Remote telephone ringer, line amplifiers, etc	600.
1 Utility Amplifier	400.
1 House Tone Oscillator	400.
1 Equalizer (w. line amplifier)	1,000.
3 Patchbays (prewired)	2,100.
1 Automatic Studio Switcher unit	3,000.
1 Stereo Compellor (audio compressor)	1,000.
3 Symmetrix telephone coupler units (specially modified)	3,000.
1 Quad stereo distribution amplifier unit	1,500.
Misc. (cable, racks, custom electronics)	2,000.
TOTAL	27,400.

VI) Transmitter Facility at Ballou Hall

1 150 Watt FM Transmitter................................	$ 8,000.
1 10 Watt FM Exciter....................................	3,000.
1 Optimod 8000 (custom modified).........................	4,000.
1 Remote Control unit (transmitter half).................	2,000.
1 Bird Wattmeter (with slugs)............................	400.
1 Audio and Power monitor................................	1,000.
Misc. (cables, rack, fans, etc.).........................	500.
Antenna, FM elements, labor (for antenna only), etc....	17,000.
TOTAL..	35,900.

VII) Misc. Equipment

Music-on-hold driver and custom telephone equipment......$	500.
Stereo Headphones..	600.
Microphones..	3,000.
Biamp 12-channel stereo mixer............................	1,300.
2 Marantz portable cassette recorders....................	600.
Tape cartridge winder....................................	600.
Tools, etc...	500.
Electronic test equipment (oscilloscope, oscillator, etc)	1,000.
2 spare REVOX reel-to-reel decks.........................	2,000.
Misc. equipment for recording (mic. stands, cable, etc)..	1,000.
Misc. around station (PA speakers, track lighting, etc)..	1,000.
IBM-Compatible computer system...........................	3,500.
TOTAL..	15,600.

VIII) Labor Estimates

Curtis Hall (Studio and Master Control Installation)....	$20,000.
Ballou Hall (Transmitter, etc. Installation)............	1,000.
TOTAL..	21,000.

IX) Summary

Net Hardware at Ballou Hall site.........................	$ 35,000.
Labor required at Ballou Hall site.......................	1,000.
Net Hardware at Curtis Hall studios......................	126,750.
Labor required at Curtis Hall studios....................	20,000.
TOTAL COST OF WMFO REPLACEMENT...........................	$183,650.

NOTE: The above estimate does not account for replacement of the record library, furniture (not related to equipment), etc.

Dan Ottenheimer: 391-3215

WMFO Technical Staff (10/17/84)

Name	Home Tel.	Work Tel.	Tufts	Solder	Prev.	Expirience
Andrew Meyer	628-7251		X	X	X	Min. Carp.
Bill Rosenblatt	277-8490	364-2000 x7308		X		Project Coord. Stud.
Bob McTeague	625-8134	391-2578		X		Project Coord. Circ.
Carol Paronis	396-3624		X	X		Operations Director
Charlie McKittrick	623-2981		X		X	Min. Carp.
Chris Donovan	396-3867	381-3435	X	X	X	Min. Wiring
Chris Guttmacher	734-9647			X		None
Dan Ottenheimer	391-3215		X	X	X	Min. Drill.
David Hirshberg	625-2064		X	X	X	Mod. Studio
Diane Andronica	522-4209			X		Mod.
Emily Phillips	623-1382		X			None
Evan O'Hara	288-6554	482-2700 x2795				None
Frank Salmon	666-5183		X	X	X	Min. Wiring
Gregg Butensky	395-3051	497-3154		X		Project Coord. Circ.
Heidi Smith	254-0863			X		None
Jane Jackson	536-9344			X	X	Mod. Studio
Joe Paradiso	332-1460	258-1475		X		Effort Coordinator
John Plummer	625-4572		X			None
Jon Mysel	776-8653		X	X	X	Min. Drill.
Judy Schwartz	492-2031	241-8600 x256		X		Project Coord. Equip
Karen Donoghue	665-8081		X		X	Min. Studio
Ken Seltzer	396-7479			X		Project Coord. Equip
Leigh Shein	576-2190	497-5054		X		Effort Coordinator
Mike Drumke	776-1941		X	X		Project Coord. Stud.
Nora Nolan	488-6382		X			None
Sarah Hood	395-4728		X	X	X	Mod. Studio
Steve Cantor	547-7962	495-4509				None
Susan Fero	566-6950					None
Sylvia Giannitrapani	782-4268	492-1000 x420		X	X	Min. Studio
Tom Galluzzo	666-5073			X		Min.

David Allen — Non god will not go on or the wrong way to be right

Thursday meeting — Accidents
Work — Tie This

WMFO FM 91.5

PARTY

* *To celebrate the complete renovation of studios*
* Location: WMFO Radio, 3'rd floor of Curtis hall
  Tufts University, Medford
  Tel. 381–3800
* Time: Saturday, Dec. 7
  9:30 pm through ∞
* Basic party essentials & on–air music provided

---Note: The WMFO studios are located in Curtis Hall on the Tufts University Campus, which is much easier to locate than one might initially think after looking at the maps on the flip side of this invitation

WMFO 91.5 FM

P.O. BOX 65, MEDFORD, MA 02153 ■ BUSINESS (617) 625-0800 ■ REQUESTS 381-3800

Original
"Unreleased"
Version

For immediate release
October 9, 1985
For further information:
Leigh Shein 254-7511
Joe Paradiso 332-1460
Dan Ottenheimer 776-9874

WMFO OPENS REVAMPED STUDIOS

The Tufts University radio station, WMFO, recently completed a year long project to rewire the station and rebuild its studios.

"With our extensive record collection and the new studios, we probably have one of the best stations in all of Boston, including the professional ones," General Manager Dan Ottenheimer (E86) said. Dan continued by stating," aside from already containing many sophisticated pieces of equipment, the studios were designed so that any new device could be added to the system."

The project was started over a year ago by alumni Joe Paradiso (E77) and community member Leigh Shein. Joe worked at WMFO as an undergraduate at Tufts and came back to work with the radio station after obtaining a job in Cambridge. " I really enjoy the creativity that MFO has, and knowing the bad shape the studios were in I realized that there must be an all staff project that would ultimately allow jocks to have even more freedom," Joe said.

The main broadcast studio has been taken apart and then physically redesigned and electrically rewired. The production studio now has many pieces of equipment for use in doing creative work. Each of the three studios now have a DA/Studio Switcher unit in them which enables the rapid switching and combining of on air studios.

MORE MORE MORE

The Interface unit in each one enables the quick addition of any additional equipment.

Relying on his Chief Engineer experience from his college radio station, Leigh Shein designed most of the rewiring work. " I really care about college radio," Leigh said, " and when I moved to Boston and heard WMFO, I knew that this was where I'd fit in."

In 1978 a fire struck the WMFO studios and they were rebuilt quickly in order to get the station broadcasting again. This apparently was not the best procedure in the long run Dan tells us," Leigh spent hundreds of hours fixing all the shoddy work done previously, while Joe designed, tested, and built new electronic devices. The two of them, with help from dozens of other staff members have brought WMFO out of the Stone Age and into the forefront of the Technological Age."

Leigh Shein comes to WMFO from NorthWestern University as well as other college radio stations. His show is on Saturday mornings from 6-10 AM. Joe Paradiso has a PhD. and works at Draper Labs in Cambridge. He does a show on Tuesday evenings from 8-11 PM.

-30-30-30-30-30-30-30-

Later "unreleased" Version

FOR IMMEDIATE RELEASE

Reconstruction at WMFO-Medford
Leaves Tufts' Radio Station Best

WMFO, a community broadcast service of Tufts University operating at 91.5 mHz. FM, has just undergone an intensive rebuilding and renovation, making it "one of the best engineered and best equipped stations in the Boston area," acording to General Manager Dan Ottenheimer, E'86.

The reconstruction process, which has been underway for nearly one and a half years, had become increasingly necessary since the 1977 fire at Curtis Hall, which houses the station. "At that point, speed, not sophistication, was important to get us back on the air," said Leigh Shein, a community volunteer, who, along with Draper Lab physicist Dr. Joe Paradiso, E'77, masterminded the rebuilding process. With the opening of the new studios this fall, "we compare very favorably to professional stations and rank highest amoung college stations in technical quality," Shein added.

Three complete stereo studios have been installed at WMFO; these studios have been built to stringent professional audio standards, and can be put on-air in any combination via a flexible switching scheme. The reconstruction has brought to the station's operation a sophisticated yet easily handled system for on-air and production creativity, with specialized recording options, convenient features added to existing equipment, and patching access that allows audio from any source to be routed to any studio and processed through any available equipment for better creative control.

Joe Paradiso and Leigh Shein achieved these technical developments buying little new equipment and spending no money on labor. All the skilled work and effort required to renovate the studios was donated by a volunteer team of students and community members. Wherever possible, existing equipment was repaired, modified, and upgraded for use in the new studios. Most special professional audio equipment needed for the reconstruction project was designed by Joe Paradiso (who also donated most of the required electronics and components to the station), and assembled by the all-volunteer technical staff. Costs were thus cut to a fraction of the expenditure necessary to procure equivalent commercial audio devices. The complex task of organizing the thousands of cables running between the studios and patchfields was coordinated by Leigh Shein; he was aided by a crew of student and community volunteers who have by now become quite proficent at the art of soldering. "This amount of dedication shows the importance of WMFO to its staff and listening audience," Ottenheimer said. Under $_____ was invested to complete the entire project; if professional labor was contracted and all equipment was purchased, costs could easily have run an order of magnitude higher.

Of course, its hardly possible to affix a price tag upon the particular attention devoted to the project by Paradiso and Shein; their creative approach has made WMFO certainly one of the most unique audio installations anywhere. The studios certainly have achieved a special personality; original devices have been incorporated such as a pitch shifter that can turn even the most timid freshman voice into an ominous growl, an intercom system which subjects the studios to a the opening notes from a random popular tune after a "zero" is pressed on the telephones, and a large indicator light which illuminates to display the word "GAS" whenever the on-air DJ has set his audio levels too high. "After all, we had to have some fun...," apologizes Paradiso.

WMFO provides an alternative programming format, featuring on weekdays "Radio Free Jazz" 9 am to 2 pm, and "Contacto", a program produced and aired by the Portuguese community from 6 to 8 pm. The majority of the programming is "Freeform", defined as "a coherent mixture of music and sounds not limited by the barriers of popularity of labels," according to Program Director David Hirschberg, A'88. WMFO is fortunate in owning one of the largest and most diverse record collections of any radio ststion, and works diligently in maintaining and expanding its library.

Dear "Tech Crew"

This noon to 11 p.m sign out of B is a bit much. No one should use this for more than 4 or 5 hours at a time. But 11 hours? That's getting a bit indolgent... ne c'est pas? Look, this is all well and fine but howzabout asking yourselves some concrete empirical questions instead of jumping into some pondful of assumptions:

1. WILL LISTENERS BE ABLE TO DETECT a NOTICABLE DIFFERENCE?

2. WILL THE STAFF EVER ACTUALLY DO ANYTHING IMAGINATIVE WITH ALL THIS STUFF?

Or a correlated question: Is this supposed to be some kind of monument? I used to build a monument to my aspirations out of service and programming but these were at least a function of the air sound. I mean; the station tends to sound lame most of the time. I know J.P. has his sound but what are you doing to pass that on. And Leigh will be fine if he ever gets over his aversion to knowing anything or remembering it. Ask yourselves this anyway. I know you will feel threatened by this sort of enquiry and I know you'll think its mighty ungracious of me to not appreciate your obviously considerable efforts. But! Form and Function are unified. You can't have one without the other, and in radio function matters more as an alternative then form. Basically this station sucks to listen to more often than it doesn't and how doyou feel you enhance or detract from this. Don't worry, I won't hold my breath waiting for an answer.

CR

Dear Tech Crew,

I'm sorry. I was all wrong. You guys are solid after all. I see what you are doing. Great Job. Now when listeners from Arlington call to complain about not being able to pick up the station from five miles distance, we just put em on "HOLD" and they can hear it through the phone. Pretty good.

Really though. I feel like my slant six Dodge has been replaced by a BIG GAUDY Caddy barge with a mess-o-gadgets n buttons and driven by a buncha scared old folks. Bad milage and breaks down all the time.

Oh well, it was good while it lasted. And at least you don't break things as predictably as Noah.

R

pioneer

PIONITE®
Decorative Laminates

Customer's Name: PARADISO
Job Description: WMFO, ETC.
Date: 4/14/87

4

DEAR JOE -

I'M SITTING HERE AT ABOUT 8 AM STARING AT THIS MFO STUFF. WE ARE LEAVING FOR LUXEMBOURG (ACTUALLY FOR THE AIRPORT) IN A FEW HOURS. I CAN'T REALLY REMEMBER WHAT I'M SUPPOSED TO DO. MORAL: "NEVER PUT OFF FOR TOMORROW WHAT YOU CAN'T DO TODAY!"

SO I'VE COMPILED & REDRAWN THE INTER-RACK AND EXTERNAL TREES BUT I'LL HAVE TO LET YOU DO THE BARRIER STRIPS. WHAT WAS I GONNA DO ABOUT THE STUDIO TREES?

ANYHOW I'LL MAKE IT UP TO YOU BY SENDING AN EXTRA HAWKWIND T-SHIRT FROM EUROPE.

BRIEFLY THE SCOOP IS: WE LEAVE TODAY FOR THE RIVIERA VIA LUXEMBOURG. AS SOON AS WE HAVE AN ADRESS WE'LL SEND IT OUT.

THANKS FOR THE BUTTONS & PRINTOUTS, TRES COOL!

GOTTA PACK NOW, BUT WILL WRITE SOON.

LOVE
[signature]

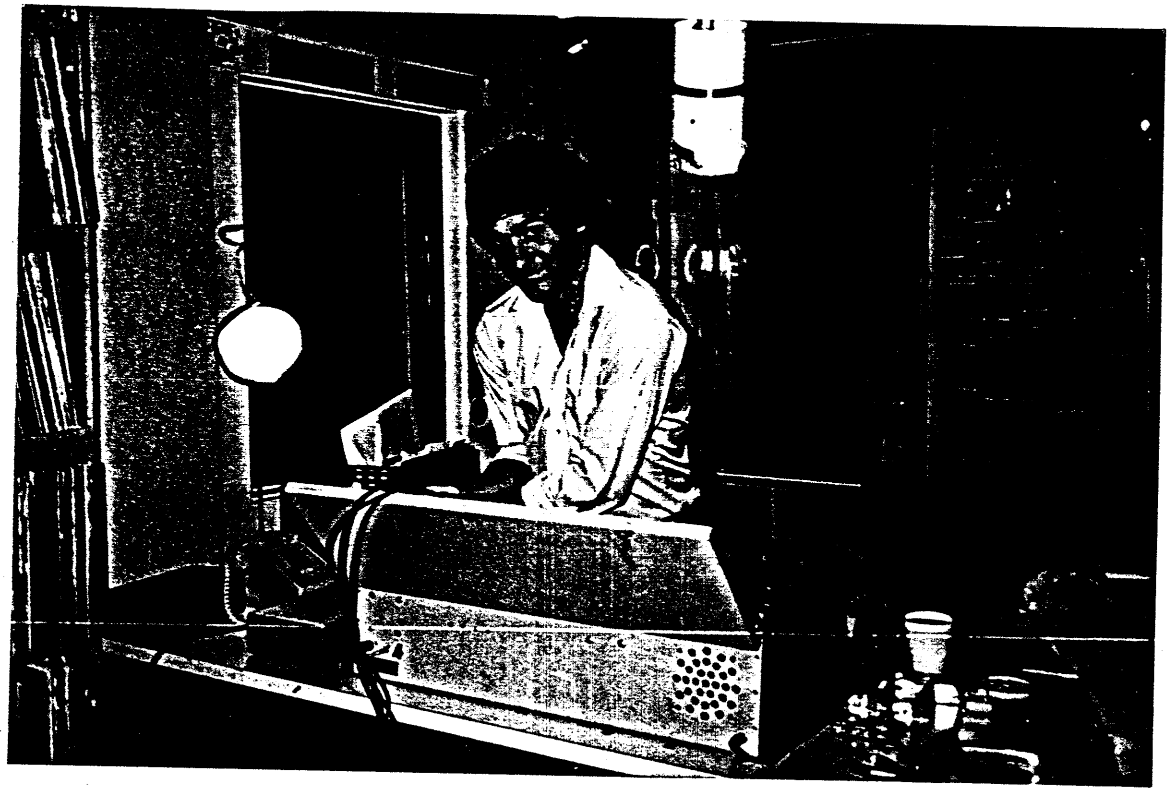
We miss you!
Betsoid
(3" not 2"!)

PIONEER PLASTICS
DIVISION OF LOF PLASTICS INC.
A LIBBEY-OWENS-FORD COMPANY

DRAWN BY: ________

SHEET ____ OF ____

Yours in Posterity,



Joe Paradiso

Leigh Shein