The HAPIRAT, a Ham Audio Panel Interface for Receiving And Transmitting

K5AM built an audio panel to manage the "rat's nest" of cables behind his transceiver and computer, and now says that he is a "happy rat."

by Mark Mandelkern, K5AM
5259 Singer Road
Las Cruces NM 88007 USA
www.zianet.com/k5am
k5am@arrl.net

Note: This is the expanded version of the article. The short version appears in QST (September 2004, pp 28-33).

CONTENTS

1. Introduction
2. Operating features
3. Design features
4. Non-features
5. Modes
6. Front panel controls
7. Setup
8. Meter
9. Indicators
10. Circuit details
11. Automatic microphone selection
12. Microphone preamps
13. Signal switching
14. Outputs
15. Computer audio interface
16. Transceiver audio interface
17. Computer control interface
18. Transceiver control interface
19. Sound file monitoring
20. Speaker and headphones
21. Test signals
22. Logic
23. Power Supply
24. Construction
25. Parts
26. Alignment
27. Results
28. Summary
29. Biographical sketch
30. References

Appendix I. Parts dealers
Appendix II. Glossary

1. Introduction. A modern ham station often involves quite a collection of cables, especially if a computer is used for digital work or contesting. These cables frequently need changing for different modes, or for operation without the computer. One often hears the accumulation of cables behind the rig referred to as a "rat's nest." This project provides all the necessary connections in all the various operating configurations without plugging or unplugging any cables. Also, it provides automatic switching for three different microphones, and the speaker or headphones. Before this panel was built, changing cables in the "rat's nest" was an unpleasant chore – with the *Hapirat* I am now truly a "happy rat."

Besides the zillions of connecting cables behind the rig, before the *Hapirat* there was a multitude of little mini-boxes and switches on the operating bench. Microphone switches, filters, preamps, monitor amps, headphone switches, meters, recording controls – all built during the last 55 years of hamming in an attempt to cover all bases and keep up with changing configurations and new modes. Still, I often had to switch cables for special modes, and reset various levels on the mini*boxes or in the computer. Something had to be done to consolidate all this clutter – the *Hapirat* was the answer!

2. Operating features. Arguably, dealing with audio signals has become a completely separate hamshack function. We cannot expect a transceiver to include all the necessary audio and control functions, especially when there are several rigs in the shack, several microphones to choose from, and several different operating modes.

Although the features listed below are all useful in a comprehensive audio panel, they are not all absolutely necessary; a simple audio panel can be built using only a few desired features. To begin operating on RTTY or PSK is actually much easier than it may appear; you only need a computer program, a small circuit, and a few cables. Helpful information is readily available:
ARRL'S HF Digital Handbook, Steve Ford, WB8IMY, ARRL, Newington, CT 06111, http://www.arrl.org/catalog
www.rttycontesting.com
www.k9jy.com
www.qsl.net/mmhamsoft
www.writelog.com
www.psk31.com

Here are the *Hapirat* operating features:

- Automatic microphone selection. The boom mike is the default, but grab the desk mike or hand mike and push the PTT button, and that mike is automatically connected to the rig. A panel switch allows optional manual selection. This automatic circuit is an update of a design previously published by the author in *QST* (reference 30.4).
- Separate gain setting for each microphone on the front panel.
- Adjustable microphone load resistance. Adjust the panel knob for the best audio frequency response with a particular mike and voice characteristics. Separate load adjustments for each of the three mikes.
- Provision for electret-condenser microphones. Pull out any one of the mike load knobs to switch in the bias voltage for that mike.
- Microphones are disabled during computer WAV audio transmissions. (Time for munching.)
- Automatic headphone switching. Merely lift the headphones or boom mike set off the hook. A panel switch allows optional manual selection.
- Panel headphone level control to balance the speaker/phones levels. This avoids the need to reset the receiver AF Gain control.
- Panel level controls for digital receiving, transmitting, recording, and voice DX spots.
- Monitoring of transmitted computer-stored SSB messages, when desired.
- Two-tone generator for amplifier linearity optimization, to minimize splatter.
- Pulse CW keyer for amplifier adjustment at 33% duty cycle, to reduce dissipation and prevent premature amplifier tube failure. This circuit is an update of a design previously published by the author in *QEX* (reference 30.2).
- Separate knobs for all functions. A total of 18 panel controls are provided, virtually all of the set-and-forget type. No need to change transmitter settings when changing microphones or modes. No need to readjust controls for recording. Fixed computer soundboard settings; no need to adjust at any time.
- Universal character; not specialized for a particular mode, operating program, computer port, or radio. Useful in any station.
- Normal radio operation when the computer is off; no need to change cables.
- Simultaneous distribution of transmit audio to six or more transceivers. No need to switch cables for different rigs. Four high-level outputs for special connecting modules; two regular outputs for easy connection without modules. More outputs can be added easily.

- Metering of all audio levels, for receiving, transmitting, monitoring, and recording. Provision for complete setup and adjustment before transmitting.
- All operating controls are on the front panel. No internal or rear panel settings, adjustments, jumpers, or switches.
- No cable-changing is required for any mode or configuration.
- Connects to all makes and models of radios; no special settings.
- Unique computer modes, with LED indicator. In other modes the computer PTT and KEY functions are disabled; this avoids undesired transmissions.
- Separate AF amplifier channel for DX packet spots with a separate speaker, if desired. This provides for packet cluster announcements at a remote location.
- More features? There is ample space on the board for adding extra features. The mode switch has two unused positions. With TTL circuits, it is easy to add modes and features, and to modify options. There is one spare control on the panel.

3. Design features. Every effort was made to choose circuits that would provide high-performance and absolute reliability.

- High-level computer and transceiver audio interfacing with connecting modules, attenuators, and isolation transformers, to avoid hum problems.
- High pass filter in the transmit audio channel. This is to improve speech clarity, restrain bandwidth, and eliminate any externally-induced hum. The filter has a design cut-off frequency of 274 Hz; it has 30 dB attenuation at 60 Hz.
- No active discrete components. Integrated circuits are used exclusively.
- No relays; all solid-state switching.
- Microphone audio does not go *through* the computer, but automatically goes to the computer in **REC** mode for recording. The rig works normally when the computer is off.
- Full-logic control of all modes. No microprocessor with pre-programmed memories. As the need arises, features can be added easily, or options modified.
- CMOS audio signal switching. No audio signals on the mode switch.
- Sufficient internal space for more features which may be added. Simple construction; easily modified as needs change. All components directly accessible: No moving parts!

4. Non-features.

- The panel does not make the selection between transceivers. That is done on my operating bench with a separate switch, which also selects VHF transverters. There is a trade-off between comprehensive application and flexibility; I have found that it is best to deal with these problems separately. The transceiver switch needs to switch only the speaker, key, PTT, and receiver line audio cables. The *Hapirat* distributes transmit audio to all transceivers simultaneously; the bench switch determines which transceiver will react to the PTT command. If VOX is used, it must be switched off for each transceiver not in use; this is always a good operating procedure.
- An SO2R provision is not included. Some contesting programs are able to deal with this matter directly. Hopefully, some of the ideas in this article will be useful in the design of an audio panel that includes SO2R.
- Simplicity? No, sorry about that; the *Hapirat* uses 107 integrated circuits. Modern life is not simple; especially not with modern ham gear, computers, and digital modes. However, the *Hapirat* is simple in operation, with an absolute minimum of panel switches. A single **MODE** switch controls all circuits. The other two switches on the front panel merely override the automatic mike and headphone circuits, and are not normally used.

5. Modes. A single 12–position rotary mode switch is employed in order to avoid a confusing situation with up to a dozen individual toggle switches. Selecting a mode tells the panel exactly what you want to do; then all the signal switches are set accordingly by the logic circuits. Ten modes are used at present; two more modes may be introduced by adding a few ICs to the logic section.

Here are the various modes of operation:

STBY. Standby; the radio functions manually. This mode engages computer standby, but not radio standby. No packet spots are heard in the spots speaker. WAV audio, digital tones, PTT and KEY signals are *not* sent from the computer to the transmitter, no matter what the computer tries to do; this mode in included to prevent undesired transmissions. Receiver line audio is sent to the computer for digital decoding, as in all other modes except **REC** and **PLAY**.

SPOT. Same as **STBY**, except that DX packet cluster spots from a logging program will be heard in the spots speaker. The spots speaker will be off in all other modes, to prevent interferring sounds while operating; this option may be varied. The packet speaker may be placed elsewhere in the home; there are reports that such a system has sometimes been connected to a 200 watt PA system in a suburban backyard, to announce rare DX spots while the OM mows the lawn. If a separate packet speaker is not connected, mode **PLAY** can be used to hear packet spots in the rig speaker. These modes can also be used for computer CD music; in mode **PLAY**, packet spots will be heard mixed with the music.

CP. Computer control enabled. The rig functions manually when desired. Also, the computer will send WAV audio, digital tones, PTT and KEY signals to the radio. The headphones or speaker will not hear WAV audio or digital tones for monitoring. This mode is always used for CW – in order to hear the sidetone generated by the radio. This mode may also be used in SSB mode with a radio that has a built-in monitor. For digital transmissions, mode **CP** is used to avoid hearing the transmitted tones; to hear the transmitted tones, mode **CPM** is used.

CPM. Computer control with Monitoring. Same as mode **CP**, plus transmit audio monitoring. When WAV audio SSB transmissions are sent by the computer, the headphone or speaker audio from the receiver is switched off and the computer audio (amplified in the *Hapirat*) is connected for monitoring. This is normally used for SSB; to hear a built-in radio monitor, mode **CP** is used.

2TONE. Two-tone test. Two tones, at 1000 and 1500 Hz, are sent to the transmitter for linearity testing. Panel controls set the level and adjust the balance. The tones are not heard in the speakers. PTT is controlled manually at the transmitter or with a foot switch. A heterodyne mixer demodulates the difference frequency of 500 Hz and provides this as a sync signal to the monitor scope for triggering.

1TONE. One-tone test. Only the 1000 Hz tone is sent to the transmitter.

PULSE. This mode enables the 33% pulse generator on the KEY line for amplifier tuning at reduced dissipation. The pulses correspond to CW dits at about 50 wpm, but with double spacing between the dits. This mode does not key the transceiver PTT line; manual control at the radio is used for tune-up.

REC. Record. Microphone audio is sent to the computer for recording WAV files. In this mode, the panel **REC** control sets the *Hapirat* output level fed to the computer; the level is indicated on the meter. This output is amplified, and uses the computer LINE input, rather than the MIC input. This often produces a better signal-to-noise ratio in the WAV file than is obtainable when a microphone drives a computer directly. The individual microphone levels are not changed when in mode **REC**; they are adjusted in mode **SETUP**. In all other modes, receiver line audio is sent to the computer for decoding RTTY or other digital modes. While recording, receiver audio to the speaker is automatically muted.

PLAY. The computer sends WAV audio to the headphones or speaker for previewing. The PTT and KEY signals from the computer will not be sent to the radio. This mode is also convenient for testing WAV files without starting a contest program. Alternatively, WAV previewing is possible in mode **SPOT**; the WAV audio will be heard in the packet speaker, rather than in the rig speaker. (CW previewing requires the sidetone oscillator in the radio; mode **CP** is used. Set the transmitter power output to zero, select a dummy load with the station antenna switch, and choose a frequency unlikely to cause interference.) This mode can also be used to play computer CD music.

SETUP. This mode is used to set the individual microphone levels; the meter reads the internal signal levels. This mode allows mike adjustment without transmitting. Whistling into the mike (a poor procedure for on-the-air testing) is allowed in this mode, but may not indicate the peak; this depends on the individual voice, whistling ability, and the mike. After setup, the mike settings are not changed; the setup procedure equalizes any differences among the mikes. The transceiver mike gain is then set while speaking into any one of the mikes. Then the **TX** control is adjusted so that the computer WAV level read on the meter matches the mike level. This three-step procedure must be followed in the proper order. Now, if a different transceiver is selected, no settings on the *Hapirat* need be changed. The computer sound controls are never changed after an initial one-time setting.

6. Front panel controls. The circuits are designed so that the nominal setting of most panel controls is 30% of full rotation, about 10 o'clock.

SPOT. This control sets the level of DX packet cluster spots heard in the separate spot speaker. This control is given a prominent position on the panel and the largest knob because in my shack it is used the most. The other controls, important as they are, are mainly of the "set-and-forget" sort. The **SPOT** control gets changed every time I move from the operating bench (low) over to the workbench (high), or into the kitchen (very high).

RX. Receive. Receiver audio level delivered to the computer, for digital decoding or recording.

TX. Transmit. Computer audio level delivered to the transmitter, during transmission of WAV audio or digital signals. This control does not affect the transmit audio level when the microphones are used; thus is serves to match the computer level to the preset mike level. This control is given a special, red, distinctive knob, which helps to locate it. In spite of all the preset level adjustments, a touch-up may sometimes be necessary when changing digital or voice modes, or computer programs. The audio transmit level is critically important for transmitting a clean digital signal; this may be the only knob on the *Hapirat* panel used during normal operation.

Both the **RX** and **TX** controls refer to receiving and transmitting with the computer, but not otherwise. This reflects the fact that the panel is mainly a computer interface, but allows normal operation without a computer, without changing cables. Receiver AF gain, and transmitter microphone levels are set normally on the transceiver.

MON. Monitor. Audio level from the computer sent to the speaker or headphones, for monitoring WAV audio transmissions from the computer; only in mode **CPM**.

2T. One-tone or two-tone test signal level.

BAL. Balance for the 2-tone test signal. The transmitter might not have a perfectly flat response for audio frequencies; this control compensates.

REC. Record. Sets the audio level from the microphones sent to the computer, for pre-recording WAV files before a contest.

SP/PH. Speaker/Phones. Up: Speaker, unless the headphones are lifted off the hook. Down: Phones. Center: Both speaker and headphones off; for answering the telephone.

PH. Phones. Headphone level; balances the levels between speaker and phones, so the receiver AF gain control need not be reset.

7. Setup. This procedure sets the proper gain levels for each microphone, for the transmitter, for WAV transmissions, for computer recording, and for digital reception. The sequence is important; the first matter demanding attention is individual mike level adjustment. Set each mike **LOAD** control to 9 o'clock (10k ohms); experimentation can be done later. In mode **SETUP**, select each mike in turn using the panel switch and adjust the mike **LEVEL** controls for mid-scale meter readings. Next, in mode **STBY**, and any one of the mikes, adjust the transmitter MIKE gain control for proper SSB operation according to the manual, driving a dummy load.

Now switch to mode **CP** and set the computer soundcard MASTER, WAV, and record LINE settings at a fixed level, about 1/3 or 1/2 scale; these levels will never need to be changed. Play a WAV CQ file using a contesting program, and adjust the **TX** control for mid-scale meter readings; verify the proper transceiver ALC and other meter readings. Note that the transmitter MIKE gain control is not used to adjust WAV transmissions; it was preset for the mikes. The *Hapirat* **TX** control affects only the WAV transmissions.

If *WriteLog* and *MMTTY* are used, send an RTTY message. If the meter reads other than miscalled, adjust the *Digital Output* setting in the program's TX options menu accordingly. For other programs, look for a similar adjustment. This one-time setup procedure assures that no adjustments will be needed when switching between SSB and RTTY.

Finally, tune into an RTTY signal and adjust the panel **RX** control for the proper level to your digital program. For making WAV recordings from the receiver, a different setting may be needed.

There is an easy way to restore your fixed computer soundcard level settings for ham radio, in case they are changed for other uses. The excellent program *Quickmix*, written by M. Saxon, is available at www.msaxon.com/quickmix.

Quickmix will remember your fixed settings for all the soundcard levels. With a shortcut for the settings file in your *Startup* folder, you will be automatically set-up for ham radio every time the computer starts.

Note on setup. I use a Dell laptop, Win98se, *WriteLog*, and *MMTTY*. Terminology and adjustments for other systems may vary.
References:
www.writelog.com
www.qsl.net/mmhamsoft

8. Meter. The *Hapirat* includes a peak-indicating meter to allow careful adjustment of all circuits in all modes. The circuits are designed so that the normal indication in any mode is mid-scale. When receiving, the meter reads the level of the receiver line audio sent to the computer; this can be adjusted by the panel control **RX**. When transmitting, the meter indicates the audio level sent to the transmitter, whether directly from a microphone, from the computer, or from the tone generator. The panel control **TX** adjusts only the level from the computer.

For direct speech from the microphones, the three mike level controls are pre-adjusted in mode **SETUP**. The test tone level is set by panel control **2T**. When recording, the meter reads the level sent to the computer, adjusted by the panel control **REC**. The station PTT switch, foot switch, or transceiver PTT switch will also switch the meter function from receive to transmit; the *Hapirat* senses the external PTT circuits.

The meter reads signals at internal points which maintain a nominal standard internal level of 3000 mVpp. The meter reads 6000 mVpp full-scale; the proper reading is 50% of full-scale in all modes. The microphone level controls operate before the meter and are adjusted in mode **SETUP**. Computer WAV audio may be pre-adjusted in mode

PLAY using the control **TX** and the meter before transmitting. With the mike levels set beforehand, the audio level to the computer for recording WAV files is set in mode **REC** using the control **REC** and the meter. The standard level 3000 mVpp appears at the jack **TO CP**; this high level allows a low setting in the computer for LINE input gain, and avoids hum problems. I use the 3rd tick on a scale of 7 for all three computer soundcard levels (MASTER, WAV, LINE), and never change them; all adjustments are made on the *Hapirat*.

While receiving, control **RX** is used to set the standard level on the meter; thus the computer receives the same level as when recording. (The various digital programs may, however, require different levels than the WAV recording program. Control **RX** can be used to compensate.)

In all modes the relevant level is set by a single control. The mike level controls are *not* used to set the drive to the transmitter, they are used to set the internal standard level of 3000 mVpp in mode **SETUP**; then the transmitter controls are set for proper operation. When playing computer WAV audio in modes **CP**/**CPM**, control **TX** is used to match this level. Control **TX** does not affect the live mike levels. This system equalizes the levels of three different mikes, and maintains the same level for computer WAV audio.

Meter summary. There is no meter switch; the logic circuits determine the metering function, depending on the **MODE** switch setting. Here is a list of the meter functions for each mode. In each case, the proper meter indication is mid-scale. Mode **STBY** refers only to computer control standby; the transceiver functions normally in this mode.

STBY. When receiving, the meter reads the receiver line audio level fed to the computer for digital decoding or recording; adjust with the panel control **RX**. When transmitting, the meter reads the microphone audio level fed to the transceiver; this is preset for each mike.

SPOT. Same as **STBY**.

CP. When receiving, same as **STBY**. When transmitting, the meter reads the WAV audio level fed to transceiver; adjust with panel control **TX**.

CPM. Same as **CPM**.

2TONE. The meter reads the test tone signal level sent to the transceiver. Adjust with panel control **2T**. Monitor the RF envelope pattern on a monitor scope and adjust the tone balance with panel control **BAL** for sharp cross-overs.

1TONE. Same as **2TONE**.

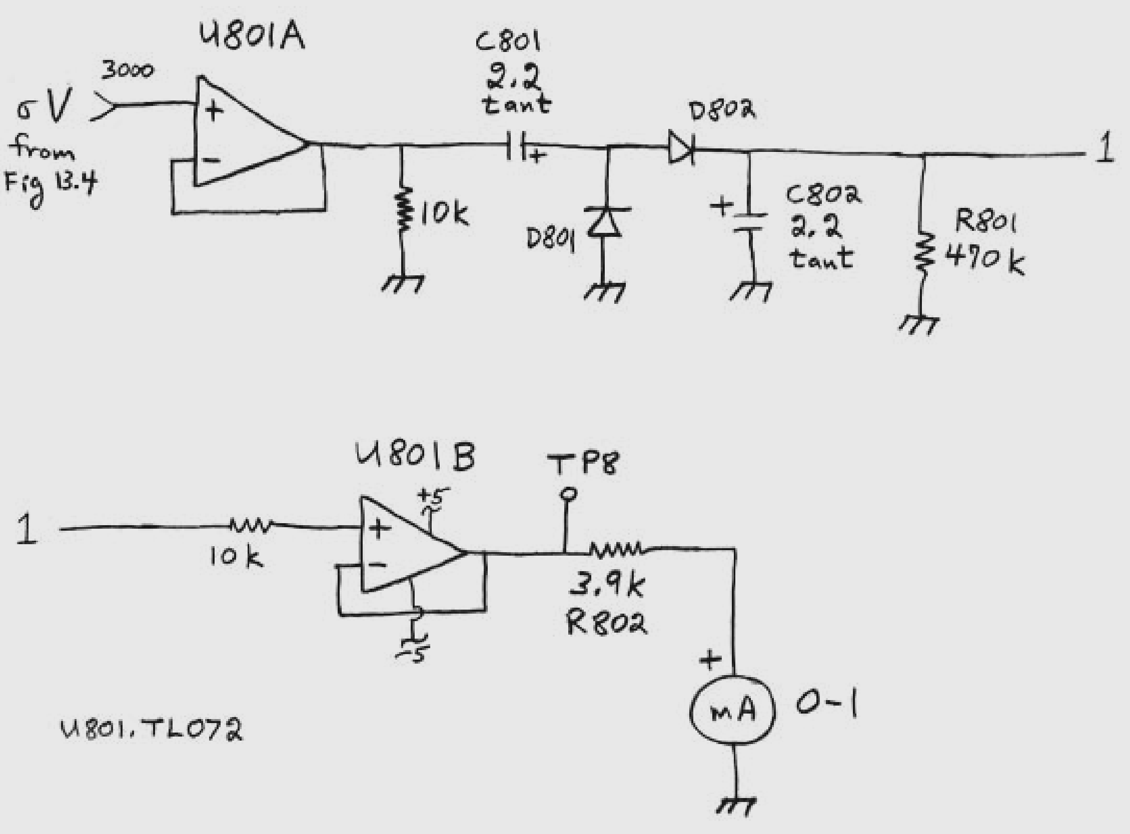
PULSE. Same as **STBY**.

REC. The meter reads the voice level sent to the computer for recording. Adjust with panel control **REC**.

SETUP. The meter reads the internal *Hapirat* microphone levels.

PLAY. The meter reads the WAV audio level from the computer sent to the transceiver. Adjust with panel control **TX**.

Figure 8; Meter circuit.



Meter circuit. Figure 8 shows the peak-indicating meter circuit. Voltage follower U801A is used to drive the voltage-doubler detector D801–D802. The op amp isolates the detector from CMOS switch U1303B in section 13, which determines which audio signal is measured. The detector has a fast attack time; a slow decay results from R801–C802, with a time constant of 1 second.

A 1 mA meter is indicated. Other DC meters, of any range up to 5 mA, may be used; the meter resistor R802 is selected accordingly. The maximum voltage output of U801B at test point TP8 is about 4 volts, so the maximum current applied to the meter is about 110% of full scale, enough to pin the needle but not enough to damage the meter.

9. Indicators. There are three LEDs on the panel.

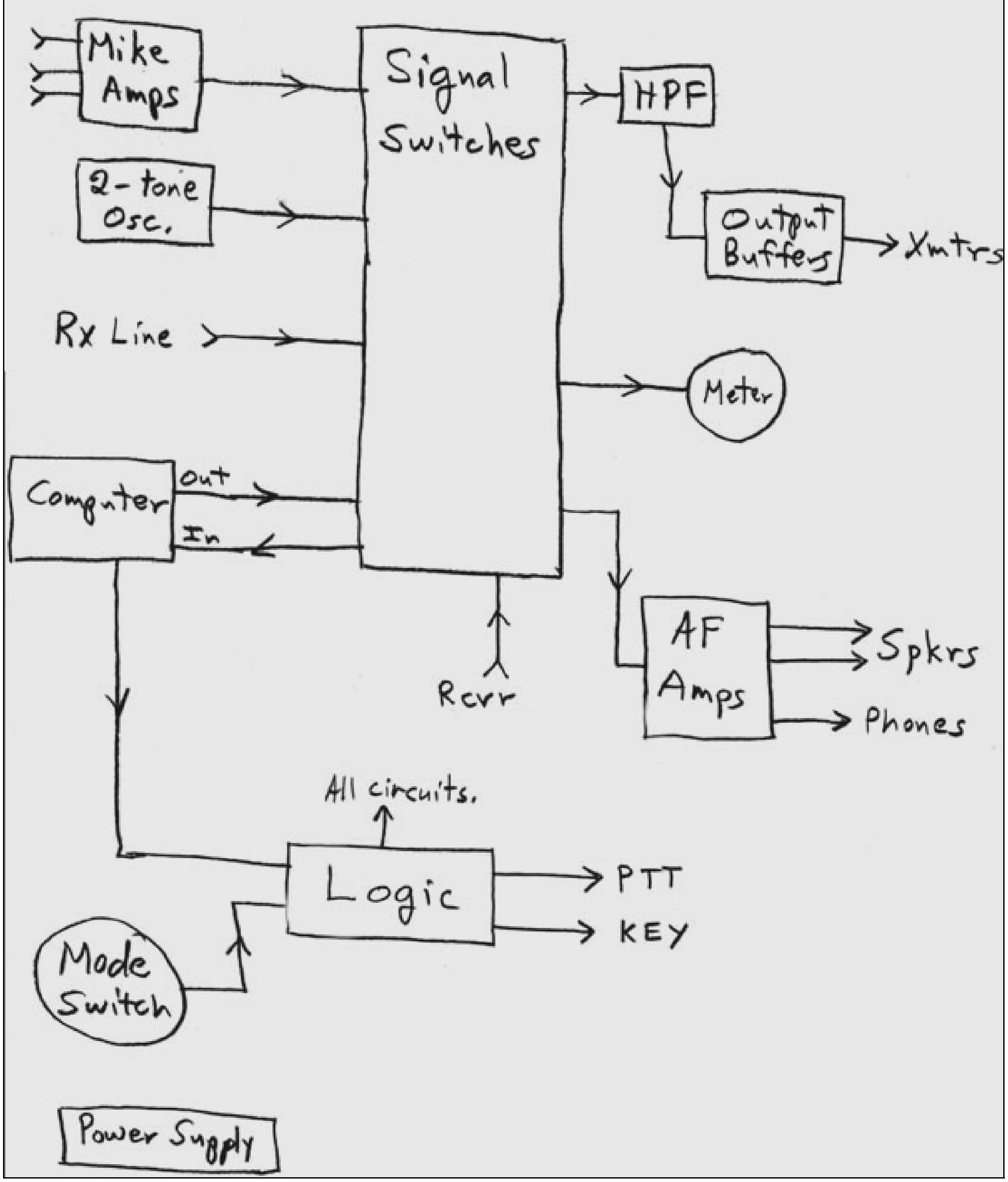
Green. Power is applied to the *Hapirat*. There is no power switch on the panel – it's always ready! Of course, the main AC switch on the operating bench cuts power to the main +13.6 V power supply, and everything else in the shack except the clock.

Yellow. This LED lights whenever the *Hapirat* is in mode **CP** or **CPM**; warning the operator not to press a keyboard F key unless a transmission is definitely desired. If computer-ordered transmissions are not intended, it is best to stay in **STBY** mode.

Red. This LED lights when the computer sends a PTT command, but only in modes **CP** and **CPM**. If the red LED comes on immediately when switching to one of these modes, the operator should quickly switch to **STBY** mode and check the contest program.

10. Circuit details. A block diagram of the entire *Hapirat* circuit is shown in Figure 10.1. The schematic diagrams for the various sections are shown separately. Not all the features are needed, so some of these separate schematics might be used for smaller projects. For example, you may wish to build only the automatic microphone selector; or the two-tone generator for amplifier tuning and linearity checking. The high-level audio signal system (for hum and RFI avoidance) may be adapted for any computer interface box.

Figure 10.1; Block diagram.



Certain special conventions are used in the schematics, mainly to avoid clutter. The most important convention is the omission (in the schematics) of the usual, and important, bypassing.

Unless otherwise stated:

- Repetitive circuits on the same page often do not have all the component values labeled. The convention is "same as above", or "same as to the left".
- Nominal signal levels are marked at convenient places on the diagrams, in mVpp.
- The power rails are bypassed to ground with 10 μF electrolytics at various points adjacent to each subsection.
- Each IC has a small 100 nF monolithic bypass capacitor soldered across the socket between the + and – power terminals.
- Each signal (or control line) rear panel jack has a 1 nF (or 10 nF) disk ceramic bypass capacitor. Each section and sub-section has similar bypassing.
- Capacitors labeled s.m. are silver mica types, with value indicated in pF.
- Unmarked diodes are small signal types, such at 1N4148.
- Electrolytic capacitors are aluminum types, with 16 volt ratings.
- Several non-conventional schematic drawing methods are used to improve clarity. So far as possible, signals flow from left to right. Inputs to a section of a circuit are shown with the angle symbol, outputs with an arrow. A distinction is made between a connection to a voltage source, and the source itself, as in the power supply section. The former is indicated with a wavy line, the latter with a small circle. For example, in Figure 17.3 a reference voltage of +3 volts is developed for use by the comparators above; this source is marked with a small circle, and may be borrowed by other portions of the circuit board. Some of the special schematic symbols used are shown below in Figure 10.2.
- In most cases, commonly used pin-outs are omitted.

Special conventions also apply to the labels on various lines:

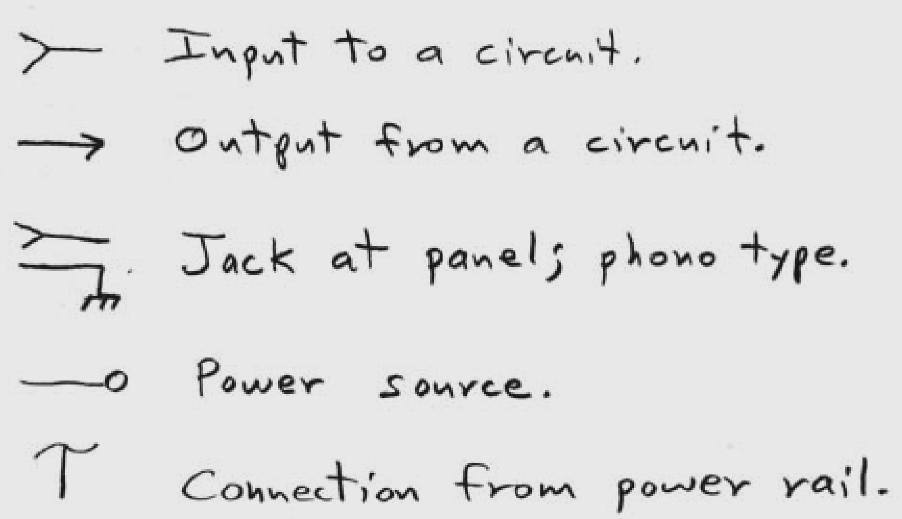
- Signal lines are labeled with the Greek letter σ as a prefix. Example: σTX is the transmit audio line to the output circuit.

Control lines are labeled in several different ways:

- α (alpha). Prefix for lines from the outside world. Unlike the logic lines, for which a high level generally indicates a “yes”, these alpha lines are usually at ground for “yes”. Example: αD is the line from the PTT bar on the desk mike.

- β (beta). Prefix for lines from the optocoupler module. These lines are also low for “yes”.
- δ (delta). Prefix for a line from an open-collector circuit. Example: Line δ2T will turn on the two-tone oscillator.
- TTL level lines are labeled with no prefix, and one or more capital Latin letters. Example: W is the control line which commands the CMOS switch to enable WAV audio from the computer, and turn off the microphone.
- Lines marked with a single digit merely indicate connections to another portion of a schematic, or to another page in a group of schematics.

Figure 10.2; Schematic symbols.



11. Automatic microphone selection. Changing microphone plugs is a nuisance that may try one's temper. The *Hapirat* has three mike jacks, six mike adjustment controls, and automatic mike switching. The center jack is for the boom-mike headset; this is the default and is used with the footswitch. When the PTT bar on the desk mike is pressed, the panel automatically switches to the desk mike input. (And the transceiver receives a closure on the PTT line.) Similarly for the hand mike, hanging from a hook at the front of the operating bench.

Photo 11 shows the three mike jacks and six controls on the front panel. The panel switch labeled **D-AUTO-H** can be used to override the automatic circuits and enable either the desk or hand mike without using the PTT buttons. This is useful when you want to use either of the alternative mikes with the footswitch or with the transceiver switch, without holding a button. It is also useful for mikes which have no PTT bar.

Photo 11. Microphone jacks and controls.

This automatic system is an update of a design previously published by the author in *QST,* reference 30.4.

The panel jacks are all Collins-type 3/16 inch jacks. This type is used as a standard for all the mikes in my shack. Any sort of jacks can be chosen to suit an individual preference. The more commonly available 1/4 inch stereo jacks would be a good choice. Patch cords can be used to adapt the standard plugs to any radio.

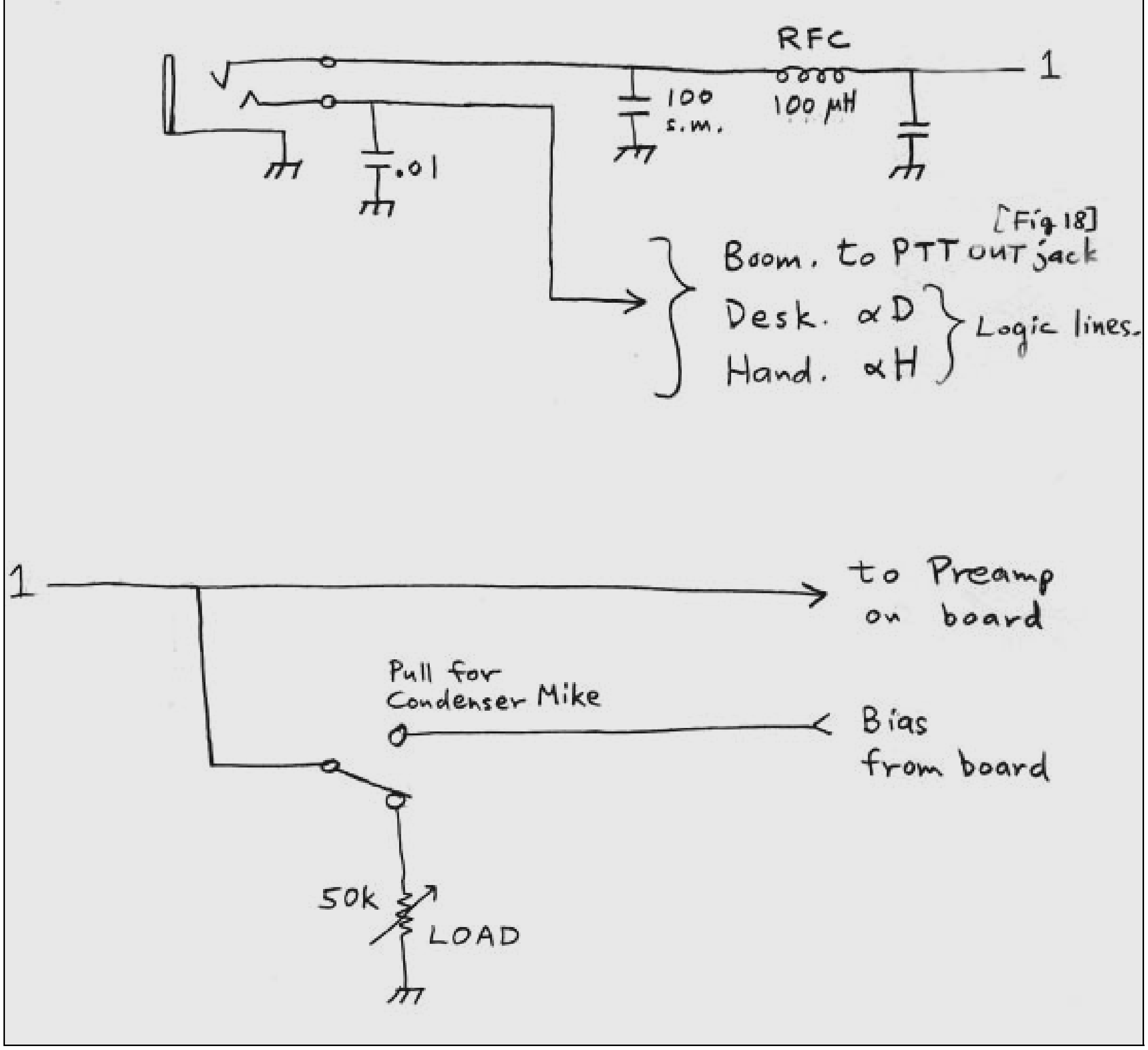
Load resistance controls. The frequency response of many microphones is greatly affected by the load resistance applied. Some microphones are advertised as 600 ohm devices, perhaps to indicate a compatibility with transceivers similarly rated. However, these mikes often produce better results with a higher load resistance. This depends not only on the microphone, but also very much on the individual voice. The *Hapirat* includes three separate 50 kΩ **LOAD** resistance controls to allow adjustment for individual microphones and voices. Modes **REC** and **PLAY** permit convenient experimentation.

The load resistance controls have several other possible uses. For example, if a microphone preamplifier, equalizer, or other audio source is used ahead of the *Hapirat*, its output may be too high for the FET op amp input. Inserting a series 47k resistor in the mike plug at the

Hapirat will allow the load resistance control to be used as an input attenuator; its range will be zero to 50% of the full source output.

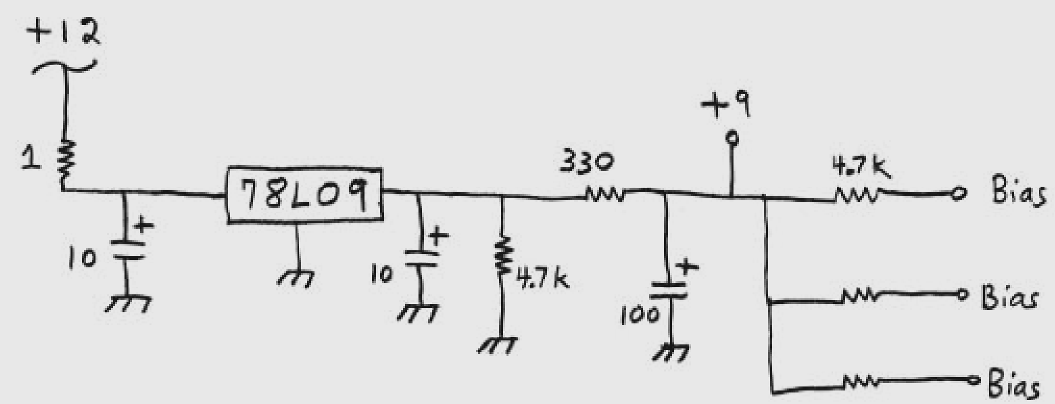
A bias circuit with a push-pull switch on the **LOAD** resistance control is included for use with electret-condenser microphones. When the knob is pulled out, a bias voltage is applied to the microphone jack, and the load resistance is disconnected.

Figure 11.1; Microphone input circuits.



Microphone input circuits. The schematic diagram is shown in Figure 11.1; three of these circuits are used. This diagram shows only the mike jacks and the components located directly at the front panel. The selection sensing is provided by the logic circuit (section 22); the switching is done by the CMOS signal switching circuit (section 13). The bias supply components for electret-condenser microphones are located on the circuit board; the schematic is shown in Figure 11.2.

Figure 11.2; Condenser microphone bias supply.



Each PTT terminal for the desk and hand mikes has a control line connected to the logic section; this will cause the mike to be connected and the radio PTT line to be closed. The boom mike is the default; it is connected whenever one of the other mikes is not in use. The boom mike PTT terminal is rarely used, but it is connected to the **PTT** output jack, in case it is needed.

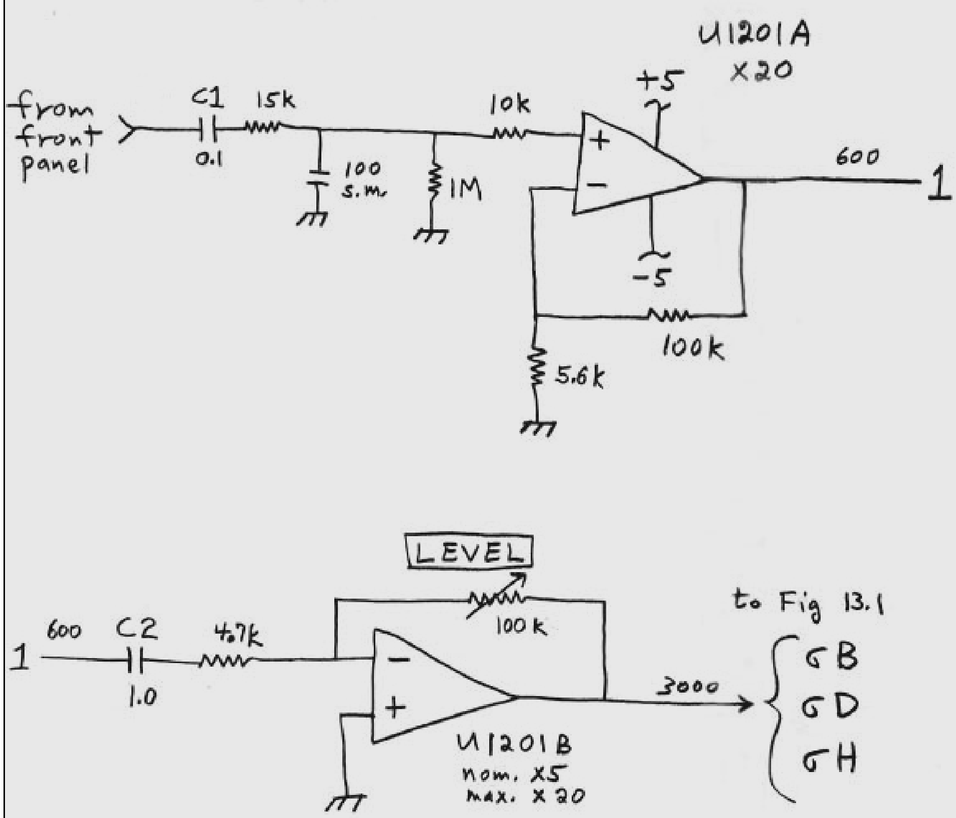
Figure 11.1 includes RF filtering positioned directly at the front panel. There is more RF filtering in the preamps on the circuit board. This schematic also includes the switch which connects either the **LOAD** control, or the electret-condenser mike **BIAS** (when pulled out).

12. Microphone preamps. Separate low-noise FET-input op amps are used for each of the three microphone inputs. Each preamp has two panel controls; **LEVEL** and microphone **LOAD** resistance. Mike switching is done later in the circuit by CMOS switches at a higher signal level; this avoids ground loops and hum pickup that might result from direct switching of the microphone inputs.

The circuit parameters are chosen for a nominal microphone level setting of 25% of full CW rotation – about 10 o'clock. The **LEVEL** control will provide up to 4 times additional gain – not to obtain higher output, but to accommodate lower output mikes. For each mike, the level is adjusted for a mid-scale meter reading. With the separate controls, no levels need be reset when an alternative mike is selected.

The six front panel microphone controls are set-and-forget adjustments; that is, until you change mikes or decide to experiment with the load resistance. There are separate controls for transmitting and for recording.

Figure 12; Microphone preamplifier.



U1201, TL072.

Microphone preamp circuits. The schematic diagram for the microphone preamps is given in Figure 12; three copies of this circuit are used. The three different output connections leading to section 13 are shown in brackets. This schematic shows only the components located on the circuit board; components located directly at the front panel were shown in Figure 11.1. The RF filtering prevents RF feedback into the preamp. The 15k resistor (with the 100 pF capacitor) serves as an RF filter; this resistor is even more effective than an RF choke – it avoids the "holes" at certain frequencies which are typical with chokes. On the other hand, a resistor could not replace the RFC at the panel (Figure 11.1), since it would be loaded by the **LOAD** resistance control.

The approximate nominal gains and signal levels are as follows:

Mike input	30 mVpp
U1 voltage gain	20
U1 output	600 mVpp
U2 voltage gain	5
U2 output	3000 mVpp
Overall voltage gain	100

The 3000 mVpp level is used as a standard in all sections of the *Hapirat.* All inputs are raised or lowered to this level, then applied to the switches and output circuits. This level appears at test point TP14 in Figure 14 below, for measuring with a scope.

The *Hapirat* uses dual supplies on most circuits. Nearly all signals are ground-referenced and DC-coupled; this avoids a need for blocking capacitors. However, two blocking capacitors are needed in each microphone preamp. C1 isolates the op amp input from the condenser microphone bias supply when it is switched on. C2 is required because of the input offset voltage of U1, which may be up to 5 mV, resulting in 100 mV dc at the output. Without blocking capacitor C2, we could have up to 500 mV dc at the output of U2, at the nominal **LEVEL** setting. The remaining circuits could handle this, but with the level control at maximum it would be too much.

The two-stage preamp uses one non-inverting and one inverting stage. The net result is one inversion from input to output; this reduces the possibility of unwanted feedback. The stage with the **LEVEL** control must be inverting, to allow a gain range down to zero.

The microphone circuits have been tested and used with many mikes, including the Heil boom-mikes, the Astatic D-104, the ElectroVoice 664 and 719, and the Icom IC-SM2 electret-condenser microphone.

Parts. The blocking capacitors should be low-noise types, not ceramic.

C1. 0.1 µF, metallized polyester, Panasonic #ECQ-E11O4KZ, DK#E1104.

C2. 1.0 µF, metallized polyester, Panasonic #ECQ-E11O5KZ, DK#E1105.

13. Signal switching. Although in the past I've used relays for microphone and computer audio switching, with few problems, for variation this time I tried CMOS integrated circuit switches. These switches are very convenient and efficient. This section uses five switches. Four more CMOS switches are used in section 20.

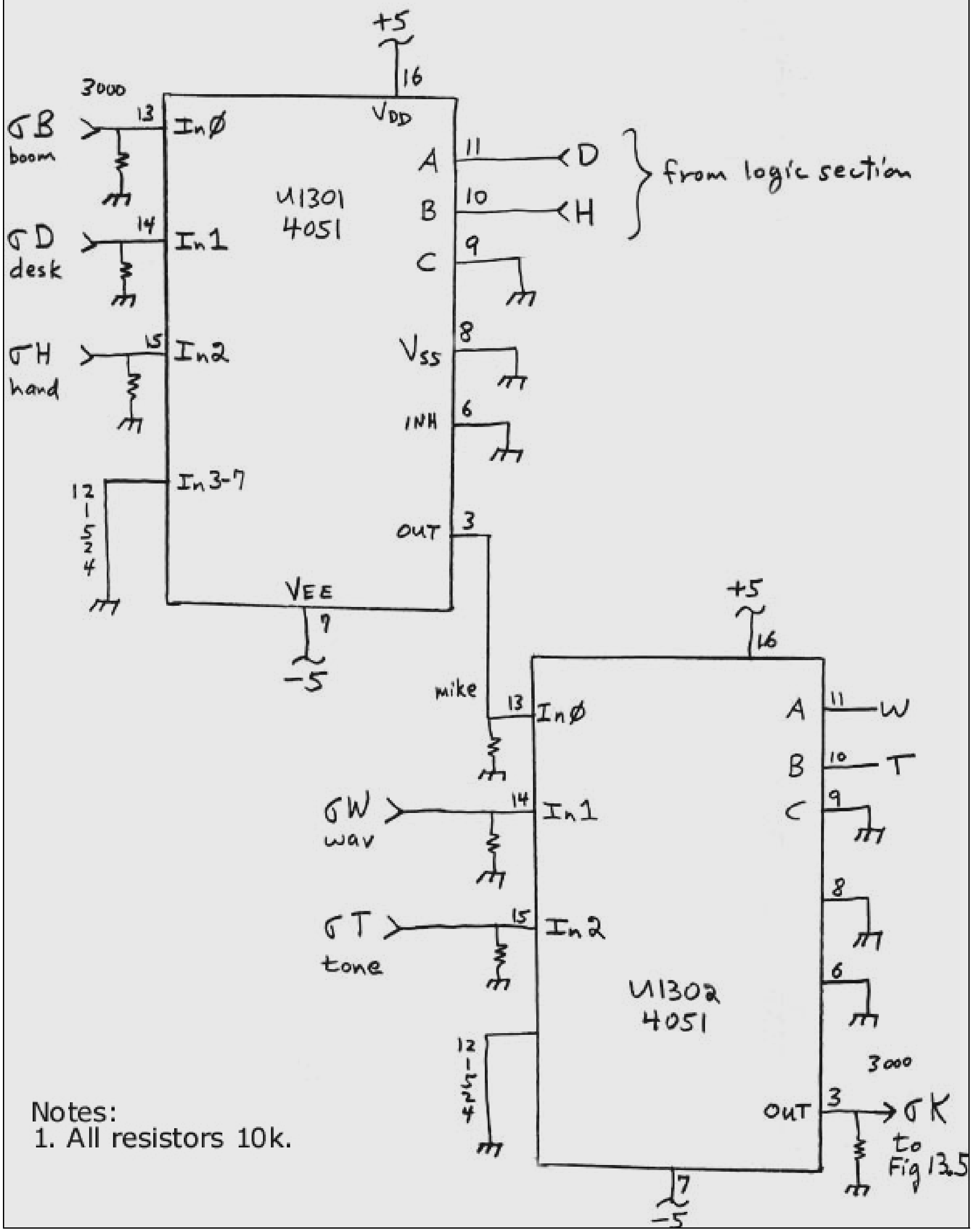
In this section, the first IC selects one of three microphone inputs, while another IC selects mike, computer, or test tone audio. A third IC contains three separate switches, two of which are used here; these switch the meter circuit input, and the receiver or recording signals to the computer. This section also contains a high-pass filter for noise reduction in the transmitted audio.

Here is a list of the signal lines, for reference.

Symbol	Description
σ1	Tone; 1000 Hz
σ2	Tone; 1500 Hz
σ3	Beat note; 500 Hz
σB	Boom mike
σD	Desk mike
σH	Hand mike
σK	Selected transmit audio
σM	Monitor audio
σMR	Receiver audio to meter
σMT	Transmit audio to meter
σR	Audio for receiver speaker
σRC	Audio for recording
σS	Packet spot audio
σT	Test-tone audio
σTX	Transmit audio to output circuit
σV	Audio to meter circuit
σW	WAV audio

The schematic for this switching section is shown in Figure 13 (five parts). The logic control lines are obtained from Section 22.

Figure 13.1; Transmitter audio.



Notes:
1. All resistors 10k.

Figure 13.2; Audio to computer.

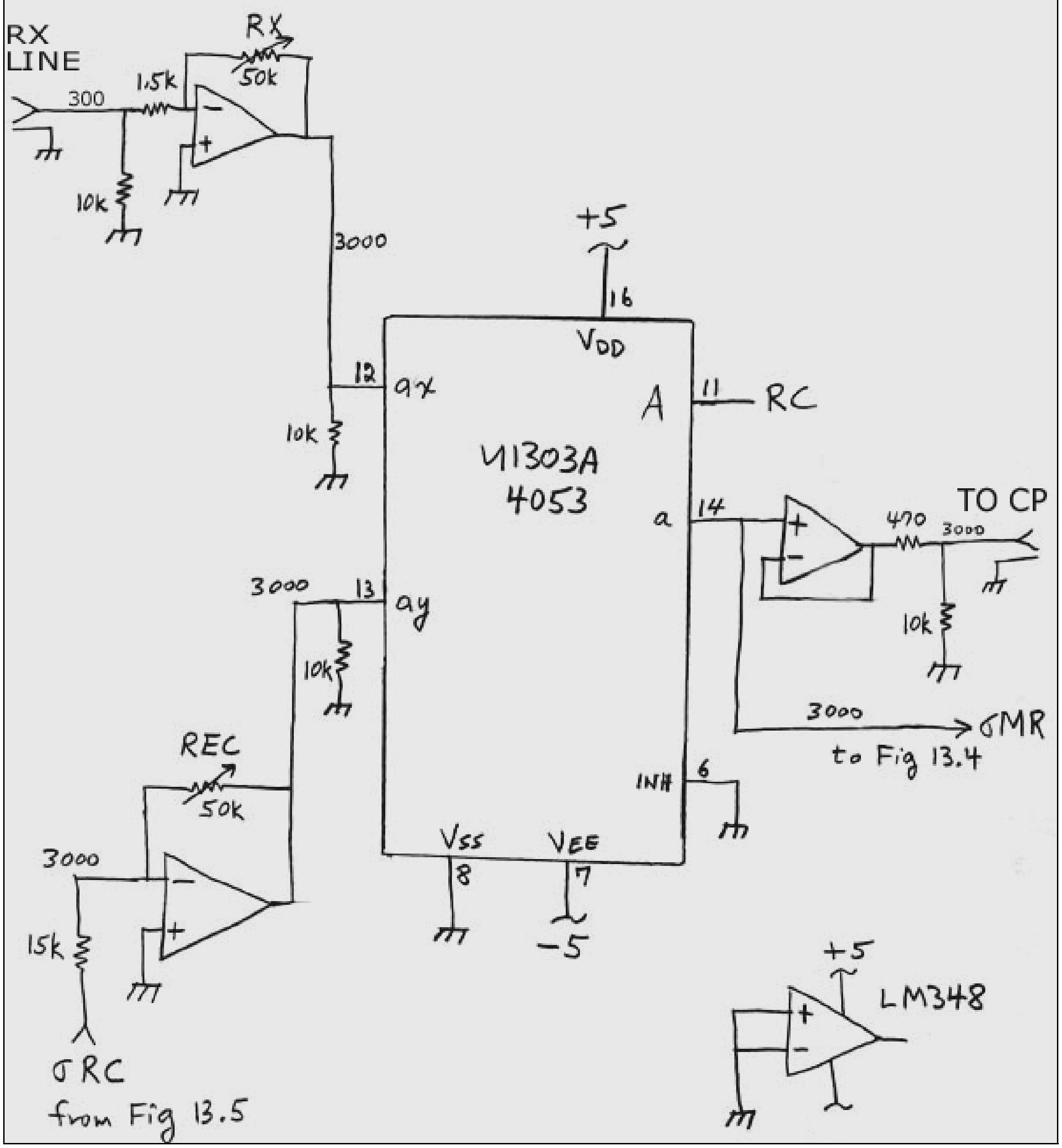
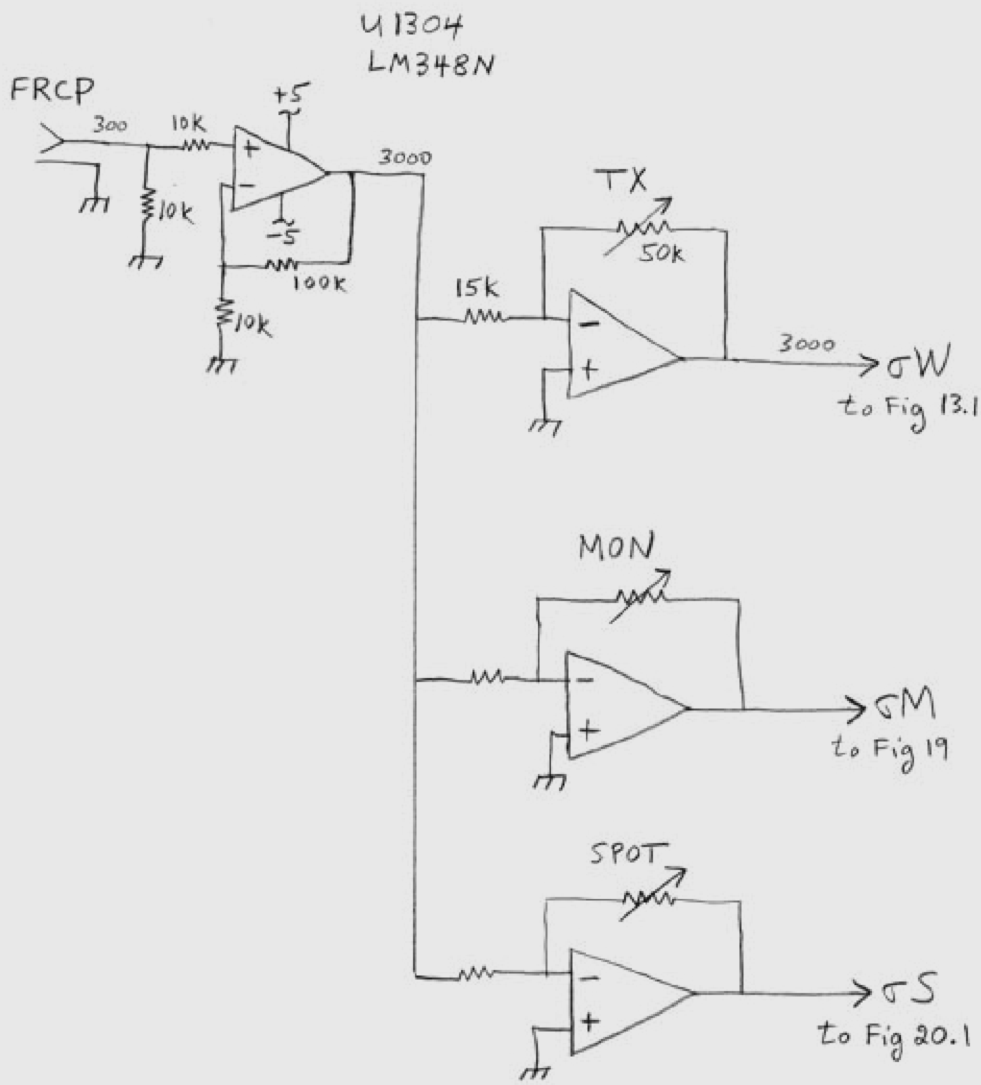


Figure 13.3; Audio from computer.



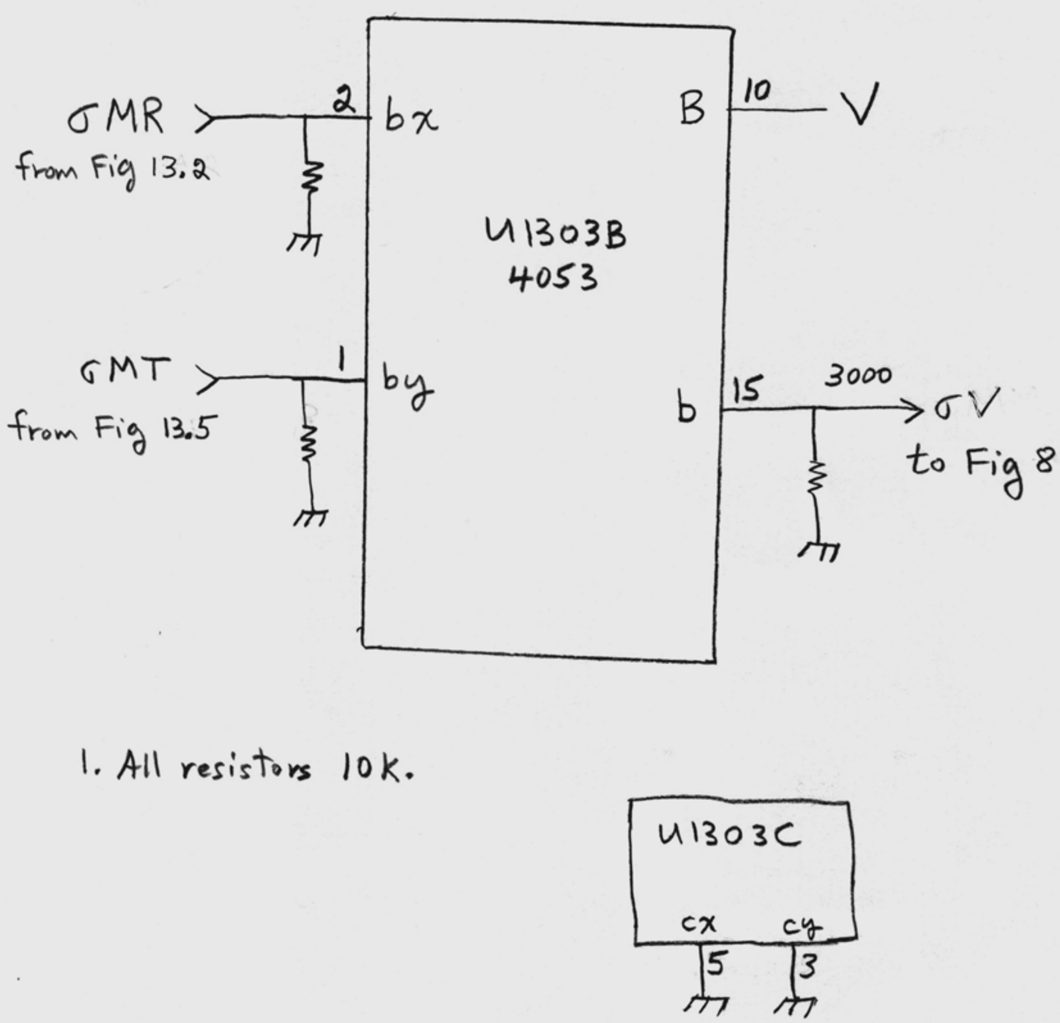


Figure 13.4; Meter selection.

Signal switching circuits. The CD4051BC and CD4053BC multiplexer/demultiplexer ICs are used here as digitally controlled analog switches. The 4051 and 4053 were chosen for audio switching mainly because they are easily controlled by TTL type logic. With V_{DD}, V_{EE}, and V_{SS}, at +5, −5, and ground, respectively, TTL-level control signals may be applied directly to the logic terminals A,B,C.

The CD4051BC is a single 8-channel analog multiplexer/demultiplexer. In this application the switch is powered by the +5 and −5 volt rails; this enables it to accept TTL-level control signals. There are three logic terminals, for binary-coded logic inputs 0,1,...,7. However, only three channels are used here, 0,1,2; this results in very simple data input, without the need for binary encoding. The binary data input values, for data lines A, B, and C, are 1, 2, and 4. Line C is held low. With the 1 and 2 data input lines held low, the resulting binary value is 0, and signal input line 0 is enabled. Input 0 may be thought of as the default. For the microphone switching IC, U1301, this is the default for the boom mike. If data line A is high, then signal input 1 is enabled; now the desk mike is connected. Similarly, data line B is for the hand mike. Thus, for a SP3T switch, only two TTL data lines are needed.

The second 4051, U1302, is also used as a SP3T switch, to select microphone, computer audio, or test tones. An early plan was to use only one IC, to choose among the five possible inputs – it did seem a waste not to use as many of the available eight inputs as needed. However, this would have required a few extra logic gates to generate the binary data inputs. Also, this method with two signal switches allows more flexibility for possible future modifications or additional features.

The third switch package, U1303, is a CD4053B, which contains three separate SPDT switches. Two of these are used: for meter switching (rx/tx), and computer input selection (digital reception or recording). This IC is very easy to use. For example, a TTL high at data input line A switches signal output a from input ax to input ay.

These ICs were chosen because of their compatibility with TTL systems. An alternative choice for a analog switch, the 4066, uses CMOS level data inputs and is more difficult to interface with TTL. Also, the TTL circuits used here are less susceptible to malfunctioning from RF pickup.

The CMOS signal switching circuits in the *Hapirat* are fairly compact; the data lines are not connected to panel switches, connectors, or the outside world. The 10k resistors at each signal input and output help to avoid excessive sensitivity and malfunctioning due to RFI. The unused inputs are grounded. No RFI problems have arisen with the *Hapirat*, even using nearby antennas that do trash my telephones, TV, VCRs, and computers. Probable factors in this happy result are the excellent LMB shielded enclosure, and the bypassing and filtering of all front and rear panel jacks.

The use of TTL to control the CD4051 series of signal switches depends on the supply connections. These must be V_{DD} = 5V, V_{SS} = 0V and V_{EE} = –5V. The CMOS packages should be handled as any sensitive device, with grounding pads, etc. Circuits are built onto the board only after the board is mounted in the chassis and the ground lugs are secured to the central aluminum support rail. I don't use wrist-grounding devices, but I keep the chassis grounded while handling sensitive devices and rest my wrists on the aluminum chassis edge. (New Mexico at times has extremely low humidity. Charged dust particles often fly about in severe windstorms; one can sometimes draw a 1 inch spark off an ungrounded antenna wire.)

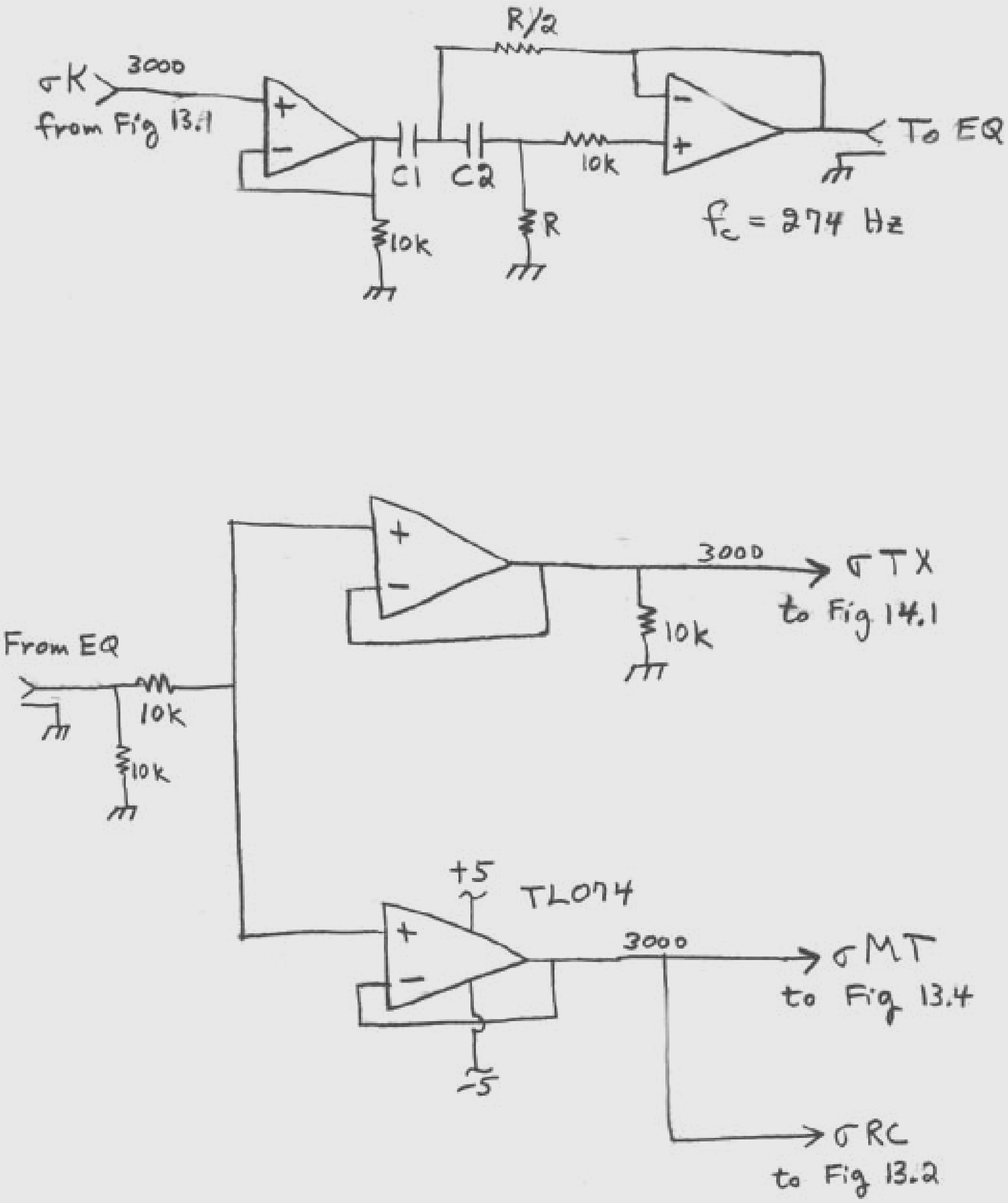
Note on load resistors. The general rule of using a 10k resistor at each switch input and output, to protect the CMOS circuits, must be modified in certain cases. Unlike an op amp voltage follower, which simply duplicates the input at the output, the CMOS switch effects what is more like a direct connection. It follows that when switches are used *in tandem*, several load resistors can appear in parallel, sometimes causing an unexpected load on a previous circuit. Thus a minimum number of load resistors is used, only enough so that a load is seen by each input and output.

Parts information. A data sheet covering both of these CMOS switch types can be downloaded from:
www.fairchildsemi.com/ds/CD/CD4051BC.pdf
or from *The TI Signal Switch Data Book* at:
http://www-s.ti.com/sc/psheets/scdd003/scdd003.pdf
These very inexpensive IC switches are readily available:
MO#511-4051 STMicroelectronics HCF4051BEY.
MO#511-4053, STMicroelectronics HCF4053BEY.

High-pass filter. Following the two 4051 switches which select transmit audio is a simple one-stage high-pass filter for reduction of hum and computer rumble. Designed for a –3 dB cut-off frequency of 274 Hz, the measured attenuation is as follows:

Frequency	**Attenuation**
60 Hz	30 dB
215 Hz	6 dB
280 Hz	3 dB

Figure 13.5; High-pass filter.



The HPF circuit is shown in Figure 13.5. The first op amp is a voltage follower which provides the filter stage with a low-impedance source. Following the filter stage are voltage followers which provide signals for the various circuits which require transmit audio: σTX for the transceivers, σMT for the meter, and σRC for recording WAV files on the computer. The circuit includes optional jacks for inserting an external equalizer loop into the circuit; a jumper connects the jacks when no equalizer is used. Inserting the equalizer at this high-level point avoids hum problems. The equalizer must of course be configured for this level.

The filter stage is a unity-gain maximally-flat Butterworth filter in a voltage follower configuration. The required value of Q is $\sqrt{2}/2$.

This configuration is the simplest to apply in regard to component selection. It also involves the simplest formulas. One first chooses the cut-off frequency f (in Hz) and the capacity C (in farads). Then the required resistance R (in ohms) is given by

$$R = \frac{1}{\sqrt{2}\,\pi\,C\,f}$$

With the convenient value of 0.01 µF for C,

$$R = \frac{22.5 \times 10^{3}}{f}$$

The typical cut-off frequency choice of 300 Hz thus requires R = 75k. Using the next-higher common value, 82k, we obtain a cut-off frequency of 274 Hz. To avoid having to find a 41k resistor for the R/2 position, we simply use two 82k resistors in parallel.

The cut-off frequency may be changed by using the inverse relation between f and R. For example, AM operators may prefer a cut-off frequency f = 113 Hz. This is obtained with R = 200k; we could use two 100k resistors in series for R, and one for R/2.

The 10k resistor at the non-inverting input does not affect the response; it serves merely to protect the op-amp; this protection is used throughout the audio panel. A FET-input op amp is used, to avoid any possibility of the op amp loading the circuit and altering the desired response.

This high-pass filter for hum and rumble reduction is in the circuit in all transmit conditions, and for recording. For a notch filter which attenuates only 60 Hz, see reference 30.1, Part 4.

Filter components: High-quality capacitors are used. Specifying a fixed common value for capacitors in all audio filters and oscillators in the station makes stocking easy; these capacitors are always on the shelf.

C1,C2. Polypropylene capacitor, 10 nF, 2% tolerance. Panasonic #ECQ-P1H103GZ, DK#P3103.

R. Carbon-film resistor, 82k, 5% tolerance.

R/2. 41k. Use two 82k resistors in parallel. This avoids the need for obtaining 1% resistors. Select the three 82k resistors from the same batch, using a DMM for the closest match; the exact value is not as important as the matching.

Ref: Walter G. Jung, *IC Op-amp Cookbook,* Howard W. Sams & Co., Inc., Indianapolis, 1974, pp 331-333.

14. Outputs. The *Hapirat* uses a special method for audio signal transmission to avoid hum problems. This involves amplification, high-level cables, attenuators, and isolation transformers to prevent ground loops. The high levels permit the use of ordinary phono jacks which are grounded to the chassis, and ordinary cables. The attenuators and isolation transformers are easily built into small modules placed near the transmitter.

This method provides the greatest flexibility; any transmitter can be driven, and any number of transmitters, with no internal modification to the *Hapirat.* Low-level outputs are also provided for use when the special connecting cables are not available, perhaps for quick hookup with a backup transceiver five minutes before a contest.

There is no limit to the number of outputs that can be provided for transmitter audio inputs. A single 14 pin quad op amp will drive 4 outputs. In this panel, 6 outputs are provided. Four of these, **A** through **D**, are high-level outputs, intended to drive the special connecting modules. Low-level outputs E and F are also provided.

Output jacks. From right to left: **A, B, C, D** unbalanced at 3000 mVpp; **E** and **F** balanced at 300 and 60 mVpp.

Output circuits. The circuits are shown in Figure 14 (two parts).

Figure 14.1; Transmit audio outputs.

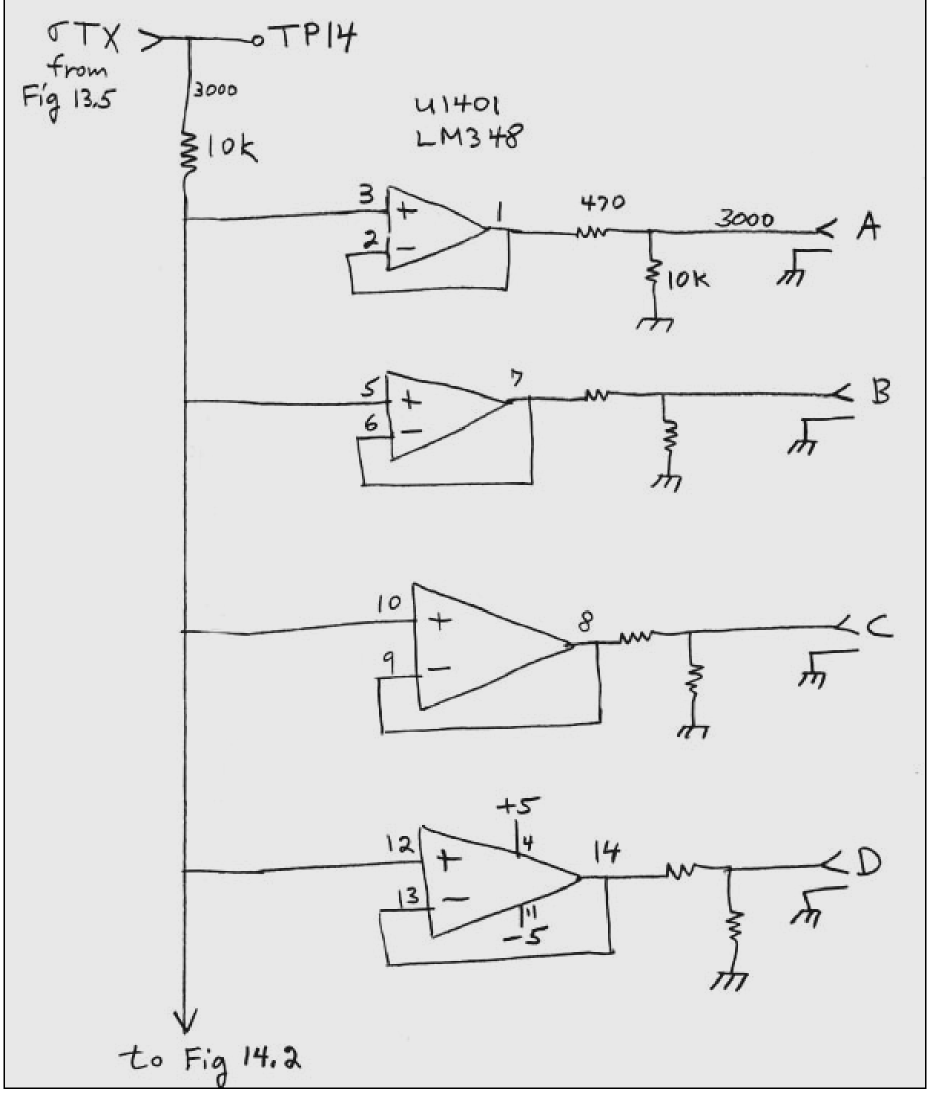
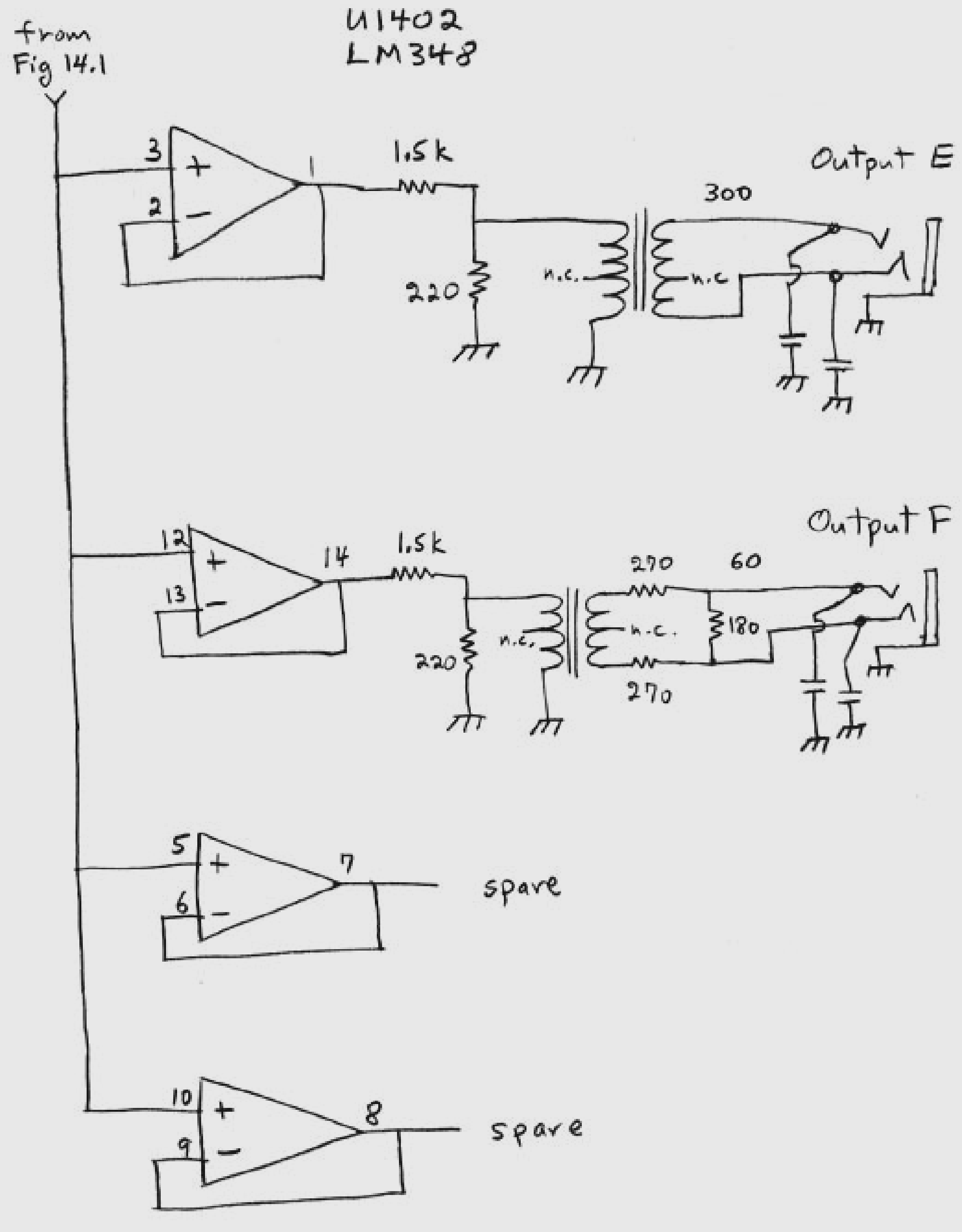


Figure 14.2; Transmit audio outputs.



For isolation between transmitters, separate voltage followers are used at each output. The four main outputs are at a level of 3000 mVpp. The 470 ohm resistors limit the current in the event of an output short circuit; the op amps have internal protection, but it is good practice to include circuit protection. These outputs are intended to be used with high-impedance attenuators in the connecting modules; these will be described in section 16.

High-level outputs **A** through **D** are single-ended, not balanced. The high-level method suffices to avoid hum problems. This method was chosen to allow use of ordinary phono jacks grounded to the chassis, ordinary single conductor cables, and transformers and attenuators at the transmitters. When well-shielded stereo cables are available and desired, balanced outputs **E** and **F** can be used. More outputs of any sort can be added easily.

Balanced output **E** is at a level of 300 mVpp. This output was included to directly drive my homebrew transceiver at it's special rear panel transmitter audio line input jack, designed for 300 mVpp. This jack was included in the 1990 homebrew design, and was part of an early version (reference 30.4) of the high-level transmitter audio cable scheme used here. The 20 dB boost in that version avoided all hum problems, even without an isolation transformer. I now use the high level output **A**, with a transformer and an attenuator at the transmitter. This 40 dB boost is perhaps more than needed, but to avoid hum problems it pays to take all precautions in the beginning, rather than try to patch things up later. This output can also be used with a simple attenuator at the PATCH input of other transmitters; it has been tested with an FT-1000D, with excellent results.

The last output **F**, at 60 mVpp, is for the usual transmitter microphone input jacks. The nominal mike input level of the *Hapirat* is 30 mVpp, so this output represents a 6 dB increase. This moderate increase can help avoid hum problems, even when not using the high-level outputs and special attenuation/isolation cables. The transmitter mike gain will be turned down slightly, of course. Too much added gain could overload the transmitter microphone input circuit, but this amount of gain is reasonable. After adjusting the mike gain settings in **SETUP** mode, this arrangement makes your modern low-output mikes look like good-old-fashioned high-output mikes to the transmitter.

The two low-level outputs **E** and **F** include isolation transformers inside the *Hapirat* enclosure; these are included to facilitate transitional situations and permit operation before special cables are made up for the main outputs. Outputs **E** and **F** are balanced, with 1/4 inch stereo jacks, and can feed an unbalanced transceiver input, minimizing hum problems. Unbalanced low-level outputs can be obtained simply by using a mono plug; the shell will ground one side of the output transformer. Connection diagrams for using these outputs in various ways are shown section 16.

This high-level transmitter audio method is analogous to that used for the channel 3 signal sent from a VCR to a TV set. The signal is much stronger than necessary to get a
picture; AGC in the TV set activates attenuators to reduce the gain. The result is an excellent signal-to-noise ratio. If only the minimal signal needed to get a picture was sent, with the TV set front-end wide open, the result would be
a very snowy picture, with more susceptibility to TVI.

Parts. A source for high-quality isolation transformers is given in section 25.

Output measurements. Measurements are given in the following table, in mVpp. The load used is 600 ohms, although for the first four outputs a much higher load impedance is intended.

Jack	**Nominal**	**Open cir.**	**Loaded**
A,**B**,**C**,**D**	3000	3000	1700
E	300	350	250
F	60	67	54

15. Computer audio interface. The **TO CP** ("To Computer") output jack on the *Hapirat* is used to feed receiver line audio to the computer for digital decoding, or to feed mike audio for WAV recording; the selection is done automatically by the **MODE** switch. In some installations, this switching is accomplished by changing cables; a prime goal in the *Hapirat* design is to avoid this. Other arrangements use a contest program to control recording, with the mike connected directly to the computer; this would conflict with the *Hapirat* requirement that the rig will work normally with no cable changing when the computer is off, or in crash condition.

Output from the **TO CP** jack is at the nominal level of 3000 mVpp. In the connecting module this is attenuated to an appropriate line level for the computer, 300 mVpp (i.e., roughly 0.1 Vrms), and an isolation transformer is used to avoid ground loops. The module circuit is shown in Figure 15(1). The circuit assumes a normal computer line input impedance of about 10k; the load resistors on the transformer help to provide a fairly constant output level under reasonable load variations.

To accommodate computers that may require a different input level, the circuit shown in Figure 16(A2) can be used.

Figure 15; Computer audio interface.

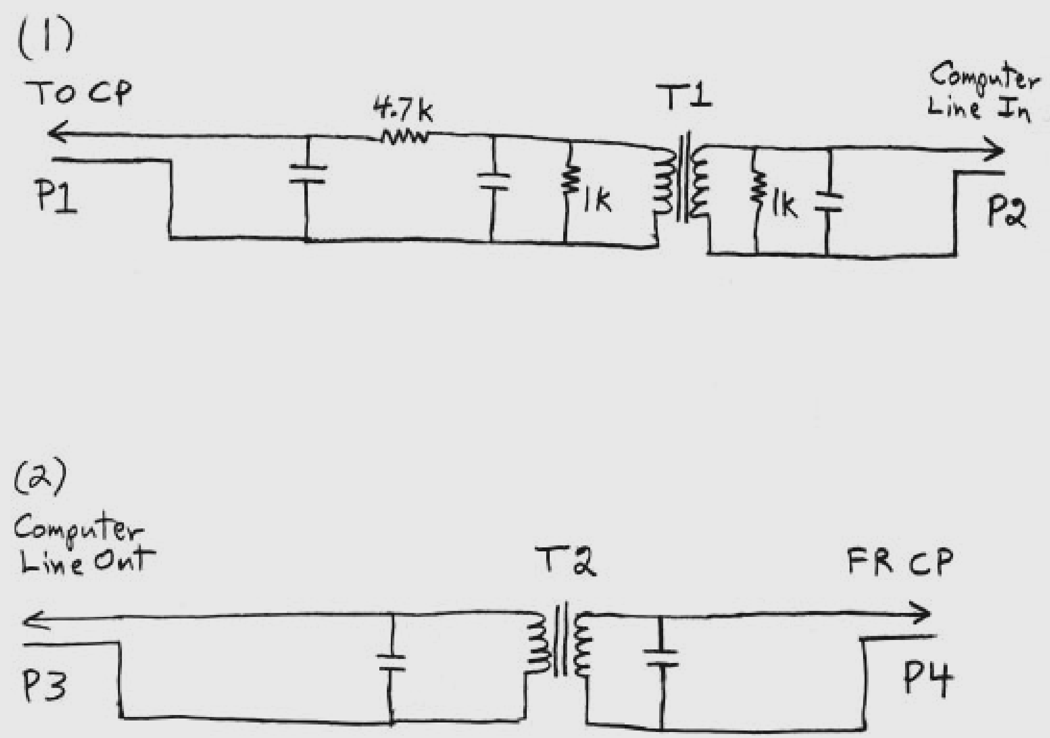


Figure 15 component details:

P1,P4. Phono plugs; 6 foot well-shielded cable.

P2,P3. To fit computer audio jacks, typically 1/8 inch stereo plugs. Use 1 ft well-shielded cable, such as RG-174. Connect the center conductor to one terminal only, usually the tip, which is usually the left audio side. Configure the contest program accordingly.

This distribution of cable lengths puts the module close to the computer, minimizing the cable length at the computer. On some computers, a connector "ground" is not a solid ground to the enclosure, but leads directly to a circuit board, with questionable shielding and grounding.

T1,T2. 600 ohm telephone transformer. A source for high-quality isolation transformers is given in section 25. The center-tap terminals found on most telephone transformers are not used; the pins may be snipped-off.

Capacitors. .001 means 1 nF. 100 means 100 pF. Ordinary disc ceramic types.

Grounds. Note that the usual ground symbol is not used in Figure 15. There are two different grounds, the computer ground and the panel ground.

Cables. Ordinary audio cables with spiral-wound shield wires do not provide as good shielding as the braided shields found in RF cables. Type RG-174 cable has a 95% tinned copper braided shield, and is very convenient for all station audio and control line connecting cables. It is only 1/10 inch in diameter, but nevertheless is surprisingly durable, due to a copper-coated steel center conductor. A very large number of these cables have been used for many years in my shack, with no failures. They even survive years of use and constant flexing as test cables on the workbench. Specs: www.nemal.com/catalog/Pg25.jpg and the following pages.

Setup. The recording level requires special attention; distortion can result if the level is too high, while the signal-to-noise ratio will be poor if the level is too low. The **REC** and **PLAY** modes will help in the adjustment. A scope connected to an unused *Hapirat* output terminal, used to preview sound files in advance of a contest, is the best way to ensure high-quality WAV transmissions. A scope is able to show peak distortion that an on-screen computer recording "scope" indicator does not reveal. For digital decoding, the program instructions should be consulted for proper level adjustment. Once a line input level is fixed on the computer, it need never be changed; the *Hapirat* panel controls and meter are used to maintain the proper levels.

Module construction. The best way to construct an audio module is in an aluminum mini-box, but so much trouble is not necessary. Empty plastic 35 mm film "cans" provide good insulation; a module can be built in only a few minutes.

Once upon a time these were real aluminum cans; I have only a few, saved from the 1940s. However, the lack of shielding with the plastic film "cans" has not been a problem. The cables are shielded and grounded on one end; RF bypassing is included. Shielded isolation transformers are sometimes seen, but this is more important in the vicinity of AC fields, as in a device with a power transformer. The *Hapirat* runs on 12 Vdc, externally supplied, precisely to avoid this problem. The modules should not be positioned adjacent to AC transformers, of course. The high-level signal transmission method is probably the main factor contributing to the success of this system in avoiding hum, noise, and RFI.

When aluminum mini-boxes are used, only one of the cables can be grounded to the box; the panel side of the circuit would be best. The other cable should be insulated from the box and connected directly to the circuit.

A film "can" is better than leaving the circuit open on the bench for two reasons. First is the obvious insulation, preventing stray shorts. Also important is the cable strain relief provided by the cable ties inside the can. Contesters want 100% reliability; this means that no wire anywhere in the shack can be left hanging loose from its soldered end; cable ties and tie-downs are essential.

The **FR CP** ("From Computer") input jack on the *Hapirat* is used to feed computer audio into the panel. This may be digital signals, contest SSB transmissions from WAV files, packet spots, or even music. The connecting module needs only an isolation transformer. The panel assumes an input level of 300 mVpp, and amplifies this as needed for the various circuits. The computer line output level is set only once to achieve this level; the panel controls and meter are then used to obtain the required outputs. The circuit is shown in Figure 15(2); the module looks the same as the lower module in Photo 15.

After all the circuits were carefully connected with attenuators and isolation transformers, I was surprised and discombobulated – hearing a slight buzz in the packet cluster speaker! What went wrong? It turned out that the laptop computer, with its switching power supply module, was previously grounded by the rig, but was now floating free. The optocouplers and isolation transformers removed *all* the grounds from the computer. A heavy lead to the main station ground bus solved the problem. Thus I learned that the goal is not to eliminate *all grounds*, but *ground loops.* The computer did require one good ground.

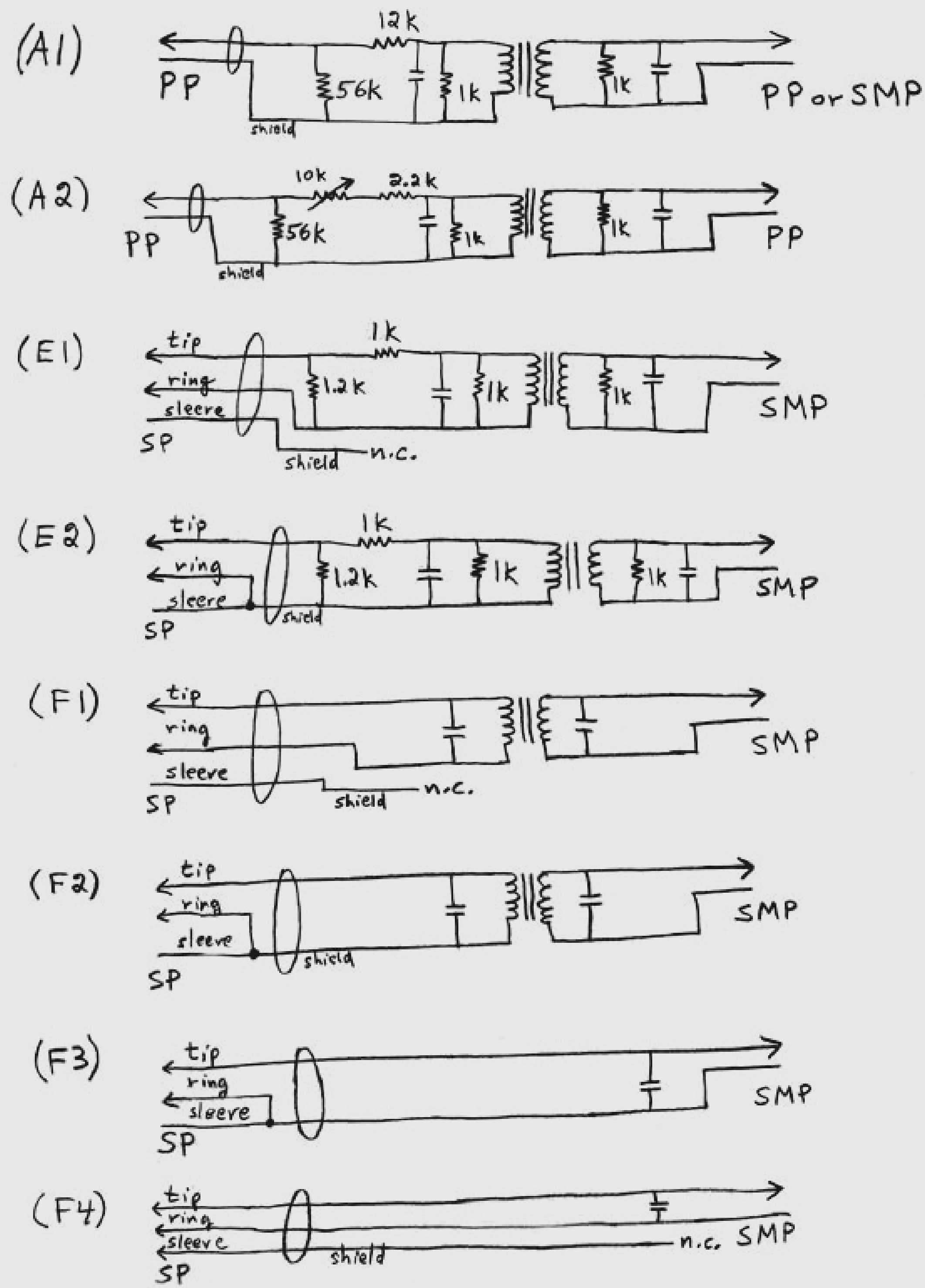
16. Transceiver audio interface. There are several ways to feed transmit audio into transceivers, depending on the *Hapirat* output jack selected. The highest output level, 3000 mVpp, offers the best protection against hum pickup and RFI, although 300 mVpp is usually adequate. With these high-level outputs, modules with attenuators and isolation transformers are used.

The *Hapirat* is designed to drive several transceivers simultaneously; the six outputs are at fixed levels. Each individual transceiver module is built to accept the fixed level and drive a specific transceiver; no readjustments are required when switching rigs.

The advantages of outputs **A**,**B**,**C**,**D** are the convenience of the grounded phono jacks and the high output, which masks any possible hum pickup. The advantage of output **E** is the balanced configuration, which may be best for long runs. Output **F**, a low-level balanced output, is provided mainly for emergency situations when a proper cable and module is not available.

Schematics for cables and modules for use with the six output jacks are shown in Figure 16. These circuits provide 60 mVpp at a transceiver mike input jack, assuming a 600 ohm load at the transceiver.

Figure 16; Transceiver audio interface.



Transceiver audio module details:

A1. For outputs **A** through **D**, 3000 mVpp, unbalanced. Attenuated to 60 mVpp for a transceiver mike input. With a phono plug on the output end, this circuit can also be used to feed transmit audio into a transceiver PATCH jack on its rear panel. Using the rear panel PATCH jack keeps the front of the transceiver clear of cables and avoids the need to acquire a special front panel mike connector.

A2. Same as A1, except a trimpot is added. This circuit is useful in cases where the transceiver PATCH jack requires a higher audio level. Output is 60 to 300 mVpp.

E1. For output **E**, 300 mVpp, balanced. Attenuated to 60 mVpp for a transceiver mike input. Stereo cable, balanced feed.

E2. Same as E1, except this circuit uses a mono cable and unbalanced feed.

F1. For output **F**, 60 mVpp, balanced. Stereo cable, balanced feed.

F2. Same as F1, except this circuit uses a mono cable and unbalanced feed.

F3. Same as F2, except no isolation transformer is used. Not recommended, but sometimes usable when a transformer is not available. The transformer inside the audio panel does not provide isolation, since the secondary is grounded at the panel by the connection inside the plug.

F4. Same as F1, except no isolation transformer is used. Not recommended, but often usable when a transformer is not available. This circuit is much better than F3, and should be used when well-shielded stereo cable is available. The transformer inside the audio panel provides a balanced drive, and keeps the panel and transceiver grounds separated.

The resistors in the attenuator circuits are chosen with values low enough to provide suitable loading so that the output will be relatively independent of load impedance. The variation in output level for loads between open-circuit and 600 ohms is not excessive.

More variations are possible, and will occur to the operator when a special situation arises. When calculating the resistor values for an attenuator, the load impedance at the transceiver, the source impedance inside the panel, and the design load impedance at the jack, must be considered.

For example, consider circuit E1. We first decide that we want the two 1k resistors at the transformer; this provides a minimum amount of loading and prevents wide changes in output level under different load conditions. Call the series resistor R. The 600 ohm impedance at the transceiver, together with the two 1k resistors, combine to present a 273 Ω load to the signal at the right side of R. The 60 mVpp spec means that we want 20% of the input signal, so R needs to be 4 x 273 = 1092 Ω. We choose 1k for R, so we'll get 64 mVpp output, neglecting the transformer loss of about 1 dB. So far we have a load of 1273 Ω for the panel jack, which was designed for a 600 Ω load. A parallel resistor of 1.2k reduces the load to 618 Ω.

Module components used in Figure 16. See section 15. Also:
PP. Phono plug.
SMP. Special mike plug to match transceiver.
SP. Stereo plug, 1/4 inch. When a stereo plug is not available for circuits E2, F2, and F3, a mono plug might be tried. Usually, the mono plug sleeve will automatically ground the ring terminal of the stereo jack.
All capacitors. 1 nF.
n.c. means no connection; the cable shield continues into the module or plug, but is not connected.

Cables: 1 ft to the transceiver, 6 ft to the audio panel. This distribution of cable lengths minimizes the ground cable length at the transceiver. On some transceivers, the mike connector "ground" is not a solid ground to the enclosure, but leads directly to a circuit board, with questionable shielding and grounding.

17. Computer control interface. The *Hapirat* was built to simplify my station arrangement by combining seven separate gadgets. However, for some functions separate devices are preferable – they can provide greater flexibility. This is the case for the computer control interface. With a multitude of computer operating systems and ham operating programs, a separate interface device allows easy changes for different systems, and adaptability to new systems. Thus the *Hapirat* panel proper accepts only normal PTT and KEY signals from an interface circuit; the interface task itself is accomplished in a module on a computer cable. Thus the *Hapirat* will operate with a variety of computers and programs, including those not yet designed.

The interface circuit uses optocouplers; this avoids the possibility of hum resulting from a direct ground connection between the *Hapirat* and the computer. Bipolar transistors were used previously for computer interface circuits with no problems, but many operators prefer the optocouplers, so I tried them here. Having heard of so many troubles with the usual optocoupler circuits, I knew that something different would be needed to ensure 100% reliable operation in all situations.

Most of the troubles reported involve the inability of an optocoupler to sink a moderate current to a low voltage, while at the same time drawing little current from a parallel or serial port. When Darlington optocouplers are used to increase the current sinking capacity, the device often will not sink the line to a sufficiently low level, due to the Darlington configuration. Recently, further problems have arisen in regard to the industry standard for parallel and serial ports. While the standard specifies current sourcing ability which is sufficient for the ham circuits usually seen, some reports state that manufacturers have not followed the standard. Thus a circuit that has worked for several years in a ham shack sometimes might not work on a new computer or with a new operating system.

The solution adopted here is to use optocouplers only to drive IC comparators, then TTL logic, and finally bipolar transistors to control the transceiver PTT and KEY lines. The ability of the comparator to sense small voltage changes while drawing negligible input current overcomes the optocoupler's infrared diode limitations. Also, the ability of the comparator to sink a considerable current to a low voltage overcomes the optocoupler's photo transistor limitations. Bipolar transistors (in TTL driver packages) are used to directly activate the transceiver PTT and KEY lines.

The comparators link the optocouplers and the bipolars. Thus the system enjoys the advantages of both methods, and the disadvantages of neither. The optocouplers are located in a module near the computer; the comparators and bipolars are located inside the panel. With care, the optocouplers may be located inside the computer connectors, making a very neat and compact arrangement.

This system has a universal character; there is no dependence on any particular type of computer port. The system will work with a serial port, a parallel port, or any other port in use today or introduced in the future.

Although many operators use command control (direct connection between the computer and the transceiver), this is only possible with later-date radios with internal microprocessors. Older radios, and homebrew radios without microprocessors, require a PTT and KEY control interface.

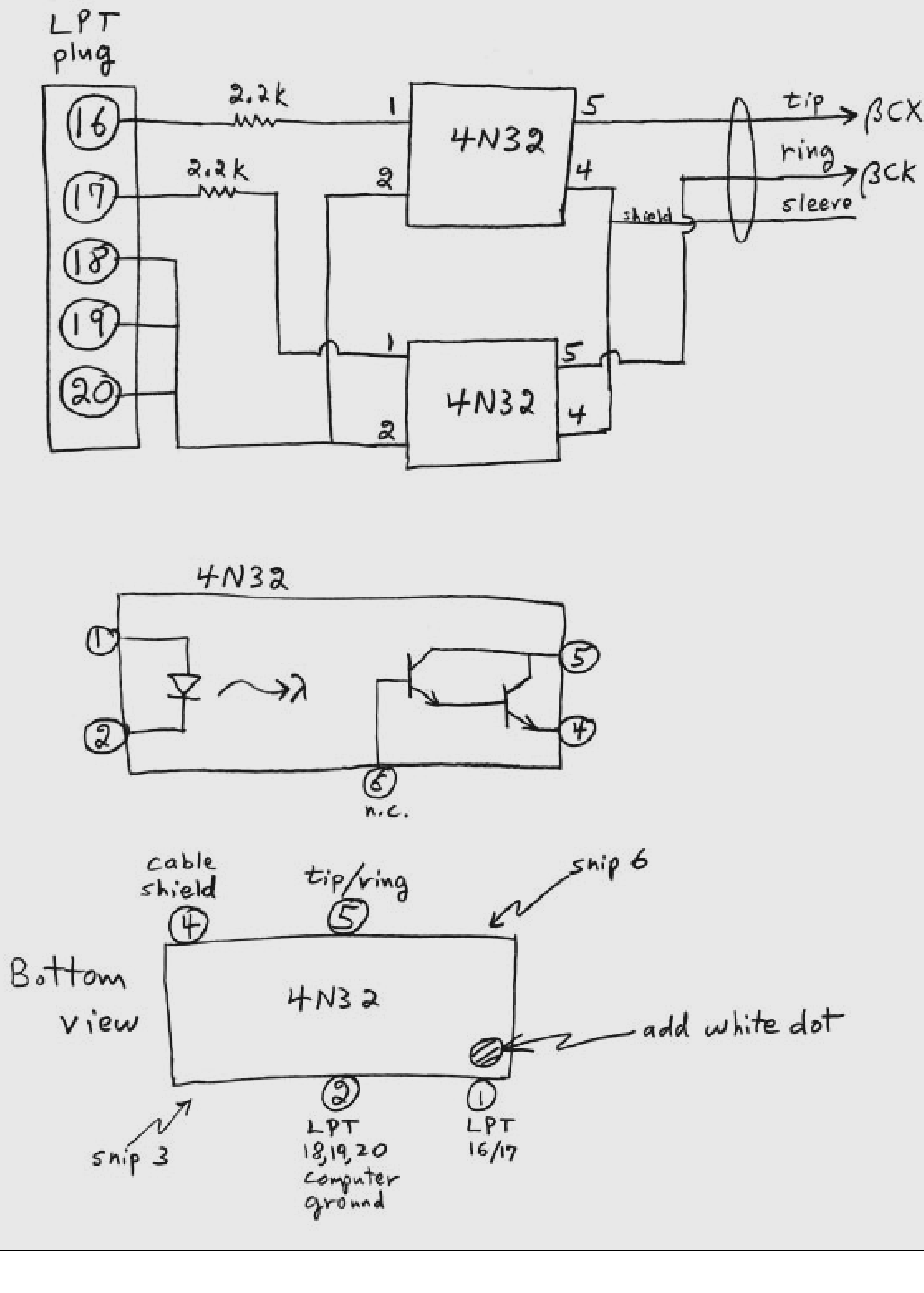
These circuits have been tested on Dell and Toshiba laptop computers using *WriteLog*. The serial port module is useful in situations where the parallel port is used for an old printer. When a free serial port is not available, a USB adapter can be used. I am currently using a laptop USB port with a type 70607 USB-serial adapter from www.pccables.com. Other contest programs may require different connections; consult the instruction manuals.

The panel requirements for PTT and KEY functions at the **CPTR** jack on the *Hapirat* are identical to those for any modern transceiver, so far as polarity and direction are concerned; the current sinking and low-level requirements are much more lenient than typical. Thus any circuit designed to drive a transceiver will easily drive the *Hapirat.*

Another advantage to having the computer interface operate through the panel is the ability of the **MODE** switch to disable computer control of the radio. This is important in many installations, where the computer or contest program may sometimes cause unintended transmissions. The unwanted transmissions may cause interference and equipment damage. With this panel, computer PTT and KEY control is enabled only in modes **CP** and **CPM**.

Parallel port module. The circuit is shown in Figure 17.1; it uses two optocouplers and is built inside a DB25 connector. Photo 17 shows the connector before the upper shell is attached. Spec sheets for the 4N29-32 series of photodarlington optocouplers are available at:
http://www.fairchildsemi.com/ds/4N/4N33.pdf
http://alds.stts.edu/datasheet/Opto/4N29REV4X.PDF

Figure 17.1; Parallel port module.



Parts.

MO#78-4N32. Allowable substitute: 4N33; other types usually have less sensitivity.

DB-25 connector, male, solder cups. CS#DB-25P.

Metallized hood for DB-25 connector, hardware included, chromated plastic with thumbscrews, CS#DBM-25ST.

Parallel port module assembly. An aluminum mini-box, or even a plastic enclosure, would serve well for the optocoupler circuits; building on a small piece of perf-board would be easy. Alternatively, the circuits can be built directly into the DB25 connector which fits the LPT port. The inside of the module is shown in Photo 17. Putting the circuit inside the connector has several advantages. The optocouplers are very close to the computer; there are no long leads that might bring RF into the computer. Also, the LPT terminals are connected only to points inside the plug, and to no further points where they could be grounded accidentally and possibly damage the computer. The exposed 1/4 inch plug does look hazardous when not plugged into the panel, seemingly connected to the LPT port, but the plug contacts are isolated from the computer.

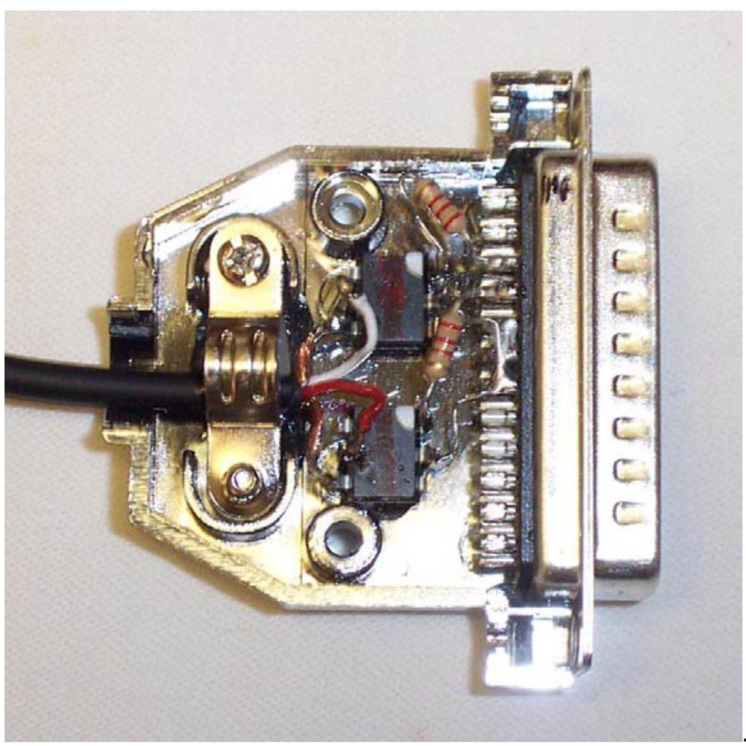


Photo 17. Parallel port plug; inside.

Connecting even a few components inside a DB25 connector can be an exasperating task, with all the parts bouncing around as you try to solder the connections. However, it becomes very easy with one simple trick: secure the main components in place first with 5-minute epoxy cement.

Small pieces of thin cardboard are used for insulation. The plug specified above has a conductive coating; this is very nice for shielding, but shorts to the shell must be avoided. Check to see where portions of the upper half shell of the connector will fall when attached. Note that the cable shield is not connected to the plug shell and should not touch; that is the whole point of using the optocoupler to avoid a ground loop. The cable shield is audio panel ground; the plug shell is computer ground.

Strip the cable jacket to leave the individual wires one inch long; excess length here makes the connections easier. Use a cable tie inside the connector clamp to take up the free space; this provides strain relief to prevent cable failure. Apply epoxy cement to the clamp, the cable, the connector insert with the terminals, and the optocouplers. Use a marker, such as white nail polish, to mark pin 1 on each coupler before cementing. Snip off unused pins to make the connections easier. Photo 17 show that the insert is installed with the needed pins up. A single #24 wire connects the three connector pins and the two optocouplers. Rather than deal with the braided shield and chance a short circuit, a single #24 wire connects to the braid, with each end of the wire leading to an optocoupler.

Parallel port module measurements. The requirements of the interface circuits inside the panel are extremely low; the module at the computer need sink only 0.5 mA. Measurements of circuit performance are given in the table below. It may seem that this is an excessive effort to require the minimum from the computer. However, these measurements were taken at room temperature. The data sheet for some optocouplers shows that the sinking ability drops drastically at high temperatures. Perhaps this temperature problem is behind reported incidents where optocouplers have been inadequate to sink some radio PTT lines. The *Hapirat* was designed to function properly at up to 50 deg C, including a New Mexico mountain contest site during the hottest days of June and the worst drought in decades; it helped win a top-ten place in the VHF contest.


K5AM Horse Mountain contest site; 7900 ft elev.
http://www.zianet.com/k5am/MiscPhotos/K5AM-Horse-Mountain-contest-site.html

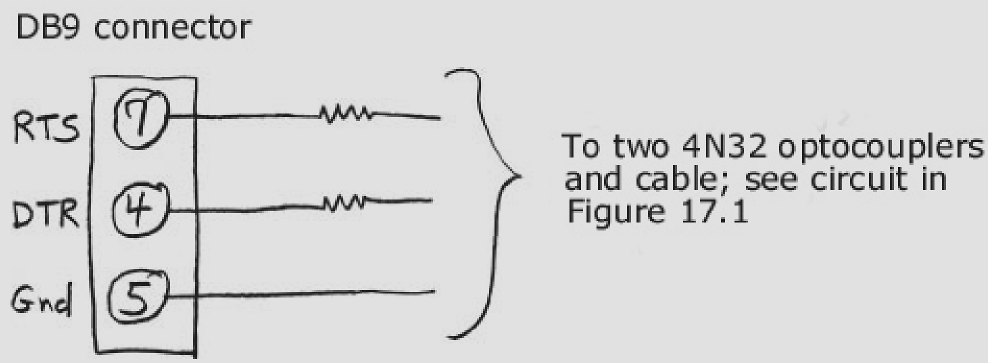
The table below shows the high-level output voltages E and currents I which are required from a computer parallel port at pins 16 or 17 (PTT or KEY), or at corresponding serial port terminals, in order that the optocoupler module will sink certain currents.

Sink current (mA)	E(V)	I(mA)
10	2.2	0.5
1	1.3	0.2

The audio panel **CPTR** control jack requires current sinking of only 0.5 mA. Thus, less than 0.2 mA is required from the computer port; circuits further along in the panel provide up to 40 mA sinking at the PTT or KEY jacks on the transceiver, much more than is required by any modern transceiver.

Serial port module circuit. This module is identical to the parallel port module, except for the connector type and pin-outs, shown in Figure 17.2. Assembly inside the smaller connector is of course much more difficult.

Figure 17.2; Serial port module.

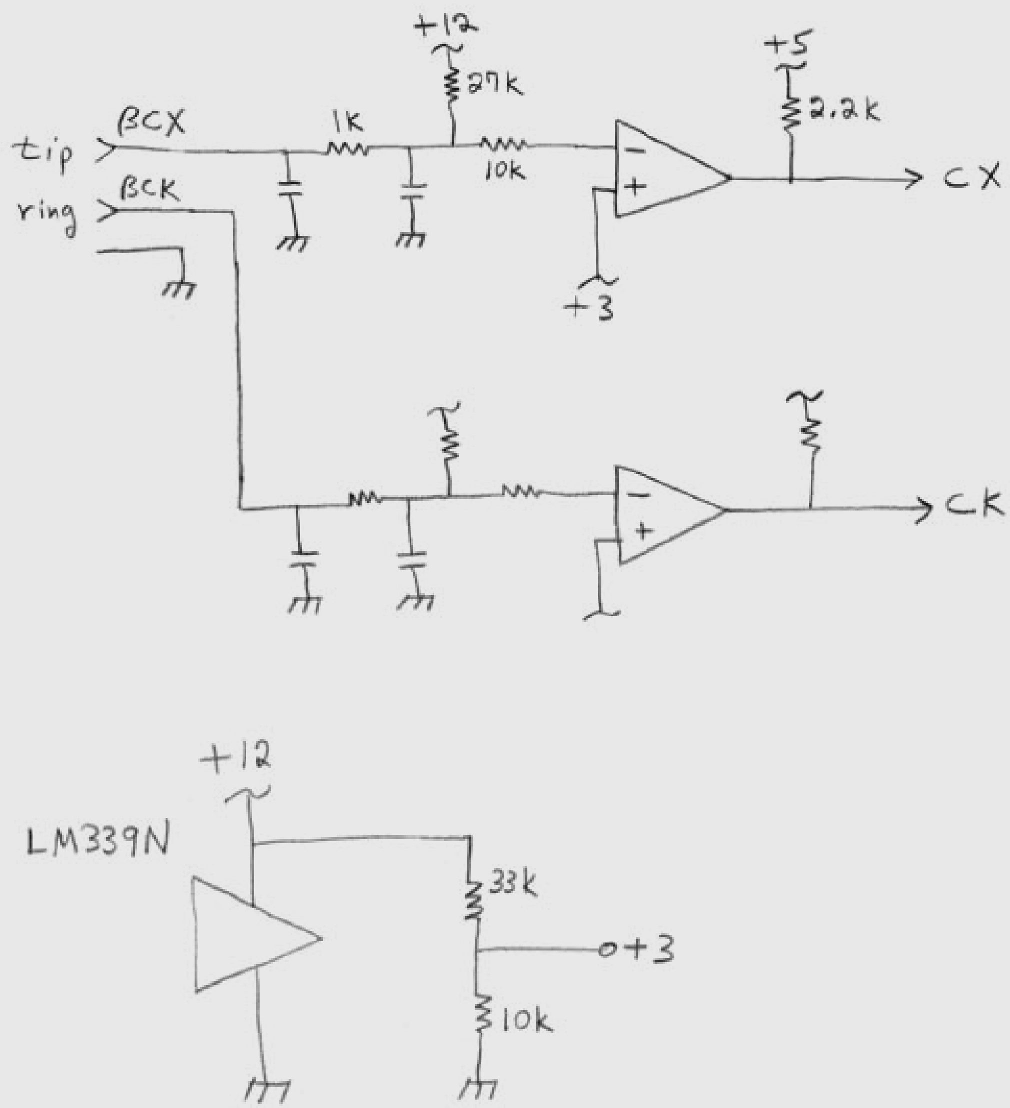


Parts. See the parallel port module parts list above; except:
DB-9 connector, female, solder cups. CS#DE-09S.
DB-9 hood. CS#DEM-09ST.

Panel interface circuit. The *Hapirat* accepts the output from a computer interface module at a 1/4 inch stereo jack labeled **CPTR** on the rear panel.

The comparator circuit inputs operate at 12 Vdc open-circuit, require sinking of only 0.5 mA for PTT and KEY signals, will tolerate a thousand ohms of resistance in the lines, and require sinking only below 2 volts. Any sort of interface circuit, for any sort of computer port, present or future, will function when connected to this jack. After the comparators the circuit uses TTL type logic; then it's easy to wire the logic gates to do anything desired. The comparator circuit is shown in Figure 17.3.

Figure 17.3; PTT and KEY comparators.



Two sections of an LM339N quad comparator are used in this circuit. The two circuits are identical; they convert the βCX and βCK control signals from the interface module into TTL lines CX and CK for the logic section. The capacitors shown are 10 nF. The minimal current sinking requirements of the comparator circuits allow for a great deal of RF filtering at the inputs. An RC filter is much more effective than a simple bypass capacitor.

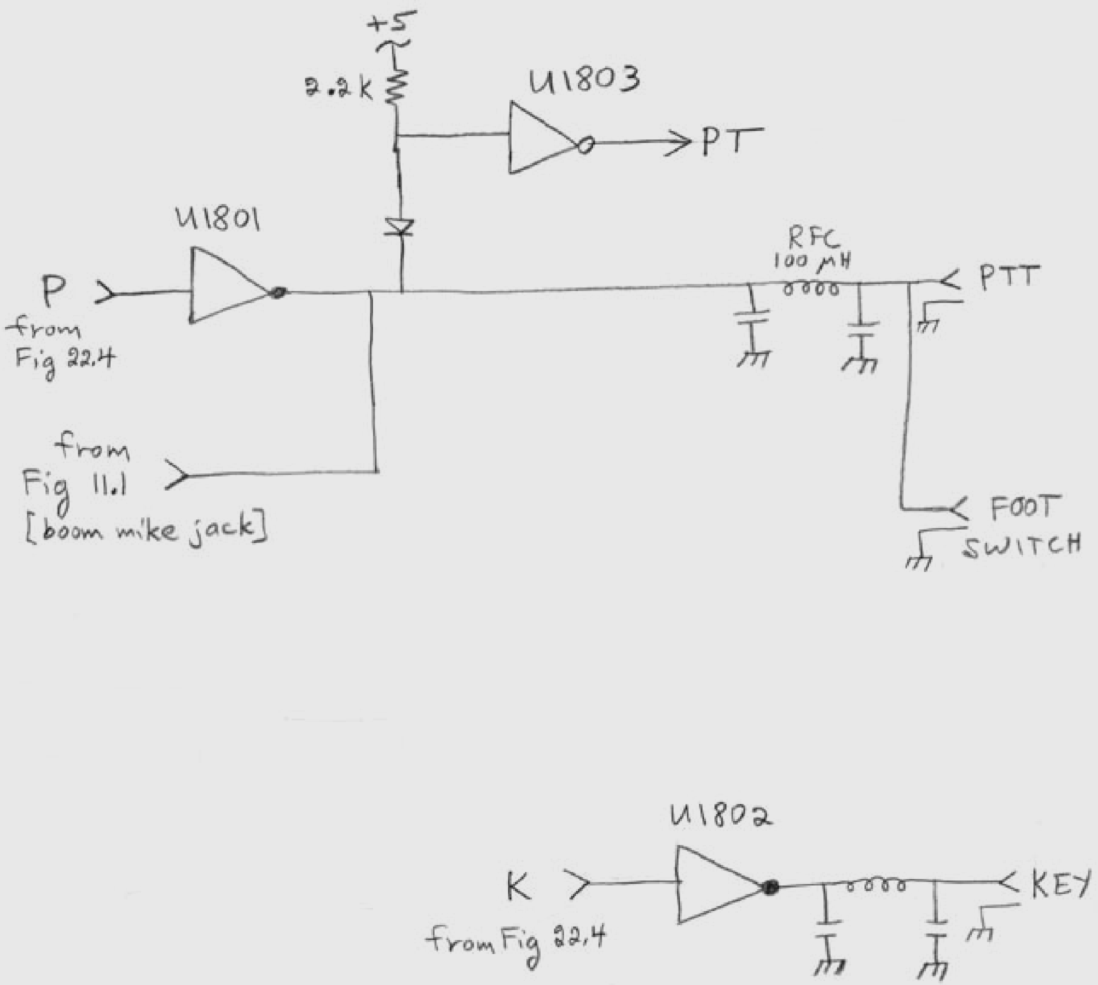
A close study of the data sheets for some types of optocouplers reveals that it would often be difficult to sink even the 1.6 mA required by a TTL gate. Using the more sensitive CMOS logic at this point would put a very sensitive device very close to the "outside world." The 339 is a bipolar device, relatively insensitive to RF, yet capable of sensing very small input changes. The parameters were chosen to provide enough sensitivity so that the optocouplers would trigger the circuits in any situation (including temperature extremes), but not to have more sensitivity than needed.

The open-collector outputs of the comparator are convenient for the level transitions to TTL. Note that line βCX from the interface module goes low for a PTT signal, while the output line CX goes high. We try to ensure that each logic line is high for a "yes" situation, to avoid having to think negatively.

18. Transceiver control interface. Discrete bipolar NPN transistors are available with quite high voltage and current ratings for use in open-collector circuits to control radios. However, modern radios are not very demanding. The transistors used here are contained in TTL packages – type 7406 hex drivers. These open-collector drivers are capable of sinking 40 mA, with open circuit voltages up to 30 Vdc. This is more than adequate for any modern transceiver. If older transmitters with higher demands are to be controlled, they should be fitted with suitable relays, either internal or in external mini-boxes.

Activating the PTT line on transceivers has been a perennial problem, discussed at length on the various e-mail lists. This system ends these problems; it provides both the computer isolation of the wimpy optocouplers and the reliability of the stalwart bipolars.

Figure 18; Transceiver control.



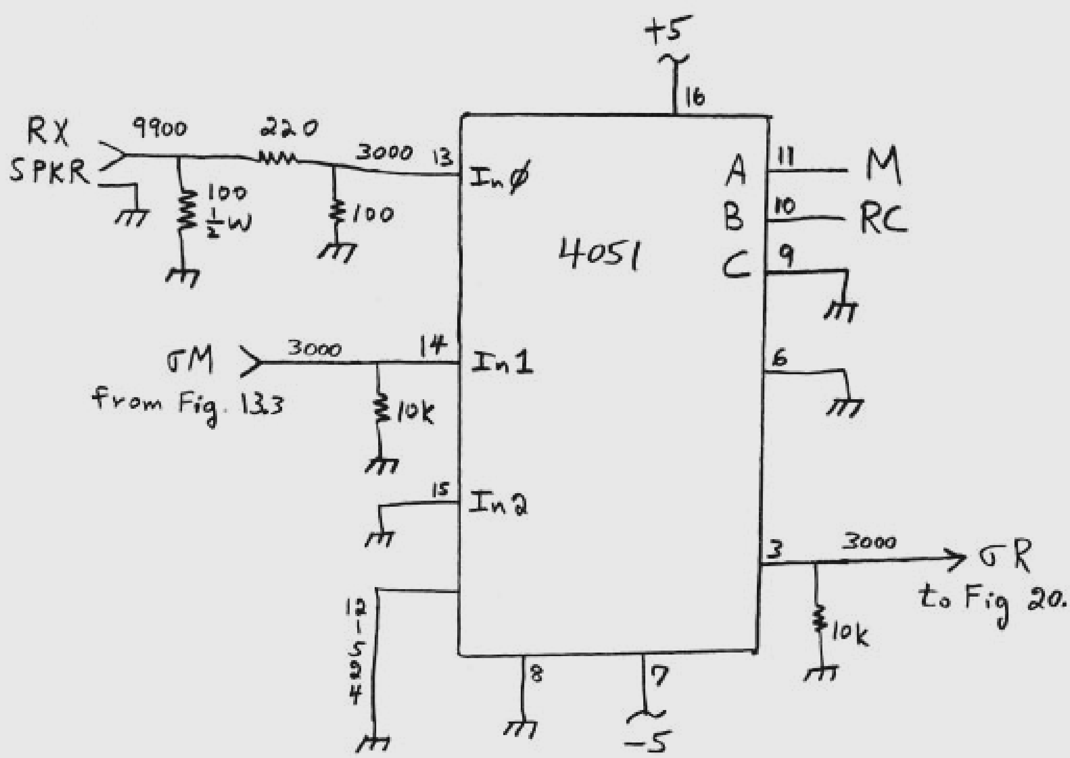
The output circuits for **PTT** and **KEY** are shown in Figure 18. TTL-level control lines P and K are obtained from the logic section, and used to generate outputs at the rear panel jacks **PTT** and **KEY**. U1801 and U1802 are both 7406 open-collector drivers. U1803 is an ordinary 7404 TTL inverter, used to sense external PTT closure from the transceiver or foot switch; line PT is used to switch the meter accordingly.

Spec sheets for the 7406 are available at
www.fairchildsemi.com/pf/DM/DM7406.html
www.fairchildsemi.com/ds/DM/DM7406.pdf
www.mouser.com/catalog/specsheets/7406.pdf

Parts. Surplus dealers often have a good supply at low prices; cf. JA#49091. When regular dealers stock certain types, prices between the different 7400 series can vary greatly. MO#512-DM7406N is very inexpensive.

19. Sound file monitoring. It is usually desired to monitor a WAV file while it is being transmitted. I need this feature to know that I have hit the correct F-button. The circuit is shown in Figure 19.

Figure 19; Sound file monitoring.



In Figure 19, a 4051 switch is used in a SP3T configuration. The output σR leads to the speaker amplifier in section 20. Normal receiver audio from the transceiver speaker output jack is connected to the **RX SPKR** jack on the rear panel. This is attenuated to a maximum level of 3000 mVpp. The normal output at σR is this receiver audio. In choosing resistors for the attenuator, the resistance of the following circuits must be considered. In this case, the net following resistance is 4.2 kΩ. Resistors for the attenuator were chosen of low enough value so that this following resistance could be neglected.

When in mode **CPM**, if the computer sends PTT and WAV file signals, logic line M switches the input to the signal σM; this is WAV audio with level set by the **MON** control on the front panel. WAV audio is also heard in **PLAY** mode. In **REC** mode, logic line RC causes a switch to ground on input 2; this silences the speaker while a WAV file is being recorded.

20. Speaker and headphones. Worse than microphone plug changing, and more frequent, is the nuisance of the headphone plug. EME operators, DX hounds, and many others need their headphones to dig weak signals out of the noise, but they may not want to wear them all day long! So we use the speaker while waiting for the fearless hams in the DXpedition boat to land on the reef. Then, when we hear the signal, we grab the headphone plug and frantically plug it into the jack, often scratching the transceiver panel.

To avoid this unpleasant situation, for over 50 years I have had a switch on the operating bench to select speaker or headphones. A little mini-box fastened to a shelf has a cable from the transceiver speaker output, jacks for speaker and phones, and a switch. The box also has a 100 ohm level control connected to the phones jack to equalize the levels; this eliminates the need to turn down the AF gain on the transceiver whenever the phones are used. This method of headphone connection also solves problems with those transceivers that have insufficient output at the phones jack to drive some headphones.

This system is included in the *Hapirat*, with an additional feature. The boom-mike headset hangs from an arm at the front of the operating bench. The arm is linked to a switch which is connected to the panel; the *Hapirat* automatically switches from speaker to headphones whenever the phones are lifted off the arm.

There is also a panel switch for manual speaker/phones selection. The center position of this switch is for silence - when the landline rings, or the XYL visits the shack for a chat; this reduces wear on the transceiver volume control.

Photo 20A. Left end of the K5AM operating bench, with three microphones. All three mikes are fitted with Heil elements.

When either the desk mike or hand mike is used, and its PTT bar is pressed, that mike is automatically connected, and the others silenced.

When the boom mike headset is lifted off the arm, a switch rigged-up to the arm triggers a circuit in the *Hapirat* which switches the speaker off and the headphones on.

The operating bench has a pine 2x4 front rail. The phones hook-and-switch assembly attaches to the rear of the rail, with the arm extending out below the rail.

Only the left side of the homebrew transceiver (reference 30.1) is in the above photo. The photo does include the front panel jacks for mike and phones. Included only for the sake of conformity, these jacks are never used; dual connections are at the rear panel, leaving the operating bench clear at the front. The *Hapirat* is designed to be placed at the far left end of the bench, keeping the cables out of the way.

The idea for an automatic headphone system is due to Mark Wilson, K1RO. When he saw the system for automatic microphone switching in the manuscript for *QST* in 1995 (reference 30.4), he asked "What about the headphones?"

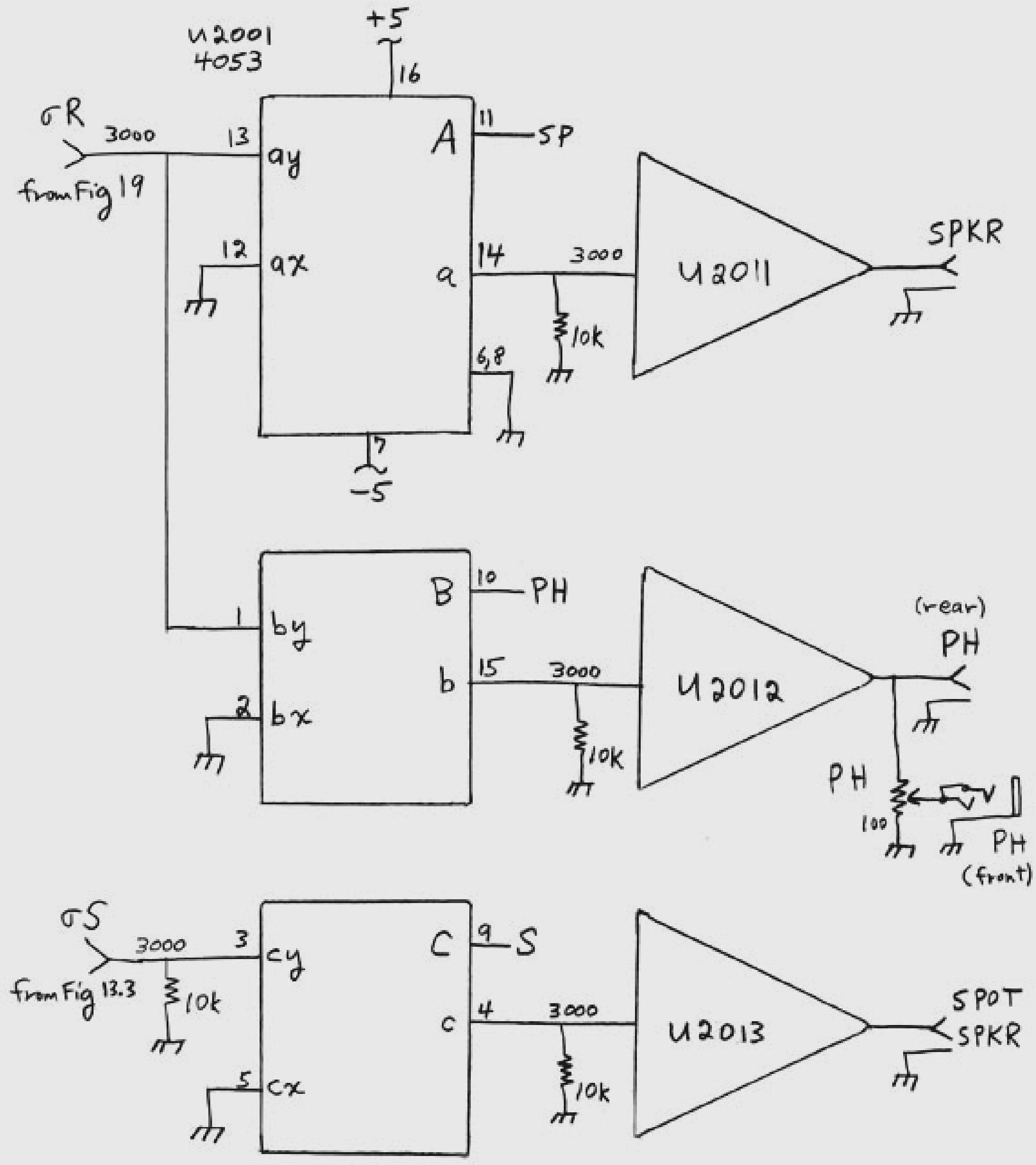
Audio channels. The *Hapirat* has three audio output channels. Two channels drive the receiver speaker and phones; the third drives a separate speaker for packet cluster spots. As an option, a single speaker can be used. Each channel uses a TDA2003V integrated circuit, which has exceptionally low distortion, especially at normal room speaker levels. CMOS switches select receiver audio or WAV monitor audio to the main channel, and control the packet cluster channel.

The audio output level to the receiver speaker is exactly the same as the input from the transceiver; there is no gain or volume control on the *Hapirat* panel. The receiver AF gain control is used as normally. Using a separate TDA2003V for the headphones avoids the relay that would otherwise be used; the cost is about the same.

The packet cluster channel also provides a convenient hi-fi amplifier for playing CDs. The five-dollar 30-year-old 12-inch speaker on my packet cluster channel sounds much better than the 1/2 inch speaker in the laptop.

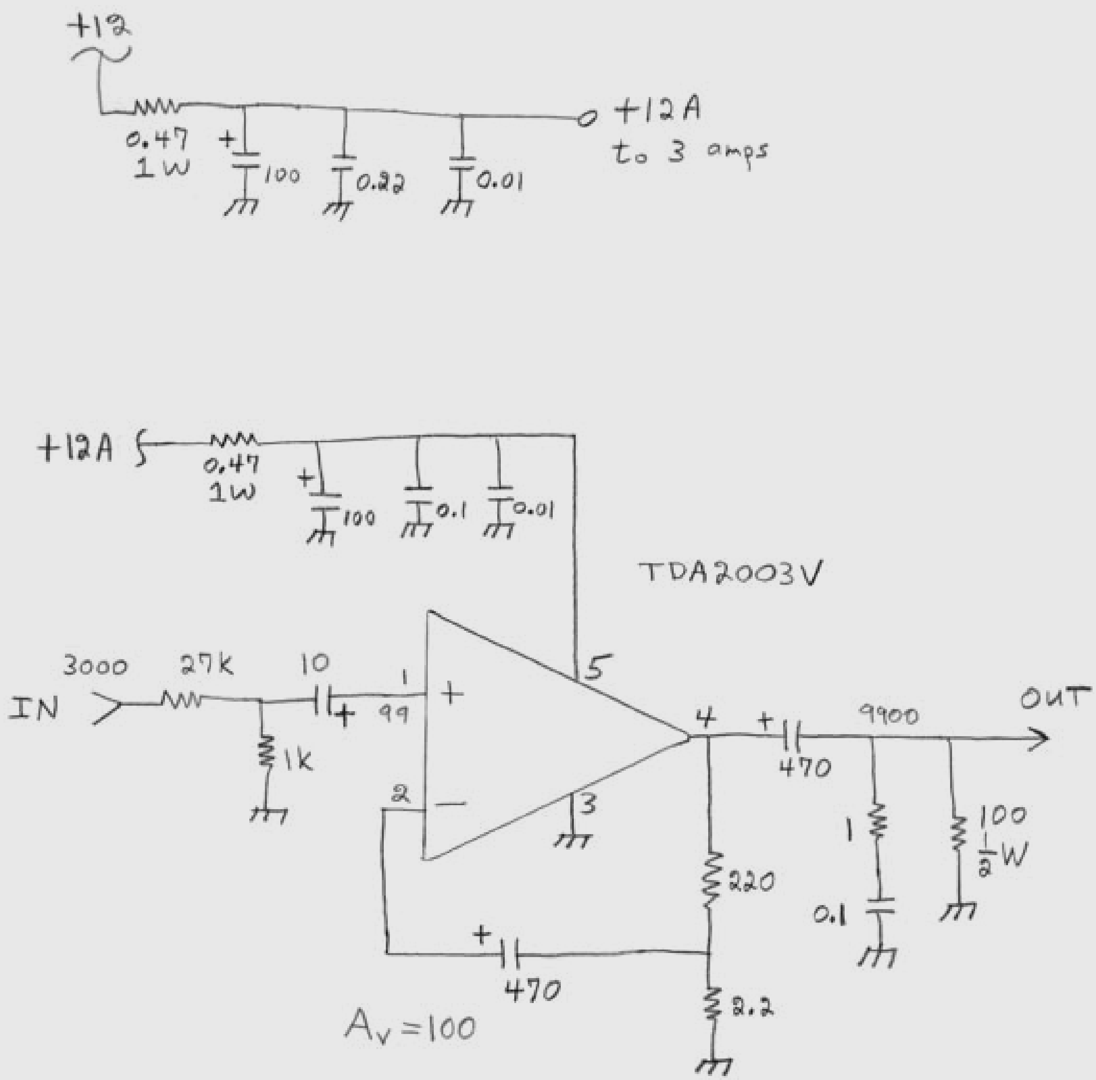
Speaker and headphone circuits. The AF output circuit is shown in Figure 20.1. A triple SPDT CMOS switch is used to control the signals into the three output amplifiers. This switch is used in the same way as described in section 13 above. When an amplifier is not in use, its input is grounded by the switch; this ensures that cross-talk is reduced far below an imperceptible level. For dual reception with a subreceiver and stereo headphones, the dual-channel CD4052 could be used for switching.

Figure 20.1; AF output.



The three identical amplifier modules, U2011, U2012, U2013, are built separately on perf-board and mounted on the main board; the module schematic is shown in Figure 20.2.

Figure 20.2; AF amplifier module.



The 100 ohm load at the output of each amplifier module has two purposes. The first is to keep the output blocking capacitor charged at all times; this prevents a "pop" in the speaker if it is plugged in when the *Hapirat* is powered-up. For this purpose, 1k, or even a larger resistor would be adequate. The second purpose is to improve the low-level IMD performance by providing a minimum load. This method is often suggested, although I do not have actual data for this IC.

The remainder of the circuit is essentially the same as shown on the TDA2003V data sheet. One change is the omission of the high-frequency roll-off components; the data sheet indicates that this is an option. The optional roll-off is not sharp; my homebrew transceiver already has a high-pass filter with cutoff at 3 kHz. Also, the data sheet shows an output blocking capacitor twice the size used here; we are not interested in deep bass music. The amplifier modules have a flat response (−1 dB) from 100 Hz to 20 kHz.

The headphone amplifier provides full output to a rear panel jack, unaffected by the front panel phones level control **PH**. This could be useful for providing receiver audio to a remote location, for connection to phones through an external level control, or for driving a multiple-headphone distribution box.

AF amplifier module parts.

Amplifier IC. MO#511-TDA2003V, 10W, 14V, from STMicroelectronics. The 10 watt listing is misleading. At 12 volts and an 8 ohm load, the output rating is 2 watts. A spec sheet is available at http://us.st.com/stonline/books/pdf/docs/1449.pdf

Capacitors. To save space, it is best to use the latest style aluminum electrolytic capacitors; these are amazingly small compared to the old ones in my junk box, and cost only pennies.

MO#140-XRL16V10	16V 10uF 20%	Xicon	$ 0.04
MO#140-XRL16V100	16V 100uF 20%	Xicon	$ 0.05
MO#140-XRL16V470	16V 470uF 20%	Xicon	$ 0.12

AF amplifier module construction. It was convenient to build the three output stages on three separate small pieces of perf-board. Each sub board is about 1 x 1.5 inches; it would not have been so difficult if more space, about 1.5 x 2 inches had been reserved. Point-to-point perf-board wiring is an easy method – especially when there is ample space.

The three amplifiers are mounted to a small heatsink, 1.5 x 3 x 0.0625 inches. This is sufficient. A test was run with one amp running at full output, steady sine wave; after one hour the sink felt quite warm, but not too hot. A good indication of sufficient sinking is that the output did not drop during this test; this means that the internal temperature limit and cut-back point was not reached. If high output levels are required from all three amps simultaneously, perhaps a sink 0.125 thick, with a few fins, would be better.

Alternative methods. In previous interface projects, I've used relays for speaker audio switching. Relays have generally worked very well, but are not 100% reliable. It takes two relays, for rx/mon and spkr/phones. One amplifier is still required for monitoring WAV files and for packet spots. An LM380 could be used to save a buck and a few components, if audio quality adequate for CD music is not a requirement. Three amplifiers are used in this project to avoid relays and to increase reliability. The results are excellent, and well worth the bit of extra work.

Gain distribution. To ensure that the receiver speaker output level matches the level from the transceiver, the overall voltage gain must be 1. Also, certain portions of the circuit have input level limitations. We'll consider 1.5 watts as a typical maximum AF output from a ham receiver. That means 9.9 Vpp into 8 ohms. To reduce this to a proper level for the CMOS switch in Figure 19, we use an input attenuator with a voltage gain of 0.313, obtaining our standard 3000 mVpp. The TDA2003V amplifier IC, with the manufacturer-recommended feedback shown, has a voltage gain $A_V = 100$. We use an attenuator at the input of the module with a voltage gain of 0.036. These three gain figures result in a net voltage gain close to 1 for the entire panel, from transceiver to speaker.

Amplifier performance. The figures given here for *total harmonic distortion* (THD), are for overall performance of the *Hapirat* circuits, including the op amps, the CMOS switches, and the IC power amplifiers. The analyzer readings also include hum and noise. The measurements for the three output channels were the same. Measurements were made at 1000 Hz, with an 8 ohm load. Data is often given at the 2 watt level, but typical room volume is usually considered to be 50 mW. With an efficient speaker, 10 mW may be enough. Since low levels are used in practice, it is more important to consider distortion at low levels than at the maximum available power. In poor amplifiers, crossover distortion is often present at low output levels.

Power output	**THD (%)**
1.4 W	0.2 clip point
1 W	0.1
50 mW	0.08
10 mW	0.09
10 mW	0.07 headphones*

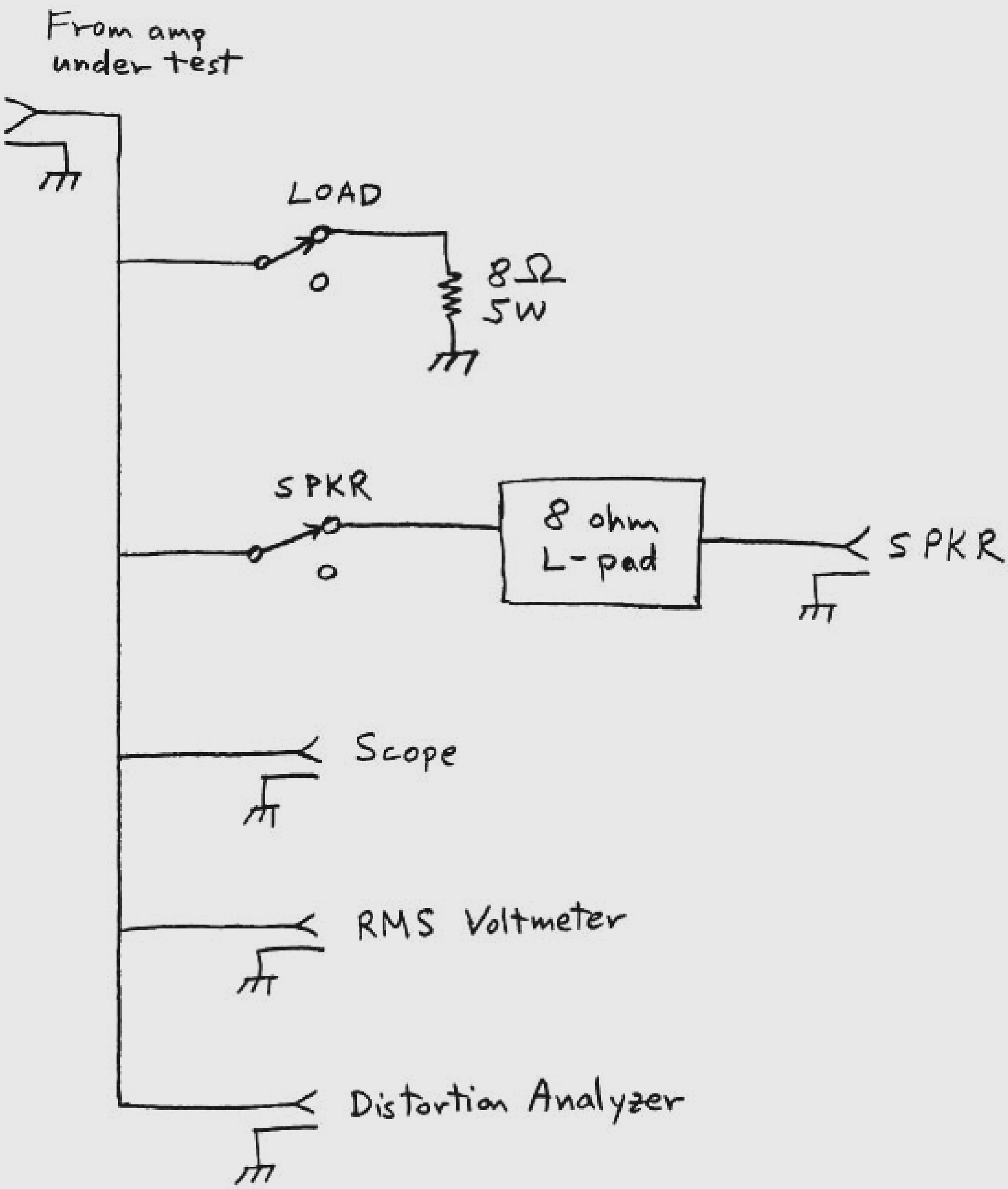
*This measurement is made at the front panel headphone jack. The receiver input level is set for 100 mW output to an 8 ohm load at the rear panel jack; then the front panel headphone level control is adjusted for 10 mW output to an 8 ohm load at the headphone jack. Typically, ham headphones have a higher impedance, so this is a worst-case test.

Slightly more power is available at the usually-mentioned 10% THD point, but this sort of measurement seems irrelevant. Although the TDA2003V is advertised as a 10 watt device, that rating is for a 14.4 volt supply and a 2 ohm load.

Audiophiles will complain that the TDA2003V integrated circuit is a crude device, and that far better systems are available for serious music. Okay. But 0.1% distortion means 60 dB down, undetectable by ear during SSB and CW communications.

Measuring amplifier performance. Each AF output module should be tested after construction, before installation. A schematic for a suitable bench audio test fixture is shown in Figure 20.3. The components are mounted in a small mini-box attached to a shelf above the workbench. The switches allow either the speaker, the test load, or neither to be connected. Test levels can be up to 20 dB above normal room volume – an ear-shattering level – so the speaker switch is necessary. The audio can also be monitored by ear at high output levels, using the L-pad.

Figure 20.3; AF test fixture.



The audio test fixture includes a test point for scope attachment. It is convenient to read the scope in Vpp; the following formula applies, where P_o is the power output into an 8 ohm load, when the scope indicates a sine wave at the level of E_p volts peak-to-peak.

$$P_o = \frac{(E_p)^2}{64}$$

Headphone hook details. Fabricating the phones hook with a switch provides a good opportunity for innovation. Readers might invent a variety of different solutions; an optical switch might be fun to try. This part of the project is not very critical; if an experimental Rube Goldberg type device malfunctions, just unplug it and use the switch on the panel. I took an easy and simple approach; a pine board is often a convenient base for an experiment. Photos 20B and 20C show the hook assembly. Surprisingly, this crude device has turned-out to be robust and reliable.

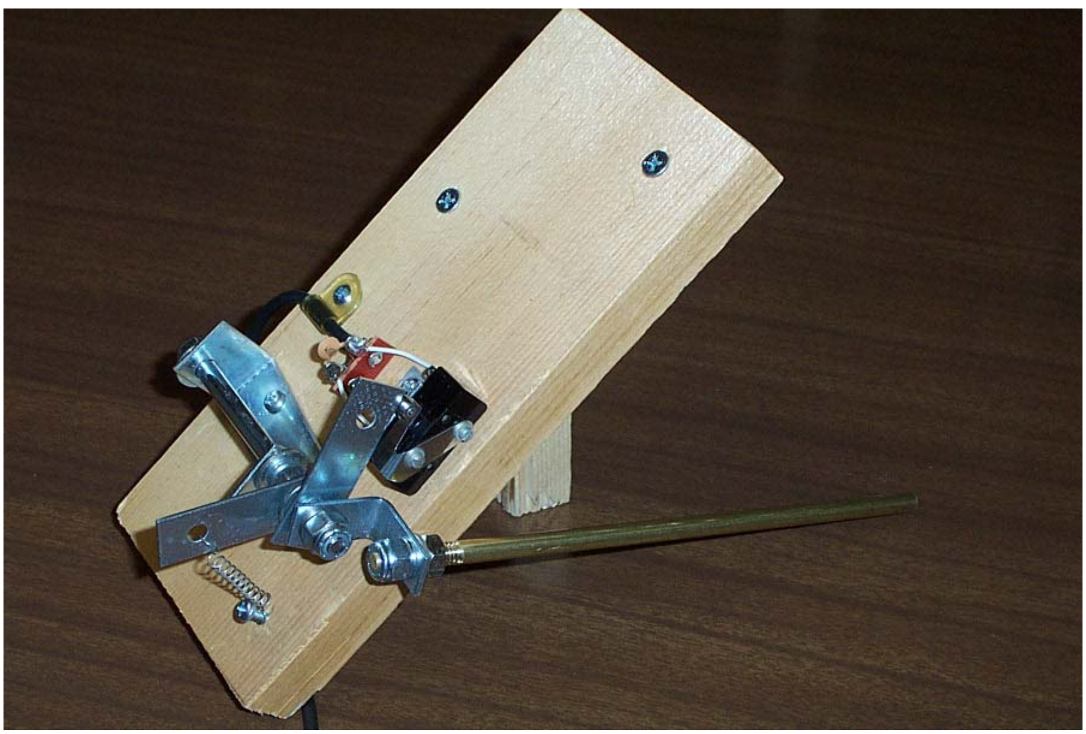


Photo 20B. Headphone hook and switch assembly.

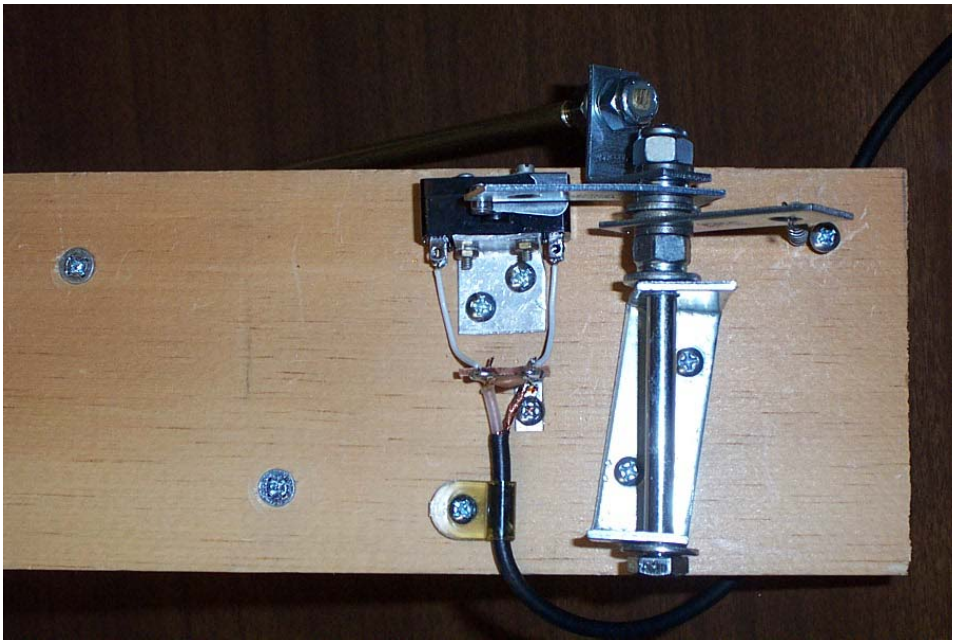


Photo 20C. Headphone hook and switch assembly; top view.

Headphone hook; parts. This simple method requires Grade 8 hardware, lock washers between all pieces, and extremely tight hardware. Locking nuts (with nylon inserts) may help.
Base. Pine board, 1x4, 8 inches long. (0.75 x 3.5 x 8 inches)
Arm. Brass rod, 1/4 inch, 7.5 inches long, thread one end for 1 inch. Cover with black shrink plastic tubing.
Saddle. Steel mending plate, 4 inch, bend into U-shape. A few holes in all the plates will need to be drilled out slightly to 1/4 inch.
Axel. Grade 8 bolt, 1/4 inch, 3.5 inches long, thread an additional length, as required.
Wings. For switch actuator and spring, two steel mending plates, each 2 inches long.
Widget. For attaching arm, steel corner brace, 1 x 1 inch.
Hardware. Grade 8 washers, lock washers, and nuts.
Spring. An inexpensive assortment of springs is useful for various small projects; MSC#79996963 is an assortment of 101 little springs in a fascinating variety of shapes. www.mscdirect.com
Microswitch. Normally closed. From the junkbox. Also found in surplus dealer catalogs; Hosfelt has a nice selection. If only a normally open switch is available, it can be mounted on the other side.
Bracket. For mounting the microswitch; fabricate from 1/16 inch aluminum sheet.
Terminal strip. 2 lug.
Bypass capacitor. 10 nF.
Cable. Shielded cable, 8 foot, phono plug.
Cable clamp. Clamp or tie-down; for cable strain relief.

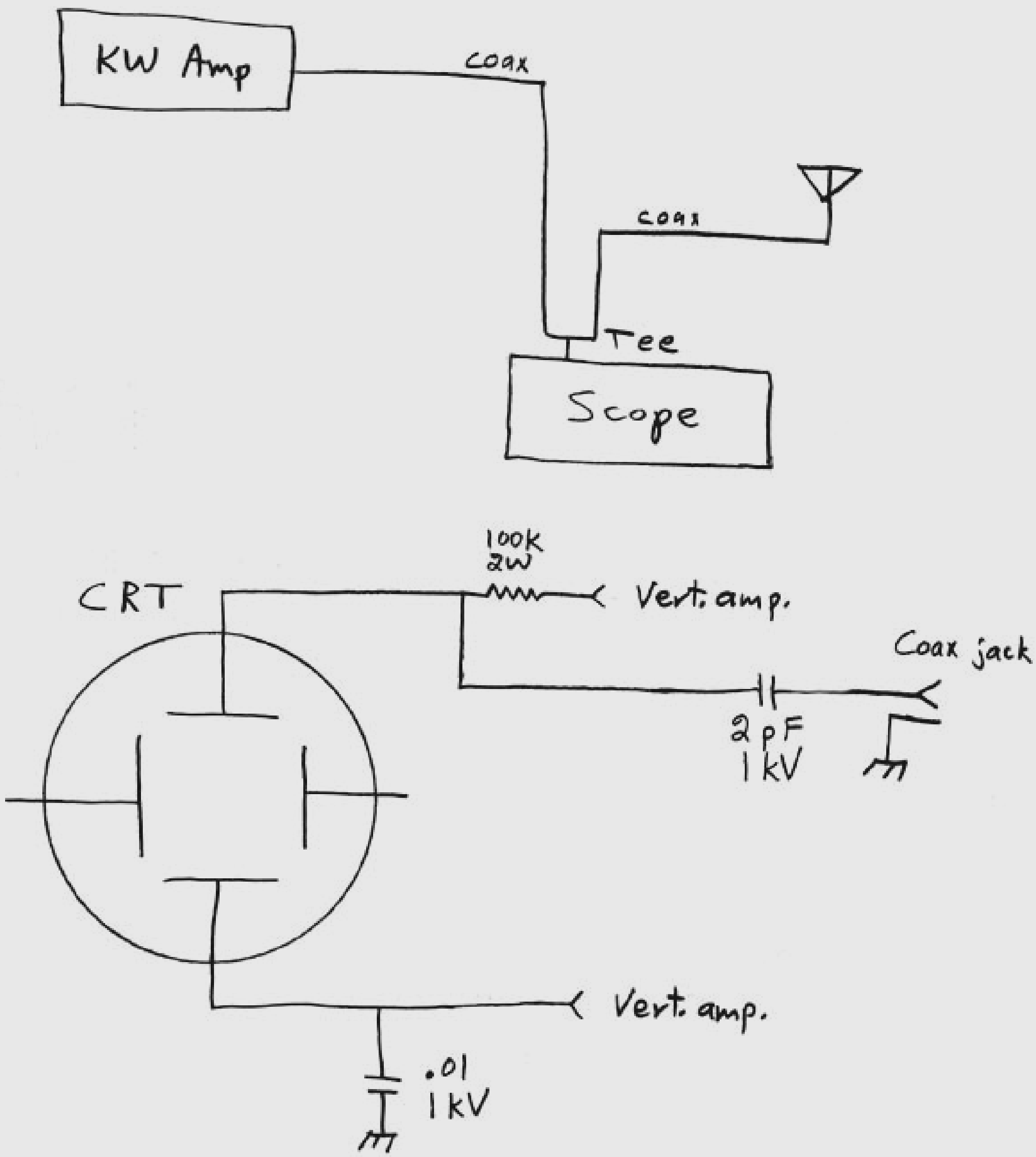
21. Test signals. The *Hapirat* provides audio tones for SSB linearity testing and low-duty-cycle CW pulses for safe amplifier tuning. It also provides a **SYNC** output for stabilizing the two-tone oscilloscope display.

Tone generator. Linearity checking of RF amplifiers to prevent splatter usually requires a two-tone audio generator. A better method to determine the characteristics of an amplifier alone involves two transceivers. The advantage of the two-tone audio, one-transceiver method is that it checks for linearity of the entire system.

The *Hapirat* includes a low-distortion two-tone generator with tones at 1000 Hz and 1500 Hz, in mode **2TONE**. The oscillators have distortion levels of 0.6%; this is far better than needed in this application. As an option, the frequencies are easily changed if desired. To obtain a proper scope pattern, the tones must be balanced at the amplifier. To adjust for slight imbalance which might result from the transceiver filter passband, the balance control **BAL** is on the front panel, along with the level control **2T**. A single tone, 1000 Hz, is available in mode **1TONE**.

Tone generator operation. Any scope can be used. The old Heath HO-10, available at flea-markets, is very popular. An excellent scope for linearity testing is the HP 120B, an old 400 kHz scope with a large 5 inch CRT, often available for only $5.00 at ham flea-markets. (Pick up several if you can, for spare parts; some of the 26 tubes inside each cost more, if you can find them, than the whole scope.) The frequency limit is no problem, because we don't use the vertical amplifier – we apply a sample of the amplifier RF directly to the vertical plates inside the CRT.

Figure 21.1; RF connection to scope.



A method for applying a sample of RF to the scope is shown in Figure 21.1. First add a coax jack to the rear panel of the scope, as close as possible to the CRT socket. Now break the lead to one of the vertical plates near the socket and insert the resistor as shown. This keeps the vertical amplifier connected well enough to provide centering adjustments, while keeping the RF out of the amplifier. Connect the two capacitors as shown. Outside the scope, attach a coax tee adaptor to the jack. A coax jumper connects the amplifier to the tee; the other side of the tee connects to the antenna. This setup can be left in the line at all times; the scope drains only negligible power from the coax line. If the pattern is too large or too small, change the size of the capacitor at the jack. The best method is to include a switch to select different size capacitors, or an air variable capacitor of suitable rating.

The peak output when two tones are used is twice that of one tone. The circuit does not compensate for this; when one tone is used the operator must adjust the level using the **2T** control and the meter.

SSB linearity testing. Connect a shielded lead from the *Hapirat* **SYNC** output jack to the scope trigger input. Set the horizontal sweep speed to 0.5 ms/cm.

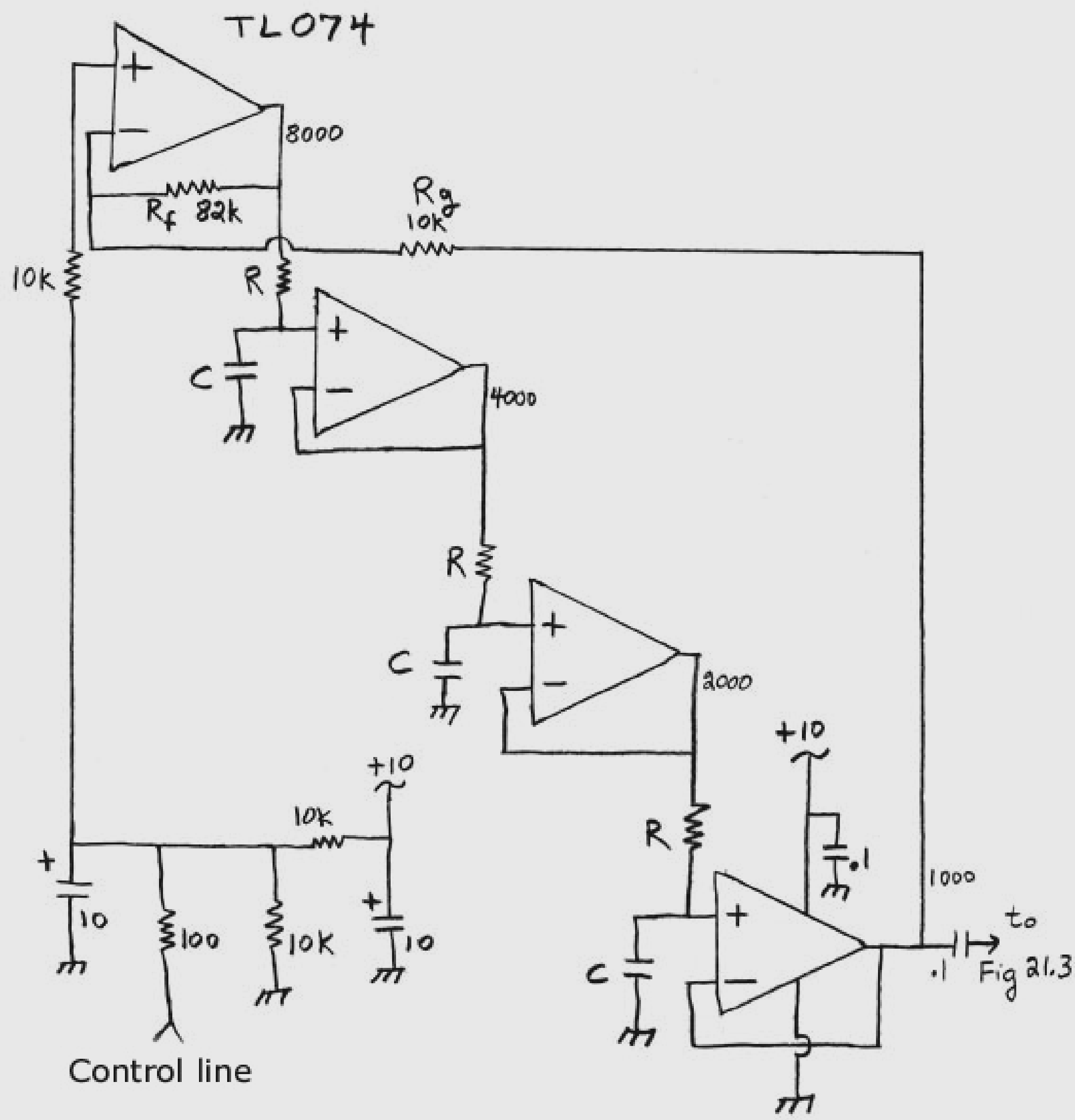
CW keying waveform testing. Connect a shielded lead from the transceiver key jack (in parallel with other connections), to the scope trigger input. Set the horizontal sweep speed to 5 ms/cm. It is convenient to build a keyline junction box using an eight phono jack plate, with all jacks tied together, RS#274-370. Leads can be run to the transceiver key jack, the *Hapirat*, the scope, a straight key, a bug, a separate contest keyer, etc. This is neater than an unruly bunch of Y-adapters.

Observing the keying waveform on the scope can be helpful in eliminating key clicks caused by poorly designed transceivers. When QSK is used, observing at the amplifier output is also necessary, because key clicks are often caused by poorly designed amplifier QSK circuits. There has been a voluminous discussion lately about key clicks caused by top-of-the-line transceivers. Correction of these faults is sometimes difficult. The homebrew transceiver has separate internal trimpot adjustments for rise and fall times; no CW waveform distortion appears during QSK operation, even on the first dit; breaking stations can be heard while sending at 50 wpm.

Tone generator circuit. To obtain very low-distortion sine waves, op amp phase-shift oscillators are used. A single op amp could be used for each oscillator, but lower distortion is obtained by buffering each RC section to reduce loading. This was rarely done in vacuum-tube days, because of the cost of the components. Today, with inexpensive quad FET op amps available, the single op amp phase-shift oscillator is obsolete.

Ref: Texas Instruments; Application Report SLOA060, March 2001. Sine-Wave Oscillator, by Ron Mancini and Richard Palmer. www-s.ti.com/sc/psheets/sloa060/sloa060.pdf

Figure 21.2; Tone oscillator.



The phase-shift oscillator circuit is shown in Figure 21.2; two of these circuits are used. The frequency of oscillation is given by

$$f = \frac{0.27566}{RC}$$

High-quality capacitors are required. Specifying a fixed common value for capacitors in all audio filters and oscillators in the station makes stocking parts easy; these capacitors are always on the shelf.

C. Polypropylene capacitor, 10 nF, 2% tolerance. Panasonic #ECQ-P1H103GZ, DK#P3103.

With this fixed value for C, the frequency of oscillation is determined solely by the value of R.

$$R = \frac{27.6}{f}$$

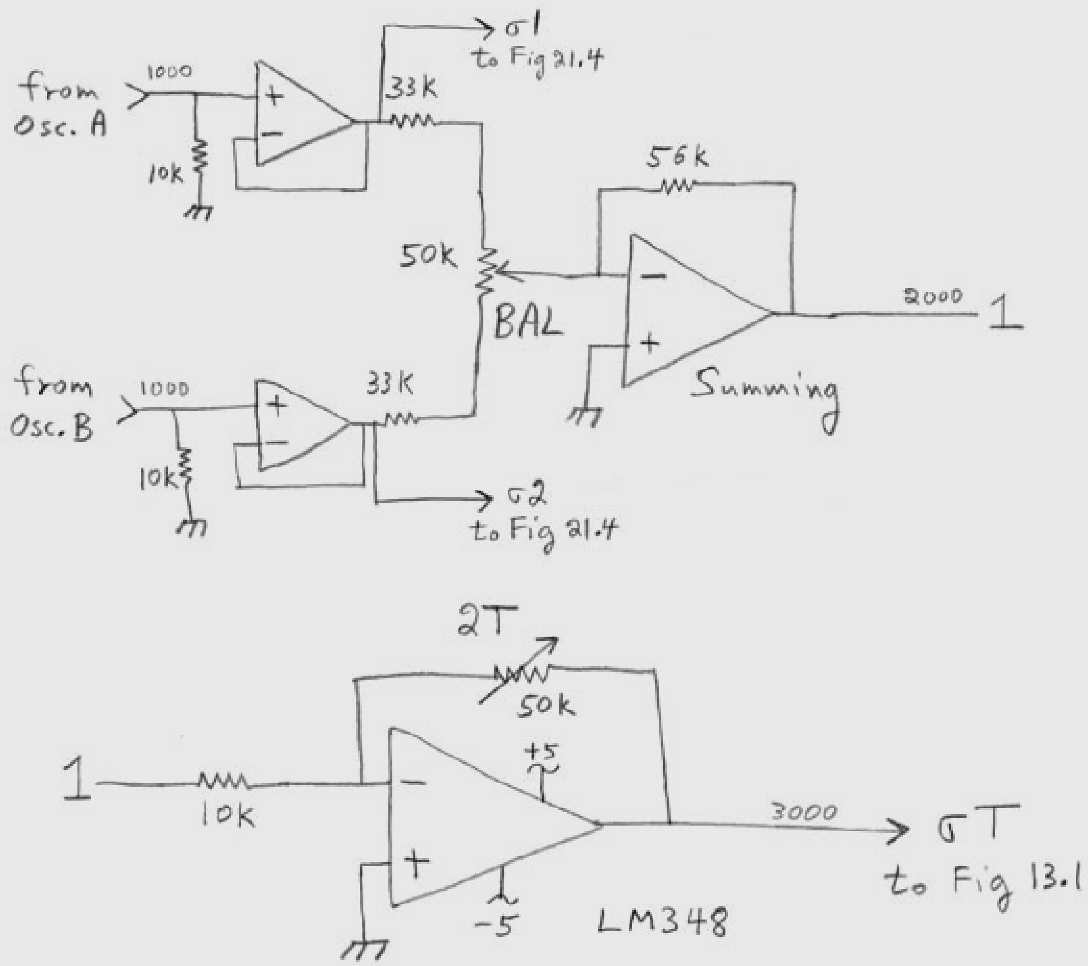
where:
f is the desired frequency in kHz
R is the required resistance in kΩ

The components for the two oscillator circuits are:

Osc.	**Freq.**	**C**	**R**	**Control line**
A	1000	10 nF	27k	δT
B	1500	10 nF	18k	δ2T

The two oscillator outputs are combined in the summing amplifier in Figure 21.3, where the level and balance controls are introduced. Each oscillator output is buffered by a voltage follower ahead of the summing amplifier.

Figure 21.3; Tone buffers.



The non-inverting summing amplifier configuration produces a virtual ground at the inverting input. With the balance control centered, the feedback resistance is equal to the input resistance from each channel, so the gain is 1 with respect to each input. Control **BAL** allows a maximum channel difference of 8 dB, more than adequate for any properly functioning transceiver.

The last section of the quad op amp provides level control with panel knob **2T**. At the nominal setting of 30%, or 10 o'clock, the stage provides a gain of 1.5, raising the combined two-oscillator output to the standard 3000 mVpp.

Note on resistor selection. The values of R affect the frequencies obtained, but exact frequencies are not at all important. The resistor *matching*, however, does affect the distortion level. One method for obtaining a good match is to buy 1% precision resistors with values of 27.4 kΩ and 18.2 kΩ. Another method is to select matching resistors from a batch of ordinary 5% resistors. This is usually quite easy, since resistors in the same batch of 100 or 200 are often quite close in value, although other batches may be different.

Note on precise tone frequencies. For the simple purpose of linear amplifier testing, exact frequencies are not required; this note on frequencies is merely for reference if a precise oscillator is needed in a different application.

We have the following measurements:

Oscillator	**f-nom**	**R-rqd**	**R-inst**	**f-calc**	**f-meas**
A	1000	27.6 k	26.4 k	1044	1030
B	1500	18.4 k	18.2 k	1515	1499

Symbol	**Definition**
f-nom	nominal frequency
R-rqd	required resistor value from the formula
R-inst	installed resistor value – measured
f-calc	calculated frequency predicted by the formula, using the installed resistor
f-meas	measured frequency

The closeness of f-meas to f-calc is the result of using three buffers, which prevent loading of the RC networks.

To obtain more precisely a desired frequency, there is an easy way to simulate a 1% resistor using two 5% resistors. Using a DMM, connect a resistor of the next higher standard value from the desired value. Then add in parallel a value about 10 times higher, trying several values until the result is within 1% of the desired value. Not only cheaper than a precision resistor, but you get it *now*! However, the capacitor has only 2% tolerance, so extra effort with the resistors may not pay-off. Selecting 2 or 3 ordinary resistors in parallel to adjust the frequency is an easy method. A trimpot is not the best way, because trimpots have more drift due to aging and temperature.

Note on distortion. For the simple purpose of linear amplifier testing, perfect sine waves are not required; this note on distortion is merely for reference in case it is needed for a different application.

The circuit consists of an op amp oscillator and three buffers. The gain is set by Rg and Rf; we will leave Rg fixed at 10k and use Rf to experiment with the gain setting. Each RC network results in a voltage loss; the output of each network is 1/2 the input. The three networks in tandem result in only 1/8 of the signal being returned to the feedback circuit. Thus a minimum gain of 8 is required for oscillation. Tests confirm this. This oscillator uses 82k at Rf, for a gain of 8.2. Reducing this to 78k stops oscillations. The output from the oscillator section is about 8 Vpp; at the last follower the output is 1 Vpp.

The raw oscillator output, before the first RC network, shows 2.2% distortion, while the circuit output, after the third buffer, shows 0.6%. This demonstrates the effectiveness of the buffers, which filter the raw oscillator output.

The distortion can be reduced further. While the circuit output looks perfect on the scope, the raw oscillator output before the buffers clearly shows clipping, about 10% of the peak waveform, but only at the low part of the wave. I could not eliminate this clipping by reducing the gain, which was already at the limit required for oscillation.

However, the exact bias setting at the non-inverting input of the oscillator was found to have a great effect on the clipping at the top and bottom of the waveform. A very slight adjustment balanced the clipping and produced a significant improvement in distortion performance. The distortion dropped from 0.6% to 0.2% when the bias was changed from (0.50) x V to (0.53) x V, where V is the circuit supply voltage, nominally 10 volts. In the event, the bias was left untouched at (0.50) x V, since anything below 1% distortion is far better than required for this application.

Two-tone scope synchronization. Watching a monitor scope when two tones are applied to the transceiver always presents a minor, but irritating, problem. The waveform seen on the scope is an envelope pattern whose frequency is the difference of the two tones, here it is 500 Hz. The envelope pattern is filled in by the much higher frequency RF waves, which appear only as solid areas.

The problem is synchronizing the scope; a precise and stable horizontal sweep speed of 200 μs/cm is required to display one complete cycle of the envelope pattern in a full trace of 2 ms. This can be quite difficult to do on a simple scope with a small sweep frequency knob. And finally, after tedious adjustment, the display usually begins to drift immediately, since the phase relationship between the two tones and the scope horizontal oscillator is always changing. It helps if the scope has a selectable low-pass filter in the trigger circuit, but for this purpose we are usually not using the full-featured shop scope, but an old, simple scope from a flea market.

To solve this triggering problem, the *Hapirat* provides a **SYNC** output which is connected to the scope external trigger input. The sync signal is a square wave at 500 Hz; it is obtained by demodulating the combined 1000 and 1500 Hz audio signal to recover the 500 Hz beat note.

A Motorola IC is used to recover the 500 Hz heterodyne; the data sheet for the IC calls it a "modulator/ demodulator". We all mix RF signals in our transceivers, and might call this a mixer, but an "audio mixer" is usually something else, so this would be misleading. (Audio mixing in the usual sense does occur when the two oscillator outputs are combined in the summing op amp.)

The idea is this: when we put the two tones into the transceiver, it is as if one is a carrier and the other is the modulation. Listen to it in an AM receiver; you'll hear the 500 Hz tone. It's something like an AM signal, but with only one sideband; we might say that the 1500 Hz tone puts 500 Hz modulation on the carrier produced by the 1000 Hz tone.

Demodulator circuit. The sync demodulator circuit is shown in Figures 21.4 and 21.5.

Figure 21.4; Demodulator.

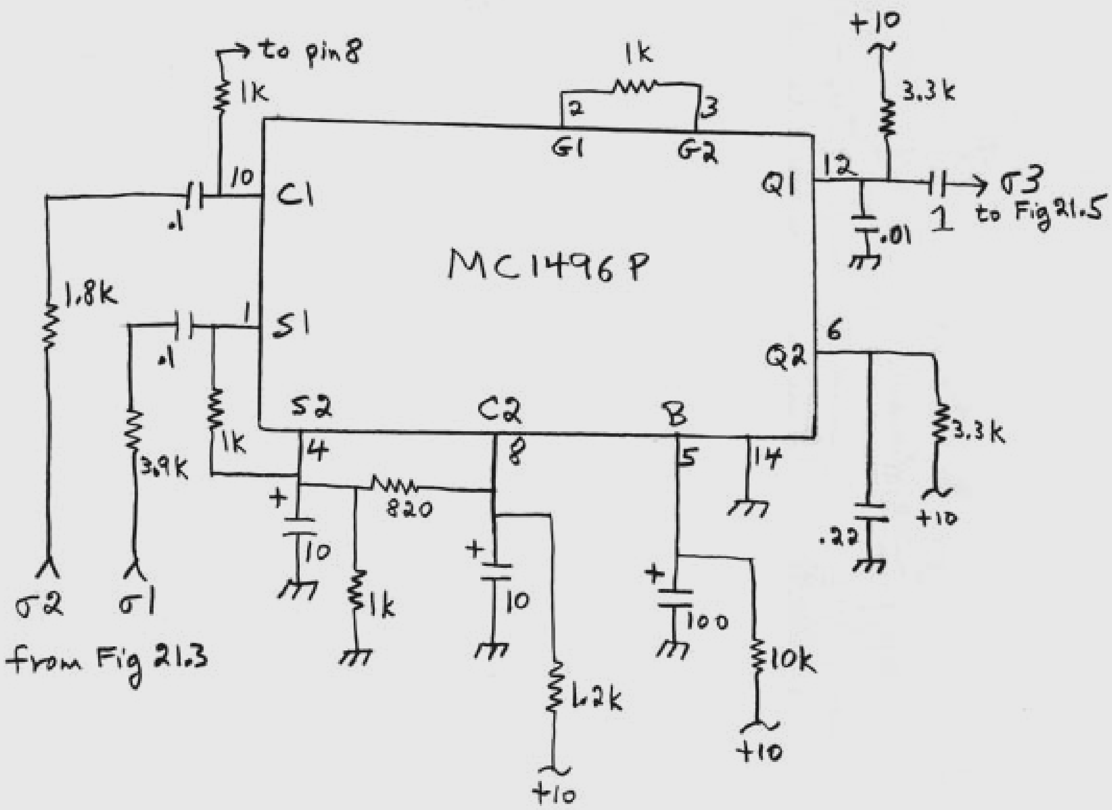
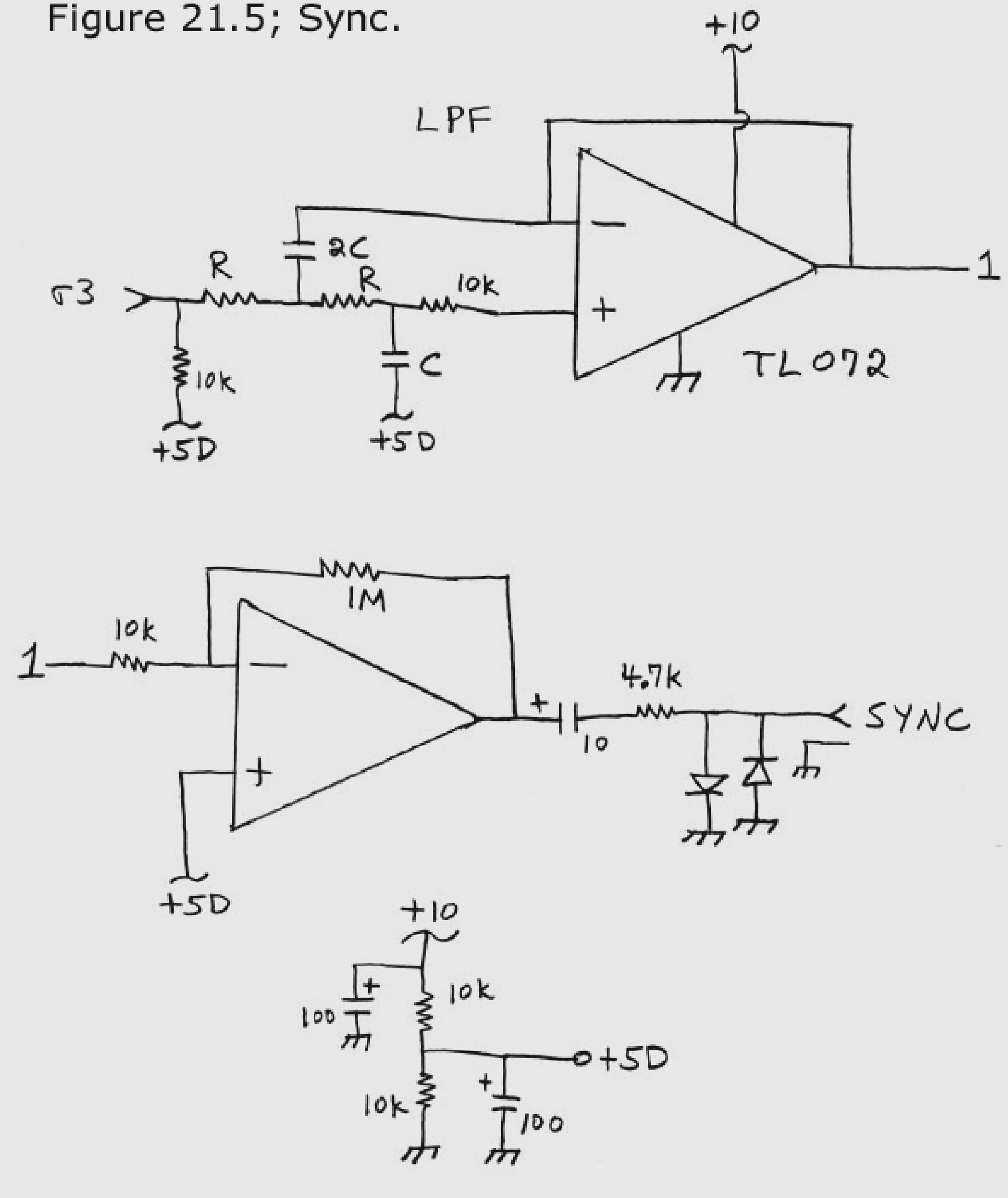


Figure 21.5; Sync.



After the MC1496P demodulator, a simple op amp low-pass filter in Figure 21.5 is used as an image-stripping filter to remove the 2500 Hz component. Then a rather high-gain amplifier raises the level; this amplifier saturates somewhat on peaks, and helps to square the waveform. Finally, a pair of diodes squares the sync signal and reduces the output to a fixed level of 1.2 Vpp, appropriate for the external trigger input of a scope.

LPF circuit. The filter stage is a unity-gain maximally-flat Butterworth filter in a voltage follower configuration. The value of Q required is $\sqrt{2}/2$.

This configuration is the simplest to apply in regard to component selection. It also involves the simplest formulas. One first chooses the cut-off frequency f (in Hz) and the capacity C (in farads). Then the resistance R (in ohms) is given by

$$R = \frac{1}{2\sqrt{2}\ \pi\ f\ C}$$

Fixing the convenient value of 0.01 μF for C, we have

$$R = \frac{11.25 \times 10^{6}}{f}$$

We want to keep the 500 Hz beat note and attenuate spurs at 1000, 1500, and 2500 Hz. Choosing a cut-off frequency of 625 Hz, we use R = 18k. The resistors should be matched as well as possible, but the exact value is not critical.

Parts.
MC1496P. Mouser has a replacement IC from NTE, but as usual for replacements, it is expensive. JA#23211 is inexpensive; it might be marked LM1496N. Data sheet:
http://cache.national.com/ds/LM/LM1496.pdf

C. Polypropylene capacitor, 10 nF, 2% tolerance. Panasonic #ECQ-P1H103GZ, DK#P3103. For 2C, use two in parallel; this results in a precise ratio of 2 between the capacitor values, better than might be found otherwise.

The **SYNC** output has very steep rise and fall characteristics, for excellent triggering in the scope. After decades of futile efforts to stabilize two-tone test patterns, the effect is quite astounding.

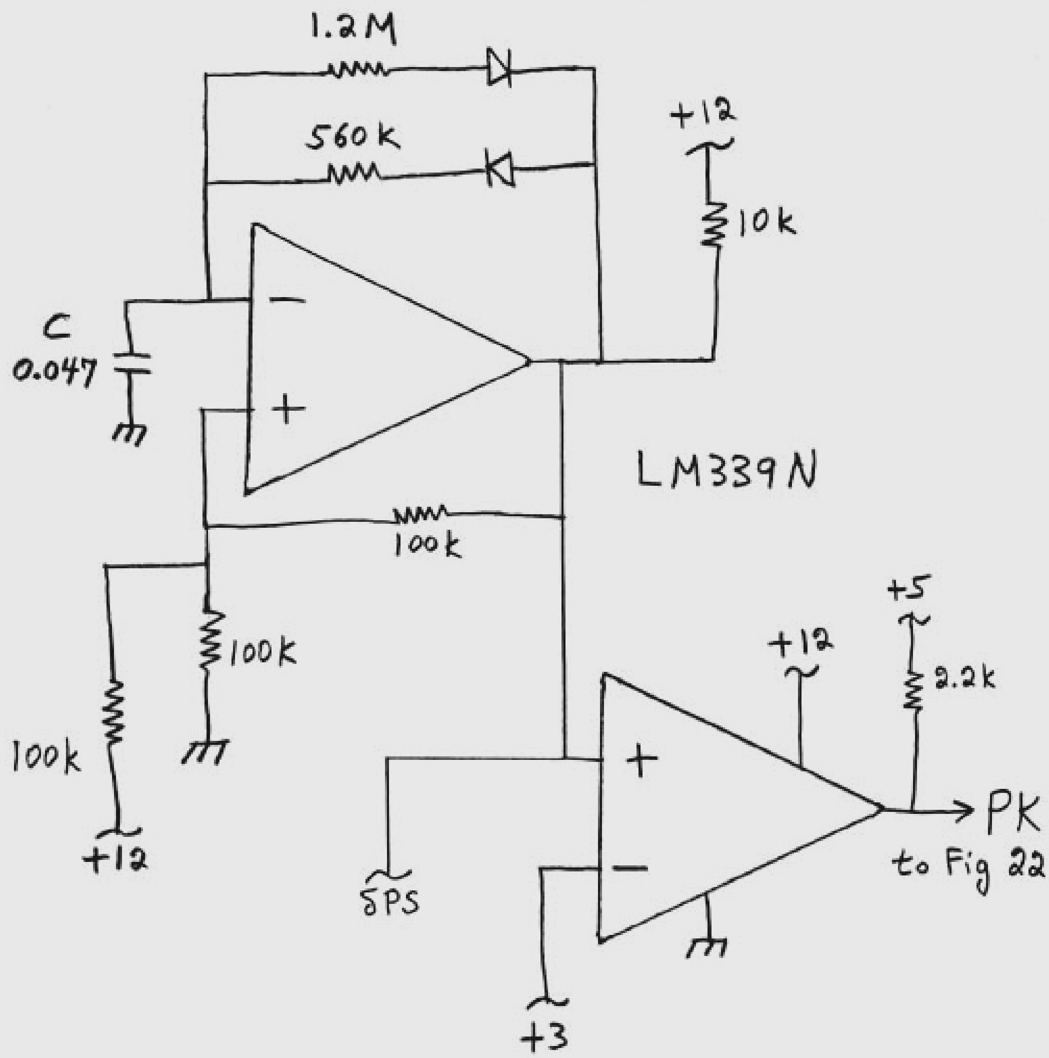
Pulse generator. To prevent premature failure of an RF amplifier or HV power supply, it is best to tune-up with CW dits at about 40 wpm. A peak-indicating output meter makes this easy. PA tube failures may be caused more often by brick-on-the-key tuning than by normal operation.

Even better than using the 50% duty-cycle CW dits is a pulse generator set for a 33% duty cycle. My homebrew transceiver has a built-in pulse generator. The TUNE switch on the transceiver panel automatically shifts the radio into CW, silences the sidetone, hits the PTT line, and starts the pulse generator. Testing at 1500 W PEP output involves only 330 W of average plate dissipation.

Tuning up with the pulse generator is easiest with a peak-indicating power meter; most of my homebrew amplifiers have built-in peak-indicating meters. I also use an Autek model WM1, which works very well at the pulse rate of this circuit. www.autekresearch.com/wm1.htm

Pulse generator circuit. The *Hapirat* pulse generator schematic is shown in Figure 21.6. The circuit could also be build into a homebrew transceiver or as a separate device.

Figure 21.6; Pulse generator.



C. Polypropylene capacitor, 47 nF, 2% tolerance. Panasonic #ECQ-P1H473GZ, DK#P3473.

Data sheet: http://cache.national.com/ds/LM/LM139.pdf
Application Note: www.national.com/an/AN/AN-74.pdf

22. Logic. It took only a few moments to choose logic methods for switching between the various modes. One has only to imagine the complexity of a 13-wafer, 12-position mode switch, the possibility of hum pickup or noise, and the trauma that would result from having to replace the switch someday. It is easy to switch modes with a logic circuit; one 12-position wafer is sufficient. TTL was chosen over other logic types because it is less susceptible to malfunctioning from any possible RF that might leak into the circuit. The original TTL family, 7400, is nearly obsolete. We use the current 74F family, which is compatible, but continue to use the old term TTL, and the plain familiar 7400 labels.

A PIC controller could perhaps be used, but this would require an involvement with programs and expensive devices. Also, it would make changes and modifications difficult.

Boolean functions determine the characteristics of each control line output from the logic section. Aside from a few inverters, only non-inverting gates, AND and OR, are used to generate the functions. These gates work well with the particular functions required, and this allows straightforward design methods - no negative thinking is required.

This table lists the logic lines used in the *Hapirat*:

Symbol	**Description**
2T	Tone 1500 Hz
C	Computer control allowed
CK	Computer KEY command
CM	Computer control with monitoring
CX	Computer PTT command
D	Desk mike
H	Hand mike
HK	Phones off hook
K	Key command
M	Monitor
MD	Desk mike mode
MH	Hand mike mode
MP	Phones mode
MS	Speaker mode
P	PTT order
PH	Phones
PK	Pulse generator key out
PL	Play mode
PM	PTT from desk or hand mike
PS	Pulse generator mode
PT	PTT order for meter
RC	Record mode
S	Spot mode
SP	Speaker
SU	Setup mode
T	Tone 1000 Hz
V	Switch meter to transmit
W	WAV audio
X	Transmit; computer command

Most of the inputs to the logic section are derived from the **MODE** switch, but not directly. The arm of the switch is grounded, so each terminal goes low when its mode is selected; 11 inverters are used to obtain the positive-thinking inputs.

Using TTL. Information on logic circuits and Boolean algebra can be found in several places. References:
1.ARRL Handbook, Newington, CT, 2000; Chapter 7.

2. *TI Logic Data Book:*
http://focus.ti.com/lit/misc/scyb004/scyb004.pdf

However, no extended study program is required. It is not necessary to understand what is inside the logic chips in order to use them, just as we need not explore the innards of a Pentium chip in order to send e-mail. Only the *function* of each type of chip need be known. The functions and pin-outs for the TTL gates used here are given on the data sheets:
www.fairchildsemi.com/ds/74/74F04.pdf
www.fairchildsemi.com/ds/DM/DM7406.pdf
www.fairchildsemi.com/ds/74/74F08.pdf
www.fairchildsemi.com/ds/74/74F32.pdf

The logic circuits may be thought of in terms of inputs and outputs. The *inputs* are the settings of the panel switches, the computer outputs, and the external PPT line. The *outputs* are the control lines leading to all the circuits. The inputs may be thought of as independent variables, the outputs as dependent variables. This enables us to think of the logic circuits as functions, in this case Boolean functions. These functions are logical combinations of the inputs; Boolean algebra is used to calculate the outputs, just as ordinary algebra is used to calculate the voltage output of a radio circuit. The symbols used for the logical operations are:

Symbol	**Meaning**
+	OR
•	AND
–	NOT

Example. When the computer wants the radio to transmit, logic line CX goes high. But we don't want the rig to obey the computer all the time, only in modes **CP** and **CPM**. In these modes, the **MODE** switch drives logic line C high. The Boolean function

$$X = C \bullet CX$$

means "C *and* CX"; this will result in PTT line closure. The circuit is in Figure 22.4; the AND function is carried out by a 7408 gate. The main logic line for PTT is P. We also want P high whenever one of the mike buttons is pressed; when this happens, line PM goes high. The Boolean function

$$P = X + PM$$

means "X *or* PM"; this will provide PTT line closure exactly when required. The OR function is carried out by a 7432 gate.

The Boolean functions generated in the logic section are:

$$\begin{aligned}
D &= (-\alpha D) + MD \\
H &= (-\alpha H) + MH \\
K &= (C \bullet CK) + PK \\
M &= (CM \bullet CX) + PL \\
P &= X + PM \\
&= (C \bullet CX) + [(-\alpha D) + (-\alpha H)] \\
PM &= (-\alpha D) + (-\alpha H) \\
V &= PT + T + PL + SU \\
X &= C \bullet CX
\end{aligned}$$

In addition, two logic operations, the implications

$$CM \Rightarrow C$$
$$2T \Rightarrow T$$

are not effected by logic gates, but by diodes attached to terminals on the **MODE** switch.

The following table lists the delta (open-collector) lines used in the *Hapirat*: In each case, the circuit functions normally when the line is open, but the circuit is inhibited when the line is low.

Symbol	**Description**
δPS	Pulse generator
δT	Tone; 1000 Hz
δ2T	Tone; 1500 Hz

Logic circuit. The logic section schematic is shown in Figure 22 (six parts).

Figure 22.1; TTL gates.

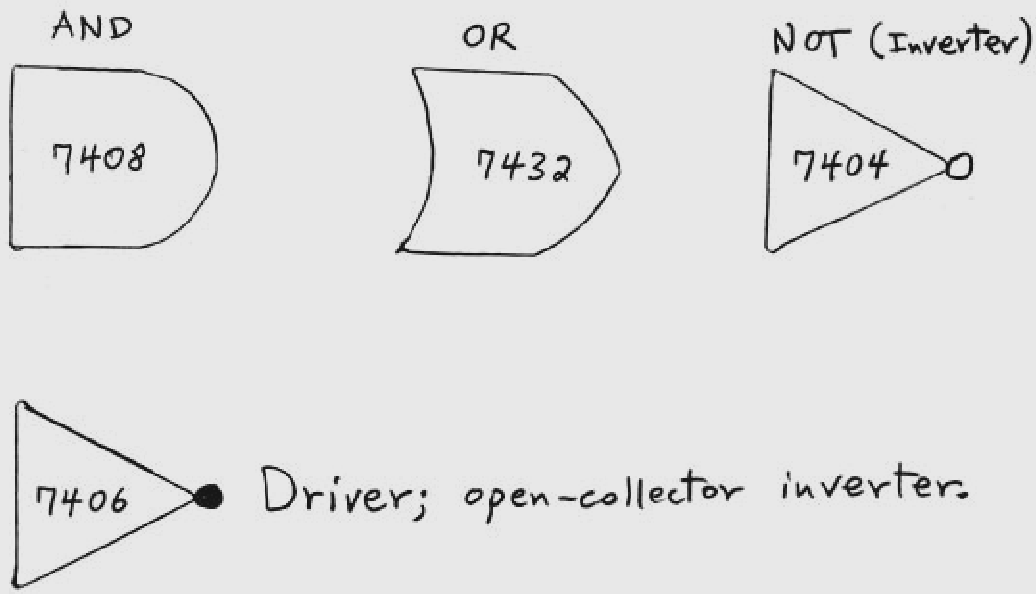
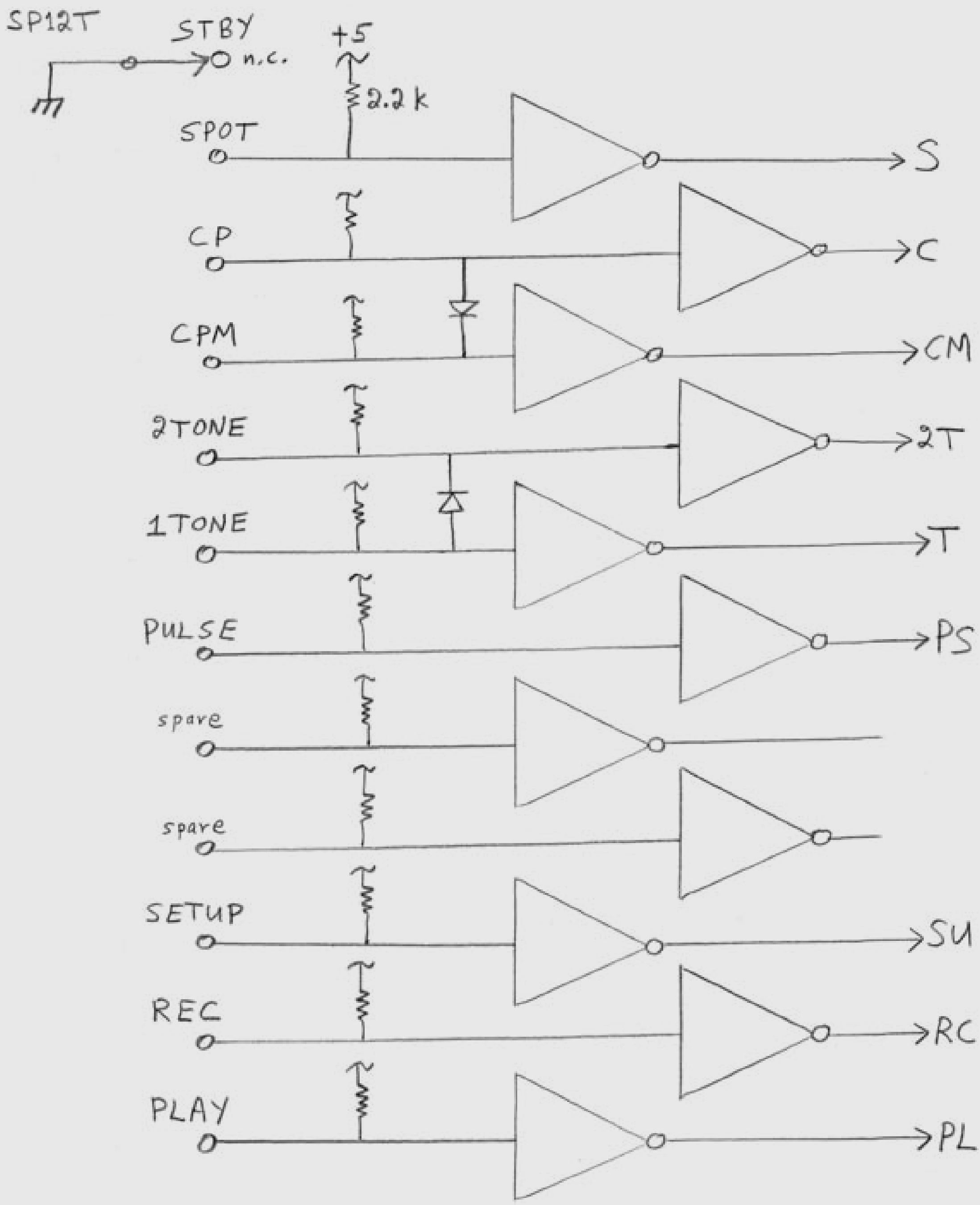


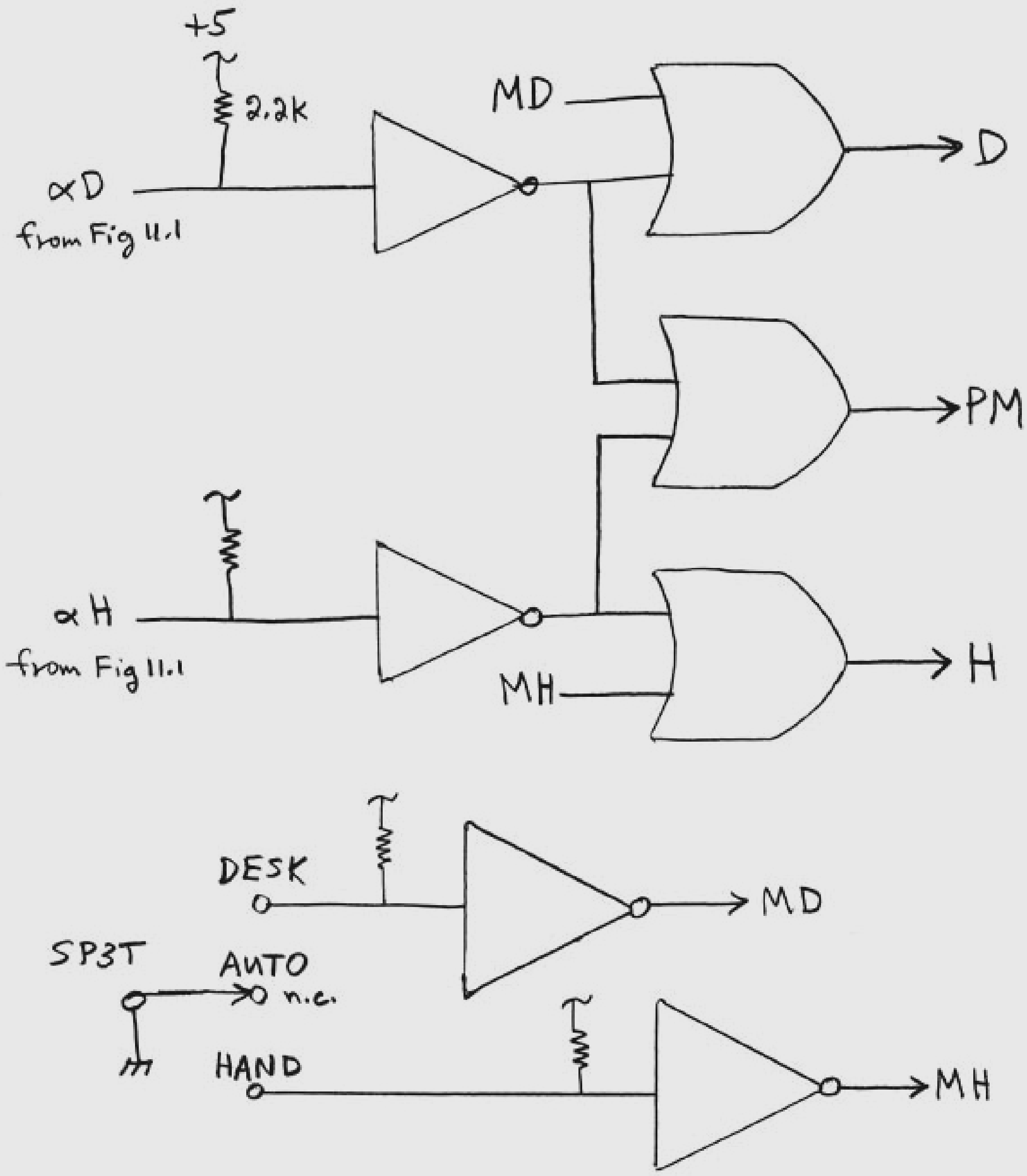
Figure 22.1 shows the symbols used for the gates. These are standard, except for the symbol adopted here for the open-collector driver/inverter, which in no case can be confounded with an ordinary TTL inverter.

Figure 22.2; Mode switch.



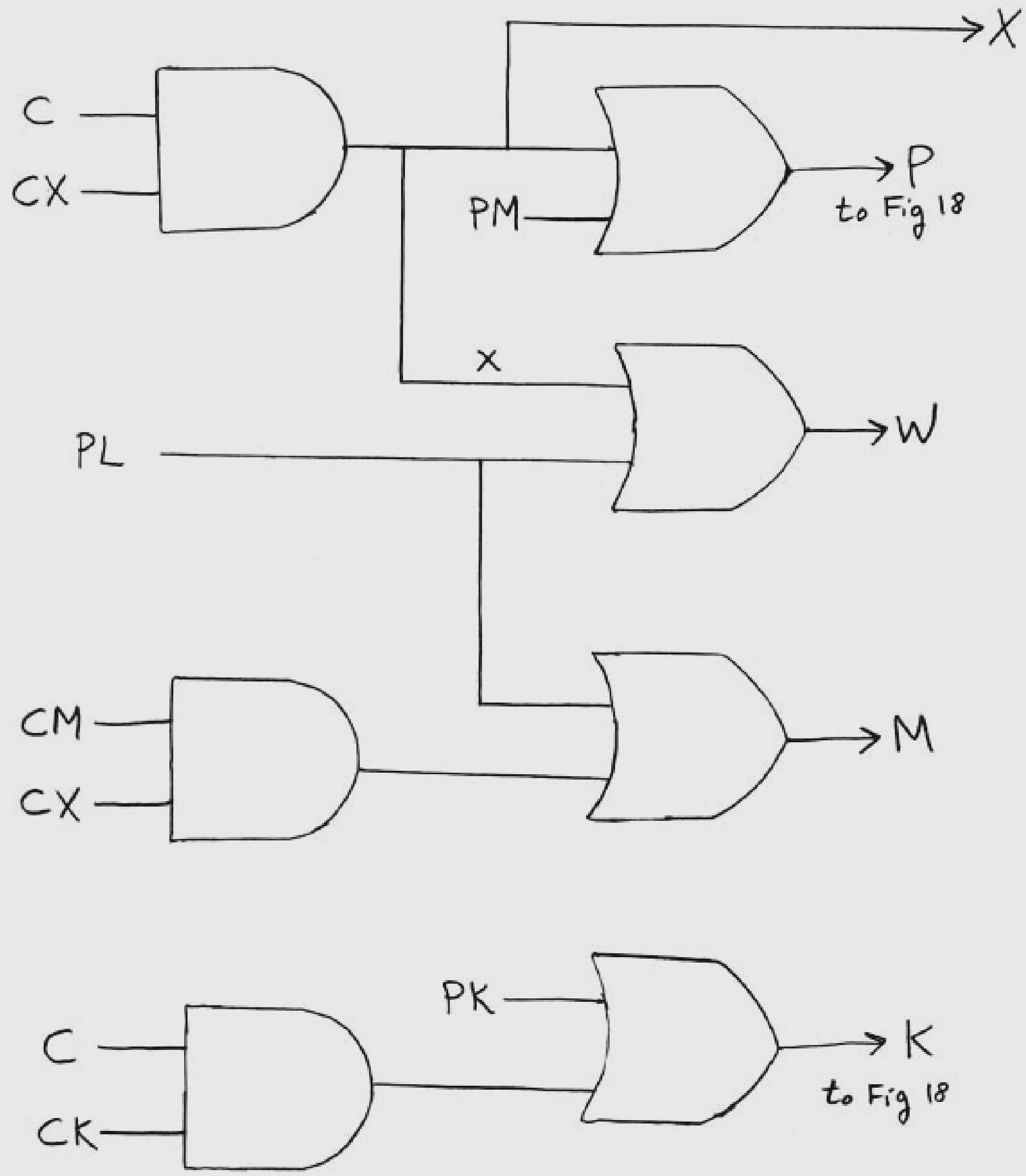
Note the implication diodes on the **MODE** switch in Figure 22.2.

Figure 22.3; Microphone selection.



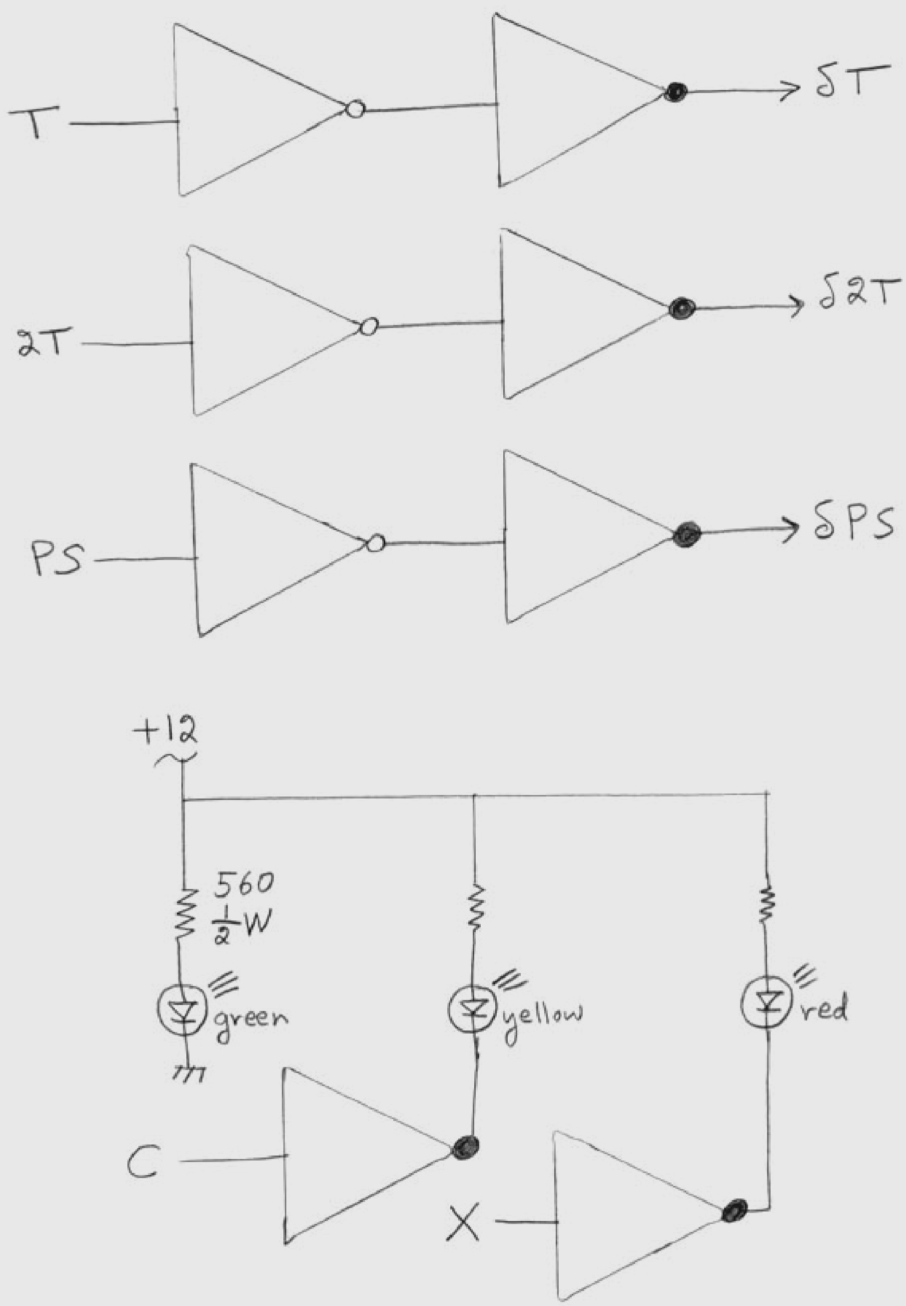
Mike selection, either automatic or manual, is carried out in Figure 22.3; the SP3T switch is on the front panel.

Figure 22.4; Function generators.



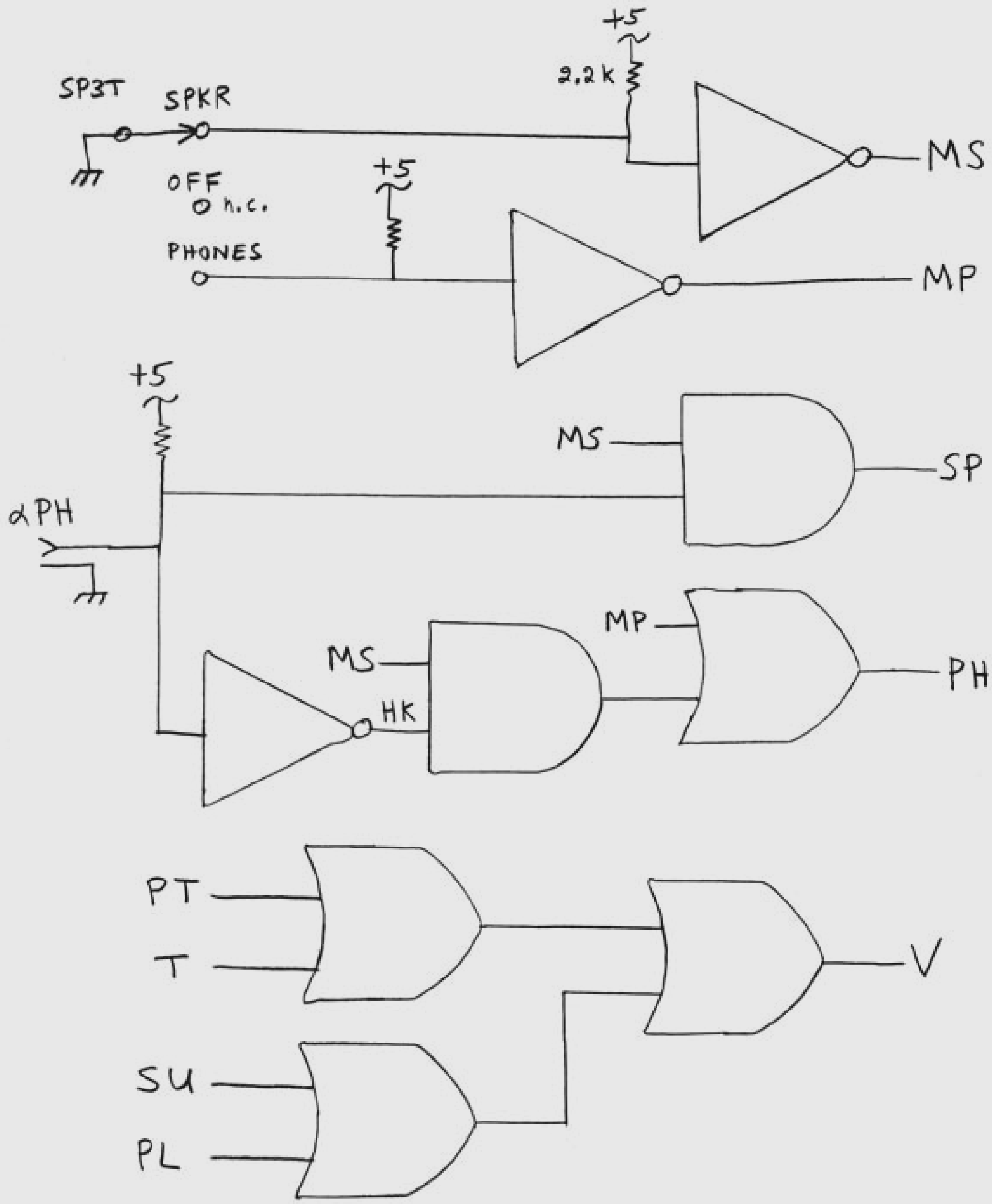
The main Boolean function generators are shown in Figure 22.4.

Figure 22.5; Drivers.



The drivers in Figure 22.5 control the two-tone generator, the pulse generator, and the LEDs.

Figure 22.6; Speaker, phones, meter.



Speaker/phones selection, automatic or manual, is shown in Figure 22.6. The jack labeled α**PH** connects to the switch on the headphone hook. Note that this line, with the special alpha prefix, is low for automatic switching to headphones. This arrangement allows manual phones selection when no hook switch is connected to the jack; otherwise a shorting plug would be required.

Parts. TTL type bipolar logic is used because it is less susceptible to RFI problems than the more sensitive CMOS types. The original 7400 series itself is nearly obsolete, but can be used. Surplus dealers, such as Jameco, often have some original 7400 types, at very low prices. Here we use mainly the current 74F series. When the large suppliers do have the 7400 types, they are likely to be more expensive than the 74F types. The open-collector driver 7406 is not a pure TTL type, but more of an interface to the outside world; Fairchild does not list a similar type in the 74F series, but supplies the DM7406.

Mouser supplies all the needed types:

7404	MO#512-74F04PC	Fairchild	Hex Inverter
7406	MO#512-DM7406N	Fairchild	Hex Inverter (open collector)
7408	MO#512-74F08PC	Fairchild	Quad 2-Input AND Gate
7432	MO#512-74F32PC	Fairchild	Quad 2-Input OR Gate

All the various bipolar types (TTL,S,LS,ALS,AS,F,FR) are input and output compatible; any of these could be used. More information is available at:
www.fairchildsemi.com/an/AN/AN-661.pdf
www.fairchildsemi.com/collateral/logic/catalog/logic_product_catalog.pdf
www.fairchildsemi.com/collateral/lsg2000.pdf

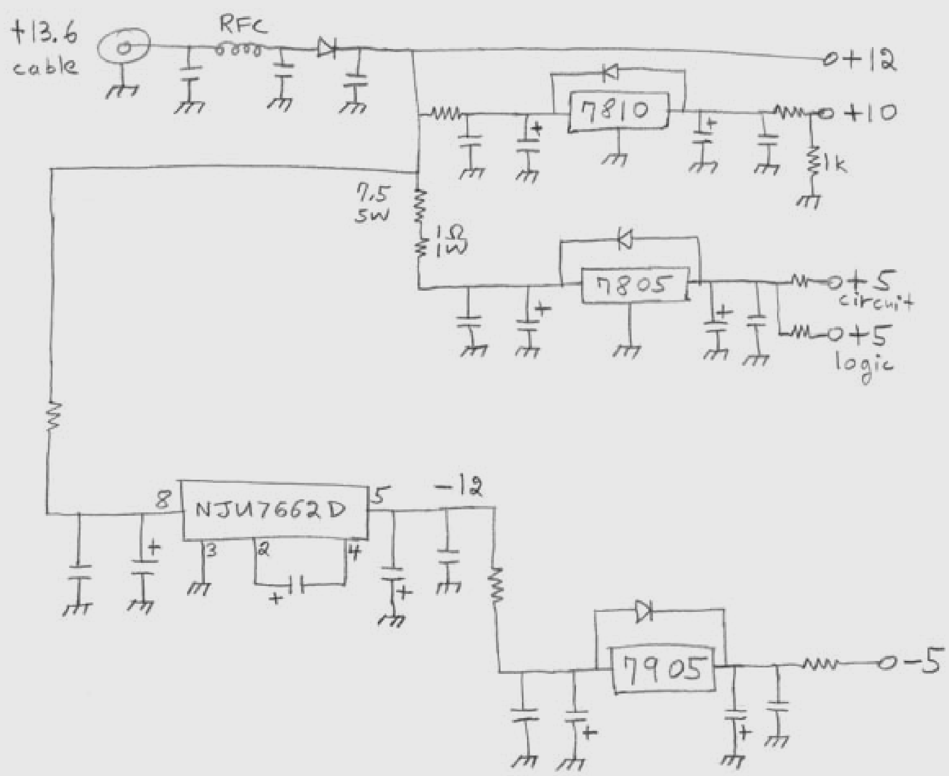
Many newer series are available with nomenclature of the form 74xxx00, but these may be intended for different voltages, for CMOS levels, or other systems, and must be avoided for this circuit.

23. Power Supply. The *Hapirat* is powered by a 13.6 Vdc supply on the operating bench which powers all the many accessory gadgets. Any well-regulated ripple-free 12 to 14 volt supply will suffice. Inside the panel, a negative voltage rail is obtained from a charge-pump voltage converter IC. The idea of a self-contained AC supply was rejected, as likely to introduce hum.

Power supply circuits. Having both positive and negative power rails available for the circuits is very convenient; it avoids the chore of having to establish V/2 potentials for the op-amp circuits and allows simple control of the audio signal switches. The resulting audio signals are ground-referenced; they swing each side of zero with no DC component. No blocking capacitors are required, except at a few special points.

An internal AC supply to provide the positive and negative power rails would bring AC fields from the transformer, and possible hum, into the circuits. Thus we definitely don't want an internal supply. A separate external positive and negative AC supply would work okay, but would be extra trouble. It is convenient to have all the accessory devices on the operating bench powered by one 13.6 Vdc supply. The simple solution adopted here is a dc-dc converter inside the audio panel. The power supply schematic is shown in Figure 23.

Figure 23; Power supply.



1. Electrolytics; 10 μF.
2. Bypass capacitors; 10 nF disk ceramic.
3. Resistors; 1 Ω, ¼ W, except as noted.
4. RFC; approx. 10 μH, must have less than 1 Ω dc resistance.
5. Diodes; 1N4001.

Negative supply; dc-dc converter. Type 7662 is an 8-DIP integrated circuit that produces nearly −12 volts from a +12 volt supply; it is rated at up to 50 mA, more than enough for this audio panel. The particular 7662 used here was able to produce 65 mA at −7.3 volts, enough to drive the 7905. This involves a device dissipation of 370 mW, well within the maximum rating of 500 mW.

In this audio panel, less than 30 mA is required from the −5 volt power rail, in any mode or state; this leaves up to 35 mA available for added features. With +13.6 volts input to the panel there will be 13.0 volts at the 7662 input. The older type 7660 should be avoided, because of its lower input voltage rating. The label −12 on the schematic diagram denotes the nominal output of the 7662; under load it is somewhat less, about −10 volts.

The dc-dc converter 7662 operates with an internal oscillator at 10 kHz. On the −12 volt line the ripple voltage is 150 mVpp; for direct use this would require more filtering in most applications. Here the −12 volts is not used directly; the 7905 regulator provides a high degree of ripple rejection. The −5 V line has only 0.6 mVpp ripple; this is 78 dB down. Also, the op amps have a supply-voltage rejection ratio of more than 80 dB. In summary, there is nothing to fear from the 7662 oscillator.

Regulators. The power supply provides +12, +10, +5, and −5 voltages. The nominal label +12 refers merely to the regulated +13.6 V input, after small voltage drops from RF filtering and the reverse-polarity-protection diode. The logic circuits use +5 V. Most of the op amps use +5 and −5 V, as do the CMOS signal switches. A few op-amps in non-critical circuits use +12 volts and ground, to lessen the load on the −5 V supply, although it turned out that there was plenty of available current at −5 volts. The tone oscillators run on a separately-regulated +10 V rail, to isolate them from the +13.6 V supply on the bench, and to keep firm control over the biasing.

In order to measure the current drains, 1 ohm resistors are used in the input and output lines of each regulator. Comparison of these currents will reveal leaky filter capacitors and other faults. When measuring currents at the 1 ohm shunts, a DMM is set to read millivolts; the readings are then interpreted directly as milliamperes. The 1 ohm 1/4 watt shunts might also, with luck, serve as fuses in certain fault conditions. Two +5 output pins are provided, with separate current shunts, for the logic section and the audio circuits.

Power supply parts.

7662. MO#513-NJU7662D, New Japan Radio (a.k.a. Japan Radio Company), DIP 8 package. Data sheet: sales@njr.com Request document "ae06046.pdf".

7810. MO#512-KA7810, Fairchild #KA-7810, TO-220 package, 10 volts. This regulator has a dropout rating of up to 2 volts; this is okay for devices operated from the usual 13.6 volt ham station power supply.
www.fairchildsemi.com/ds/KA/KA7810.pdf
In some situations, a low drop-out regulator would be better: MO#511-L4940V10, STM #L4940V10, TO-220 package, 10 volts. This regulator has a maximum dropout of 0.4 volts for moderate loads.
http://us.st.com/stonline/books/pdf/docs/2141.pdf

Power supply construction. The 7805, 7810, and 7905 are all TO-220 types. The last two carry a very light load, and do not need heat sinking. The 7805 is heat sunk by fastening to the aluminum side rail.

In lieu of a DC connector for the +13.6 input, this panel has a 6 foot length of RG-174 coax attached. At the end is a 2-pin Jones plug. This is the standard method for my shack. Distribution boxes with 2-pin Jones sockets, fuses, short leads to the station ground bus, and LEDs are located at several places on the operating bench. Other builders will use their own favorite methods, of course.

24. Construction.

The *Hapirat* is built into a 3 x 17 x 11 inch aluminum chassis behind a standard 3.5 x 19 inch rack panel. A black rack panel was chosen – to match the other 18 rack panels used in my homebrew station. The total height of these panels so far in 55 years of homebrewing is only 12 feet. (The *Hapirat* project, 3.5 inches in one year, barely exceeded the average rate.) Photo 24A shows how the audio panel fits into the homebrew station. Formerly, all the jacks were at the rear of the old audio devices, keeping the front of the operating bench completely clear. The compromise adopted now, with all the cables at the far left end of the bench, includes the convenience of the front-panel jacks.

Photo 24A. K5AM homebrew station. The *Hapirat* is above and to the left of the homebrew transceiver. Above the *Hapirat* is a transverter and a 200 watt tetrode amplifier for 144 MHz. Above the 6-digit clock is the HF section of the transceiver. Above the old HP scope, on rack panels, are two 50 MHz transverters, vintage 1963 and 1989. In the upper right corner is a small 50 MHz amplifier built while in high-school in 1951, still in constant use today. To the right is the 6 foot rack with kW amplifiers for 6 meters, HF, and 2 meters, vintage 1961, 1971, and 1995.

The panel height for this project was chosen to fit all the controls. Even then there was trouble; the old junk box surplus mil-spec controls would not fit, and I had to look for small imports. This worked out happily, however. I was especially lucky to find small controls with push-pull switches.

The depth of the box was chosen to fit the shelf. The circuits might fit into a panel with less depth, but I like the 11 inch depth, even for small projects – they fit nicely onto the "1x12" wooden shelves that I use at the operating bench. Then it's easy to reach all the rear panel connections from the passageway behind the bench. Although only about half the circuit board is used, the two constraints, panel space for the required knobs and depth for easy rear connections, determine the enclosure size. Also, it’s nice to have extra space for future additions.

Photo 24B.

The construction uses LMB *Omni Chassis* components. The special feature driving this choice is that both the top and bottom plates are removable – this is not possible with a standard chassis and cover. The circuit board is permanently mounted, so this feature ensures that both the top and bottom of the board are easily accessible. This eliminates the problems of movable boards and stress-relief for movable wires to prevent breaking. I call this construction method "No Moving Parts".

The *Hapirat* circuit board is shown in Photos 24C and 24D.

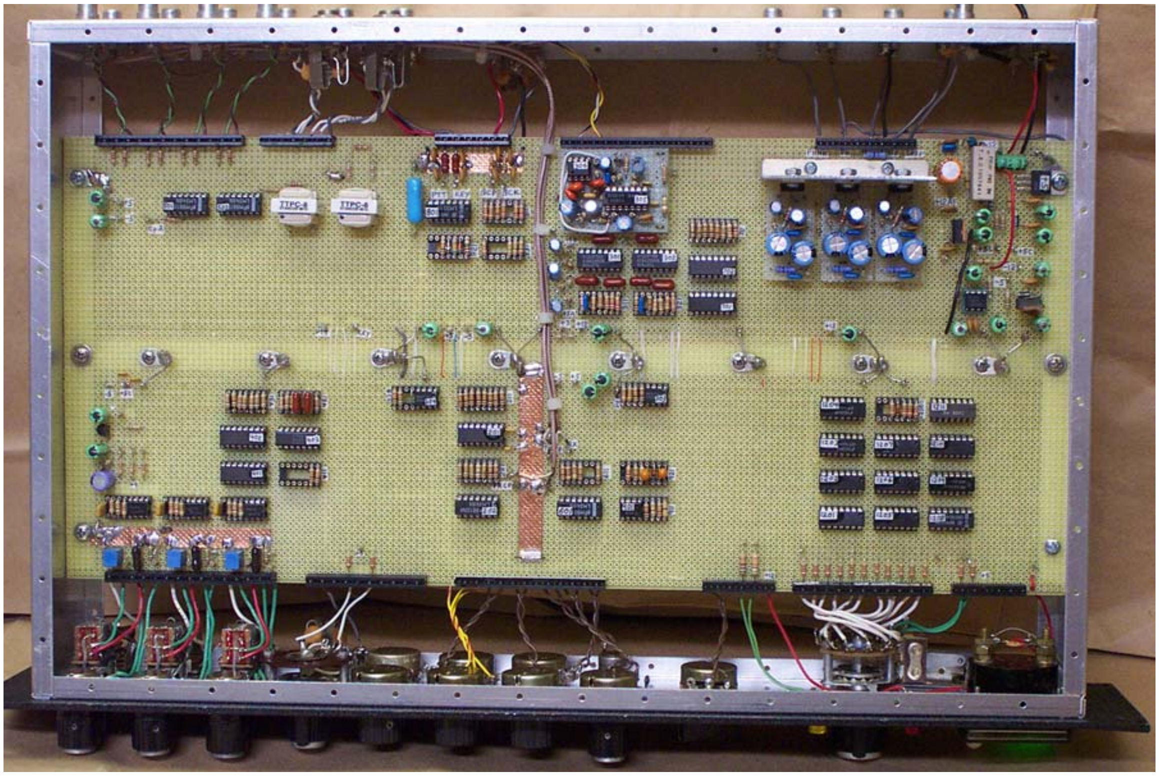


Photo 24C. Top view.

The copper foil at the lower left in the microphone preamplifier section is bonded to the aluminum chassis side rail, and provides a low-impedance path for the RF filter components. A second copper foil near the center is bonded to the aluminum chassis central rail, and provides a low-impedance RF path for the sensitive computer audio circuits.

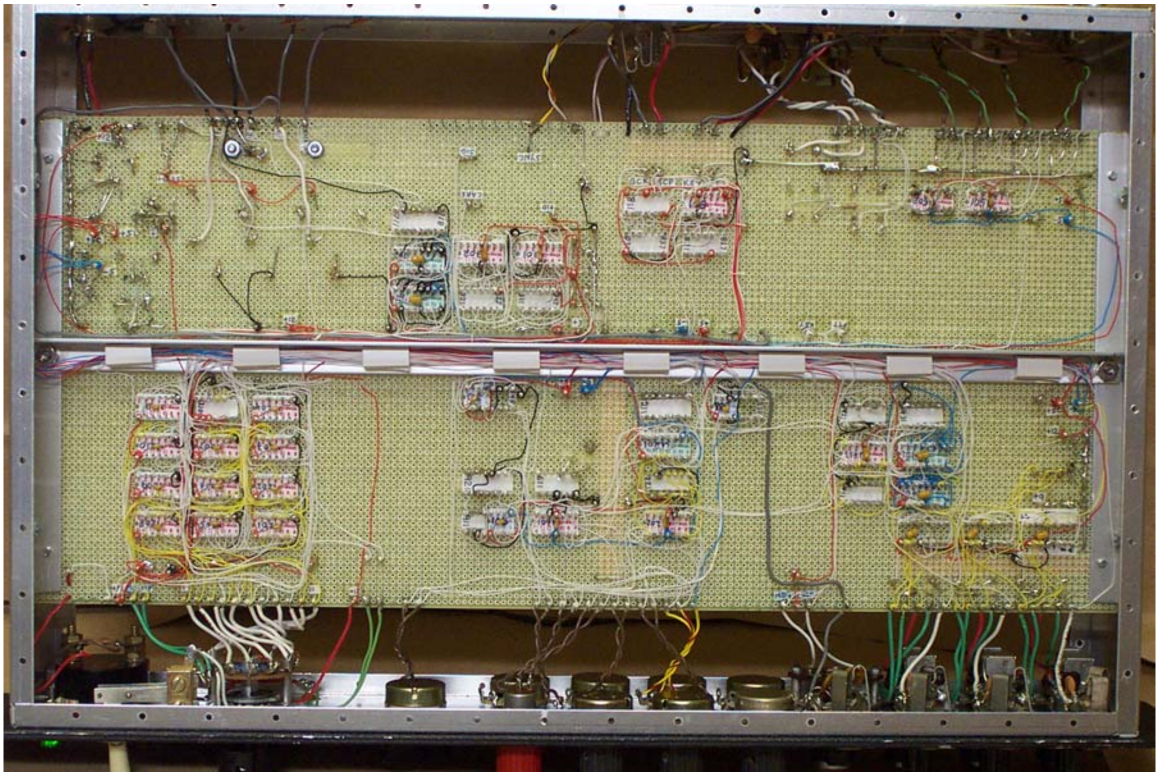


Photo 24D. Bottom view.

The LMB *Omni Chassis* series provides numerous pre-punched holes for fastening and results in an excellently shielded enclosure. Below is a list of materials needed to assemble the chassis and panel. LMB stock numbers are listed. The sides are 40 mil thickness aluminum; the covers are 50 mil; the front panel is 125 mil. The chassis is perfectly rigid when fully assembled with the grounding bars and circuit board, even without the covers. Screws are provided for the 104 holes in the covers. The fit and finish is so good that I use only 4 screws for the top and 4 for the bottom, and yet it seems perfectly RF-tight.

Chassis Parts:

Quant.	**LMB#**	**Type**	**Size (inch)**
1	S-317	Chassis front/rear (pr)	3 x 17
1	S-311	Chassis sides (pr)	3 x 11
2	C-1117	Top and bottom covers	11 x 17
1	350	Front panel	3.5 x 19 Specify black texture finish.

The board is supported by aluminum angles at the sides, and also an angle running from side to side under the center of the board. This center support serves mainly as a substantial common ground for all the circuits. Ample space remains at the front and rear for installation, connection, and servicing of all connectors and panel components, while observing the *"no moving parts"* rule.

Circuit board parts.
Board: 16.9 x 7.5 inches. Cut from 17 x 8 inch per board, Vector #169P79WE. DK#V1007, MO#574-169P79WE.
Support bars. Aluminum angle stock, 1/2 x 1/2 x 1/8 inch. Drill and tap to provide good connections for the ground lugs.
2 pcs. 7.3 inches
1 pc. 16.9 inches

Special care may be required in preparing the aluminum angles. Some hardware stores sell angles made from what I like to call "non-conductive aluminum." The aluminum, of course, is a good conductor, but it has a plastic coating. All mating surfaces must be scrapped clean. As I found out the hard way using a ground bus on the test bench, even an alligator clip might not penetrate the coating.

Terminals. The terminal strips used at the edge of the board are fashioned from inexpensive single row headers available from surplus dealers. They are often found with 12, 25, or 36 pins, each 0.025 inch square, 0.5 inch long, on 0.1 inch centers. I remove alternate pins; this leaves ample space for soldered connections. A 25 pin header makes a 12 or 13 pin terminal strip. The strips can also be cut to any needed length. First the leads from the front or rear panel are soldered to the terminals under the board, holding the pins tight to the board; then the wire-wrapping is added. The pulled pins are saved and used in other portions of the board, mainly as ground pins and power-rail pins.

In this project, the board is never moved, so the leads to the panels do not need strain relief, and can even be solid wires. When these terminals are used in other projects for a board that may have to be lifted or turned, then stranded wire is always used. In that case, each lead, with its insulation, passes first through two holes in the board (drilled out to fit), before connecting to the terminal.

Construction procedure. The following procedure has always worked well for me. First build the enclosure, including the panels, with all the controls, switches, labels, connectors, etc. This establishes the specifications for the project, and is completed even before the circuits are designed. It requires only a block diagram and a rough idea of the sort of circuits to be used to execute the specified functions. Now install the circuit board. Building a board first, and then later trying to fit it into an enclosure can lead to difficulties.

On the board, first build the power supply; this will enable each circuit to be tested as it is built. In this panel, the power supply is adjacent to the DC input cable, and at the side, for convenient heat sinking of the 7805 regulator. Now the logic section is built; this will supply control lines for testing the individual circuits. Finally, the individual circuits are designed to meet the specs, and built onto the board.

Photo 24E. Microphone preamplifier.

Preferred positions are assigned according to need. The microphone preamps are adjacent to the jacks – to ensure short leads and avoid hum pickup. The other sections of the circuit are spaced around the board, leaving space for modifications and additions. Testing each section as construction proceeds is important. Early detection of troubles in a new design makes necessary changes easier. Photo 24E is a close-up of the microphone preamplifier section. The copper foil is bonded to the aluminum chassis side rail at the left, and provides a low-impedance RF path for the filter components.

Several sections of the circuit are identified in Photo 24F.

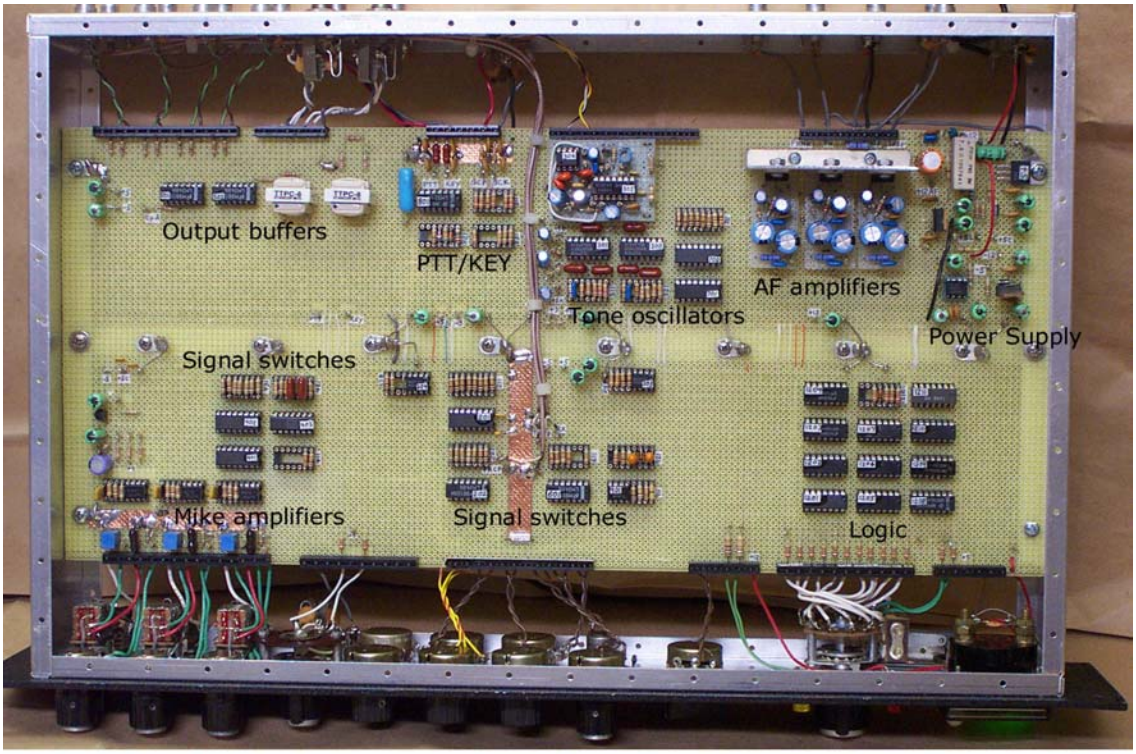


Photo 24F. Circuit section identification.

The rear panel is shown in Photo 24G. Several extra pilot holes have been drilled, ready for more connections or new features. Note the lock washers under the phono jacks; they ensure a good ground contact. Placing the lock washers on the outside, while not so pretty, makes secure tightening easier. The lock washers are not usually provided with the jacks; consult your local hardware store. Before mounting a connector, both sides of the hole are wire-brushed with a small battery-operated Dremel tool.

Photo 24G. Rear panel.

Wire-wrapping. Most of the *Hapirat* circuit is wire-wrapped. Besides the ICs, most of the other components, especially the 1/4 watt resistors and the small capacitors, are plugged into DIP sockets. This is easier than point-to-point perf-board wiring, and saves space. Most important is the ease of selecting components and making changes. Portions of the circuit with large components, including the entire power supply section, are wired point-to-point on the perf-board.

Photo 24H. Wire-wrapping is easy – if your XYL helps. Thanks, Lisa!

25. Parts. Every effort has been made to use readily-available parts. In a few cases where special parts were used, substitutions are possible.

Connectors. In a contest station, reliability is of the utmost concern. One of the most common failure points is at the connectors; it's too bad we can't just solder the whole station together.

This project at first used three 1/8 inch stereo jacks on the rear panel where 1/4 inch jacks appear now. The small jacks are quite handy, and ready-made cables are available everywhere. Hours and hours were spent with catalogs and websites trying to find high quality jacks, plugs, and cables. After several months there were no failures, but the jacks just didn't have the right feel; the plugs might have come loose. It seemed best to change to high-quality 1/4 inch jacks, even 40-year-old junkbox jacks, before trouble set in.[1]

[1]Is this being over-cautious? Well, at first I thought perhaps it was, although in 55 years of contesting one does receive quite a few visits from Mr. Murphy, and caution increases over the years. Then, a few months later, I read a report from a leading contester: placing only 2nd in a world-wide contest, he lost by less than 1% of the top score. He reported that he missed part of the contest when a 1/8 inch plug came loose from his computer.

Mike connectors. I use Collins type 0.210 inch jacks on the audio panel and the homebrew transceiver, with matching 0.206 inch plugs on all the old microphones in the shack. Patch cables are made up to fit other rigs. This is the simplest way to standardize mikes for an entire station – any mike can be used with any rig. A full selection of modern-type microphone connectors for making patch cords is available at Hosfelt.

Collins type 3/16 inch mike plugs and jacks are readily available, at various prices – and corresponding quality:
HF#224D.
MO#502-S-260. Switchcraft jack.
MO#502-S-12B, Switchcraft plug.
SSN. See catalog or website.

Bargains are sometimes found at flea markets. Any other type of mike jack could also be used, of course. The main idea is to use only one type for all construction in the entire ham shack, and to make patch cords as necessary. Then any mike can be used with any rig. The simplest method for this panel would be to use the inexpensive and ubiquitous 1/4 inch 3-circuit stereo phone jacks and plugs. It may be that only old-timers can appreciate the slim, elegant, WWII military, Collins/Drake 3/16 inch plugs, which establish a clear distinction between microphone plugs and headphone plugs.

Knobs. Most of the knobs are from junked Tektronix scope plug-ins, obtained free from a local surplus junkyard.
Spot level control knob. MO#5164-1500, round, pointer, 1.13"D, 0.57"H.
Mike load control. MO#5164-1900, Davies Molding, black, unlined, 0.50"D, 0.625"H; white dot added with nail polish. This small knob contrasts with the larger mike level knob above, and is easy to pull for

the electret bias voltage. Thus, panel labels for these six knobs are unnecessary.
Mode knob. Large, round, bar, white indicator, inset, from junkbox.
Mike mode knob. Same as *Mode*, except smaller.

Meter. Small DC panel meter, 1.75 inch square. Any range from 5 mA down is suitable. Resistor R802 is selected so that the meter indicates full-scale with 4 volts at test point TP8. Meters of higher range often have internal shunts which may be removed to obtain a more sensitive meter.

New, high-quality meters are expensive, even at surplus dealers; it is best to look for a meter at a ham flea market. An option, of course, is an LED bar graph, but most old-timers will insist on a bouncing needle. I was lucky to find a Simpson 200 microampere meter for 25 cents at a hamfest in Alamogordo, New Mexico, 15 years ago; it has been sitting on the shelf ever since – waiting for this project.

There is a high-quality 1.75 inch Triplett meter in the MPJ catalog at a reasonable price. It is for 50 mA, but meters in this range usually have internal shunts which can be removed. The result is often a meter for 5 to 10 mA. This meter might be worth a try.

Op amps. For routine audio work, the general-purpose dual LM1458N and quad LM348N are very satisfactory; these are improved 741-types. Avoid the single-supply types like the quad LM324N and the dual LM358N, because these have cross-over distortion.

For the microphone amps, the TL072 FET-input type has lower noise. It also has a higher input impedance; this may be useful for experimenting with old high-impedance crystal mikes, perhaps changing the mike load potentiometer to 5 MΩ. The higher input impedance also makes this type preferable for filters and peak-hold circuits.

Potentiometers. Linear taper, except audio taper for certain controls, such as **MON** and **SPOT**, which are generally set far below maximum.
Mike level pot. MO#313-1000-100K, Alpha/Xicon, 100K, 16 mm, carbon comp.
Mike load pot with push-pull switch. MO#313-1601-50K. Alpha/Xicon, 50K, 16 mm, carbon comp.
Other level pots. MO#313-1000-50K Alpha/Xicon, 50K, 16 mm, carbon comp. For audio taper, see Mouser catalog or website.

Headphone level pot. 100 ohm, surplus, wirewound if possible, small enough to fit. Check the value at minimum - old potentiometers often have too high a minimum resistance. MO#313-2401-100, wirewound, 5 watt, 24 mm, looks suitable, but more expensive.

Switches.
Speaker/Phones switch. Old-fashioned Switchcraft toggle switch, large plastic bat handle, SPDT, center off, from various surplus dealers. A similar switch is available as SSN#(SWL)323177, but it is momentary on one side. It might be modified, or parts of two switches could be re-assembled for the desired action. These are very fine switches, smooth operating and pleasant to use; my homebrew transceiver has nine of these on the front panel. Alternative: standard mini toggle switch, SPDT, center off. *Mode switch.* Rotary, single wafer, SP11T. No stop; the 12th position which does nothing is used for **STBY** mode. *Mike mode switch.* Rotary, SP3T.

Transformer; isolation. Telephone transformer, 600 ohm to 600 ohm, Stancor TTPC-8, HF#56-8444.

Wire and cable. Connections only a few inches in length from the edges of the circuit board to the front and rear panels are made directly with hook-up wire. The board is never moved or turned, so these wires require no strain relief.

Teflon-insulated wire can be used to avoid insulation melting from soldering iron heat. Less expensive, much easier to strip, and soldering iron resistant, is irradiated PVC wire. Alpha #7055-WH005, 22AWG, UL1429 WHT, MO#602-7055-100-01.
www.alphawire.com/pages/pdf/171.pdf

Three longer signal runs from the circuit board to the rear panel, for computer in, computer out, and receiver line audio, are made with miniature teflon-insulated coax, RG-178B. If special care is taken not to melt the insulation, and heat clamps are used, ordinary RG-174 is adequate.

Wire-wrapping sockets. The cost of logic chips for this project is minimal. The wire-wrapping sockets cost more, if reliable, high-quality types are used. MO#575-293314, #575-293316.

26. Alignment.

The *Hapirat* is designed to involve no internal adjustments or settings. All circuits must be tested, of course; the best way is stage-by-stage testing as construction proceeds. The proper signal levels are marked on the schematic diagrams, in mVpp. In case of larger-than-usual component value variation, some adjustments can be made by changing resistors in the sockets.

Some stubborn perfectionists may prefer to include trimpots in a dozen or more spots, for precise adjustment. That would normally include me, but I exercised some restraint this time. Here, however, are a few suggested places for refinement.

Meter adjustment.
Full-scale. In lieu of R802, a fixed 2.2k resistor in series with a 5k trimpot may be used for a more precise full-scale reading, although this is not necessary. The series resistor is suggested to prevent meter damage in case of improper adjustment. To adjust, select mode **2TONE**, advance control **2T** fully clockwise, and adjust the trimpot for a full-scale reading. The values suggested, 50% of calculated value or a bit more for the resistor, and 100% or something close for the trimpot, provide a wide adjustment range, and an overload limit of 100%. This method has been used for several dozen meter circuits in other equipments.
Half-scale. Due to component variation, the meter may not read exactly mid-scale to indicate the fixed 3000 mVpp level at test point TP14. This is not a critical issue; the operator should make a note of the proper reading, for example 45%, and set the panel controls accordingly. For an exact setting, U801 can be reconfigured as an inverting amplifier with a trimpot.

27. Results. The *Hapirat* has been used with excellent results for nearly a year; it is a joy to have it on the operating bench. The station operates exclusively using the author's homebrew transceiver, but the audio panel has also been tested with Yaesu, Kenwood, and Icom equipment. Microphones used include the Heil boom-mikes, the Astatic D-104 crystal, the ElectroVoice 664 dynamic and 719 ceramic, and the Icom IC-SM2 electret-condenser microphones.

There is no ready-made circuit board available for the *Hapirat*. Readers might wish to fabricate a board, perhaps even using surface mount components. However, the wire-wrap method used here allows for convenient changes and additions. Building “from scratch”, with no manufactured board, parts kit, or pre-programmed CPU, will provide a level of satisfaction rarely found in today’s plug-and-play world, will usually produce a learning experience, and will help preserve the traditions upon which ham radio was founded.

28. Summary. After years of wrestling with the rat's nest of cables behind the rig, I am now really a happy rat. This panel doesn't eliminate all the cables, but it organizes them – they all connect to this one panel. The *Hapirat* replaces a large number of interface gadgets made over the years. Readers are certain to notice places for circuit improvements; please send suggestions to the author: k5am@arrl.net

29. Biographical sketch. Mark Mandelkern, K5AM, was first licensed in 1948 as W9ECV in Wisconsin. His ham station is completely homebrew; for details, see www.zianet.com/k5am. Mark's main ham operating activity is 6 meter DX; 134 countries. He also enjoys HF DXing and contesting; he has DXCC confirmed on 10 bands and a number of top-ten contest awards.

Mark is emeritus professor of mathematics at New Mexico State University; his mathematical research is in topology, constructive analysis, and the foundations of mathematics.

30. References. The references listed in this section are for articles by K5AM. Other references are given within the text.

30.1 Homebrew transceiver (8 articles).

A High-Performance Homebrew Transceiver: Part 1, *QEX*, Mar/Apr, 1999, 16-24. [General plan]
www.arrl.org/members-only/tis/info/pdf/990304qex016.pdf

A High-Performance Homebrew Transceiver: Part 2, *QEX*, Sept/Oct, 1999, 3-8.
[IF board; notes in *QEX*, Nov/Dec 2000, p.60.]
www.arrl.org/members-only/tis/info/pdf/990910qex003.pdf

A High-Performance Homebrew Transceiver: Part 3, *QEX*, Nov/Dec, 1999, 41-51.
[RF board; corrections in *QEX*, Jul/Aug 2000, p.59, and in *QEX*, Nov/Dec 2000, p.60.]

www.arrl.org/members-only/tis/info/pdf/991112qex041.pdf

A High-Performance Homebrew Transceiver: Part 4, *QEX*, Jan/Feb, 2000, 47-56. [AF board]
www.arrl.org/members-only/tis/info/pdf/000102qex047.pdf

A High-Performance Homebrew Transceiver: Part 5, *QEX*, Mar/Apr, 2000, 23-37. [Logic board, etc; corrections in *QEX*, Nov/Dec 2000, p.60.]
www.arrl.org/members-only/tis/info/pdf/000304qex023.pdf

HF Circuits for a Homebrew Transceiver, *QEX*, Nov/Dec, 2001, 20-42.

A High-Performance AGC System for Home-Brew Transceivers, QEX, October, 1995, 12-22. [Corrections in *QEX*, Jul/Aug 2000, p.59.]

Evasive Noise Blanking, QEX, August, 1993, 3-6.

30.2 Pulse keying for amplifier tune-up.

A Luxury Linear, QEX, May, 1996, 3-12. (Photos also in QST, Jul 1996, p.19.)

Design Notes for "A Luxury Linear" Amplifier, QEX, November, 1996, 13-20.

30.3 Station description www.zianet.com/k5am/ncj/ncj.html.

30.4 The AMSAFID: An Automatic Microphone Switcher Amplifier Filter Integrator Distributor, *QST*, November, 1995, 47-49.

Appendix I. Parts dealers.

CS. Circuit Specialists, 220 S. Country Club DR., Mesa, AZ 85210, 800-528-1417. www.circuitspecialists.com

DK. Digi-Key Corporation, 701 Brooks Ave S, PO Box 677, Thief River Falls, MN 56701-0677, tel 800-344-4539 (800-DIGI-KEY), fax 218-681-3880. www.digikey.com

HF. Hosfelt Electronics, 2700 Sunset Blvd, Steubenville, OH 43952-1158, tel 800-524-6464, fax 800-524-5414; hosfelt@clover.net, www.hosfelt.com catalog@hosfelt.com

JA. Jameco Electronics, 1355 Shoreway Road, Belmont, CA 94022-4100, tel 800-831-4242, 415-592-8097, fax 800-237-6948, 415-592-2503. www.jameco.com

LMB. LMB Heeger, Inc, 6446 Flotilla, Commerce, CA 90040,: tel 323-728-5108, fax: 323-728-4740. www.lmbheeger.com

MO. Mouser Electronics, 2401 Hwy 287 N, Mansfield, TX, tel 800-346-6873, fax 817-483-0931; sales@mouser.com. www.mouser.com

MPJ. Marlin P. Jones Associates, Inc., Box 12685, Lake Park, FL 33403-0685, tel 800-652-6733, 561-848-8236, fax 800-432-9937, 561-848-8299. www.mpja.com

SSN. Surplus Sales of Nebraska, 1502 Jones Street, Omaha, NE 68102, tel 402-346-4750, fax 402-346-2939. www.surplusales.com

Appendix II. Glossary.

CRT. Cathode ray tube; found inside oscilloscopes, including those used to monitor RF amplifiers to ensure linearity. The CRT is unique in its ability to respond to high frequency RF signals applied directly the vertical plates. For this application the CRT has not yet been superceded by flat screens.

Electret-condenser microphone. The electret microphone is a modified version of the classic capacitor (or condenser) microphone, which exploits changes in capacitance due to mechanical vibrations to produce voltage variations proportional to sound waves. Whereas the condenser microphone needs an applied voltage, the electret has a built-in charge. The few volts of required bias are to power a built-in FET buffer, not to create an electric field.

HPF. High pass filter.

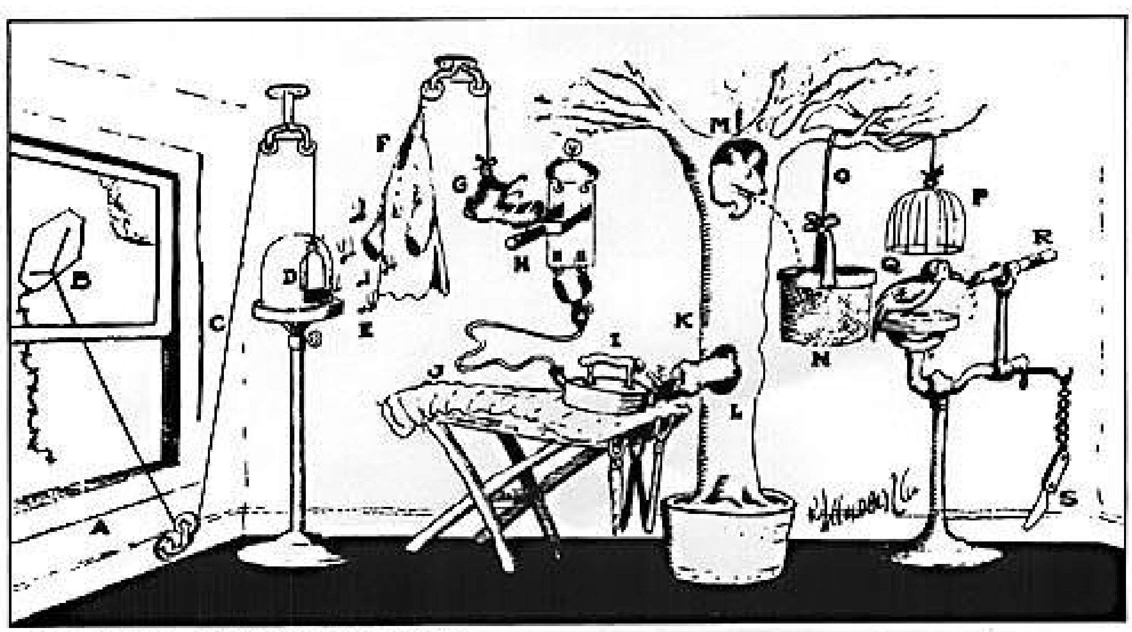
LPF. Low pass filter.

mVpp. Millivolts peak-to-peak. For a sine wave, this is about 2.8 times the RMS millivolt level.

RFI. RF Interference. Faulty operation in any device caused by poor design, especially lack of grounding, shielding, and filtering to prevent unwanted RF energy from entering sensitive circuits. In a ham station, a typical case involves RF energy from the antenna entering a speech amplifier in a transceiver, causing distorted and unintelligible transmissions. See also TVI.

Rube Goldberg type device. Rube Goldberg (1883-1970) was a Pulitzer Prize winning cartoonist, sculptor, and author. His cartoons were, as he said, symbols of man's capacity for exerting maximum effort to accomplish minimal results. Rube believed that there were two ways to do things: the simple way and the hard way, and that a surprising number of people preferred doing things the hard way.
www.rube-goldberg.com

Rube Goldberg's simplified pencil-sharpener:



Pencil Sharpener RUBE GOLDBERG (tm) RGI 038

Open window (A) and fly kite (B). String (C) lifts small door (D) allowing moths (E) to escape and eat red flannel shirt (F). As weight of shirt becomes less, shoe (G) steps on switch (H) which heats electric iron (I) and burns hole in pants (J). Smoke (K) enters hole in tree (L), smoking out opossum (M) which jumps into basket (N), pulling rope (O) and lifting cage (P), allowing woodpecker (Q) to chew wood from pencil (R), exposing lead. Emergency knife (S) is always handy in case opossum or the woodpecker gets sick and can't work.

Hams who have been operating for years on RTTY with a one-transistor interface circuit may view the *Hapirat* as a Rube Goldberg type device.

SO2R. Single-operator, two-radio. A contesting method whereby one operator uses two radios simultaneously.

TU. Terminal unit. Electronic device used to detect RTTY signals, for printing on paper with a large machine; used by hams before digital computer programs became available. The TU shown below was built in 1964, and uses 19 vacuum tubes. It uses sharp filters constructed from 88 mH telephone coils. The large dial in the center is for variable shift; one tone goes directly to one filter, the other tone is heterodyned to the frequency of the second filter. The built-in cross-hair scope is used for receiver tuning and shift tuning. This system was very useful for printing stations whose shift was not precisely the standard 850 cps used at the time. This TU was used for the first 50 Mc RTTY meteor scatter contacts (ref. *QST*, Feb 1965).

TVI. Television interference. Often related to ham radio signals, this is nearly always caused by poor design in the TV receiver, especially lack of selectivity, grounding, shielding, and filtering to prevent unwanted RF energy from entering sensitive circuits.

Vpp. Volts peak-to-peak, as read on a scope. For a sine wave, this is about 2.8 times the RMS voltage.