Modifications of the Heathkit HW-8 QRP Rig

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Back in 1979 I had built an HW-8 and gave it to another ham in our family. This HW-8 came back into my possession during the pandemic, and after using it during a "Straight Key Night," I decided to modify the rig to overcome what, to me, were annoyances and drawbacks.

This write-up describes that "modification" as a collection of projects which are ranked by increasing levels of difficulty. The "easy" modifications can be done with little more than a soldering iron and a modest toolkit. The more involved modifications are best done with access to an oscilloscope and function generator. (Paraphrasing Byron Goodman in a January 1949 QST article, "You don't need to own the equipment; you just need to know somebody who does.")

In its original (unmodified) state, my HW-8 drew 91 ma from a "12 VDC" supply while receiving. To support portable operation, the modifications keep the battery drain at that level.

The modifications are:

1) Allow the use of a pair of small stereo headphones.

2) Bring the audio pitch of "optimal signals" down to around "A 440 Hz."
Also, as part of this, improve the quality of the "sidetone."

3) Provide a "100 KHz Calibrator."

4) Provide an output to drive an external frequency counter.

5) Provide improved audio bandpass filtering

6) Provide "Full Break-In" operation

Modification #1: Allow the use of small stereo headphones

The HW-8's final audio amplifier stage (Q201 etc.) was designed to drive older-style headphones which have a resistance of around 200 ohms. Modern stereo headphones have a much lower impedance - typically around 10 to 25 ohms when the "left" and "right" earpieces are driven in parallel.

Many HW-8 owners install a 'buffer' amplifier to boost the available audio output. But the "buffer" adds to the 'supply current drain,' which is contrary to my goal of keeping the "receive mode" battery drain at low levels.

The battery current stays at low levels if we put the HW-8 audio output through an impedance-matching output transformer. A check with DigiKey or Mouser shows that suitable transformers (such as a Hammond model 149C) cost around $20-$25, but there is a less expensive approach.

We can use a pair of low-cost "600 ohm CT to 600 ohm CT" transformers. My modification uses a pair of Triad TY-145P transformers (Jameco PN 630459), which cost $6 each. (A suitable headphones jack is Jameco PN 1766139. When purchasing a "3.5mm" headphone jack, check to confirm that its threaded bushing is long enough to allow installation on the HW-8 back panel. In this regard, the Jameco PN 1766139 is barely adequate.)

These small transformers are inexpensive, but they have a lot of DC resistance in their windings. We can bring the resistance down to tolerable levels by connecting two transformers in parallel.

On the "Triad" transformers, one winding has around 45 ohms of DC resistance (measured from one corner pin to the other). The other winding has around 58 ohms of resistance. We use the "lower resistance" windings to drive the stereo headphones.

Using a fine-tip marker, label a corner pin of each transformer's "45 ohm" winding as "GND." (Label the same corner pin on each transformer.) We wire up the transformers like this:

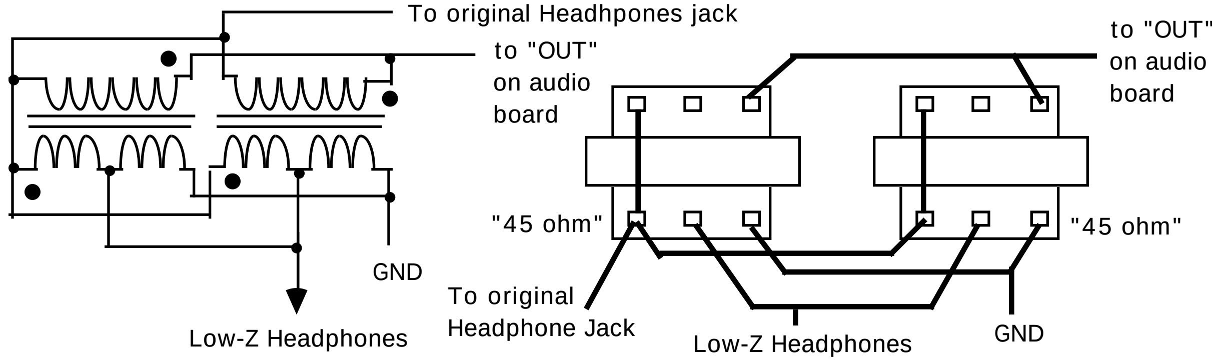


Figure 1: Wiring for a pair of Triad TY-145P Transformers

With the transformers' windings connected in parallel (as shown above), the stereo headphones are being driven by a winding with a DC resistance of around 11 ohms - i.e, the resistance of this "secondary" is a bit lower than the resistance of the headphones themselves. And with the 1/4" jack now wired to the junction between the two sets of windings, a "200 ohms" set of headphones appears to be 800 ohms at the output of the "audio amplifier" board.

I wired up the transformers according to Fig. 1, and checked their operation with an audio frequency sinewave generator and oscilloscope. However, lacking this equipment, you can temporarily wire up your transformers to a 1/4" phono plug, a "3.5mm stereo" jack, and your headphones, to make sure that things are working properly.

Perhaps you purchased some other "600 ohm CT to 600 ohm CT" transformers (rather than the TY-145P transformers). If you find that the resulting audio volume (in the headphones) is very weak, it's likely that the phasing of one of the windings is "backwards" in comparison to the TY-145P units. In this case, try reversing the polarity of the wiring to the "55 ohm" windings as shown in Figure 2:

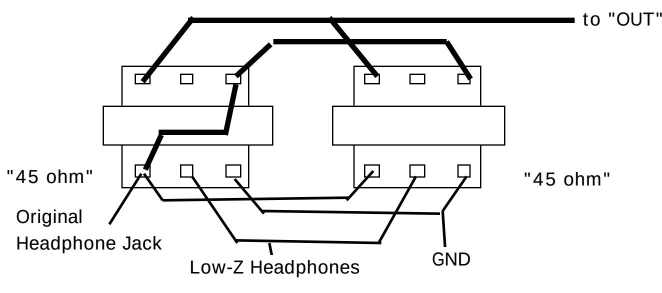


Figure 2: Alternate wiring to try if "Fig 1" wiring doesn't work.

The pair of Triad TY-145P transformers (as purchased from Jameco) provides a low-frequency cutoff of around 40-50 Hz, which is fine for CW work.

After confirming that your transformers are able to drive the stereo headphones, the transformers can be glued onto the side panel of the HW-8, above to the "Audio Amplifier" board. I suggest using a "clear silicone rubber" glue so that, if need be, the transformers can later be removed.

Modification #2: Bring the audio pitch of "optimal signals" down to "A 440 Hz"

The original HW-8 is set up for a fairly high amount of frequency shift. Many older hams might have a hearing loss that tends to drop off at these higher frequencies. Also, when some other stations appear to be near the frequency of the station that you're attempting to copy, it becomes difficult to distinguish that station from the QRM if everybody is "close to the same pitch."

This modification is in three parts. First, bring the "xmt frequency shift" down to 440-450 Hz. Second, bring the "peak frequency" of the bandpass filters down to 440-450 Hz. Finally, bring the pitch of the sidetone oscillator down to 440-450 Hz.

For this step, it's handy to have access to a musical instrument that can play "A above middle C." Another option is to have an audio-frequency sinewave generator driving a spare pair of headphones, and set the generator's frequency to around 450 Hz. An oscilloscope can also be used.

<u>Reducing the XMT Frequency Offset</u>

Note: This step requires a physically small trimmer capacitor with around 10 or 15 PFd maximum capacitance. I used a "3.5 to 13 PFd" trimmer (Jameco PN 134818).

Remove "12v" power from the HW-8.

Locate D11 (near the front panel, close to the "21 MHz" selector switch). Then locate C55, which is the small ceramic cap adjacent to D11. (If you have an HW-8 assembly manual, the "C55" step is at the bottom left of page 28.).

Lift one end of C55 (leaving the other end attached to the PC board).

Solder a short length of "tinned solid" wire to one side of the trimmer capacitor. (I used wire that was clipped from a 1/4W resistor.) For the other side of the cap, prepare it by "tinning" it with solder.

Drop the trimmer's "tinned solid wire" into the hole formerly occupied by the "C55" lead. Connect the other side of the trimmer cap to the now-elevated free lead of C55.

Adjusting the trimmer:

> <u>Option 1</u>: If you have built the "uA733" amplifier (on page in Figure 7 of this write-up), then connect a frequency counter to the "freq counter" output jack. Apply "+12v" power to the HW-8 and have the radio driving a dummy load. Run "7.0 MHz" Put the frequency counter in "1.0 second gate time" mode. Key the transmitter, and adjust the tuning until the counter reads 7.00000 MHz (least-significant digit is "tens of Hz"). Then go to "key-up" and note the reading. Find a trimmer cap setting that results in "key-up" being 7.00044 MHz after "xmt" has beein re-adjusted to 7.00000 MHz.
>
> <u>Option 2</u>: Use a separate ham-band receiver with a few feet of wire as the antenna. Apply "+12v" to the HW-8, set it up for "7.0 MHz" operation, and connect its antenna connector to a dummy load. Have the separate receiver set for SSB. Tune the receiver so that you can hear the HW-8 local oscillator signal. Adjust either the HW-8 or your receiver so that this local oscillator signal is "zero beat." Then key the HW-8 and note the audio pitch of your receiver. Readjust the trimmer until (after readjusting for zero beat in key-up) the "key-down" audio pitch is around "A 440 Hz."
>
> <u>Option 3</u>: Using a digital oscilloscope that provides "frequency of the trigger signal" display, connect a "x10" probe to the "not-grounded" side of R51 (a 100 ohm resistor). Apply +12v to the HW-8, and have the "antenna" connector hooked to a dummy load. Adjust the oscilloscope until you can see the "frequency" displayed. Then use the "trimmer adjustment" steps of "Option 1" above.

Set the audio bandpass filters for "440 Hz peak."

Let's assume that you plan to continue to use the original bandpass filters. In the original HW-8 design, the two audio bandpass filters appear to have center frequencies around 700 Hz. (To identify this, I drove the junction of R19, R21, C34, C35 from function generator via a hacked 1Meg resistor.) On my unit, the broad filter was centered at 725 Hz, and the narrow filter was at 674 Hz, as judged by observed phase shifts between the function generator and the "filter amplifier output."

Add 1000 pfd across C34 and C35 (across the two 1800 PFd caps).

Add 570 pfd (470 pfd in parallel with 100 pfd) across C36 and C37 (the two 1000 PFd caps). Resonant peaks (as measured by phase shift) are now at 500 Hz for the "broad" filter, and 455 Hz for the "narrow" filter.

Set the Sidetone Oscillator at "440 Hz" and clean up its output

This modification assumes that you are happy to continue to use the original HW-8 "Vox" control of the "Transmit/Receive Relay." It also assumes that you are happy to have a very loud amount of sidetone, relative to the headphones volume of "normal receiver" operation.

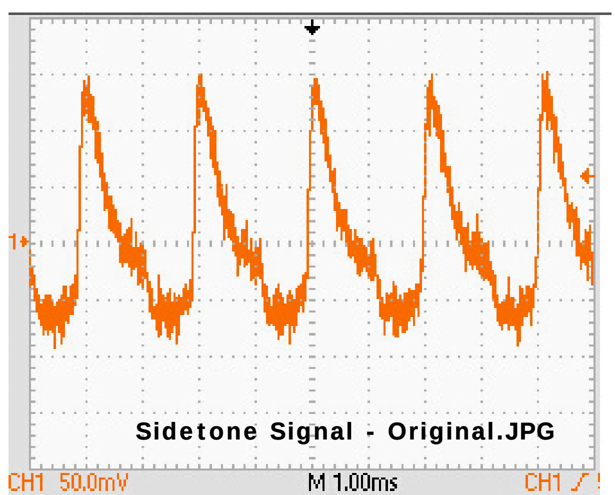
If you are going to be adding "full break-in" to the HW-8, do not bother with this modification. The "break-in" modification uses a separate dedicated "sidetone oscillator."

An ideal "sidetone" frequency would be around 440 Hz (matching the "XMT vs RCV" frequency shift).

A check of headphone voltage of the original HW-8 shows that this almost the case - the sidetone oscillator is running at 500 Hz.

However, the sidetone waveform has a lot of high frequency content, which I find to be annoying.

Figure 3:
Sidetone as observed at headphone jack with the original HW-8 circuit



The frequency can be pulled down to 440 Hz by raising the value of C109 up from "0.022 uF" to around (500/440) x 0.022 = 0.025 uF --> on the backside of the board, add 3300 pfd in parallel with C109.

Try to eliminate most of the 2nd harmonic by making the oscillator's waveform symmetrical. (A symmetrical square wave does not have even harmonics.) Experimentally I find that the waveform at Pin 10 is symmetrical when R73 is increased by around 5 Meg. I lifted one end of R73 from the board, and installed a 4.7 Meg resistor between the "free end" and the board foil.

In the original HW-8 circuitry, a 1-pole low-pass-filter was formed by R76 (1.00K) and C112 (0.1uF), for a cutoff of 10,000 rad/sec = 1.6KHz. We set this filter's cutoff to around 400 Hz --> We feed C112 with something that is at least 4.0K. Use 5.1K --> cutoff is 1960 rad/sec = 310 Hz. (This resistor could also be 4.7K without any significant change in filter's behavior.)

Add in a second capacitor, to make this a pair of low-pass filters in succession. The filter's RC 'tau' is C x (1.00K || 5.1K)   If C = 0.47uF then 'tau' = 0.39 msec --> cutoff freq = 400 Hz.

The filter is implemented by lifting the "pot end" of R76 up from the board, installing a 5.1K resistor from the "free end" of the 1.0K resistor and the resistor's original "circuit board foil" connection.

On the component side of the board, probe among the nearby components to find a resistor lead that is tied to "ground."   With this lead becomes the "ground point" of your hacked "0.47uF" capacitor. (Another option is to "hack solder" to the "ground" pin of the "sidetone volume" potentiometer.)   The free end of your 0.47uF cap goes to the junction of the "R76 1.0K" resistor and the new "5.1K" resistor

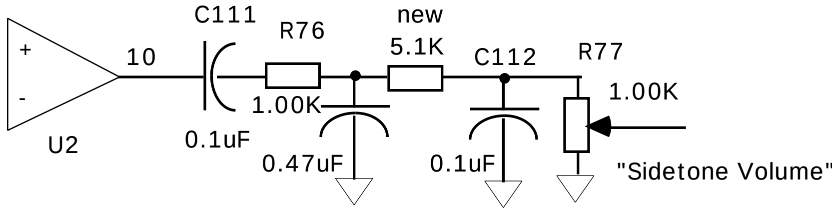


Figure 4: Output Filter for Reworked "HW-8 Sidetone Oscillator"

The resulting waveform (at the "Sidetone Volume" pot) is considerably improved.

Sidetone Signal - Revised.JPG

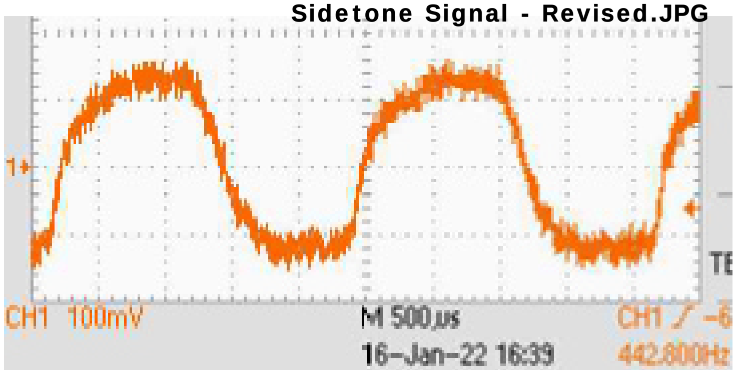


Figure 5:
Sidetone as observed at top of "Sidetone Volume" pot with the Revised Oscillator Circuit and "Output Filter" of Figure 4

We need to take note of a subtle problem with this circuit: The "lower arm" of the "Sidetone Volume" potentiometer is directly in parallel with the headphones! Even if using older "high impedance" headphones, it might be problematic to connect the R77 pot wiper directly to the headphones jack if the "sidetone volume" pot is adjusted for low volume. Therefore, a further improvement would be to remove C113, and in its place install a 1.0K resistor. Note that the wire from the "BB" point still goes directly to the 1/4" headphones jack.

Modification #3: Provide a "100 KHz Calibrator."

It is handy to be able to check the accuracy of the HW-8 frequency dial. For this, it's nice to have a pushbutton on the back panel which, when pressed, enables a "100 KHz" marker.

This particular marker generator uses "CD4000 series" logic, which can be powered directly from the "12V" supply. If a small terminal "5v" regulator was used, a comparable generator could certainly be made using 74HC logic. The advantage of using "CD4000" is that the logic can be left wired up to "12v," with the pushbutton merely enabling the quartz crystal oscillator.

In construction, I used "dead bug" assembly on an unetched single-sided piece of PC circuit board.

In building this "marker generator" you may find that it is useful to have an oscilloscope.

The inputs of any unused gates should be tied to either the chip's Vdd supply pin, or tied to ground. If you are not including a uA733 buffer amplifier (to drive an external frequency counter), then tie the CD4069UB chip's "Pins 9, 11, 13" to ground. Also, if you are not going to do the "full break-in" modification, tie the CD4093 chips "Pins 1, 2, 5, 6, 12, 13" to ground.

Part	Jameco PN
CD4069UB	849761
CD40174	893582
CD4093	13400
CD4518	13565

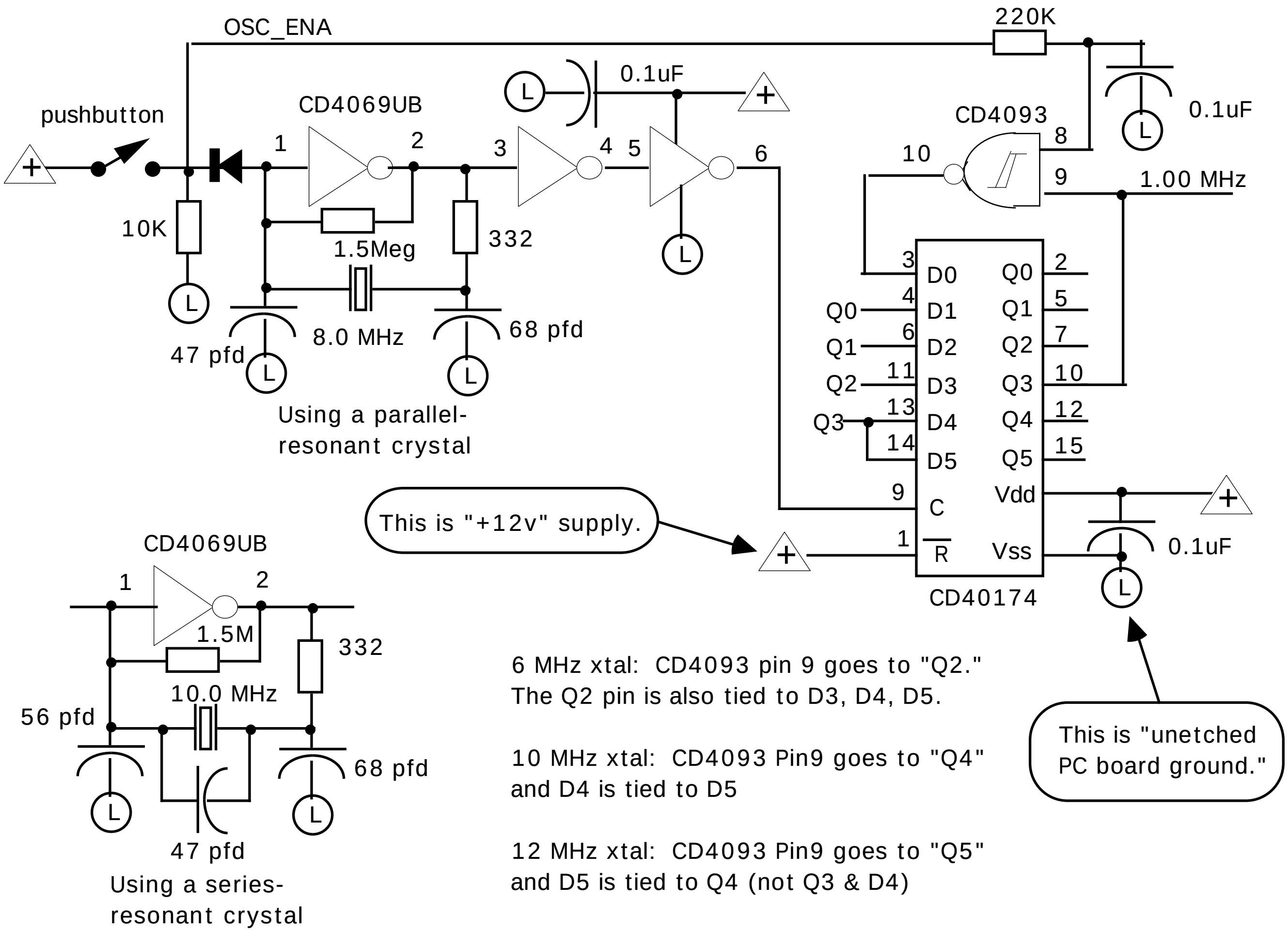


Figure 6A: Crystal Oscillator and "First Divider" for "100 KHz Ref"

When selecting a crystal, try to select a crystal that was specified for "Parallel Resonance" operation. The "single logic inverter" oscillator circuits (two examples shown in Fig 6A) operate with the crystal in parallel resonance. A crystal that is specified for "parallel resonance" will be at its nameplate frequency when used in a circuit like that with the 8 MHz crystal. However, if the crystal is specified for series resonance, then its parallel-resonant mode will tend to be at a higher frequency than the "nameplate," which then requires some additional capacitance directly across the crystal (as shown in the "10 MHz" circuit). For my "10 MHz in series resonance" crystal, when the circuit was lacking the extra "47 pfd capacitor, the frequency was around 3 KHz too high. With that "47 pfd" cap added, the frequency was around 1.5 KHz too high.

Margins: Old-timers may remember that CD4000 logic runs faster as the supply voltage is raised. Therefore, safety margins can be checked by reducing the supply voltage below the "nominal" level.

With an 8.0 MHz xtal, Johnson counter works down to "supply = 7.0 VDC." With a 10.0 MHz xtal, the Johnson counter works properly with supply voltages as low as 8.5 VDC. (The counter worked with a 12 MHz xtal at "12 VDC," but I didn't check the supply voltage margin.) Note that the CD4093 schmidt trigger is needed to insure that all "counter" FFs start out in the same state (in this case, all FFs will be initialized to '1') before allowing the "ring counter" action to commence.

The Johnson counter provides a 1.00 MHz square wave. In my HW-8 this is applied to a CD40192 for a further "divide by 10." The "Borrow" output is used as the "marker" feed because it is highly asymmetrical.

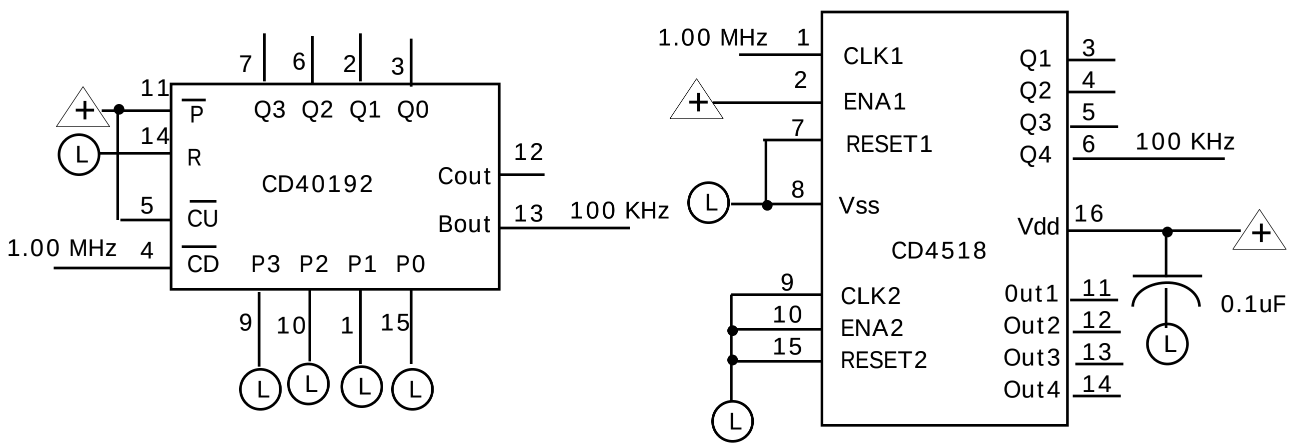


Figure 6B: Two "Divide by 10" Circuits for "100 KHz Reference" Generator

Modification #4: Provide an output to drive an external frequency counter.

The 14-pin uA733 differential amplifier and the 8-pin TL592 have been obsolete for around 15 years, but you can still find DIP-package parts without difficulty if you hunt around.

This "differential input / differential output" chip is designed for "12v supply" operation. Normally it was powered from "+/- 6V" dual supplies, but for our purposes it is a good "fit" as a buffer for driving an external frequency counter.

The drawback is that this chip consumes nearly 20 ma of supply current. However, this is made manageable by setting up a timer so that the chip is powered for around 15 or 20 seconds after the back panel "crystal calibrator" button is released.

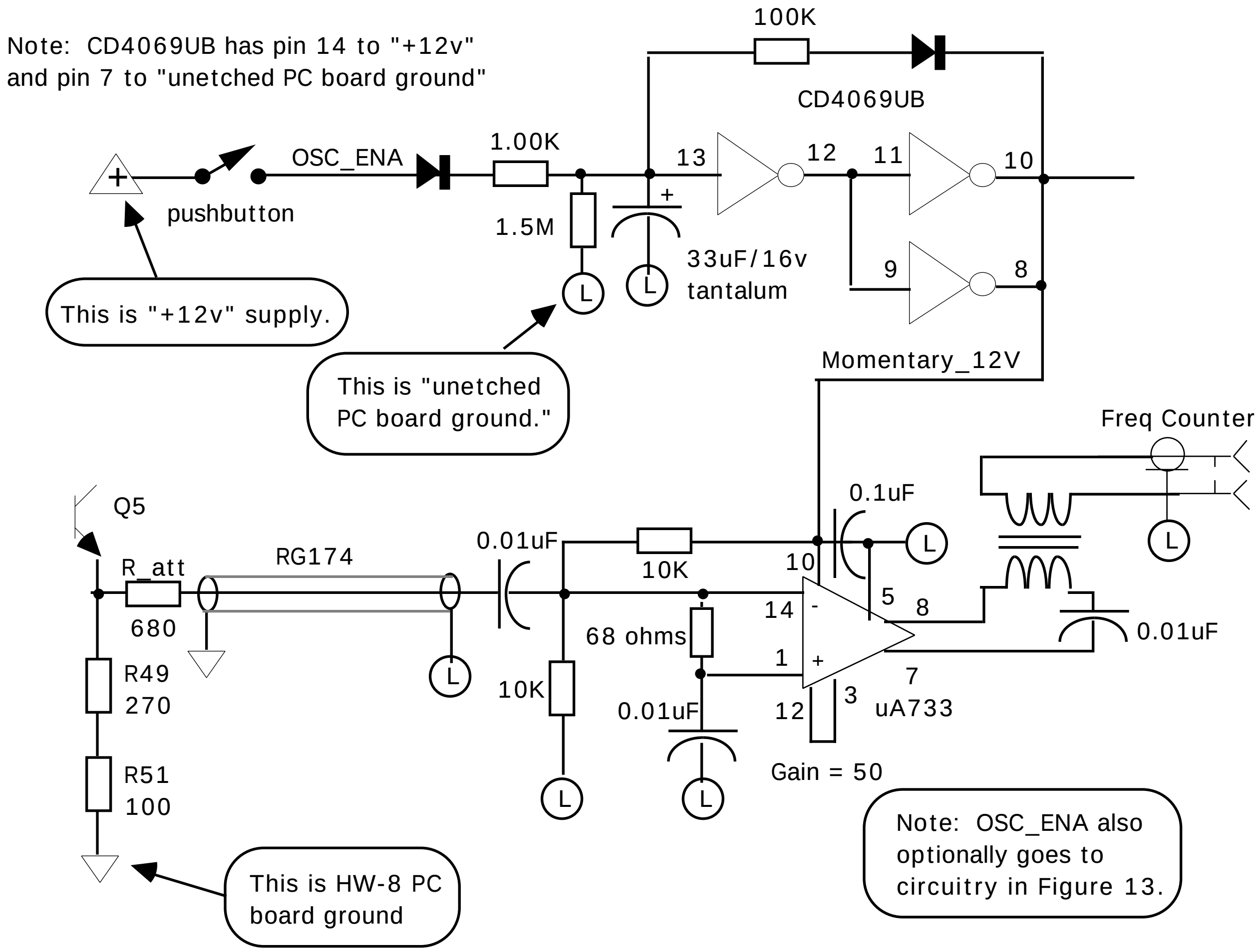


Figure 7: Buffer Amplifier to drive External Frequency Counter

The "R_att" attenuation resistor is chosen so that the AC voltage at either Pin 7 or Pin 8 of the uA733 is around 2 volts pk-pk with the chip wired u for a "differential gain" of 100. The chip is operated as if it has a "+/- 6 volt" supply by having both inputs held at a "midpoint" level by the pair of 10K resistors.

My frequency counter is "on the ragged edge" of operating properly if a 3-foot-long skinny coax cable is used to connect it to the "Counter" jack on the HW-8. So, a small ferrite toroid (around 0.5 inches outside diameter) is set up as a 1-to-1 transformer (using two strands of 28 AWG bifilar, a total of around 25-30 turns through the toroid opening). The 0.01uF cap is needed between the transformer and the uA733 outputs due to a DC offset voltage that exists between Pin 7 and Pin 8.

Modification #5: Provide improved audio band-pass filtering

The HW-8 was designed with simple bandpass filters that provide a "6 dB per octave" roll-off on either side of resonance. When operating in the "Narrow" mode, a loud station that is several kiloherz away ends up being attenuated by only "12 dB per octave" away from resonance. In addition, the original bandpass filters have poor audio dynamic range - they are easily driven into distortion by strong signals, making it difficult to hear adjacent weak signals.

The revised circuitry provides 18 dB/octave roll-off on each filter section. When operating in "Narrow" mode, if a strong station is well-separated from the passband, it is attenuated at 36 dB/octave. In addition, the new circuits have very good audio dynamic range.

The original HW-8's "wide" BPF has Q = 1 and Av = 5, and the "narrow" BPF has Q = 2 and Av = 1. In the original circuit only one output of the HW-8's product detector is used. If the HW-8 was powered from a "plugged into the wall" supply, residual supply hum could find its way into the audio signals.

In this redesign, the MC1496 product detector now feeds a *differential* amplifier so that we reject most of any residual supply hum that may leak through from a "wall-powered" supply.

The "wide" filter is now a "1dB Chebyshev" 2.5 KHz LPF feeding a 200 Hz "1dB Chebyshev" HPF. For "Narrow" bandpass, the "fc = 550 Hz LPF allows for a received cw signal to shift by "three musical half steps" above "A 440 Hz." (When the HW-8's VFO is powered by ZD1 9.1v zener and R33 and "12v" is from a battery, my HW-8 shifted its tuning by this amount after a long transmission.) The "350 Hz" filter allows for "four musical half steps" below the "A 440 Hz" tone.

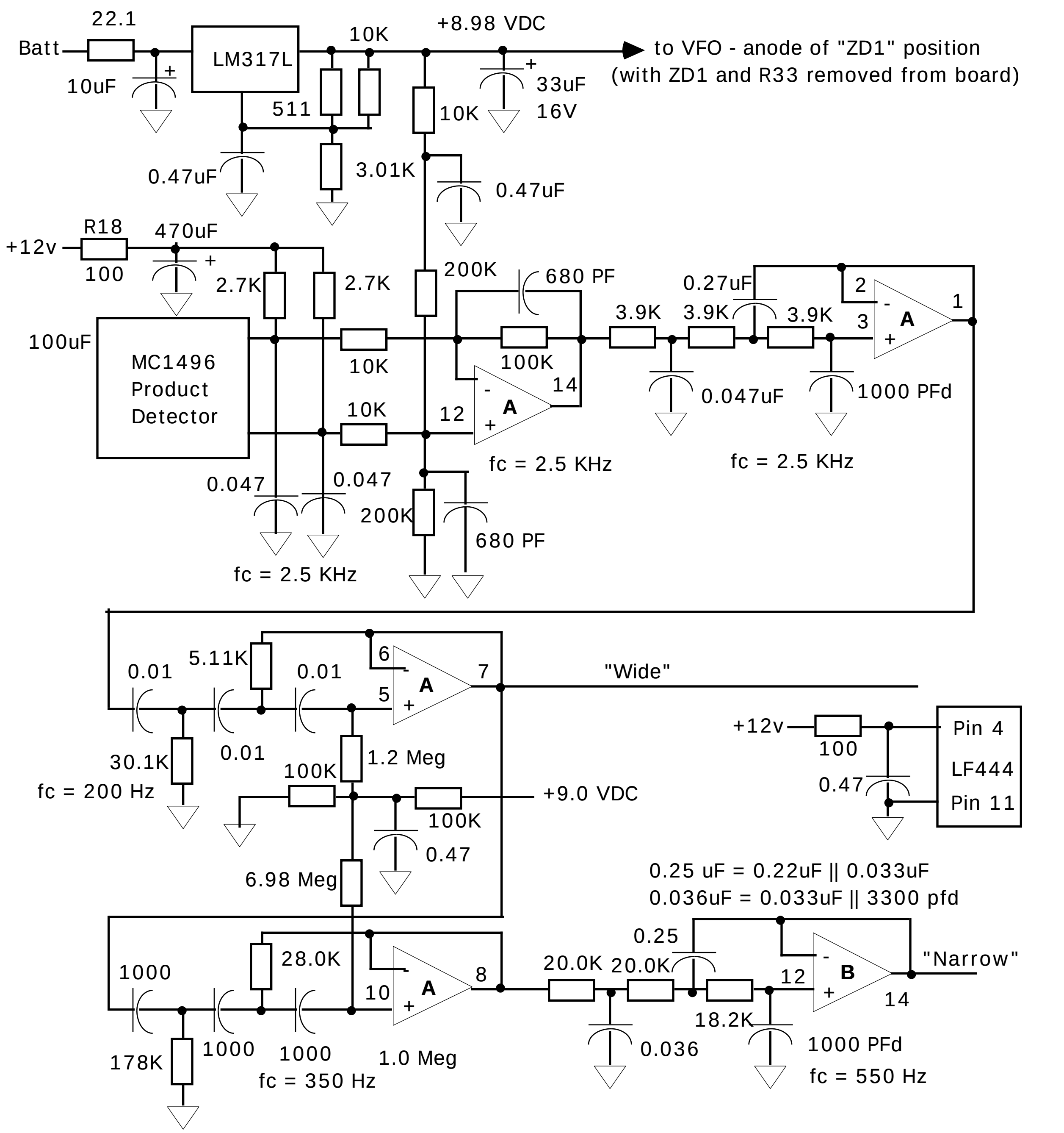


Figure 8: "Improved Audio Filter" circuits

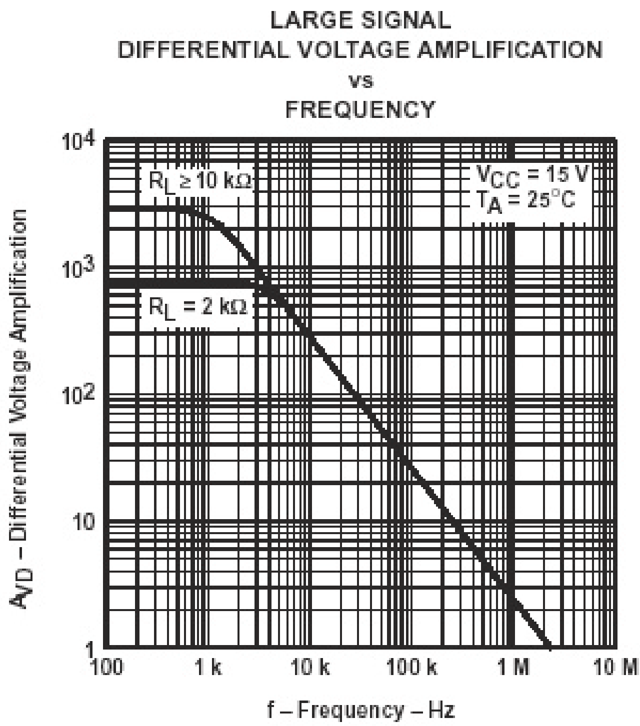
Filters per "Stout & Kaufman, Handbook of Operational Amplifier Circuit Design" ISBN 9780070617971

To minimize any "tuning shift" that occurs (during transmitting while using "battery power"), the HW-8's VFO is now powered from a "9v" supply that coming from the LM317L regulator. (With the resistor values in Figure 8, the output on my particular LM317L is 8.98 VDC.

With no load (not hooked to the VFO etc), the "8.98v" supply starts to drop (goes down to 8.95v) when the battery drops to 10.8 VDC. Normal operation with an LiPo battery calls for holding the battery above 11.5 VDC.

Designing the new "audio gain" circuitry:

The original HW-8 has only one amplifier stage providing high gain. (This "high gain" makes it difficult to achieve "click-free" QSK). We can estimate the gain of the HW-8's "original design" high gain stage from the LM3900 data sheet. The amplifier has some DC feedback for biasing the output, but otherwise is running "open-loop."



Open-Loop Voltage Gain of LM3900 (data sheet's Figure 3)

The LM3900 data sheet lists the low-frequency voltage gain as 2.8 V/mv. This "Fig. 3" curve shows that, at typical audio frequencies, the gain is around 2500.

In the original HW-8 the over-all gain from the MC1496 to the "audio gain" pot is
(gain of wide filter = 5) x (2500 in hi-gain stage) = 12,500

In the reworked audio string we might target the over-all gain as
(gain of MC1496 & diff ampl = 20) x (gain of hi-gain ampl string) = 12,500 -->
high-gain ampl string's gain = 625 if original gain is adhered to.

Actually, we get adequate gain when the cascaded gains of the "**B**_pin_7 and **B**_Pin_8" stages is around 250. (The "**B**" notation refers to "IC number **B**".)

For each of the new audio amplifier stages: Feedback 'tau' sets cutoff = 15,000 rad/sec = 2.3 KHz.

First ampl has gain = 68K/(2.2K + 3.3K) = 12.3
Second ampl has gain = 68K/3.3K = 20.6

Before going to the front panel switch, both "Wide" and "Narrow" signals are passed through RC low-pass filters so that stray RF (picked up in the wiring between the perfboard and the Wide/Narrow switch) does not get into the output stages of the "**A**_Pin_1" and "**B**_Pin_14" op-amps. The stray RF also doesn't get to the input of the "**B**_Pin_8" op-amp.

If the "front panel audio pot" is wired between the second stage and the "audio board" then strong signals will push the output of the "**B**_pin_7" amplifier into clipping. Therefore, we put the front panel "audio" pot between the first stage and the second stage.

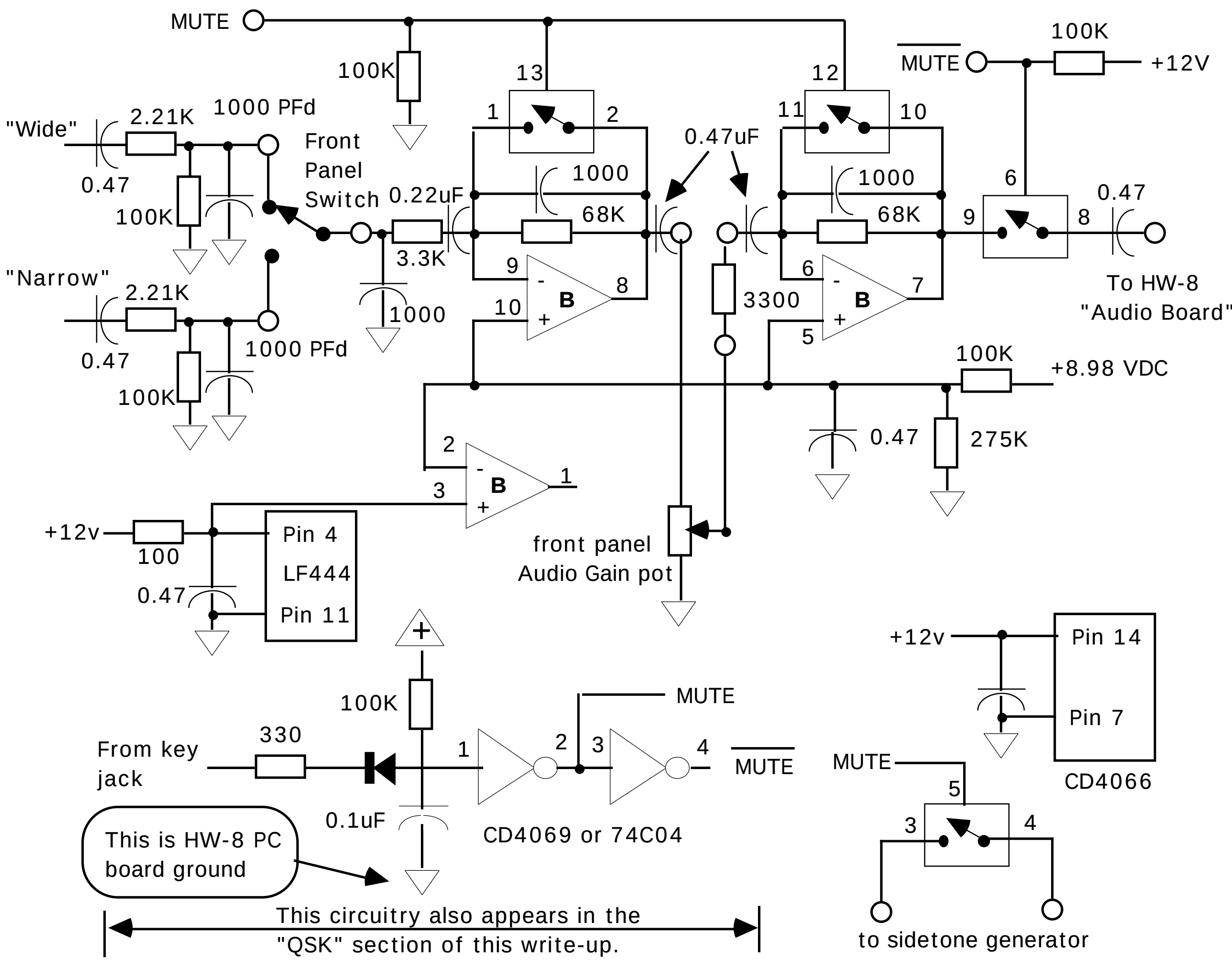


Figure 9: High-Gain section of Reworked Audio Amplifier for HW-8

At the front-panel "audio gain" pot, the pair of 0.47uF caps form a one-pole HPF with cutoff = 270 Hz.

The output of an LF444 can swing between 2.5v and 10.5v. The average of these is 6.5 volts, which is the "bias point" of the "+" inputs of the "Av = 12" and "Av = 20" op-amps.

If the HW-8 is not going to be adapted for full-break-in (QSK) then the CD4066 can be omitted. However, the CD4066 works well with these stages to provide muting during "key-down." The "R_on" of CD4066 is 400 ohms max. --> when muted, total gain of the two amplifiers is less than 0.022

The 100K pull-up and pull-down at MUTE and $\overline{\text{MUTE}}$ (CD4066 pins 5,12,13 and pin 6) allow a quick installation into the HW-8 for evaluation without a working QSK break-in system being in place. The "IC **B**" "Pin 2 and Pin 3" hookups to "6v" and "12v" keep the unused ampl. "Pin 1" output from oscillating.

Physical construction notes for the new audio filters & high-gain ampl:

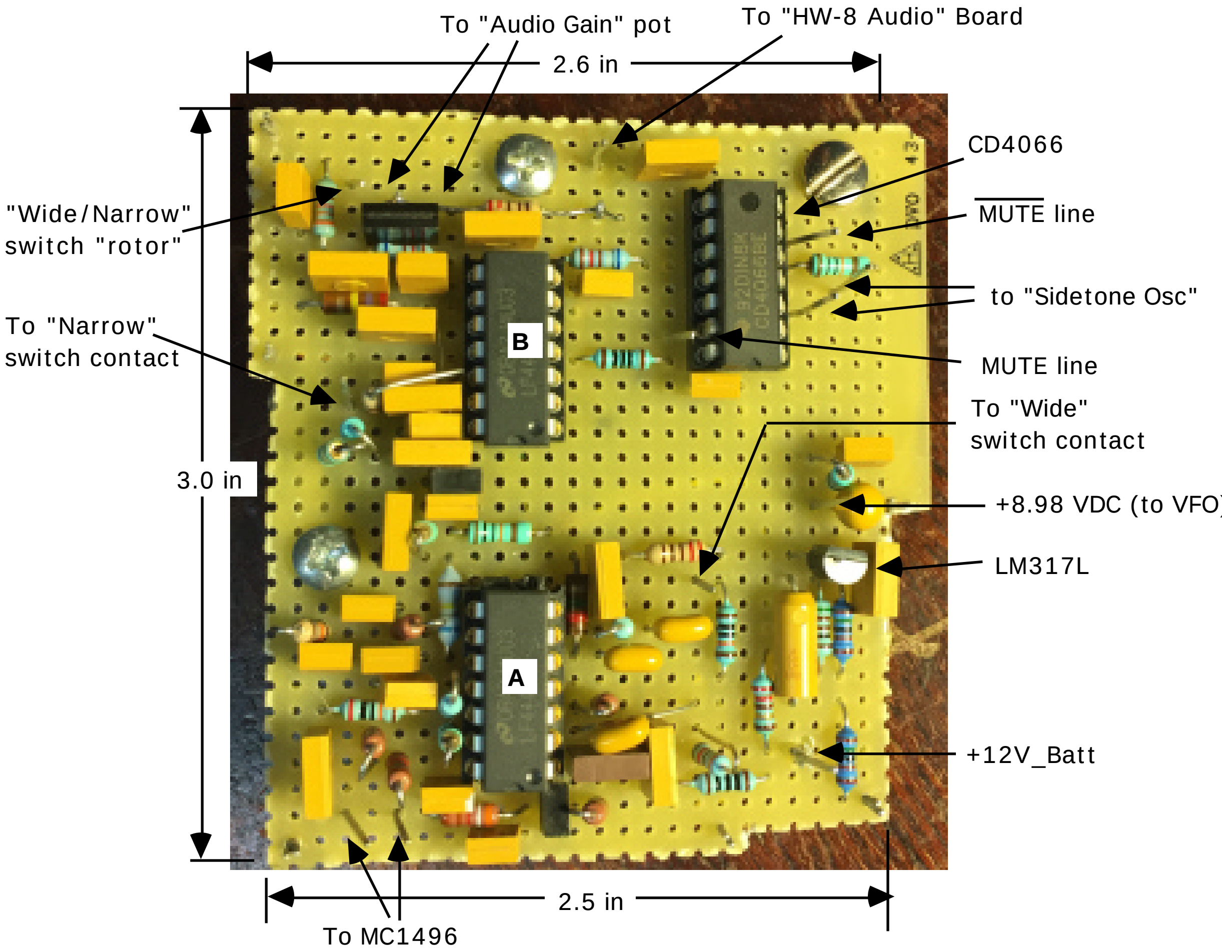


Fig 10: Showing the "Perf Board" construction of the "Audio Filters and Audio Gain" Circuits (This board holds circuitry of Figures 8 & 9) Chips "A" and "B" are both LF444 (Texas Instruments)

The audio assembly is built on a 2.6" x 3.0" piece of perfboard. Notches were cut out of sides of this particular piece of perfboad to provide mechanical clearance to tuning caps, clearance to the front panel "Audio Gain" pot, and clearance to the original audio board itself.

The perfboard is held onto the side panel of the HW-8 via three aluminum F/F 6-32 hex spacers. Before putting any components on the perfboard, the board is placed against the outside of the side panel, and its three mounting holes are used as templates for drilling 9/64" holes into the side panel. (Have the HW-8 in an orientation where the drill shavings do not fall onto the radio's circuit board.)

Building the perfboard: A bare wire is routed around the periphery of the perf board. This wire is soldered to the "solder lugs" at each mounting hole - this ring of wire provides easy access to "chassis ground" for perfboard circuitry. (A solder lug is on the backside of the perfboard at each of the three mounting holes.)

Before installing the perfboard into the HW-8, it's worth checking the functionality of the high-pass and low-pass filters (with a function generator and oscilloscope), plus checking the functionality of the "high gain amplifier" stages.

It will be necessary to remove the three green "0.1uF/100v" caps from the HW-8 circuit board. These caps will mechanically interfere with the bottom edge of the perfboard (near the two points that go to the MC1496 outputs). On the next page it will be noted that these caps (C26, C27, C31) have been pulled from the board. C26 and C27 have been replaced with 0.1uF/50v ceramic caps - these caps are on backside of the board.

View of the HW-8 Side Panel:

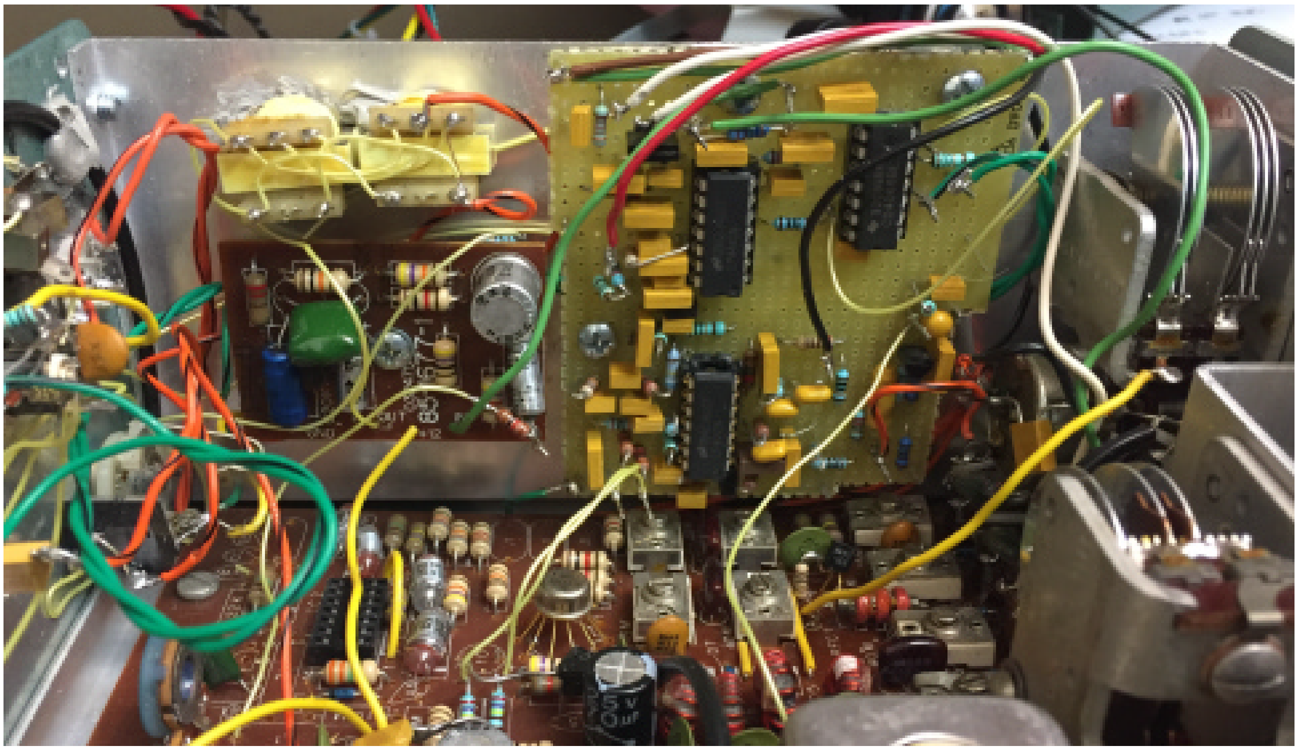


Figure 11 - Perfboard assembly is installed in the HW-8

Note how one edge of the perf board was "notched away" to provide mechanical clearance to the on/off switch assembly (on the back end of the audio gain pot). The other edge of the board was slightly notched out to provide clearance to the "Audio Amplifier" board.

The original "C32" (100 uF electrolytic) has been replaced by a 470 uF/25v electrolytic.

The "2.7K 1/2W 5%" resistors at R16 and R17 have been removed, and replaced with 2.75K 1/4W 1% resistors (mounted partly on end, with the long leads being the ends of the resistors going to the MC1496 chip). The original C29 has been replaced with a 0.047uF/5% cap, and a 0.047uF/5% cap is also soldered from "long lead of R17" to the "ground" side of the "C29" 0.047uF cap.

Remove the 10uF cap at C33. Remove the LM3900 from its socket.

A "twisted pair" set of wires connects the "MC1496 side" of R16 & R17 to the "differential input" pins of the perf board assembly.

The cathode end (banded end) of ZD1 has been lifted from the board. The corresponding end of R33 (470 ohms 1/2W) is also lifted up from the board. A wire runs from the perfboard's "+8.98v" pin to the PC board's hole for the cathode end of ZD1.

The "Wide/Narrow" switch's cable can have its vinyl jacket stripped back until the jacket is close to the perfboard assembly. Unused wires (Brown, Green) can be soldered to a "ground" pin on the perf board. The other wires (red, black, white) are soldered their connection points on the perfboard.

The jacket for the "audio gain pot" cabling is completely stripped away. The pot wires go directly to the perfboard assembly. A short wire brings "+12v battery" from the "On/Off Switch" assembly over to the perf board (orange wire with black tracer).

Modification #6: Provide "Full Break-In" operation (QSK)

Modifications #1, #2, and #3 are fairly straightforward, and can be done even if you don't have a lot of experience with building electronic circuitry.

It's strongly recommended that you build and install Modification #5 (the improved audio bandpass filters) before you move onto installing this modification. Yes, you can try to "make do" with the original HW-8 "LM3900" circuits (per notes in the appendix), but the results are less-than-ideal.

This modification should only be done if you have a lot of confidence or experience. If this is your first time taking on a project like this, it would be helpful to have an "Elmer" to offer guidance.

You should have access to a good-quality solder sucker, a good oscilloscope (at least 2 channels), and a function generator.

Before beginning this modification, you need to obtain the key item: a SPDT reed relay with a coil voltage of somewhere between 5 volts and 12 volts. The Coto model 7141-12-1001 is a suitable choice (Mouser catalog 816-7141121001, price around $22). This is a "dry contact" relay.

At the time of this writing (2022) Jameco offers a "dry contact" SPDT reed relay at modest cost. If you have the chance, avoid "mercury wetted contacts" and use a "dry contact" reed relay for the "QSK" modification.

If you find a possible relay for sale at a swapfest, you can confirm "mercury wetted" by shaking the relay - you'll hear the mercury droplet rattling from end to end.

> *A cautionary note: Mercury-wetted relays are sensitive to the relay's mounting orientation.* These relays are marked with an arrow saying "UP" for a reason. These relays work great if that arrow is pointed "skyward," and they generally work properly when they are horizontal. However, when upside down (so that the "UP" arrow is pointing towards the floor) the "NO" and "NC" contacts become shorted together. This will be very bad if you've got the rig upside down for testing, and you key up the transmitter!

One potential advantage of mercury-wetted reed relays: Relays can develop a problem where a contact is "mechanically closed but electrically open-circuit." If this happens, it is due to surface (oxide?) contamination on at least one of the contact surfaces. But it may be possible to circumvent the issue if the reed relay is temporarily oriented upside-down (so that all contact surfaces are bathed in mercury) and the relay coil is energized (to expose the "Receive" contact surfaces to the mercury). This must be done *without keying the transmitter*, but the "OSC_ENA" diode in Figure 13 can allow you to do this.

One other cautionary note: Many reed relays have a built-in "catch diode" across the coil. (The Coto coil referred to, above, is one of these "diode included" relays.)

If you have found a possible relay at a hamfest or flea market, you might want to check its operation by driving the coil with a function generator, driving the "COM" contact via a small battery, and watch voltages at the contacts with an oscilloscope.

This shows how I tested the relay which I then used on my HW-8:

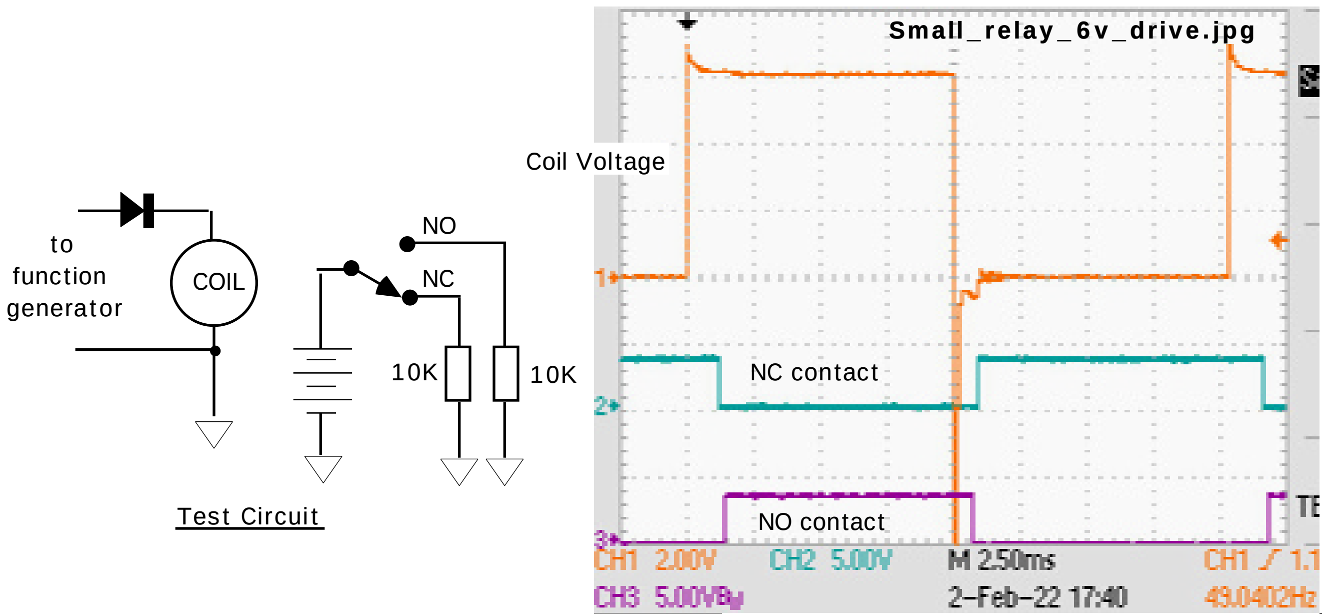


Figure 12: Testing a "No Name" SPDT reed relay

In running this test, I found that (a) the coil should have at least 5 volts of drive - it wouldn't switch at all if coil driven to 4 volts. And, (b) when the coil is driven "hard" (at 6 VDC) the actuation time is quite fast, at 1.5 milliseconds. Finally, I was able to confirm that, from the standpoint of making acoustic noise, my relay was really quiet. (My rig's operating desk is just off from the bedroom, and a quiet setup is necessary if I'm trying to have a QSO while my wife is sleeping.)

Preliminary modification steps:

So, let's assume that you have found a suitable reed relay, and you have become acquainted with its operation by operating it in the test circuit of Figure 12.

Your next step is to build, and test, the new "relay driver and transmit keyer" circuitry, as shown in Figure 13. I built my circuitry "dead-bug style" on the piece of unetched single-sided PC board.

The "Schmidt trigger NAND" gates are CD4093 (you might have already used one gate of this chip in the 100KHz Calibrator). Diodes are 1N4148. If your reed relay has a 12V coil then you should omit the "330 ohm" resistor that is shown in series with the coil.

QSK Circuitry (Note - an editable version of this is in Appendix B at the end of this document):

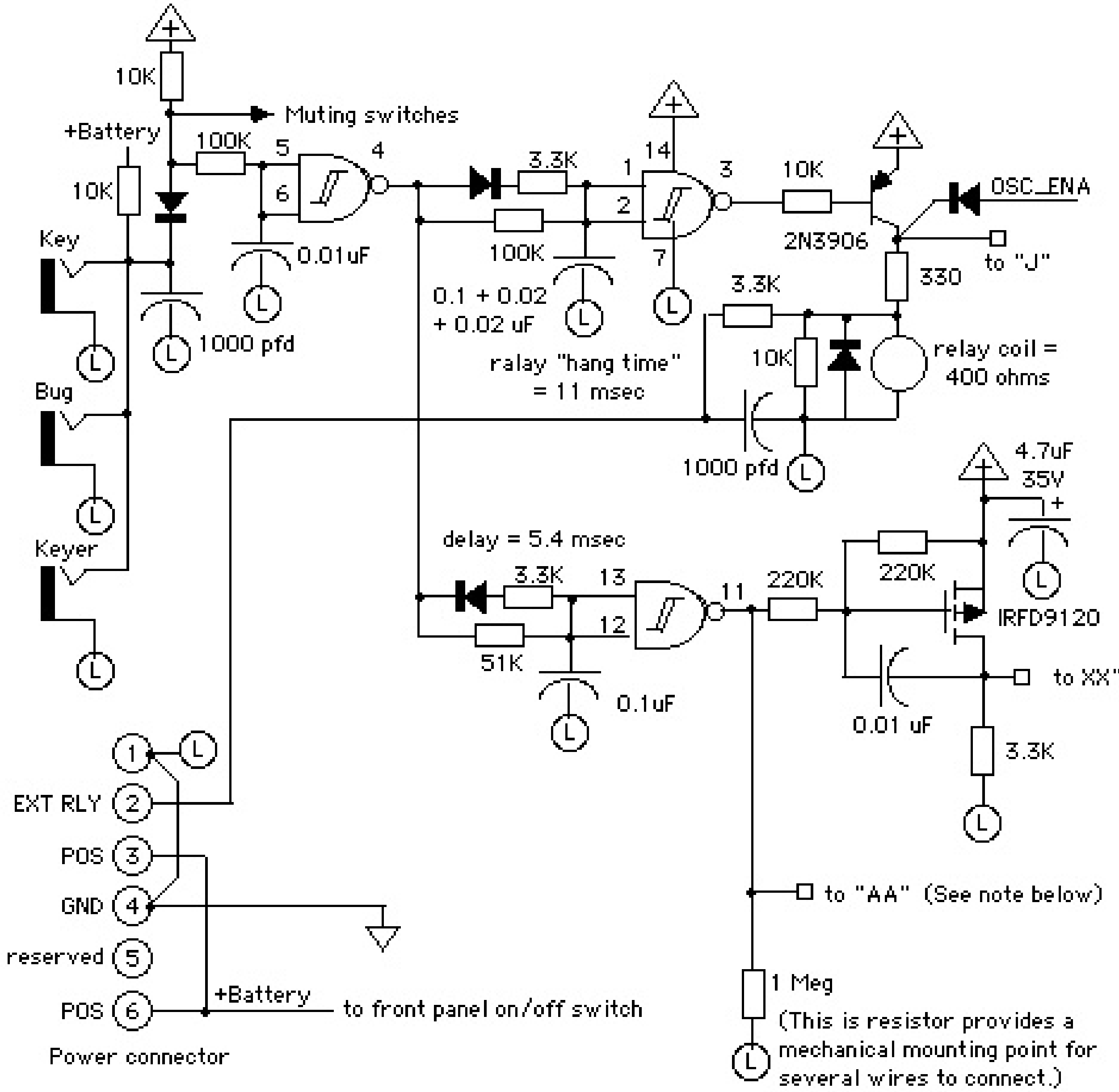


Figure 13: The new "Break-In Control and Transmitter Keying" circuitry

Note 1: The "AA" hookup is done if (and only if) you are using the LM3900's sidetone oscillator. If you are building a new sidetone oscillator (per Figure 17) then you do not need to run a wire to the "AA" point on the HW-8 circuit board. **Note 2:** The "OSC_ENA" line and diode allows you to exercize the reed relay (with the rig physically oriented upside down) to "re-wet" the relay contacts. Try this if the receiver appears to be dead after doing a key-down transmission ("RCV" contact is mechanically closed but electrically "open"). *If you do this, first disconnect the key from the rig!*

A comment: As part of the work on my HW-8, I removed the HW-8's original power connector (Molex), widened the hole slightly, and installed a Tyco/AMP PN 350711-1 connector, using "JB Weld" epoxy. The new power plug is Tyco/AMP PN 350715-1. This was strictly due to my personal preference.

My "power connector" has Pin 2 assigned to providing a control signal for additional "relay driver" circuits that might be in a small external power amplifier.

After building up this circuitry, I tested it using the oscilloscope.

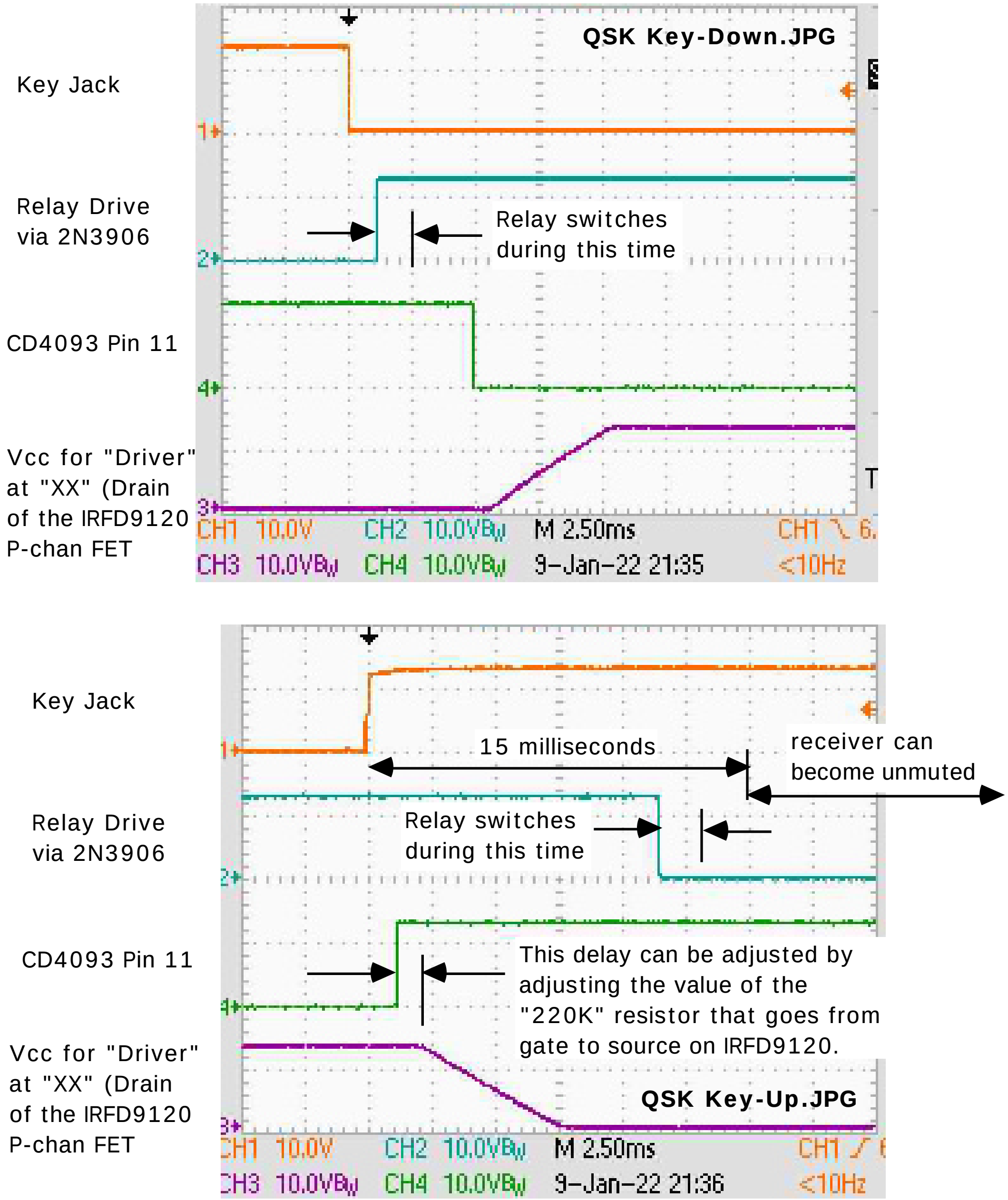


Figure 14: Behavior of new Keying Circuit at Key-Down and Key-Up

In my implementation, the P-channel FET has a turn-on threshold of Vgs = -2 to -4 volts. If you find it awkward to purchase the IRFD9120, then an equivalent is an IRFU5305 (Jameco PN 2308439). If your junkbox has a "logic threshold" P-chan FET (such as BS250P - Jameco PN 2294935), then the smaller turn-on threshold requires a shift in the gate resistances. The gate-to-source resistor drops from 220K to something around 180K, and the "NAND Pin 11 to Gate" resistor will be around 330K.

As noted in the "Key-Up" waveform, there is a bit of delay between the "CD4093 Pin11" signal going high and the "FET driver power" signal starting to ramp downwards. The "Key-Up" waveform shows what is probably an optimal amount of delay (around 1 to 1.5 millisecond). The fact that this delay exists (is not zero) is an indication that the FET is being turned on adequately for providing the 15ma needed by the HW-8's Q8 driver. If this delay is around 5 msec or longer, then the "gate-to-source" resistor's value should be reduced until a delay of 1 or 2 milliseconds is seen.

At this point you can use "clear silicone rubber" to glue the reed relay onto the sidewall of the HW-8. Find some "moderate mechanical compliance" plastic foam cushioning to form a layer between the relay body and the aluminum sidewall of the HW-8. If you can find some "pink ESD" foam (such as is used to package static-sensitive components), that material works well. Your goal is to have a mechanical cushion that can keep the HW-8 sidewall from acting as an acoustic sounding board.

> If your reed relay has an "UP" arrow, make sure that the arrow actually points "up" away from the HW-8 circuit board.

If you wish, now is a good time to yank the back panel's "RCA jack," enlarge the hole, and install a BNC connector for the antenna. You can also disconnect the hookup wires from the "KEY" 1/4" jack, and remove the fiber washers (so that the threaded bushing of the jack now goes to "panel" ground)

After the relay's glue has cured, we come to the "take the plunge" step:

Make sure that the solder sucker is clean and working well. Then, use it to remove the "K1" relay, and all of the circuitry in the "Keying," the "Break-In Delay," and the "Relay Driver" circuitry. (This is Q11, Q12, Q13, and associated resistors and capacitors). The 2N3906 collector is wired up to the relay's coil, and the relay's "COM" point is wired up to the antenna connector. The "NC" contact goes to the "M" point on the board (coax to the receiver's input), and the "NO" contact goes to the "L" point (coax going to the bandswitch assembly).

Use an ohmmeter to confirm that, with the HW-8 sitting on a table top, the relay's NC and NO contacts are NOT shorted to each other. (This is a potential problem if you are using a mercury wetted relay.)

At this point in the modification (before building the "audio muting" circuits of Figure 16), your "headphones audio" is going to have very loud clicks and thumps during keying.

If your oscilloscope has good bandwidth, you should be able to confirm that the transmitter's keying characteristics look nice.

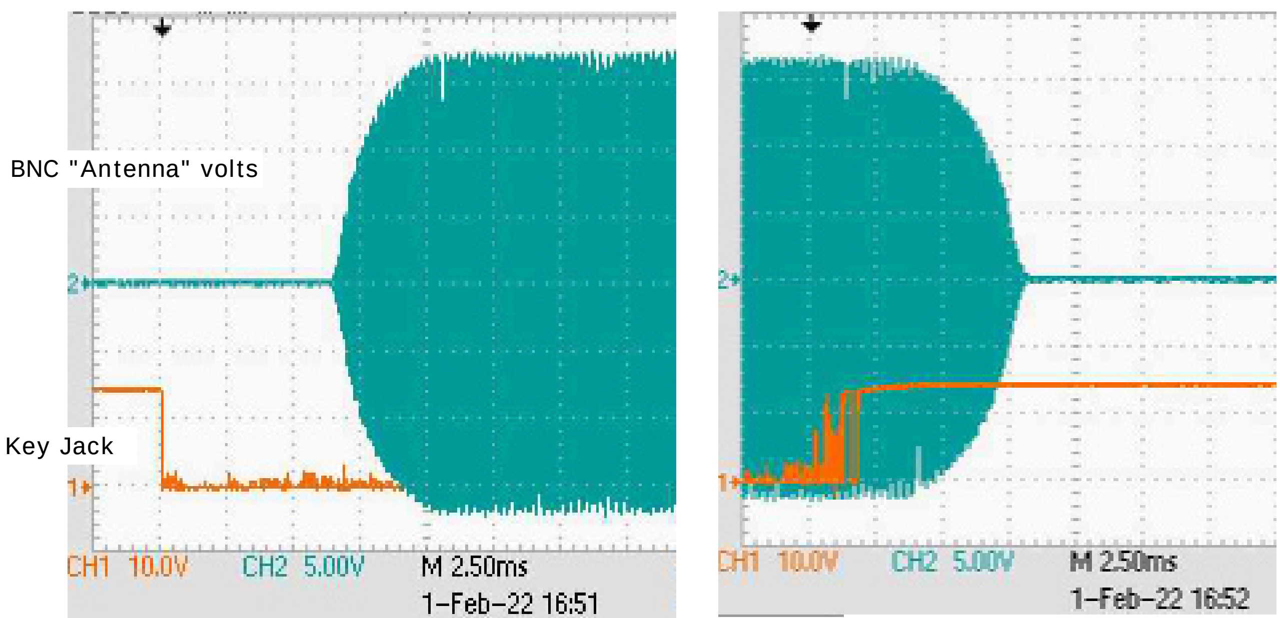


Figure 15: Signals (Key-Down and Key-Up) at 7 MHz
(17v 0-pk --> 2.9 watts to the 50-ohm dummy load)

Note: Doing the same "key-down / key-up" test on 21 MHz shows 15v 0-pk --> 2.25 watts to the load.

After the "proper keying sequencing" has been confirmed (per Figure 15), you can add the "audio muting" controls of Figure 16.

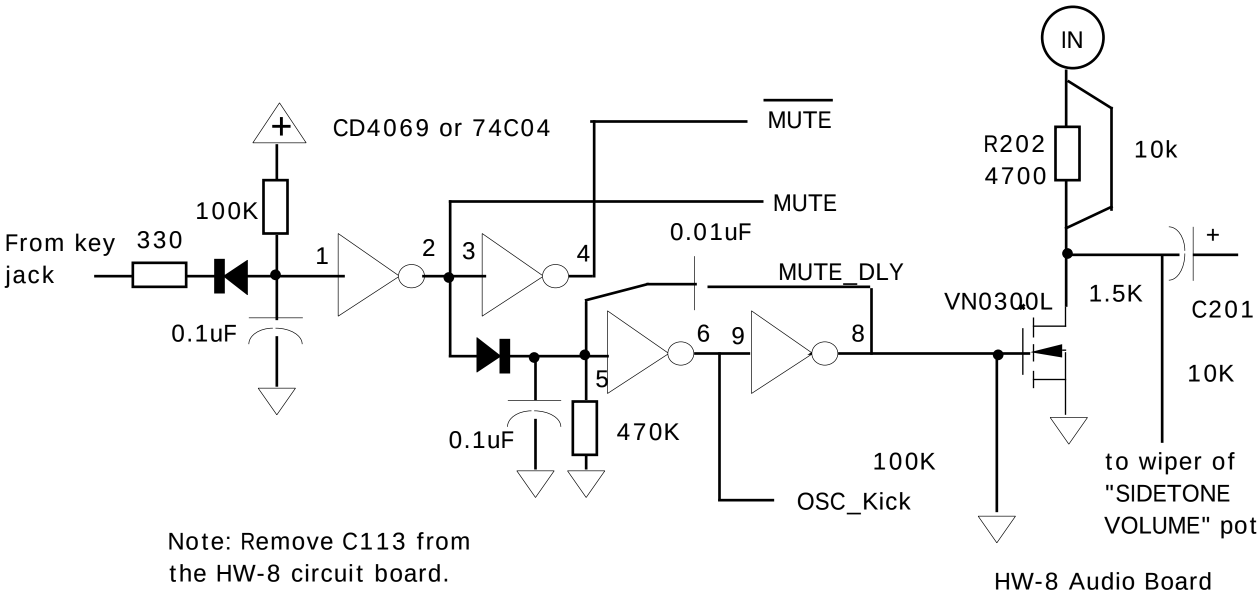


Figure 16: Muting Circuitry for Receiver Audio Path
The VN0300L, the 100K "gate pulldown," and "10K across R202) are on the side panel "audio board." This scheme requires a new dedicated sidetone generator.

I am showing the N-channel FET (and the associated "MUTE_DLY") circuits in Figure 16 because they are in my HW-8. But the FET (etc.) was added when I was attempting to deal with "clicks and thumps" in the original LM3900 circuitry. My suspicion is that you will not need the FET (nor the "MUTE_DLY" signal) if you first build up the "perfboard" assembly of Modification #5.

An important part of Figure 16 is that the sidetone oscillator's "audio volume" pot is now going into the HW-8's "headphones amplifier" on the "Audio Board." In other words, the sidetone pot is no longer being wired directly onto the headphones jack, which lets you set the "sidetone" volume at a comfortable level. As part of this, remove C113 (a 0.1uF 100v film cap) from the HW-8 circuit board.

Not explicitly noted in the schematic: The N-channel FET is tied to the "ground" of the little audio amplifier circuit board, and is not tied directly to the "ground foil" of the larger HW-8 board.

For the FET I used a VN300L but you could use a STQ1NK80ZR (Jameco PN 2308479) or some other "TO-92 package" N-channel part. I would avoid using a FET with a large chip (such as parts in TO-220 packages) because the gate-to-channel capacitance of this FET might inject a bit of "thump" noise into the audio amplifier, and we want to keep those residual thumps and clicks as small as possible.

In implementing the hack on the audio board, I added 10K in parallel with the R202 4.7K resistor, and I lifted the "negative end" of the 2uF cap at C201. This uplifted end of the cap is tied back to the board with a 1500 ohm resistor. And, precariously attached to the cap-resistor junction is one end of a 10K resistor. (The other end of this resistor will be tied to the new sidetone oscillator, which can sit on the piece of unetched PC board on the back panel.)

Sidetone Oscillator:

If you build the "perfboard" assembly of Modification #5, then you could *probably* continue to use the original HW-8 sidetone oscillator. Tie the $\overline{\text{MUTE}}$ line to the "AA" point on the HW-8 circuit board.

If you try to work with the original HW-8 bandpass filters and high-gan stage (per the Appendix), then you will need to build the separate sidetone oscillator. The LM3900 has significant "cross-talk" between amplifier sections, and if the oscillator section of the LM3900 is running it injects lots of crud into the "high gain" section, making proper "fast-action" muting difficult to obtain.

But for the cleanest end result, build the "perfboard" audio assembly, <u>and</u> build this sidetone oscillator. The frequency of the sidetone is very stable (does not vary with battery voltage), and the tone has a clean "attack" and "decay" characteristic. In other words, this sidetone oscillator's signal does not inject addition "click" noises into your headphones.

If you have built the "perfboard" and you build this oscillator, you can pull the LM3900 chip from its socket, which saves you 10ma in battery current.

If you try working with the original HW-8 bandpass filters and high-gain amplifier, then the "Receiver Muting" circuitry <u>requires</u> you to move the "sidetone oscillator" function out of the U2 LM3900 chip. To disable the LM3900 oscillator, remove the 0.022 uF at C109 and replace it with a jumper wire. (This ties U2 Pin 11 to ground.) Also remove R73 and R74 (the two 10 Meg resistors). The D21 diode can stay on the board because you have already removed the "Break-In Delay" circuitry involving Q11, Q12, Q13, and their related components. Also, while you are at it, remove C111, C112, and R76 (the parts that are between U2 Pin 10 and the "Sidetone Volume" potentiometer).

Our new sidetone oscillator should be able to produce a waveform that is a clean sinewave, at a frequency close to 440 Hz. It should be keyed "on" and "fully off," and it should inject virtually zero ripple into the "12v supply" when the tone is not being generated.

In addition, the oscillator should permit independent, non-interacting adjustments of
- frequency (via R_in or "R_trim" - make this adjustment first)
- decay time at key-up (via R_d - make this adjustment after finalizing "R_in" values)
- rise time at key-up (via R_osc - adjust this after setting "R_d" value)

The circuit shown here is a "Bi-Quad" arrangement of two integrators. Figure 17 shows the circuit with "tight tolerance" components (the "0.01 uF" caps are +/-2% NPO, and the "R_in" resistors are 1% film). Figure 18 shows a circuit which uses 5% tolerance resistors and 10% tolerance capacitors.

The "OSC_Kick" circuit speeds up the "oscillator start-up" process. At "key-down" the "OSC_Kick" signal falls from "+12v" to "ground," and the resulting "12 volts change on the 330 PFd cap" forces the output of the first op-amp to quickly rise to +0.4 volts.

The quad op-amp should be an LF444, although a TL064 would also work. Either part draws only around 1 ma of supply current from the "+12v" supply when the oscillator is muted.

The CD4066 switch appears in both Figs 17 & 18, and in Figure 9. It is "on" when key is down.

The 0.47uF "output" capacitor is mounted with the quad op-amp on the "single-sided unetched" circuit board that is on the back panel of the HW-8. From there, a jumper brings the signal to a now-unused pad of "C111" on the HW-8 board, bringing the signal to the 15K resistor at "R76."

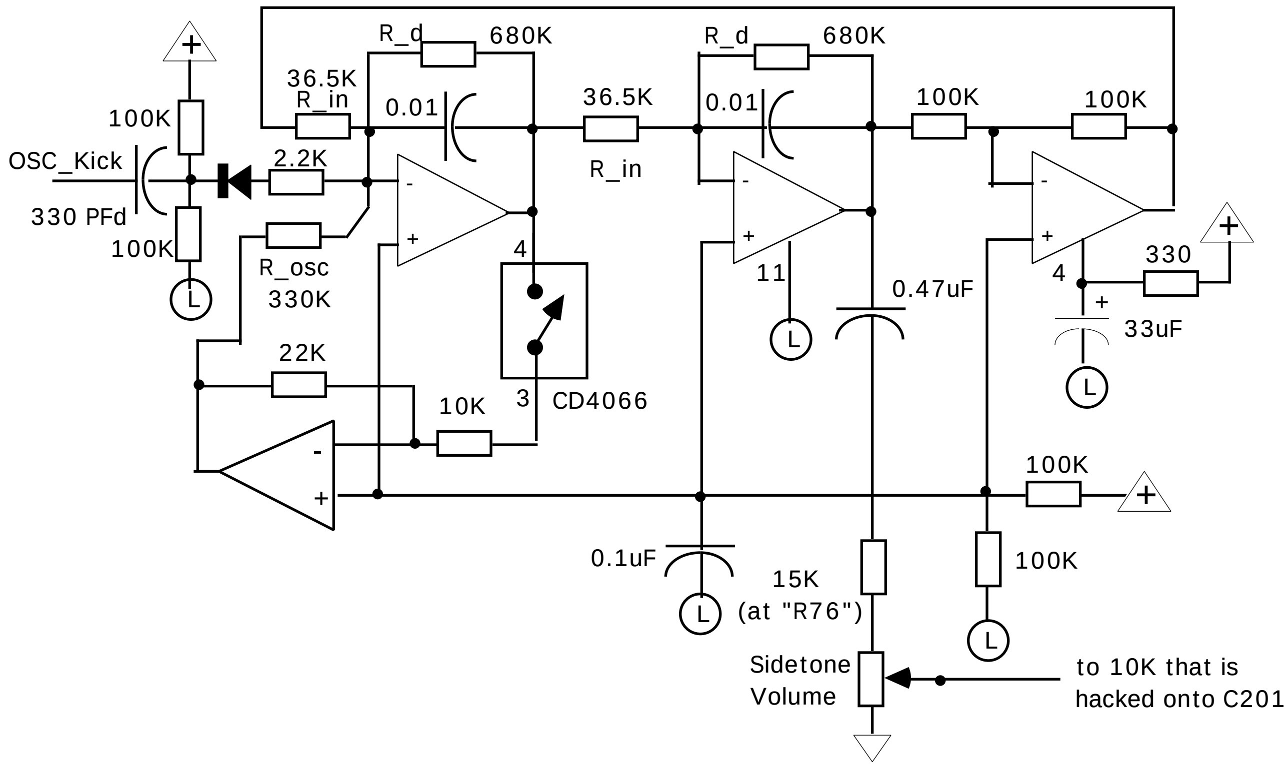


Figure 17: Sidetone Oscillator with "tight tolerance" values

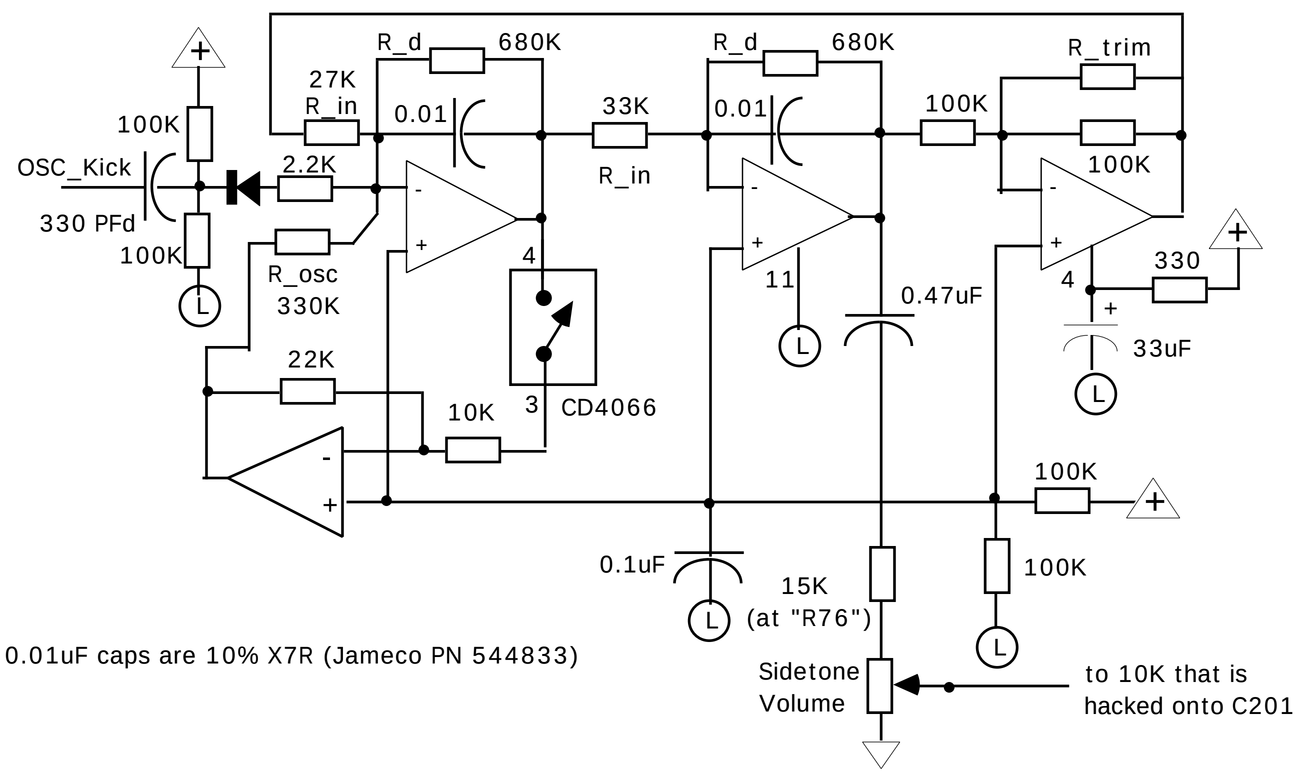


Figure 18: Sidetone Oscillator with "Commodity Tolerance Part" values

If a 30K resistor is available for the first op-amp's "R_in" resistor (instead of 27K), then use 30K.

With a 27K resistor at the first op-amp's "R_in" and nothing installed at "R_trim" the oscillator will probably be running at around 540 Hz. (With worst-case-tolerance parts, the frequency might be as high as 620 Hz.) Connecting a resistor decade box in place of "R_trim" you should be able to bring the frequency down to 440 Hz without difficulty. For the "540 Hz" initial frequency, the likely value of R_trim will be around 200K. Note that if you don't have "200K" in your junkbox but you have a 270K resistor and a 1 Meg resistor, then install those two in parallel. The result will be "good enough."

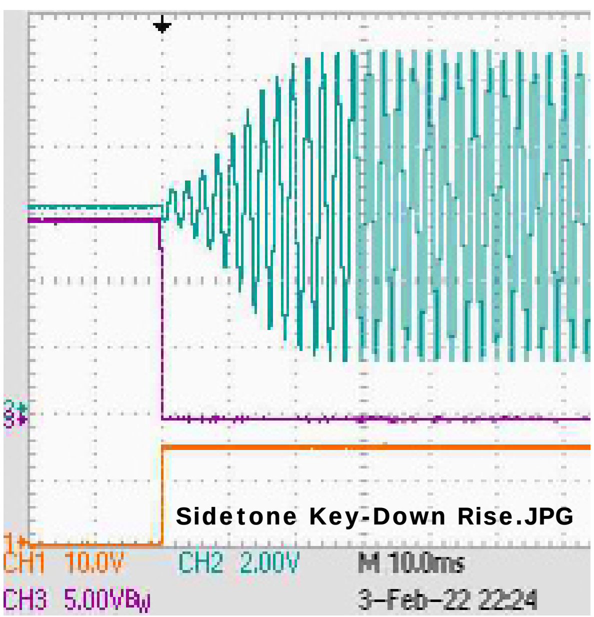
It is important that the two 0.01uF caps should not be "cheap." The best choice would be "COG" caps (also known as "NPO"). The COG capacitors always come back to having the same value after soldering.

The next-best choice is a pair of X7R caps. (Figure 18 references a possible "X7R" capacitor.) These capacitors will shift slightly in value after going through a "soldering" operation, but the amount of shift is "tolerable," especially in this application. Use capacitors that are rated for 50V or 100V.

Avoid using "physically tiny" capacitors with Y5V and similar dielectrics, and avoid capacitors that are rated for low voltages like "16V." These capacitors might be good for bypassing the "+5v" supplies on logic chips, but they will "jump" from one value to another value as they go through repeated soldering.

The resonant frequency of the "Bi-Quad" circuit is set by 1/(R_in x C) in "radians/sec" (assuming that the first two op amps use identical "R_in" and "C" values, and assuming that the third op-amp's circuit operates with a gain of -1.00). For 440 Hz = 2765 rad/sec we have an "R_in x C" product of 1/2765 = 0.362 milliseconds. If C = 0.01 uF then R_in = 36.2K, which is very close to a "standard 1% resistor value" of 36.5K.

<u>Behavior of the sidetone oscillator</u>:



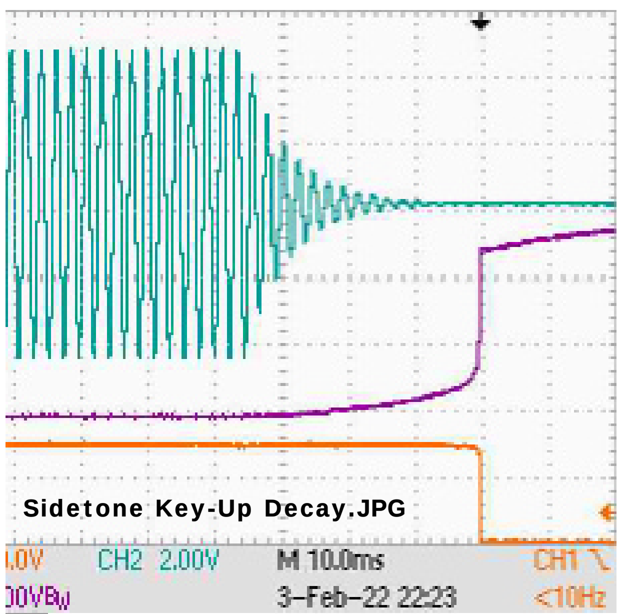


Figure 18: Sidetone "Turn-On" and "Turn-Off" (10 msec/div)
Yellow = "Gate Signal for FET on Audio Board
Violet = "OSC_kick" signal
Blue = Sidetone Oscillator's "2nd R_in & C" op amp - 2v/div

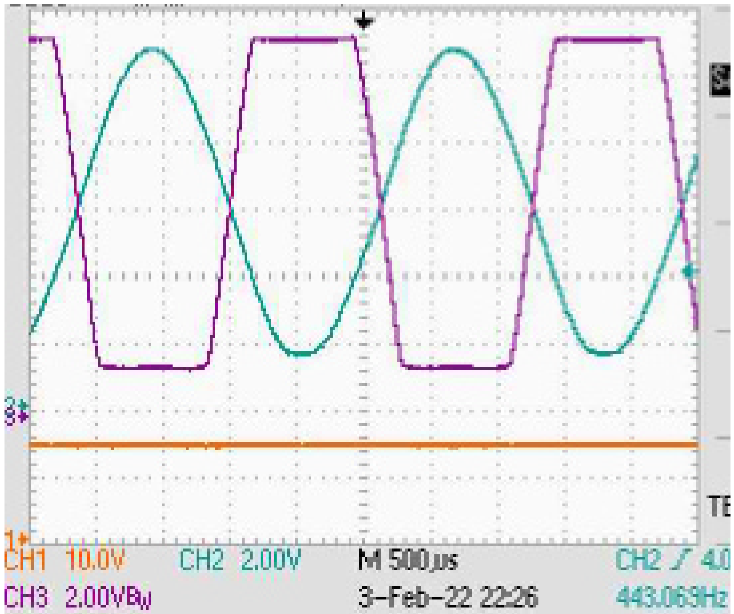


Figure 19: Sidetone Oscillator waveforms - Detailed View (0.5 msec/div)
Yellow = "Gate Signal for FET on Audio Board
Violet = "R_osc" amplifier's signal
Blue = Sidetone Oscillator Output 2v/div, Frequency = 443 Hz

Conclusions:

In implementing these modifications, most of the circuitry was built "dead-bug style" on an unetched single-sided circuit board measuring 1-3/4 inches by 5-1/2 inches.   The receiver's "muting switch" was built on a piece of "perf board" measuring around 1.5 inches by 1.2 inches.

The "Break-In" operation is not perfect, but it's good enough for somebody (like me) who tends to work CW at speeds not exceeding 25 wpm. At 35 wpm you will still be able to hear the other station during the spacing between words, and probably during the spacing between letters.

The headphones audio has no evidence of "clicks" or "thumps" during "break-in" work. And that's while using some good stereo headphones! Again, referencing Byron Goodman W1DX back in March 1948: When using an HW-8 with this modification, the CW enthusiast no longer "needs to have shock-resistant ears" in dealing with receiver overload.

My HW-8 was reworked with a small mercury-wetted reed relay. As noted above, this creates a risk of "blasting" the receiver's front-end circuits with "transmitter" voltage if the HW-8 is held in unusual physical positions. Experimentally, I find that I start being vulnerable to potential risk if bottom of the rig's case is elevated more than 60 degrees above horizontal (with the broad edge of the front panel remaining horizontal). If I was troubleshooting something with both the "case top" and "case bottom" removed, and I had the rig upside-down on the bench (so I could poke into the PC board foil with a scope probe tip), I would definitely be at risk.

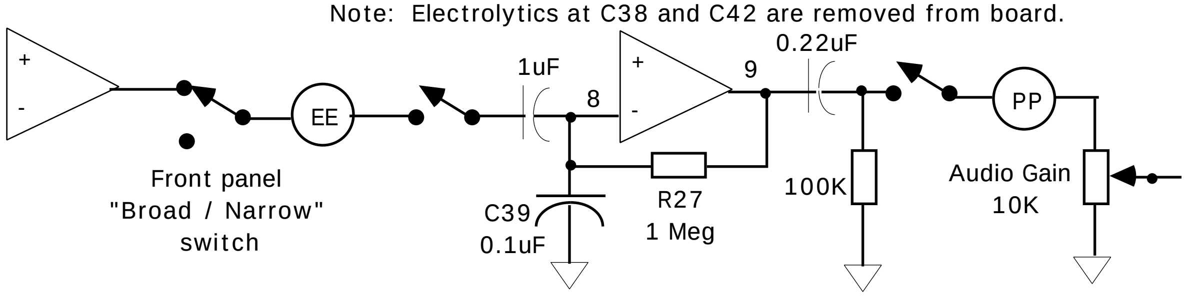
Appendix: Providing QSK while using the original "LM3900" Audio Circuits

In my original attempt at providing my HW-8 with full break-in, I retained the "LM3900" audio circuits at U2. The "clicks and thumps" in the headphones were *almost* eliminated, but there were some operating conditions (probably related to stray RF) where annoying clicks would reappear.

But, if you don't mind the drawbacks of using the original HW-8 audio circuitry, you can try following the information in this section to "tame down" the LM3900 circuits.

Our difficulty is that most of the audio gain is being delivered by a very unconventional high-gain amplifier in the "U2 Pin 9" stage. We need to somehow keep this amplifier from having "wild behavior" during "muting" and recovery. A good first step is to isolate the amplifier from other circuitry with a pair of analog switches.

Note: Electrolytics at C38 and C42 are removed from board.



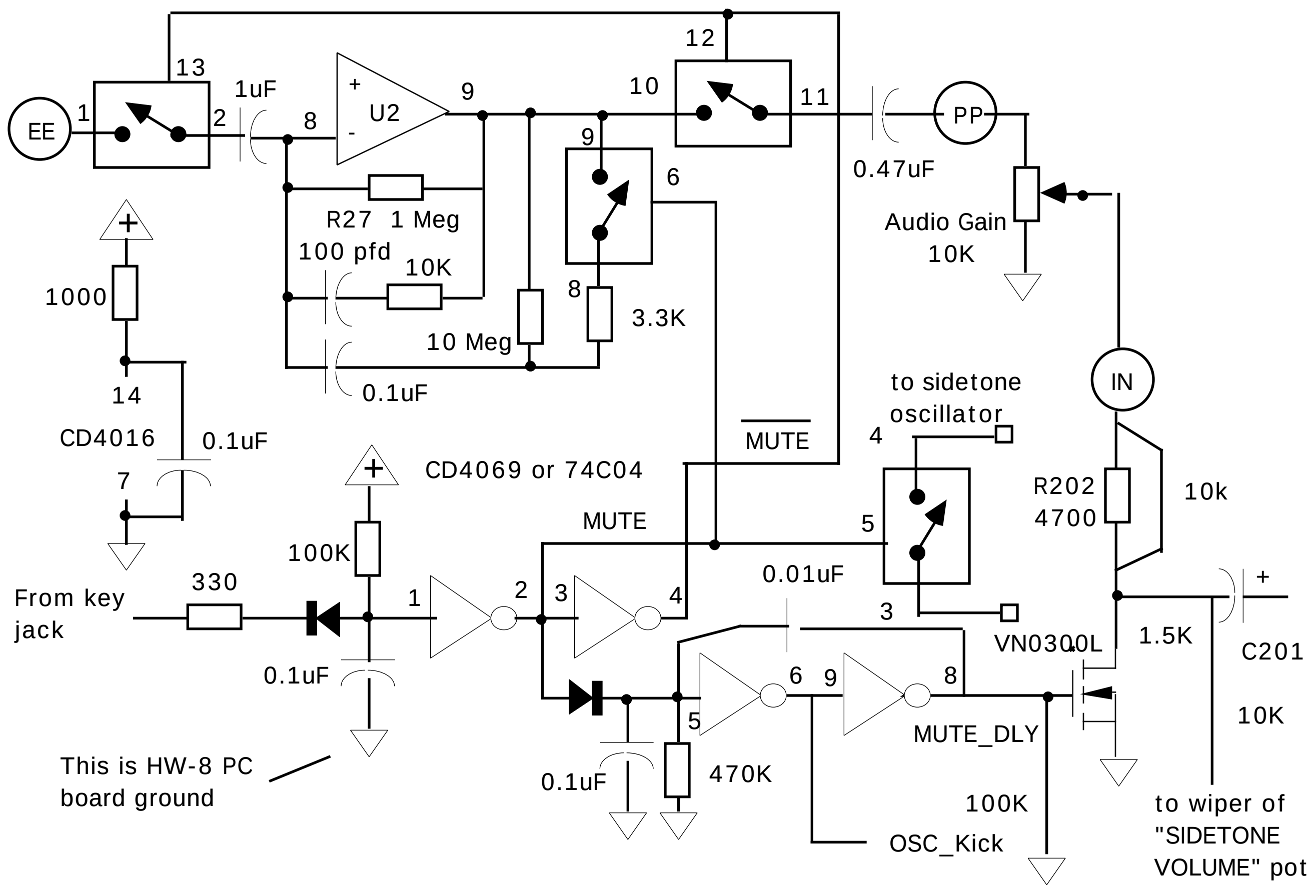
Note: Q14 is removed, as is the 22K resistor at "R29"
C39 is eventually removed in a subsequent step.

Appendix Fig A-1: Preliminary schematic for muting the "high-gain" amplifier stage

In the original HW-8 circuitry, the turn-on of Q14 causes the abrupt "snap" in U2 "DC average" signal levels, which is the cause of the loud "click" in the headphones. Remove Q14, and plan to install two muting switches, as shown here. Reduce the value of the stage's input capacitor from 2 uF to 1 uF to reduce the total energy associated with muting/recovery. As part of this work, it's confirmed that the 0.1uF cap from Pin 8 to ground (i.e., C39) causes some unnecessary (unwanted) amplifier "ringing" at 1.8 KHz. So, we remove C39 from the circuit board.

Progressive iterations ended up producing the circuit that is on the next page. Among other key findings, it was discovered that there is a lot of cross-talk between the various stages of the LM3900 chip at U2. The sidetone oscillator (output on Pin 10) injects lots of noise into the "U2 Pin9" high-gain stage. So, we will disable the LM3900 sidetone oscillator, and build up a dedicated oscillator.
The "muting" switches are on a small perf-board. The CD4016 quad analog switch (Jameco PN 12722) is in a 14-pin DIP socket (Jameco PN 37197). Note that this chip is powered from "+12v" via a 1000-ohm isolation resistor. ("+12v" is picked up from "Relay Sequencing" circuitry.) The "1uF" and "0.47uF" capacitors are on the perf-board assembly.

Note that some of this "muting" is mounted on the HW-8's little side-panel "audio amplifier" PC board.



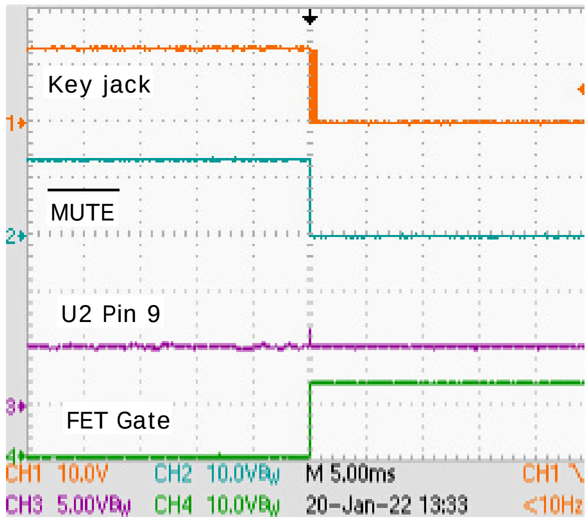
Appendix Fig A-2: Muting Circuitry for Receiver Audio Path

The VN0300L, the 100K "gate pulldown," and "10K across R202) are on the side panel "audio board."
This scheme requires a new dedicated sidetone generator.

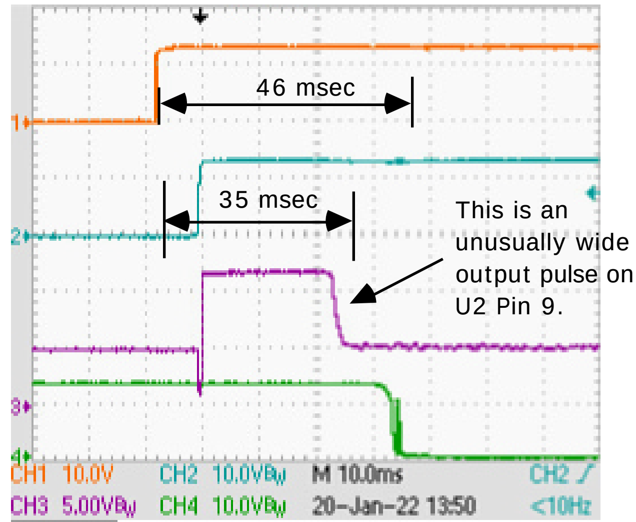
At key-up, the final residue of "thumps" (in "recovery" of the high-gain stage) are suppressed by having a "low on-resistance" FET act as a shunt attenuator. Note that this scheme allows the sidetone "Volume" pot to now be tied into the audio amplifier, instead of directly driving the headphones.

Continuued on the next page.

Performance of the new muting scheme:



U2_Pin9_Muting_KeyDown.JPG



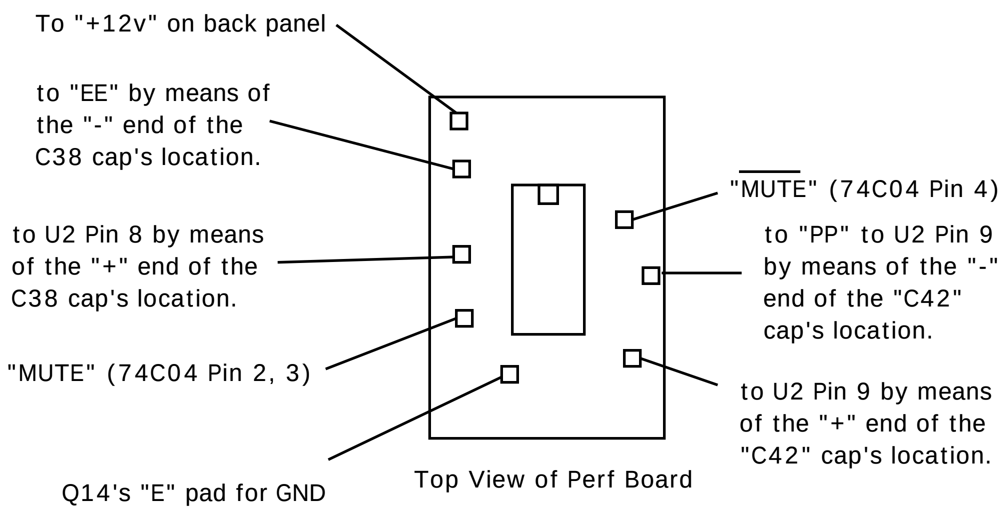
U2_Pin9_Muting_Key_Up.JPG

Appendix Fig. A-3: Timing for "Receiver Muting" functions

The "recovery pulse" of U2 Pin 9 (when $\overline{\text{MUTE}}$ goes high) has a width that can vary a lot between one "key-up" event and the next. The right-hand chart shows an unusually-wide pulse. This pulse occurs when the CD4016 "amplifier output" switch turns on. (The root cause of this pulse is unidentified.)

Timing shows that the receiver could become unmuted at 35 msec after "key-up." However, actual circuit components cause the audio to become active at 46 msec after key-up.

Physical Layout of the Muting Switch for the High Gain Amplifier stage:



Notes:

For clarity, I refer to the "muting circuit sequencing" CMOS inverters as a 74C04, but a CD40106 "hex schmidt trigger" is probably a better choice. A CD4069UBE can also be used.

Wires for the "muting" functions are soldered to "perfboard wire stakes" on the perfboard. The "sidetone oscillator switch" wires are soldered directly to pins 3 & 4 of the IC socket.

Appendix B: Nisus Graphics version of Fig 13: