Up one level
Back to Home

The Jet Propulsion Labratories "Smart VOX"

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The following is a condensation of an article titled: '"Smart" Squelch for SSB' published in 73 Magazine for August 1982. The original article was written by Frank S. Reid (W9MKV) and David A. Link (W9YAN) and describes a circuit which responds to normal human speech while ignoring other forms of audio. The audio-operated squelch circuit described is similar in principle to Motorola's "Constant SINAD" squelch that is used in the Motorola Micom HF SSB Transceiver, and is documented in the service manual part number 6881025E95 (the "A" version is dated 1975, the later "B" version manual is dated 11-76). The Micom squelch board part number is TRN6175. If anyone can scan the information on that Micom squelch board we'd like to add it here.

Following the article condensation is a description of the changes made by Jan Tarsala WB6VRN and myself, Randy Hammock KC6HUR. The changes were made while adapting the Reid and Link "Smart" Squelch to convert it into a "Smart VOX" that we use to control the retransmission of NASA Select audio over JPL's W6VIO repeaters.

Note from WA6ILQ:
The original 73 Magazine article is here, courtesy of Frank Vondra WBØQQK, who scanned his copy for this web page. A reduced size article is here. Grab the larger file if you have the disk space; it's slightly better resolution.
The article includes a PC board layout, and comments that PC boards are available, but remember this was over twenty years ago... and both W9MKV and W9YAN are no longer in the QRZ data base. The PC board shown is single sided and would not be hard to duplicate or the circuit could be built on a piece of perfboard.

Discriminating the Human Voice

Lingusitic research shows that people normally speak about three syllables per second. The squelch works by detecting voice-band energy (500-3,000 Hz) which is varying at a rate of 0.4 to 3.5 Hz.

The circuit is a type of FM detector. It is insensitive to amplitude variations throughout the range where the input stage is not driven to saturation but background noise is strong enough to saturate the limiter. The squelch works properly with most speaker-level signals. It can be connected directly to the receiver's detector output, adjusting the gain of input buffer amplifier U1a as necessary.

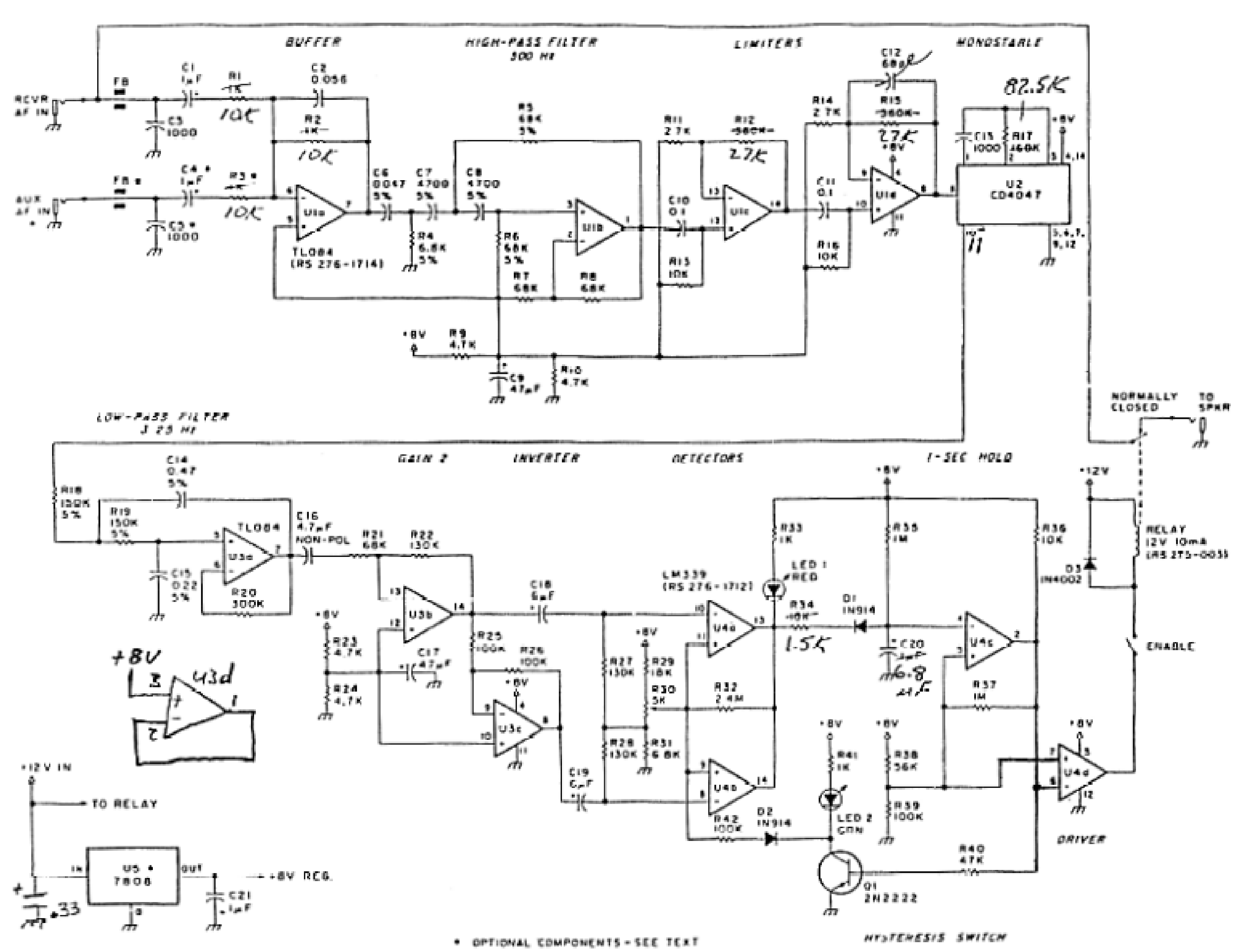

Fig. 1. Schematic diagram.

Click here or on the schematic for a vertically oriented larger version suitable for printing.

Circuit Description

The IC U1a is a unity-gain summing amplifier, input buffer and low-pass filter with a 3 kHz cutoff. U1a drives U1b, a third-order high-pass active filter with 3 DBA cutoff at 500 Hz. High-performance FET-input operational amplifiers are used so that the active filters could use high resistances and small capacitors. The TL084 quad op-amp chip is equivalent to the National LF357 which may be more readily available.

U1c and U1d are limiter amplifiers with a combined gain of 85 DBA (See Circuit Changes #2). U1d's output is voice-band audio turned into constant amplitude square waves. The square waves trigger CMOS monostable multivibrator U2. Output of U2 is a train of .33 MS pulses (See Circuit Changes #3), one for each audio cycle. The average voltage of U2's output is proportional to the input frequency. U2 and the following low-pass filter form a frequency-to-voltage converter, i.e. FM detector.

Active low-pass filter U3a cuts off at 3.25 Hz, the best compromise between noise falsing and the rate at which people speak syllables. Note that U3a has no bias network even though the amplifier uses a single polarity supply. U2's average pulses keep the output of U3a at 5 to 6 volts with normal noise input from the receiver. R17 which sets U2's period, can be varied to keep U3a quiescent output voltage near the center of its range (See Circuit Changes #3).

On very quiet channels, there may not be enough pulses from U2 to keep U3a properly biased (See Circuit Changes #3). False detects may occur as U3a's output goes in and out of it's linear range. You can inject extra noise or a low-level tone into the squelch circuits auxiliary input to achieve he desired results for your application.

U3a's output is AC coupled to U3b, which amplifies with a gain of two and thence to U3c, a unity-gain inverter. U3b and U3c together form a phase splitter with a gain of two. The phase splitter provides positive going outputs for positive and negative frequency deviations of the receiver audio.

Comparators U4a and U4b detect the rate-of-change-of-frequency signals from the phase splitter outputs. If the voltage at the inverting input of U4a or U4b exceeds the reference voltage set by the squelch-threshold control (R30), then the low-going level at the comparators' paralleled open-collector outputs discharges C20 through R34 and Dl. The discharge time constant is 10 ms. C20, R35 and comparator U4c form a time-delay circuit which holds squelch open during its one-second period (See Circuit Changes #1). Each detector output longer than 10 ms resets the timer for another one second.

U4c's output is the squelch-open signal (active high). U4c turns on hysteresis-switch transistor Ql (which lights LED2) and activates output driver U4d. As shown, U4d's output goes high to unsquelch. Since a normally-closed relay is used, the speaker is enabled when the relay is turned off or if the power is removed from the squelch circuit. To reverse the sense of the output, swap the inverting and the non-inverting inputs of U4d. (Jumpers are provided on the PC board.)(See Circuit Changes #6.) U4d's open-collector output can drive a relay in the speaker as shown, or a gated amplifier, analog gate, optoisolator, TTL or CMOS logic circuit. The comparator output can sink 50 mA maximum.

This squelch circuit incorporates a feature called hysteresis, which is a deliberate change in sensitivity betweeen squelch-closed state and squelch-open state. If you design the circuit so that it is more sensitive after opening than before it functions much better in the real world. Without hysteresis, the squelch may drop out while someone is talking. If there is too much hysteresis, squelch threshold becomes hard to adjust properly. Detector U4a and U4b have two levels of hysteresis. Positive feedback resistor R32 prevents comparator oscillation and lowers the threshold during a detect. Ql conducts while squelch is open, further reducing the threshold voltage via R42 and D2. R42 determines the amount of hysteresis. The 100k value shown for R42 provides smooth squelch operation.

Circuit Adjustment

LED1 lights whenever the detector is active. Listen to a voice signal and adjust the R30 threshold control until LED1 blinks for every spoken syllable, then make fine adjustments as necessary.

Circuit Changes

(from the original published article / circuit in 73)

1. Hang time:
   C20 -- From: l uF   To: 6.8 uF
   R34 -- From: 10k   To: 1.5k

   These changes were made to accommodate the audio from NASA Select being retransmitted via the repeater. This will keep the repeater from "chattering" due to the VOX cutting in and out during short delays in the speech.

   The 6.8 uF capacitor along with the 1M (R35) charging resistor will yield a 6.8 second delay before the VOX will drop the PTT. Changing R34 to 1.5k very nearly maintains the original 10ms detect discharge time constant. Alternatively you could make R34 a potentiometer and adjust it until the setting matches the needs of your audio source.

2. Limiters:
   C12 - Remove
   R12 - From: 360k   To: 27k
   R15 - From: 360k   To: 27k

   These changes effectively drop the gain of the limiters from 85 dB to 40 dB. C12 was not really needed.

   In working with the original circuit design, it was discovered that there was entirely too much gain. This over abundance of gain made the circuit susceptible to internal and other stray noise. Also, based on the fact that we will be hitting the circuit with a maximum level of zero dBm with a dynamic range of 30 dB, the 40 dB gain of the redesigned limiters would provide saturation for all anticipated input signals.

3. Monostable and Low-Pass Filter:
   R17 - From: 160k   To: 82.5k

U2 output - From: Pin 10 To: Pin 11

These changes shorten the output pulse width of the monostable and reverses the polarity of the output.

When analyzing the circuit for use on the NASA-Select audio to feed the W6VIO repeater, it was found that if there was no signal on the input to the VOX, the non-inverting input of U3a (low-pass filter) would eventually drop below the acceptable level for it's common-mode input voltage. By tying the input of the low-pass filter stage to the inverted Q output (pin 11) of the monostable, U3a's non-inverting input level would be held within the acceptable range for it's common-mode input voltage (the maximum level being Vcc+). The pulse width of the monostable was shortened so that when there is noise on the input to the VOX circuit, the common mode input voltage on the non-inverting input of U3a would remain within acceptable limits (between +4v and Vcc+).

4. U3d (unused / extra op-amp):
   The inverting input was tied to its own output while the non-inverting input was tied to Vcc+. These changes lock the opamp down and keep it from becoming unstable and generating noise on the power bus.

5. Monostable:
   Added a 0.1 uF cap across U2 pins 7 and 14 for noise suppression.

6. Output Driver (U4d):
   The connections to U4d Pins 7 and 6 were reversed to cause the output driver to conduct on detect rather than release as shown in the original schematic.

7. Threshold Control:
   The threshold control was wired backwards from what is shown in Fig. 3 Component Layout. Since the original article was written describing a squelch circuit, the original orientation of the control would make it "feel" like a squelch control where turning the knob counter clockwise tightens the squelch.

   When being used as a VOX, the reversal of the control wiring now makes the control "feel" like a sensitivity control where turning the knob clockwise increases the sensitivity.

8. Input Audio Buffer:
   Rl,2,3 - From: 1k To: 10k
   This changed the input impedance of the entire circuit to prevent loading down the line level audio source. The original circuit was designed for a low-impedance speaker-level source, and we were driving it with a line-level audio source.

Note from WA6ILQ:
The original article used a red led for LED1, labeled as "Detected" and a green led for LED2 labled as "Unsquelched".

The SPST toggle switch that is in series with the relay coil could be "flipped over" and replaced with a center-off SPDT switch. Tie the armature to the coil connection, one pole going to the IC output, and the other pole going to ground. This would give you a test feature for essentially free: the center position disables the receiver, the ground position simulates a signal, and the other position provides normal operation.

The output driver IC could feed the base circuit of an open collector NPN transistor through a 1K resistor. That collector could feed, through a suitable connector, the COR input of the Receiver #2 port on a repeater controller. The repeater controller would be configured as if port #2 was a remote base, with the standard DTMF commands for remote base off, remote base monitor and remote base transcieve... however the last one would not be needed as the audio feed is listen-only.

If you build this circuit I recommend that you make R34 a 10-15k potentiometer (label it "Dropout Delay") and mount it on the front panel. This will allow you to adjust how long your transmitter stays keyed to match your audio source.

From the email from Randy, KC6HUR, that accompanied the above writeup:

For the most part, the "Smart VOX" does work well but needs some tweaks. While it actually ignores static, tones and music, it generates a detect on the start transient of the sound. So, for a tone (i.e. an "over beep", an IMTS idle tone, or a DTMF string), it will trigger on the onset of the tone but will drop out. It also triggers on static crashes that come over the shuttle audio.

There should be some way of limiting the transient response such providing a delay circuit that holds off keying the PTT until two to four syllables are accepted. This would of course demand a audio delay module (so you actually hear those opening syllables) which I would highly recommend anyway.... Arcom makes a good one that has a built in audio mute.

Lastly, there are two schools of thought on the length of hold (dropout delay) on the PTT output. One wants the Smart Vox to assert PTT for 6-8 seconds after last audio detect, allowing those 6-8 seconds of audio to be heard on the output. The other school feels that should be the responsibility of the hang-timer (a.k.a carrier delay) on the repeater. The latter method would prevent a lot of the noise that comes over the shuttle audio from being transmitted. For example, using the first method you'd hear 6-8 seconds of the "wakeup music" that is used to wake the astronauts each morning. Using the second method you'd hear a chirp, then the repeater's courtesy beep or carrier delay timer. The "Dropout Delay" potentiometer referenced above would control this duration.

Back to the top of the page
Up one level
Back to Home
