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The QRSer: A CW Operating Aid

Here's how you can put solid-state brakes on too-fast CW!

I'll admit it—I *like* CW! In fact, most of my operating time is spent on CW, and I've put a lot of effort into increasing my copying speed. Even so, there are times when a call sign or contest exchange is sent too fast for easy copy. In a contest, milliseconds add up, providing an incentive for the big guns to crank their keying speeds to stratospheric levels when sending CQ. As often as not, the call sign has an unfamiliar prefix that makes it even harder to decipher. Consider copying a call like "HH5SH" at 50 WPM and you'll see what I mean!

The polite request to "PSE QRS" works wonders during a normal "ragchew," but who wants to slow down a good operator when they're in the middle of a good run or working a pile-up? Sure, I can listen on frequency until I get all the information I need, then make my contact and move on, but that takes time and success is not guaranteed.

The Past and Present

In the days of reel-to-reel tape recorders, the problem was easily handled: Just record the exchange at $7\frac{1}{2}$ inches per second and replay it at $3\frac{3}{4}$ inches per second. Now, an indecipherable 50-WPM transmission is reduced to a manageable 25 WPM. But most tape recorders available today have only a single recording and playback speed.

Fortunately, technology comes to the rescue. Solid-state audio-recording devices, widely used in answering machines and memo recorders, are the basis of the voice keyers so common in phone contests. Why couldn't such a device be used to capture a few seconds of received audio at one speed and play it back at a slower speed? Replaying the captured segment at a lower speed would make it much easier to recognize a call sign or other transmission. A few seconds is long enough to record an entire contest exchange. That's how the QRSer was born.

The circuit is based on a device made by Information Storage Devices, the ISD1110P.[1] This chip requires only a handful of external components to record and play back 10 seconds of voice-quality sound. The recording is not continuous, as with a tape recorder. Rather, the sound is sampled at a rate determined by either an internal or external clock. The samples are stored and reconstructed into an audio signal during playback. Samples are stored as *analog* levels, and the resolution of the storage and retrieval of the sampled waveform is equivalent to that of an 8-bit ADC.

The trick in the QRSer is to use different clock frequencies for record and playback. With the ISD1110P, an internal clock running at 819.2 kHz normally controls sampling. This results in a sampling rate of 6.4 kHz. But the internal clock can be overridden by an external one. Suppose the chip is driven at 800 kHz while the data is sampled, but at 400 kHz for playback. Now it will take 20 seconds to play back the information that was recorded in 10 seconds—just what we did with the tape recorder!

If the design team at Information Storage Devices ever sees this, they'll probably emit a collective groan. The sampling frequency controls the ISD1110P filter passband—normally 2.6 kHz—optimized for voice recordings over ordinary telephone lines. The pitch of the reproduced sound is reduced by the same proportion as the clock frequency. Halving the clock frequency cuts the digital-filter bandwidth and the filter's center frequency in half, but obviously cannot affect the rejection properties of the internal analog filter. ISD does not recommend changing the clock speed between record and playback because the quality of the reproduced sound is degraded somewhat, but for CW reproduction, the fidelity is quite good. It might take some practice to get used to copying CW at a lower pitch than normal, though.

[1]Notes appear on page 36.

Longevity

One of the drawbacks to tape recording is that the tape eventually wears out. The ISD1110 is all solid state, but like other nonvolatile memory devices, it, too, has a finite life expectancy. The specified life of the memory cells is 100,000 write cycles. For a 10-second chip, this translates to about 275 hours. This is a lot of operating time, and some chips in this family can be used as an "infinite tape loop," so that the last 10 seconds of received audio is always available for review. I didn't like this approach. It would be too easy to leave the device running and exceed the IC's write-cycle specification. Instead, the QRSer uses momentary-contact switches to select the record and playback functions. It is easy enough to know when to start recording and playback always starts at the beginning of the recorded message.

Circuit Description

Refer to Figure 1. Power is derived from your station's 13.8-V supply; most stations today have one or more such supplies available. Voltage regulator U1 provides 5 V for the logic chip and the ISD1110. I used a 1.5-A regulator, which is loafing at the maximum drain for this unit, because the idea of regulator failure putting 13.8 V on the recorder chip was unappealing. U1 needs no heat sink. D1 ensures that accidentally reversing the power leads won't damage anything.

The circuit provided by the ISD1110 application note requires little modification for this purpose. All of U2's address lines are grounded, so recording and playback

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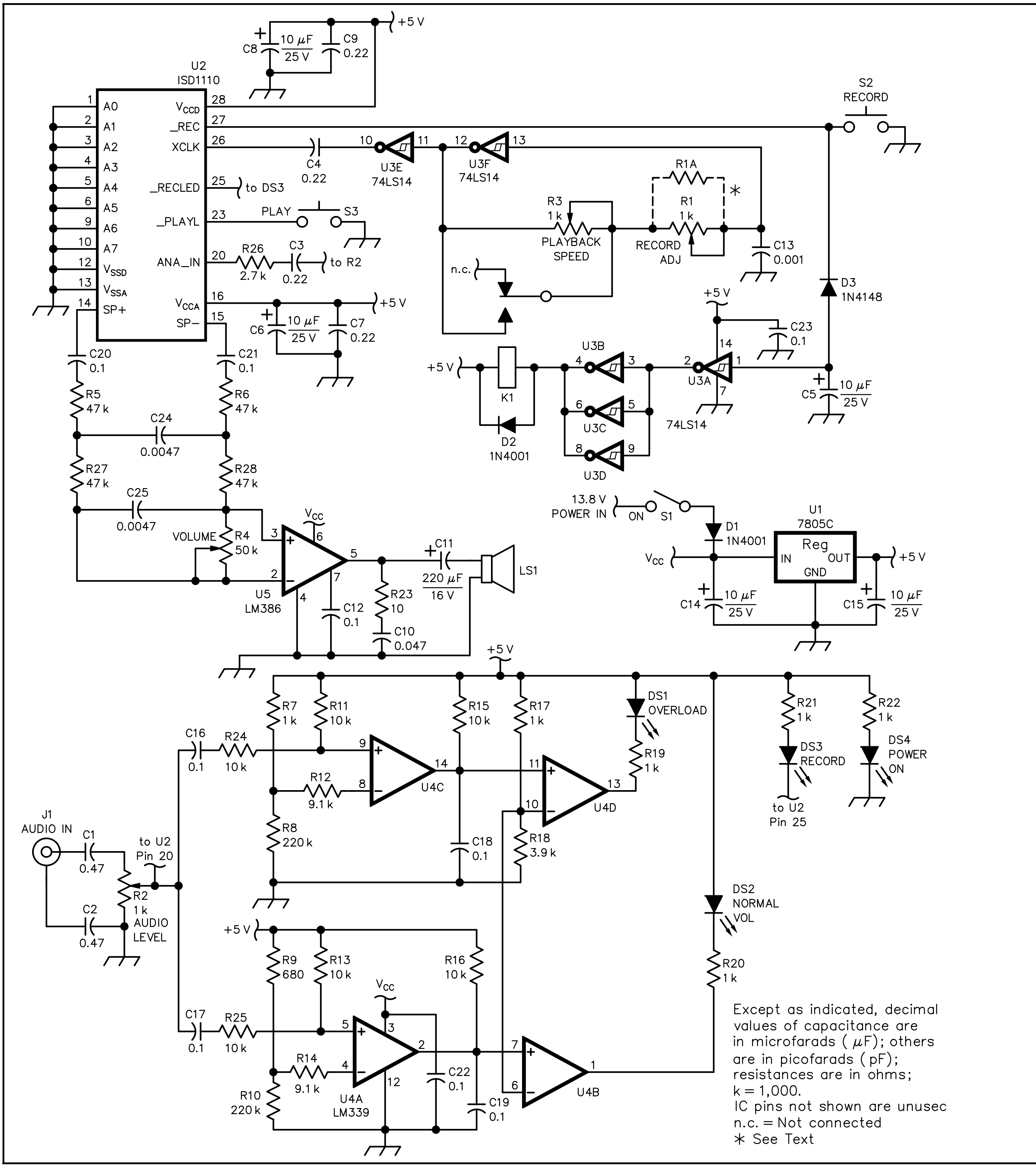
always start at the beginning of memory. The audio preamp intended for microphone-level signals is not used.[2] Instead audio is fed to U2's **ANA_IN** pin from the **LINE OUT** connection of my Kenwood SP-31 speaker. This signal is not affected by plugging headphones into the speaker front panel and has plenty of amplitude to drive the recorder chip. Because the unit is powered by the station power supply, both sides of the audio input are capacitively coupled across the speaker to avoid grounding issues. R2 sets the input level.

The signal amplitude applied to the analog input of U2 must be kept below 50 mV peak to peak. Two comparators set the audio level at nearly maximum. As shown in Figure 1, the negative input of U4C is held



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at about 25 mV below the 5-V supply, while the positive input is the sum of the supply voltage (5 V) and half the audio level. As long as this sum is higher than the voltage at U4C's negative input, the output transistor in U4C is turned off and C18 charges to 5 V through R15. This places the positive input of U4D at 5 V. The negative input of U4D is at 4 V, so the **OVERLOAD** LED (DS1) is not lit.

When a negative peak of the input signal exceeds –50 mV, the output transistor of U4C turns on, rapidly discharging C18. U4C turns off again after the signal peak, but C18 can only recharge through R15 because the LM339 outputs are open collectors. Until the voltage across C18 exceeds 4 V, the output of U4D is at ground potential and DS1 is lit. The time constant of R27 and C19 is 1 ms, so for signals above a couple of hundred Hertz, the LED stays on constantly until the peaks no longer drop below –50 mV. A similar comparator with the trip point set at about 15 mV controls the **NORMAL VOLUME** indicator, DS2. Because the LM339 has rather high bias voltage and current, the exact switch points may vary by several percent from the design values, but the comparator works well enough for this purpose. U2's analog-input impedance is about 3 kΩ, so the audio input to U2 is half the attenuated signal level (ie, 30 mV peak to peak at the threshold for lighting DS3, and 50 mV at the threshold for DS2). If an oscilloscope is available to set the signal level, this part of the circuit can be eliminated, but there is a certain appeal for some of us in watching the LED flash in sync with incoming CW.

A sound bite is recorded by pressing S2. The **RECORD** LED (DS3) lights while recording. When the record memory is full, DS3 extinguishes and the chip automatically goes into power-down mode. In addition to activating the recorder chip, S2 provides a discharge path for C5, grounding the input to U3A. This causes U3A's output to go high, energizing K1. K1's contacts bypass the **PLAYBACK SPEED** pot, R3. This makes oscillator U3F run at a speed determined by the time constant of C13 and R1. Recording can be stopped at any time by releasing S2. When the switch is released, C5 charges through the internal pull-up resistor in U3A. This holds K1 closed until the internal debounce circuit on the ISD1110 **RECORD** pin has timed out. If this isn't done, the clock might switch to its lower speed before recording stops, causing an abrupt and unwanted pitch change at the end of the recorded message. D3 prevents C5 from holding pin 27 of U2 low.

Trim pot R1 (**RECORD ADJ**) allows closely matching the record speed to U2's internal clock. It appears that when S2 opens, U2 ignores the external clock for a few milliseconds before recording ceases. This produces an effect similar to allowing the external clock to change frequency during recording (ie, a short but noticeable pitch change). This is a blessing in disguise since it provides an easy way to set the record clock to the design frequency of the ISD1110 (more on this later). This optimizes signal quality while recording. The exact resistance needed across U3F to give the required clock frequency depends on the tolerance of C13 and the internal pullup at U3 pin 13. In my prototypes, this resistance is about 750 Ω. The FAR PC board[3] provides pads for adding a fixed-value resistor (R1A) in parallel with R1. Using an 820-Ω resistor at R1A and a 10-kΩ trim pot for R1 makes adjustment of R1 less sensitive.

Pressing S3 plays back the recorded message. Because K1 is not energized, the time constant of C13 and the series combination of R1 and **PLAYBACK SPEED** pot, R3, determine the oscillator speed. R3 allows adjusting the playback speed from the recording speed down to less than half the recording speed.

U2's audio output is designed to deliver a mere 22 mW to a 16-Ω speaker and does not provide any means of controlling the playback volume. Because the output tones of interest are going to be at the low end of the speech spectrum and the efficiency of small speakers is poor, more audio power is needed to deliver a good listening level. Audio output is provided by U5, the ubiquitous LM386. For optimum signal-to-noise ratio, U5 is driven by the speaker outputs using a balanced circuit; R4 is the **VOLUME** control. U2's speaker outputs are isolated from the LM386 at dc, so U5 can be powered from the higher, unregulated supply voltage. Be sure to use an LM386N-4, designed for a maximum supply voltage of 18 V. If you use an LM386N-1, power it from the 5-V regulated source. This increases U5's power output over that available if its supply voltage were limited to 5 V.

The audio output of U2 contains a noise component at 1/160 of the clock frequency. Normally, this frequency is above the filter passband at about 5 kHz. However, reducing the clock speed by a factor of two moves the noise into the passband of U2's output filter. R5, R6, R27, R28, C24 and C25 remove this noise component and clean the reproduced sound.

Building the QRSer

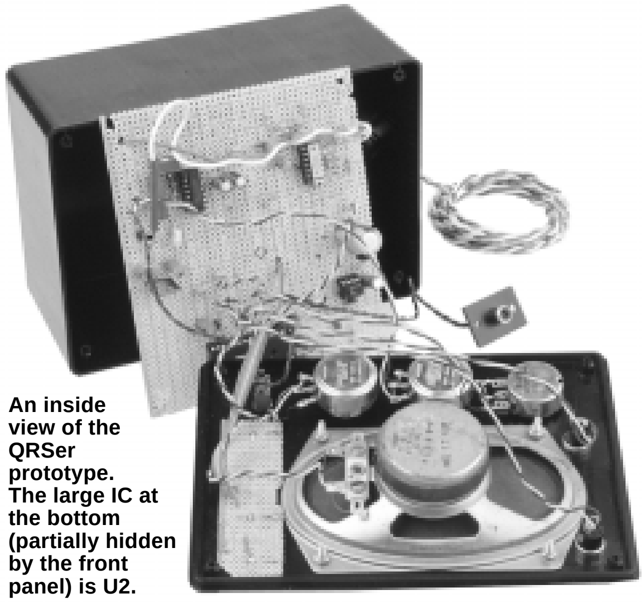
A PC board for this project is available from Far Circuits (see Note 3). A semi-kit including U1 through U5, sockets for U2 through U5, K1, and the PC board is available from me.[4] All other components are available from Digi-Key. Most parts can also be obtained from RadioShack, but the ISD1110 and 74LS14 are special-order items. To save a few bucks, the best place to find switches, pots and hardware is in your junk box or that of a friend.

There's nothing tricky about building the QRSer. Prototypes I assembled on perfboard work well. Point-to-point wiring of this project is less work than it might initially appear—most of U2's 28 pins are either unconnected or grounded (U2 pins not shown in Figure 1 must be left floating.). It's important to provide separate power leads to the digital and analog power pins of U2, and to bypass these leads to ground as close to the IC as possible using a 10-µF electrolytic capacitor and a 0.1-µF ceramic capacitor. If this is not done, the recorded sound suffers from noise. The 74LS14 and LM339 power pins are bypassed at their sockets with 0.1-µF ceramic capacitors.

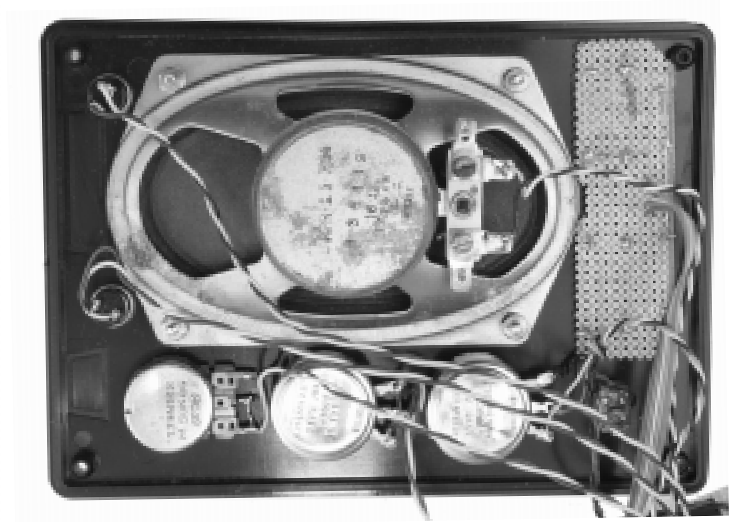
My prototype is built in a 5×7×3-inch (HWD) RadioShack project box (RS 270-

Figure 1—Schematic of the QRSer. Unless otherwise specified, resistors are 1/4 W, 5% tolerance carbon-composition or film units. DK part numbers in parentheses are from Digi-Key Corp, 701 Brooks Ave S, Thief River Falls, MN 56701-0677; tel 800-344-4539, 218-681-6674, fax 218-681-3380; http://www.digikey.com. Equivalent parts can be substituted; n.c. indicates no connection.

C1, C2—0.47 µF monolithic
C3, C4, C7, C9—0.22 µF monolithic
C5, C6, C8, C14, C15—10 µF, 25 V electrolytic
C10—0.047 µF ceramic
C11—220 µF, 16 V electrolytic
C12, C16-C23—0.1 µF ceramic
C13—1000 pF polystyrene
C24, C25—0.0047 µF ceramic
D1, D2—1N4001 50-PIV, 1-A diode (DK 1N4001MSCT)
D3—1N4148 (DK 1N4148MSCT)
DS1, DS3—Yellow LED (DK HLMP-1719QT)
DS2—Red LED (DK HLMP-1700QT)
DS4—Green LED (DK HLMP-1790QT)
J1—Phono jack
K1—SPST miniature relay, 5-V dc coil (DK HE206)
LS1—8 or 16-Ω loudspeaker, 3-inch minimum diameter (DK P10187); see text.
R1—1 kΩ or 10 kΩ trim pot; see text.
R1A—820 Ω; see text
R2—1-kΩ linear- or audio-taper potentiometer
R3—1-kΩ linear-taper potentiometer
R4—50-kΩ to 500-kΩ linear- or audio-taper potentiometer
S1—SPST toggle switch
S2, S3—Normally open push-button switch, panel mount
U1—UA7805KC 5-V, 1.5-A positive-voltage regulator (DK 296-1974-5)
U2—ISD1110P ChipCorder (DK ISD1110P)
U3—74LS14N hex Schmitt-triggered inverter (DK 296-1643-5)
U4—LM339 quad comparator (DK 293-1393-5)
U5—LM386N-4 audio amplifier (DK LM386N-4)
Misc: PC board (see Notes 3 and 4), enclosure, hardware



An inside view of the QRSer prototype. The large IC at the bottom (partially hidden by the front panel) is U2.



The back side of the front panel. Most of the panel is occupied by the oval speaker. A strip of perfboard (right) holds the four LEDs. Across the bottom are the three pots and toggle switch. The **RECORD** and **PLAY** pushbuttons are to the left.

1807). The speaker and controls occupy most of the space. This enclosure provides enough room for a 3×5-inch oval speaker firing through the front panel. Because the playback frequency is reduced by the same ratio as the CW speed, it's best to use a fairly large speaker and enclosure. A 2-inch speaker I tried first did a very poor job of reproducing the 300-Hz output tone resulting from the halved 600-Hz CW tone I normally use.

Good things come in small packages, but be careful about trying to stuff the QRSer into a small enclosure. The reed relay is very sensitive to magnetic fields. If it is too close to the speaker magnet, the relay may be held closed even when the coil isn't energized. If your project works perfectly until you put it into the box, this is one place to look! Although I've not experienced any problems with prototypes without D2, adding the diode across the relay coil to dampen the inductive kick when the relay is deenergized is a good idea. You'll have to place D2 on the foil side of the PC board.

Another way to package the QRSer is to build it into an existing speaker enclosure and use a relay actuated by S3 to switch the speaker between audio from the rig and the output of the LM386. This makes it unnecessary to turn down the receiver volume in order to hear the recording. Both sides of the speaker should be switched using a DPDT relay to avoid grounding problems. Be sure to install a diode across the coil of the added relay.

The only adjustment required before putting the QRSer into operation is to set R1 so that the frequency of U3F matches that of the internal clock in U2. If a frequency meter is available, measure the frequency at pin 10 of U3 and adjust R1 to set it at about 820 kHz. Otherwise, your "piano-tuning skills" will come in handy for this procedure. Supply a constant audible signal to the audio input. Hold S2 closed momentarily then release it. Press S3 and keep it pressed until the recorded message ends. Unless you got lucky and the frequency match is perfect, the reproduced sound will be steady until the end of the recorded tone, then will shift up or down. Moving R1 one way moves the shift higher in frequency; moving it the other way shifts it lower in frequency. Repeat the **RECORD/PLAY** cycle adjusting R1 each time until the frequency shift at the end of playback disappears.

Operation

Using the QRSer is easy. Connect a suitable signal source to the audio input and apply power. I use twisted-pair pigtails that pass through a grommet-protected hole in the rear of the enclosure, but you may prefer to install jacks and use cables. If the audio-input source is unbalanced, the high side must be connected to C1 and the grounded side to C2. Unless the signal source is of very low amplitude, it should be possible to adjust the **INPUT LEVEL** control so that DS2 lights to the dits and dahs of incoming CW. DS3 should not light except, perhaps, on noise peaks. If an oscilloscope is available, use it to check the signal level at pin 20 of U2. The peak-to-peak signal amplitude should not exceed 50 mV.

Press the **RECORD** pushbutton to record up to 10 seconds of audio. Press **PLAY** and adjust the **PLAYBACK SPEED**. The recorded message can be replayed as many times as desired by releasing the **PLAY** switch and pressing it again; the message will always start at the beginning of the 10-second interval.

Acknowledgements

I want to express my thanks to my dad, W1BSO, for getting me involved with Amateur Radio more years ago than I like to remember.

Notes

[1]Information Storage Devices, 2727 North First St, San Jose, CA 95134; tel 800-677-0769, fax 408-544-1789; **http://www.isd.com/**.

[2]This input has a maximum signal-input level of 20 mV peak to peak and its gain is controlled by a fast-attack/slow-decay AGC. However, I found that recordings were noticeably noisier when the input level was reduced much below its maximum value. Because I planned to take audio input from across an 8-Ω speaker, it would be necessary to attenuate the signal considerably to bring it below 20 mV. There isn't much sense in attenuating a signal just to amplify it again. The FAR PC board brings preamp pins 16-18 to pads for the builder who wishes to modify the present design to accomodulate very low level audio inputs.

[3]PC boards for this project are available from FAR Circuits, 18N640 Field Ct, Dundee, IL 60118-9269; tel 847-836-9148 (voice and fax). Price: $7.50 plus $1.50 shipping for up to four boards. Visa and MasterCard accepted with a $3 service charge.

[4]A semi-kit consisting of all semiconductors, sockets for U2-U5, K1, and a PC board is $23 postpaid; $15.50 postpaid without the PC board (see Note 3). Contact Charlie Cheney, K1LDZ, 319 Highland St, Northbridge, MA 01534; **charliec@ieee.org**. Price: $27.50

Charlie Cheney, K1LDZ, was first licensed as KN1LDZ in 1959. He's held an Extra class license since 1992. He's interested in restoring and operating vintage (tube type) ham gear, and in building homebrew equipment. For DXing and contests, however, he's "not averse to using more modern equipment." His favorite mode is CW (including mobile). Charlie holds a BS from Boston College and an MS from the University of Chicago. He works as the principal software engineer for QuadTech Inc (formerly GenRad Instruments) writing firmware for electronic test equipment. You can contact Charlie at 319 Highland St, Northbridge, MA 01534; **charliec@ieee.org.**

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