allows the input amplifier to run at full gain with weak signals. When a signal reaches the set point, (just below the AGC 'knee'), the gain of the input and first IF amplifiers are adjusted so that the signal level, at the input to the 3089, does not exceed the set point. What you will see on the signal meter is that once the set point is reached, the signal meter will stay at the same reading, no matter how strong the signal becomes.

This is done by placing an error amplifier between the AGC output of the 3089, and the RF, IF amplifier (controlled by Q6). For stability, the DC gain of the amplifier is higher than needed to close the loop, but low enough to prevent wild output swings in the OP AMP. In this case, a gain of 22 is enough. Since the AGC output from the 3089 is negative going for strong signals, a non-inverting amplifier is used. P2 adjusts the AGC set (operating) point. A 22 µf cap on the base of Q6, provides the basic AGC loop stability and AGC speed. Power to the circuits is supplied via a 78M08 three terminal regulator

Construction

Building this receiver on a PCB is a must. The board is 4"x5". Any other method is bound to prove time consuming. The board layout and parts list are not published here to conserve valuable page space. The actual construction is straight forward. Simply insert the parts in the proper places and solder. The layout is a bit tight in places. If you stuff the parts in the order of their physical height, there should be no problem. The completed board will fit into one of several project boxes from Radio Shack. If you would like a circuit board for this project, drop me a line. If there is enough interest, say 10 or more, we'll have **Fred** at **FAR Circuits** make up a batch.

Alignment

If you have tuning equipment, an RF signal generator with variable output level and a 'scope will help in the adjusting and possible trouble shooting of the receiver. First, apply power, turn up the volume and place receiver in the CW mode. Adjust P1 to just below the point of maximum hiss. Adjust P2 until the output voltage on pin 7 of U7 goes to maximum. Connect an antenna or signal generator to thye antenna input. Tune around until you hear at least a hint of a signal. Tune T2 and then T1 for maximum signal. The tuning of T2 is sharp and that of T1 is very wide. Tune for a real strong signal. Adjust P2 so that the audio is not distorted. When it becomes muffled, you have gone to far. You will indubitably have to play around with the setting of P2 until you find the point of best AGC action. It will be easiest to set the AGC control if you can tune in SW broadcast stations. Those stations on the 49 and 40 meter band prove to be a good test here in the Northeast.

Additional Information

For $2.00, I will send you an information kit that will include the complete parts list with vender part numbers and phone numbers, full size schematic, X2 PCB parts placement and PCB artwork.

Steven Weber, KD1JV
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Gorham, NH 03581

The R2 Single Sideband Direct Conversion Receiver, Revisited: The R2a

Glenn Leinweber VE3DNL

[In this article Glenn presents some significant improvements to the audio channel in Rick Campbell's R2 receiver. Here is a chance to spend some time at the workbench and to squeeze even higher performance out of an already fine receiver. -WB6TPU]

Rick Campbell's R2 receiver prompted the question, "Can it be prodded to perform even better?". A good deal of simulation, and breadboarding showed a few minor flaws, some of which Rick has already corrected. My main interest is in improving the audio chain. The goal was to achieve an overall noise figure 10dB or under. A new method of tuning the audio phase shifter is introduced that allows sideband suppression of 50 dB. A lower noise audio preamp, a sharp, tunable audio filter, and a more powerful no-adjustment audio driver are included as well.

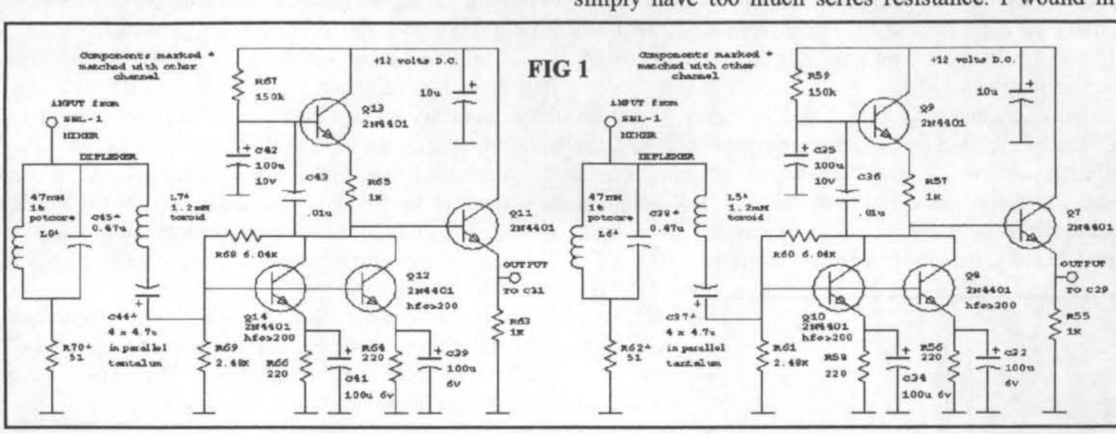


FIG 1

Low-level Audio Preamp

The old common-base audio preamp, having a noise figure of 5dB is an obvious target to improve the noise figure. A common-emitter configuration, with shunt feedback gives 50 ohms input resistance (preferred by the mixer) and a 2dB noise figure. Two paralleled 2N4401 transistors results in better noise performance at such a low input impedance. A gain of 60 masks most of the noise in the op-amp phase shifter that follows. Common collector ouput stages provide a low output impedance needed to drive the audio phase shifter.

A simpler diplexer between mixer and preamp shaves a little off the noise figure too. See the schematic in figure 1. Rick's R2 diplexer has steeper stopband slope, but about 2dB loss. The TOKO 10RB inductors simply have too much series resistance: I wound my own on small potcores, 14x8 mm. The potcore bobbins are very easy to wind, but its tricky to get the inductance exact - too much pressure from a mounting screw can change inductance dramatically. Potcore mounting hardware from the manufacturer is recommended. The 1.2mH inductor should have low internal capacitance - 35 turns #32 wire in a single layer on a FT37-77 ferrite toroid.

The diplexer's low-pass frequency at 6700 Hz., and high-pass frequency at 170 Hz are outside the audio passband (350 - 3500 Hz.). This means that unwanted phase shifts from mismatched diplexer parts are less troublesome, relaxing the need for extremely accurate component values. With the tuning method described below, some of the bad effects of mismatched components can be tuned out. Nevertheless, diplexer components were matched with the aid of a commercial Maxwell impedance bridge.

Quadrature Phase Shifter

The phasing R's and C's were all scaled so that resistor values were much smaller. Otherwise, the Johnson noise from those warm, large value resistors adds so much noise that preamp gain would have to be much higher to achieve the same noise figure. A return to a low-noise bipolar op-amp (LM837) gives lower noise than the FET-input TL074 with these lower resistor values. The Motorola MC33079P should work equally well. The circuit is show in figure 2.

Most low-noise op-amps have extended frequency response as a side-effect. My circuit layout was slightly unstable, oscillating at 3 Mhz. Running at low power supply voltage doesn't help, but oscillations are most likely caused by too much stray capacitance between the inputs and ground. A 10pF feedback capacitor is usually enough to correct the problem, and doesn't add noticeable phase error. A simpler USB/LSB sideband switching arrangement means a few less op-amps.

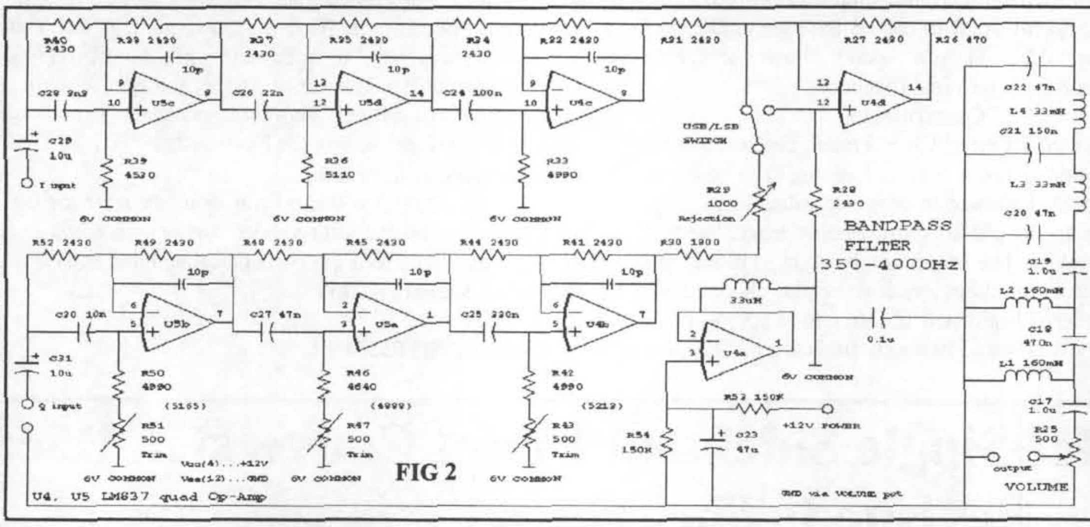


FIG 2

This phasing circuit is capable of high sideband suppression; a minimum of 58 dB (figure 3). As Rick has mentioned, actually doing this well is very very difficult, mostly because of component tolerances. Capacitors C24, C25, C26, C27, C28, C30 should all be of the same type. The Panasonic P-Series (polypropylene) capacitors have tight tolerance, and good temperature stability. Philips 460 series are even better, but unavailable in the larger sizes. Polystyrene capacitors are good too, but only small values are available. Almost all the resistors should be 1% precision resistors.

Trimming these components is considered by many to be too difficult, partly because of the need for a quadrature signal generator. I'm a believer in building your own test equipment; a very simple and accurate quadrature generator was developed that uses two common CMOS chips. Figure 4 illustrates the circuit. This circuit can be built temporarily on a protoboard to trim the three phasing trimpots R43, R47, and R51 for best alternate sideband suppression. The signal generator must supply two signals of the same frequency, same amplitude, but with exactly a 90 degree phase relationship. Since each of the three trimpots to be tuned affects some audio frequencies more than others, these two signals must be frequency agile: 300 - 4000 Hz. Two square waves in quadrature are easily generated with a logic shift register. However, the harmonics of those square waves make it very difficult to determine the null point for the fundamental frequency of interest. Sharp low-pass filtering in the following audio stages can eliminate those harmonics. Simply listen for minimum signal of the resulting sine wave.

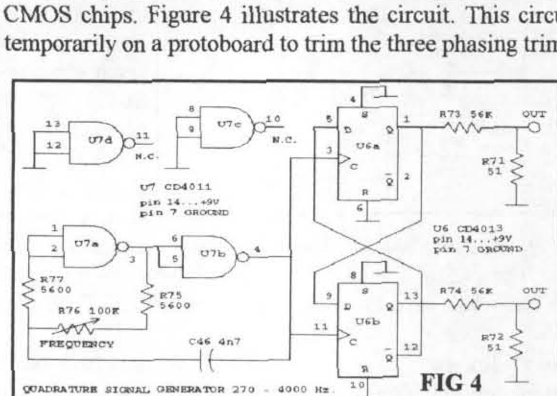


FIG 4

U7 is a simple square-wave oscillator whose frequency can be adjusted with R76. It oscillates at four times the output frequency. The dual flip-flop U6 is connected as a two-bit shift register. Outputs at pin 1 and 13 are at the same frequency, but shifted 90 degrees in phase. R72 - R74 divide the output voltage down to a small value (with a 50 ohm output impedance) that the preamps can handle. Since there is so much gain in the R2's audio chain, the possibility of ground loops can adversely affect trimming adjustments. Power the signal generator from a separate power supply, perhaps a 9v battery. It draws less than a few milliamps.

You might wish to set the three trimpots with an accurate ohmmeter first. Theoretical values of resistances from the non-inverting inputs of op-amp U5b, U5a, U4b to 6V COMMON are listed on the schematic diagram in curly braces.

Apply the two outputs of the quadrature generator to the two diplexer inputs. Now adjust the SCAF frequency (R15) to pass only the fundamental frequency, and eliminate all the harmonics. You can do this by ear, or look at the voltage at C5 with an oscilloscope. You should be able to adjust R15 to hear (or see) a sine wave. Flip the USB/LSB switch to the side giving the lowest amplitude. Adjust R29 for minimum signal. Now you can go to the three 500 ohm phasing trimpots.

Iterative tuning will be required, since each trimpot affects the setting of its neighbor a little. R51 will trim the highest audio frequencies: try nulling for best sideband rejection at 3 KHz. R47 trims mid-frequencies: try for a null at 700 Hz. R43 trims the lower end: 320 Hz. At each of these frequencies, adjust the SCAF cutoff frequency to pass most of the fundamental frequency, but reject harmonics of the square wave. Rock R29 back and forth as well to help find the best null. The signal generator provides a large enough signal that a deep null should be achievable.

Figure 3 shows that this quadrature circuit can give excellent sideband suppression. However, mismatching in the local oscillator quadrature hybrid, and an unbalanced R.F. splitter can easily degrade sideband rejection.

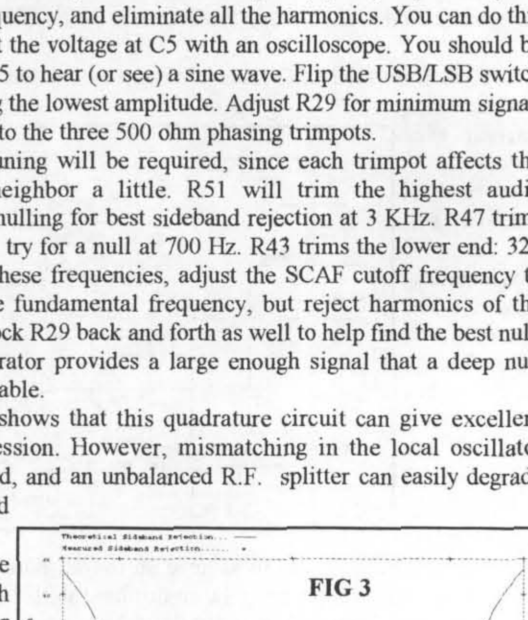


FIG 3

The Butterworth band-pass filter following the audio phasing combiner U4d passes audio from 350 Hz to 3500 Hz. Again, the TOKO 120 mH pre-wound coils have been replaced with hand-wound potcores. The 120 mH 10RB TOKO coils have a Q of three at low audio frequencies...a little too low to be useful. However the R10B type 33mH coils have acceptable Q for the low-pass part of this filter. Bandpass filter components need not have high accuracy.

Audio Amplifier

From the volume control, Q6 amplifies the audio signal by about 100. Good noise performance is needed here, especially at low volume. The 2N4403 PNP transistor gives a noise figure of about 1dB, with a 500 ohm source resistance, and biassed at about 1ma. Only exotic, expensive low-noise op-amps can give better performance than this $0.16 transistor. However, the integrated circuit amplifiers are often better at rejecting noise and hum present on the power supply.Q5 is added to isolate Q6 from power supply hum, noise and feedback. A non-critical op-amp follows (U1a) with a gain of 16, and provides nearly rail-to-rail output voltage swing. Figure 5 shows the audio amplifier, SCAF and the power output stage.

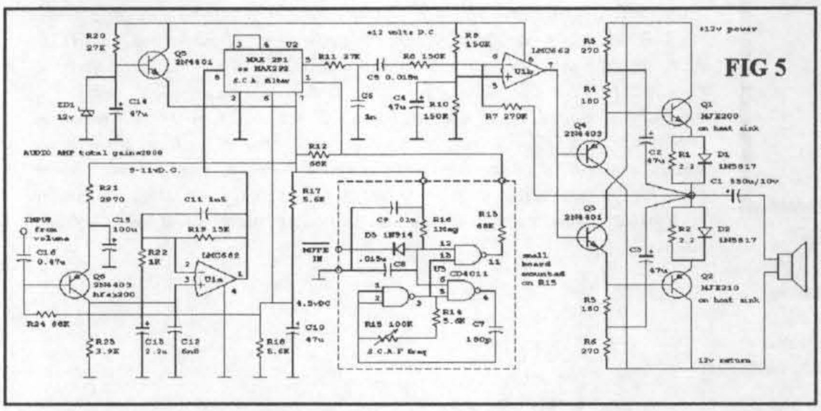
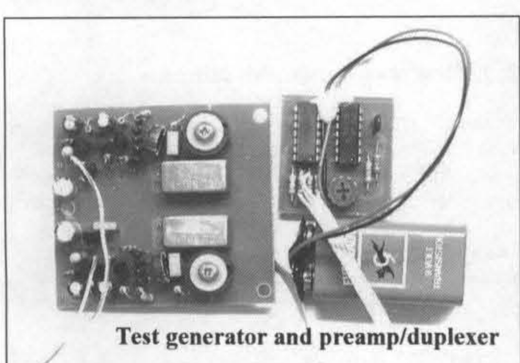


FIG 5

Switched Capacitor Audio Filter (SCAF) U2

These devices are so easy to apply - they're impossible to pass up. And the variable lowpass tuning that they make possible is a great bonus. Unfortunately, these devices are relatively noisy, necessitating their use at high amplitudes. You have a choice of pin compatible filters to plug in here:



Test generator and preamp/duplexer

MAX292 - Bessel filter for best rise & fall shape with no ringing whatsoever. (not recommended)

MAX291 - Butterworth filter for flat frequency response and a little ringing. (recommended)

MAX293 - Elliptic filter for steep stopband slope but more ringing. (recommended)

MAX294 - Elliptic filter giving even steeper slope

All are available from Digi-Key for about $6.

Instead of using the built-in oscillator, the SCAF filter chip is driven from an external variable-frequency oscillator (R15, a front-panel variable resistor sets the frequency). U3 oscillates at exactly 100 times the cutoff frequency, providing a continuously variable filter from 350 Hz to 4000 Hz. This oscillator can be gated on and off with one of the CMOS logic inputs. With no clock, the SCAF filter stops in its tracks, holding its output voltage constant. This makes an extremely clean mute. R16, C8 and D3 have a fast attack, and slow decay (about 10msec) appropriate for break-in keying. The R16-C8 time constant can be easily changed if you need a mute with a longer tail.

Power Amplifier

Because the SCAF filter works best with high-level signals, the power amplifier needs little voltage gain, but lots of current gain. All the common integrated circuit power amps have too much voltage gain to be useful here. Q3, Q4, Q1, Q2 have a composite voltage gain of one.This circuit self-biases to a quiescent current (class AB) of about 16 ma.

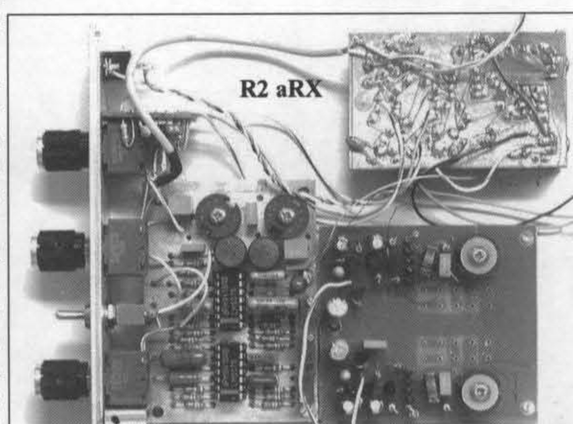
Since the SCAF filter must run from a lower supply voltage, a tiny bit of gain is needed. Op-amp U1b supplies this gain, and reduces distortion to a very low level. The combination of a rail-to-rail op-amp with the bootstrapped output stage (C2, C3) result in an output swing nearly equal to the supply voltage. Be warned: with no AGC this amplifier will make you jump when an unexpected QRO signal arrives in the passband.

The high currents drawn by the amplifier must be routed carefully to avoid howling oscillations. Ground loops are difficult to avoid when dealing with such high overall audio gains. The collectors of Q1, Q2, Q3, Q4 and resistors R3, R6 should be connected to the supply voltage separately from the rest of the receiver. The "ground" lead of the speaker should be connected to exactly the same point as the collector of Q2 and Q4. A grounded phono plug connection to the speaker is asking for trouble, if the chassis is attached to any other part of the receiver. Fortunately, the SBL-1 mixer ground-isolates the local oscillator and radio-frequency input stages, so you shouldn't have to worry about ground-isolating these inputs. A headphone jack, if used, should be ground-isolated too.

The LMC662 CMOS op-amp sets an upper power supply limit of 16v. High performance eclipsed low-power operation as a design goal - 100mA total current is drawn from the supply - more at high audio levels.

Conclusions

High-gain audio amplification needed by direct conversion receivers will always be difficult to deal with. Microphonics in the low-level preamp are a problem. Hum pickup by the diplexer's magnetic components and ground loops are problems too: a ferrite-free diplexer is a tempting future project. While the square-wave quadrature generator allows optimum trimpot tuning, the procedure is still not for the faint-of-heart: it should be attempted by experienced homebrewers.



R2 aRX

Excellent sideband suppression of 50 dB has been achieved with a little extra tuning. The resulting receiver has contest-quality characteristics, with good dynamic range and few spurious responses. The sharp cutoff audio filter can be set very wide (when listening after a CQ) or narrow, for QRM elimination. A bandpass response would be better, but would require another front panel control to set the centre frequency. Break-in muting is excellent, when the time-constant is matched with the transmitter.

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