Quadrature Detection: Biaural Reception of Radio Signals
Notes compiled with aid of ARRL HB 2005 Ed: by Dannie Ray Jackson
{ I / Q Signal Phasing }

A radio receiver amplifies the weak rf carrier signal and then down converts it to an intermediate frequency (i.f.) from which a diode detector or balanced mixer detector circuit takes the audio intelligence off of the rf carrier and amplifies it for final output to a speaker.

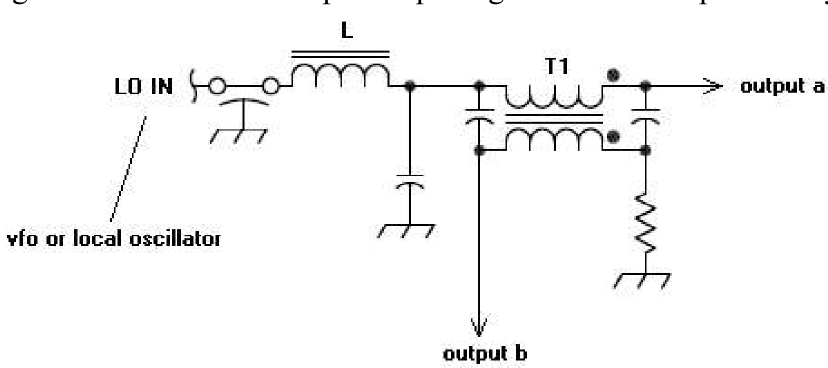
The intermediate frequency then is a mid point between the high frequency rf carrier and the lower frequency audio signal that is impressed upon the carrier via modulation of the carrier amplitude with audio intelligence.

It is possible to down convert the rf signal carrier all the way down into the audio frequency spectrum (without detection of the audio) and have an audio spectrum intermediate frequency or i.f. that can pass directly into a PC sound card and be treated as a radio frequency carrier. And so in the realm of Digital Radio there is a mode called Software Defined Radio. The Pc then is the detector / demodulator section with agc.

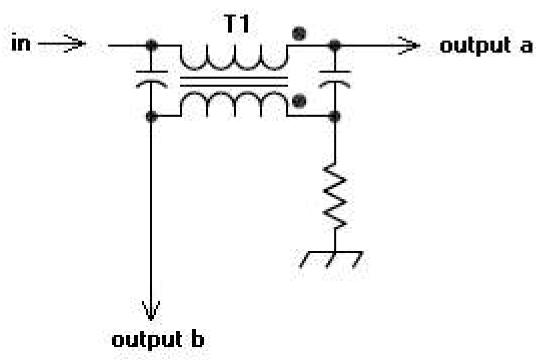
In SDR as it is called, the down converted radio signal is reduced down to a audio frequency intermediate frequency that can be sent into the sound card via the line level input and acted upon by Digital Signal Processing (DSP) software that does the jobs of detection as well as adding a bfo or beat frequency oscillator to detect single side band as well as AM and even FM.

Because of the gain of the sound card a simple receiver that uses direct conversion at the antenna input without the need for rf pre amplification can be used. Hence the circuitry is almost as simple as a regenerative receiver. The receiver set up concepts and the look of the software however appear to be very high tech.

Since direct conversion can be used the idea of the circuit is very similar to the idea of a regenerative receiver. The difference is that we have the incoming signal split off into two separate mixers that have a vfo or local oscillator signal also split into two, 90° out of phase signals. We'll look at the phase split signal and then explain "why" it is needed.



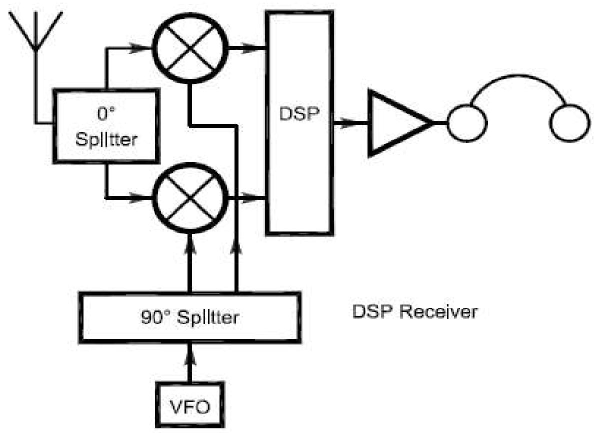
T1 is the phase shift splitter.



To make sure you understand how the vfo or local oscillator signal is split 90º out of phase, we will look at T1 which does a simple job of splitting the signal and phase shifting the two split outputs.

Here T1 is usually a bifilar wound transformer coil on a toroid core.

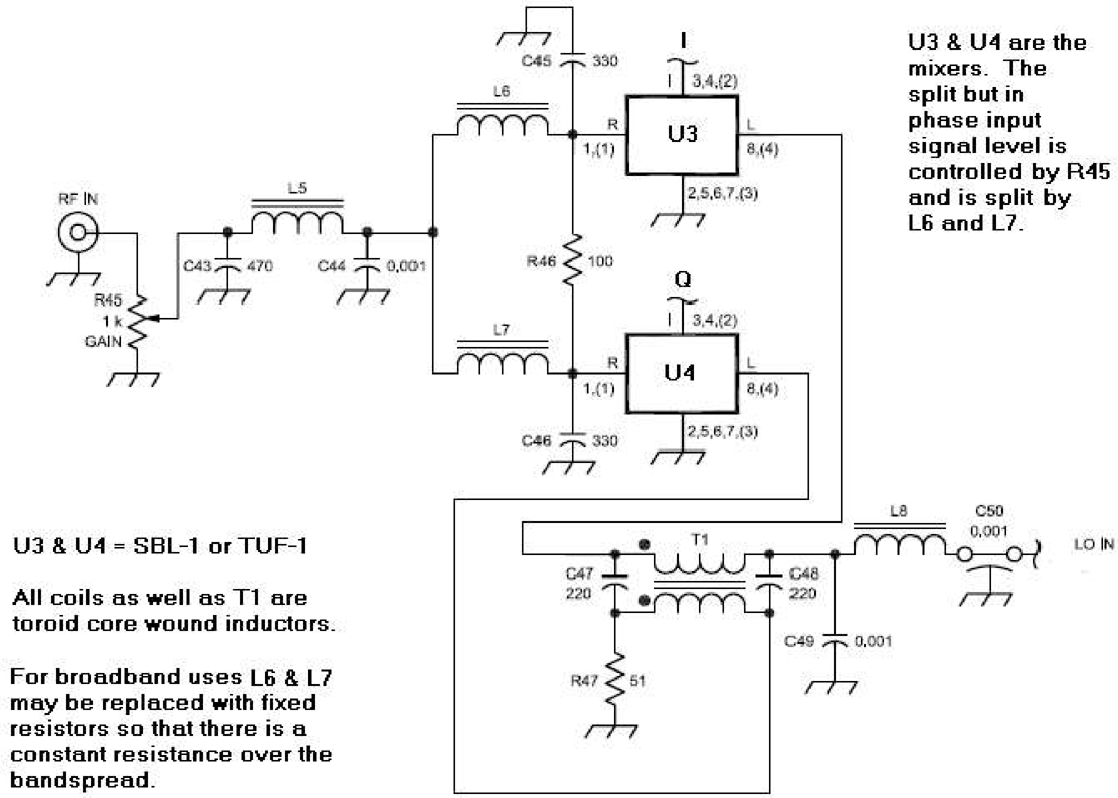
The incoming signal from either a vfo or a crystal oscillator is split by the transformer and one of the split off signals travels through the transformer to output a. The other half is taken off of the other winding that has its signal induced 90º out of phase with the signal of output a. and hence is output at b.



Both of the outputs go to separate mixers that combine with a split in two but in phase radio signal from an antenna. The LO IN signal to T1 is at the same frequency as the radio signal and beats directly as the fundamental beat frequency. Two 90º out of phase outputs in the audio frequency spectrum are created by this scheme. These two 90º out of phase signals will go to the right and left channel inputs of the PC sound card.

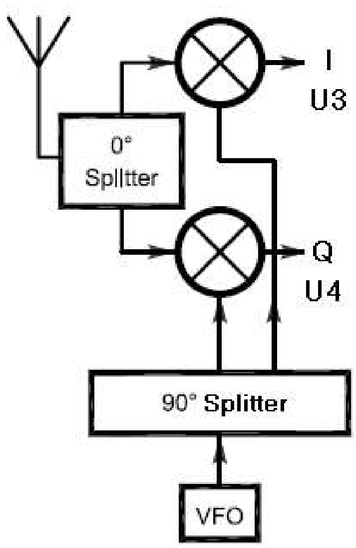
The reason for this splitting and phase shifting scheme is that information about the right and left audio side bands can be separated out of this arrangement by the software of the PC. This then makes it possible for the Software Defined Radio software to demodulate and give us AM, Single Sideband as well as CW and FM radio intelligence in the speakers of the PC. {The process is known as Digital Signal Processing or DSP.}

The reason I explained it this way was to first show you how easy it is to produce what is known as output I, and output Q, or I / Q port signals. When the signal is converted to two 90º phase shifted components we have what is known as quadrature detection: as contrasted to diode detection of AM or balanced mixer detection of single sideband.



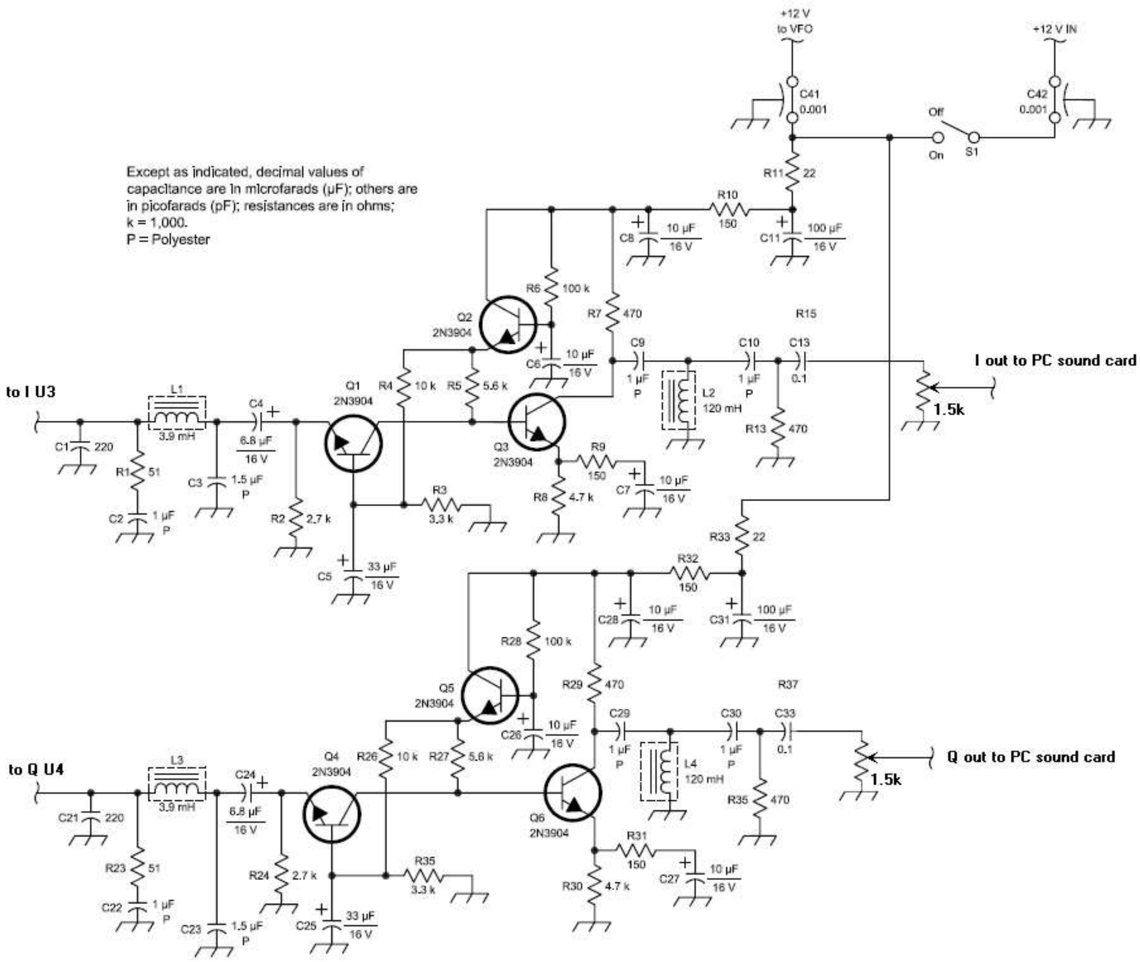
This little receiver was designed and built by Rick Campbell, KK7B. It was first described in the March 1999 issue of QST. It consist of a few more sections, the above is merely the front end input section. It however can be used as a stand alone simple receiver section or as a detector section in a larger super heterodyne receiver scheme.

Here T1 is shown beneath the mixers so as to get all of the circuit on this page with enough room and size for the details of the parts values. A list of all parts will be given later. The circuit is quite simple and can fit on a printed circuit board area of about 3"x 3" minus the vfo or local oscillator.

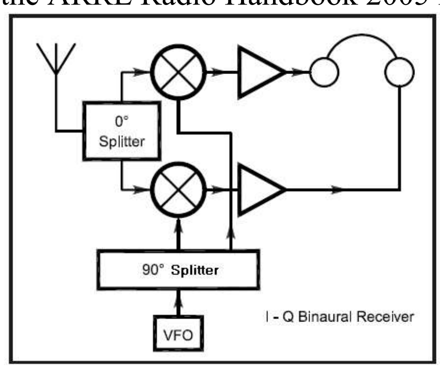


The incoming signal is split but is in phase or has a 0º phase difference. The Local Oscillator Input (LO In) from a vfo or crystal oscillator or even a phase lock loop oscillator goes to T1 and is split and phase shifted 90º and then goes to mixers U3 & U4 after which the resultant i.f. products are shifted also 90º from each other and exit as I & Q.

Please remember to enlarge these circuits on your PC to see the details of the component parts and to print out a copy of this text with your printer if you seriously feel like you wish to home brew this circuit.

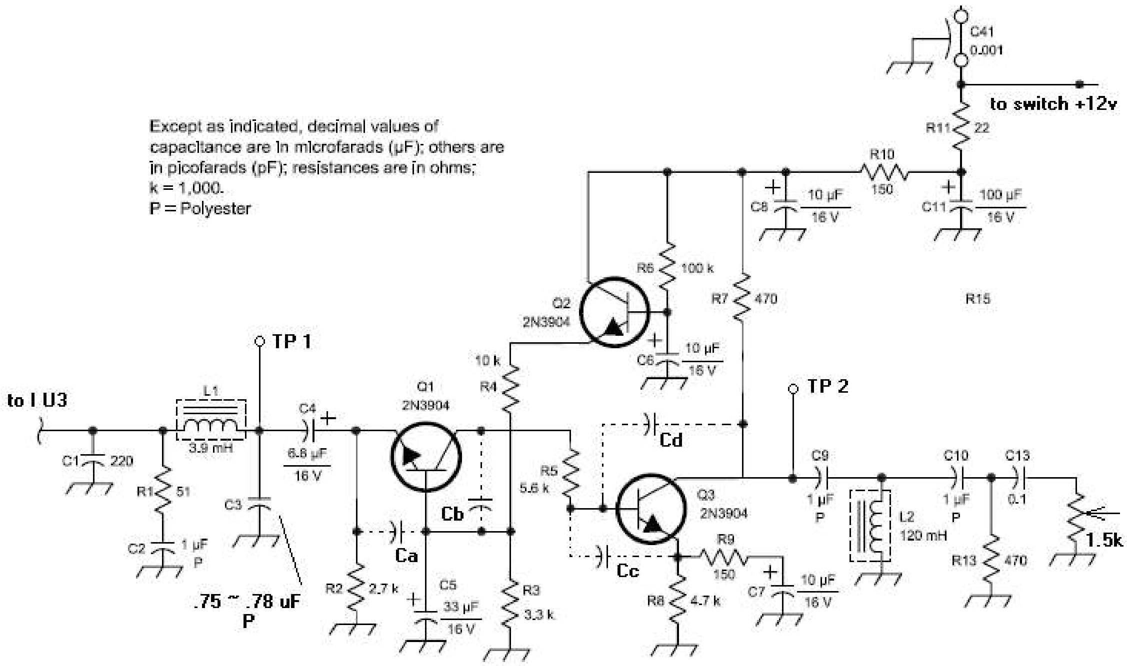


Here is the original biaural receiver circuitry that follows the mixers. Here I and Q are amplified by two identical stereo audio amplifier sections. This circuit was originally followed by a higher power audio amplifier section so that the audio can be sent to a pair of head phones. The original idea here was to allow the brain to do its own job of processing two stereo versions of a tuned to station where the brain is capable of precessing stereo sound information. {This circuit description is offered in Chapter 14 of the ARRL Radio Handbook 2005 Edition under "A Binaural I-Q Receiver" page 14.66}



The block diagram of a binaural I-Q receiver that allows the ear/brain combination to process the detector output, resulting in stereo-like reception.

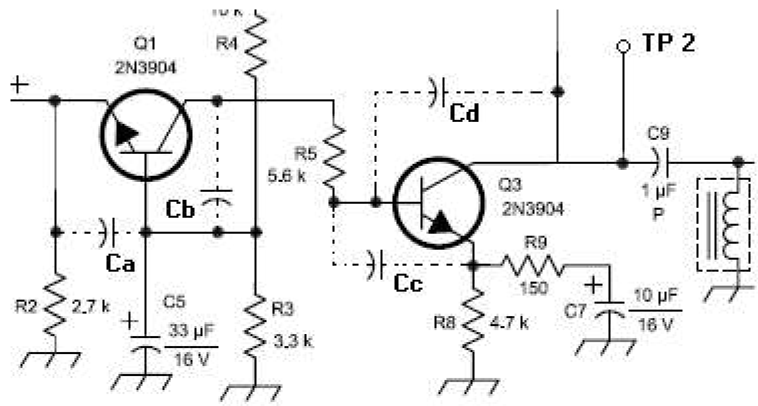
This is the original scheme behind the circuit that led to this simple to construct idea to obtain quadrature signals which are what PC based SDR programs require to operate as a radio receiver section.



Here is an exploded view of one of the audio sections. With both audio sections being identical, you can think here in terms of stereo and since you will utilize the stereo line level inputs you have to look at the circuit as both an audio circuit and as a radio band pass circuit. Thus, the broader the band pass the better. How we improve the band pass is via feedback to improve the circuits square wave characteristics with a good square wave response of at least 3 kHz minimum. With a good square wave response of at least 3 kHz the 11th order sine wave response will be 33 kHz: defining how far the sine wave response will be before sine wave reproduction accuracy ceases. A trailing sine wave from an amp with this response will be down to 50 kHz typically before the amp ceases to pass any higher frequencies. {This may not always work in every circuit idea used.}

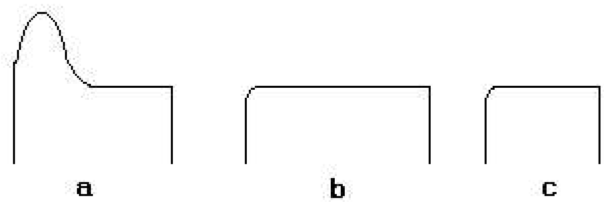
If the 11th order response is at 33 kHz then with the two 90º signals acting as a plus and minus or upper and lower side bands, the spectrum bandwidth then is +/- 33 kHz or 66 kHz wide.

Here is the spectrum display of the beta version of Winrad 0.90 SDR software by Alberto (call i2phd) at http://www.weaksignals.com and represents how the spectrum of the software may look when used with the I Q receiver. [3 kHz upper side band in blue.]



Feedback methods.

After the construction of this circuit it will need to be checked for its square wave response and so an audio generator is connected to TP 1 (test point 1) through a 560 Ω resistor and the square wave function is selected for around 750 Hz to start. The test signal is taken off of TP 2 and viewed on an oscilloscope.



At around 750 Hz or somewheres below you will see that the wave shape at TP 2 is somewhat similar to the square wave at **a.**

Advance the frequency until you start to see a wave like that of **b**. As you advance the frequency you will have to increase the the horizontal sweep rate to keep the square wave in a proportional size since it gets smaller with frequency. At **c**. is a depiction of the wave trace getting smaller as you go up frequency.

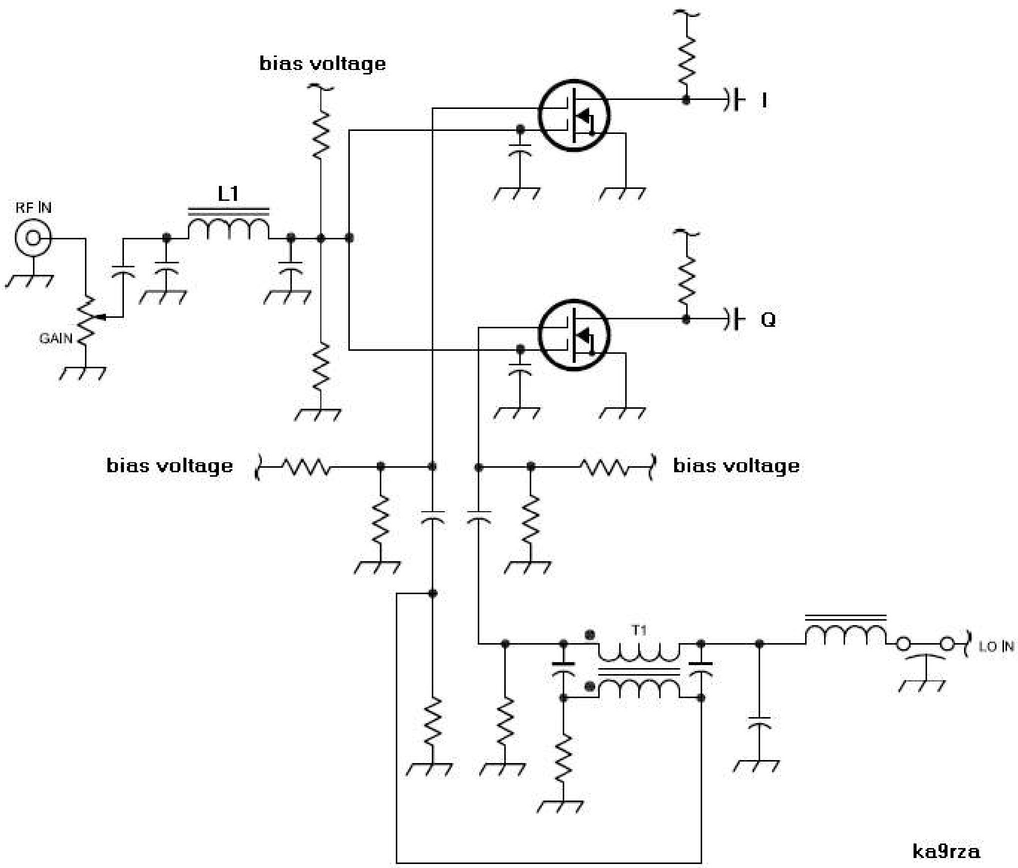
As you increase the frequency at some point you will start to loose the nice squared corner in the right up corner of figures b. and c. The frequency just before you loose this corner is your square wave response limit. The left hand corner after you leave the audio bass frequency region of figure **a**. begins to look like the left hand corners in figures **b.** and **c.** And so it is the other corner that you keep an eye on.

The method that you use to apply feedback is via the use of capacitors for **Ca, Cb, Cc,** and **Cd.** A capacitance substitution box helps. Or you may use a 1 µf to 10 µf capacitor in them all and use a series resistor with each. First of all using a capacitor and resistor as the equivalent of Ca you adjust the resistor until you have effected a better higher frequency for your square wave response limit. In testing this you may use a carbon composition potentiometer as your resistance substitution since it is adjustable. When you find the value that gives you a better higher frequency on your square wave then you measure the resistance of the potentiometer and use a fixed resistor of the closest value measured on the pot. Continue to experiment with the other feedback capacitor positions until you have a better response. If your circuit already works to 3 kHz then no need to do all of this.

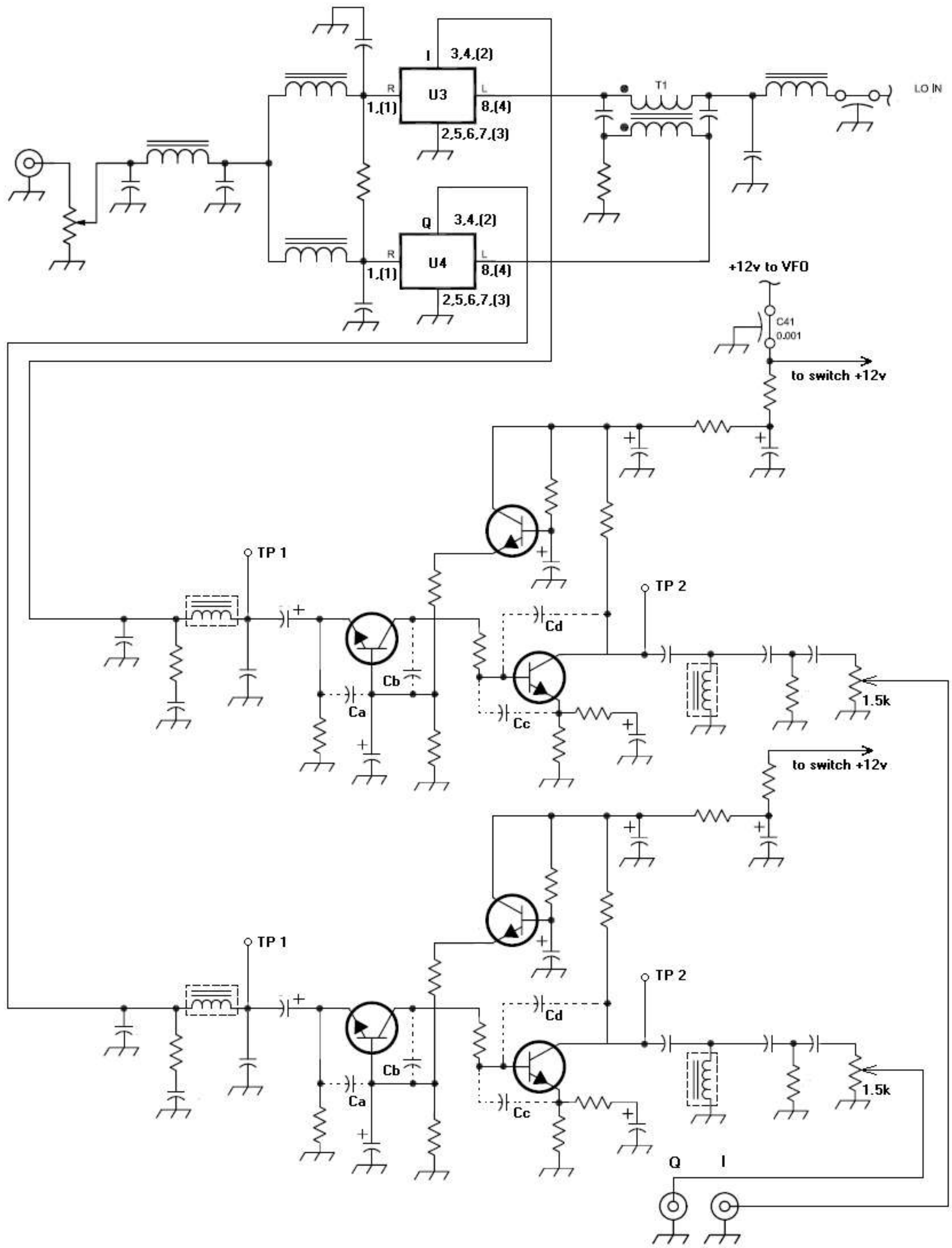
After looking at the ideas for circuits for quadrature detection on the Web and reading numerous articles and finding most of it allusive and cloaked in mystery I resorted to a more simpler circuit idea such as this one of Rick Campbell, KK7B. With it I could better explain to myself and others the circuit ideas for quadrature detection and make it all very simple instead of the secret initiate society thing some want to make it out to be on the Internet: to sell you on the idea of buying an overly complex and expensive chip with expensive registered software to go with it. Or a chip that has DSP software pre programmed into it. Which I suppose is fine if you want a portable digital mode radio.

I did find a very interesting kit called the Soft Rock 40 by Tony Parks which uses a similar simple idea but with a few added complexities but not too complex. This is a surface mount component type kit.

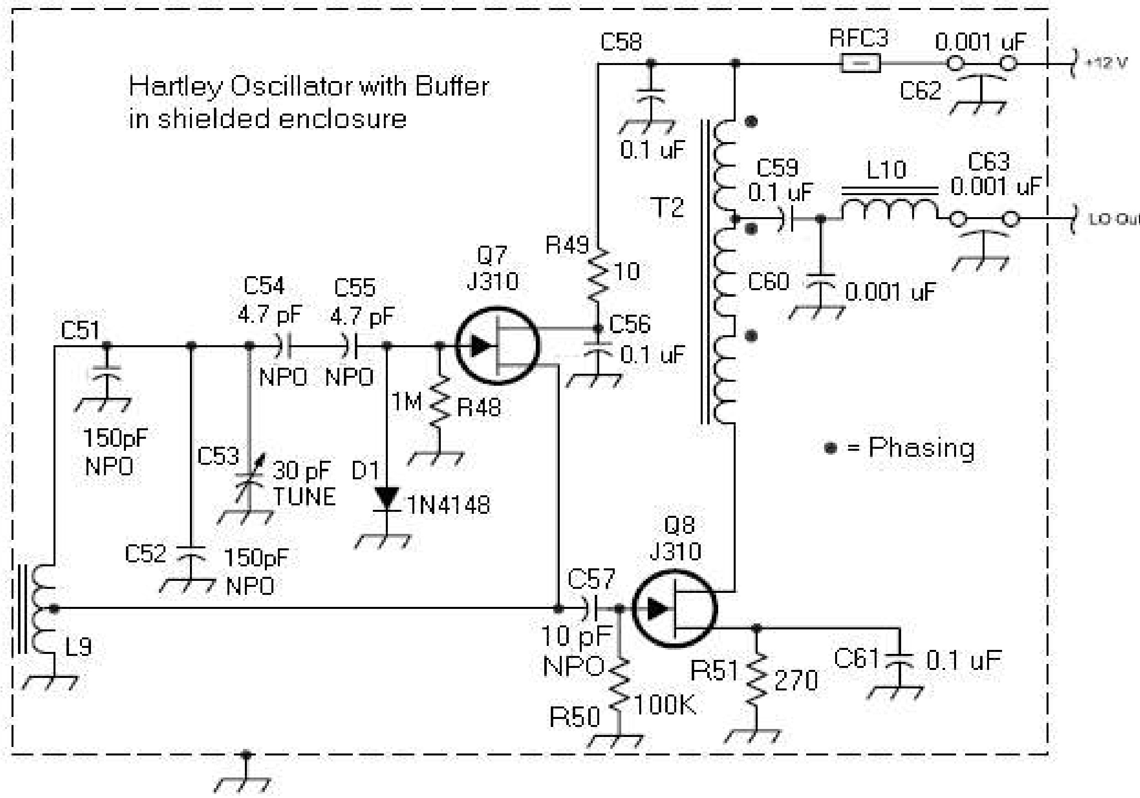
I however like kits that do not have surface mount devices in them so that it is allot more effective at making repairs. Also I like to use DIP sockets with standard sized DIP chips so I can plug in the chip and if it ever blows I can unplug and pop in another one. Surface mount has its right full place in portable hand held devices, PC's and in satellite circuity it however is impractical for use in the house where size is not a problem. Us people who wear glasses do not like to strain to see an electronic part. And some of us are not going to rush out to buy a stereo microscope to build and work on surface mount circuitry.



Mosfet Quadrature Mixer



If you come up with a good circuit as a spin off of this one or have come up with a method of feedback with certain defined values for Ca, Cb, Cc, and Cd and wish to share these items then email me a pdf copy of your diagram and description and I will expand this text with as much information as is helpful in the future. Including hints and tips you may have. Daniel Jackson wavelengths@netzero.com



VFO for the circuit that operates from 7.0 t o7.3 MHz
see ARRL HB 2005 Ch 14

Last things: As these types of receivers become more developed and more common their use in the general public will catch on and make a good item for marketing to the shortwave listener as well as CB'ers not to mention the interest amateur radio operators have in such items.

There definitely will be a market for such receivers but those of you who keep it simple and inexpensive will realize more of a market potential so do not go adding PC control to to the vfo or serial interfacing of the PC to the receiver board. The only connections to the receiver and the PC should remain as they are, the line level inputs to the sound card and nothing more. All of the rest of the set up is in the software and there are already some very high tech freeware versions of software to choose from so do not go buying the software and that which is in existence now, will remain floating around a long time even if others come along with some expensive to obtain newer versions.

So do not be overly complex, make a stable vfo for this or if you want, make this a detector to place inside of another receiver at a take off point before the crystal filter. Please consult the ARRL 2005 Handbook for more details on the Biaural Receiver.

Dannie Ray Jackson ka9rza
2058 Turkey Creek Road
Barbourville Kentucky USA 40906
wavelengths@netzero.com

Notes:

The oscillator must be shielded if you use it on the fundamental frequency so that reception of the signal does not feedback into the antenna circuitry.

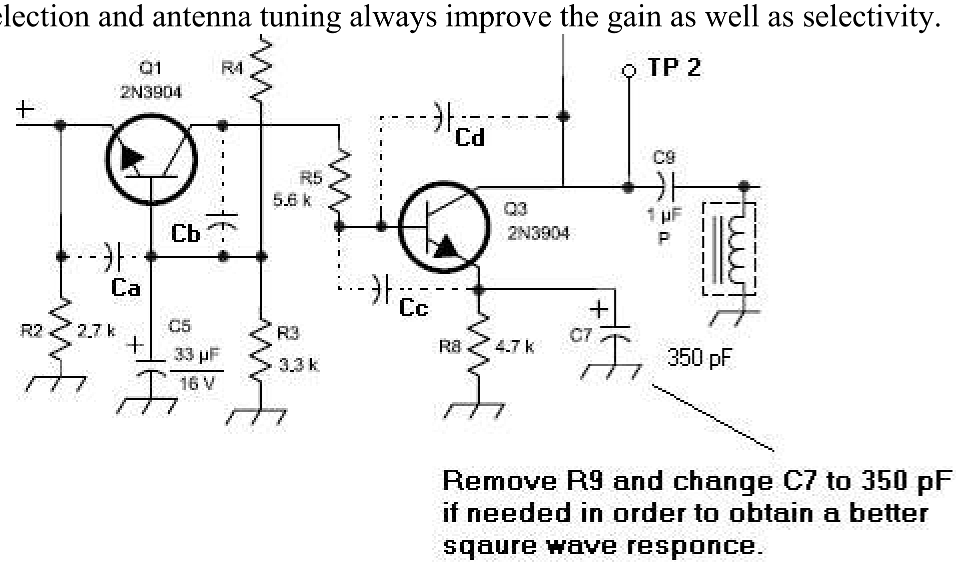
If the vfo is used on the fundamental, then it can be read directly with a frequency counter.

A vfo made to operate at a lower frequency can be more stable once you work out the thermal characteristics of the chosen capacitors. Just use a crystal oscillator / mixer to convert the signal up to the fundamental and use a frequency counter as your frequency readout. Use a variable capacitor with strong thick plates and wide spacing of plates to reduce expansion of mechanical parts due to heat. Some very small 5 and 6 digit frequency counters can be obtained for $50 to $70 at places such as Coppers Electronics.

I have a 1972 era receiver that contains two solid state pll controlled vfo's and the circuitry is very simple as compared to other schemes I have seen for pll control of a vfo. In a future version of this text I will place the schematic as well as parts list herein. I use the receiver daily and it does not drift.

If you want to get technical and add in allot of front end selectivity you can do so with a 4 section ganged capacitor and make band pass circuits for each band on the antenna input. This will give you a pre selector that will peak up the rf allot.

In a broad banded input to a front end mixer or transistor, the gain of the device is spread out over a wide range of in coming frequencies from the antenna that can span many mega Hertz of spectrum and this is all being amplified by the preamp and hence the gain is spread out over undesired frequencies. The transistor has a fixed amount of energy usage and gain; and all of this undesired energy does subtract from the desired frequency energy. Total gain of the stage is never realized in a broadband type input. So pre selection and antenna tuning always improve the gain as well as selectivity.



If you can not make an improvement in the square wave with lets say Ca then move onto Cb etc.

For the ultimate in “receiving purposes” no antenna can out perform a one meter diameter low noise loop antenna. It places a minimum of 20 dB noise rejection at the antenna end as well as directivity and even selectivity since it requires variable capacitor tuning. The dipole can't do this neither can a beam antenna.

Send me your ideas and I will expand this text. Include circuits and your name and call letters. You do not have to be a ham either, you can be a hobbyist.

I have another paper that deals with theoretical Mosfet circuits for quadrature detection where there are ideas for T1 that can be explored for feasibility and some variations on the idea of a Mosfet based quadrature mixer circuit that can be tried in order to perfect a simple and replicate-able mixer circuit that has a little gain in it as well as low noise. “Analyzing Theoretical Mosfet Quadrature Mixers.” On page 7 of this article is one of the circuit ideas.

Daniel

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