Third-Method Narrowband Direct Upconverter for the LF / MF Bands

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Introduction

Previous designs for upconverters from audio generated from a soundcard to RF have been published at [1] and [2],   but these single conversion designs all suffered from a level of sideband and carrier suppression that was barely acceptable for use on the LF bands.   Unwanted sidebands, although only perhaps 2kHz away from the wanted signal (for a 1kHz drive tone) and typically 40dB to 50dB down could easily be out of band.   Carrier leakage at 1kHz away for the same tone , unless great care has been taken over optimising carrier leakage and drive levels, may be only -30 to -40dB

Such out of band transmissions are unacceptable, especially when used with high power amplifiers, so the design here uses a different method of upconversion that ensures sideband suppression and carrier leakage are not important.

Third Method Sideband Generation

This is a technique familiar from the early days of SSB where the input audio is first downconverted to baseband.   In that a 300 – 3300Hz speech band is mixed with tone of 1800Hz and folded back on itself to give a range of frequencies from –(minus)1500Hz to +1500Hz. By using quadrature mixing, ie. mixing the audio with 0° and 90° phase shifted versions of the 1800Hz tone to give two I and Q channels, the complete audio band can subsequently be recovered and put back together again properly at the next stage of upconversion to RF in a quadrature mixer

The advantage of this method is that there is now no need for a narrow 3kHz bandwidth filter at RF or IF to separate out one sideband.   At baseband, two identical 1500Hz Low Pass filters on the I and Q channels do the job.   The other main advantage is that as the IF is now symmetrical about DC , carrier leakage lies in the middle of the passband with unwanted sidebands directly overlaying the wanted ones.   With a poor sideband rejection or carrier leakage figure of just 20 to 30dB, this level will now go unnoticed when used to generate digital modulation. Even with voice, carrier leakage only appears as a weak 1800Hz tone superimposed on the audio.   The unwanted sideband is usually even less noticeable as it moves in amplitude at the same syllabic rate as the wanted voice signal.   Another advantage of the third-method over conventional phasing SSB exciters is that there is no need for a wideband audio phase shift network to generate 0/90° phase shifted versions of the voice signal. The only quadrature component needed is the first downconverter LO at 1800Hz.

Narrowband Version for Soundcard Data Modes

This upconverter design is specifically for soundcard generated data modes where the audio tone is generated at a user specified tone frequency whose signal bandwidth is no more than one or two hundred Hz.   Most software allows arbitrary selection of the generating tone frequency to permit this restriction. The block diagram is shown in Figure 1 and the complete circuit diagram in Figure 2.

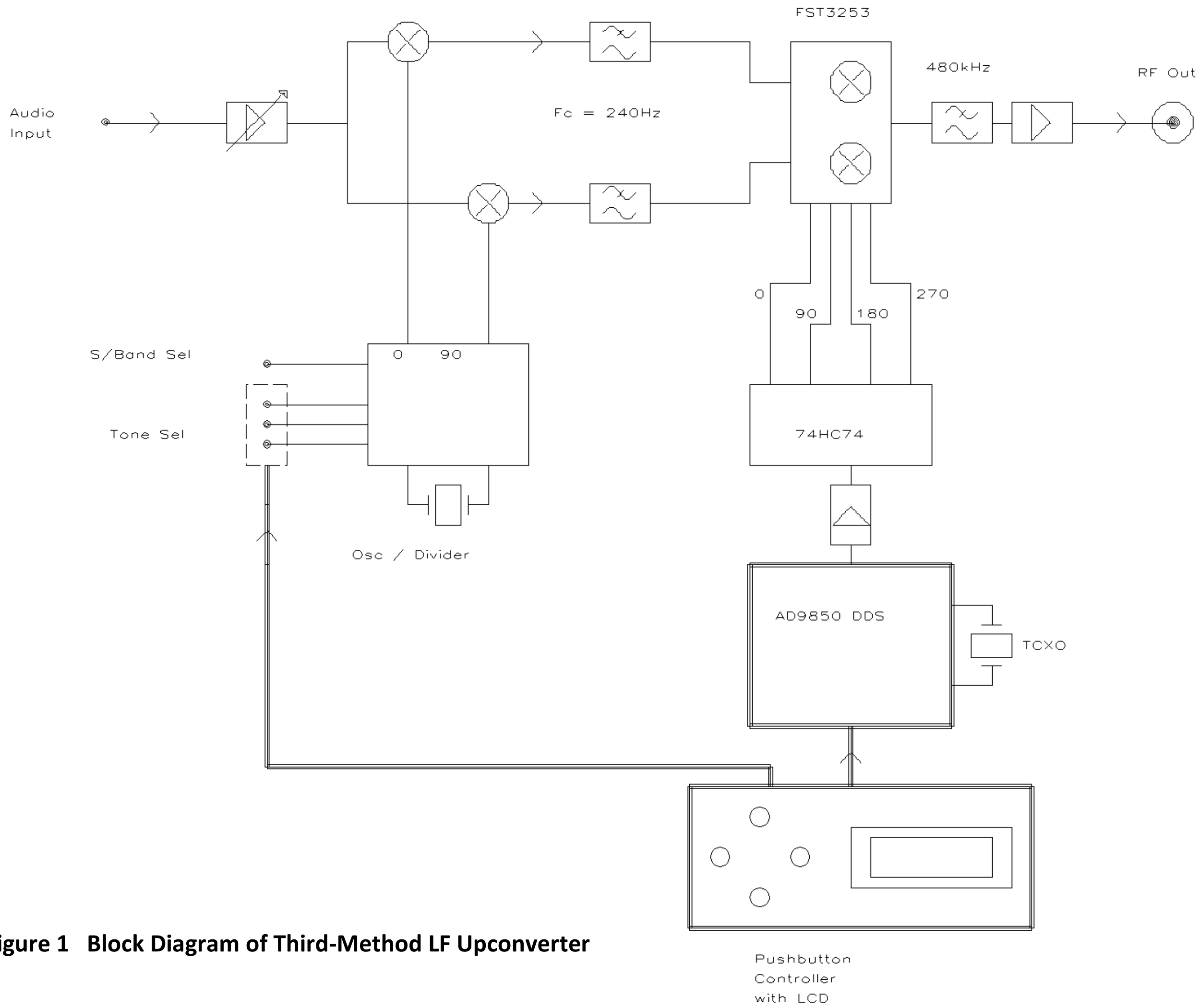


Figure 1 Block Diagram of Third-Method LF Upconverter

Input Mixer / Tone Generator

The first stage of the frequency converter consists of two single-balanced mixers based around opamps whose function is to translate the incoming audio tone down to I and Q baseband channels. The local oscillator generating the quadrature drive for the two mixers is generated by using a PIC as an oscillator/ divider, where a 12.288MHz master clock is divided down to give eight possible tone frequencies. Three pins on the PIC are used to select which one of these is to be generated. The frequencies with their selection codes are listed in Table 1. They may be selected from a binary switch, hard wired, or sent as a three bit parallel control word from another controller.

Code	Frequency Hz
000	600
001	750
010	800
011	1000
100	1200
101	1500
110	1600
111	2000

Another pin on the PIC selects the polarity of the downconversion, LSB or USB, by swapping the phase of the I/Q output drive. While this is something unlikely to be needed during routine operation, it does save having to check and make sure of the overall upconversion process during the building and testing; the link can be either installed or left out at the final test.

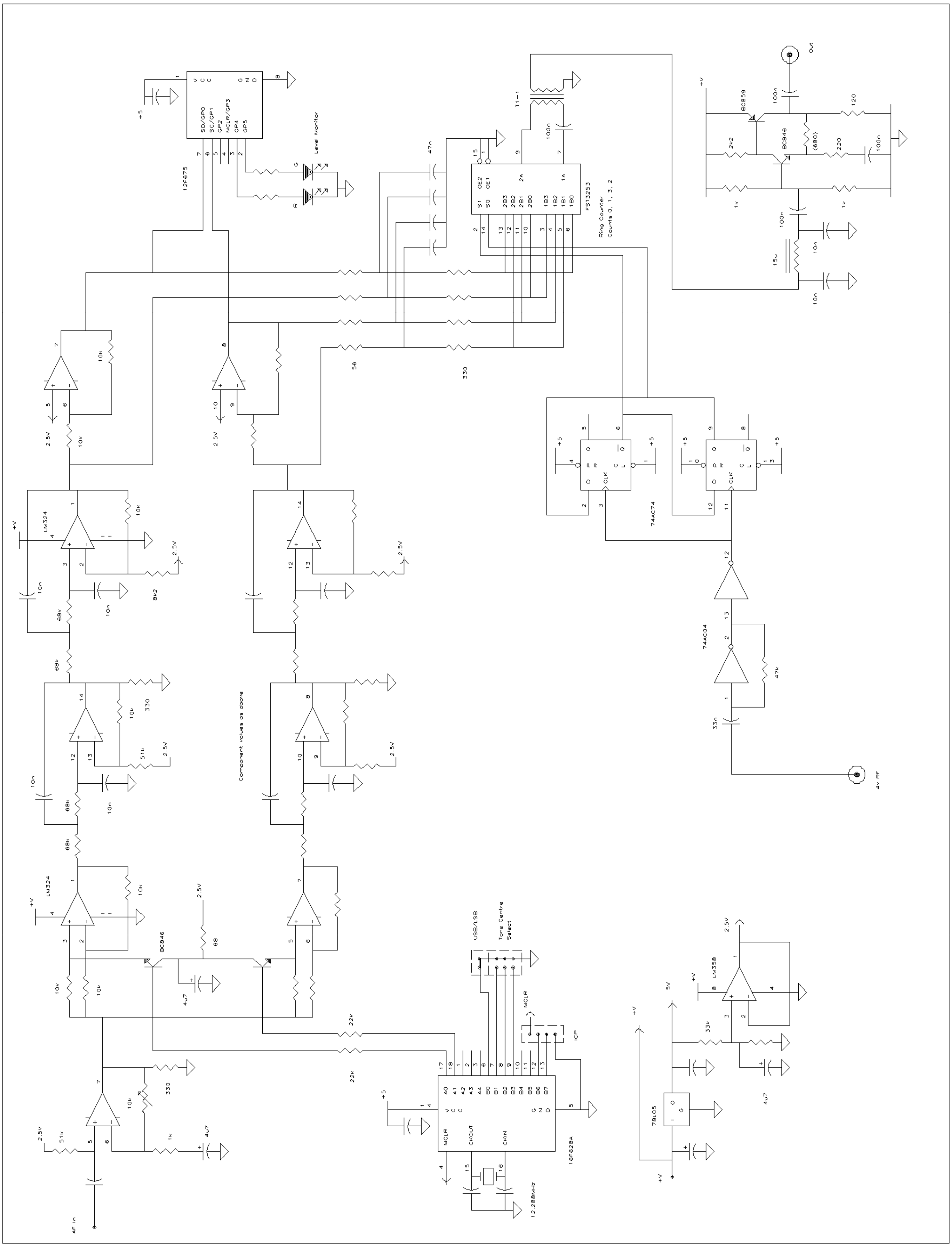


Figure 2 Complete circuit diagram (a higher resolution version can be found at [4])

Instead of the12.288MHz crystal, if a continuously variable audio centre tone is desired, the clock signal can be replaced with a variable source such as a DDS, or even a simple LC oscillator if high stability is not important. Ideally the PIC firmware should then be changed, setting its configuration fuses for 'External Clock input'. In practice, however, just applying the signal to the *ClkIn* pin is sufficient. The PIC code for generating the switch selected quadrature tones can be found at [3].

The audio frequency mixers use opamps, switched to amplify by +1 or -1 depending on the polarity of the square wave drive signal. The switching element is a bipolar transistor, but note the connections of this. At first sight they appear to be upside down, with collector and emitter swapped – and this is very definitely correct! A bipolar device used this way has very low forward gain – sometimes less than unity, but when saturated has exceedingly low $V_{CE(SAT)}$ of just a very few mV, compared with the 0.1V typical of normal operation. The low gain is no problem as a few milliamps of base drive is more than enough to fully saturate at the low audio levels and high impedance used here.

All the opamp stages are biassed to a 'virtual ground' voltage of 2.5V. This is chosen to be at mid rail when it reaches the final bus-switch mixer device which runs from a 5V supply.

RF Upconversion

The output from the audio mixers pass into a pair of identical fourth order low pass filters with a 3dB cutoff of 240Hz. This limits the bandwidth of the possible input to 240Hz either side of the carrier, ie 480Hz overall.

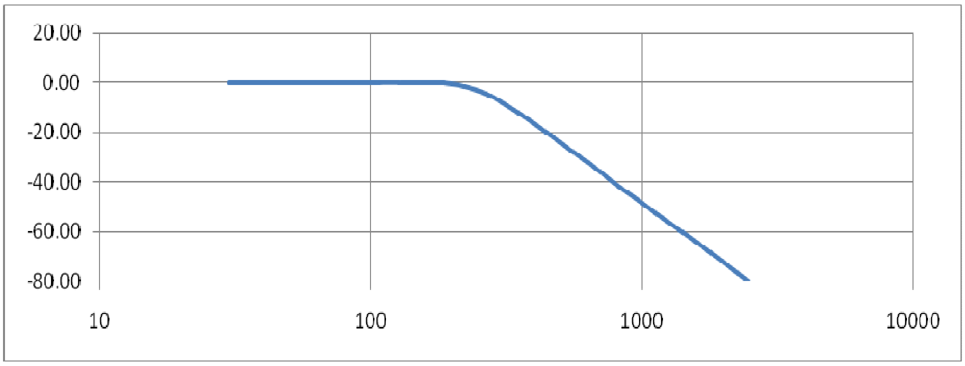


Figure 3 Lowpass Filter Response

The output from the LP filters then go into a quadrature switching FET mixer for upconversion to the final RF. A double balanced design using an FST3253 dual four-way bus switch means the output transformer does not need to be centre tapped.

The mixer output is passed though a low pass filter with a cutoff at 480kHz, followed by a feedback stabilised amplifier stage. The gain of this can be adjusted slightly by altering the 680Ω resistor (value shown in brackets). Output power is around +7dBm before compression becomes significant. For use at output frequencies below 200kHz additional low pass filtering will be needed somewhere in the system. Being a switching mixer, all products associated with odd harmonics of the LO appear in the output so a 480kHz low pass is unsuitable for carrier frequencies of 160kHz or under.

Local Oscillator and Controller

An AD9850 DDS driven from a 12.8MHz Temperature Compensated master oscillator forms the carrier source. This generates at four times the final output frequency, passing though a buffer stage to drive a 74HC74 ring counter for the 0°/90°/180°/270° drive signals. Note that the ring counter generates the sequential codes 00/01/11/10 for the 4-way switch, so the drive signals are not treated in the FST3253 in the same order they appear in other bus-switching mixer designs that use different mixer topologies – like the Softrock.

A PIC controller programmes the DDS, with an LCD displaying the centre frequency of the output signal. As this is a transmit source, the frequency won't need continuous twiddling as would be needed for a receiver. Frequency selection is therefore performed using pushbuttons in 1Hz steps rather than by a rotary encoder suited to higher resolution selection. A Digit-Select button selects the decimal digit to be altered using Up and Down buttons. Pressing and holding Digit-Select saves the current frequency in the memory. Eight memories are provided, numbered Men-1 to mem-8, selected by cycling though using the fourth, 'Mem' button. Pressing and holding this makes the current memory the default when the unit is next turned on.

To allow the input tone to be selected from the front panel, three wires are taken from this controller to the tone generator. To change the input tone centre frequency , apply power to the unit while pressing either the Up or Down button, releasing it only when the LCD shows *"- Set –"*. The tone frequency can then be cycled through its eight possible values using these buttons. The selected tone is stored and normal operation resumed by pressing the 'Mem' button.

PIC code for driving the AD9850 DDS and setting the tone generator can be found at [3]

Drive Level Monitor

To assist in setting the audio drive level correctly, another PIC with internal A/D converter is used as voltage threshold monitor, showing the result on red and green LEDs. It looks at the voltages on the I and Q inputs to the second mixers, comparing these against internally stored thresholds. If the mixer input voltage on either channel is above 4V or below 1V (ie. more than 3V peak-to-peak) the red LED is flashed to indicate an overdrive. (This threshold is actually defined by the maximum swing of the opamp outputs). If the voltage into the mixer is less than 2V or greater than 3V (ie. more than 1V pk-pk) the green LED is illuminated to show satisfactory drive level. Below 1V pk-pk, no LED illuminates. The drive level can be adjusted over a 10:1 range with the 10k input gain control variable resistor. An input drive level of 80mV to 0.5V RMS is suitable for nearly all soundcards and audio interface options. PIC code for the level detector can be found at [3]

Results

Measurements of sideband suppression and carrier leakage are difficult with a third method converter, as these all lie on top of each other. One way to get the values is to drive with a single tone that has been intentionally set off-frequency by a small amount but still falls in the +/-240Hz bandwidth.

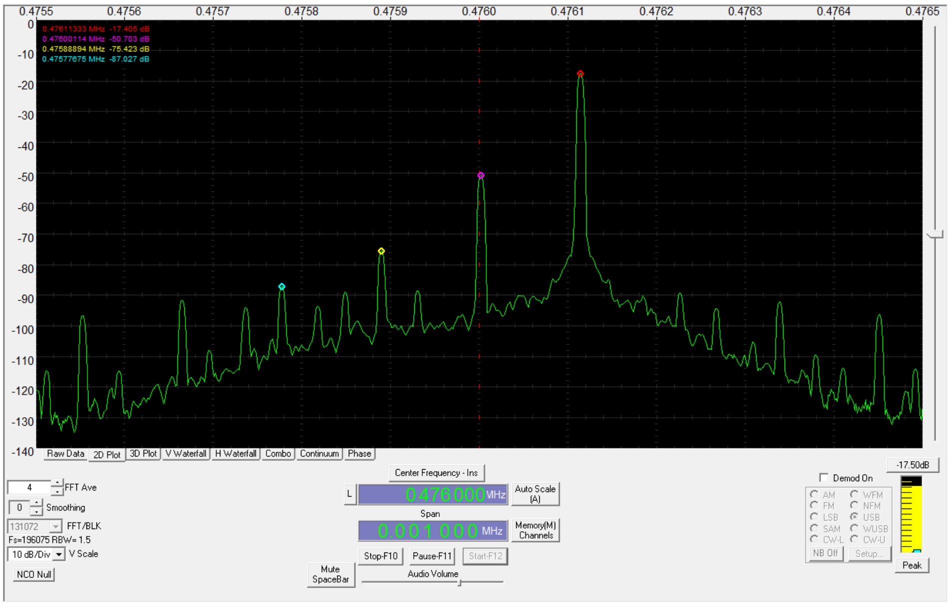


Figure 4 output spectrum when driven by a tone deliberately set above nominal frequency, to show sideband and carrier leakage

Figure 4 shows the output at 476kHz with a drive tone of approximately 1613Hz (113Hz above the designated 1500Hz input). The input level was set so that the red LED was just short of flashing, ie. to the maximum permitted. It can be seen that the carrier leakage at the display centre is 33dB below the wanted (upper) sideband output. When correctly tuned, for most practical modulation types, this carrier leakage lying in the middle of the modulation passband would be completely 'invisible'. The lower sideband is over 50dB down on the wanted, hence even less significant.

Figure 5 shows a WOLF BPSK signal driven 36Hz off frequency. The carrier leakage is barely visible as a fractional dB 'glitch' on the edge of the main PSK lobe. The opposite sideband leakage is completely masked. However :

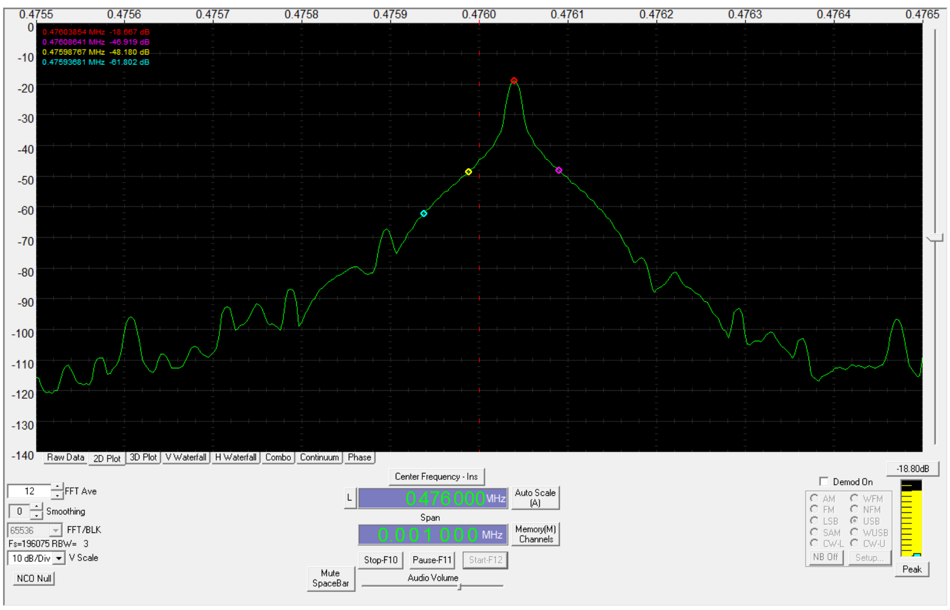
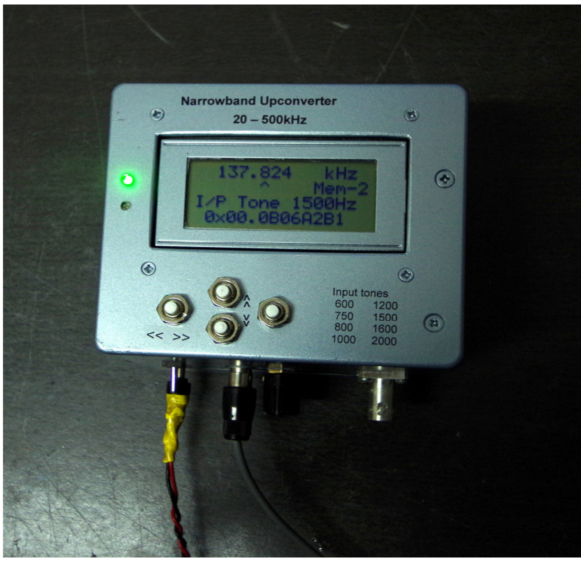


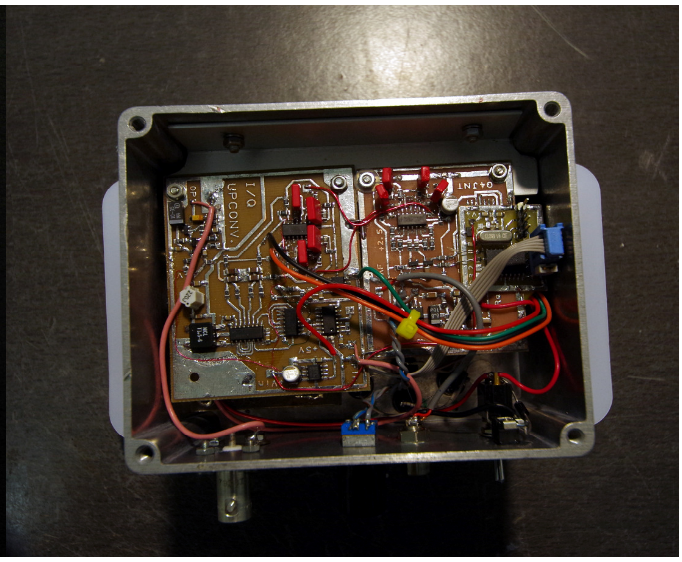
Figure 3 WOLF BPSK delivered as a tone 36Hz high, showing carrier leakage is virtually invisible. The harmonic conversion artefacts can be seen .

The peak that is visible about 100Hz below the centre is an interesting artefact of image cancelling conversion. With square wave local oscillator drive, all odd harmonics of the LO result in conversion of any harmonics of the audio. So the peak that is visible is the third harmonic of the audio converted by 3* LO (which suggests a bit of overdrive may have been present). Now, for a converter set for upper sideband conversion, the phase of the harmonics of the LO scale with the harmonic number. So, if we say the I channel has 0° phase and the Q channel 90°, then 3.I = 0° but 3.Q = 270° which is the same as -90°. So, since the relative phases are now swapped, the opposite sideband 'wins' For an upper sideband conversion, the lower sideband of the third harmonic is enhanced; the upper sideband of the 5th, lower sideband of the 7th and so on. Some of the output plots in [1] and [2] shows these products (although they are very studiously ignored in the text).

Construction

Figures 5 and 6 show my breadboard version, with the converter PCBs visible. For convenience in their design the two separate converters were placed on different PCBs; the LPF ended up being split between them, just to add confusion. The PIC tone generator is also a separate PCB, visible on the right hand side showing the ribbon cable carrying the tone-set code from the main controller. The DDS, controller and level detector are hidden from view under the converter PCBs





References

[1] IQ Upconverters for SDR Transmitters http://www.g4jnt.com/IQConverters.htm

[2] Single Conversion LF Upconverter http://www.g4jnt.com/LFUpconv.pdf

[3] PIC Code for all three devices http://www.g4jnt.com/LFUpconvPicCode.zip

[4] Higher resolution version of Figure 2 http://www.g4jnt.com/lfupcnv3.gif