# Challenges of SSB: the development of the QSX all-mode HF transceiver kit

**Hans Summers, QRP Labs, http://qrp-labs.com**

## Introduction

Last year I spoke at QRP ARCI FDIM about the design of the QCX 5W CW transceiver that was produced for the 2017 Youngsters On The Air summer camp, and subsequently went on general release in August 2017.

This year, my presentation is about the evolution of the design to a new radio, called QSX, which extends the operating modes to include SSB (and more). The jump from CW to SSB involves many significant challenges. This article is about the challenges, the problems, and the solutions; with the design decisions which led to an all-band all-mode 10W HF transceiver combining performance, features and low cost all in one.

I hope that you will find the discussion interesting, and learn something from the outcome of all my mistakes, blood, sweat and tears. Perhaps some of the techniques discussed will be useful in your own projects too.

## The QSX 10W All-mode all-band HF transceiver kit

This article discusses some of the challenges and solutions when designing a SSB transceiver. Practically speaking, the specific design in question is the new QRP Labs QSX 10W all-mode all-band HF transceiver kit. Let's start with a brief description as the basis for the rest of the discussion so that we know what we're talking about.

QSX (for **Q**RP Labs **S**SB **X**cvr) is a very ambitious project which challenges the traditional concepts of amateur radio products: namely, that you can have quality, or cheap, but not both. QSX will deliver a kit radio that is simultaneously:

- High Performance
- Feature packed
- Low cost

It's a combination which has never been seen before at the target price. Achieving this requires very careful design, with attention to optimizing the performance to cost ratio in every part of the design.

The kit was initially designed for the 2018 Youth On The Air (YOTA) summercamp "buildathon" hosted in South Africa in August 2018 by SARL (South African Radio League). They built a mono-band QSX transceiver kit for 40m. They did not, at that time, end up with a working kit on the day, because the firmware development was not completed in time. However, as you will see later, firmware updates are easy.

QSX is based on Software Defined Radio (SDR) techniques with a powerful internal 32-bit ARM processor handling all the required Digital Signal Processing (DSP). Why SDR? That's a good question which we'll come to shorty.

The inclusion of the Digital Signal Processing makes possible a huge array of features. QSX supports SSB, CW, AM, FM and Digital Modes. PSK31 and RTTY can run on the radio (no PC required). You can plug in a USB keyboard, or use the internal CW to PSK31/RTTY translator. Or, you can connect the QSX to a PC via a single USB cable, and it will appear to the PC as a high performance 24-bit USB soundcard, for use with SDR programs or any digital mode programs directly (no other connections between PC and radio required); at the same time, providing a virtual COM port for CAT control via the PC.

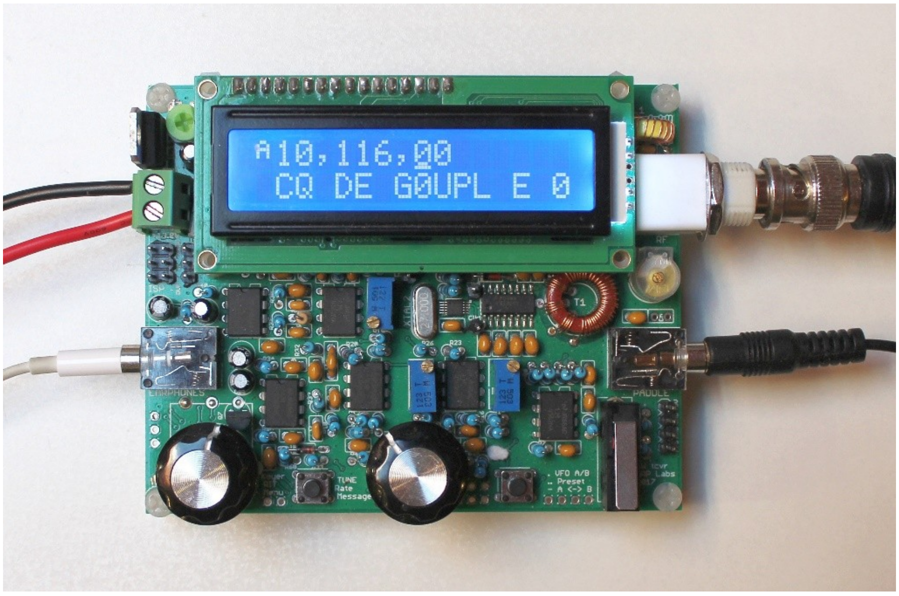
When using the more conventional CW and SSB modes, the QSX includes AGC, noise reduction, notch filter, speech compression (SSB) and full break-in QSK (CW). There is an iambic keyer, CW decoder, message and frequency memories.

An array of built-in test and alignment equipment makes setting up the QSX very easy, with no additional equipment. QSX also includes an SWR bridge.

The QSX will initially be available as a single-band radio for 40m, subsequently an optional 10-band filter module will be available which plugs in to provide coverage from 160m to 10m including 60m. A smart extruded aluminium enclosure will be available which is drilled, cut and silk-screen printed ready for the QSX boards (unlike my hand-made hacked-up version pictured below!).

## Legacy: QCX, the 5W monoband CW transceiver

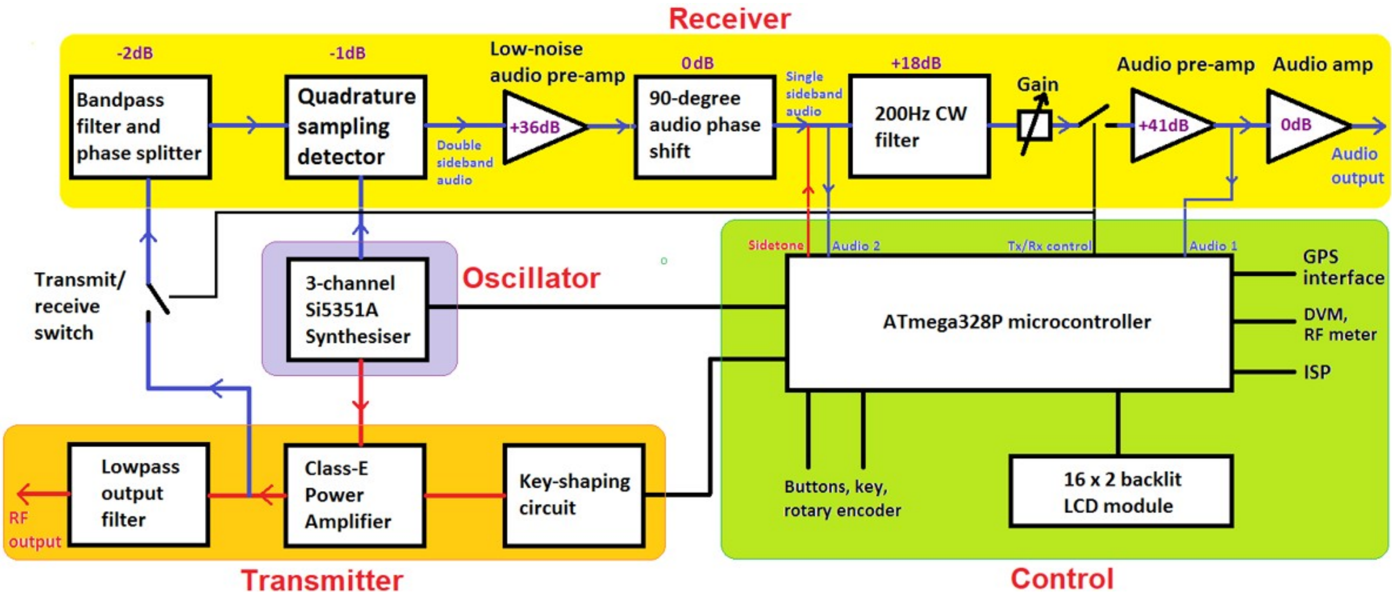
The QCX kit is a tremendously popular mono-band CW transceiver with 5W output power, available for 80, 60, 40, 30, 20 and 17m. This transceiver was designed initially for the 2017 YOTA summercamp buildathon hosted by the RSGB in UK, August 2017. Since then, well over 7,000 of these kits have been sold.




The QCX kit combines very high performance with loads of features, all at the very low price of $49. It includes an iambic keyer, CW decoder, message and frequency memories, and built-in test and alignment tools. A ground-breaking large feature set for a kit radio with such low cost.

After the success of the QCX CW transceiver, it seemed natural to consider SSB as the next step. The initial design of the new QSX radio focused on what could be done to the QCX, to make it SSB-capable? This is where the challenges start! It turns out to be much more difficult to design an SSB radio, than an CW one – particularly if you are determined to define high performance as a fundamental requirement.

Here’s a block diagram of the QCX CW transceiver:




The synthesized Si5351A provides an accurate and stable oscillator signal for both the transmit and receive sections. The transmitter is a Class-E power amplifier followed by a low pass filter for the band of operation; a key-shaping (RF envelope shaping) circuit ensures clean keying with no key-clicks. The Class-E power amplifier is a key feature of the radio, operating with high efficiency typically over 85%, it means that the three little BS170 MOSFETs in parallel are capable of providing 5W output power without needing a heatsink, reducing cost and board area.

For the receiver, the Si5351A produces two signals in quadrature, this means the two signals are on the same frequency but with a 90-degree phase offset. The way the Si5351A is configured for this was described in my FDIM 2018 conference proceedings article last year; you may download a PDF of that article here: [http://qrp-labs.com/images/news/dayton2018/fdim2018.pdf](http://qrp-labs.com/images/news/dayton2018/fdim2018.pdf)

The receiver section is an all-analogue design consisting of a band pass filter, followed by a double-balanced Quadrature Sampling Detector (a.k.a. Tayloe Detector), which offers very high performance, with high dynamic range and high 3$^{rd}$ order intercept (IP3), as well as low insertion loss. +36dB audio pre-amps amplify the I & Q signals which then pass through a 90-degree active all-pass audio phase shift network. The resulting signals are summed, which has the magical result of reinforcing the wanted sideband, but cancelling the unwanted one. Once everything is adjusted properly (using the built-in alignment tools), 50 or even 60dB of unwanted sideband cancellation are easily possible.

The narrow 200Hz CW filter follows, centered on 700Hz; then a conventional potentiometer as the gain control, and final amplification to a level suitable for driving earphones.

An ATmega328 microcontroller with 16 x 2 LCD module, rotary encoder and two buttons provides the brains of the rig, it configures the Si5351A synthesizer but also does much, much more, sequencing the transmit/receive switchover, providing the iambic keyer and CW decoder, memories, and all the built-in alignment tools via sampling of the audio using the microcontroller's built-in 10-bit ADC peripherals.

## Evolution to SSB

Now that's a brief description of the QCX… next follows consideration of how we could convert that to SSB operation, which is in fact a list of reasons why it is difficult.

## Power Amplifier

The first problem is the power amplifier. The QCX transceiver has a highly efficient Class-E amplifier. It’s perfect for a low-cost CW transceiver because only small transistors are needed and no heatsink. But it’s also highly non-linear; harmonics can be removed with an effective 7-element Low Pass Filter, but this is unsuitable for an SSB transmitter. In an SSB transmitter any non-linearity creates intermodulation distortion, which widens the frequency bandwidth, it is known as “Splatter” because your signal splatters onto adjacent frequencies. This is a waste of power and of course, hugely disruptive to any users of nearby frequencies. For an SSB transmitter we need a proper Linear Power Amplifier, which is considerably more complex.

## Wide-band 90-degree phase shift

QCX is a CW transceiver utilizing direct conversion and the phasing-method unwanted sideband cancellation. This requires a 90-degree phase shift of the audio and oscillator (or RF) frequencies. The oscillator is covered, using the Si5351A in quadrature generation mode. The audio phase shift is accomplished using four op-amps in what is known as “active all-pass” configuration. The frequency range over which the 90-degree phase shift is accurately maintained need not be large since CW is received at 700Hz audio with a ~200Hz wide filter.

However, in an SSB receiver, the 90-degree phase shift needs to be accurate over a much wider range, for example 300-3000Hz. To do this with the “active all-pass” network requires a larger number of op-amps.

## SSB Exciter

In a CW transmitter, the oscillator signal only has to be keyed, amplified, filtered, and connected to an antenna. In an SSB transmitter we need a full SSB exciter; this circuit block converts audio to RF, and is supposed to simultaneously ensure that both the unwanted sideband and carrier are greatly attenuated. Following the QCX architecture, this circuit calls for a 90-degree phase shift, covering 300-3000Hz. Perhaps the demodulator phase-shift circuit could be dual-purposed but this adds the complication of switching. This SSB exciter is an entire circuit block which just doesn’t need to exist in a CW transceiver.

## Filtering

Since the QSX will cover both CW and SSB (and less commonly used modes on HF: AM and FM), at least two different filters are needed. QCX contains a compact four-op-amp CW filter centered on 700Hz with ~200Hz bandwidth. But for SSB we will need audio filtering to create the desired bandwidth, say 300-3000Hz (of course, making it variable would be very nice). Again, quite a complex analog circuit block.

## Automatic Gain Control (AGC)

QCX does not have an AGC circuit. CW operators differ in their opinion over whether AGC is essential or not, for CW operation. But you will have a much harder time convincing an SSB operator that a high performance SSB transceiver doesn’t need AGC! So, there’s another analog circuit block to add to the list.

## Speech Compression

Speech compression can add considerable effective power to an SSB transmission. It’s more than an AGC for transmitters, to adjust to the volume of your voice. A proper speech compressor adds punch by increasing the average power of your SSB transmissions. It’s well worth having, particularly so if you have a high-performance product in mind; but again, it’s another complicated circuit block.

Overall, evolution of the QCX CW transceiver into an SSB transceiver seemed like a difficult task. The resulting circuits would add significant complexity, parts count, board area and of course, COST.

The solution turned out to be Software Defined Radio (SDR).

## Software Defined Radio (SDR)

"Software-defined radio is a radio communication system where components that have been traditionally implemented in hardware are instead implemented by means of software on a personal computer or embedded system" (Wikipedia https://en.wikipedia.org/wiki/Software-defined_radio).


We may typically imagine an SDR as some kind of box, to which we connect the antenna to, and a bunch of cables connect it to a computer. On the computer we run an SDR software package to control this little box, all the normal functions associated with the front panel of a radio transceiver are now available with mouse clicks on the computer screen. Additionally, there's a fancy panadapter display of a wide chunk of band (typically 48, 96 or 192kHz).

I've never been attracted to the idea of having a radio which can only work when tethered to a PC. Luckily, that's not the only way to do SDR – the Wikipedia definition (above) says so too: "or embedded system"! An embedded system is where a microprocessor or microcontroller is embedded into a product and provides a specific function (as opposed to the general purpose nature of a PC). In an embedded SDR system, the radio itself contains a processor that is purposed to handle all the Digital Signal Processing (DSP) requirements.

This is actually not new, many modern transceivers are internally an SDR; the radio still has the conventional knobs and buttons etc., and does not require a PC to operate, but internally a powerful processor implements the SDR. There are good reasons why manufacturers are taking this route:

- It lowers the cost and hardware complexity
- Offers high performance
- Allows lots of nice features to be included
- Permits new functionality to be delivered later to owners, via firmware upgrades

It wasn't my intention originally to design an SDR-based transceiver to evolve the QCX transceiver into QSX, including SSB and CW together. But SDR became inevitable.

Modern microcontrollers aren't just a fast processor. They also include memory, a multitude of I/O ports and peripherals like timers, converters, and serial ports (conventional USART serial, I2C, and even USB). These permit a large amount of functionality to be implemented in a design, all at very low cost and component count.

It was an enlightening moment when I added up the cost of a high performance 24-bit Analog to Digital Converter (ADC), and a 24-bit Digital to Analog Converter

(DAC) chip, and the incremental cost of the powerful processor (compared to the ATmega328 processor used in the QCX). Then I compared that to the cost of all the op-amps and analog circuits that this digital processing would replace. The result was a clear WIN for the digital (SDR) option, just on cost alone. Then the mind boggled at all the additional feature possibilities that powerful processing and built-in Digital Signal Processing (DSP) could offer.

## Is it really SDR?

A short discussion is warranted about the type of SDR used here.

The SDR in the QSX processes the I & Q channels baseband (audio) produced by the Quadrature Sampling Detector. These two channels are converted to a digital representation by Analog to Digital Converters (ADC). The 90-degree phase shift is applied digitally by Digital Signal Processing (DSP), which allows the unwanted sideband to be cancelled (AM and FM can also be demodulated. Then the digital audio is converted back to analog by a Digital to Analog Converter (DAC). The transmitter operates a somewhat similar procedure in reverse.

Digital Signal Processing by the microcontroller in the embedded SDR performs the 90-degree audio phase shift and also other functions that would be handled conventionally in analog circuit blocks; these include filtering, AGC and speech processing.

But, there's another type of SDR which is viewed by some enthusiasts as the only "real" SDR. This is called Direct Digital Conversion (DDC) SDR. In these receivers, there is no analog hardware doing the conversion from RF to baseband (audio). A very high-performance ADC operates directly at the antenna, converting the entire RF spectrum to a digital representation. Then a fast computer applies DSP to tune and demodulate the desired narrow slice of this entire spectrum. This is then converted in a DAC back to analog audio and amplified for presentation to the operator in his headphones.

This more sophisticated SDR is used for example, in Icom's IC7300 transceiver. It is said to be "real SDR" because it converts the analog signals to digital, one stage closer to the antenna: there is no mixer converting the RF to baseband, this is done digitally.

However, in reality this method definitely does NOT eliminate all analog circuit blocks. It just eliminates the analog conversion from RF to baseband. You STILL, ideally, need a set of band pass filters ahead of the receiver. Or even if you are not chasing optimum performance, you definitely need a low pass filter ahead of the ADC which limits input signals to below the Nyqvist frequency. At the far end, you are still converting back to analog and amplifying to provide audio to the operator's headphones. When implementing a transmitter, you will still need Transmit/Receive switching, several stages of amplification, filtering, and a Linear Power Amplifier, all analog circuit blocks, not digits in a computer.

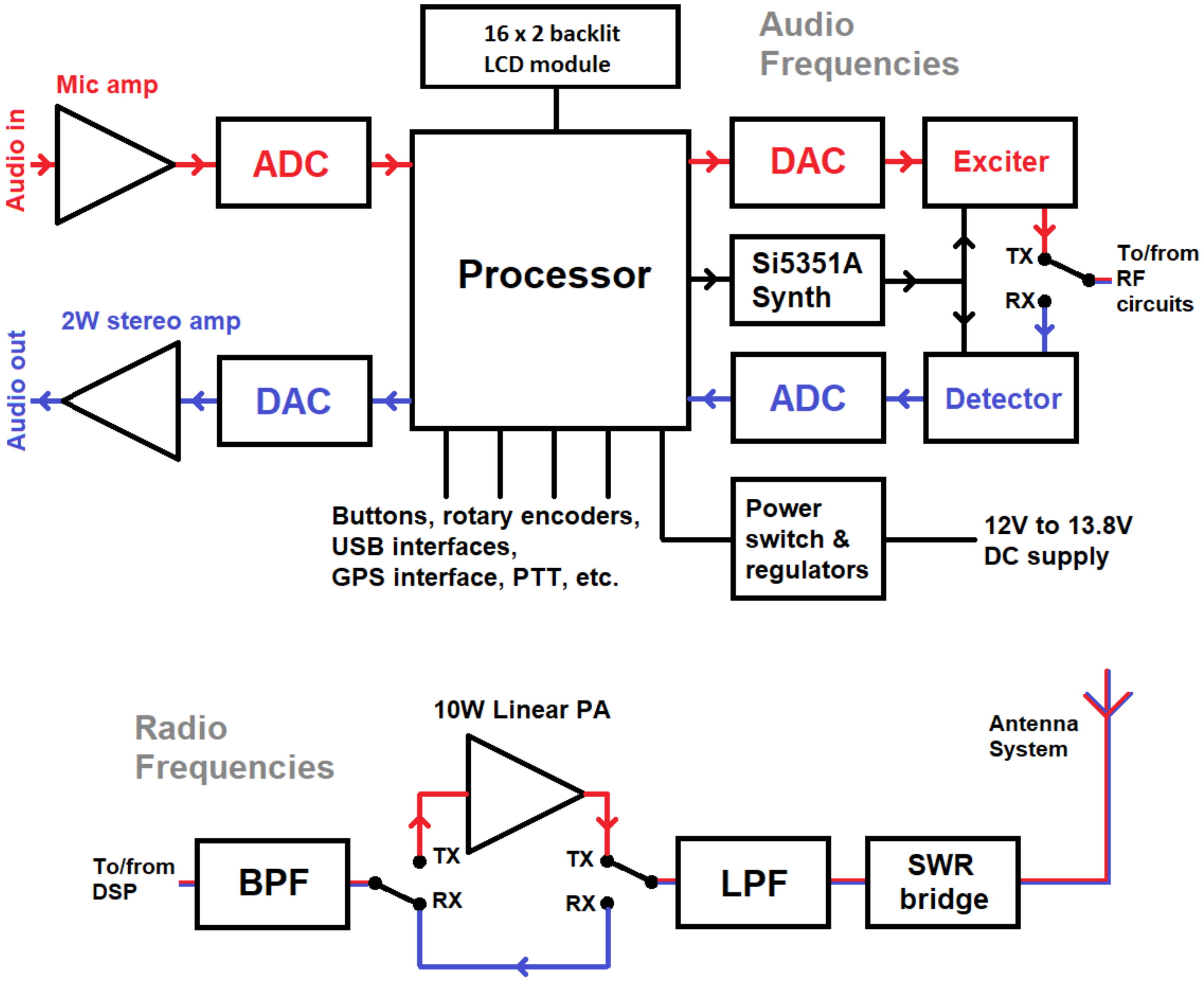
So, really, "what is SDR" is a matter of degree. The conventional conversion to baseband, THEN Digital Signal Processing that has been called SDR for the last two decades, is still real SDR. You are still replacing quite a number of analog hardware circuit blocks by digital processing in a computer (whether embedded or general purpose).

The significant disadvantages of the DDC approach to SDR are the requirement for very high performance, fast ADC chips (which are expensive and power-hungry), and very fast computer processing (which is also expensive and power-hungry).

In the QSX transceiver, since the aim is very high performance at a very low cost, with today's technology QSX is incompatible with the DDC approach.

# QSX architecture

Here's a block diagram of the QSX architecture.




The upper part of the diagram shows the Digital Signal Processing that implements the SDR. The processor used is from the STM32F4 family by ST Microelectronics; it is a powerful 32-bit ARM processor running at 168MHz clock speed.

On the receive signal path, the Quadrature Sampling Detector (QSD) is similar to that in the QCX. It is clocked by the Si5351A Synthesizer chip in quadrature mode, just like in the QCX. The output I & Q channels of the QSD go straight into a high performance 24-bit ADC chip running at 48,000 samples per second (abbreviated to 48ksps). These digital samples are handled in the magic of the Digital Signal Processing (DSP), which creates a corresponding stream of output samples, now representing Single Sideband (SSB) and also at 48ksps, that are fed out to a high performance 24-bit DAC chip to convert back to audio. A 2W stereo audio amplifier chip provides plenty of audio to the earphones.

On transmit, a similar thing happens but in reverse. Signals from the microphone are amplified and fed into an ADC. The DSP magically produces I & Q channels, whose 48ksps samples are converted to analog by a DAC, then feed into a Quadrature Sampling Exciter to produce SSB RF at the desired operating frequency.

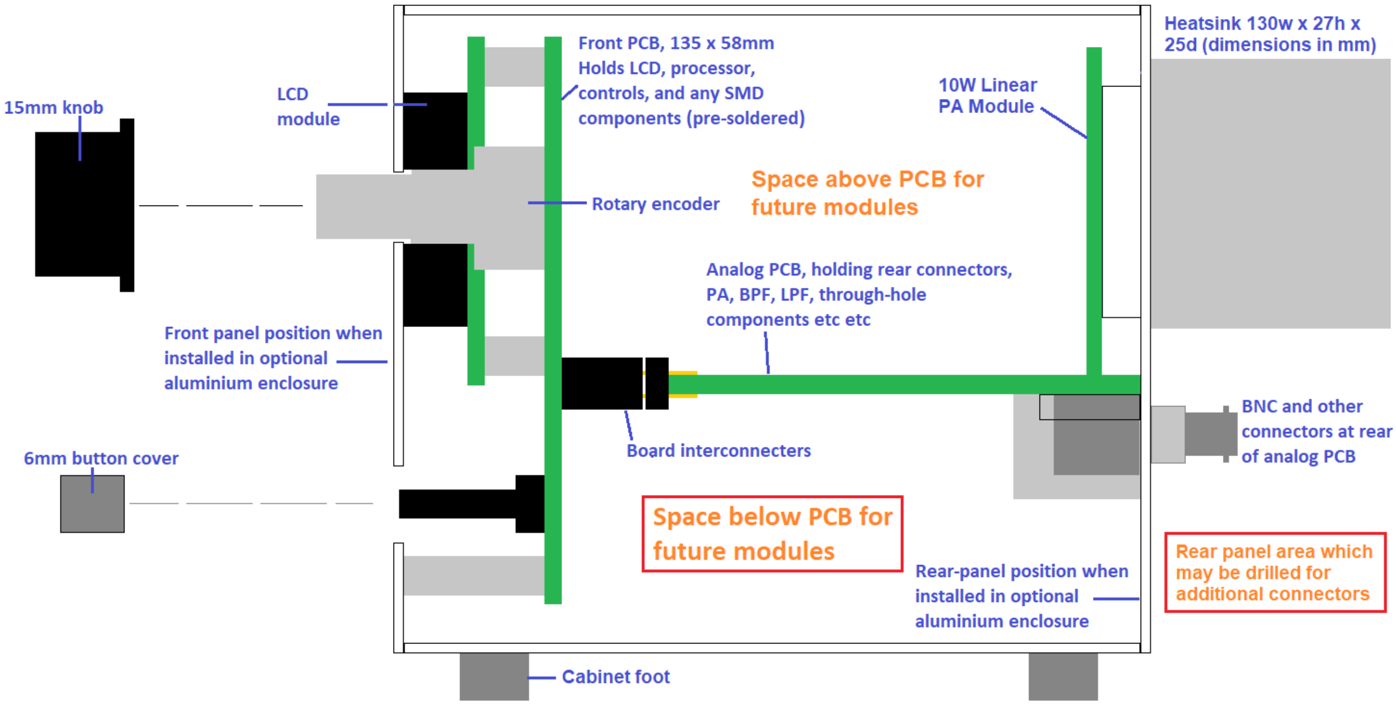
The lower part of the diagram shows the RF part of the transceiver. The Band Pass Filter (BPF) and Low Pass Filter (BFP) are used bidirectionally on both Transmit and Receive. There is an SWR bridge circuit that allows the microcontroller to measure Forward and Reflected power and calculate SWR. The transmit 10W Linear Power Amplifier (PA) block deserves a lot of attention. So does the transmit/receive switching. More on these, later.

## Mechanical design


The QCX CW transceiver design ended up a little inconvenient to fit into an enclosure. Many different solutions were found by creative constructors. An enclosure solution in which all components and controls remain on the PCB as designed, was difficult because the heights of the LCD and controls were all different so wouldn't easily fit into a front panel. Furthermore, the connectors being on the two sides wasn't very convenient either.

Markus DL6YYM ([https://www.bamatech.net/](https://www.bamatech.net/)) did design a two-piece aluminium enclosure (pictured right) that fit the QCX board perfectly and has been very popular; the enclosure kit includes shaft extensions for the rotary encoder and gain potentiometer, and also plastic button extensions to reach the small buttons on the board.

With the QSX design, I wanted to consider the enclosure right from the start. I wanted to avoid these subsequent enclosure headaches completely. Therefore, from the start, the mechanical design of QSX was chosen to permit easy mounting in an optional cut/drilled/silk-screen printed aluminium enclosure.




This enclosure is 145w x 68h x 73d mm and is formed of identical extruded aluminium top and bottom half pieces which have slots for holding PCBs, and front/rear aluminium panels. The QCX design is composed of a number of PCBs. One is a faux front-panel PCB that fits 12mm behind the real aluminium front panel. At this distance, the LCD fits snugly behind a rectangular cut-out and all of the other front panel controls protrude through holes in the front panel, all at just the right height.

A horizontally-mounted PCB fits into the slots on the lower extruded aluminium half piece. All connectors are board-mounted along the rear edge of this PCB and fit through holes in the rear panel. A large heatsink is also bolted to the rear panel.

This assembly permits ALL controls and connectors to be board-mounted with NO wiring at all. There are also no holes in the top and bottom extruded sections, only in the front and rear panels. Manufacture and shipment of the enclosure is therefore optimally economic. It took a lot of head-scratching to come up with this arrangement.

Learning the lessons from the former QCX which didn't easily fit an enclosure, I think it is well-worth planning ahead to ensure a design is mechanically suitable for a planned enclosure. This often applies equally to ANY project: your own homebrew designs too! It pays to think in advance, how you are going to put it in an enclosure later.

# Some design decisions

In this section I want to talk about some of the design decisions implemented in the QSX transceiver. At all times bear in mind the three objectives:

- High Performance
- Many features
- Low cost

Remember my objective was to combine three in a transceiver with a performance & features to price ratio never seen before. These objectives drive every aspect of the design. Everything must be optimized to meet these criteria.

## User interface

The user interface has been kept deliberately simple. Whilst a radio covered in knobs and buttons looks very nice, every one of those knobs and buttons adds to the cost and the size (which also means cost due to larger PCB area and enclosure). They do nothing for the actual raw performance of the radio. On the other hand, we need to have enough controls to make it reasonably usable.

Accordingly, the user interface follows on from the QCX CW transceiver, using the same popular 16 x 2 alphanumeric LCD display. It uses two rotary encoders and four push-buttons, one of which performs the power on/off function. Each rotary encoder also has a push-button on its shaft, making a total of six buttons. Since each button can be used for multiple purposes depending on whether pressed once, twice, or with a long press (as on QCX), in fact there are a large number of transceiver functions that can be easily activated with various simple button presses.

It also turned out to be possible to obtain push-buttons that are soldered to the faux front-panel PCB, and have long enough shafts to protrude through front panel holes, thereby satisfying mechanical requirements to fit the enclosure.

## What, no volume potentiometer?

That's right! No volume potentiometer! Originally, I had planned to have a volume potentiometer. But a number of factors conspired against this intention. The first was the aforementioned mechanical considerations. I could not easily find a low-cost potentiometer which has a shaft-length compatible with the other controls and a slickly fitting front panel.

Another consideration was the taper of the potentiometer. Ideally an audio gain pot needs a logarithmic taper because this is the way the ear perceives sound levels. But logarithmic track potentiometers are much more expensive than the usual linear track ones.

Finally, I was also keen on a stereo audio output. In most common use cases the Left and Right audio channels would be identical, after all, Single Sideband isn't stereo! But having separate control over Left and Right channels permits several interesting functions to be included. Binaural reception is one. A secondary receiver, is another – so that you can have Left and Right ears tuned to different frequencies and monitor both at once. Or, there may be people with a hearing imbalance who would like to be able to adjust the relative volume in each ear to compensate a hearing defect.

There were also design complications. The STM32F4 microcontroller contains two 12-bit DACs, which could be used for the output Digital to Analog Conversion. 12-bits is enough resolution. However, they are also needed on transmit, to provide the I & Q signal outputs. A switching circuit to route the outputs to the transmit exciter, or to the receive output audio amplifier, would add complexity – but the final nail in the coffin is that it wouldn't be possible to produce a sidetone during CW, as well as the necessary I & Q transmit outputs, from only two DACs.

Taking all this into consideration, the ideal solution was to output the audio with 24-bit resolution to a dedicated 24-bit stereo DAC chip, and control the volume level digitally in the DSP. This also allows a logarithmic control function. At low cost, this solution solved all the issues: logarithmic control, stereo output, Left/Right balance adjustment, and the mechanical matching of the controls to fit the front panel. Use of a 24-bit DAC chip permits a wide volume adjustment range, while still retaining enough effective bit-depth of resolution to provide sufficient audio dynamic range of the represented signal.

## Microphones

Microphones are a difficult topic! There are many different styles of microphone and connectors. Low cost, high cost. In the end, I opted for a solution which provides two different microphone options, each with their own independent buffer amplifier and gain adjustment, feeding separate ADC inputs of the microcontroller.

The first is a 4-way 3.5mm jack socket on the front panel, which can be used with ordinary cellphone-style earphones having a microphone built into the cable. I bet everyone has several of these lying around, the average lifetime of cellular phones seems to be around 2 years. Upon casually opening one drawer here I easily uncovered five or six different units.

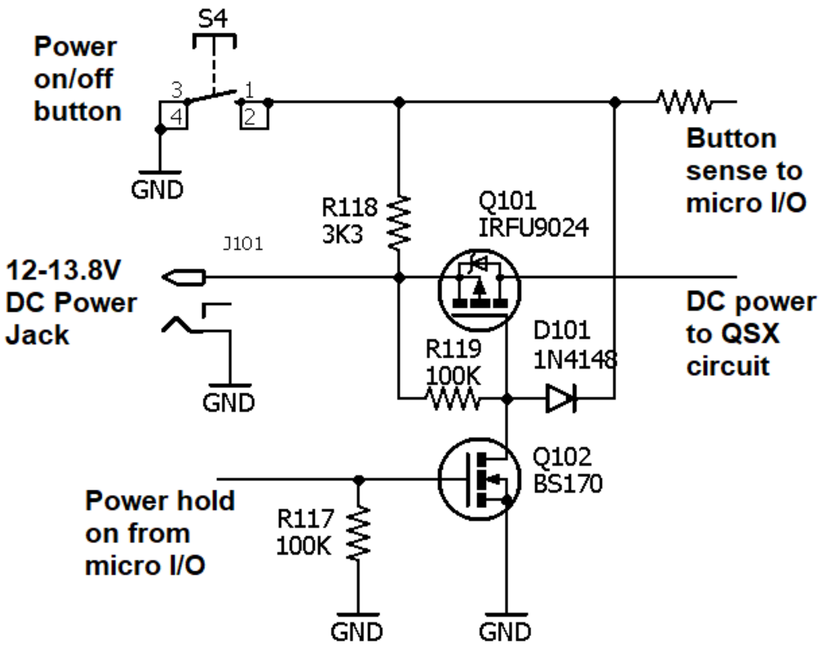
Secondly there is an RJ45 connector on the rear panel. There appear to be multiple "standard" pinouts for these, for various Kenwood, Yaesu etc microphones. They all seem to have in common, the same pin numbers for ground and audio. Different functions provided include Up/Down buttons, Push-To-Talk (PTT) button and even numeric keypads with serial data. In QSX the non-audio and ground pins are connected to processor I/O pins so can be configured to suit particular microphone styles.

Provision of these two microphone possibilities seemed that it would cover a large range of existing microphones.

## Soft power-switch and reverse polarity protection

The QCX 5W CW transceiver had no on/off power switch but in QSX I felt that I'd like a power switch. QSX takes performance and features to the next level and a power switch may be included in people's expectations.

Still on the theme of mechanical design to match the planned optional enclosure, it proved impossible to source any toggle switch or other kind of electrical switch that would be simultaneously possible to solder to the PCB, as well as be in physically the right place and height to appear on the front panel of the optional enclosure. I therefore designed a "soft switch" to replace the toggle switch.




This on/off power switch is the right-hand button of the four push-buttons on the front panel. When pressed, it grounds the gate of a P-channel MOSFET, allowing it to pass current from the DC input power connector, to the rest of the QSX circuit. The microcontroller boots up and the first thing it does, is take over from the right-hand button, switching on Q102 to hold the P-channel MOSFET gate low, to keep the current flowing when the power on/off button is released.

Once on, the microcontroller is control of the power switch. Power remains on as long as voltage is still supplied to the DC power jack, until the microcontroller decides to turn it off. To decide that, it can monitor the power on/off button. When the operator presses it again, it can then switch off the power to Q102, allowing the Q101 transistor to switch off.

In this way, external power to the QSX can be left switched on and consumes no current; and is controlled by the push on, push off button on the front panel.

An additional and important advantage is that on the power-down button being pressed, the microcontroller can save all the current settings of the radio such as the operating frequency, volume etc., into a non-volatile storage memory (EEPROM), and THEN power down. This means that on next switching on the QSX, it can restore its state exactly as it was when switched off. Just like a "real" old analog radio with all the knobs! When you come back tomorrow and switch it on, it's exactly as you left it!

This is a really useful circuit, and also has the benefit of providing reverse polarity protection to the radio.

# About DACs and ADCs

An SDR requires four different conversions between analog and digital domains. Two Analog to Digital Converters (ADC), and two Digital to Analog Converters (DAC). Some of these conversions must operate simultaneously on two channels, such as the I & Q signals coming from the Quadrature Sampling Detector (QSD). The requirements:

- Receiver ADC, two channels I & Q from the Quadrature Sampling Detector
- Receiver DAC, two channels Left and Right to the headphones
- Transmitter ADC, mono from the microphone
- Transmitter DAC, two channels I & Q to the Quadrature Sampling Exciter

Since there are two microphone options, we can simplify and say that the transmitter ADC should also have two channels, one for each microphone; this will avoid any need for hardware switching between the two, we can just implement the switch digitally inside the microcontroller.

Now the very important question is, how much performance is needed? Performance costs money. Good engineering is all about choices. Trade-offs. We don't want to purchase any more performance than is necessary to achieve the overall design goals. But we can't purchase too little performance either or we won't meet the design goals.

## Dynamic range

ADCs and DACs have dynamic range. In radio communications systems, dynamic range is a very important parameter. It turns out that the theoretical maximum dynamic range available from an ADC is 6dB per bit of resolution. The best theoretical dynamic range that can be obtained for various popular converter resolutions is:

| | | |
|---|---|---|
| 8-bit: | 48dB | |
| 10-bit: | 60dB | (the ADC in the ATmega AVR microcontroller is 10-bit) |
| 12-bit: | 72dB | (the ADCs and DACs in the STM32F4 microcontroller are 12-bit) |
| 16-bit: | 96dB | |
| 24-bit: | 144dB | |

Unfortunately, there's a popular misconception that this is all there is to it. It isn't! In ANY converter, the bottom few bits are quite simply, noise. The maximum theoretical performance is never obtained. Most ADC and DAC chips never even come close to the maximum theoretical performance. If your SDR uses a 24-bit ADC, it sounds good to boast about, but by itself, this parameter is largely meaningless. This also applies just the same, to PC USB soundcards.

ADC and DAC chips designed for audio purposes quote the dynamic range in their performance specification. This is the critical parameter, worth more than the number of bits!

## Nyqvist Frequency

The Nyqvist frequency is the highest frequency which may be accurately sampled for a given sample rate. It is half of the sample rate. Commonly used sample rates in audio ADCs are 44.1kHz (for CD audio), 48kHz, 96kHz and 192kHz.

## Over-sampling

Over-sampling is a technique where a signal is sampled much more often than required by Nyqvist, so that each desired sample is made up of multiple actual samples. A process called "decimation" divides down the number of samples, and by a process somewhat like averaging, the noise is also reduced. The effect is to increase the dynamic range of an ADC. However, it requires decimation by a factor of 4 to obtain an effective increase in bit-resolution of the ADC of only 1 bit. Practically speaking in this application, over-sampling does not gain us much.

## What performance ADC?

Finally, after understanding the basics, the question is, what performance converters are needed? By far the most critical conversion is the Receiver ADC, which converts I & Q channels from the Detector into digital representation. The performance of this device determines the overall performance of the entire receiver. If an ADC input is overloaded, the output digital representation cannot reflect the input. The resulting demodulated audio becomes severely distorted. The gain of the pre-amps in front of the ADC must bring the audio level to a suitable level that weak signals are detectable; at the same time, very strong signals must not overload the ADC. Since the ADC sits in the signal path before the selectivity is applied (which is in the DSP), it must deal with a relatively large chunk of band which could contain very strong and very weak signals.

This receiver ADC is therefore one of the most important and critical devices in the entire design. Modern HF transceivers should generally have a dynamic range of around 100dB. The ADC must therefore offer at least this level of dynamic range. For the QSX, a 24-bit stereo ADC chip was chosen with a specified dynamic range of 110dB. As a point of comparison, the chip used in the very highly regarded Elecraft KX2 and KX3 transceivers (which are also embedded SDRs with a similar architecture) has a dynamic range of 103dB.

When listening to audio, even an expert musical human ear cannot perceive any difference in quality when more than 12-bits of DAC resolution are used. This applies to Hi Fi audio. The level of quality of SSB audio on HF bands is far below Hi Fi so the requirements on the DAC are even lower. Hence the 12-bit ADC and DAC peripherals contained in the STM32F4 microcontroller are plenty adequate for handling all of the other conversions.

A complication arose with the receiver output DAC which drives the headphones. During transmit, the STM32F4 microcontroller's own two 12-bit DACs are busy generating I & Q signals for the transmit modulator (Quadrature Sampling Exciter). When operating CW, we also need to generate another audio signal for the sidetone. I could have used an I/O pin or Pulse Width Modulation (PWM) output to achieve this. But then I would have needed some switching circuits to switch the output audio amplifier between the DAC output and sidetone. As discussed previously, I also found it an advantage to use a rotary encoder and digital volume control within the DSP, rather than a potentiometer as analog volume control. The rotary encoder fits the mechanical requirements very well, and allows the DSP to emulate a logarithmic gain control, in stereo (separate Left and Right channels) and with an amplitude balance adjustment between Left and Right.

For the receiver audio output, I therefore chose to use a 24-bit stereo DAC chip with 96dB dynamic range. This has more than enough dynamic range to allow a wide range of volume control without sacrificing the dynamic range of the received audio. For example, if the volume can be attenuated 24dB and still leave 72dB of dynamic range for the signal; 72dB is the equivalent of the 12 bits resolution, beyond which the musical expert's human ear cannot distinguish any difference in Hi Fi quality. Bear in mind that we are only talking about audio volume here; Intermediate Frequency (IF) gain is provided within the DSP.

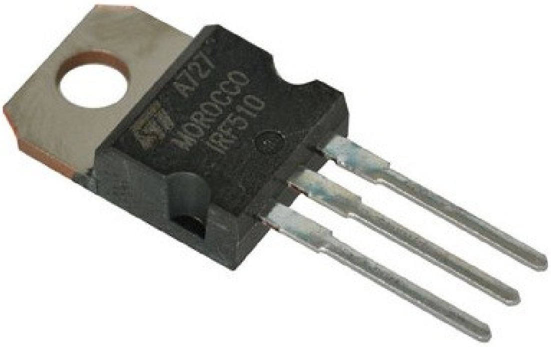
The two 12-bit ADCs inside the STM32F4 microcontroller are used for the two microphone inputs on transmit, and the 12-bit DACs used to drive the I & Q channels that feed the transmit modulator (QSE). 12-bits is plenty for SSB! Using the internal 12-bit ADC and DAC peripherals is a great way to reduce parts count and save costs.

# The 10W Linear Power Amplifier

The power amplifier is one of the most interesting and challenging parts of an SSB transceiver design. This is particularly the case when you want the power amplifier to cover the whole of HF from 1.8MHz to 30MHz. Ideally, we are looking for:

- 10W output, that doesn't change from 1.8MHz to 30MHz
- High Linearity
- Big enough heatsink to not overheat on high duty-cycle digital modes like FT8
- Easy to set-up
- Unconditionally stable
- Robust in the face of high SWR etc
- Low cost!

A popular range of RF MOSFETs made by Mitsubishi include part numbers like RD06HVF1, RD06HHF1, RD15HVF1. These are used in many commercial transceivers. They work well BUT, their big problem is that they are expensive!

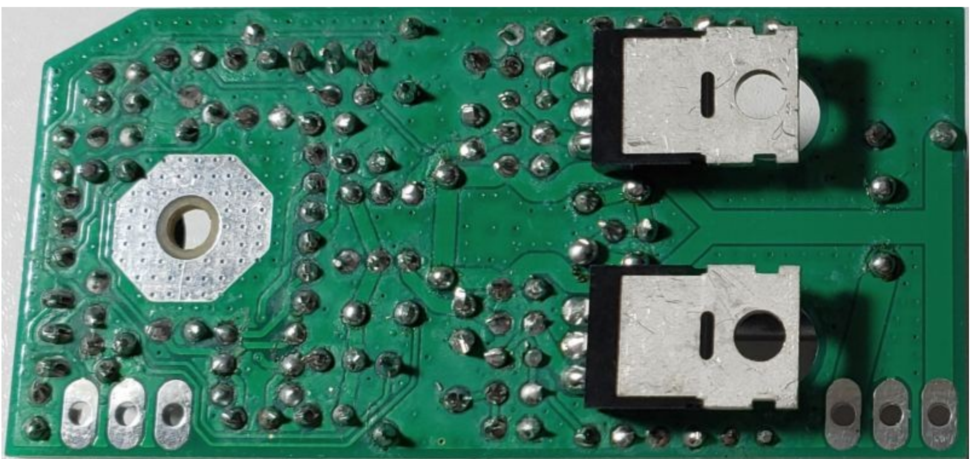



The IRF510 MOSFET is a low-cost device intended for switching applications. It has been used in power amplifiers but has gained something of a poor reputation, for poor performance: low gain at the upper end of the HF spectrum leading to low power output, and also poor IMD performance. A big disadvantage of the IRF510 is higher input capacitance than the RD15HVF1, which means that a lower impedance driver is required. Another problem of the IRF510 is that the metal tab is drain. The RD-series transistors have the metal tab as source, which is usually grounded, so that makes it very convenient to use with a grounded heatsink.

My challenge was to see if the IRF510 can be made to work properly in this application. If it could, this represents a significant cost-saving over the RD-series of "proper" RF transistors.

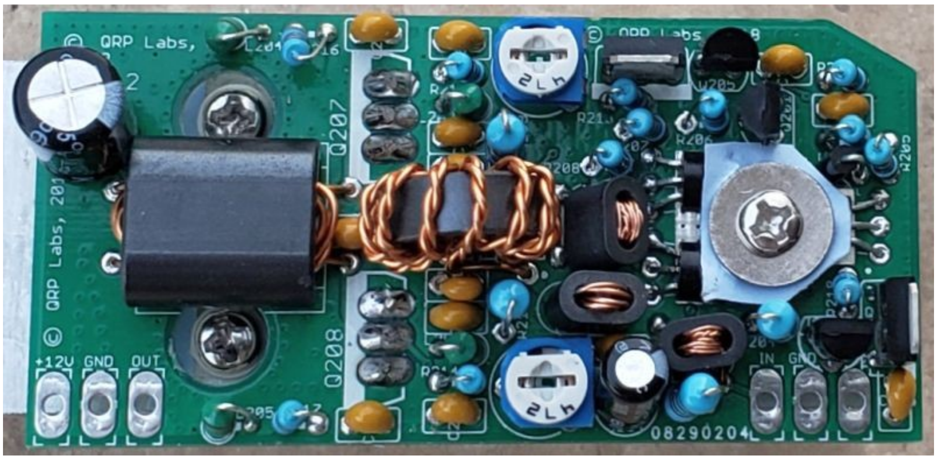
After a great deal of experiment, it turns out that it IS possible to design a fine HF PA using a pair of IRF510s in push-pull for Class A/B operation, resulting in excellent performance at low cost! The driver circuit is critical and must be able to supply enough power at low enough impedance to overcome the relatively large gate capacitance of the IRF510s. I used a pair of BS170 MOSFETs in push-pull as the driver stage, this part of the circuit is taken from the famous SoftRock transceiver board.

The other tricks necessary to make this happen, and I do mean absolutely necessary, are a very well laid out PCB, with plenty of ground-plane on both sides, strapped together by vias at frequent intervals. Critical leads must be kept short. This is particularly true for the IRF510. Wire has inductance. Each untrimmed lead of an IRF510 has an inductance of something around 10nH, which is quite significant at 30MHz. My initial lash-ups of the PA with wires over unetched PCB for development purposes, worked fine at 7MHz. But anything above that, it went off into its own horrific self-destructive oscillation.



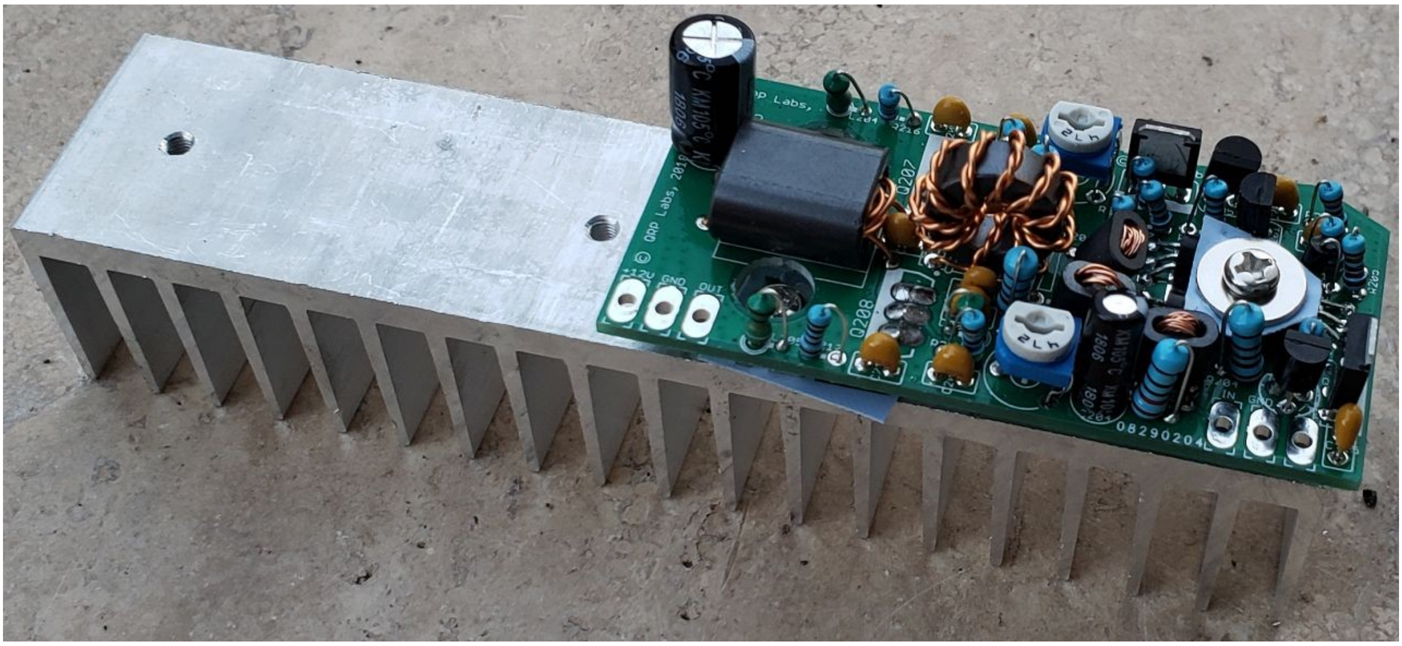
The IRF510 transistor leads are bent 90-degrees directly at the transistor body, and soldered to the PCB with only about 1mm of leads between them. The circuit is also electrically and mechanically kept as symmetric as possible.

Insulating pads must be used between the IRF510 and the heatsink, since the IRF510 metal tabs are internally connected to the drain, not the source (which is grounded).

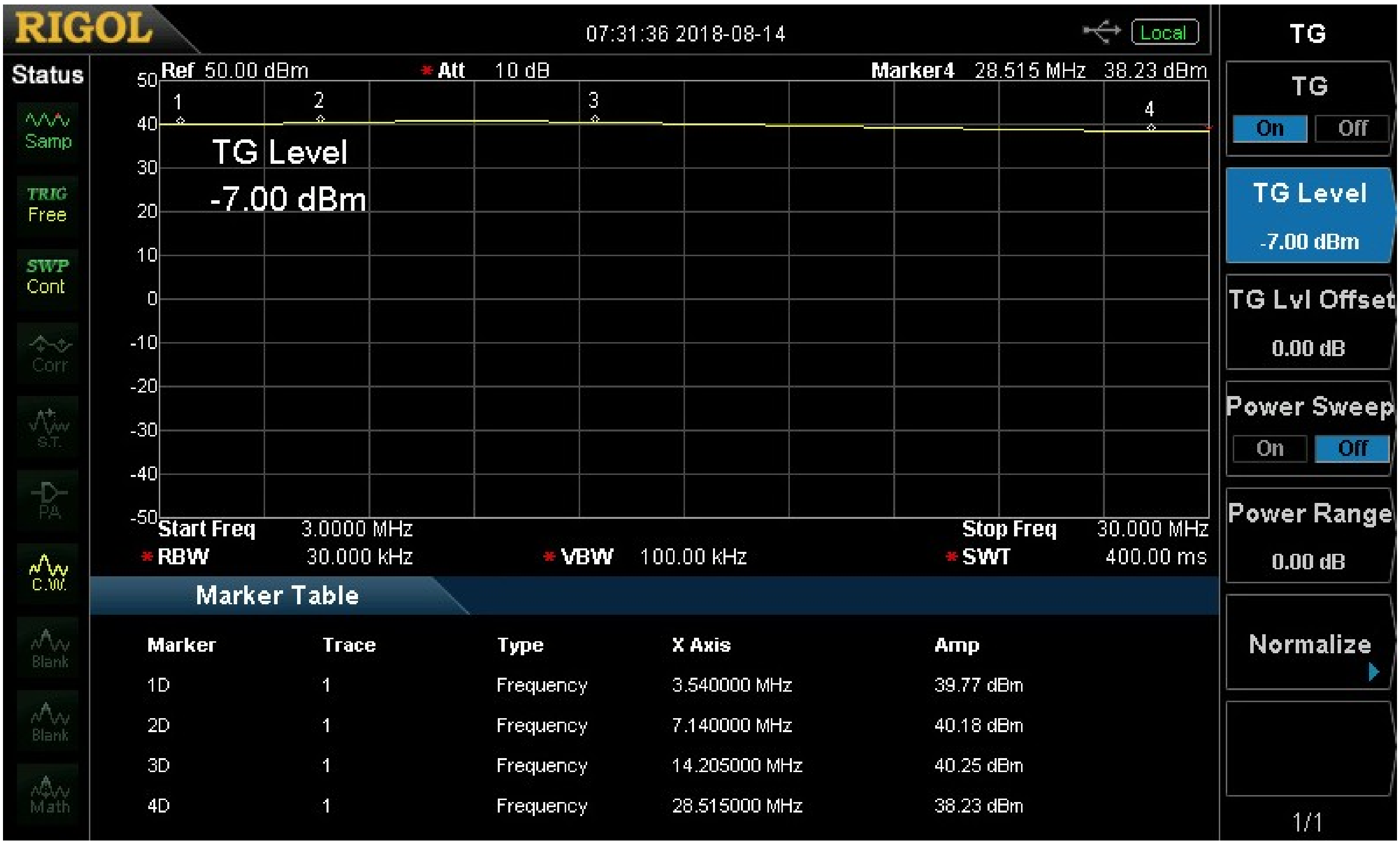



Don’t be tempted to use two separate half-size heatsinks and no insulation hardware! Those heatsinks are RF-hot and DC-hot, if they touch anything else you will get short-circuits and fry something; even if they do not touch anything else, they are adding capacitance to the drain circuit and they’ll act as two nice little antennas, wanting to radiate energy back to the amplifier input.

A substantial heatsink was used, measuring 130 x 28mm and with fins 25mm deep. This is large enough to not overheat even on continuous key-down for 1 hour at 10W full power output. It also designed to fit very nicely bolted to the back of the optional QSX enclosure



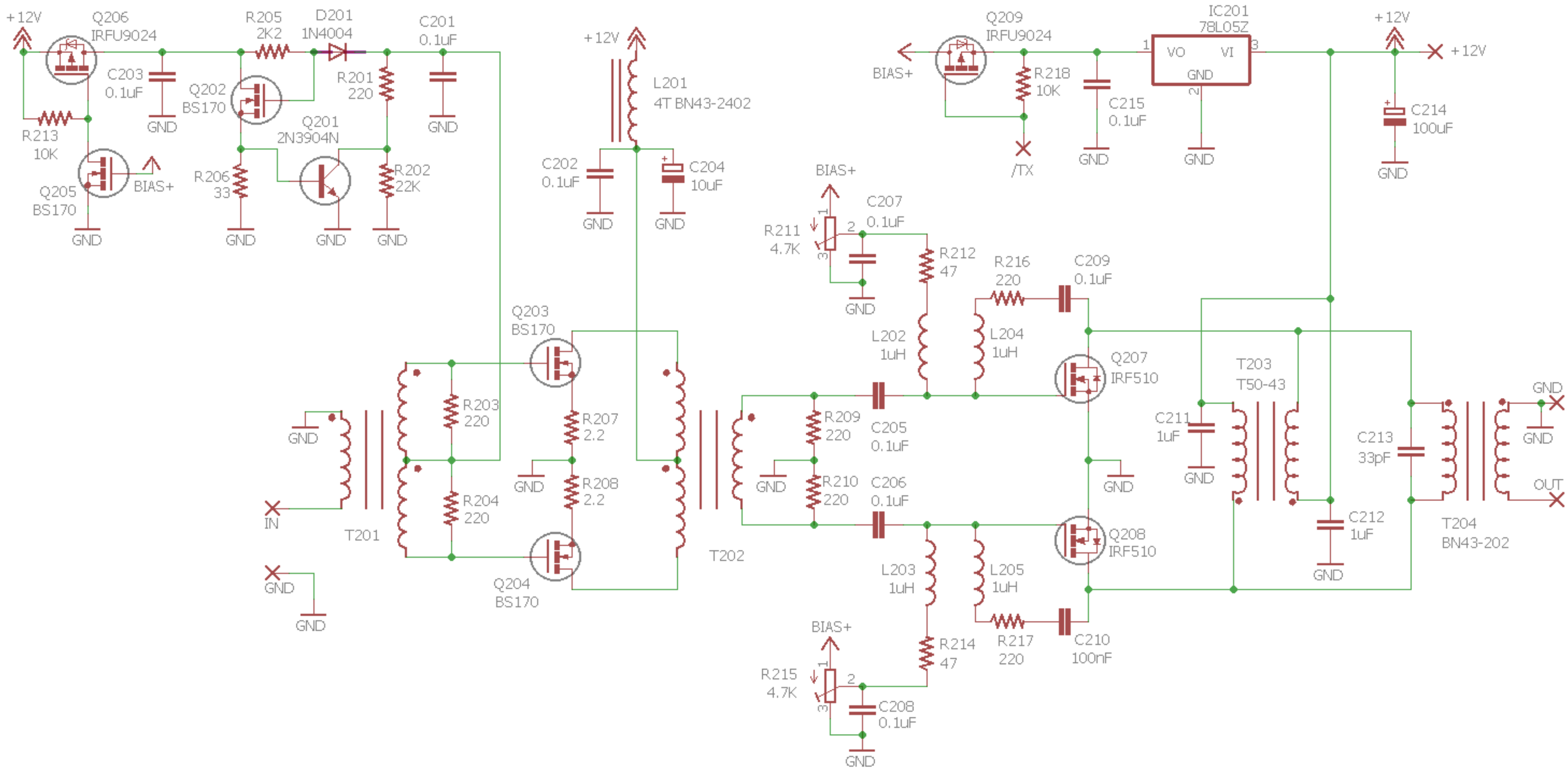

The performance of this low-cost Linear HF amplifier is really remarkable. Gain is 26dB +/- 1dB from 1.8MHz to 30MHz.




Testing showed very good linearity; on 80m the 2$^{nd}$ harmonic was a -38dBc which is a good indication of linearity; and the measured 2-tone IMD3 was -30dB relative to 10W PEP.

The amplifier also behaved very well in stress testing, operated at full-power into open load, shorted output, and various mismatches – all without any oscillation or instability, or damage or overheating. I do encourage you, to NOT give up on the humble IRF510 MOSFET!

More details are at [http://qrp-labs.com/linear](http://qrp-labs.com/linear)



+12V
Q206
IRFU9024
R205
2K2
D201
1N4004
C201
0.1uF
C203
0.1uF
Q202
BS170
R201
220
R213
10K
Q201
2N3904N
R206
33
R202
22K
Q205
BS170
BIAS+
GND
+12V
L201
4T BN43-2402
C202
0.1uF
C204
10uF
Q209
IRFU9024
IC201
78L05Z
VO
VI
GND
R218
10K
C215
0.1uF
/TX
+12V
C214
100uF
R211
4.7K
C207
0.1uF
R212
47
R216
220
C209
0.1uF
L202
1uH
L204
1uH
Q207
IRF510
T203
T50-43
Q203
BS170
R203
220
R207
2.2
R208
2.2
R204
220
IN
T201
Q204
BS170
T202
R209
220
C205
0.1uF
R210
220
C206
0.1uF
C211
1uF
C213
33pF
C212
1uF
Q208
IRF510
L203
1uH
L205
1uH
R214
47
R217
220
C210
100nF
R215
4.7K
C208
0.1uF
T204
BN43-202
OUT

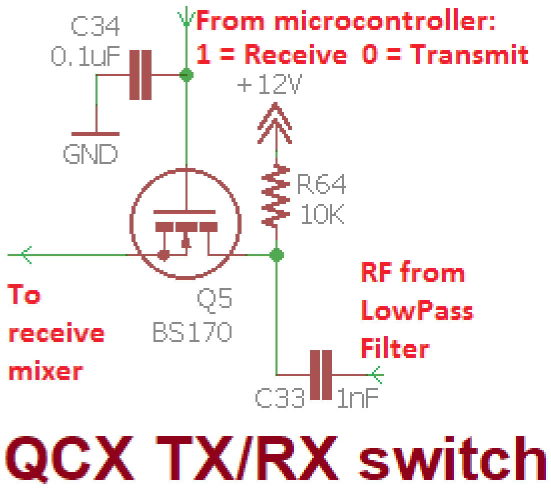
## Transmit/Receive switching

Transmit/Receive switching is another area where I spent a lot of time at the workbench!

My starting point here is an absolute determination to avoid using a relay for the Transmit/Receive switch. There are many reasons to NOT use a relay:

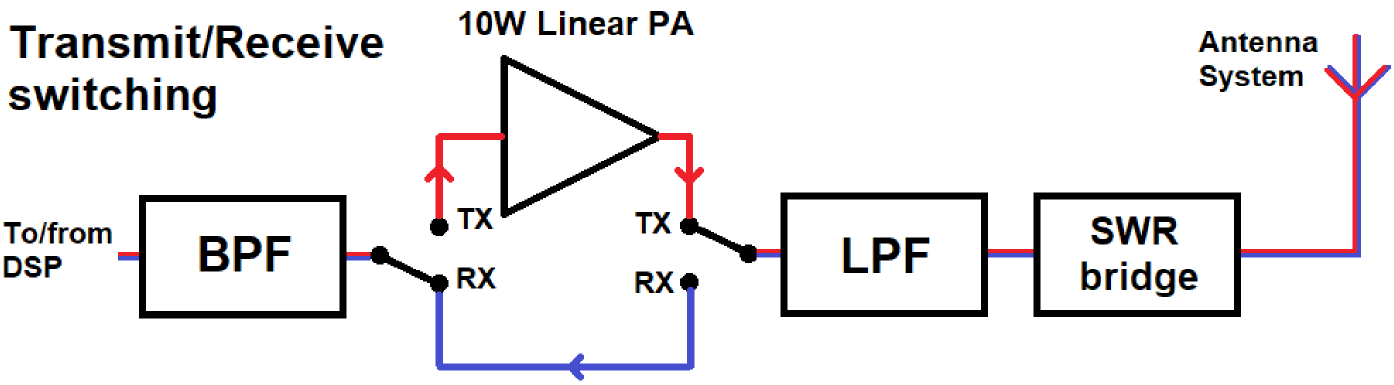
- Reliability: Relay datasheets quote a lifetime of as little as 30,000 operations. It's a mechanical component and sooner or later, there's a good chance it will fail.
- Isolation: Low-cost relays don't have wonderful RF characteristics anyway, in particular the isolation in the "Off" state. To get better performance requires higher cost relays. High cost is the enemy.
- CW full break-in: Here's a very important one. Full break-in operation on CW (a.k.a. QSK) is a feature which some modern brand-name rigs don't do very well, but many CW operators love. The QCX CW transceiver has nice full break-in operation and after 2 years I've got very used to it. If the relay had to switch back and forth on every CW symbol, the radio would clickety-clack like an old teletype machine, and those relays would wear out fast. Not to mention that the slow switching speed would wreak havoc with break-in at fast operating speeds!

Particularly for the sake of providing high-speed full break-in CW, relays are out of the question. I did not want to sacrifice CW performance, just because the QSX radio can do other modes SSB, AM, FM etc.




My starting point was the Transmit/Receive switch of the QCX CW transceiver, which is a simple BS170 MOSFET switch (see right). This turned out to be woefully inadequate for QSX, though perfectly satisfactory for the QCX. In the QCX transceiver, the only function of the Tx/Rx switch is to stop high voltages reaching the receiver during transmitter operation.

But recall the RF section of the QSX block diagram:




The QSX transmit/receive switch needs to have the function of a DPDT switch (Double pole double throw). During transmit, the isolation along the Receive signal path (the lower line) must be very high. If the isolation is less than the gain of the transmit signal path stages (pre-driver, driver and PA), then a positive feedback situation arises. An amplifier plus positive feedback equals an oscillator. (Which of course, is what all PAs aspire to be anyway., and the more self-destructive, the better).

The simple MOSFET switch in the QCX might just about be marginally good enough on the 40m band. But isolation performance deteriorates as the frequency increases. This is natural enough since the MOSFET has a certain drain-source capacitance, even in the OFF state, which leaks RF, more and more as the frequency goes up. We need isolation to at least exceed the amplifier chain gain, and preferably by a considerable margin so that we don't need to worry about what unpleasant effects a low level, phase-shifted feedback might cause.

So began a great deal of research, and here I shall present the brief summary of what I learned and the final design I arrived at, which could be useful in other radios too.

I spent a lot of time examining the schematics of other famous brand radio transceivers, and how they implement their solid state transmit/receive switching. PIN diodes are usually used. To my surprise, I found design faults in EVERY schematic I researched. Clearly these faults did not, in most cases, prevent good performance – but they were sub-optimal, in some cases, significantly so.

A great inspiration for Transmit/Receive switching is the Transmit/Receive switch of Don W6JL https://www.qrz.com/db/w6jl which 137dB of isolation. He runs a 600W CW Transmitter, and separate Receiver, always switched on, even during transmit; the isolation is so great that no receiver muting is required, and the sidetone is implemented by receiving the 600W transmission through the 137dB of isolation! There's a lot of components in that circuit and shielding too – overkill for QSX, but very educational to study! Don wrote an in-depth article which you can read here: https://www.funkamateur.de/tl_files/downloads/hefte/2017/w6jl_improved_qsk_system_mar_2016.pdf.

An ordinary diode has a P-type doped semiconductor region and an N-type doped semiconductor region. A PIN diode has a thick undoped intrinsic semiconductor region sandwiched between the P and the N layers. Briefly, this makes the diode relatively slow to react. When an AC signal passes through the diode at a high enough frequency, and a small DC current biases the diode "On", the current swings of the AC (RF) current are much too fast to be able to switch off the diode. So, the RF current is ignored and the diode remains on, effectively operating as a very small value resistance. In the zero or reverse biased "Off" state, the diode acts as a high resistance, though it still has a small capacitance which effectively limits the isolation at RF, getting worse as the frequency increases.

"Proper" PIN diodes are expensive. However, I found people like Don W6JL are using the cheap 1N4007 rectifier diode as a "poor man's PIN diode". The internal construction of a 1N4007 is quite similar to a PIN diode, with a relatively wide undoped intrinsic layer, which helps it to withstand high reverse voltages. The 1N4007 sits at the top of the popular 1N400x range of diodes with a reverse voltage rating of 1,000V. The important point is that the same thing that makes the 1N4007 suitable in rectifier service for high voltages, also makes it behave as a PIN diode!

I spent a lot of time testing and confirmed that they do indeed, behave like just like a "proper" PIN diode. Not quite as good, but certainly a fraction of the cost! Follow the theme here: cheap IRF510 MOSFETs in the 10W Linear PA; cheap 1N4007 diodes as PIN diodes in the Transmit/Receive switch. Low cost, a primary objective!

Even with a “proper” PIN it is unlikely that more than 40dB of isolation can be achieved, so in typically applications multiple PIN diodes are used in series or shunt configurations to cascade the attenuation.

The rules governing PIN diodes are:

- The “ON” state requires a small forward bias current, typically 10mA is enough
- The “OFF” state requires a large reverse bias voltage, which should be larger than the peak voltage of the applied RF waveform

The latter condition does not seem to be universally agreed upon and is an open question requiring more research on my part. In all my experiments and testing, I certainly found that the reverse bias voltage must be higher than the peak RF, otherwise the isolation provided by the diode is very low. This agrees with Don W6JL’s documentation.

I tried numerous combinations of “PIN” diodes in series and shunt configurations, along with MOSFETs. I eventually dismissed MOSFETs because the off isolation was poor and the internal substrate diode tends to interfere, which I found impossible to avoid. I also chose to use a quad SPST CMOS switch type FST3125 which has very good isolation characteristics. My final configuration uses a combination of “PIN” diodes to switch the PA output to the LPF, and to protect the CMOS switches which have a signal handling voltage range of only 0-5V. MOSFETs are only used for switching bias currents.

The schematic shows my final configuration. The transmit amplifier stages (pre-driver, driver, and PA) are not shown in full, just symbolically by the triangle shape.

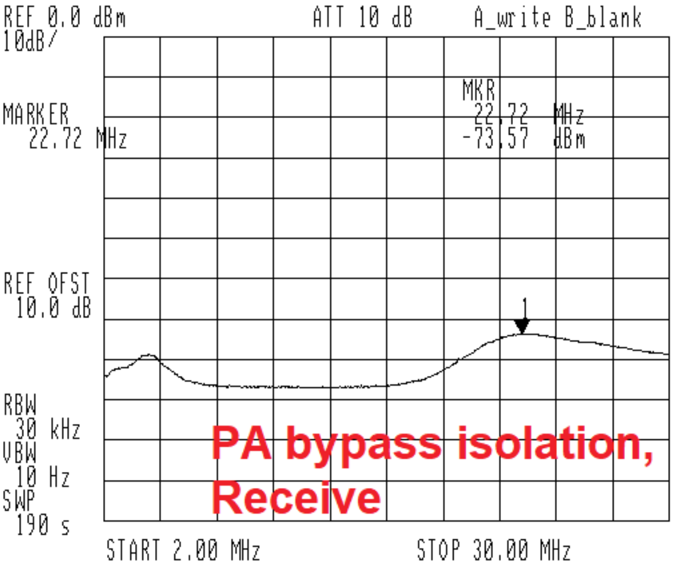
The control signal from the microcontoller is “low” for Transmit, and “high” for Receive. The series PIN diode switch formed by D4 and D5 is biased “Off” by a large reverse bias that is larger than the peak of the RF waveform from the PA output. This is achieved by rectifying and voltage-doubling the RF waveform; almost no current is drained so this does not affect the PA output amplitude. It’s a neat and simple, but highly effective, way to ensure a higher reverse bias voltage than the peak of the RF waveform!

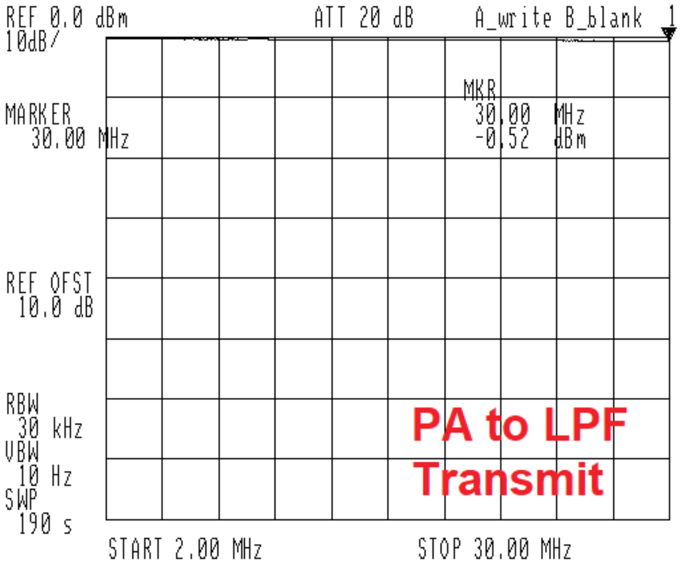
The spectrum analyser measurements show 75-85dB of PA bypass isolation in Receive, which is plenty.

On transmit, a fraction of dB is lost through the single PIN diode switch leg formed by D3.

The PA is isolated from the Receive signal path during receive by the same PIN diode, D3; the isolation ranges from 50dB on the 160m band to 28dB on 10m; however, the isolation along this path is not critical; it is only needed to maintain 50-ohm impedance through the receive path without unwanted loading from the inactive PA output.

Overall this solid state Transmit/Receive switch achieved all the objectives, high performance and low cost, and is fast enough to permit full break-in operation.



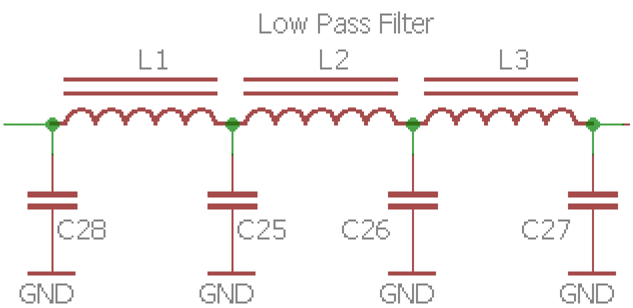

## Test and alignment tools

The included test and alignment tools in the QCX CW transceiver proved to be very popular indeed. It is possible to set up and align your QCX transceiver using all the built-in test functions and no additional equipment. For the QSX, is was clear to me that again the test equipment should be included in the radio. Furthermore, I should add a couple of new features! The list (at time of writing) is:

- Signal Generator
- Frequency counter
- DVM
- RF Power meter
- SWR meter
- Inductance meter
- Spectrum analyser function for filter adjustment

### Inductance meter

It became clear from the QCX experience that the inductors in the Low Pass Filter (LPF) are quite critical on some band versions of the radio. The LPF design has been popular for several decades, it is a 7-element design by Ed W3NQN, published for many years on the G-QRP Club web site's technical pages and used in the QRP Labs Low Pass Filter kit http://qrp-labs.com/lpfkit .

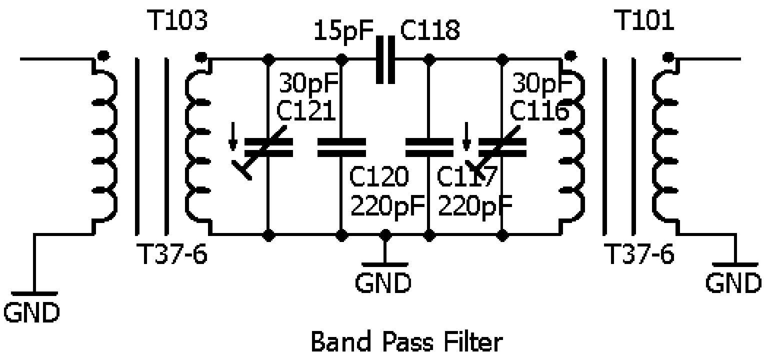



Component tolerances can lead to the filter cut-off being lower than the operating frequency, which causes attenuation at the operating frequency. Use of an inductance meter to measure the actual inductance of L1, L2 and L3 prior to installation in the filters would overcome these variations and optimize performance. But not everyone owns an inductance meter – and even if they do, then there's the debate about how valid the readings are, when performed at several hundred kHz whereas the inductor will be used in HF circuits.

Therefore, let's include an inductance meter inside the radio, to help when assembling the kit! The QSX includes a resonant circuit formed of a known capacitance and the unknown inductance; the microcontroller sweeps the signal generator through a range of frequencies to determine the resonance, then calculates the actual inductance. In this way the constructor can check the inductance values of all the toroids before installing them.

## Spectrum analyser function for filter adjustment

I'm particularly proud of this one which I think came out very nicely.




The QSX transceiver uses a double-tuned resonant circuit Band Pass Filter (BPF). This is a more complex, and higher performance filter than the simplified single-tuned transformer arrangement in the QCX CW transceiver.

There are two adjustments, the 30pF trimmer capacitors shown here as C121 and C116. If these are adjusted with a simple on-screen peaking measurement as used in the QCX CW transceiver, then it is quite easy to adjust the filter incorrectly. If the two resonances are too far apart, they create a double-peaked filter characteristic; each peak having a too-high insertion loss.

A spectrum analyser with tracking generator is the ideal way to align such a filter. But not all of us are lucky enough to possess such an expensive piece of test equipment. I have a 25-year old one, which still works beautifully; I could justify its still-significant n'th hand expense as I use it a lot in the course of my work for QRP Labs.


So, the idea came to me to try to build a simplified spectrum analyser with tracking generator, into the QSX kit (a.k.a. scalar network analyzer). This would allow adjustment of the BPF filter trimmer capacitors and also provide lots of fun, and an introduction to how these instruments work, for those majority of constructors who don't have one.


The QCX CW transceiver achieves its BPF peaking function by injecting a low-level signal into the receiver input, using the Si5351A's third output as a signal generator mode; then measuring the signal amplitude arriving through the receiver signal path and displaying that on-screen (see right). It is very useful but falls short of the actual filter characteristic trace plotted on a real spectrum analyzer.

The same principles can be applied in QSX; there is a signal generator which can be injected at a low-level into the RF input, and the receiver, whose signal strength can be accurately measured. Both the signal generator, and the receiver, can have their operating frequency swept. Which is of course, exactly what a spectrum analyser does!

The remaining problem to be solved is how to display the resulting filter characteristic as a nice trace; remember that the QSX has a low-cost minimalist 16 x 2 alphanumeric display, not a fancy graphic display. Well, 8 characters can be custom-defined with your own choice of pixel-map, 5 x 8 dots to configure as you please. So, for the spectrum analyser function, these eight custom-characters are reprogrammed dynamically on every sweep of the spectrum, to represent the filter passband trace. It actually looks really good in practice. The analyzer sweep and update occurs around 2.5 times per second which is fast enough to be able to adjust the trimmer capacitor and see the result in real-time intuitively.


Initially I plotted the trace as black pixels on the empty background, it looked cool but not *very* cool… the problem being the gaps between the alphanumeric character rows and columns, which cannot be filled in. Then I noted that the gaps between rows and columns are about 1 pixel sized. If I consider the display area to be 23 x 17 pixels, instead of the real 20 x 16 pixels, and invert every pixel, so that the trace is now yellow-on-black: then the formerly unwanted row and column gaps now look just like the graticule of a spectrum analyser display! It was necessary to re-scale everything to 23 x 17 pixels, and also for the sake of time saving, to not bother to sweep the three frequencies that falls under the vertical graticule lines.


The image (above right) explains the display labelling. The QSX left knob adjusts the center frequency, while the right knob is used to adjust the frequency per horizontal division. So, you can center on the amateur band whose BPF you are trying to tune, and zoom in or out to get a wider or narrower view of the filter passband shape. The vertical scale is adjusted automatically to fill the screen and the dB per vertical division shown in the top right. When the system is able to auto-detect the peak and -3dB points it interpolates to calculate the -3dB bandwidth and shows it at the top left. An additional nice feature is the row of yellow pixels on the bottom of the display area. These pixels are coloured yellow to show the extent of the ham band, so that you can easily see how you need to adjust the filter center frequency.


## DSP Magic

I don't intend to say too much about the Digital Signal Processing (DSP) side of things, to accomplish all this SDR magic. Not least of all, because at time of writing I can't say that I have it all so neatly arranged in my head that I would be able to express it remotely coherently. Aside from that you could fill a whole book talking about DSP.

But DSP is like a box of kids' magic tricks. You may not be a real magician. But if you go and buy this box of tricks, you can sure look like one. In this case, Google is your friend, and can find you many fine examples of DSP in action. That's my state right now…

The most important requirement for the DSP in the QSX SDR receiver is the 90-degree phase shift between the I and Q channels. After the phase shift, adding the two sides together eliminates the unwanted sideband, leaving SSB.

Other DSP functions used include:

- Audio filtering
- Fast Fourier Transform
- Noise reduction
- Notch filter
- CW amplitude detection for the decoder
- Speech compression
- AGC

## Other interfaces

The STM32F4 microcontroller has many other interface peripherals built in, and lots of Input/Output pins which can be programmed for useful functions.

Of particular note is the USB support (Universal Serial Bus, not Upper Sideband!). There are two USB interfaces. In the QSX one is used in the USB Device mode, and one in USB Host mode.

### USB Host

The USB host (standard USB type-A connector) can be used to plug in a USB flash drive. This is used for firmware upgrades to the QSX. So, there is no need to own any special programming hardware, or any special software on the PC. Firmware updates are extremely simple. All you need to do is copy the firmware file to a USB flash drive, plug it in the back of the QSX, and use the update the firmware from the setup menu.

A standard USB keyboard can also be plugged in to the QSX, and it makes it possible to send CW from the keyboard, and to operate PSK31 and RTTY without any PC. Just the radio/keyboard alone.

### USB Device

The USB device (standard USB type-B connector) is used to connect to a PC. The QSX is made to appear to the PC as two separate devices: one, a virtual Serial COM port which can be used for CAT control; the other, a 24-bit stereo sound card, which can be used for raw I and Q channels (to use an SDR program on the PC) or it can be used for demodulated audio to operate with any digital mode software package such as WSJT-X for FT8.

### Other digital facilities

The STM32F4 also has serial ports, both traditional USART and I2C; these can be used to connect to more external devices and sensors; and plenty of digital I/O pins that can be programmed for your particular purposes.

The QSX exposes all of these functions via a BASIC scripting language. Another huge topic all of its own.

Suffice it to say: with such a powerful microcontroller onboard to handle the DSP, there is a truly staggering array of possible uses the radio could be put to!

## Single-band vs Multi-band QSX transceiver

The QSX transceiver will initially be available in a monoband 40m version, with an optional extruded aluminium enclosure. Subsequently it will be upgradable with a 10-band filter board, with individual Band Pass Filters per band to maintain high receiver performance, and enough Low Pass Filters to ensure harmonic content is attenuated comfortably meeting regulatory standards.

## QSX transceiver more information

All available information on the QSX is on the QSX web page http://qrp-labs.com/qsx. The target price for the QSX all-mode all-band 10W HF transceiver, including 10W Linear, 10-band filter module, enclosure and all necessary hardware fittings, is around $150.

## Conclusion

I hope this article has generated some ideas for you to use in your own homebrew projects, and provides some insight into the design process of this ambitious transceiver kit. This article necessarily only scratches the surface of all the intricate details. More can be learned from the manuals and other information available at http://qrp-labs.com/qsx . I hope that you have found this article interesting and educational.

73 de Hans G0UPL
http://qrp-labs.com