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Table of contents

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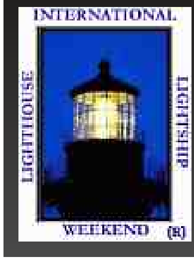
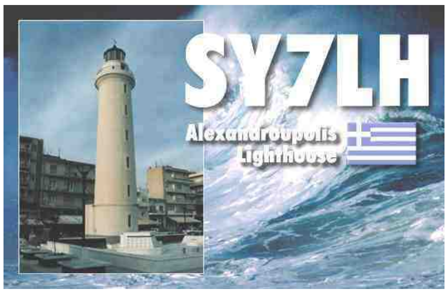
INTERNATIONAL LIGHTHOUSE/LIGHTSHIP "WEEK-END"

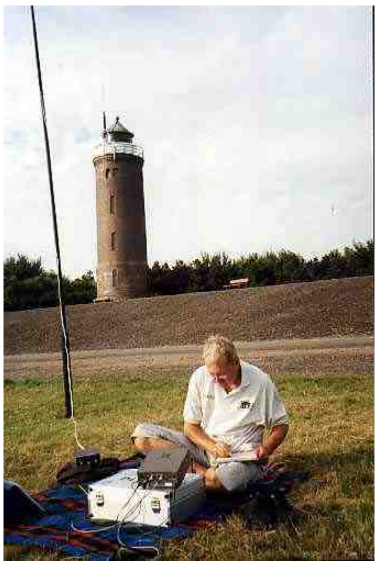


By Kevin Mulcahy VK2CE

It all started in 1994 during a wet wintry evening when two members of the AYR Amateur Radio Group in Scotland, John GM4OOU and the late Mike GM4SUC, after a club meeting were talking about creating an event in the summer when club members could get out on a sunny weekend and play radio. Various themes were considered; ports, airports, historic Scotland sites, the Firths of Scotland, castles etc. but it was finally decided that lighthouses of Scotland would be ideal.

Following research it was discovered that the lighthouses of Scotland were controlled by the Northern Lighthouse Board in Edinburgh who were not only responsible for the lighthouses of Scotland, but also around the Isle of Man. Approval was sought and obtained from the Northern Lighthouse Board to establish amateur radio stations adjacent to their property. In February 1995 an invitation was sent to all Scottish clubs and the Isle of Man club to join in the fun of a weekend, to be called the Northern Lighthouse Activity Weekend, by establishing an amateur radio station at a lighthouse during the third weekend in August. This first year's event saw 11 stations established at lighthouses, operating primarily on the HF bands, with each station making approximately 750 QSOs over the weekend.





The following year the Scottish clubs were involved in a weekend activity with the theme of Scottish Firths (river estuaries), so two years elapsed before the next Northern Lighthouse Activity Weekend. During this period Anne-Grete OZ3AE enquired through a letter to Practical Wireless if there was any lighthouse activity on amateur radio. Following discussions with her it was decided that Danish stations could join in the fun of the weekend. Quickly Germany, South Africa and France asked to join, so the name of weekend was changed to The International Lighthouse/Lightship Weekend . It was at this time that John, GM4OOU, due to pressure of work, had to cease his connections with the event.

The weekend became an annual event taking place over the third full weekend in August and has slowly grown in popularity and in 1999 there were 204 lighthouse/lightship stations in 36 countries until 2008 when 406 stations in 50 countries took part.

Full statistics and guidelines for participation can be found at on the ILLW web site at :

http://illw.net

The main reason the event has become so popular is because it is NOT a contest. It is a relaxed fun weekend without the pressure of a contest. The guidelines are simple and the onus is on the operators to act within the spirit of the weekend which is simply to expose amateur radio and the plight of lighthouses to the public. This is why it is important for the ham station to be as close to the lighthouse/lightship as possible and with the controlling body's approval.

A few years ago the International Association of Lighthouse Keepers decided to have an annual open day for lighthouses all around the world to encourage visitors to visit at their lighthouses. They decided that no better day could be decided upon other than the Sunday of the ILLW. This move has been highly successful as the media have become involved in quite a few of the countries involved in the event.

This year's event takes place on 15-16 August 2009 so if you haven't done so already, find a lighthouse nearby and get a group together or do it solo and fire up a lighthouse station. In most cases if you don't intend operating from within the lighthouse itself or one of its cottages, you really don't need to get any approval. Most first time entrants are so enthused with the event that they return year after year. A report from the Burlington ARC, Canada summed their first participation in these few words:

"The greatest delight of the day was the active participation of the visiting children who showed a remarkable interest in the whole idea of amateur radio, especially the use of Morse Code.

It was an honour and a delight to participate in this adventure and we look forward with increased enthusiasm to next year's participation."

As you can see from the website, Mike Dalrymple passed away in December 2005. He was the Treasurer of the Ayr Amateur Radio Group and one of their members has taken on Mike's roll as the PR man and main co-ordinator. The event is now dedicated to Mike's memory as is the official web site:

http://illw.net

Where you will find the event guidelines, an on line entry form and lists of participating lighthouse since 1999.

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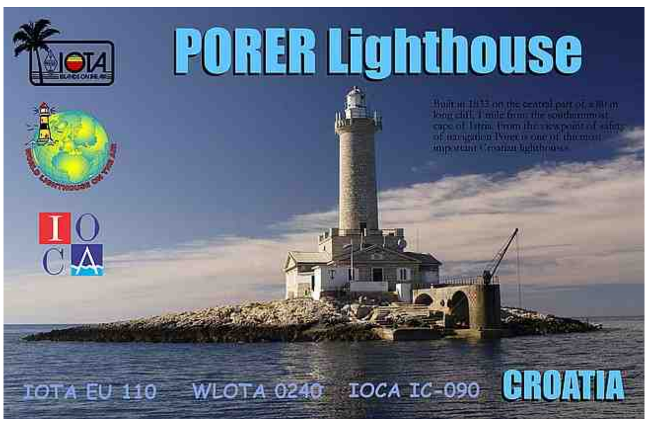

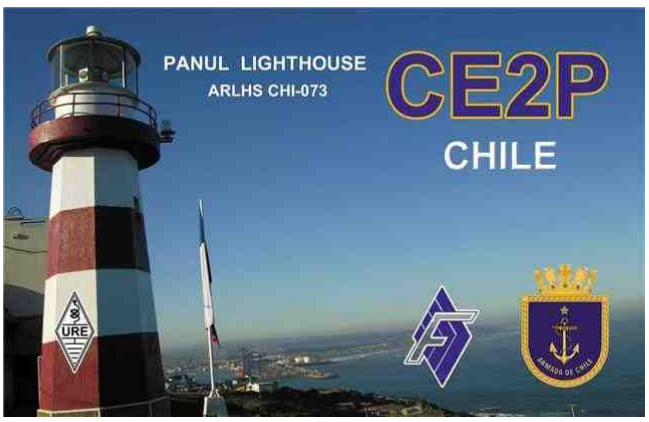

HI-FI or Enhanced Single Sideband (ESSB) for Amateur Radio By WB8EVI

For years I was one of those operators who believed you had to have a Heil DX microphone cartridge to have the best sound for chasing those far away stations. I owned a Heil Proset with the DX microphone cartridge in it, and I worked my fair share of DX stations with it--it worked. Then about a year ago, I started listening to 14178, 3851, etc, to some of those HI-FI audio guys. Gee some of them sure sounded good to me, and they did not take up too much bandwidth either. They also were not fatiguing to listen to for long periods of time, unlike some operators who have rather limited transmitted audio frequency response. So after careful consideration of my available spare funds, meaning I had some mad money, I decided to take the plunge and give ESSB a try. Besides, I was bored and needed a new radio project to work on. My goal was to sound the best that I could, while still staying within a reasonable bandwidth limitation, and a reasonable budget.

I started listening to those people that sounded good to me and took notes on what equipment each of them used. I spent weeks doing this and I believe it was time well spent. Then I checked out every website I could find, looking up the equipment they were using. I decided to start purchasing one piece at a time, paying cash as I went, to make it relatively painless. Buying one piece at a time also allowed me time to become familiar with how each piece worked. I am the kind of guy who likes to see something before buying it, if possible. I also like to support local dealers, as long as their price is within reason. My local Guitar Center is the place to go! They have lots of equipment to see, several locations within driving distance, a great website to do research on, and will compete with anyone’s price. I also found the sales people very friendly and helpful with suggestions once I explained what I was trying to accomplish.

I have always wanted to try a condenser type of microphone. Most professional or studio microphones are this type. They have a wide and flat frequency response. The downside is that they require what is known as a phantom power supply. Usually this is 48 volts, and is supplied by the microphone preamplifier. With $400 burning a hole in my pocket, I went to my local Guitar Center and came home with an AKG Perception 200 condenser microphone, a Behringer T1953 preamplifier, a small microphone desk stand with boom, and a high quality Monster audio cable with XLR connectors on each end. Oh yes, most professional microphones use XLR connectors, and have balanced output. Note that there is no PTT (push to talk) switch on these microphones, since they are not designed for Amateur Radio use. I chose to use a foot operated switch to key my transmitter with. I have a Heil footswitch, and think it is perfect.

The next step was to interface my new equipment to my Yaesu FT-1000MP transceiver. Professional audio equipment levels are way too high to directly connect with “consumer” audio equipment. So an interface is needed to drop the level down to where we can use it. W2IHY sells what he calls an I-box. A lot of people use those, and they are excellent units for converting from professional audio level down to consumer audio level. I decided I would rather try building an interface circuit myself. I have built three different interfaces so far, and I have learned a lot from each of them. I scrapped an old Telebit Worldblazer desktop modem and salvaged a very nice 600 ohm transformer out of it. Do not try to use the inexpensive transformer sold at your neighborhood electronics store. They do not have very good frequency response, at least not the one that I tried. If you try the modem route, find an external modem, not the internal modems that are on a card. From what I have experienced, the physically larger transformers tend to sound better. I have an old Heathkit phone patch, and someday I may try the transformer in it; maybe even try making an interface out of the whole unit since it does have a nice meter. I added a couple of RF chokes, resistors, capacitors, a level pot, a switch for “ground lift” and a couple of connectors with my modem transformer. All this went into a plastic box—do not use a metal box! You do not need the extra RF shielding, and you probably will not want the grounds connected together as this creates unwanted hum. I wired in a switch to connect the grounds and I can certainly hear the difference with the switch closed, and the grounds connected. If you do not want to build your own interface, the W2IHY box is an excellent choice. You can also get a professional DI box, such as the Behringer Ultra-DI 100, at your local Guitar Center. This Behringer unit requires a 9 volt battery or 48 volt power, since it has a buffer amplifier in it. I tried using an Ultra-DI 100. It worked fine, sounded great, was very quiet, and was about $35 at the time. My homemade box has both ¼ inch TRS (tip, ring, sleeve, with 3 conductors like a stereo headphone jack) and phono jacks installed on it for both input and output. I like having a little extra flexibility for interfacing with my equipment. For the balanced audio portion of the audio chain, you need to use either XLR or ¼ inch TRS jacks. The interface box to transceiver circuit is just two conductor unbalanced, so I used phono jacks for that.

Ready for my first test; I wired the AKG microphone into the Behringer preamplifier with an XLR cable, and the output of the preamplifier to the input of my interface box with a TRS cable. Next I ran the output of my interface box into the phone patch input jack on the rear of the Yaesu transceiver, using a phono to phono cable. Then I unplugged my Yaesu microphone from the front of the transceiver. I now use a Heil footswitch to key the transmitter with, plugged into the rear of the transceiver. Not only did this setup work, but it even sounded reasonably good to me, while monitoring my transmitted signal with headphones. Wow, this condenser microphone is much more sensitive than the Yaesu desk microphone I was using! I can hear my dog walking across the floor upstairs. I also can hear the washer and dryer running in the room above me. Microphone placement, and level controls suddenly became very important.

The next piece I purchased was a Behringer Feedback Destroyer Pro DSP1124P for about $100. This unit has two channels, like the preamplifier does. It also has both XLR and TRS input and output jacks, again like the preamplifier. Each of the two main channels has 12 individual channels that can be used as either a feedback destroyer (like a heterodyne remover in a receiver) or as an audio equalizer. I may try using one of the main channels on receive audio to remove heterodynes someday, but for now I use all 24 channels for transmit audio equalization. I adjusted 12 channels for what sounded good in my monitor headphones plugged into the transceiver, and the other 12 channels to cutoff all audio above 4 KHz. Remember, I do not want to be so wide that I cause trouble. I just want to sound good.

Time to talk about balanced audio interconnect cables. Either XLR or ¼ inch TRS connectors will work fine. I tried using an inexpensive patch bay mounted in my audio rack, with the input and output of every piece in the audio chain going to a ¼ inch TRS jack on the rear of the patch bay. This allowed me to monitor with headphones between each piece, to inject tone to set levels, or to patch around a piece of equipment all from the front of the patch bay. So I only used an XLR interconnect cable between the microphone and the preamplifier. This I did not have wired through the patch bay. Since the rack of audio gear is not far away from my HF linear amplifier, I thought it would be wise to get good quality, well shielded audio cables for everything. Monster makes very impressive cables, and at about $25 each, this really adds up. They are very high quality, and guaranteed for life. I have a few of them. Live Wire cables look almost as good to me, and are a little more reasonably priced. I have a few of those also. Then I discovered that Radio Shack has a very nice cable for $14 with ¼ inch TRS connectors on each end. These are also an interesting purple color. This is a good thing. When you have a large pile of cables behind a rack of equipment, it is nice to have a few cables that are a different color, so you can tell them apart easier. I soon realized that I should have purchased a better patch bay. The plugs kept falling out of the jacks on the rear of the unit I had. After fighting with it a few weeks, I removed the patch bay, and now I have enough spare cables to last me for many years.

Now that I could limit my audio to a reasonable bandwidth with the equalizer, it was time to bypass the internal microphone circuitry in the Yaesu FT-1000MP altogether for improved frequency response of my transmitted audio. Cringing only slightly, I drilled a hole in the back panel of my transceiver, using some of that blue painter's tape and a paper towel to catch the metal filings. I was much too lazy to try and remove the back panel. I installed a phono jack, and wired it to the input of the balanced modulator via a 220 uf capacitor. This step is not for the faint of heart. It requires soldering directly to a pin on the balanced modulator chip in the transceiver. There are several ways you can damage your transceiver doing all of this, so extreme caution is needed. Next some menu changes had to be made in the transceiver. The EDSP feature must be shut off, etc. The ALC meter still works, so adjusting the transmitted modulation level is easy, using the level pot on the interface box. I tend to run it near the end of the normal ALC range. I did discover that when transmitting CW, my output power varied by what sounds my microphone pickd up! So when operating CW, I must now open the circuit from the microphone somewhere or flip off the phantom power to the microphone. This is easy, as the Behringer microphone preamplifier has a switch for phantom power, and another switch for selecting inputs. If I build another interface box, I may add a switch there, so it will be within easy reach of my operating position. When on the air with this setup, I received many excellent audio reports, so I must have been doing something right.

Next piece was an audio compressor, to give my transmitted audio a little more punch. My son gave me a Behringer compressor as a Christmas gift. I have the one with four separate compressors in it, but I am only using one of them. You can save money and only buy a stereo one, because you only need one channel. Street prices for Behringer units are $100 to $149, depending on the model. DBX makes a nice one for $150, and a really nice one for around $250 (that one would have been my first choice). Many compressors also have other useful features, such as a noise gate, and a limiter. So there are many devices to choose from out there. Like anything else, you can spend a thousand dollars or more if you feel the urge to splurge.

Then I saw a used Behringer Virtualizer Pro at Guitar Center one day for only $50. This is half the price of a new unit. I had to have it, and brought it home with me. This device does A/D conversion on the input. Then it has something like 71 different effects you can use to alter your sound, and finally does D/A conversion for output. I used it to add just a tiny bit of reverberation. I just added enough to notice it if you turned it off, but not enough to be very noticeable or annoying. This added a little polish and a studio quality to the sound. Evidently my unit had a problem, and would not keep the settings that I programmed into it when power was removed. So out it went; maybe to be replaced someday.

Almost done; I only added one more piece of audio processing equipment. I picked up a used CBS Laboratories Audimax III. This piece of equipment was used as a limiter in a commercial radio station many years ago. They were very popular back in the 70's. So this is now the last professional audio processing piece in my transmitter audio chain. It feeds my homemade interface box. The interface box drops the professional level down to consumer level and isolates the ground between the audio rack and my transceiver. It also has the gain control that I now use to set my transmitter modulation level. The interface box sits close to my transceiver, so that I can touch up the level easily when needed, such as when I am using my Collins 30S-1 linear amplifier to get through a pile up.

So the final configuration is: microphone, preamplifier, equalizer, compressor, limiter, interface box, and finally the transceiver. I did add a Monster Pro 2500 power conditioner to the rack of equipment. This $200 unit controls the AC input power to all the other pieces of equipment in the rack. It has a digital voltmeter, lots of jacks, and even a power up delay circuit for some jacks. A little extra filtering and protection on your AC power gives great piece of mind. I use a similar unit manufactured by Furman for my computer equipment.

How does all this sound? I like it a lot. Do I get good audio reports? You bet I do. "Mike, you sound like a broadcast station." "You have great audio." Do I get through DX pileups? I sure do, and I only have a vertical antenna with four buried radials, no tower or directional antenna with lots of gain. It seems to me, that if you sound a little different than everyone else, it increases your changes of being picked out of the pile up by the DX station. Being loud and clean sounding doesn't hurt either. I have been running this way for over a year now, and I have never been told my signal is too wide or that I am splattering. I honestly have not received one negative comment about my transmitted audio, but frequently receive unsolicited positive remarks about how I sound. I will not say my audio is perfect, and if I am trying something new or fiddling with the adjustments, you may hear me and think I sound terrible. Hopefully you will tell me if that happens so I can correct it immediately. Experimentation is part of the fun of Amateur Radio, so I will be making changes and trying new equipment as long as I am able to.

Total cost was about $1200 including everything that I was using. You can easily spend more or less to fit your budget. You don't need a patch bay. You can live without the reverb unit, etc. If I had to do it all over again, I would probably buy a preamplifier, then one box that does the rest of the processing, or maybe one box that even includes the preamplifier. There are several to choose from, with many of them for under $500. I may get away from the "overly sensitive" condenser microphone one of these days, or perhaps try to modify mine to be more directional. I did try a lesser expensive condenser microphone, but was not pleased with how it sounded. I may try a more expensive one someday. A lot of people use the Heil PR-40 dynamic microphone. For about $75 more you can get the Electro Voice RE-20; for yet another $100 the Electro Voice RE-27, or the Shure SM-7B. Any of these I believe would be an excellent choice. I have heard them all on the air, and they all sounded fine to me. They are all also used in professional broadcast studios. Those Electro Voice microphones tend not

to distort much by proximity effect when you talk very close to them. So if you like to talk close to your microphone, one of those would be an excellent choice. A shock mount for the microphone is also a good investment. It isolates the microphone from vibrations. For the $160 I paid for my AKG microphone, I consider it a bargain. I like the way I sound with it. Although being a condenser design, it does require the phantom power supply.

You do not have to tap into the balanced modulator if you would rather not modify your transceiver. Just adding a little external processing, and going into the phone patch jack should make you sound better than the stock microphone plugged into the front of your transceiver. In fact, just processing the audio from any microphone should make it sound better. You can feed your processed audio into your standard microphone jack if you need to. Some ESSB operators go the extra mile and replace the filters in their transceiver for even wider bandwidth on transmit and receive. I do not wish to do that, and did not find it necessary to achieve my goal. I want less bandwidth on received audio most of the time, to make it easier to pull DX stations out of the noise during heavy QRM. Speaking of receive, it is possible someone may tell you they do not like the way you sound with your new wider processed audio, although it has not happened to me yet. It could be that the receiving station is using a very limited bandwidth receiver, and your processed (now slightly wider than normal) transmitted signal is too wide for their receiver passband. You can demonstrate this yourself. Tune in an ESSB station, and switch your receiver IF filter, if you can. At 2 KHz it does not sound very pleasing to listen to, but is much better at 2.4 KHz, better yet at 3 KHz, etc.

So if you want to give this a try, there are lots of websites to give you more info. Just run web searches for ESSB and HI-FI SSB. You will easily find several sites to get you started. I am also happy to talk about it if you hear me on the air.

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Digital speech within 125 Hz bandwidth (DS@125)
By N6IEF - Mike

Objective

To modify and write code needed to convert analog voice into narrow band digital modulation.

Why do this?

The bandwidth of voice is about 2400 Hz. When speech could be reduced to 125 Hz, the gain would be 12.8 dB (19.2X). Processing gain by a computer is cost free. This project receives weak signals 9 dB below SSB (Single Side Band) noise floor of the radio.
NOTE: For those who have not worked with digital modulation, there is a prequel at the end which should be read first.

Generating of the transmit phonemes

A phoneme is to speech as the alphabet is to reading or writing. Since each person sounds different from another, it is clear that the computer must recognize the unique phonemes used by only that person while operating this software. The software must be able to teach itself the phonemes so that it can recognize that person's voice, which is done by reading words shown on the monitor into the microphone while holding down the space bar of the keyboard.

The code used

The 45 phonemes are represented by a code made up of 1's and 0's. The code is similar to a court recorder typing out steno, which can be read back. All code groups start with 1 and end with two or more 0's. Since phonemes are grouped by the shape of the mouth, tongue and lips, the codes used in one group of phonemes should be as different as possible from other groups. Some phonemes are longer than others and they should have a longer code. Of the 53 codes, only 45 are used with eight as spares. This code is exactly the same Varicode used in PSK-31, (Phase Shift Keying with 31 Hz bandwidth).
100, 1100, 10100, 11100, 101100, 111100, 1010100, 1011100, 1101100, 1110100, 1111100, 10101100, 10110100, 10111100, 11010100, 11011100, 11101100, 11110100, 11111100, 101010100, 101011100, 101101100, 101110100, 101111100, 110101100, 110110100, 110111100, 111010100, 111011100, 111101100, 111110100, 111111100, 1010101100, 1010110100, 1010111100, 1011010100, 1011011100, 1011101100, 1011110100, 1011111100, 1101010100, 1101011100, 1101101100, 1101110100, 1101111100, 1110101100, 1110110100, 1110111100, 1111010100, 1111011100, 1111101100, 1111110100, 1111111100
As shown, the code is the fastest speed for each phoneme. By adding one or more extra 0's to any code, the length of that phoneme is stretched by increments of 1/125 of a second. This is very important because voice speed is constantly changing. The original 45 phonemes are expanded to many new phonemes.

The software summary

Voice received through the computer's microphone is converted into numbers, amplified to a constant level, converted into 16 bands of frequency, cut into three parallel 24 mS sections of time, compared in a two-stage process to a library of 45 phonemes that have been made by the operator of the radio, converted to a digital code, stretched to fit the operator's real speech, and sent to the radio in a modified QPSK-125 format (Quadrature Phase Shift Keying with 125 Hz bandwidth) to be transmitted.

The transmit sequence

The transmit sequence starts with the pressing of the space bar on the computer keyboard and continues until the space bar is released. The computer speakers' D/A (Digital to Analog) converter is forced to zero. The AGC (Automatic Gain Control) is un-frozen.

The 400 mS synchronizing alternating series of ones and zeros is sent to the transmit section of the PSK program. This 125 Hz BPSK code is used by the other computers' receiver section of the PSK program to re-synchronize the 125 Hz clock. This insures that the receiver section of the PSK program is sampled in the middle of each code digit and is not sampled during the transitions.

The sampling 66,000 Hz clock starts the A/D (Analog to Digital) converter from the microphone input of the computer. Each clock cycle makes the A/D output a 16-digit signed number. Each number goes to the AGC (Automatic Gain Control) array and the AGC level adjustor.

The AGC is used to amplify the weak signal from the microphone to about 90% of the maximum value for the 16-digit signed number. This is done by TBD (To Be Determined) method. It will use the normal fast attack and slow decay, but it will be frozen when the space bar is not pressed.

Some of the numbers from the AGC level adjustor go to 32 FIR (Finite Impulse Response) low-pass filters. A FIR low-pass filter has a frequency F and a number of taps N and a sampling rate. The problem with filters is the time difference, DPD (Differential Propagation Delay), between the outputs of high frequency filters and the outputs of low frequency filters with the same input to both. The 17 F frequencies for the FIR filters are 8000 Hz, 6083 Hz, 4625 Hz, 3517 Hz, 2674 Hz, 2033 Hz, 1546 Hz, 1176 Hz, 894 Hz, 680 Hz, 517 Hz, 393 Hz, 299 Hz, 227 Hz, 173 Hz, 131 Hz, and 100 Hz.

A first order attempt to solve the DPD problem is to use different sampling frequencies for each group of two FIR filters. The numbers from the A/D are at a 66,000 Hz rate. When every fourth number is used, the new sampling rate is 16,500 Hz, or 66,000 Hz divided by 4 is 16,500 Hz. The 16 divide-by numbers are 4, 5, 7, 9, 12, 16, 21, 28, 36, 48, 63, 82, 110, 145, 190, and 251.

For example, the divided-by-4 sampling rate is used by the two highest frequency FIR low-pass filters, 8000 Hz and 6083 Hz. Both FIR low-pass filters need to have the same number of taps N to insure that their output numbers are available at the same time, or zero DPD. By subtracting the output numbers from these two FIR low-pass filters, new numbers are created at the same sampling rate. These numbers are approximately the instantaneous amplitude of the sound between the two frequencies. In the same way the other numbers are made by two FIR low-pass filters for each of the other 15 frequency bands, with each associated sampling rate. NOTE: Each set of two FIR low-pass filters has the same sampling rate, and taps N, and their DPD is zero, so their output numbers can be subtracted.

The DPD between frequency bands is not zero, but this doesn't matter because the numbers between frequency bands are never used together.

This complicated process is being done to change the time-amplitude energy of voice into the time-frequency energy of speech. Some people say that there are 44 phonemes and one extra phoneme for no sound. Dividing the A/D sample clock rate of 66,000 Hz by 1584 makes the phoneme sample interval. This interval is 24 mS. After the start of the phoneme sample interval, the absolute values of the next 12 numbers from each of the 16 frequency bands are examined for the largest value. This is called the peak search process. Just before the end of the interval, say at count 1583 of 1584, the 16 peak numbers are put into the phoneme sample array. The phoneme sample array can be visualized as a blue transparency bar-graph with 16 vertical columns, but it actually is a 16 by 1 array of numbers. This process re-synchronizes the DPD problem to the original 66,000 Hz sample clock of the microphone input D/A.

In order not to miss a phoneme, the above procedure is repeated in parallel, two other times by starting at counts 528 and 1056 from the original 1 to 1584. This insures a new phoneme sample array every 8 mS. The 24 mS time interval is used to detect each of the 45 phonemes, even when the phoneme lasts longer. To reduce the chances of receiving part of one phoneme and part of another phoneme, a new set of 16 peak numbers is started every 528 numbers or 8 mS. Overlapping numbers insure that a phoneme is not missed.

One of three parallel phoneme comparators takes its phoneme sample array and compares it to one of 45 arrays of 16 numbers from the phoneme library, visualized as a yellow transparency bar-graph. By subtracting one array from the other array, visualized as overlapping the yellow and the blue transparencies, the differences are visualized as blue and yellow and the common part of the bar-graph is visualized as green. To amplify these 16 differences, they are multiplied by themselves to make them all positive numbers and these 16 positive numbers are added together to make the single error number for that comparison. In the same way, the next array of 16 numbers from the phoneme library is subtracted from the original phoneme sample array until all 45 arrays from the phoneme library are used. The phoneme code for the three smallest error numbers of the 45 possible error numbers is sent to the guesser along with their error numbers and code sizes from the phoneme library. Although this process takes some time, the output rate should be the same as the input rate of 24 mS. Since there are three peak detectors with three comparators staggered 8 mS apart, a phoneme code with its error number and code size is sent into the guesser every 8 mS. The code size is a number from three to ten, which is the number of ones and zeros in that phoneme code.

The guesser is used to determine what code should be sent to the output Q. The guesser is like a Q with three levels. Three phoneme codes and their error numbers enter the back of the guesser and work their way down to the front of the guesser. So there are always nine phoneme codes in the guesser. Whenever three codes are entered, three other codes are removed. When there are three of the same phoneme codes in the guesser, the error number of that phoneme code in the front of the guesser is divided by three. When there are two of the same phoneme codes in the guesser, the error number of that phoneme code in the front of the guesser is divided by two. After the divides, the phoneme code and the code size of the smallest error number of the three in the front of the guesser is sent to the output Q. This happens every 8 mS.

The output Q is a buffer that is used to fix problems that happen when one phoneme transitions to another phoneme in our speech. The output Q is used to sort the phoneme codes into groups, like sorting cards into suits. When the phoneme code sent to the back of the output Q is the same as any of the two previous phoneme codes in the output Q, the new phoneme code is moved forward to that same phoneme code group.

One phoneme code is removed from the front of the output Q as each digit of the phoneme code is sent to the transmit part of the PSK program. But before a new phoneme code group is sent to the transmit part of the PSK program, the number of phoneme codes in that group is checked to see that they are more than the minimum number for that code size. When they are less than the minimum number, the group is removed from the output Q.

An extra zero is sent to the transmit part of the PSK program as each extra phoneme code beyond the phoneme code size is removed from the output Q. An example would be the phoneme code of 10100, which is different from 10100000 because the sound of the second code last 3/125 of a second longer. Although there only 45 fundamental phoneme codes, there are hundreds of extensions. No extra zeros are sent to the special phoneme code of 100, but the code could repeat when needed.

When the output Q does not contain enough of the phoneme codes, each digit of the code is still sent to the transmit part of the PSK program, but the output Q does not move to the next phoneme code until all the digits of that code are sent. This part is TBD, but there must be a way to make the fill in code slots equal to the slots removed by the non minimum number code.
Code sizes (Minimum number) are 3 (2), 4 (2), 5 (3), 6 (4), 7 (4), 8 (5), 9 (6) and 10 (7).
At the start of each transmission sequence, when the space bar on the computer keyboard is pressed, the guesser and output Q are filled with a quantity of the code 100, the special code for no-sound, because the computer takes some time for the numbers from the microphone A/D to be processed. At the start of a transmission, these leading 100 special codes are removed from the output Q and the ones and zeros of the rest of the real phoneme codes are sent to the transmit part of the PSK program.
Each digit of the phoneme code is sent serially at a 8 mS rate. This is the same rate at which the error numbers enter the guesser and the same rate at which the audio code modulates the radio transmitter. At the end of each transmission, the space bar on the computer keyboard is released, all 100 special codes on the back of the output Q are removed and the special end code of 1111111111 is sent to the output Q and then to the transmit part of the PSK program. This sets the squelch of the other computers' receiver section of the PSK program.
With today's computers having 3 GHz clocks and quad processors, twelve billion operations can be done every second. Speech recognition software in 2004 did not have this computer power and did not work very well. In the event the guesser makes a mistake, our brains deal with the occasional anomalous sound from the computer's speaker. Words may sound mispronounced, but we should know what they mean.
This transmit sequence may look like speech recognition software, but it has two differences. First, speech-to-text software requires the ability to handle spelling and meaning. An example would be the homonyms "to," "two," and "too." Most of the code for speech recognition software would not be used. Second, speech recognition software has no time limit from sound to text. The transmit sequence of this software requires a minimum fixed time delay.

The receiver sequence

The receiver sequence starts with the release of the space bar on the computer keyboard and continues until the space bar is pressed. The microphone A/D is forced to zero. The guesser is not allowed to send more codes to the output Q.
After the 400 mS BPSK signal re-synchronizes the 125 Hz clock and releases the squelch, the ones and zeros coming from the receive part of the PSK program are sent to the phoneme comparator. The first one after two consecutive zeros starts a new phoneme code. The first code of ones and zeros assumes a 100 special code for no-sound has been detected. Since the phoneme code is sent serially, each digit goes to the phoneme code library one at a time where half of the library is eliminated with each digit after the first one. When the next digit is received, half of the half of the library is eliminated and so on until two consecutive zeros are detected. That is when the phoneme code is found. Then four phoneme arrays (audio clips) are found from the phoneme library. The first phoneme array is called the main array. It is ((the code size – 2) X 8 mS) long and has ((the code size – 2) X 528) numbers. The next phoneme array is called the zero array. It is 8 mS long and has 528 numbers. The next phoneme array is called the third array. It is the same as the zero array, but each of the numbers is divided by three. The last phoneme array is called the two-thirds array. It is the same as the third array, but each of the numbers is multiplied by two. Normally a .wav file would be used for an audio clip, but that won't work for 8 mS to 64 mS sound clips with 528 to 4224 numbers in each array. A new way to send the numbers to the speaker D/A will be made by a TBD method.

When the first two consecutive zeros of the present phoneme code are detected, each of the numbers in the present third array and each of the numbers in the previous two-thirds array are added in the first blender array. Then each of the numbers in the present two-thirds array and each of the numbers in the previous third array are added in the second blender array. Then the first blender array is sent to the sound card D/A buffer of the computer, followed by second blender array, followed by the main array of the present phoneme code. When another zero is detected after the first two zeros of the present phoneme code, the zero array of the present phoneme code is sent to the sound card D/A buffer for each extra zero.

The two 8 mS blender arrays are used to ease the transition from one phoneme to the next phoneme when played on the computer's speaker.

Then the next detected phoneme code is sent to the sound card D/A buffer and so forth. The sampling rate for the D/A is 66,000 Hz because 66,000 Hz was used to make the original phoneme code arrays in the look-up library. Although this example uses one set of phoneme voice clips for each phoneme code, the computer contains 11 other sets of phoneme voice clips, which can be selected by the operator pressing one of the F1 through F12 keys on the computer keyboard.

Making the operator’s phonemes sequence

Before doing the transmit sequence the phoneme library arrays must be known. This is a one-time only event, which must be done before the computer is connected to the radio. The operator says words into the microphone that are displayed on the computer monitor, while holding down the space bar on the keyboard. The same microphone and A/D converter from the transmit section are used to make the numbers of the phoneme, which are then applied to the same FIR filters. After the start of the phoneme sample interval, the absolute value of the next 12 numbers from each of the 16 frequency bands are examined for the largest value. This is the same peak search process as in the transmit section. Just before the end of the interval, say at count 1543 of 1584, the 16 peak numbers are put into the phoneme sample array. The phoneme sample array becomes the library value for that phoneme. But this library value might be wrong. So the word should be repeated and averaged. When the change in the average is small, then there is enough information to use the array. This needs to be done for all 44 phonemes. The no-sound phoneme is the only exception. No testing is required. Any DPD problems are exactly the same in both the transmit sequence and the making operator’s phonemes sequence, which negate each other.

Making the library sequence at the distribution

The main, zero, third and two-third arrays used in the library of the receive section needs to be made. Twelve different people should record the 44 phonemes. This will be done in the lab with audio spectrum analyzers and high tech computers. Each of the numbers in an array must start and end at zero crossing with a positive slope. This is to prevent discontinuities when any two sets of numbers are connected then played into the computer speaker. After the main phoneme arrays are made, the zero arrays are made. This could be done in the lab by changing individual numbers in the zero array for best sound when connected and played on the computer’s speaker. The third array and the two-thirds array are easy to do.

Conclusion

At this time this project has died of apathy. Since I was the only one in the world who wanted to do it, this project has ended on 1-1-2009. Please contact me at mike.lebo@gmail.com or 858-278-5851 or Skype (Michael E. Lebo) if you have any questions.

A prequel to digital speech within 125 Hz bandwidth

Phase Shift Keying (PSK) is composed of two parts, Bi-Phase Shift Keying (BPSK) and Quadrature Phase Shift Keying (QPSK). This is a narrative on how BPSK-31 and QPSK-125 works. It also explains some of the properties of voice and how they relate to digital speech.

In the beginning of radio the carrier was modulated by turning it on and off, On-Off Keying (OOK). This became known as CW, when used with Morris code. Phone was added when the modulation was changed to AM. This form of modulation has a carrier and a upper and a lower sidebands. An improvement was made when one of the sidebands and the carrier were remove, which allowed better use of the radio spectrum and better efficiency of the transmitted power. This signal sideband (SSB) modulation is what we use in our radios today for point to point communication. By connecting our radios to computers, other forms of digital modulation can be created.

The details of how a radio is connected to a computer are complex and not covered here, but it happens somehow.

Inside a computer are numbers. A program could make numbers that represent a digital audio sin wave and send these numbers at a sample rate to the hardware of the audio sound card, where they are converted into a AC voltage. The part of the sound card that does this is the Digital to Analog converter (D/A). Then this AC voltage is used to modulate the transmitter part of the radio, which creates a constant carrier and that is sent to the antenna. If the program was modified to only make number when a key on the keyboard was pressed, lets say the notorious "any key", and the operator was to press the key using Morris code, a CW transmission could be sent. This is silly to use a computer and keyboard as a telegraph key, but is show the point that the computer could modulate the audio which in turn modulates the radio.

Two things are needed to turn this silly program into BPSK-31. First each key on the keyboard needs to be given a number or code. This is called the Varicode. The rules for this code are that the first digit must start with a one and the last two digits must be zeros. There 128 codes used in this computer keyboard transmission. They include the alphabet, both upper and lower case, numbers, punctuation, special characters and typing operators like carriage returns and line feeds. The number of digits in each code ranges from three to fourteen, with the most used keys given the smaller number of digits, just like Morris code. NOTE: Some of you use the Caps Lock key while typing. In today's text messaging world, this is called shouting or yelling. Capitol letters have more digits than lower case letters and take more time to transmit. As each key is pressed on the keyboard, its Varicode is sent to a buffer for storage and the ones and zeros are then sent to the phase shift modulator serially at a 31.25 Hz rate. Why pick 31.25 Hz? It turns out that there is a 8,000 Hz oscillator in each sound card. No matter what kind of sound card or what kind of computer or what operating system is used, they all have this 8,000 Hz oscillator. 8,000 Hz divided by 256 is 31.25 Hz. 31.25 Hz is the rate that each of the digit in the buffer is removed.

The other thing needed to turn this into BPSK-31 is to reverse the phase of the audio made by the numbers in the computer. This is easy to do by negating each of the numbers sent to the audio sound card when a one is removed from the buffer and not negating each of the numbers when a zero is removed from the buffer. But there is big problem. When there is an abrupt transition in the voltage, distortion accrues in the form of harmonics. The only number that does not make harmonic distortion when negated is zero. So the values of the numbers are slowly reduced to zero at the transition from a one to a zero or from a zero to a one, and slowly enlarged to full value after the transitions. A formula has been developed to adjust the slops of the reduction and enlargement so that minimum distortion happens.

Now lets take a look at the receive part of BPSK-31. Again the details of how a radio is connected to a computer are complex and not covered here, but it happens somehow.

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Now lets take a look at the receive part of BPSK-31. Again the details of how a radio is connected to a computer are complex and not covered here, but it happens somehow.

The sound from the radio enters the computer through the sound card. The voltage of the sound goes to a Analog to Digital converter (A/D), where numbers are made at the sound card sample rate. These number go to three pieces of test equipment and the rest of the program.

The first piece of test equipment is the audio spectrum analyzer. The horizontal scale of this graph shows the frequency with a full range of about 100 Hz to 3,000 Hz on a liner scale. The vertical scale of this graph shows the amplitude at that frequency on a logarithmic scale. The noise seen on the display rolls off at about 300 Hz and about 2,400 Hz. These frequencies are made from the hardware filters in the radio. A strong signal could be seen as vertical lines above the noise. The computer mouse could move the cursor to the signal and the scale could be changed to zoom in on that signal. Now a very important thing happens. As the bandwidth is reduce by zooming in, the noise level is reduced. This is because only the noise within the new bandwidth is seen. This is always less the 100% of the noise. The closer you zoom in the lower the noise. If you zoom into a bandwidth of 50 Hz, over 98% of the noise will be reduced..

A signal, which was previously below the noise, now could be much larger than the noise, because the signal strength does not change when the bandwidth is reduced. This is why BPSK-31 is so good. The spectrum analyzer is not a good tool for finding signals below the noise floor.
The next piece of test equipment is the waterfall. The horizontal scale is the full frequency scale of the spectrum analyzer. The vertical scale shows a straight line. The intensity at each point of the straight line is the average peak amplitude at that frequency. The peak amplitude is the signal plus the noise, which is always larger the noise even if the signal is less then the noise.

After the averaging time of about one second, the line is frozen and lowered down vertically. Then a new line is made. This process is continually repeated. It looks like a waterfall as the lines move down the screen. When all the lines are looked at together, the peak intensities form vertical lines. This is a good piece of test equipment to see signals that are below the noise floor. Again the computer mouse could move the cursor to a vertical line on the waterfall. The courser is actually the center of a narrow band digital software filter that the PSK program uses to pass the PSK signal and reject all other noise and signals. The cursor is also the frequency of the audio sin wave made by the computer generated numbers.
The last piece of test equipment is a vector scope (phase scope) It shows the phase of the detected signal. In order to measure the phase of two signals, they both must be on the same frequency. The PSK program has a built in Automatic Frequency Control (AFC) that puts the phase detector at the same audio frequency as the received signal. This test equipment allows us to see if the signal is BPSK or QPSK.

The phase detector creates a zero when the phases are equal and a one when the phases are opposite. These ones and zeros are used by the PSK program to find the Varicode, which is used to display the received message characters on the monitor. It is important for the computer to check the phase in the middle of the digit not during the transition. This is done with a second AFC at 31.25 Hz that uses a zero crossing detector and a delay.
When a new transmission starts, a series of ones and zeros are transmitted to synchronize the 31.25 Hz receive clock and release the receiver squelch. The series of ones and zeros are used because they have the maximum amount of zero crossings. At the end of all transmissions a special code is sent the receiver to set squelch, which stops the displaying random characters made by noise. When the transmitter buffer is empty the series of ones and zeros are sent until the next key is pushed and a Varicode is put into the transmit buffer.

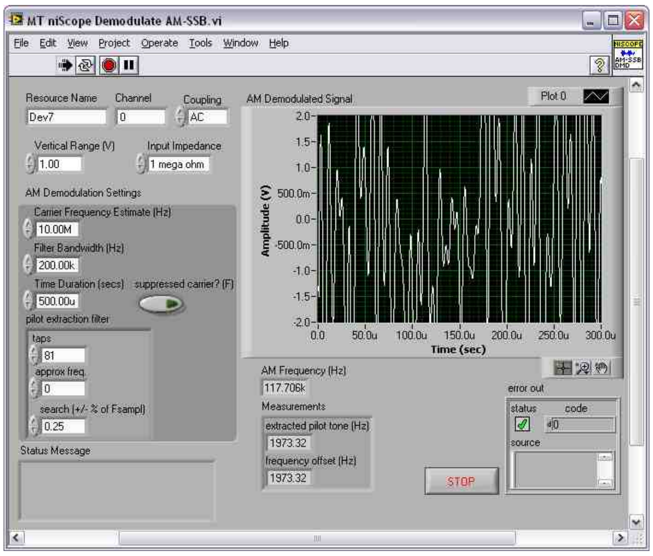
QPSK-31 is the same as BPSK-31, but there are four phases that are 90° apart. The phase 0° could be used for Varicode digits 00. The phase 90° could be used for Varicode digits 01. The phase 180° could be used for Varicode digits 11. The phase 270° could be used for Varicode digits 10. This would double the transfer rate, but this isn't what is done. Instead and error correcting algorithm is used to fix errors in the received code. I won't try to explain how it works, but the net result is that when a digit is detected, its value is checked against the previous four values and a final decision is made, if this digit is a one or a zero. The result is that one out of five digits could be wrong and the received code will fix itself to 100% copy and the transfer rate is the original 31.25 Hz.

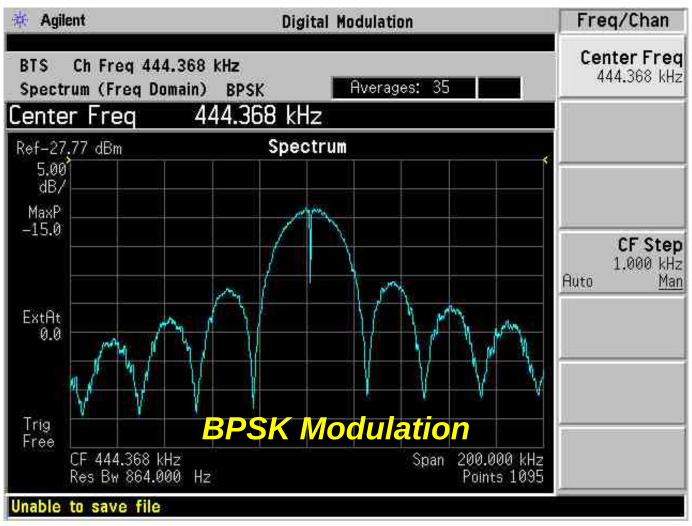
To get from QPSK-31 to QPSK-125 the 8,000 Hz oscillator in the sound card is divided by 64 to get 125 Hz and the received software bandpass filter is increased from 50 Hz to 150 Hz.

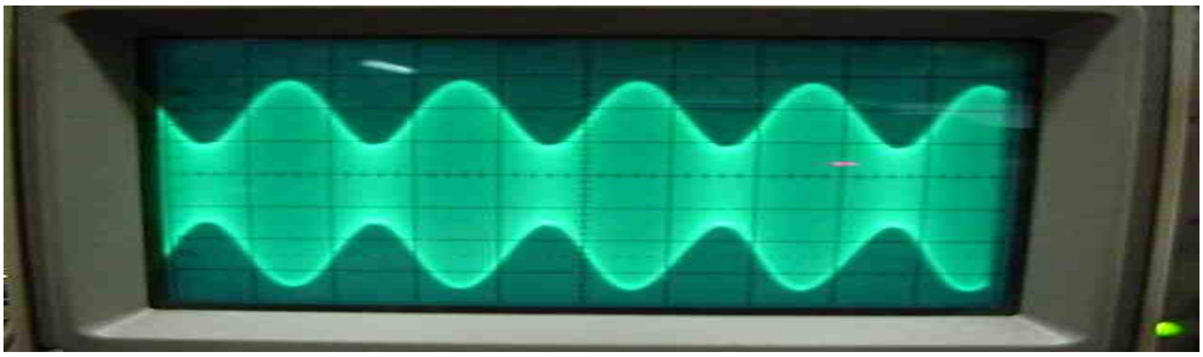
Voice has at least four properties, volume, timing, speech and pitch. Our radios have processors that try to keep a constant volume so my project doesn't transmit the volume. Timing is very important and needs to be transmitted. We are constantly changing the speed at which we talk. Timing is not accounted is speech recognition software. Pitch is not need to be transmitted to understand speech, but is is essential to recognize the voice of the operator. A complete branch of applied science is devoted to speech. Unfortunately they don't have their act together in that some of them say there are 44 fundamental blocks of speech called phonemes and others say there are only 40 phonemes. I know that the most important phoneme is no-sound. The word "at" could not be said without a no-sound between the "a" and the "t".

It is obvious that what is needed to be done is to assign a Varicode number to each phoneme. The shortest phoneme should be given the Varicode number with the shortest digits. For example the Varicode 100 is given to no-sound. There are only 45 of the 126 Varicode numbers used and maximum length is ten digits. The worse case is 15 phonemes per second, but because most phonemes have a small number of digits, the average worse case is about 20 phonemes per second. I am not sure how many phonemes per second we use in are speech, but I think that should be good enough for slow talking.

73's ! Mike N6IEF







THE DX NEWS

From the Web (tnx opdx, 425 dx news, arrl...)

9A800, CROATIA

The Radio Club "Varazdin" (9A1HDE/9A7A) will be active as 9A800VZ until the end of the year. First activity will be during the CQWW WPX SSB Contest (March 28-29th). Activity is to celebrate the 800th anniversary that King Andrija II gave to the people of Varazdin the status as a free and royal town. More details are available on QRZ.com. QSL via 9A7A.

C9, MOZAMBIQUE

Members of the Texas DX Society will be active as C91TX from Bilene, Maputo Province, between March 25th and April 5th. Bilene is 180km from the capital city of Maputo. Activity will be on all bands 160-10 meters with 3 active stations, one station each for CW, SSB, and the Digital modes. They will participate in the CQWW WPX SSB Contest (March 28-29th). Operators mentioned are: Cal/WF5W, Paul/W5PF, Madison/W5MJ, Bill/K5WAF, Dale/KG5U and Jim/N4AL. QSL Manager for C91TX is W5PF. Depending on internet access, they hope to have an online log search page once operations start, but don't expect daily updates. Texas DX Society C91TX Web page at: http://www.tdxs.net/c91.html

FJ, ST. BARTHELEMY (NA-146)

Oscar/EB1HF and Pablo/EC1DPM will be active as FJ/homecall between April 7-12th. Activity will be only on the digital modes, principally on RTTY, on the 20/15/10 meter bands. They will have two Icom 706s into dipoles and a 2 element yagi (by Super Antenna). Their QSL Manager is EC1AE. QSL only direct, (w/SAE + 2 USDs) or (w/SAE + 1 IRC) VALID CN01 IRC, to: Oscar Gancedo, P.O. Box 921, E-33080, Oviedo, SPAIN.

JW, SVALBARD

Just a reminder that Francois, F8DVD, will once again be active as JW/F8DVD from the Longyearbyen ARC located on Spitsbergen Island, 78 degree North, and about a thousand kilometers far from the geographic North Pole (EU-026, WAZ 40 and Grid Square JQ78TF), April 19-25th. Activity will be on all HF bands both CW and SSB. This will be Francois's 6th time on the island. He will use the Longyearbyen Radio Club's equipment which consists of an ICOM 751 and a 500w. into a 5 el. yagi on a 30 meters high tower. QSL via his home callsign, through the French REF-Union Bureau or direct, w/SAE + 1 IRC (or 1 USD for EU, and 2 USDs for elsewhere) to: Francois Bergez, 6, Rue de la Liberte, F-71000 MACON, France.

OH0, ALAND ISLAND (EU-002)

Anne, OH2YL, will be active as OH0/OH2YL between April 9-14th. Activity will be on the HF bands using CW and SSB. QSL via OH2YL.

P2, PAPUA NEW GUINEA (IOTA Op)

Luis/ CT1AGF, Derek/G3KHZ, Steve/G4EDG and Hans/SM6CVX, plus Gordon/G3USR, will activate three Papua New Guinea IOTA islands between October 22nd and November 9th. Islands mentioned are Tanga Island (OC-102), Green Island (OC-231) and Woodlark Island (OC-205) . Activity will be on 160- 15 m. using CW, SSB and RTTY modes. Gordon/G3USR will be exclusively on SSB. The team will be using new single band vertical dipoles for 30-15 meters and a ground plane for 40 meters.

PJ2, NETHERLANDS ANTILLES (Contest Station Available)

Geoff, W0CG (PJ2DX), resident caretaker of PJ2T, announced that nobody in the PJ2T club (CCC - Caribbean Contesting Consortium) is interested or able to operate the Curacao station for the IARU HF World Championship in July. Thus, the place is available if you would be interested in coming there as a Single- Op, or to organize a group operation. With sufficient lead time, Geoff could arrange for you to get a callsign of the form PJ2* (except that PJ2A, PJ2R, and PJ2T are unavailable). If you wanted to do a group operation that would include Geoff for a little bit of the operating time, you would be able to sign PJ2T. Two by two callsigns are NOT allowed for visitors to Curacao, sorry. The QTH has two bedrooms, two baths, and four beds available. (Geoff will sleep in another room attached to the house.) There's tons of info about the place at: http://www.pj2t.org E-mail at: ghoward@kent.edu

SPECIAL EVENT

Look for special event station R150A to be active through the end of April. Activity is to celebrate the 150th anniversary of the birth of A.S.Popov. Operations will be from the Central Museum of Communications in St.Petersburg. Activity over the past week has been on 40/20 meters CW and SSB. QSL via RK1A.

SPECIAL EVENT (Dayton HamVention)

Look for special event station W1AW/8 to be on the air during the Dayton HamVention (May 15-17th). They are currently looking for volunteers to operate the station for 4 hour shifts. Volunteers will receive an on-grounds parking pass, an admission ticket and a hat. Sign up in advance is STRONGLY recommended -- first come first served on requested shifts. This WILL fill up quickly. Contact via E-mail Jerry Bodey, N8OWV at: jerrybodey@sbcglobal.net

VK9L, LORD HOWE ISLAND (Update)

As this was being typed, there was some VK9LA activity on the air. The last of the DXpedition team should arrive on the island by March 24th (total of 16 operators). The operation with 7 full HF stations from 2 separate sites, as well as one 6m station will be on the take air until April 3rd. Modes of operation are SSB, CW and RTTY. For suggested frequencies, see OPDX.899. The QSLing chores will be handled by VK4FW. An online QSL Requesting Service is now available at:
http://www.odxg.org/qsl.htm
Any QSL CARDS managed by ODXG inc., or Bill Horner, VK4FW, can be sent to you utilizing this system. This is a FREE service if you request your cards to be sent via the BUREAU. A minimum payment of 5 USDs via PAYPAL is required if you request your cards to be mailed DIRECT TO YOU. QSL cards will be printed by UX5UO. For more updates, details on the DXpedition and how to donate to this operation, please visit the ODXG and VK9LA Web pages at:
http://www.odxg.org
http://www.odxg.org/vk9la.htm
ADDED NOTEL: Lord Howe operators Stan/SQ8X and Pete/SQ9DIE have added a daily blog daily on Twitter.com. Look for regular updates on how the DXpedition is progressing at:
http://twitter.com/vk9la

ZD8, ASCENSION ISLAND

Karol, G0UNU, will be active as ZD8KR between May 4-11th. Activity will be mainly on 20 meters CW and SSB. More details will be forthcoming closer to these dates (on (QRZ.com)). QSL via G0UNU, by the Bureau or direct.

THE DX CALENDAR
BY SM3CVM, Lars
http://www.sk3bg.se/

APRIL 2009

- 31/3 CHAGOS ISLANDS; VQ98JC and VQ9JC AF-06
from Diego Garcia by ND9M. He is active as VQ98JC through the end of the year, after that he will use VQ9JC. Expect him to be QRV in his spare time, typically on 12-16.30 UTC (from Sunday to Thursday) and on 12-17.30 UTC (Fridays and Saturdays). Occasional overnight operating will allow him to remain QRV until 1 UTC. QSL via ND9M.

- 31/3 ANTARCTICA; VKØBP
from the Davis Basis (main island only) by VK2ABP/VK2MRP. QSL via VK2CA (see QRZ.com)

- 31/3 CANADA; VC3Y
A special callsign of the "York Region Amateur Radio Club" activated during the 50th anniversary of the club. VE3SST, VE3ZF, VE3VO and VE3IRT are going to activate the lighthouse on Bluffers Park Island (CISA new) with this call on Mar 21/22. QSL cards via bureau or direct to VE3CWO.

- 1/4 DOMINICA; J79BXI NA-101
by SM0XBI. He will operate SSB only. QSL via home call, bureau preferred. QSL via bureau to homecall.

- 1/4 BAHAMAS; C6ANM NA-001
on 160m-6m. He prefers QSL via LoTW or direct to WA2IYO.

- 3/4 HAITI; HH4/K4QD and HH4/AF4Z NA-096
They will work in CW/SSB/RTTY on 160m-10m and also on the lowbands if there is power available during the night hours. They are planning to take part in the CQWW WPX SSB Contest (Mar 28/29) from the clubstation HH2JR using a special 4V4 callsign. QSLs via homecalls.

- 4/4 SOUTH COOK ISLANDS; E51COF OC-013
from Rarotonga by NL8F. He plans to operate SSB on 10, 15, 20, 40, 80 and maybe 17 metres. QSl via K8NA.

- 11/4 GUADELOUPE; FG/F4EBT NA-102
from Basse-Terre. He plans to operate holiday style on, SSB on 80/40/20/17/15/12/10 meters.

- 28/4 SOLOMON ISLANDS; H44MS
by DL2GAC. He plans to operate SSB with a focus on 80 and 40 metres. QSL via home call, direct or bureau. DK9FN (CW operator) and DL2NUD (EME operator) will join DL2GAC (H44MS) in late February for a 2-3 week operation from Temotu Province. DK9FN will go back home on 16 March. He plans to operate CW only on 160-6 metres, hopefully as H4ØFN (the callsign he used back in 1999). Whatever callsign he will be using, the QSL route is via HA8FW (bureau preferred).

Low Power Space Station Contact

AMSAT-UK

By Trevor, M5AKA

Callum Graham MM3YCG recently had a contact with astronaut Mike Fincke
who was onboard the International Space Station (ISS). It is believed that Callum is the first UK Foundation license holder to have a contact with an Astronaut onboard the Space Station. The UK Foundation licence limits users to just 10 watts output in the Amateur bands from 1.8 to 440 MHz. (1 watt at 135 kHz and 10 GHz).

Callum was using 10 watts from a Yaesu FT-847 feeding a Sharman Multicom X-50 collinear at 30 feet. He got his license in July 2007, thanks to the Mid Lanark Amateur Radio Society. Since then he has had many contacts through the Amateur Radio Satellites and shown what can be done using low power. Callum has posted a video of his ISS contact on YouTube, see link below.

Mike Fincke holds the Amateur Radio callsign KE5AIT but for this contact he was using the ISS callsign of NA1SS. Mike was running the ISS Kenwood D-700 set up for cross band working on 437.8 and 145.8 MHz and putting out typically 5 watts output on 2 metres. The Kenwood doesn't run its full rated power because convection cooling doesn't work in a zero gravity environment.
Most of the astronauts on the ISS are licensed Radio Amateurs. If you've not heard the ISS before try listening on 145.800 MHz FM, you'll find rigs with the wider filters for 25 kHz channel spacing work best.

The amateur equipment on the ISS can operate in many different modes on 145.800 MHz such as cross band FM repeater, AX.25 Packet or Slow Scan TV. Websites such as ARISS or the ISS Fan Club are good sources of information regarding which mode is in use when you listen. You can find out when the ISS is in range by using the N2YO Real Time Satellite Tracking website.

URL's
You can see a video of the MM3YCG –NA1SS contact on YouTube at
http://www.youtube.com/watch?v=aqUR3g5C9Jw
Mid Lanark Amateur Radio Society: http://mlars.org.uk/
N2YO Real Time Satellite Tracking: http://www.n2yo.com/
ISS Fan Club: http://www.issfanclub.com/
ISS Repeater Tips:
http://www.southgatearc.org/news/february2008/iss_repeater_tips.htm
Amateur Radio on the International Space Station (ARISS): http://www.ariss.org/
AMSAT-UK website: http://www.uk.amsat.org/

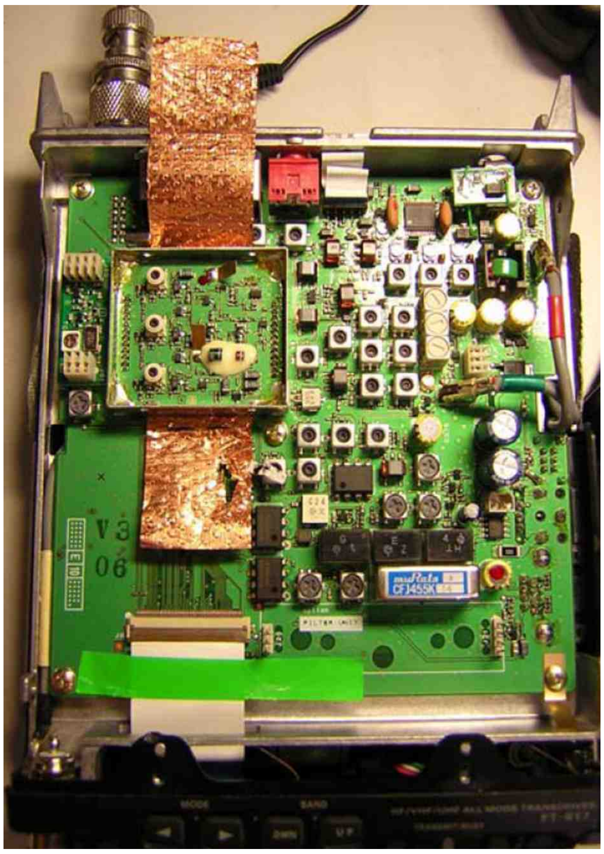
Easy cure for microphonic VHF VCO in Yaesu FT-817
By Erik Finskas OH2LAK

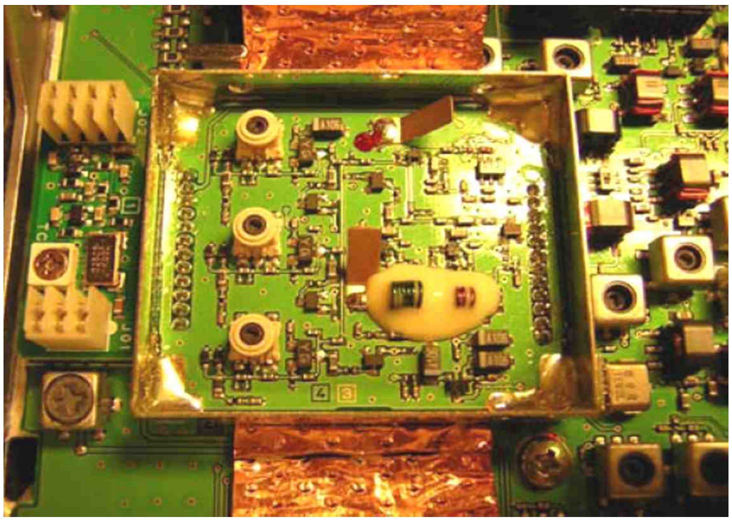
Some manufacturing series of the Yaesu FT-817 has problem with a microphonic VCO, at least on 2m band, when using radio's internal speaker. The PLL tin box including the VCO components is too close to the internal speaker, and with high volume or suitable audio signal the PLL box starts to resonate with the speaker resonation. This effect disturbs highly CW operations as it garbles the sidetone, or just creates howling sound from the speaker when the volume level is increased enough.

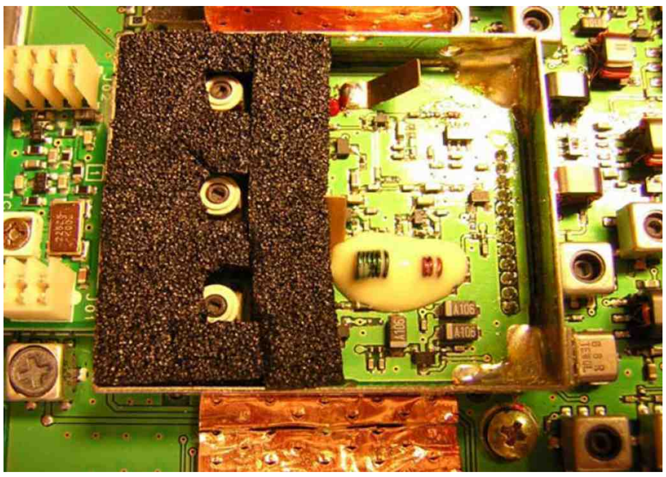
Fix is to insert muffle in to the PLL box. I've used antistatic foam mat used for IC's, and it works well.

Open the top side of the FT-817 by removing screws on the side and top and back. When the hood is loose, carfully lift it and disconnect the speaker cable when reachable.

The PLL box has copper tape around it, peel the tape open carefully to be able to re-use it, then open the PLL box. It might be soldered from some of its corners, remove the solderings carefully.







Inside the PLL box there are three coils, which cause the problem. Cut a suitable piece of muffle material, eg. the IC mat and place it around the coils as seen in the picture. Close the lid of the box, solder it from all four corners to improve grounding and restore the copper tape on it. Reassemble the top cover and you're done!

73's ! OH2LAK

HMY-2K8 - A multiband HF transceiver :
By Dr. R. RAJASEKHAR, VU2HMY

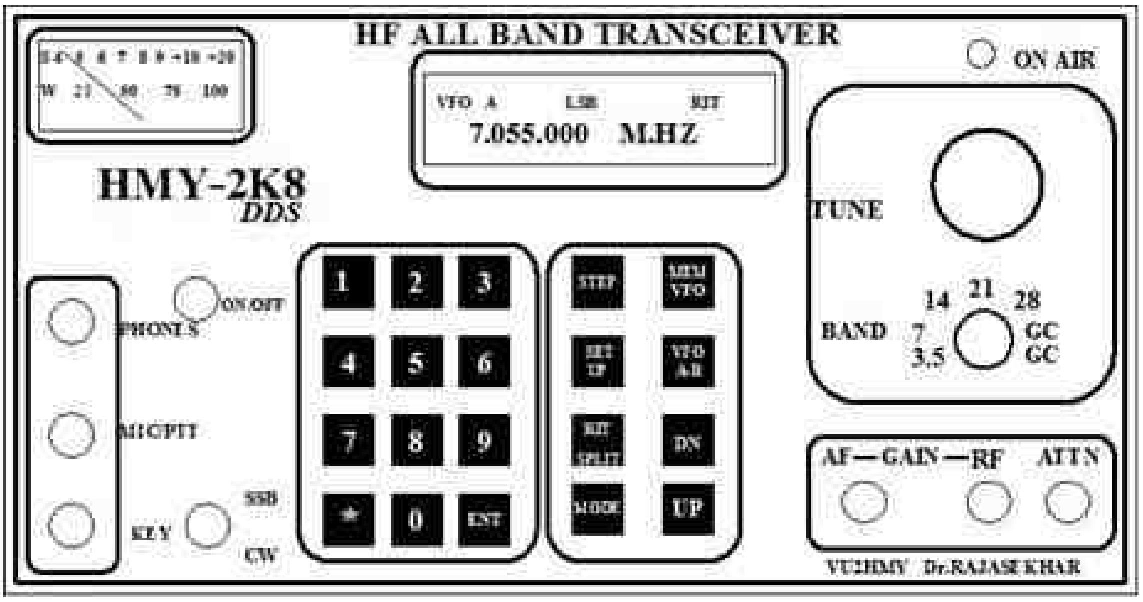


Fig1 : The front panel layout used for the multi band transceiver (8.5 X 4.5 inches).

TECHNICAL SPECIFICATIONS :
Frequency coverage: Ham Bands
3.500 - 3.600 M.Hz (80 M)
7.000 - 7.100M.Hz (40 M)
14.000 – 14.350 M.Hz (20 M)
21.000 - 21.350 M.Hz (15 M)
28.000 – 28.350 M.Hz (10 M)

Frequency control :
Ver. 1: Direct Digital Synthesizer (DDS) with 1 Hz step continuously variable VFO with 20 memory channels. 2 line LCD display. Dual VFO, Split , RIT, Key pad /rotary encoder for frequency entry.

Ver. 2: 5 Band Heterodyne VFO with PIC frequency counter to reduce cost of the project.

Receiver :
Single conversion receiver with 10.000 M.Hz Cohn filter.
Low noise figure, 2.2 K.Hz SSB band width, 1.5 W audio O/P.

Transmitter :
RF O/P - SSB - 90 W (DC PWR I/P)
CW - 60 W (DC PWR I/P)
Built in CW side tone, CW delay, pwr meter, MOS FET Push Pull PA.

Dimensions : 8.5 X 7 X 4.5 inches

Genesis :

Being a home brewer I was hesitated to buy a commercial HF transceiver soon after getting my ticket in1988. Even my guru, late T.K.Seshandam VU2WC was not allowed me to do so. But he was kind enough to give me a set of pcb's of VWN QRP TX . I was successful to come on air with VWN QRP effectively along with L board RX on 40M. Those days are really good for HF communication. With low band noise, excellent propagation conditions we could work hours together daily on 40 M with 7 W AM signals. After that I have homebrewed RM96 and ATS1 and came on air effectively with SSB signals on 40 M. Even though there is a feeling of missing a lot of activity on other HF bands Viz 20 M, because all the above rigs are mono banders. If I want come on other band I have to construct another rig of same circuit! Then I could get a used ICOM - IC720 commercial TRX. It worked well for about 6 months and gone QRT. I just sent it to few service centers and was not able to get it repaired due to the non availability of spares for PLL & Logic boards. Then I secured one BEL524 in dead condition and able to repair it and came on air on all HAM bands with an home made out board DDS VFO. But servicing such surplus equipment is not that much easy due to their concealed and modular construction. Then I was thinking of home brewing a multi-band SSB/CW TRX using indigenous components freely available in VU land. I searched for the circuit schematics even on the internet. But ….in vain! Then I could download and made simple 40 M band SSB TRX circuits.

Using bilateral switching technique using switching IC 74HS4053 by KD1JV using NE602 and PY2OHH using TA7358. Both performed well and gave me good results on air. But the cost of NE602/SA612 is around Rs.300/- ++ and its rare availability in VU land, whereas the cost of TA7358 is Rs.15/- and freely available.

Finally, I was decided to design a multi band SSB/CW TRX using Toshiba IC TA7358 with band switching using diode switching arrangement which can give 90 W DC PWR input from 80 to 10 Mtr bands. I have incorporated CW delay and side tone for easy CW operation. I have designed pcb (measures 7 X 3.5 inches) lay out using EXP PCB design soft ware and printed the board which accommodates all stages Viz Bal. Modulator/ demodulator, RX/TX mixer, 5 bandpass filter, 6 pole Cohn filter, Tone circuit for CW, CW side tone, mike amp, audio amplifier, transmitter driver and TRX change over. I have used home made DDS VFO using up-conversion to cover 80 – 10 M bands. In other version a pre mixed heterodyne VFO has used to minimize the cost of the rig. I hope this tiny rig shall meet all the demands of an average VU ham with all its sophistication at reasonable cost.

The circuit :

The circuit is simple and strait forward. IC1 TA7358 is used as RX/TX mixer, IC2 TA7358 is used as Product detector / Bal. modulator and IC3 as a bi-lateral switch to switch both the ICs, IC1 and IC2 to particular inputs / outputs in transceiver operation by applying DC voltage in Key down / press of PTT condition.

Receiving chain :

The incoming RF signal is amplified in broad band RF amplifier and fed to the band pass filter by applying appropriate switching voltage from the band switch from front panel. The strong local signals can be attenuated by VR1 from front panel. Here the signal is filtered in the band pass filter and fed to IC1 TA7358, RX mixer through the switching IC, (IC3 sec A).

Here the incoming signal is mixed with local oscillator signal (VFO) and converted to 10 MHz IF frequency and fed to the SSB filter (6 pole 10 MHz Cohn filter) through switching IC (IC3 sec B) and fed to IC2 Product detector through switching IC (IC3 sec C). In the product detector the signal is beat with Carrier oscillator signal to get resultant audio signal which is further amplified by transistors Q2 and Q3 and sent to audio amplifier IC4 LM380 to deliver sufficient audio from speaker. The AF gain can be adjusted by the potentiometer VR3 (VOL) from front panel.

Transmitting chain :

Voice received by the condenser microphone is amplified by transistor Q4 and sent to IC2 Bal. modulator. Here carrier frequency is modulated and 10 MHz DSB signal is produced and further amplified by Q2 and fed to the SSB filter through switching IC (IC3sec C) to eliminate unwanted side band. The mike gain can be set by the preset VR2 and appropriate side band X-Tal is selected by the mode switch SW 2 from the front panel. The 10 MHz SSB signal from filter is fed to the TX mixer IC1 through switching IC (IC3 sec B) and mixed with Local oscillator signal (VFO) to get required transmitted frequency. The signal is further amplified by 2 stage broad band RF amplifier and sent to band pass filter through switching IC (IC3 sec A).

Appropriate band pass filter is selected by applying +12V to the switching diodes from the 5 way band switch SW3 from front panel or from logic out put from DDS VFO in case of using DDS VFO. Few milli volts of RF of TX signal from band pass filter is further amplified by 3stage broad band HF driver amplifier for about 1 – 1.5 W. The driver amplifier is having good linearity through 80 – 10 mts and the gain of the amplifier can be adjusted by changing the value of damping resistors R53, R54, and R57 to get adequate drive level to the final amplifier. Initially one can come on air with this 1W power and can work few stations to get reports and to align the transceiver.

VFO :

I have used DDS VFO using AD9851 along with PIC16F628 in up conversion mode to cover 80 to 10 Mtrs bands in one set and in another set a pre mixed heterodyne VFO is used to reduce the cost of the transceiver. If one wish to operate on single band, can use simple colpits oscillator such as RM96 VFO which is very stable in operation. I don’t want to describe more about VFO, because one can choose his VFO according to his taste and requirement. There are varieties of VFO circuits available in hand books or on the internet.

CW operation :

Sine wave tone around 900 K.C from an oscillator consisting of Q13 and Q14 is fed to mike amplifier through SW1, SSB/CW switch in key down condition. At the same time the side tone is amplified by IC 6 LM386 and heard in speaker LS 2. CW delay circuit provides sufficient delay for proper CW operation. The delay time can be adjusted by the preset VR7 in the base circuit of Q 12.

SSB filter :

Six x-tal Cohn filters is used for selective band width of around 2.5 K.C. Select all the six x-tals with in 100 Hz tolerance to each other to achieve proper band width and audio quality. Select carrier oscillator x-tals with + and - 1.5K.C of filter frequency for LSB and USB operation.

Final MOS FET push pull broad band amplifier :

Circuit of popular IRF 510 push pull amplifier is used for final RF amplifier which is capable of delivering of 90 W DC power input over the frequencies between 3.5 to 30 M.Hz with 1 – 1.5 W of drive. Double side glass epoxy PCB has to be used for proper operation and to achieve stability especially at higher frequencies. A large heat sink (6 X 3. 5 inches) with fins should be used. The amplifier draws 3 – 3.5 A for the maximum voice peak with 25V of operation. The O/P of the PA is connected to the SO239 antenna socket through change over relay contacts with a peace of 50 ohms thin coax. The O/P of the PA is sampled and the PWR level is indicated by VU meter mounted on front panel.

Construction :

Soon after finishing soldering all the components, check for shorts and solder bridges between the tracks. I have selected a cabinet of FLD (front loaded) tape deck available from electronic shops which measures 8.5 X 7 X 4.5 inches (Almost the size of commercial TRX). On front panel volume controller, Tuning knob, Attenuator controller, mode switch, band switch, key pad, on/off switch, jack sockets for PTT/MIC, Phones, Key and VU meter are fixed. The main board is fixed on the chassis and on the back panel the PA, SO239 antenna socket, relay and two fuse holders are fixed. Inter connections are done with multi strand hookup wire. 25 V line to PA has to be wired with thick wire used for car wiring capable of carrying 5A. All audio connections should be made with 1+2 shield cable and the RF interconnections are done with thin coax cables RG174 and RG58C/U. Shields with thin ms sheet should be provided for SSB filter, VFO and to the PA. PTT switch and microphone are housed in a small plastic box such as cell phone charger case. Condenser microphone is wrapped with few layers of soft cloth or sponge to avoid unwanted low frequencies entering into mike such as breath. 1 + 2 thick shield cable which is used for public address system should be used for microphone.

Alignment :

To align the receiver, the RF signal from signal generator or the incoming signal from antenna should be used. Both the coils for each band in the band pass filter are peaked by Teflon alignment tool for maximum signal strength. Then adjust the carrier frequencies of LSB and USB X-Tals for exact beat note. This completes the receiver alignment. To align transmitter, connect a 6 V low current bulb to the O/P of driver amplifier and apply power to TX line by pressing PTT or in key down condition. As you shout into the microphone or depressing the key, you will observe glow in the bulb. By observing brilliance of the bulb set the mike gain preset VR2 to adequate level. If necessary alter the values of damping resistors R53, R54, and R57 in driver amplifier to get adequate drive. Keep this level 20 % low in CW mode to protect the finals from thermal run away by adjusting CW level preset VR8. Now disconnect the bulb and connect the driver amplifier O/P to relay, with this one can work (QRPP) few stations and get reports initially.

Connect a milli ammeter in series with PA and apply 25 V than adjust idling current of MOSFETS to draw 40 mA (20 mA each) by adjusting potentiometers VR4 and VR5 . Then disconnect milli ammeter and connect 5A FSD ammeter and connect the driver amplifier O/P to the I/P of the PA. Now connect 50 ohms / 100W dummy load or an external antenna to the rig. By shouting into the microphone, check the current drawn by the PA. It should be 3 – 3.5 A at 25V of drain supply. If not, re-adjust the mike gain preset VR2. Too much mike gain leads distortion of transmitted signal and shift in operating frequency.

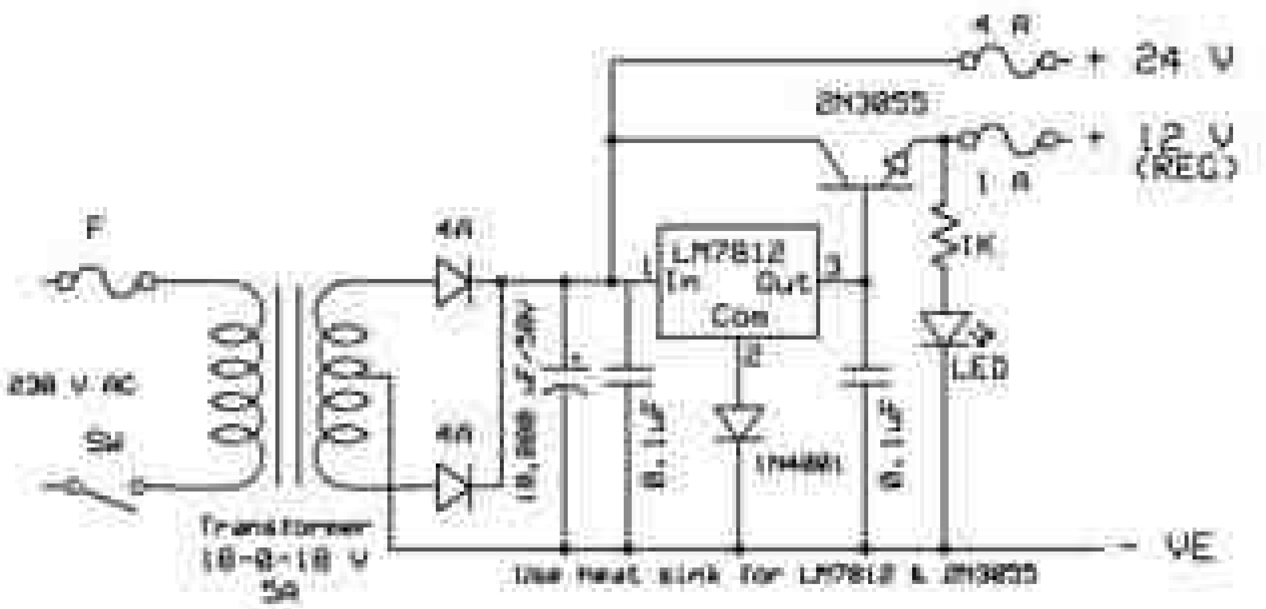
Things to remember :

1. All the resistors should be of 1 % tolerance and capacitor C6, C8, C11, C13, C16, C18, C21, C23, C26, C28, C56, C57 are styroflex to obtain stability and to avoid drift in operation. Use only 74HC4053 high speed switching IC to achieve good results especially in higher bands.
2. SSB filter should be shielded with a metal case and grounded.
3. Use 1 + 2 shield wire for audio and RG174 & RG58C/U for RF inter connections.
4. Good quality heat sinks should be used for Q8, Q9, IC5 and a large (6 X 3.5 inches) heat sink for FETs. Use heat sink compound applied both the sides of mica washers.
5. Shield VFO and the PA with small boxes made with soft ms sheet.
6. Beware of static damage while handling FETs, microcontrollers etc. Ground your soldering equipment and unplug from mains while soldering such devices.
7. Use 50 ohms/100W dummy load and carry your initial testing of your rig. Don’t shout haaaalo--- haaaalo---- halo on air and create QRM to others. Identify your self on band to get critical reports and certainly these reports shall help you to improve the performance of the rig.

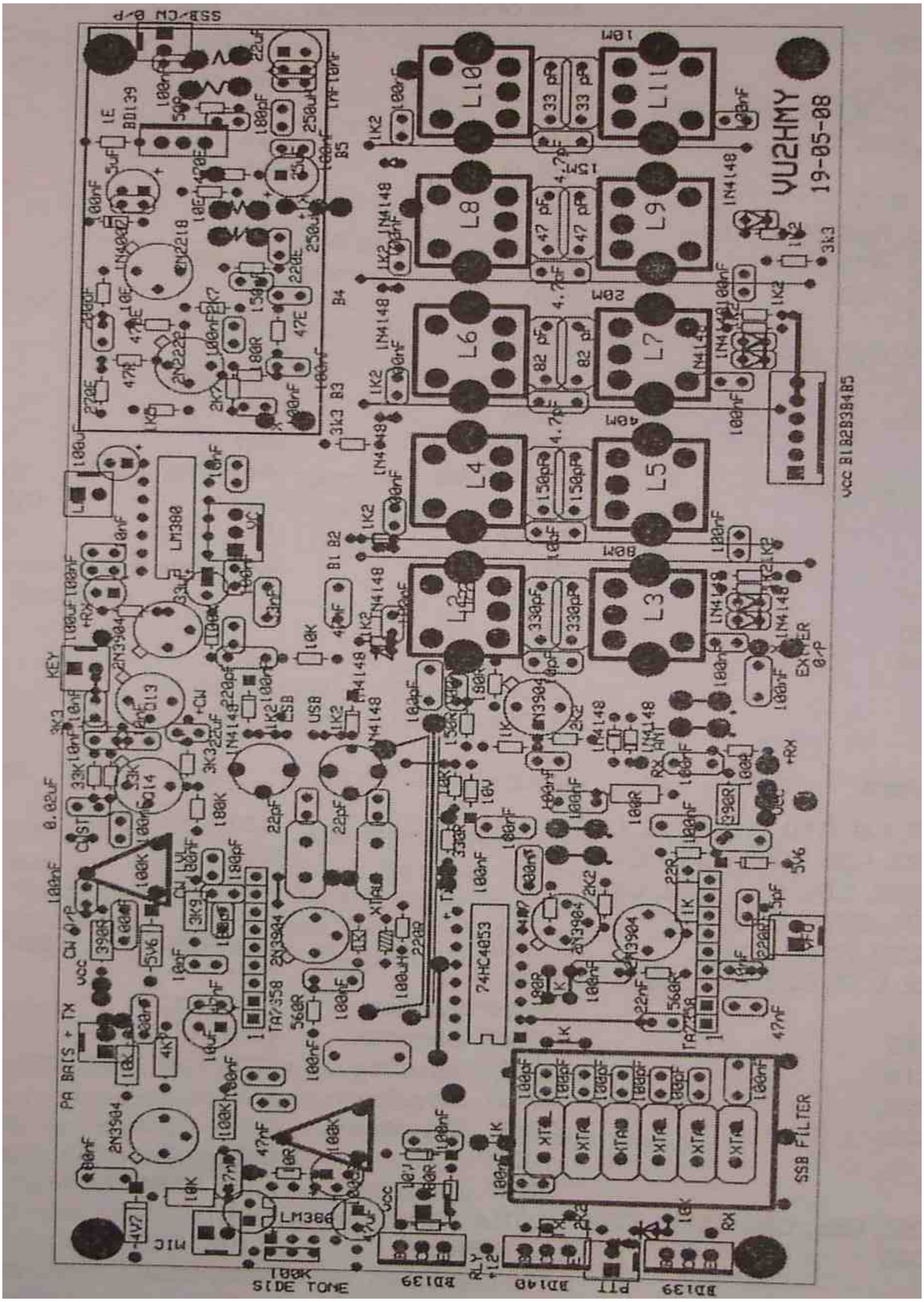
See the component Layout on the next page

Power supply :

The exciter requires 12 V / 1 A depends upon the type of VFO / Display and PA needs 24 V / 4 Amp to get maximum RF O/P. I have got 35 W DC PWR I/P even with 12V car battery and had several contacts. The following circuit diagram can be used for PSU. Use rated fuses and good heat sinks for LM7812 and 2N3055.



Power supply unit for HMY2K8 - Multi band transceiver


VU2HMY
19-05-08
SSB FILTER
LM380
74HC4053
SIDE TONE
SSB/CW O/P
10M
15M
20M
40M
80M
MIC
KEY
PTT

List of components:

Resistors:

R1, R38	2K2	R52	10R
R2, R22, R23, R35, R39, R61	1K	R53	220R* Adj
R3, R40, R43	180 R	R54	10R* Adj
R4	150 R	R55	470R
R5, R42, R70	100 R	R56	10R ½ W
R6 – R15, R26, R27, R46	1K5	R57	220R* Adj
R16, R17, R65, R69	3 K3	R58, R59	33R
R18, R21, R30	10K	R60	47K
R19, R31	100K	R62, R63	1K5
R20, R28, R32, R64	4K7	R66	180K
R24	3K9	R67, R68	33K
R25, R34	390R	R72	15K
R29	220R	VR1	1K Lin Pot
R33	220R * Adj	VR2, VR6 – VR9	100K Preset
R36	22R	VR3	100K Log Pot
R37	560R	VR4, VR5	4K7 Preset
R41	4R7		
R44	330R		
R45, R50	2K7		
R47, R49	47R		
R48	270R		
R51	470R		

Note: All are ¼ W Unless otherwise specified

Capacitors:

C1 – C5, C9, C10, C14, C15, C19, C20, C24, C25, C29, C35 – C41, C47 – C52, C58, C59, C62, C64, C66, C71, C73 – C75, C77, C82 – C88, C100, C104 -	0.1 uF
C7, C12, C17, C22, C27, C92 -	10Pf
C6,C8	330pF styroflex
C11, C13	150pF Styroflex
C16, C18	82pF Styroflex
C21, C23	47 pF Styroflex
C26, C28	33pF Styroflex
C30	100Pf
C31	22nF
C33, C55, C65, C93, C96 – C98	0.01uF
C67, C80	0.01uF/63V
C68	100 uF/25V
C69	33 uF /25V
C70	100 uF

C32, C54, C60	0.047uF
C34	10pF
C42 – C46	82pF
C53	10uF
C56, C57	- 180PF Styroflex
C61	0.033uF
C63, C72	220pF
C76	4.7uF
C78, C81, C94, C95	22uF
C79	0.001uF
C89	100 uF/63 V
C90	0.01uF /63V
C91, C101	1uF/63V
C99	0.02 uF
C102, C103	47 uF
VC1, VC2	22 pF Philips Trimmers

Diodes:

D1 – D12, D16, D17, D22, D23	1N4148
D20, D21, D24, D25	1N4001
D13, D14	6.1 V / ½ W Zenar
D18, D19	9.1 V / ½ WZenar

Transistors:

Q1	BF494	Q9	2SC1162 (or) BD139
Q2, Q5, Q6	2N3904	Q10, Q11	IRF510
Q3, Q4, Q13, Q14	BC549	Q12	BD140
Q7	2N2222A	Q15	BD139
Q8	2N3866		

IC'S:

IC1, IC2	TA7358	IC5	7805 Regulator
IC2	74HC 4053	IC6	LM386
IC4	LM380		

X-TALS:

X1 – X6	10.0000 MHz	* See text
X8	10.0015 MHz	* See text
X9	9.99850 MHz	* See text

Switches:

S1 - DPDT, S2 - SPDT, S3 - 1 Pole 5 Way, S4 - SPST

Relay1 - 12V DPDT, Relay2 - 12V SPDT Mini (If DDS VFO is used)
Condenser mike – 1, LED's Green – 1, Red - 2, Speaker - 8 Ohms/ 2W - 1
Mini speaker / Piegeo element - 1, Fuse holders – 2, Fuse 1A -1, Fuse 4A – 1
6 pin RNC connector – 1, 5 mm Stereo Sockets – 2, 5mm mono Socket -1
VU meter – 1, Heat sinks for BD139, 2N3866, 7812 and 6 X 3.5 X 1 inches for PA.

Coil winding data:

L1, L13 – Bifilar, 10 turns on 10 mm torroid with 36 SWG
L12 – 50 turns on ferrite dumbbell with 45 SWG
L14 – Pri- 30 turns and Sec – 4 turns with 36 SWG on 10 mm torroid.
L15 - Pri – 6 and Sec 4 with 36 SWG.
L16, L17 – 25 turns on 10 mm torroid with 36 SWG.
L18 – Bifilar 10 turns on 12 mm torroid with 28 SWG
L19, L20 – 9 turns 6 mm dia
L21 – Bifilar 10 turns on 25mm torroid with 20 SWG
L22 – Pri – 2 turns and Sec – 3 turns on HF BALUN core formed by six 10 mm Torroid cores with 20 SWG Teflon hook up wire (used in submersible motors)

Coil winding details for Band pass filter : Slug tuned 10 mm former (Preferably with slots for split winding). 36 SWG or thinner enamel copper wire.

Turns

BAND	COIL	PRI	SEC	COIL	PRI	SEC
80 M	L2	3+2	8+9+9+8	L3	8+9+9+8	3+2
40M	L4	3+2	5+6+6+5	L5	5+6+6+5	3+2
20M	L6	3+2	3+5+5+3	L7	3+5+5+3	3+2
15M	L8	1+1	2+4+4+2	L9	2+4+4+2	1+1
10M	L10	1+1	2+3+3+2	L11	2+3+3+2	1+1

Acknowledgements :

I am thankful to VU2RVK and VU2WMJ for their great support especially monitoring my signal and giving reports. I thank VU2RM and VU2SV for their critical suggestions to improve the final performance of the rig especially on improving CW oscillator. I also thank our UV hams VU2NR, VU2VWN, VU2RM, VU2ATN, VU2IF, VU2EM, for their great contribution of designing and publishing successful homebrew rigs and kept many hams on air.

I thank VU2PAL, VU3ITI, VU2RJN, VU2AF and all net controllers who are controlling nets for year together and keeping hams active in spite of their busy schedules.

Guardian angel of my happy HAM home, my XYL Dr. Ashalatha and my harmonics Hannu and Minnu were more keen on my academic and research activities rather than domestic chores and kept me scintillating with vigour always and never allowed any situation that would dampen my enthusiasm for ham radio.

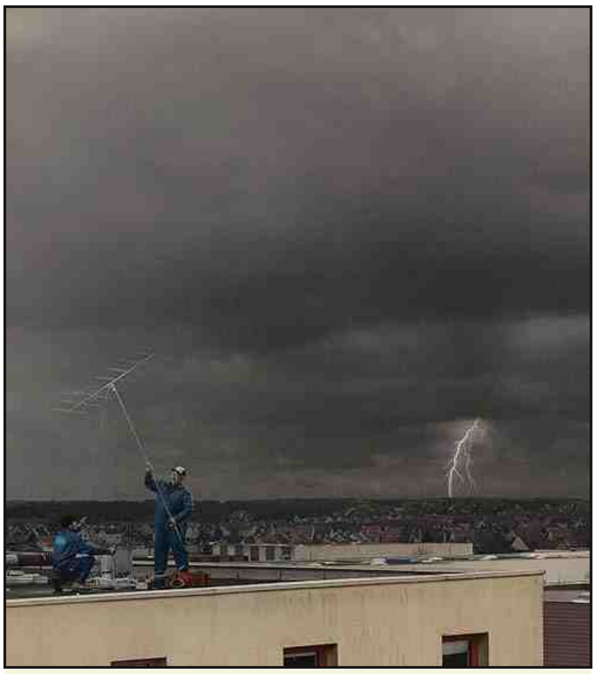
73. Happy home brewing.

QSP : http://vu2sgw.blogspot.com/

Courtesy :VU2HMY, OM Sekhar.(9704269922), vu2hmy@yahoo.co.in

COMIC'S HAM

Have some fun



Bravous Men !