

YAESU FT-1000MP Mark V

ESSB Modifications

Rev 1.0

DID YOU KNOW THAT THIS



CAN DO THIS



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About the Author's

Mike Schlamp (W5CUL) was first licensed in 2003 as AD5OX. Soon after receiving his license he applied for and received his grandfather's call sign through the FCC vanity system. Despite recently obtaining his amateur license, he has been around the hobby and other venues of communications for many years. His first introduction into the hobby was at the young age of 5 as a SWL'er, and later was tutored by his grandfather in receiver and transmitter theory. It wasn't until his induction into the armed services that Mike started working on communication equipment that did not exclusively utilize tubes. His grandfather was a little biased when it came to IC (rice box) technology. During his tenure in the active service, he repaired everything from AM/FM/SSB transceivers to microwave & satellite transceivers. The last four years, prior to transitioning out of the active service, found him deep into cryptograph cellular systems. This experience ultimately drove him into working for SWBT upon his departure from the service. He now works for AT&T Labs, formerly SBC Labs, Inc, in their Broadband Transport Group in Austin, Texas. He is married and has 3 boys, one Eagle Scout, and two runner-ups. His hobbies are Scouting, Amateur Radio, and Wood Working, in that order.

Jim Thomson (VE7RF) was first licensed in 1972 as VE7AJP at age 16. Prior to that he was an avid SWL'er between 1969-1972. In 1987 he obtained his first two letter call sign, VE7RQ. He obtained his current call sign, VE7RF, in the mid 90's. He spent some time working as a commercial radio op for the MOT in 1974. Later, after attending his local college, he worked for Canadian Telephones and Supplies, a subsidiary of the then British Columbia Telephone Co. as a Central office equipment installer. The last 27 years has been spent in Network Operations for both the former BC Tel Corp., and now the current Telus Corp, as a Network tech. During his 30 years of employment, he has seen a transition from electromechanical to electronic analog, to the current digital /fiber networks now widely in use today. Much of his amateur equipment consists of a combination of commercial radios, modified commercial products, and home designed and built linear amplifiers with mating HV supplies. He currently uses 2 x FT-1000-D's, and 2 x FT-1000-MP-MK-V's, all 4 of which are heavily modified. He also has six linear amplifiers. His interest in ESSB dates back only 5 years. Currently, 14 pieces of audio rack gear are employed. Other hobby's include jazz, and some gardening. His Spouse and two grown children support his efforts to enhance and promote amateur radio.

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Yaesu Mark V ESSB Modifications

1. ESSB and how it Pertains to the Mark V

ESSB, or Enhanced Single Side Band as some like to call it, has been around for many years. Never really gaining much popularity, but practiced by a close niche group of Amateur Radio Operators. The real intent of ESSB is to increase the fidelity of SSB communications by increasing both the receive and transmitted SSB bandwidth to something beyond that of the typical 2.4 – 2.6 KHz band-pass. In doing this, you achieve a full and much smoother sounding audio that is a pleasure to listen to, as well as transmit. There are many aspects of ESSB, however, the explanation of ESSB to the Nth degree is beyond the scope of this document. For a much more profound and detailed explanation of ESSB, to include how ESSB is viewed in both the Amateur and regulatory worlds, I highly suggest that you read John Anning's web site: <http://www.nu9n.com/>

So why this document, and how does ESSB pertain to the FT-1000MP Mark V? Well, the Mark V product lines are probably one of the easiest radios to convert to ESSB capabilities. However, one would never know this unless they happen upon another operator who is using their Mark V for ESSB. Try this out; using Google, perform a search for "ESSB Audio". A plethora of web sites will be presented to you. Most of these web sites reference the more popular ESSB capable rigs such as the Kenwood TS-850/DSP-100 combo, or the TS-870 with its capabilities straight out of the box. There is even work being done by WZ5Q (<http://www.wz5q.com/>) on the TS-950SDX to increase it's transmit and receive bandwidth capabilities. And there are a few other manufactures model's referenced out on the NET that are either ESSB capable or can become so with some modifications. But it seems these rigs are less popular in the ESSB world, and less is published about them on the Internet than the Kenwood's. The Mark V falls into this latter category, hence the creation of this document.

The ease of which to convert the MARK V for ESSB, versus that of the Kenwood product lines is pretty significant. However, one must not discount those rigs that have served the ESSB community so well. Below is a bulleted synopsis of those rigs that have served us so well and where they stand today. But before I go into this, I would like to tell all the owners of the rigs referenced below, please do not take offense, as this is just but one mans view.

- The granddaddy of them all is the Kenwood TS-850/DSP-100 combo. It has some of the smoothest audio around. However, the combo is extremely rare to find, and is in high demand within the ESSB community. This makes it quite expensive to acquire, whether as a package deal or thru separate purchases. Particularly speaking to the DSP-100, I have seen bidding wars on Ebay take the final winning price to well beyond that which would be considered reasonable. The modifications that must be performed to make this ensemble ESSB capable are extensive from the perspective of component changes; just take a look at KA0KA's website. However, the plus side to this is that OEM filter replacement is not required in order to achieve ESSB.
- The runner up is the Kenwood TS-870. It has a transmitted bandwidth of ~3.8KHz, less modifications, but externally EQ'd. If you have ever heard this radio on the air, it does sound very good, but still does not provide the full fidelity of that of the TS-850/DSP-100 combo. This radio's true claim to fame is the fact that straight out of the box, no modifications necessary, it's receive audio bandwidth is flat to roughly 6KHz. It's popularity within the ESSB world is average, thus the going used price for this rig is still reasonable.

- The ESSB capabilities of TS-950 series, in the hands of the master (WZ5Q), is predicted to rival the TS-850/DSP-100 combo; work is still being done. However, the component changes and other modifications are just as extensive, if not more so. This is not something that a typical Amateur Radio Operator will want to take on; Mike is on an elevated plane when it comes to such tasks. I would suspect that once Mike publishes his modifications on his website, the interest level for the TS-950 series for ESSB work will begin to rise. This series of Kenwood radios already holds a high value on the used market, as it is also a very popular rig in the DX community.
- The new high-end Yaesu 9000 is capable of up to 4kHz of transmitted SSB bandwidth. It's about time we start seeing a major manufacturer come out with a rig that has a little better fidelity. These rigs even come equipped with a standard 3 pin XLR connector along with an 8 pin DIN connector for interfacing a microphone, your choice. Unfortunately, one of these rigs will set you back \$12K, enough said there. And I do not think the new Icom 7800 can be considered as it is brick walled at 2900Hz and modification would be very difficult at best.
- Final mention should be given to the SDR-1000, FlexRadio. This rig is in a whole different world from that of the traditional "hardware" based radio systems. It uses software exclusively for all of its filtering, both transmit and receive. It has an incredible software adjustable transmit bandwidth of up to 20kHz, and a receive swath to match. If you were just getting started in the Amateur Radio Hobby, did not already have a rig, and were extremely interested in ESSB, then I would strongly suggest considering this radio. The caveat is that you will need a dedicated PC to run the radio, which equates to some amount of additional cost if you do not have an up to date computer system. But so does modifying any of the above mentioned rigs. It will cost you roughly \$2.5 - \$4K, radio and PC, depending upon options. Again, this type of radio is new, and will take some time to gain a strong foothold in the Amateur Community. I would definitely keep an eye and an ear out for this one, as it appears to have great potential within the hobby!

So how do these rigs compare with that of the featured rig in this document? Well, the Mark V, straight out of the box, using both Transmit and Receive DSP, is only capable of roughly 3kHz of transmit bandwidth, and 3kHz of receive bandwidth. Both transmit and receive need to be externally EQ'd in order to achieve a flat response. So far this doesn't sound so good. However, despite this radio's obvious shortcomings straight out of the box, to make this radio ESSB capable one only needs to replace two OEM filters with ones having a wider band-pass, tap four individual circuits, and install 4 RCA Female Jacks. Believe it or not, that is all there is to it. This simplicity, on top of the radio's general availability within the used market place for a reasonable price should make it a desirable radio for one to enter the ESSB world. After performing the modifications outlined in this document, its transmit bandwidth will be flat from 30Hz to roughly 5.8 – 5.9kHz, measured at a -6db point. The receive, dependant upon filter selection via the menu system, will also be flat out to ~5.8kHz at a -6db point for SSB, and ~7kHz at a -6db point for AM. A used Mark V will run you \$1400 - \$1700, no optional filters installed. And best of all, it is a "sleeper" in the ESSB world, flying just under the ESSB radar; or at least until this document goes out. So if your interested in ESSB, and you have one of these rigs, you are in luck. If you do not, they are readily available, and reasonably priced.

The remainder of this document will walk you through all phases of the required modifications to the Mark V for ESSB operations. It is highly suggested that you read this document in it's entirety prior to making any decision as to whether you personally want to do this. Such things as available materials, personal skill sets, breadth of knowledge, and operational practices should be considered before

making a decision. As you will see, there are many pictures through out this document facilitating all modifications.

Time for the usual disclaimer: Jim and I have performed these modifications to our own radio's, and have not had any problems with our radio's. So it is noted that Jim nor I will assume any risk or liability in any damage caused to your radio by you or anyone you ask or contract to perform these modifications. If you, or whom ever you ask or contract, either detours from this modification procedure, or lacks the proper skills or breadth of knowledge to properly perform this type of work, the risk is all yours. It should also be noted that we are not liable for any personal injury or harm that may be incurred by anyone who performs these modifications.

One last note before getting into the details of the modifications. The FT-1000MP plain and the Mark V (200 Watt Model & Field) are both very similar in circuit design and menu systems. I performed this work on a Mark V, 200 Watt model. Thus all of the pictures are that of a Mark V. However, it has been noted by the co-author VE7RF (Jim) that this work has been done on FT-1000MP's, resulting in the same enhanced bandwidth for both transmit and receive. Keep in mind though, that the naming convention for each board and its associated components will not be the same as the Mark V. Basically Q3046 that designates the Mark V BM, will not be the same name for the FT-1000MP's BM.

2. Required Materials

As with any project, the first step is to read and familiarize yourself with the project outline, which is what you are doing right now. The second step is to decide if the project is for you, and within your skill sets. The third step is to gather all of the parts and tools required to complete the project. The next couple of paragraphs will list in detail all the parts required for the project at hand. There is also a listing of tools needed to facilitate the construction phase. And considering that not everyone owns or has access to the same test equipment, the equipment list is broken out into required and optional (nice to have).

2.1 Parts

The following parts are required for completion of this project. Part numbers and sources for the parts are listed below:

- RCA Male Plug, 4ea – Markertek.com, NYS352, \$0.79ea
- RCA Female Jacks, 4ea – Markertek.com, PFCM, \$0.72ea
- Canare Star Quad Audio Cable, 14' (suggested) – Markertek.com, L-4E6S BK, \$0.36 per foot
- XLR Male Connector – Markertek.com, A3M, Switchcraft, \$1.99ea
- XLR Female Connector, 1ea – Markertek.com, A3F, Switchcraft, \$1.99ea
- Jensen Transformers:
 - JT-11P-1 (1:1) for BM Interface, \$70.00ea
 - JT-11-FLCF (1:1) for Receive Interface, \$36.99ea
 - JT-DB-E (12:0) For Microphone or Phone Patch Interface, \$73.08ea
- Project box (Aluminum or Plastic) – This can be picked up at most Fry's stores, or ordered through any of the Electronics Parts Houses.
- 6.8K Resistor, 10ea* – Digi-Key# P6.8KBACT-ND, \$0.67 total
- 2K SMD Resistors, 10ea* – Digi-Key# RR12P2.0KDCT-ND, \$1.40 total

- 220uF 25v Capacitors, 6ea – Digi-Key# P1181-ND, \$0.67ea
- 100uh Inductor, 1ea – Digi-Key# M8146-ND, \$0.30ea
- 26 – 30AWG wire, color jacketed – preferably 4 separate colors (one for each added trace)
- Inrad #707-C 5.8Khz 455Khz Filter, \$125
- Inrad #710 – 6Khz 8.215Mhz Filter, \$115
- Inrad #716-B – 2.8Khz 8.215Mhz Filter, \$155 (Optional, see Paragraph 3.1.1)
- Inrad #715-C – 2.8Khz 455Khz Filter, \$115 (Optional, see Paragraph 3.1.1)
- Solder & Flux

*Denotes minimum order allowed

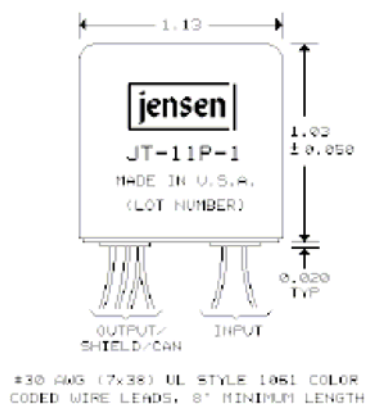
2.1.1 Ordering from Inrad

You should know that Inrad is not too fond of transmitted wide band audio, such as the band-pass we will accomplish today. So when ordering from Inrad, save yourself some grief, and either order off of their web page, or avoid the subject while ordering over the phone. If pressured, I would stick to a story of usage for AM. I am not a proponent of weaving stories, I pretty much speak my mind, and so I ordered off of the web. <http://www.qth.com/inrad/>

It should also be noted that the alpha character at the end of their part numbers (707-“C”) signifies that the filter should be mounted to an interface board for a specific radio. In the case of the 8.215Mhz filter for the Mark V, the OEM filter is solder directly to the IF Board. So when ordering the 6Khz 8.215Mhz filter from Inrad, be sure to omit the alpha character, and only request #710. The plus side of this is that the cost of the Inrad interface board is lumped into the overall cost of the filter, so they should send you the interface board without the filter mounted on it anyways. I say this is a plus as we can use this board to re-utilize our OEM 8.215Mhz filter, see Paragraph 3.1.1.

2.1.2 Ordering from Jensen

When ordering the specified transformers from Jensen Transformers, I should note that there are two distinct mounting form factors to choose from. Let's use JT-11P-1 as an example. When you order this 10K:10K transformer and ask for the specific part number of JT-11P-1, you will receive the transformer with a mounting L-bracket and two screws. This particular form factor looks just like the picture below:



The L-Bracket can be mounted between the input and output leads and then used to mount the transformer to your project box. The other form factor has a 3/8" threaded bushing mount and can be used to mount the transformer through a plate or radio frame...etc. In order to receive this form factor you need to specify "TB" at the end of the part number, JT-11P-1-TB. When ordering this form factor you will receive a transformer that looks like this:



Base your form factor on how you plan to mount/use the transformer.

One last note about the JT-11P-1 in particular, when looking at Figure 3 further on in this document, you will see an RC network depicted. You do not need to buy those parts separately; they come with the JT-11P-1 at no additional charge. Just be sure to mention that you do require the RC network, and the transformer will ship with the parts in a small plastic bag.

2.2 Tools

The following tools are required in order to perform the modifications:

- FT-1000MP/Mark V Operators Manual – This can be found on WZ5Q's web site
- FT-1000MP/Mark V Service Manual – This can be found on WZ5Q's web site
- Phillips Screw Driver, #2
- Phillips Screw Driver, #1
- Mechanical Pencil, or pencil with a good sharpener
- Both large and Small Wire cutters
- Needle Nose Pliers
- Wire strippers or sharp knife
- Simple DIP Extractor – Probably the most important tool there is during the disassembly and re-assembly of the FT-1000MP/Mark V (see Picture 1)
- Drill Press or good steady hand and hand drill
- Hammer
- Small Nail or Center Punch
- Fine Grade Metal File, or Dremel with sanding stone, can also use below drill bits to de-burr
- Drill bits – Either 5/64" through 1/4", or a Unibit
- Soldering Iron – Low wattage with a small tip (No big Bertha's needed here)



Picture 1: Dip Extraction Tool

2.3 Test Equipment

As with any tinkering within our hobby, it usually calls for test equipment. The below list of test equipment has been broken out into two categories, required and optional. When viewing the optional category, please do not get discouraged if you do not own or have access to any one piece mentioned, or all the pieces for that matter. In most cases, the procedures that would benefit from usage of such test equipment can also be performed using a second receiver, on the air marker stations such as WWV, or your DVM and or wattmeter. Some of the most clever and resourceful technicians that I have ever run across hailed from the Belgium Army. If you have ever interacted with that countries armed forces, you would understand that they are financially strapped and under equipped when compared to their American counterparts, much less their European counterparts. Yet they are able to perform some of the most critical alignments on their communications equipment using marker stations in conjunction with a second transceiver as a frequency counter/spectrum analyzer, DVM's as Oscilloscopes, and light bulbs as watt meters. I was extremely impressed with their ability to improvise and overcome!!

2.3.1 Required

This list is what you will definitely need to get by:

- Multi-Meter/DVM
- Watt Meter
- Audio Signal Generator – I use “Test Tone Generator”, a software package that converts your computer audio card into an Audio Signal Generator. You can download a 30 day trial version from this website: <http://www.timo.esser.dsl.pipex.com/ttg.htm>
- Audio Spectrum Analyzer – I use “SpectraPlus”, a software package that converts your computer audio card into an Audio Spectrum Analyzer. You can download a 30 day trial version from this website: <http://www.spectraplus.com/>
- 50 ohm Dummy Load

2.3.2 Optional

This is a list of optional equipment that you can use for more extensive alignments:

- RF Signal Generator, or second transceiver with a pigtail antenna
- Frequency Counter
- RF Spectrum Analyzer

- Oscilloscope
- RF Millivoltmeter

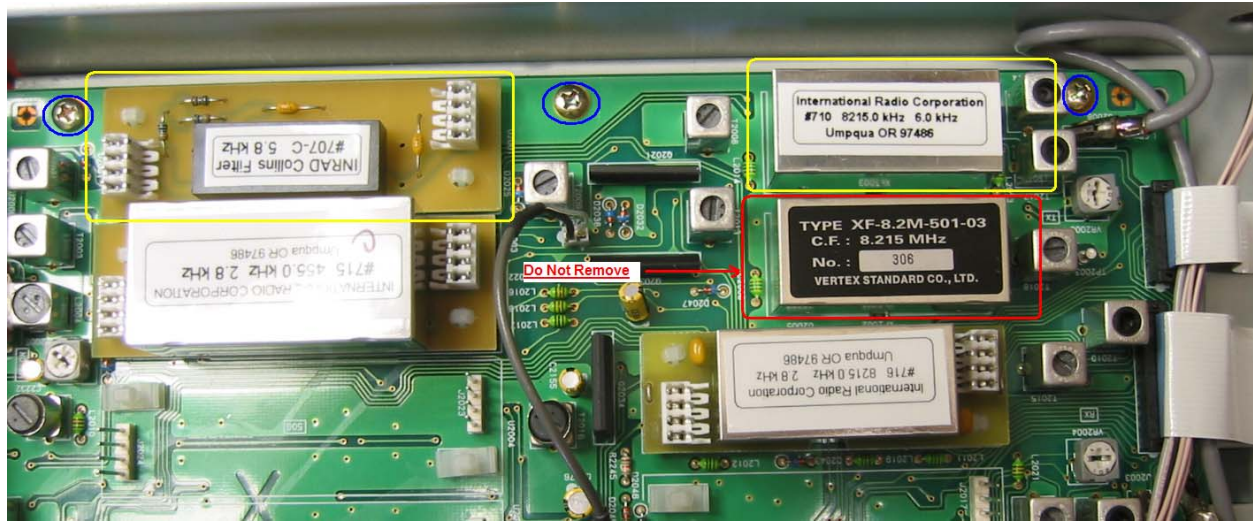
3. Construction

Before beginning construction, please take the time to ensure that you have all of the required parts, tools, and test equipment on hand. There is nothing worse than having to stop in mid stride because you forgot to purchase a part or locate a tool; something that has happened to me more than once. You should also, dependant upon your skill level, allot enough dedicated time to perform all of the work. I would suggest one day for the installation, and another to align and play with your new radio. Yes, I did purposely say “new radio”.

3.1 INRAD Filter Installation

The first step is to remove the covers off of the radio, both top and bottom and place them to the side. The #2 Phillips screwdriver is used for the larger screws that are visible on both the top and bottom covers. The #1 Phillips screwdriver is used for the two screws located down inside the heat exchanger that hold two sections of the top cover down, and the small screws on the top cover that contains the speaker and access panel. Use a pair of pliers or a magnet to remove the small screw from inside the heat exchange venting. This is to keep from dropping the screws inside of the radio. When removing the top, pay close attention to the speaker that is mounted to the underside of the cover. You will find two wires, red & black, that are attached to the speaker terminals. Pull those off of the speaker terminals and tuck them inside the compartment from which they came from, and then place the cover off to the side.

Below is a picture of the IF Unit with the 6Khz filters already installed. The filter to the right, outlined in yellow, is the Inrad #710 6Khz 8.215Mhz IF filter. This is the location where your OEM 2.6Khz filter currently resides. The one to the left, outlined in yellow, is the Inrad #707-C 5.8Khz 455Khz Collins filter. This is where your OEM 2.3Khz Collins filter currently resides. Close attention to how the Collin's filter is installed is imperative. If placed in the wrong direction, the filter will not work properly. Prior to continuing on, this would be a good time to remove the OEM Collins filter and install the new INRAD #707-C Collins Filter. Since the plastic pillars that hold the board upon which the filter is mounted has two fingers that jut out from the top forming an inverted V, you will need to use needle nose pliers to pinch the fingers closed so that you can remove the filter board. Once you have one side pinched closed, using either the DIP extraction tool or some other tool capable of grabbing underneath the filter board, pull that side of the filter board upwards until it reaches half way up the pillar fingers. Then move on to the other side and perform the same procedure, being careful not to cause the opposite side to reseal it self such that the pillar fingers grab hold of the filter board again. Once you have both sides released from the pillar fingers, then grab hold of both sides of the filter board and pull straight up. The filter board should come out of its slot, and the connection pins should still be straight.



Picture 2: Inrad 6KHz Filter Installation

In order to replace the OEM 2.6KHz 8.215Mhz filter, you will have to de-solder it from the IF Unit. This requires that you remove the IF Unit from the radio. This is a rather easy task, but a slow process. Looking at Picture 2, you will see the ribbon cables used to send control signals to the IF Unit and their associated connector blocks. These connector blocks are two parts interlocked within one another that form a pressure lock that holds the ribbon cable in place, while at the same time forming an electrical connection between the block and the cable. Using the DIP extraction tool, pull the top edges of the connector as shown in Figure 1. This will unlock the top portion from the bottom portion of the connector block. Do not pull too hard, as the two pieces are not meant to come apart. Once the top portion of the connector is unlocked from the bottom portion, you should be able to remove the ribbon cable from the connector body. To remove the ribbon cable out of the connector body you must hold onto the ribbon cable as close to the blue tab as possible and slightly rock the cable left to right while applying an upward force. When doing this, pay close attention to the top portion of the connector block. If it should fall down and thus re-lock itself, you should immediately stop pulling on the ribbon cable and unlock the connector once again before resuming removal of the ribbon cable.

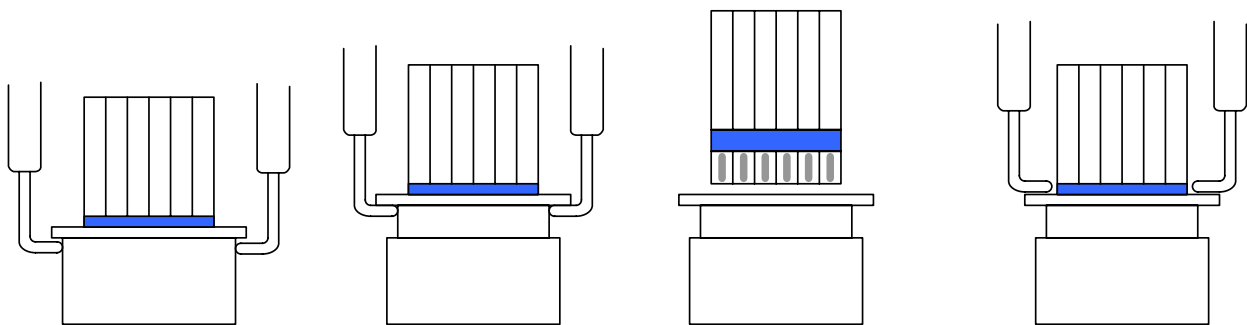
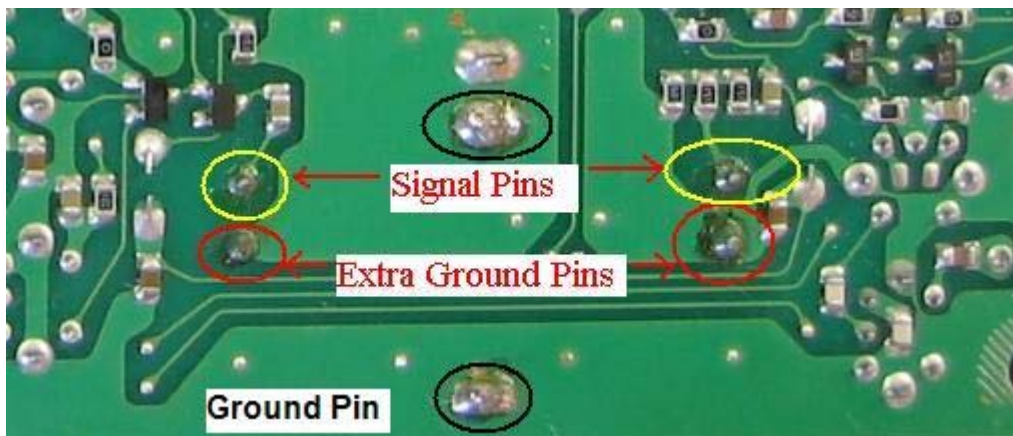


Figure 1: Ribbon Cable Removal and Replacement using DIP Extractor

After all the ribbon cables have been disconnected from the IF Unit, you should then disconnect the remaining coax and wires that are connected to the Board. The coax, such as that in the top right of Picture 2 is real easy to disconnect. It has two metal wings for which you can grab a hold of, and in a twisting motion, pull the cable out of its socket. The remaining wires are connected via blue/grey

connector blocks. However this time, both the top (blue) and bottom (grey) portions of the block are meant to come apart. Again, by grasping the top edges of the blue portion with the DIP extractor and providing an upward force, the two pieces should separate. A slight left to right rocking motion may be necessary to help it along. Do not pull the wires themselves. That has the potential of separating the wires from the blue connector thus costing time in repairing the situation.

After removing all cables, you are now ready to remove the screws used to hold the board in place. Looking at Picture 2, you will find an example of the screws that hold the board in place. These screws require the usage of a #2 Phillips screwdriver in order to keep from damaging the heads.



Picture 3: Pin location on Backside of IF Unit

Once you have the board out, place it upside down on the static mat, and orientate it versus Picture 3. Looking at the picture, you will see the thru hole connections circled in yellow and in black that need to be de-soldered in order to remove the OEM filter from the board. Find those connections on your board and de-solder them one at a time either using a de-soldering tool or the ole flux & solder wick method. Patience is a virtue at this point, and should be exercised. Too much heat for too long will damage the board. So do take your time and insure that your soldering iron is not set for greater 500 degrees.

Once you have the filter out, there are two more thru holes that you need to open up. You can find these in Picture 3 circled in red. These two holes are just extensions of the ground plane of the IF Unit and need to be opened up in order for your new Inrad filter to be placed onto the board. Close inspection of your new 6Khz 8.215Mhz Inrad filter will reveal two ground pins that will fit into these holes very nicely. This is also a good time to clean up any excess solder on any of the thru holes associated with the filter placement. Once all thru holes have been fully opened and cleaned, you are ready to install the new Inrad filter. Unlike the Collins filter, the Inrad crystal filter can be installed in either direction and still work great. Place the Inrad filter on the IF Unit such that the pins run thru the board and are visible on the backside of the board. Using a folded cloth, lay the board down on the static mat upside down such that the Inrad filter is resting on the cloth, with the rest of the IF Unit lying flat on the static mat. The idea here is to ensure that the filter is held flush to the IF Unit with the board lying upside down. Apply a little flux to the pins of the filter and solder the pins to the board, leaving a nice smooth curvature to the solder joint. Once completed, cut off the pin excess. Be sure to hold the excess piece before you cut it off, or the clipping can end up landing inside of the radio. Attempting to cut off the large tabs on either side of the filter, circled in black in Picture 3, does not need to happen. The concern here is only the long narrow pins grounding themselves to the chassis once the IF Unit is remounted to the radio.

Now that you have completed the filter install, you can go ahead and remount the IF Unit back to the radio chassis and re-install all of the ribbon cables to include coax connections. Again, the DIP extraction tool comes in real handy when re-installing the ribbon cables. You should first ensure that the connector block is fully open, insert the ribbon cable back into the block by pushing down on the cable until the silver contact portion of the cable is no longer visible. Using the DIP tool as shown in Figure 1, push the top block of the connector downward until it locks into place. After the ribbon cables have been re-installed, go ahead and re-install the rest of the cables onto the IF Unit.

***NOTE** - In the attempt to reduce redundancy, the remainder of this document will not further reference the procedures or tools used for removing and or re-installing either the ribbon cables or other supporting cable, to include the Boards. It is now assumed that the reader has a good understanding and has gained experience in these procedures.*

3.1.1 What to do with your OEM Filters

You may have noticed that the parts list called for the Inrad 2.8Khz filters. These are optional and if purchased, can be placed in the 2.0Khz slots as per the Operational Manual indicates. If you decided not to purchase these filters, you can still use the OEM filters you just removed, by placing them in the appropriate 2.0Khz slots.

When you ordered the 6.0Khz 8.215Mhz filter from Inrad, you should have requested the following part number, #710. Since the part number is not followed by an alpha character such as "C" or "B", this means you are asking for the filter minus the interface board. In other words the filter should ship to you not mounted to an Inrad interface board. However, since the interface board is included in the price of the filter, Inrad ships the interface board as well. This is actually a plus for us, because if we are to re-use the OEM 8.215Mhz filter, we need such a board in order to place the filter into the 2.0Khz slot.



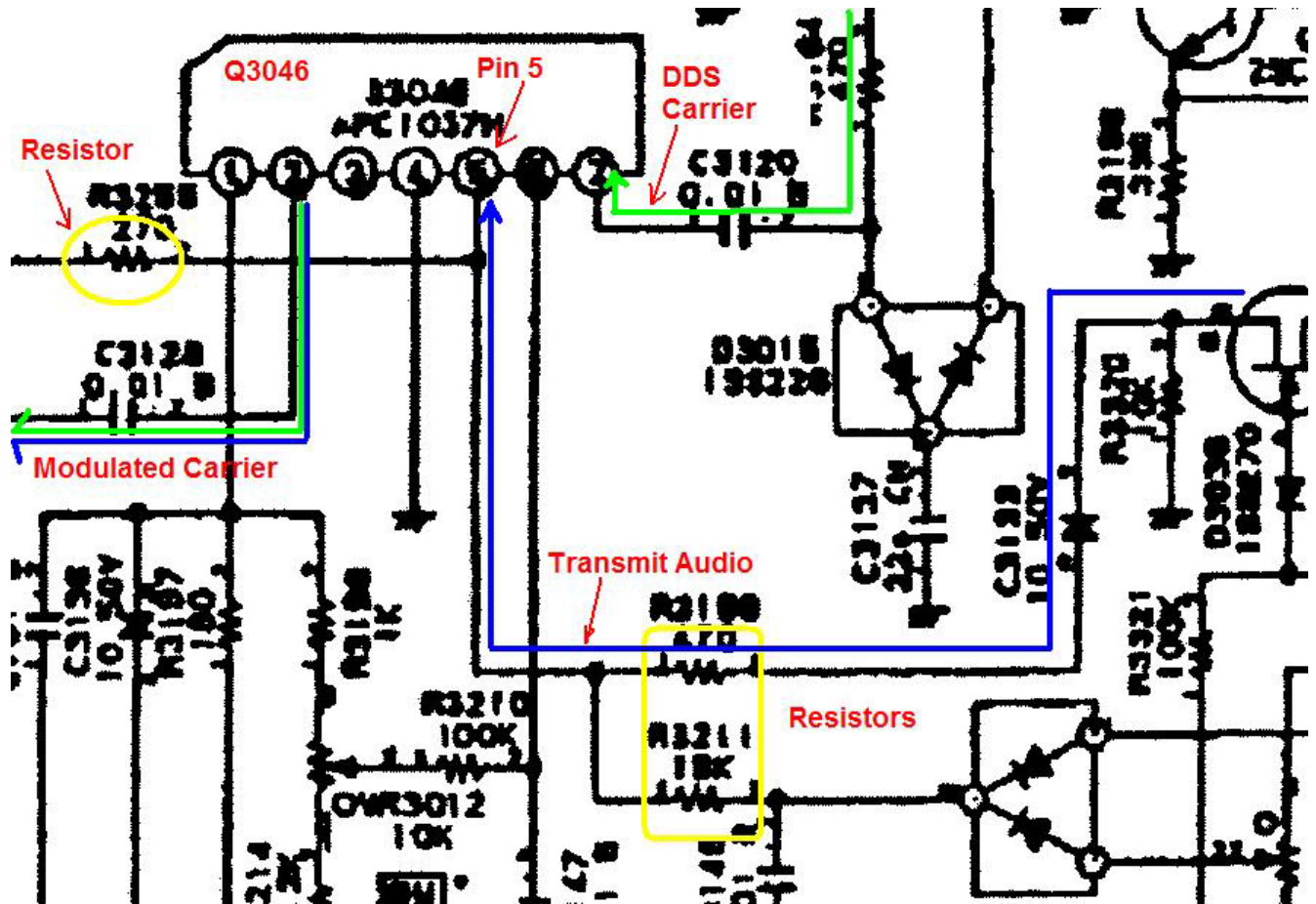
Picture 4: OEM Filter Mounted on Inrad Board

As was mentioned earlier regarding the 6Khz 8.215Mhz filter, the OEM XF-8.2M-262-01 filter can be mounted in any direction on the interface board and still work fine. So I would clean up any excess solder that may still be left on the OEM filter pins, insert the filter into the board, and solder the filter to the board. Once completed, you will be able to install the filter into the optional 8.215 Mhz 2.0Khz slot. What you need to know about engaging and using the filters can be found in Paragraph 4.3.2. It should also be noted that the “262” designator on the OEM filter indicates that the filter is 2.6Khz in bandwidth.

If you decide to install the optional 2.8Khz filters instead, then store the OEM filters in a safe place just incase you decide to sell the radio.

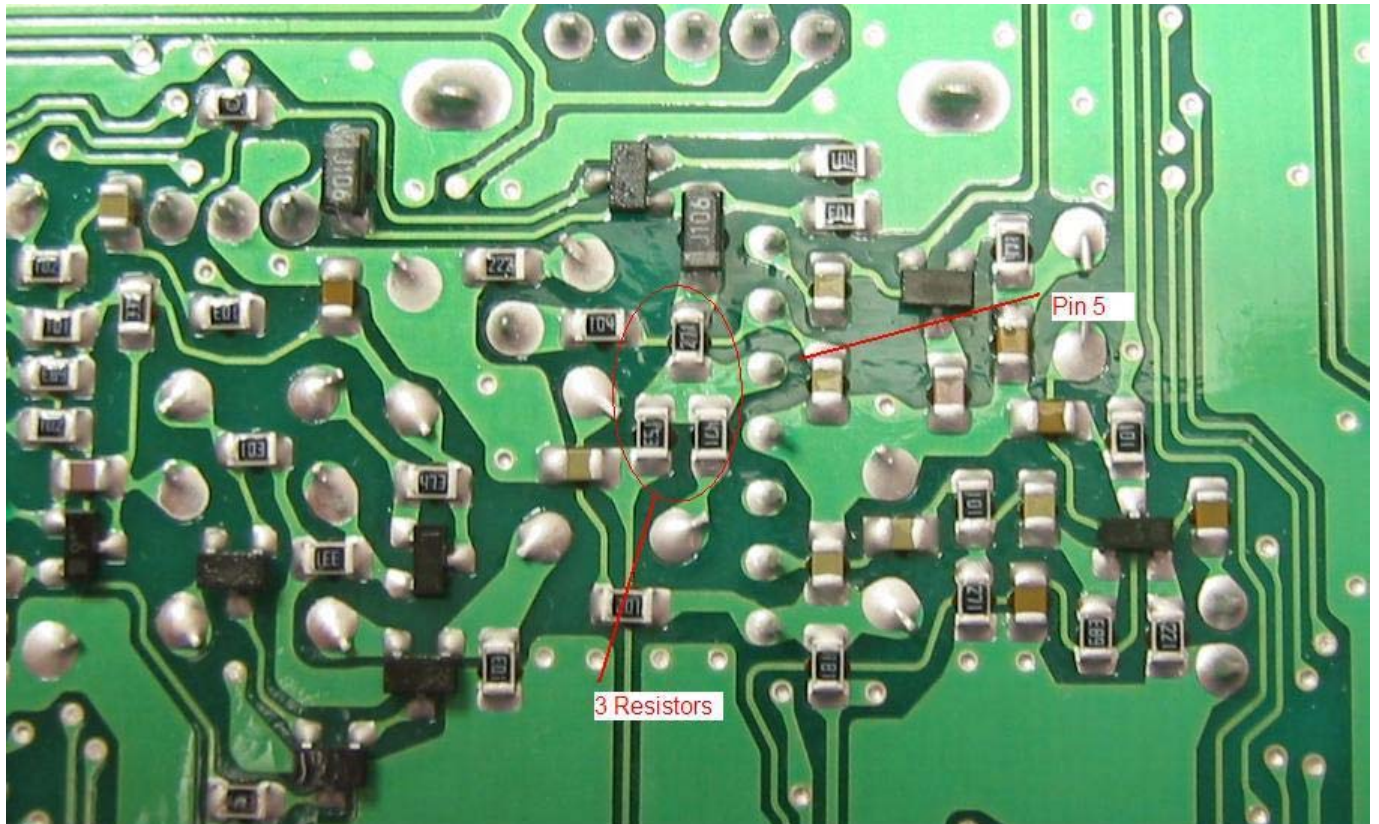
3.2 Balanced Modulator

Unfortunately, the transmitted audio, prior to being presented to the Balance Modulator, is passed from the Microphone Connector on the front panel of the radio, through a High Pass Filter and a Low Pass Filter. These filters perform bandwidth limiting such that the audio band-pass being presented to the BM is roughly 80Hz to 3.1Khz at a -6db point. You can bypass these filters by installing a wire tap directly to the BM, and terminating this wire tap to an RCA Jack on the back of the radio. In doing this, you will have bypassed the bandwidth limiting filters as well as the MIC gain amplifier, yet have gained the ability of transmitting the full audio band-pass allowed by your new filters. Although, it should be noted that you will have to use external audio equipment to control the audio bandwidth & level being presented to the BM, see Paragraph 6.1. In addition, you will also need to ensure that your microphone gain on the front panel of the radio is set fully counter clockwise; turned all the way down. Now the radio's ALC meter still registers the audio level being presented to the BM, as the pick off point for that circuit is after the BM has processed the audio. So you still have the ability to use the ALC meter deflection as a good reference point when determining if you are overdriving the BM.



Schematic 1: Balanced Modulator (Lower left of AF Unit Schematic)

The BM actually resides on the AF Unit. Looking at Schematic 1, we can see that the audio coming from the MIC OP-AMP is presented to the BM, Q3046 pin 5. Notice also that there are 3 distinct resistors that terminate on pin 5 as well. This arrangement of resistors can be used to visually help determine the location of pin 5 of the BM on the underside of the AF Unit; see Pictures 5 & 6.

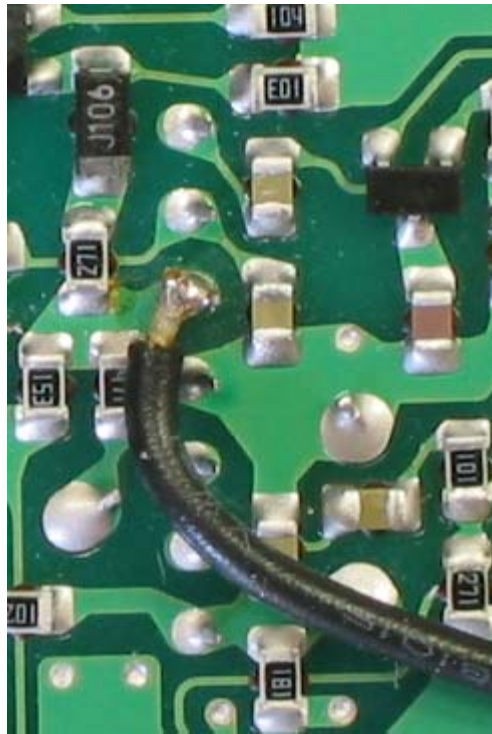


Picture 6: BM location on backside of AF Unit

In order to perform this modification, we will need to remove the AF Unit from the radio. Out of all of the circuit boards that will be modified for our new audio, this one will be the most time consuming removal as the back plate of the radio needs to be removed as well. I would suggest removing the back plate first, then moving on to the removal of the AF Unit.

***Note:** The back plate needs to come off anyways, in order to drill the holes for the RCA Jack or Jacks. Metal fragments and electrical circuits just do not co-exist all that well.*

There are two ways that you could jumper to pin 5 of the BM. The first, which is shown in Picture 7, is to directly solder a wire to pin 5 of the BM; the black wire. If you are the least bit worried about RF getting into your audio, you can solder a 100uh 1/8w inductor to pin 5 of the BM and then attach your wire to the other end of the inductor. Prior to soldering the wire, ensure that the wire is clipped long enough to route underneath the AF Unit and to the back of the radio.



Picture 7: BM Pin 5 Wire Tap

Now it should be said that if you were to look hard on the Internet you will find another operator who has done this very same BM Pin 5 modification for the sole purpose of transmitting a wider AM signal. In his modification, and thus the second method, he used a shielded cable in order to bring the transmit audio to the back plate of the radio. If you feel more comfortable in using this method, then by all means do so. However, it should be noted that there are several of us who have performed this modification using an unshielded wire, and none of us has experienced any trouble with our transmitted audio. Well, I should say that none of us has experienced problems due to the usage of an unshielded wire.

In any case, when using a shielded cable you will need to ground the shield to an AF Unit ground point. This technique can be done in one of two ways. The first is to actually tap BM Pin 5 from the top of the AF Unit. Looking closely at Picture 5 you will see a small burn mark right in front of Pin 5 of the BM. That is where I had actually performed this modification a year ago; unsteady hands at the time, hence the burn mark. The signal lead was attached to a 100uf 25Vdc capacitor which was attached to pin 5 and the ground lead was attached to the DDS Carrier cover, just above the BM in Picture 5. You can still see the solder point for the shield ground on the cover. I took the cover off in order to complete the solder joint. The cover is square, so it will fit in any direction.

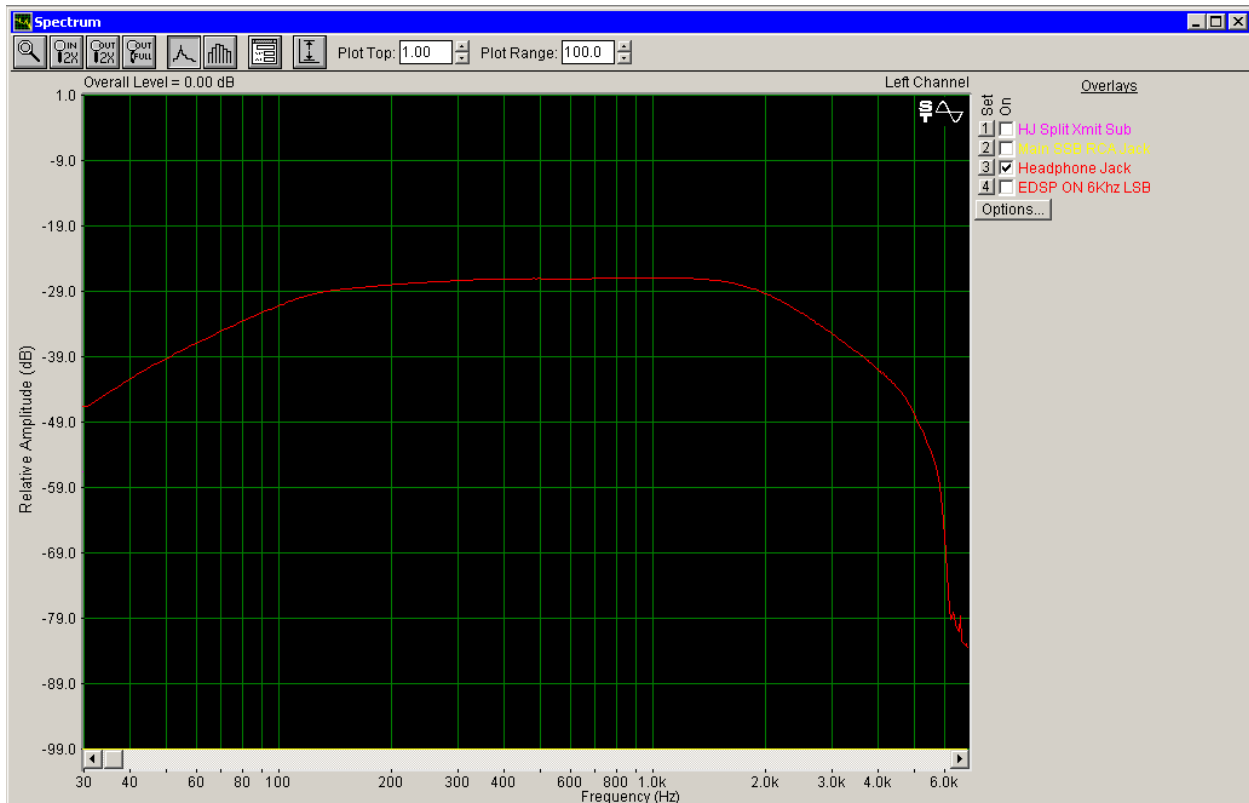
The second technique of attaching a shielded cable is to perform this on the bottom of the AF Unit. You solder the signal lead in the same location as you would the single wire; see Picture 7. The ground shield of the cable can be attached to Pin 4 of the BM, the pin just below Pin 5 in Picture 6. If you perform either of these techniques, be sure not to ground the cable shield to the chassis of the radio. Terminate the center conductor to a coupling capacitor that is affixed to the RCA Jack on the back-plate, treat it as a shield ground for an unbalanced cable, in other words leave the shield un-terminated on the RCA Jack end, see Paragraph 3.4.1.

Usage of a coupling capacitor is extremely important and a requirement for this modification! If any DC voltage is allowed to flow to Pin 5 of the BM, the BM will become unbalanced and you will transmit a full-blown carrier signal while in SSB mode.

This pretty much concludes the instructions on how to directly feed the Balanced Modulator. The next several paragraphs tell you how to get the most out of your receiver with your new filters.

3.3 Receive Audio

The stock receive audio of the Mark V is not capable of properly processing and presenting an ESSB audio signal. This is due to a High Pass Filter and Low Pass filter that the audio signal is routed through straight from the selected Product Detector on the way to the audio OP-AMPS for final amplification. These filters perform bandwidth limiting such that the audio band-pass at a -6db point is roughly 90Hz – 2.8Khz, dependant upon operational mode and filter selections. Just look at Picture 8.



Picture 8: Audio Through Headphone Jack

The red graph is the audio pulled from the headphone jack while transmitting an audio sweep from 10Hz- 6Khz through the Main Transmitter via the BM interface, with the monitor circuit turned on. We are actually seeing two things that have affected this graph, the first is the 6Khz transmit filters we just installed (the test was done after filters were replaced). And the second, and the reason why we do not see a wider response out of the headphone jack, is the affects of the high and low pass filters in our audio circuit. This second part is what we are now trying to over come.

With this in mind, there are two distinct methods for which to improve the audio response of the Mark V. The first method listed involves component changes on the AF Unit. The second method is to affix wiretaps directly to the Product Detectors for both the main and sub receiver boards, and bring those unfiltered audio signals out to RCA jacks mounted on the back panel of the Mark V. Each method has its pro's and con's:

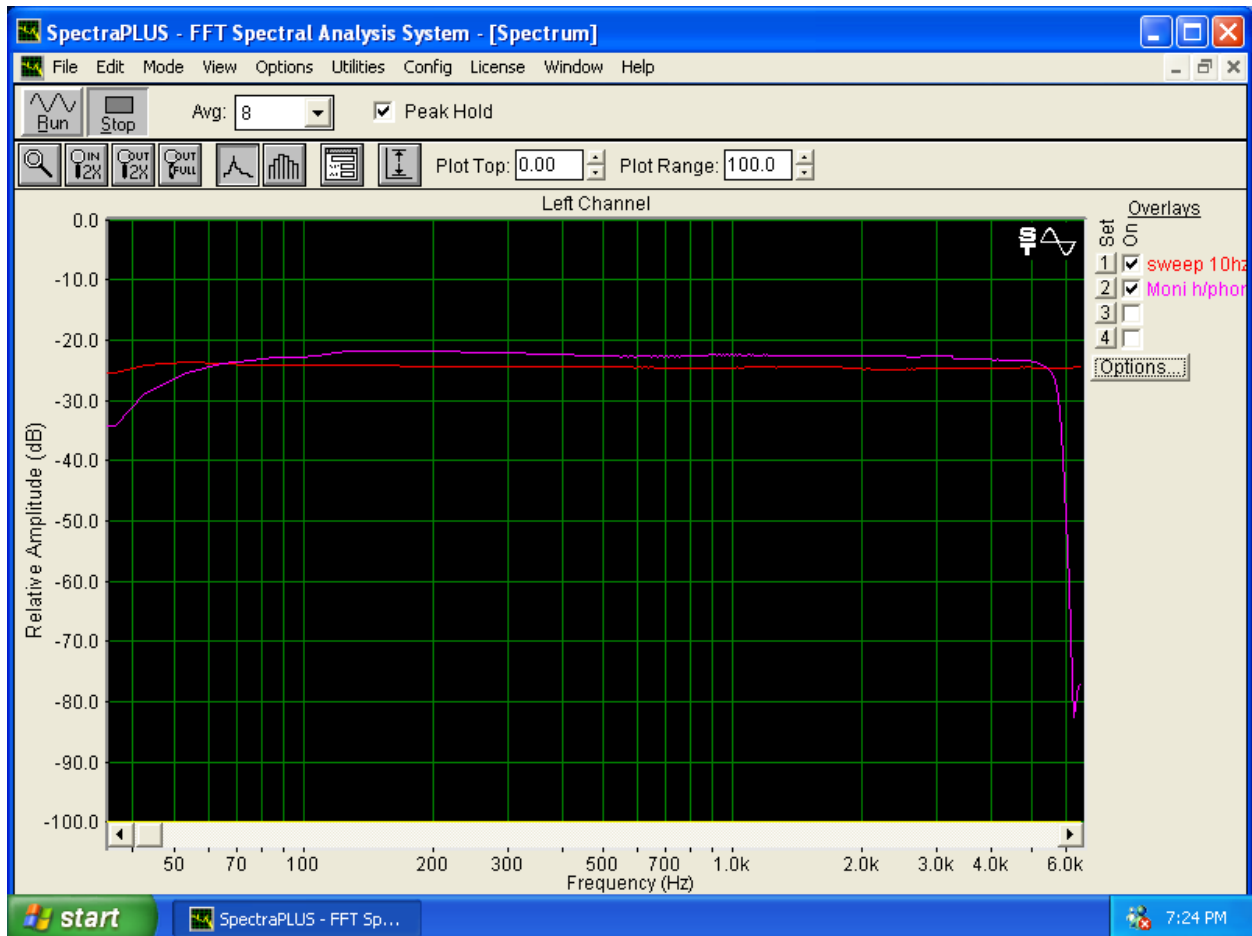
- **Component Changes Pro-** Allows you to utilize all of the OEM audio jacks (headphone, AF out, Speaker...etc) to listen to the extended receive bandwidth.
- **Component Changes Con** – It is rather difficult to perform, as well as difficult to change the radio back to a stock condition. If at some point you want to sell the radio, and do not revert the radio back to stock condition, then you are limiting your re-sale to a finite customer base; ESSB proponents only.
- **RCA Jacks Pro** – Easy to implement, and you gain additional receive bandwidth by picking the audio right off of the Product Detectors. The OEM audio jacks all work as they did before, slightly improved, yet still bandwidth limited audio. Minus the RCA Jacks themselves, it is real easy to revert back to a stock condition.
- **RCA Jacks Con** – You will have to drill holes in the back plate of the radio in order to mount the RCA Jacks. This type of modification has traditionally dropped the resale price of most rigs. The audio signal coming out of the Detectors is much lower than line-level, thus must be amplified to a line-level before it can be used in any further audio shaping or amplification equipment. This will require usage of an external audio pre-amplifier.

As can be seen, a decision needs to be made as to which method would be preferred. I personally installed the RCA Jacks for two reasons, the ability to quickly revert back to stock, and the additional bandwidth gained. You should also know that I had spare audio equipment lying around that could cleanly boost the low level audio signal coming off of the Product Detectors to a line-level. Now this is not to say that you couldn't do both routes, wiretaps & component changes, and have the best of both worlds. Ultimately, the choice is yours.

***NOTE** – If you are only performing the component changes, then you will not require several of the parts listed in Paragraph 2.1. Be sure to examine the parts list closely, unless of course you just like excess parts lying around.*

3.3.1 Component Changes

Originally, I did not perform the component changes, preferring and implementing the product detector method. However, I have since gone back and made the necessary component changes on both the AF and RX2 Unit boards, and have found the results of which to be quite satisfactory. Albeit, changing the components will not give you the same bandwidth as directly tapping the product detectors. But it does allow you to realize expanded receive audio out of the OEM audio interfaces of the radio. To give you an idea of what I am talking about in regards to improving your receive audio response, versus that of Picture 8, below is a clip from SpectraPlus taken during an audio sweep of a Mark V straight from the headphone jack.



Picture 9: Audio Sweep after component changes

Roscoe (VK3KRH) performed this sweep after he completed the component changes on both the Main and Sub Receiver. The red line is an audio sweep through his mixer pointed straight back to SpectraPlus, basically a test sweep. The purple is an audio sweep through the Mark V while transmitting. The audio was actually pulled from the headphone jack of the Mark V with the monitor circuit turned on. There is a huge difference between this graph and the one shown in Picture 8.

Now I should warn you that this modification does require replacing SMD resistors. So I must say something about working with Surface Mount Devices (SMD). You will definitely want to use the ole solder wick method when removing these components. The use of a solder sucker will more than likely result in the SMD getting sucked up into the solder sucker itself. Also, extreme patience needs to be exercised when removing these components. I would remove solder from one side of the SMD and then the other, in an alternating fashion, back and forth. I strongly suggest using flux during this process, just dab the solder wick prior to placing it on the side of the SMD. Eventually the SMD will come loose, just be cognizant that this will happen unexpectedly, thus not losing track of the SMD. When soldering the new SMD, I would pre-apply solder to one of the pads only. Then with tweezers place the new SMD on the pad. Swap the tweezers out with a toothpick, and use the toothpick to place downward pressure on the SMD itself. Then reheat the pad until the SMD sinks into the solder. This will tack the SMD to the pad and then you can work on soldering the other side of the SMD. This method works well, as long as

the new SMD is the same size as the old SMD's. If the new ones are larger in size, then you will have to mount them in a slanted manner in order to build the solder up from the pad to the bottom of the SMD.

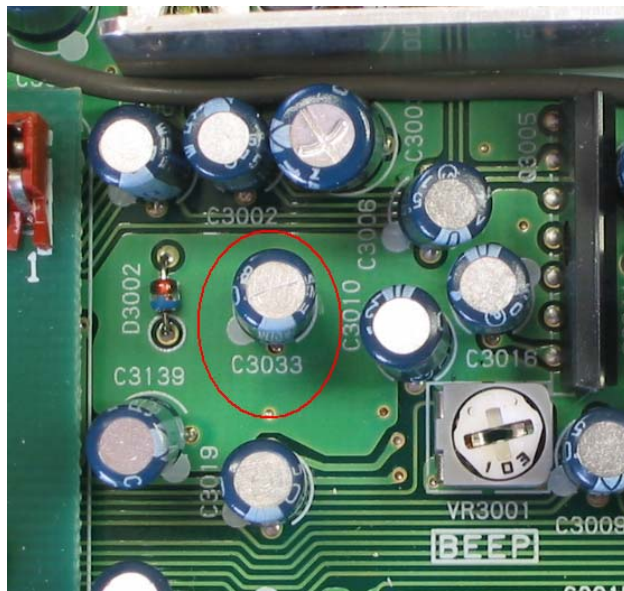
3.3.1.1 SSB Main Receiver Component changes

Here is a list of those components and their values, and along side of them, the specifications for their replacements.

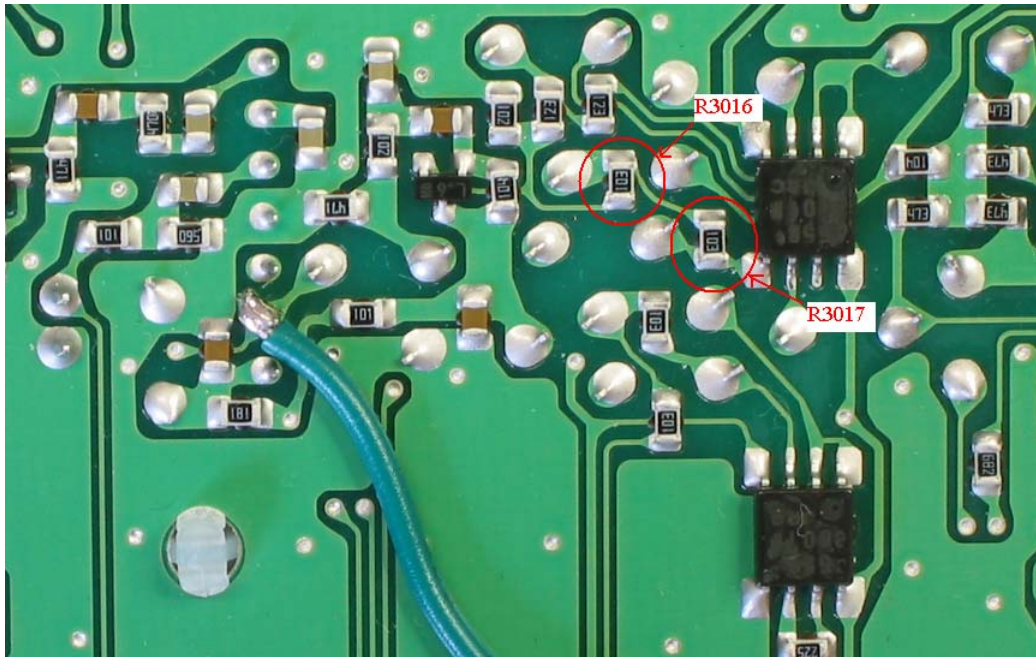
Component	Orig Value	New Value
R3016	10K Ohms	2K SMD, 1/10W, Package Size 0805
R3017	10K Ohms	2K SMD, 1/10W, Package Size 0805
C3033	10uf	Panasonic SU Series 220uf 25Vdc

Table 1: AF Unit Component List

Looking at the below pictures of the AF Unit, you will see the components that need to be changed circled in red.



Picture 10: AF Unit Capacitor Location



Picture 11: AF Unit Resistor Locations

As we did with the OEM filter exchange, we will have to de-solder the original components and then solder in the new ones. Careful attention has to be given when de-soldering Surface Mount Devices. It is very easy to lift a pad if not de-soldered carefully. When you have completed this do not mount the AF Unit until you have read Paragraph 5.3.

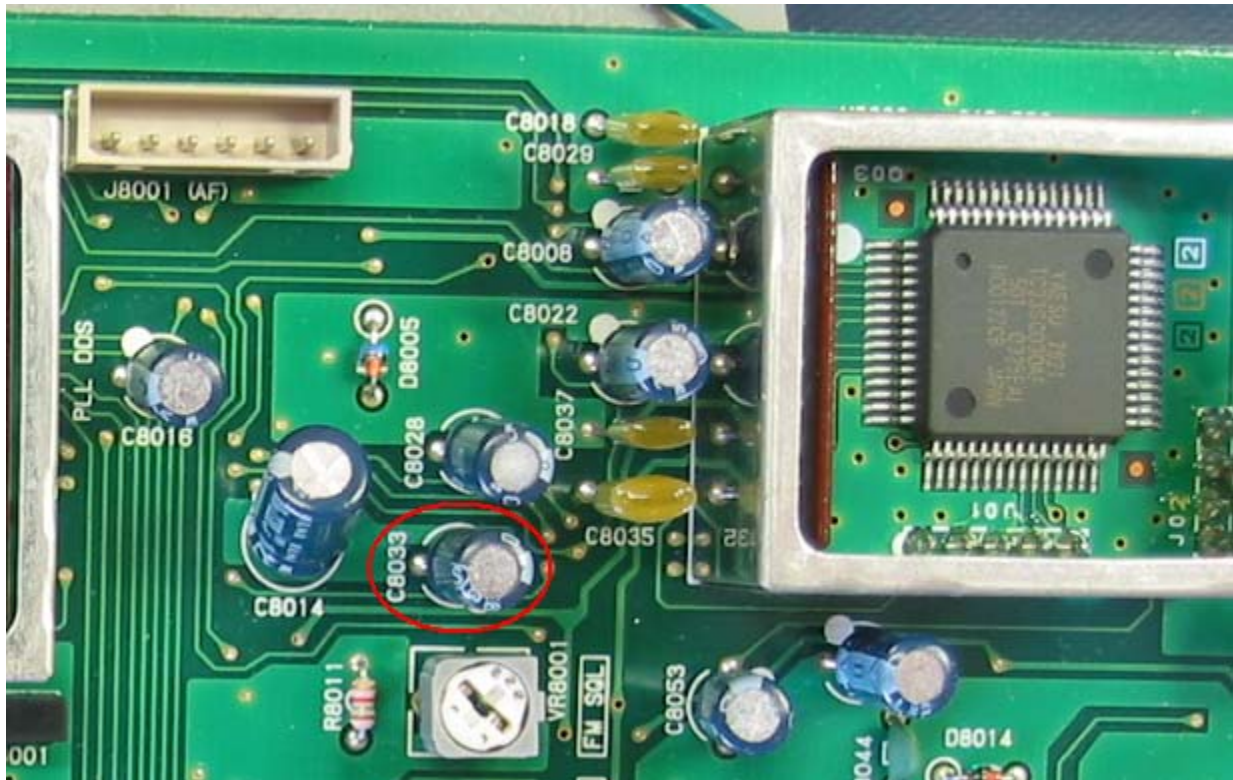
3.3.1.2 SSB Sub-Receiver Component changes

Here is a list of those components and their values, and along side of them, the specifications for their replacements.

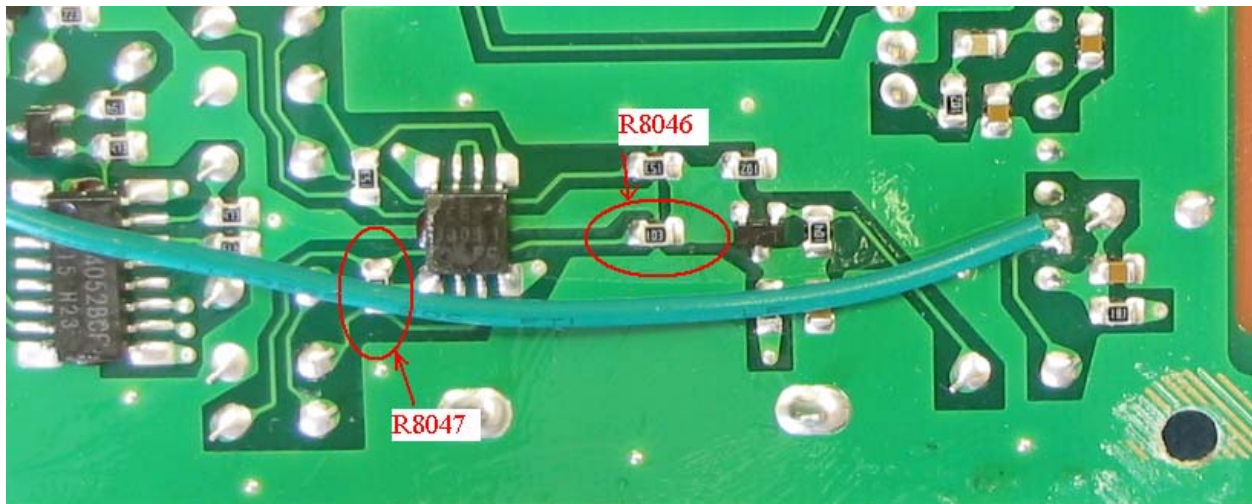
Component	Orig Value	New Value
R8046	10K Ohms	2K SMD, 1/10W, Package Size 0805
R8047	10K Ohms	2K SMD, 1/10W, Package Size 0805
C8033	10uf	Panasonic SU Series 220uf 25Vdc

Table 2: RX2 Unit Component List

Looking at the below pictures of the RX2 Unit you will see the components that need to be changed circled in red.



Picture 12: RX2 Unit Capacitor Location



Picture 13: RX2 Unit Resistor Locations

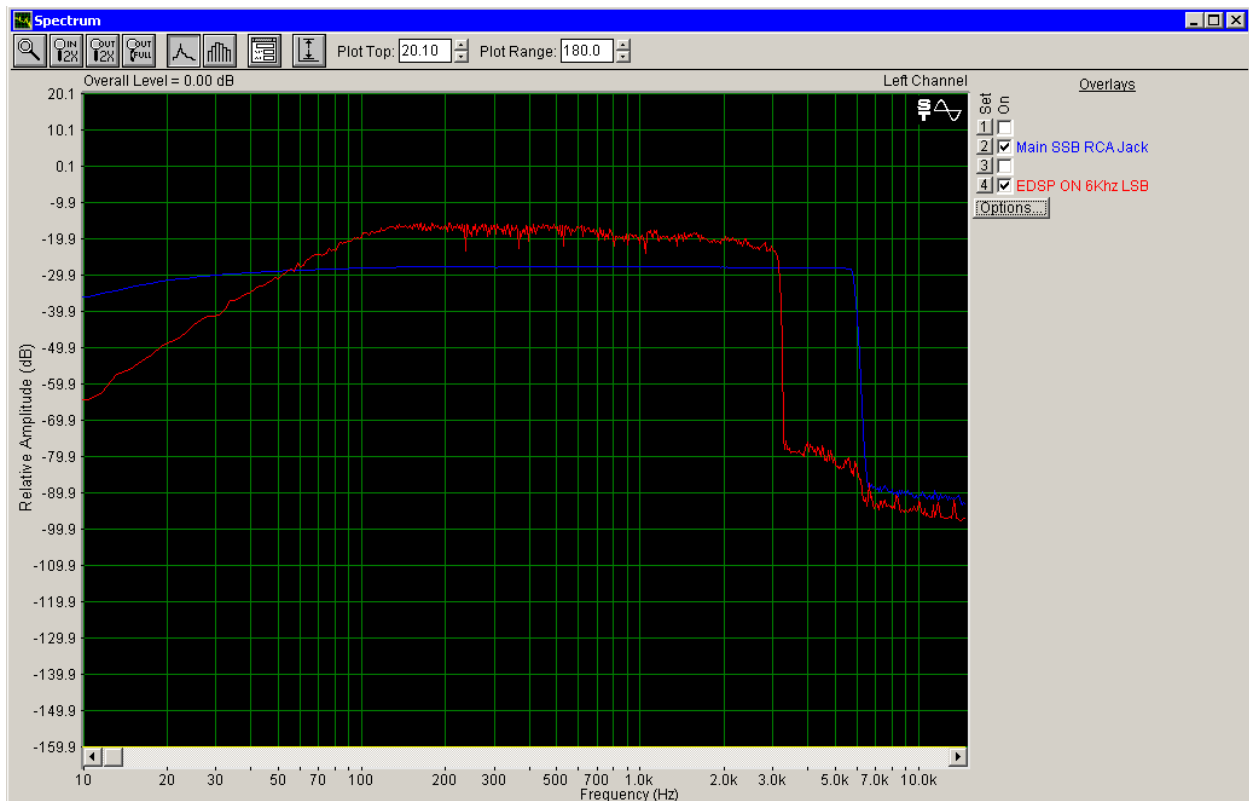
Again, we will have to de-solder the original components and then solder in the new ones. Careful attention should be given when de-soldering Surface Mount Devices. And by the way, sorry for inadvertently covering up R8047. When you have completed this you can go ahead and remount the RX2

Unit back to the frame of the radio. This is of course if you decide not to wire tap the SSB Product Detector as well.

NOTE – In the next paragraph you will be presented with a SpectraPlus audio sweep graph that is definitely better than the one depicting the component changes. I suspect the reason why there is less low end when changing out the recommended components, is that there are still several 1uf coupling capacitors in the receive audio signal path that have not been changed out. At the time of this writing, I chose not to change them all out. So I cannot definitively say that it would be required in order to achieve the same low-end response as you would with directly tapping the product detectors.

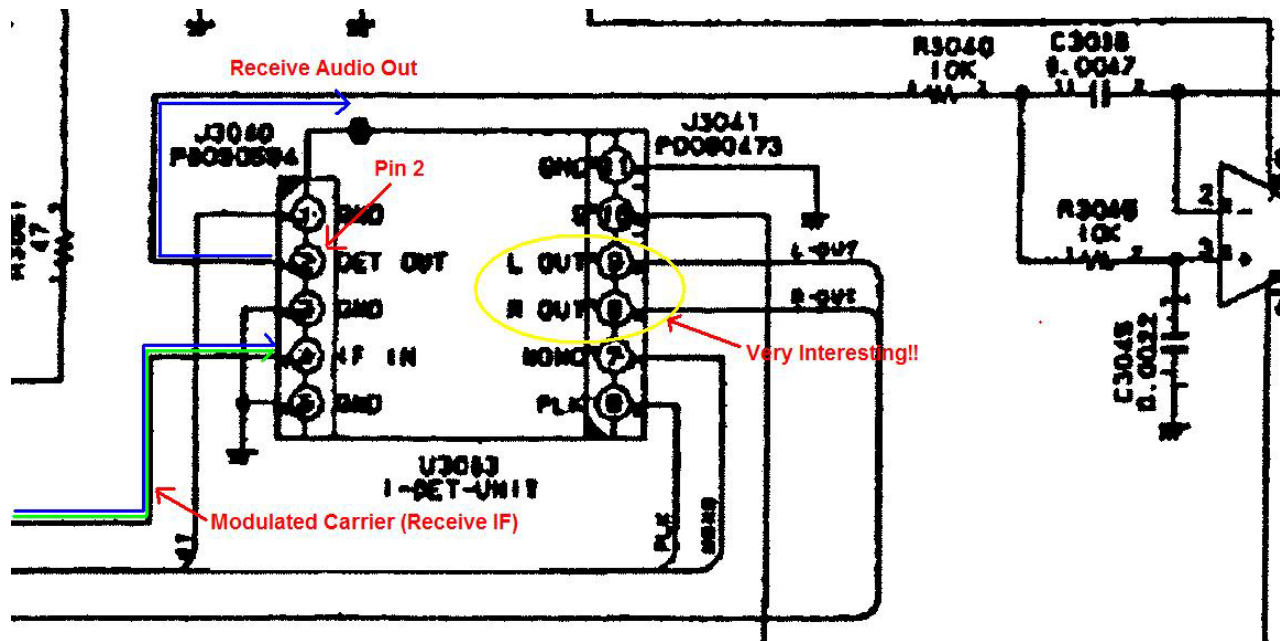
3.3.2 Direct Analog Product Detector Interface

The next couple of paragraphs will walk you through the circuit description for the Analog SSB Product Detector's and locations where we need to jumper them. At the end of this work, you should be able to realize the below audio improvement:



Picture 14: Sub RX, Analog SSB Product Detector Output

The red trace is a transmitted audio sweep through the Mark V with EDSP turned on, 100Hz-3100Khz w/ 6Khz transmit filter engaged, Menu 5-0 set for 455 as in Paragraph 4.3.1, and the sub-receiver setup as discussed in Paragraph 4.3.3, with monitor turned on. The menu setting tells the radio to pass the receive IF straight from the roofing filter right into the Inrad 6Khz 8.215Mhz filter and then onto the OEM AM Filter, thus providing you a wider listening band-pass. The blue trace is the same transmitted audio sweep, but this time the audio is being presented directly to the BM, and then picked off at the Sub-Receiver Product Detector with the Monitor Circuit turned on. The difference is quite substantial. If you

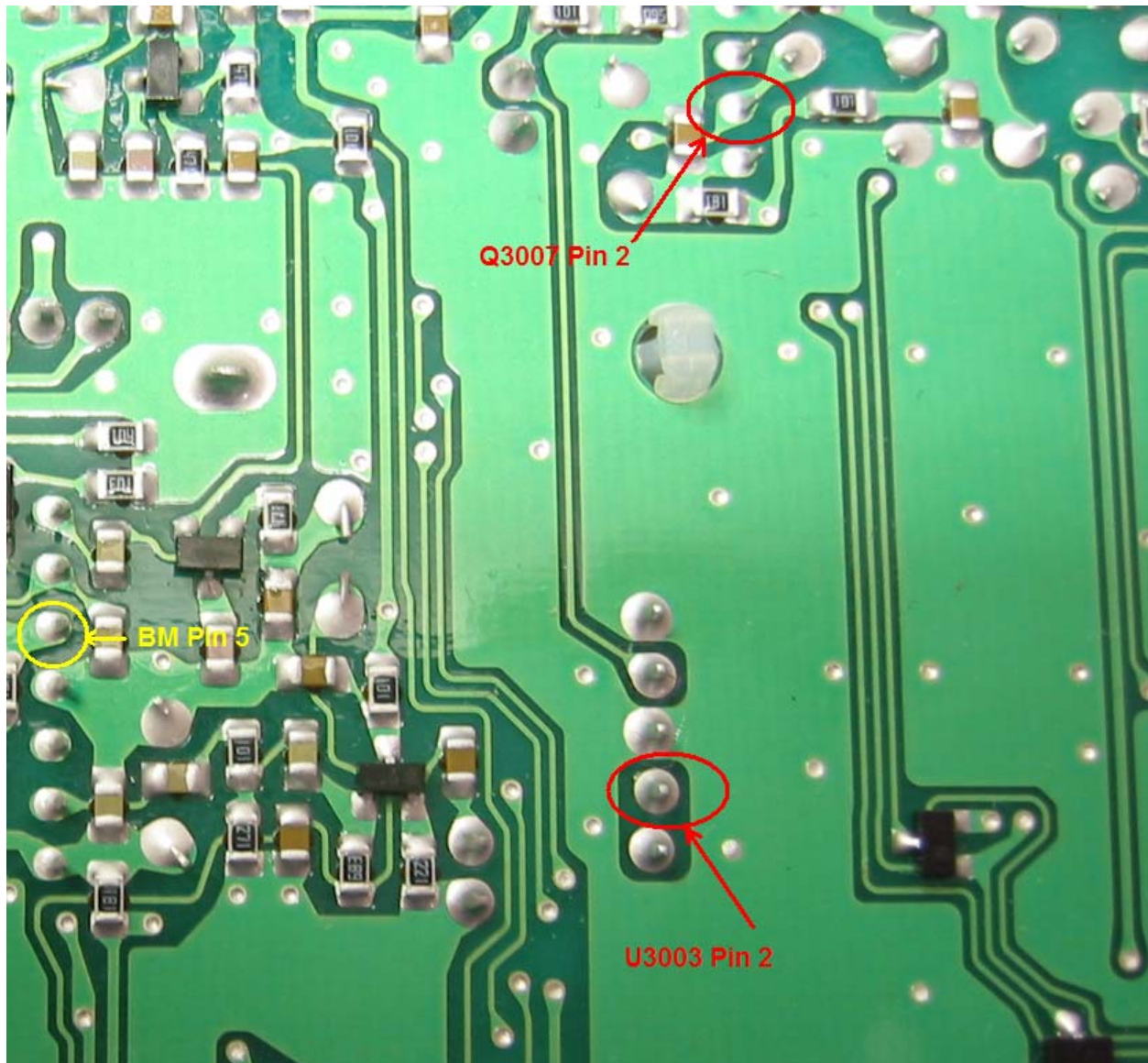


Schematic 3: Main AM Detector (Upper left of AF Unit Schematic)

In this circuit, the Receive IF is split off prior to being presented to Q3007, and is sent to the Main Receiver AM Detector, U3003 pin 4. There is no switching of this signal due to operational mode, SSB or AM, thus the Receive IF is always presented to U3003 regardless of the operational mode chosen. Since this circuit is strictly used for AM demodulation, there is no DDS Carrier required to demodulate the target signal. The resulting audio is sent out to buffer Q3013-1 via Pin 2. Interesting enough is Pin 8 & 9 of U3003. It is suggestive of a possible Left and Right Channel output. However, if you trace this lead, you will find that it terminates on a female plug that goes nowhere. The next time I am inside the radio, I will investigate this circuit and see if it provides any additional AM functionality to the table.

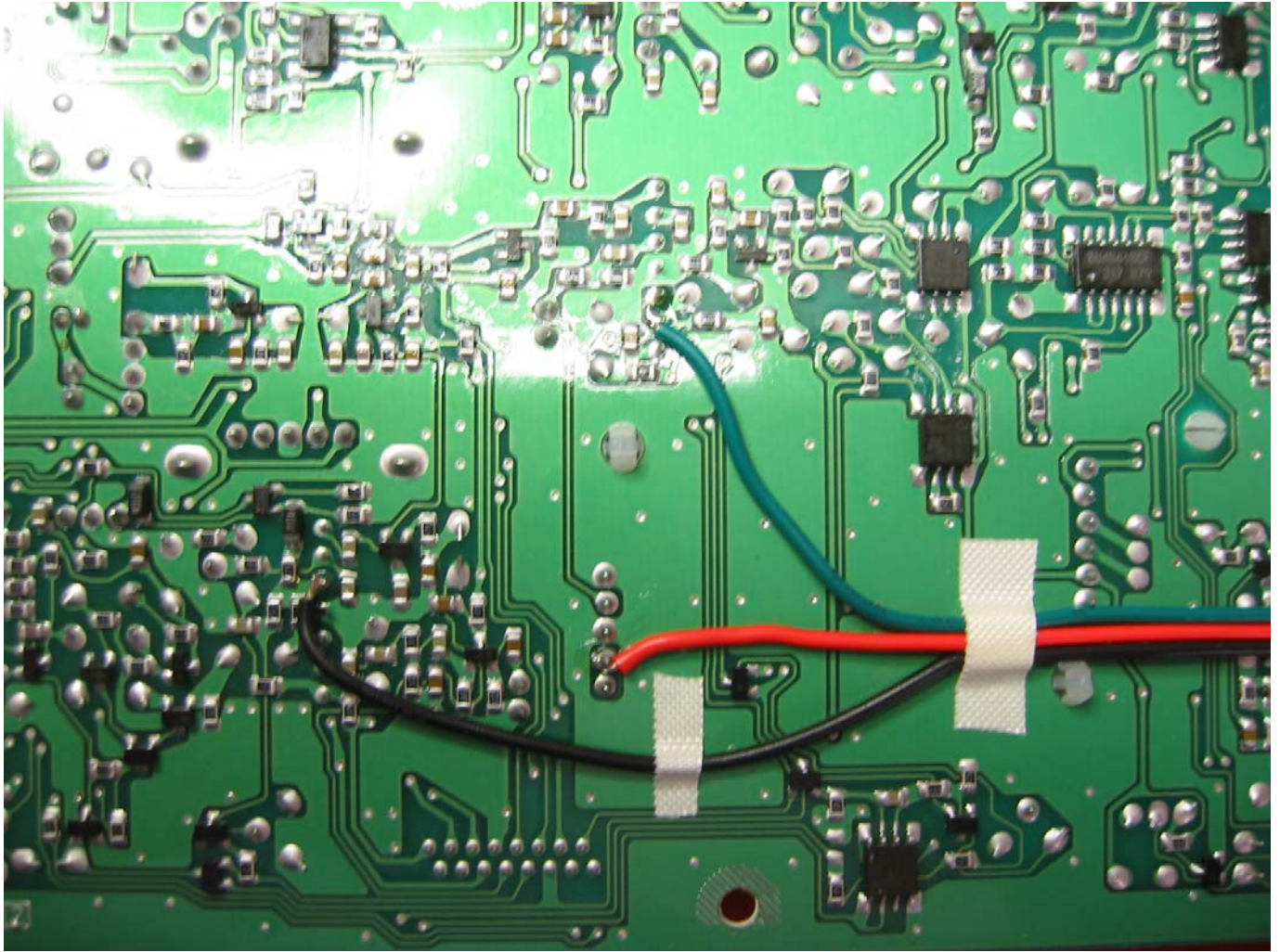
Notice that in all cases the audio is coupled to the next stage via a capacitor. Dependant upon the circuit, the capacitor uf value fluctuates, yet the voltage rating remains the same, 50Vdc. It is in this author's opinion that the voltage rating of the capacitor's involved are higher than what is truly required, and was used due to availability of parts, thus reducing manufacturing complexities. If this is the case, then we can drop the voltage level of our Panasonic SU Series 220uf coupling capacitors to 25Vdc, thus reducing the size of our coupling capacitors.

Below is a picture of where we will be tapping the AF Unit for both the SSB Product Detector and the AM Detector circuits. Pin 2 of Q3007 and Pin 2 of U3003 are circled in red.



Picture 15: Main SSB & AM Audio Pin Location on AF Unit

As we did with the Balanced Modulator, we will tap each of these pins with an individual wire. It was mentioned in the parts list that you should obtain wire with different color jackets. This is extremely beneficial in keeping track of which wire is carrying what signal. Looking at Picture 16, you will see where the Main SSB Product Detector (Green Wire), the AM Detector (Red Wire), and the BM (Black Wire) have been tapped.



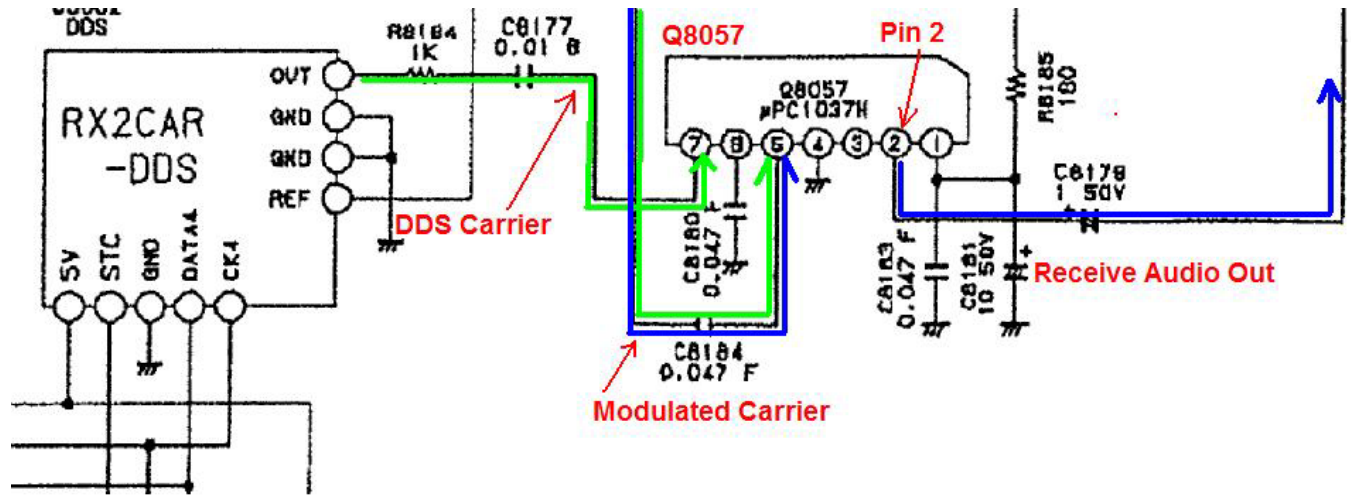
Picture 16: Main SSB Product Detector and AM Detector Taps

Once you have completed tapping these circuits that will conclude all of the audio work that is required on the AF Unit, see the below Note. Notice in the above picture that the wires were grouped into a bundle and taped to the AF Unit. This will help to keep the wires stationary. When you have completed routing the wire off to the side of the AF Unit, you can now re-install the AF Unit, but not the back-plate at this time.

Note – Prior to installing the AF Board, there is also a modification discussed that you may consider that can help reduce the sensitivity of the Modulator Balance adjustment, thus facilitate in achieving better carrier suppression, see Paragraph 5.3.

3.3.2.2 Sub SSB Product Detector Direct Interface

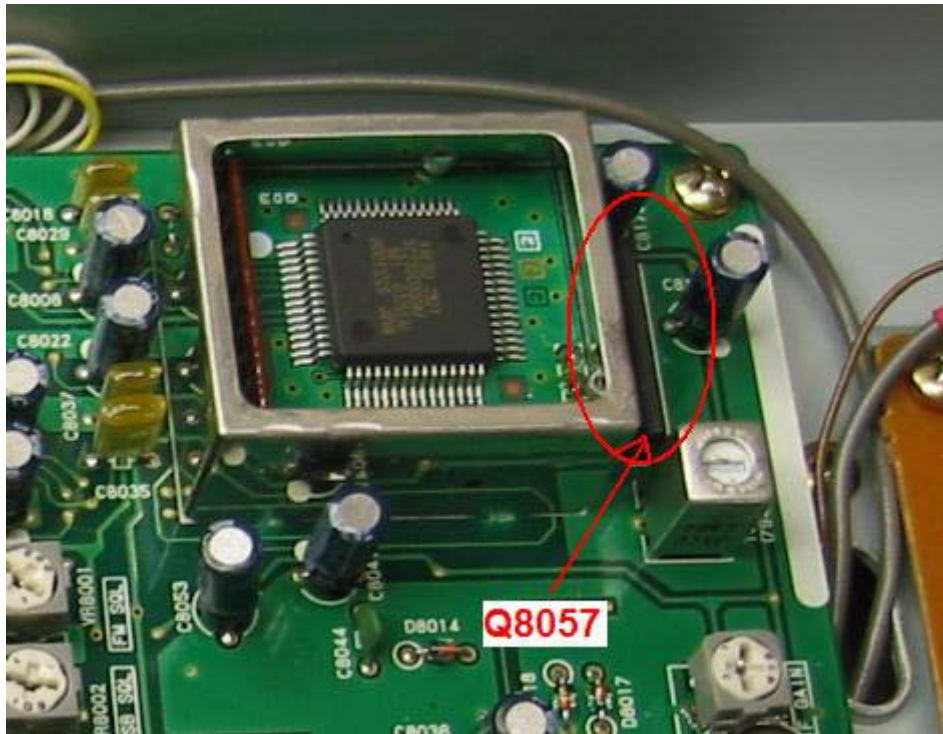
The next SSB Product Detector that needs to be tapped resides on the RX-2 Unit which is mounted on the same side of the radio as the TCXO. Again, you will have to remove the board from the radio, as we will be accessing the backside of the board.



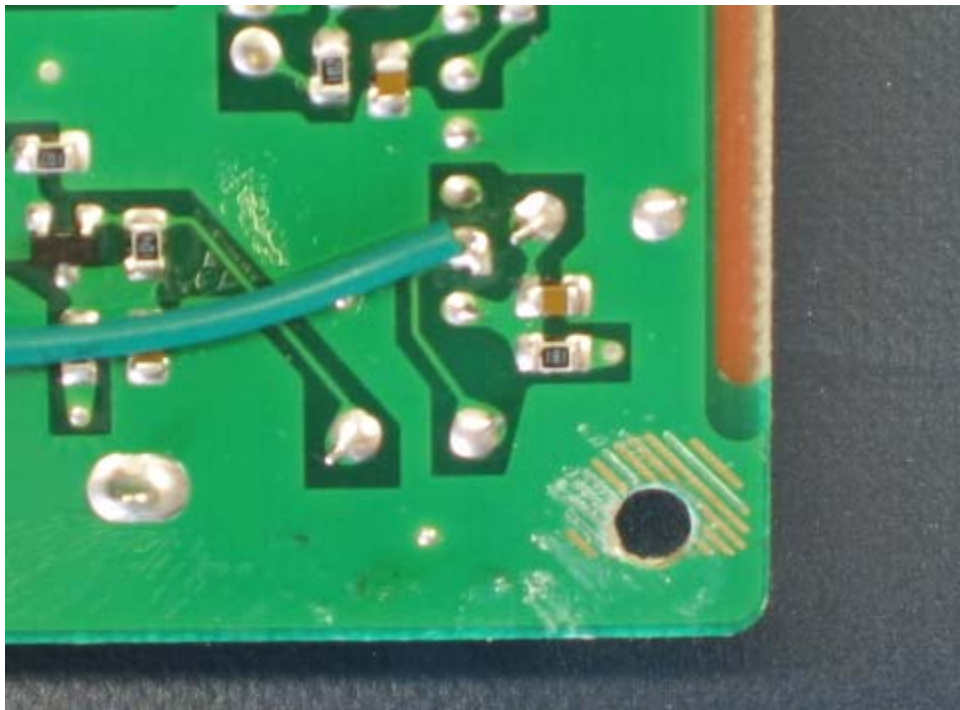
Schematic 4: Sub Receiver SSB Product Detector (Lower Right of the RX-2 Unit Schematic)

As can be seen in Schematic 4, the Sub Receiver uses the same IC package, uPC1037H, as does the Main Receiver to demodulate the SSB signal. In this circuit, the Receive IF is presented to Pin 5 of Q8057, the DDS Carrier is presented to Pin 7, and the subsequent target audio is sent out on Pin 2. Again, pin 2 is what we are interested in.

Looking at Picture 17 you will find Q8057 circled in red, and in Picture 18 you will see where the tap needs to take place. Again, I used a green-jacketed wire for this, as I did not have any other colored jackets to work with. In doing this, I kept both the Main & Sub SSB signals on the same color.



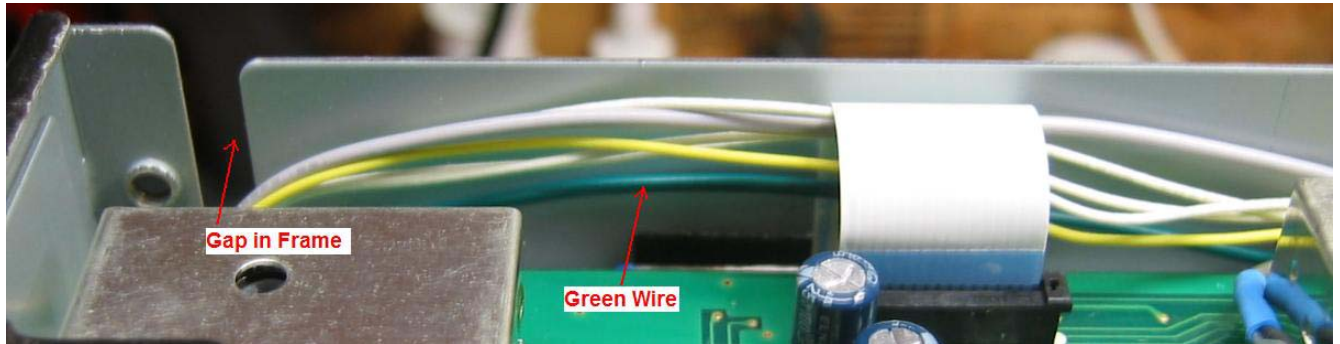
Picture 17: Q8057 (Bottom Left of RX-2 Unit)



Picture 18: Pin Location on backside of RX-2 Unit

Once we have tapped Pin 2 of the Product Detector Q8057, we need to route the wire towards the bottom side of the radio, so that we can interface into a mounted RCA Jack. The routing of the green wire is

pretty simple. Looking at Picture 19, you can see that the wire was routed along the bottom edge of the board (in relation to a radio resting position), following another bundle of signal leads. It is taken to the bottom of the radio via a gap in the frame of the radio. As with the taps for the Main AF Unit, be sure that your wire is clipped long enough to make it to its intended destination. Once the tap has been made, you can remount the RX-2 Unit.

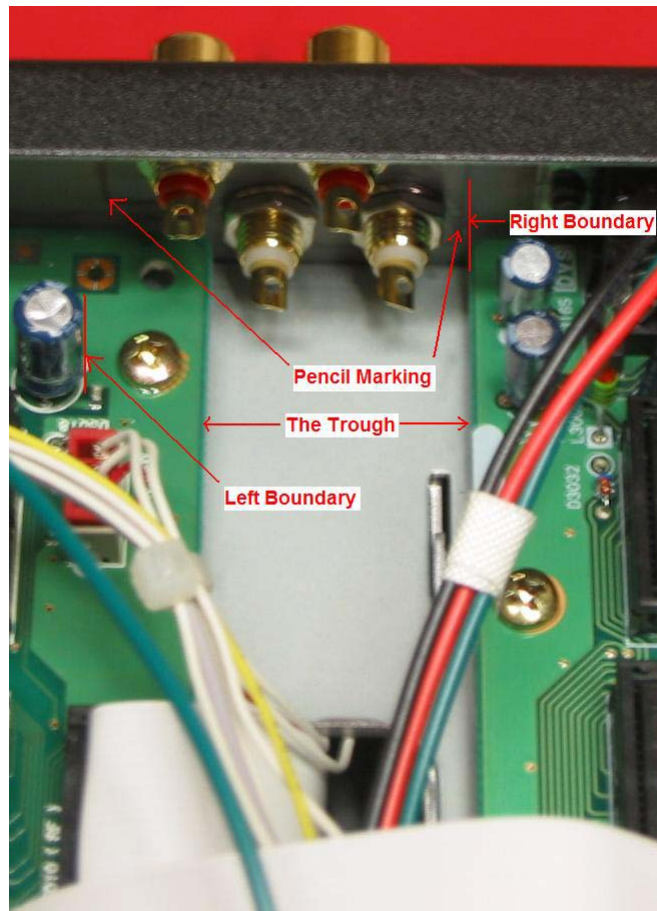


Picture 19: Signal Lead Routing for Sub-Receive Tap

3.4 RCA Jack Installation

Well, now that we have completed all of the circuit board modifications, and have 4 wires hanging about, where do we terminate them? I terminated the signal leads to 4 mounted female RCA jacks on the back plate of the radio, strategically placed so that they and their associated coupling capacitors would not physically interfere with any other component in the general area.

In order to mount 4ea Female RCA Jacks, we need to find a spot on the back-plate where there is enough room such that the Jacks, with the RCA Male Plugs mated, will not interfere with each other. By the same token, there should be enough room for the coupling capacitors to be mounted on the backside of the Female RCA Jacks, such that the size of the capacitors does not interfere with any of the other components within the radio. Fortunately for us there is such a spot on the back-plate of the radio. Looking at Picture 20, we can see that there is an actual trough formed between the AF Unit, and the Local Control Unit. This trough, along with the natural bow in the ribbon cables that bridge between the two boards, provides us enough room to mount the Female RCA Jacks, and not worry about the coupling capacitors interfering with any of the other components in the area.

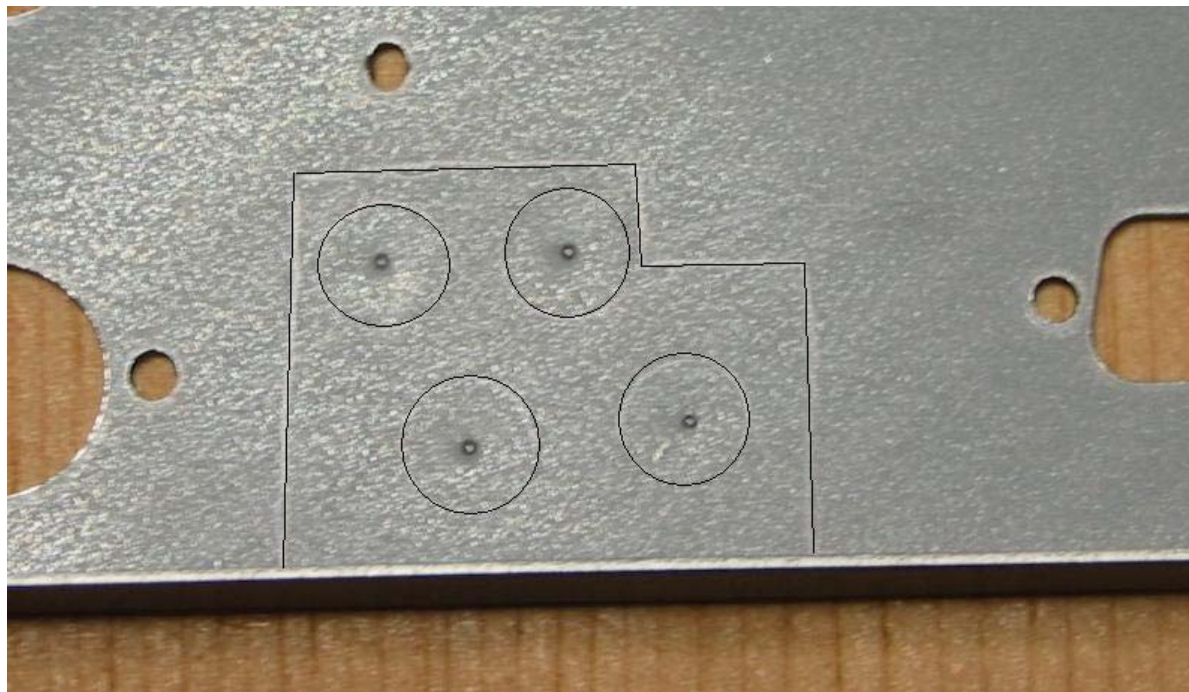


Picture 20: Mounted Female RCA Jacks

Since we took the back-plate off during our work on the AF Unit, in order to mark where we need to drill our holes for the Female RCA Jacks, we need to re-seat the back-plate to the radio, but not fully secure it to the radio; use only a couple of screws that were used to hold the plate to the frame of the radio.

***NOTE** – When seating the back-plate, it may be easier if you loosen the screws to the AF Unit so that it has a little play in movement. This will help you to align all of the I/O connectors on the AF Unit with their corresponding holes in the back-plate as you attempt to temporarily tack the back-plate back to the frame of the radio.*

Also looking at Picture 20, you can see how the left & right boundary for mounting the RCA Jacks was derived, as well as pencil markings on the back-plate. Using the same boundaries and a pencil, draw the boundaries on the back-plate. Your drawing should look something like Picture 21. I have taken the liberty of drawing over the pencil markings within the picture so that it is easier to distinguish the overall shape.



Picture 21: Back-Plate Marking

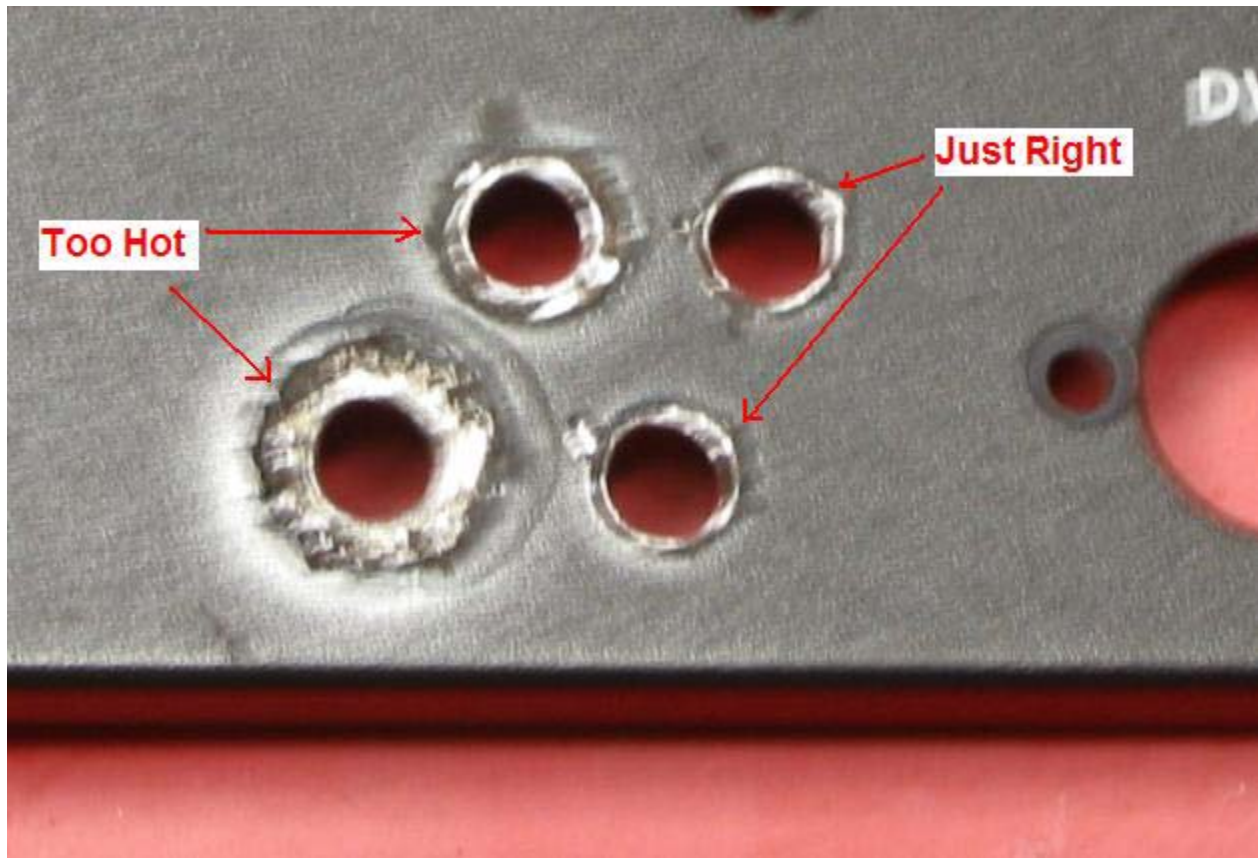
Once completed with marking up the boundaries on the back-plate, you need to remove the back-plate and lay it out as in Picture 21. Next, we need to secure our Female RCA Jacks and Male RCA Plugs and mate them together in an opposite manner. What this means is to insert the Male RCA Plugs into the backside of the Female RCA Jacks. This will allow each assembly to stand up on their own. We then arrange the RCA assemblies within our marked boundaries, such that neither of the assemblies is touching each other, yet still reside within the marked boundaries. Picture 22 gives a good visual of this procedure.



Picture 22: RCA Jack Arrangement

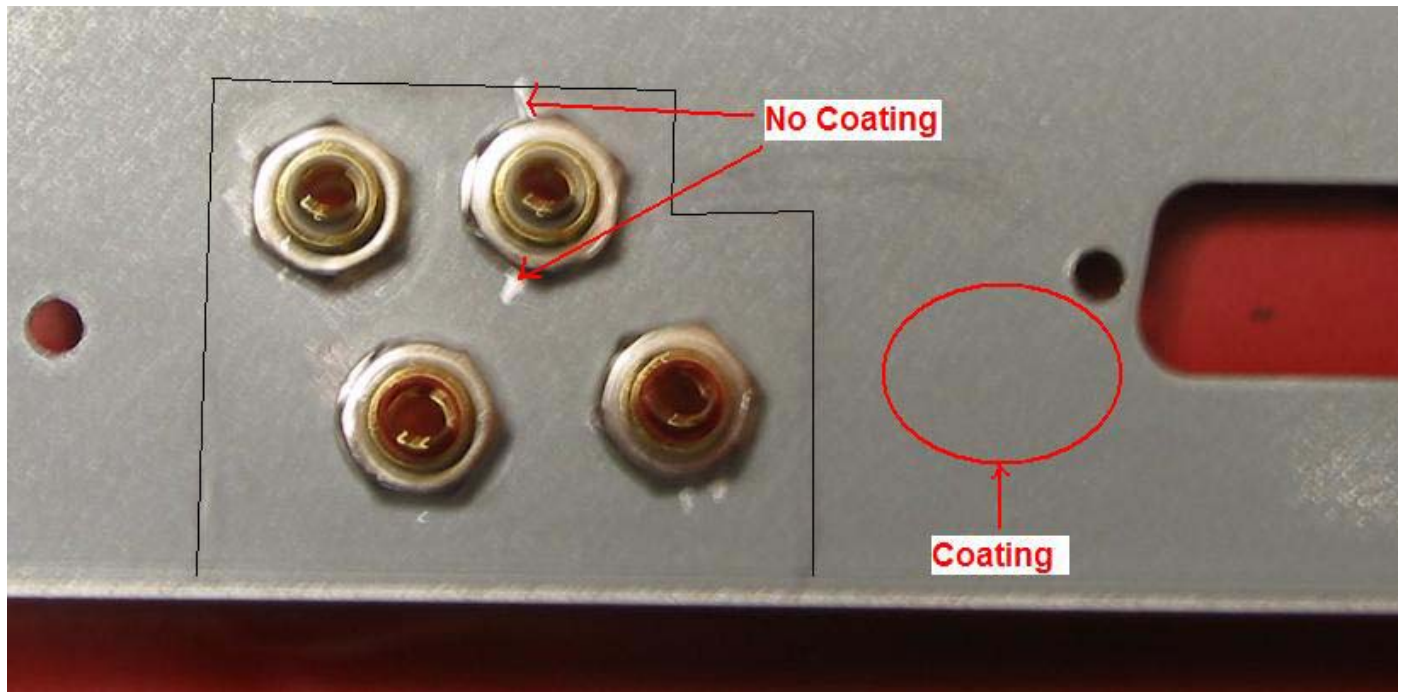
When they have all been arranged such that they meet the proper criteria, using a pencil, mark a small portion of where each RCA assembly sits. Now you can pull them all off, and using one assembly, mate it with each individual mark and draw the complete circumference of the assembly. Next, using a small nail or center punch, locate and place the nail or center punch in the center for each circle, and tap the nail with a hammer. This should produce an indentation in the metal back-plate of the radio, see Picture 21. This indentation is important. It will ensure that when you lay the drill bit onto the plate it will not walk off from center, yet dig right into the plate where you want it.

The threaded portion of the Female RCA Jacks are $\frac{1}{4}$ " in diameter, so this is the largest bit that we need. But if you have ever drilled holes into a piece of metal, then you understand that you need to start small and work your way to the target diameter. So you should start with a small bit, drill your hole, and then move on to the next bit size, and keep doing this in succession until you achieve a $\frac{1}{4}$ " hole. Or once you have your small starter hole completed, you could use a Unibit and in one continuous action, create a $\frac{1}{4}$ " hole. Although, in both cases, I strongly suggest that you use some kind of cutting oil, or some other oil to help the cutting action, and to keep the heat down. It should be noted that when using individual drill bits, attempting to prematurely skip to a larger diameter, or operating the drill at a real high rate of speed (RPM), you will either dull the bit and or temper the metal plate. Tempering the metal plate will generate a lot of heat, and do not forget that you have several coats of paint on the other side of this plate. What will happen is the paint will bubble up and chip away, as can be seen in Picture 23.



Picture 23: Tempering the Back-Plate

Once the holes have been drilled, you need to clean up your work. Taking your dremel tool with a sanding stone, or hand-file, you need to round the inside edges of your holes, knock off any metal filings that may still be attached, and remove a portion of the protective coating that is on the back side of the back-plate so that we get a good ground for the RCA Jacks. And if you tempered the paint on the front side of the back-plate, like I did on a couple of the holes, you will need to sand them down so the RCA Jack will mount flush to the plate. Once completed, you can now mount the Female RCA Jacks to the back-plate of the radio. As can be seen in Picture 24, the ground rings that came with your Female RCA Jacks are not used, as the locknut & RCA Jack itself provide the grounding by coming into contact with the metal back-plate. Also, notice the center pins, they are facing down in relation to our back-plate placement. This will make it easier to secure the coupling capacitors.



Picture 24: Mounted Female RCA Jacks

Now that you are finished with the installation of the Female RCA Jacks, you can now mount the back-plate back on the Mark V. As was mentioned earlier, it would be wise to loosen the screws that hold the AF Unit to the frame of the radio. This will allow some movement of the board as you are attempting to align the holes in the back-plate to the I/O connectors of the AF Unit. Once the back-plate has been mounted, and all of the I/O connectors have been re-secured to the back-plate, then you can re-tighten the screws holding down the AF Unit.

This should conclude all of the board modifications and back-plate construction work necessary to implement ESSB for the Mark V. The next paragraph will show you how to mount the coupling capacitors and attach your signal leads.

3.4.1 Hooking up our new Signal Leads

Now that we have installed the RCA Jacks to the back-plate of the radio, re-mounted the plate and all of the circuit boards, we need to terminate our signal leads.

If you remember, the Black Wire is the transmit audio, the red is the AM receive, and the two greens are our SSB receive. Before we actually hook-up our signal leads to the RCA Jacks, we need to make a determination as to what RCA Jack should have what signal terminated on it. Looking at Figure 2 you will see how mine is setup. The figure is presented as though you are looking directly at the back of the radio with the radio in the upright position.

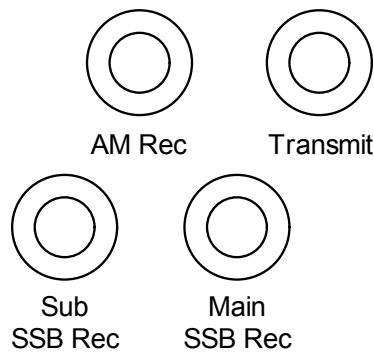
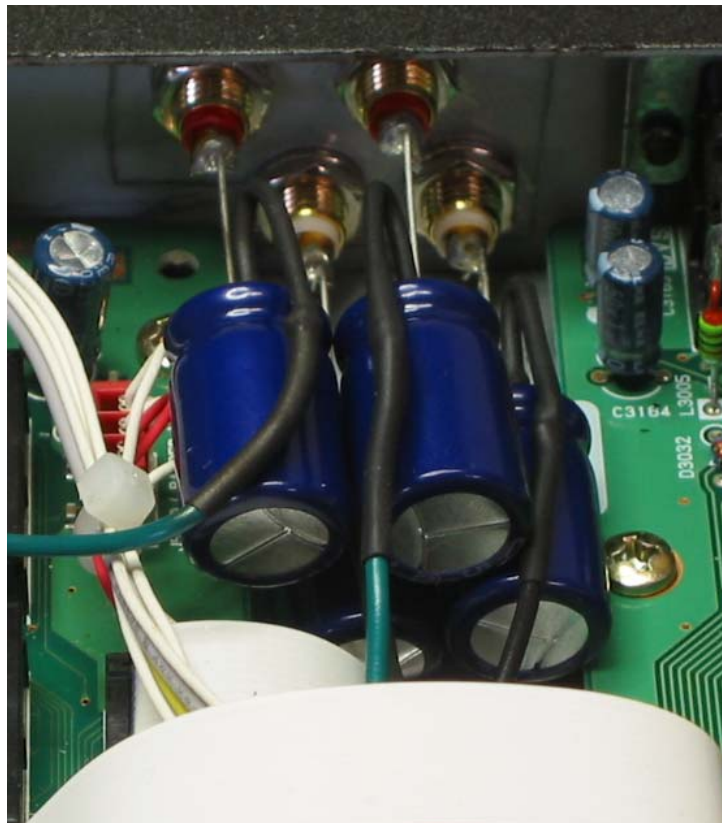


Figure 2: Signal Locations

There is no rhyme or reason as to why I designated what signal should be routed to which Jack. The only thing to remember here is that once you zero in on your own routing preference, record it.

As can be seen in Picture 20, the center pins of the RCA Jacks are facing up, somewhat. This will help facilitate mounting your coupling capacitors to the Female RCA jacks. Using the appropriate diameter heat shrink tubing, and avoiding any excess in either signal wire or capacitor leads, solder the signal wire to the capacitor and the capacitor to the center pin of your target Female RCA Jack. Do this for each of your signal wires. When completed, your work should look something like what is depicted in Picture 25.



Picture 25: Signal Wires Mounted to Female RCA Jacks

Once you have all of your signal leads connected, finish putting your Mark V back together. Next we will go over interfacing audio equipment into the BM.

3.5 Interfacing to the BM

When referencing professional audio equipment, it is understood that their audio output level is considered to be "Line Level" or +4dbu. However, the BM expects a signal level somewhere around 20db down from line level, so we do need to step the signal level down coming from our audio gear somewhat. Although, our biggest concern should be preventing any RFI or grounding issues from affecting our transmitted audio. We can ensure that we limit if not completely stop this from happening by using a 1:1 line-input transformer to go from our audio rack to the BM. The Jensen JT-11P-1 comes in handy for this application.

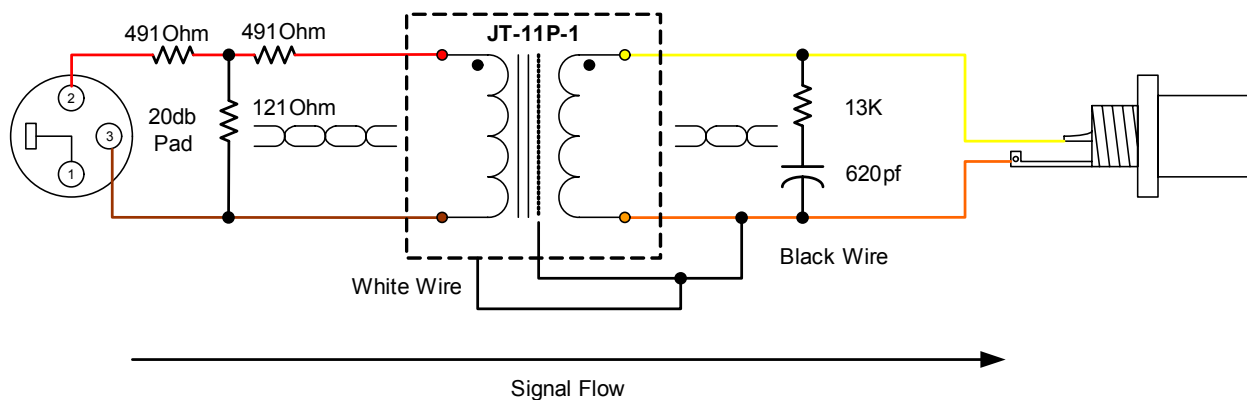


Figure 3: Jensen JT-11P-1 configuration for Interfacing to the BM

Looking at Figure 3, we can see that there is a 20db pad going into the transformer, and an RC network present on the output of the transformer. The 20db pad will help in our efforts to step the line level down such that we do not overdrive the BM when attempting to keep our audio gear at unity gain. The RC Network on the output of the transformer should be used to prevent transients and proper phasing. Also, Figure 3 is drawn from the perspective that JT-11P-1 will be mounted inside of an aluminum project box. When interfacing the XLR to your rack gear, the cable shield needs to be left floating at the rack gear end, see Figure 4. In other words do not terminate the Canare shield to pin 1 of the XLR connector that plugs into the audio rack gear.

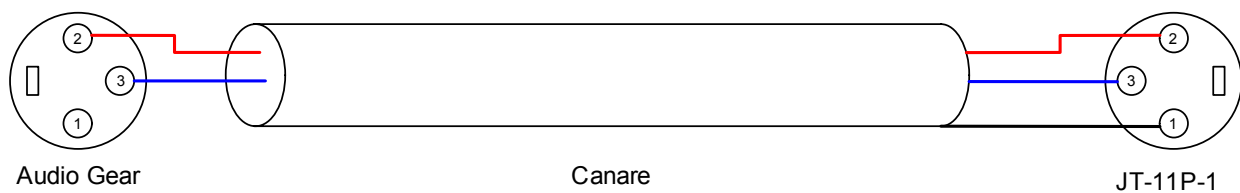


Figure 4: XLR Cable for Audio Gear to BM

Connecting the shield to pin 1 at the rack gear end will create a ground loop, thus cause hum and quite possibly RFI problems. A standard RCA to RCA type cable will have to be built in order to connect this arrangement to your Female RCA Jack that is connected to your BM, see Figure 5.

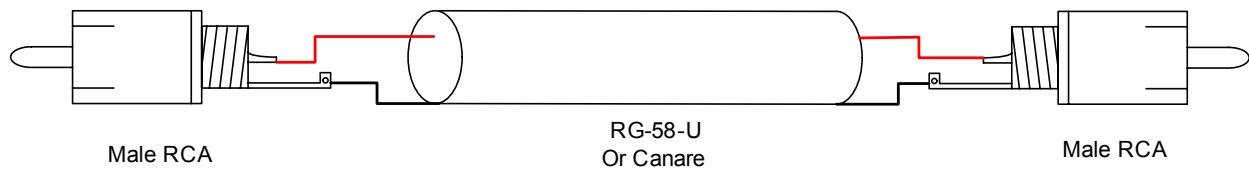


Figure 5: RCA to RCA Cable

One last comment about building this into an aluminum or metal project box, you will need to place 4 rubber feet on the bottom of the box. This will help keep the box isolated from any other piece of equipment or metal object if placed on top of such.

***NOTE** – Although the above transformer has been tested, and will work perfectly in the configuration depicted, I should mention that there is another Jensen Transformer that should work just as well with less padding required. The JT-10KB-D (4:1) should work perfectly in the above application, and will only need about 6-8db of pad to step the audio level down enough to keep from over driving your BM while maintaining your rack gear at unity gain. You may not even have to place any additional padding in-line, yet use a mixer to step the audio level down, see Paragraph 6.2.*

3.6 Various Audio Cables for our Receive

Now that we have 4 RCA jacks mounted on the back plate of the radio, what kind of cables do we need in order to interface those jacks with our external Microphone Pre-amp. Well, the answer is easy. We need to build some RCA to XLR cables. Now this may sound easy, but one must keep in mind that XLR is synonymous with a balanced cable, and RCA is synonymous with an unbalanced cable. So the cables that we need to build must take an unbalanced receive output from the radio and feed a balanced input, our MIC pre-amp.

There are two methods that can be used to do this task, the quick and dirty, and the elaborate method. Either method will work, but it should be noted that the quick and dirty does lend itself more so to possible EMI & RFI problems than the elaborate. This is not to say that the quick and dirty method will absolutely create these issues, but that it has the possibility to do so. If using the quick and dirty method you run into EMI and or RFI problems, you will probably need to invest the money into the elaborate method.

3.6.1 Quick and Dirty

What I have termed as the “quick and dirty” is basically wiring the physical interfaces (RCA & XLR) such that they will pass a signal from one physical format to another without the need for isolation transformers or other passive devices. This is basically a direct shot from the radio to your audio gear.

Earlier I said that the output of the RCA jacks we just installed were unbalanced. This means that we have a +signal lead coming from our Product Detector and a ground lead which is our RCA jack casing mounted to the metal back plate. In a balanced environment there is a +signal, -signal (audio return), and a shield ground. When directly interfacing the unbalanced world to the balanced world, we will build the below cable.

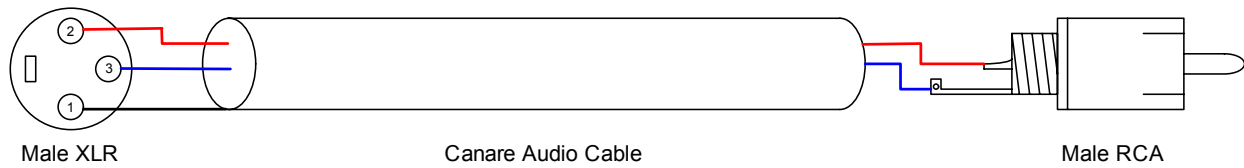


Figure 6: XLR to RCA Cable

It should be noted that Canare wire is built differently than your standard audio cable. You still have a copper mesh shield, but each separate signal lead uses a twisted pair. When you peel back the cable jacket and ground shield, you will find two white wires in a twisted pair configuration, as well as two blue wires in the same configuration. The twist in the wire provides protection against induction of foreign signals into the signal being carried. When soldering these wires to either an XLR or RCA connector, keep the integrity of each pair intact. In other words, strip the tips of the two white wires, twist and solder them together first, then solder the pair to your designated pin on either connector. Do the same with the blue wires. In order to keep consistency when building audio cables, choose a color code and stick to it!! My choice has been Blue is +signal, White is – Signal. There is no rhyme or reason as to why I picked this, only that I stick to it thus avoiding the creation of out of phase audio cables.

If you do not want to feed the balanced input of your audio equipment, instead choosing to feed your audio equipment through its unbalanced connector, then the cable depicted in Figure 7 is what you need to build. Although, it should be noted that most audio equipment that have tip ring jacks are actually balanced inputs. Look closely on the labeling near or around the connector and see if it shows a diagram depicting a +signal, –signal, and chassis ground. If it does, then more than likely it is a balanced input. If this is the case, then substitute the XLR shown in Figure 6 with a tip ring connector. Keep in mind, tip is +signal, ring is –signal, and the rest is chassis ground. If your audio equipment truly offers an unbalanced input then you can build the below cable to interface your receive audio into your MIC pre-amp.

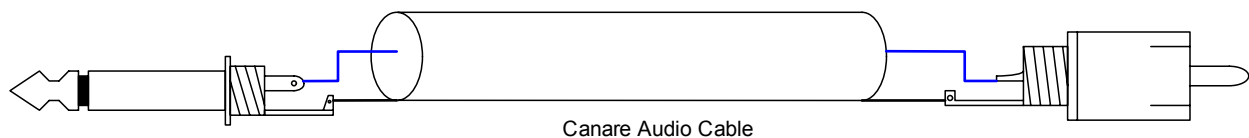


Figure 7: Unbalanced Rack Gear to Unbalanced RCA

3.6.2 Elaborate

The elaborate method involves the purchase of a Jensen JT-11-FLCF (1:1) transformer. The object is to isolate the ground planes between the radio and the audio equipment, while at the same time inductively coupling the target signal across a transformer. Below is a figure depicting how the transformer should be deployed, given that you wire directly to the transformer. It should be noted that the usage of Canare wire is recommended in the construction. I would also connect the shield of the cable to Pin 1 of the XLR, then connect the shield between the two cable sets, yet do not terminate the shield on the Male RCA Plug. This will extend the ground from your audio equipment up to the radio, but not terminated to the radio. This will help in preventing EMI & RFI from conducting itself onto the signal leads.

Now this can be built into a small project box with XLR connectors for both the input and output, see Figure 8. If you do this, you will need to make a cable such as depicted in Figure 6. The plus side to building this into a project box, using XLR connectors, is that it will protect the transformer leads from being pulled out of the windings. Also, given a large enough project box, this arrangement could be used

for placing multiple transformers into one box. When building the box, and if you use an aluminum project box, as is shown in Figure 8, ground both XLR Pin 1's to the project box. When using the cable from Figure 6, this will extend the ground from your audio rack all the way up to the radio, but not terminated to the radio. Again, this will help prevent EMI & RFI from conducting itself onto the signal leads. And remember, place 4 rubber feet on the bottom of the box to keep it isolated from any equipment that it may be placed on top of.

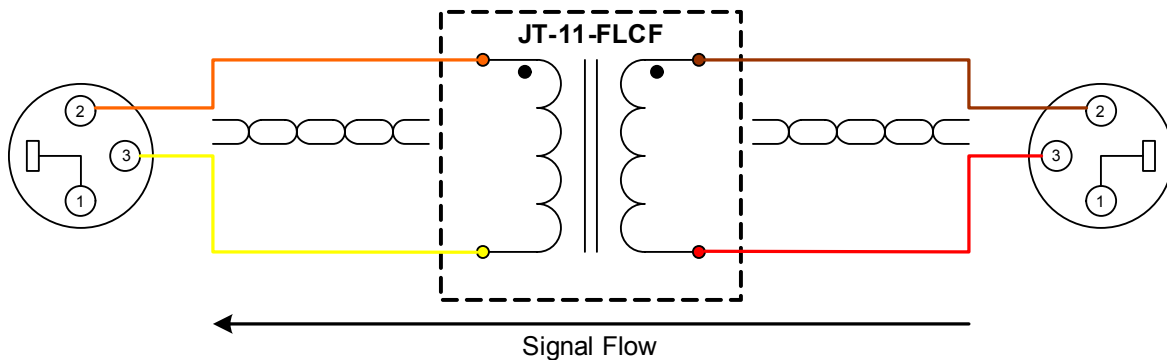


Figure 8: Jensen JT-11-FLCF Receive XLR Balanced to XLR Unbalanced

One final note before we move on, if you feel that your station, to include your Audio Rack, and computer are grounded properly, then I would try the “quick and dirty” method first. If you run into audio hum problems, then I would purchase a JT-11-FLCF.

4. Menu Settings

Now that we have completed the installation of the filters, tapped all of the circuits, and brought our audio signals out to the back of the radio via RCA Jacks, we now need to make some menu adjustments in order to achieve our ultimate goal, 6Khz audio.

In order to gain access to any of the 9-x menus, you need to depress and hold the “Fast” and “Lock” button of the radio while powering it up. Once the dial lights up, you can release the two buttons. Now by accessing the menu system as you normally would, you should be able to scroll to the “9-x” menus. A word of caution, any playing around in these menus with out knowledge of what you are doing has the potential of at least mis-aligning your radio such that you will not be able to use it. I strongly suggest that if you feel compelled to scroll through each “9-x” menu, that you record each sub-menu setting as you come to them, and keep the recording in your manual. Accessing the “9-x” sub-menus is performed in the same fashion as accessing any of the other sub-menus.

4.1 EDSP Menus

We have several menus within our Mark V that govern the usage of EDSP, Enhanced Digital Signal Processing. However, we have only made modifications for our analog filters, not the DSP engine, and thus need to stick with analog in order to utilize those filters to their fullest potential. With that being said, then there must be some menus that need to be changed in order to ensure that we operate in analog only. Make the following changes to the designated menus:

- **Menu 0-9 edSP:** This must be turned off, thus disabling EDSP all together

- **Menu 2-9 notchH:** This menu selects how the Notch filter works. Set it for “Select”. This will enable the analog notch to work during analog operations of the Mark V, and both analog and DSP when menu 0-9 (EDSP) is turned on.
- **Menu 4-4 tr-EdSP:** This selects one of 4 preset equalization patterns for your voice while using EDSP. Since we have external audio processing gear, I think we can do a much better job than these limited settings. I highly suggest that this menu be turned off, even when using EDSP.
- **Menu 5-9 t-FiL:** This selects either the 2.4Khz or 6.0Khz digital band-pass filter for transmission. Since we have EDSP turned off, selecting either of these settings has no affect to our analog transmission. However, I would suggest setting it for 6Khz just in case you revert back to EDSP.
- **Menu 7-7dSP-ndn:** This selects the DSP audio band-pass for transmission. Since we have EDSP turned off, their setting will not affect our analog transmission. I would suggest keeping these at their widest settings for transmit only (100Hz-3100Hz) and turned off for receive.

The biggest thing to remember here is Menu 0-9, it must be turned off! The remainder of the menus listed can maintain their current settings, or be updated as per the suggestions. Doing this will not affect your analog transmission, and will make it a lot easier to quickly utilize the EDSP capabilities of the Mark V if you choose to do so in the future, see Paragraph 6.4.

4.2 Carrier Offset

When our radio first came from the factory, its carrier offset was adjusted for the pass-band of the stock OEM filters. So it stands to reason that since we have replaced those filters with a wider pass-band, that we need to adjust the carrier offset of the radio in order achieve the full potential of that pass-band.

4.2.1 Standard Carrier Offset Menu 8-9

The first menu you should access is 8-9, which is the standard Carrier Offset menu. Make the following adjustments:

- r-LsbcAr – 0.000
- t-LsbcAr – 0.000
- Proc-LSb – 0.000
- r-uSbcAr – 0.000
- t-uSbcAr – 0.000
- Proc-uSb – 0.000
- t-A3-cAr – Leave at factory setting
- Proc-A3 – Leave at factory setting

This will completely zero out any carrier offsets that you originally had within this menu, minus those for AM. The service manual refers to this menu as being the carrier offset fine adjustment. So this must mean that there is a course adjustment lurking in the radio somewhere. You will find that course carrier offset adjustment within menu 9-0.

4.2.2 Hidden Carrier Offset Menu 9-0

This is where the real work will take place. By adjusting these sub-menus in the following manner, you will place the lower edge of your new band-pass right on dial frequency for receive, and the transmit

band-pass will be offset from dial frequency in accordance with your operating mode; LSB or USB. Bring up menu 9-0 and make the following adjustments within the sub-menus:

- r-LsbcAr – 1.500
- t-LsbcAr – 1.530
- Proc-LSb – 0.040
- r-uSbcAr – 1.500
- t-uSbcAr – 1.530
- Proc-uSb – 0.040
- t-A3-cAr – Leave at factory setting
- Proc-A3 – Leave at factory setting

What you have just done is set the carrier offset for each mode. The Mark V already has a built in analog carrier offset which translates into accommodating 2.6Khz of pass-band at a -6db point starting at or around 200Hz. By adjusting the USB/LSB transmit from your factory setting to +1.530, you have basically placed the -6db point of your new filters close to +30Hz from dial frequency.

I would not adjust the transmit carrier offset any closer than 1.520, as this will adjust your transmit band-pass such that your carrier intersect point with the filter skirts will be greater than the -30db point of your filter skirt. This will hurt the radio's ability to properly suppress the carrier frequency, as well as suppress the opposite sideband while transmitting audio in either LSB or USB. I know you want to get some lows in that audio, but anything less than 30Hz will do you no good anyways; read John Anning's website for a better understanding. Also, menu 9-0 is the course adjustment, so any deflection in this menu will move the carrier offset quite a bit. If you need to make small minor adjustments to your carrier offset frequency, you can accomplish this through menu 8-9.

4.3 Filter Settings

In order to obtain the most out of your newly installed filters, you will need to make some changes on how your filters are provisioned within the menu system. You can also use these menu changes to enhance an unmodified Mark V's capability.

4.3.1 Menu 5-0

Menu 5-0 selects the filters used when the NOR button is depressed on the front panel of your radio. Default setting straight from the factory had this set to "OFF", which routes the receive signal through the 2nd IF Filter and then to the 3rd IF Filter. When we had our OEM filters installed, this meant that the receive IF passed through the 2.6Khz 8.215Mhz filter first, then went to the Collins 2.3Khz 455Khz filter next. As you can imagine, this narrowed up the receive quite a bit, ultimately giving you about 2.4Khz of analog receive band-pass. We can open this up by utilizing either of the following settings:

1. **8.2 – 455:** According to the manual, this setting will pass the receive signal straight from the roofing filter to the OEM 6Khz AM filter of the Mark V. However, what really happens is that the signal is passed from the roofing filter, by-passing both the 2nd and 3rd IF filters, heading straight for the product detector. Basically this gives you a 6.5Khz-0-6.5Khz listening band-pass, for a total of 13Khz that straddles zero beat. In other words, you will be listening to both LSB & USB all at the same time. Now it should go without saying that this setting will allow any strong

signal within that pass-band to affect your AGC. If this is not a problem for you, and the band is quiet except for the target signal that you want to listen to, then this setting will ultimately give you the widest listening advantage; ~6.5Khz.

2. **445:** This setting will pass the receive signal from the roofing filter to your new 6Khz 8.215Mhz Inrad filter and then onto the OEM 6Khz AM filter. This should provide you roughly ~5.9Khz of listening band-pass, and will help limit strong signals beyond the 2nd IF from grabbing your AGC. It should also be noted that this setting does eliminate the unused sideband.

Now with this menu turned on, and provisioned for either of the above settings, the NAR1 button becomes defaulted for the following action:

- The receive signal will pass through the roofing filter onto the 2nd IF filter (6Khz Inrad), and then onto the 3rd IF filter (5.8Khz Collins), thus providing a little more selectivity as well as help with the AGC in regards to strong adjacent signals.

This pretty much sums up this menu setting and how you can better utilize it in conjunction with your new filters. It should also be mentioned that these settings could also be used for an unmodified Mark V to increase the listening band-pass.

4.3.2 Menu 5-1 & 5-5

Menus 5-1 & 5-5 are used to turn on your optional 2.0Khz IF slots. If you decided to re-utilize your OEM filters, as per Paragraph 3.1.1, or use the Inrad 2.8Khz filters then you will need to “enable” these menus. The benefit to doing this, in conjunction with suggested settings in Menu 5-0, is that this enables your “NAR2” button thus giving you one more available filter selection at your fingertips.

When you utilize these slots, with menu 5-1 & 5-5 enabled, and menu 5-0 setup as per the last paragraph, the NAR2 button can be used to select the filters in the optional 2.0Khz slots. So this allows you the following options:

- NOR – Wide receive dependant upon menu 5-0 setting
- NAR1 – Cascades your new IF filters, providing you a little better selectivity
- NAR2 – Usage of the filters installed in your 2.0Khz optional slots with even better selectivity

Even if you have not modified your Mark V for ESSB, and do have filters installed in your optional 2.0Khz slots, these settings will work in the same manner. Did you know that you have the ability to select either of two filter settings for your sub-receiver as well? Take a look at the next paragraph.

4.3.3 Filter Selections for the Sub-Receiver

Did you know that you could use similar filter arrangements for the sub-receiver as you do for the main receiver? This ability is something that goes unnoticed by most Mark V operators.

In order to activate any of the filter selections (NOR, NAR1/NAR2) for the sub-receiver, perform the below steps:

- Depress the “SUB” button – This should cause the dial frequency displayed for the sub-receiver to start flashing.
- Depress either NOR or NAR1 – This will select that filter setting for the sub-receiver.

When you depress the filter selection, the sub-receiver window should stop flashing and the filter selection should be engaged. That is how easy it is to implement. You should also know that when selecting NOR for the Sub Receiver, you are selecting a 12Khz-roofing filter that straddles across zero beat. So you will be hearing 6K-0-6K. The NAR1 will select the Murata 2.4Khz filter that resides on the RX2 Unit. Selecting NAR2 is no different than selecting the NAR1, 2.4Khz Murata filter.

***NOTE** – When monitoring your transmission from the main transmitter, using the transmit monitor circuit, the sub-receiver actually picks off the transmitted RF off of the Main, decodes it, and presents it to your headphones or speakers. To gain the widest monitoring band-pass possible use the above procedure to select the NOR filter setting for the sub-receiver, as well as the NOR for the Main.*

4.3.4 Menu 1-2 – IF Shift & Width

Another weapon in your arsenal when we talk about selectivity is Menu 1-2. This menu is used to set the tuning step size of both the IF Shift and Width knobs on the front panel. Its default setting is 10Hz, however, we want to change that to 20Hz. Doing this provides us the ability to cut the pass-band of our receiver, when the NOR button is selected, from roughly 5.8Khz down to about 3.5Khz in incremental steps using the IF Width Knob. With the default setting we would only be able to cut the pass-band down to roughly 4Khz. It should also be noted that using both the IF Shift and Width knobs in conjunction with each other, you can actually narrow the receive pass-band, and then move that adjusted pass-band as a whole, on top of the target signal thus cutting out a lot of QRM. This works great when trying to dig out a weak signal with some strong adjacent signals on either side. Using SpectraPlus to watch your received audio, you can play with the IF Shift and Width and actually watch your adjustments in real-time.

4.4 Increasing your Main VFO Display Accuracy

What would you say if I showed you how you could get 0.625Hz accuracy from your Main VFO Display? I would say, “Ok, I am game, how do you do it?” Well let’s make a few menu adjustments and then I will explain how it all works.

Set the bulleted menus as follows:

- **Menu 1-0 diAL SPd:** According to the manual, this sets the tuning rate of the VFO dial when the “Fast” button is depressed. In reality it sets the normal tuning rate of the VFO dial in non-Fast mode. Set this to 2 as opposed to 4. This will allow a tuning rating of 312Hz per 1 revolution of the VFO dial. When the “Fast” button is engaged, the tuning rate will jump to 3.12Khz per revolution.
- **Menu 1-3 A-StEP:** This is the tuning steps taken when you spin your Main VFO Dial. Set it for 0.625Hz.
- **Menu 1-5 b-StEP:** This is the tuning steps taken when you spin your Sub VFO Dial. Set it for 0.625Hz.
- **Menu 1-9 cLAr-StP:** This is the tuning steps taken when you utilize the Clarifier knob. Set it for 0.625Hz.

- **Menu 3-1 diSP-rES:** This sets your tuning display resolution for both the Main & Sub Receiver. Set this for lowest setting of 10Hz.
- **Menu 3-2 EtS-SEL:** This sets which function has control over your expanded tuning scale above the main frequency display. Set this for Display.

Now looking at your display, while rotating your main VFO dial, you will see orange indicator lights on the expanded tuning scale turning on and off as you rotate the dial. So what does this do for you? Well, it actually provides you the ability to accurately adjust the main VFO dial at 0.625Hz steps.

In Menu 1-3, we set the tuning step of the main VFO dial to 0.625Hz per step. Yet the lowest granularity we could set the main frequency display was 10Hz. So there is definitely something missing here. What is missing, and not quite spelled out in the Mark V Operators Manual, is the ability to use the expanded tuning scale to increase that granularity to 0.625Hz. If we take a look at the expanded tuning scale on our radio's display, starting from the left of the center marker, we can count 15 yellow dots across the scale. If we were to slowly move the main VFO dial, we will see orange indicator lights spread across the expanded tuning scale underneath each yellow dot. However, we do not see any change in our dial frequency. It is not until we have moved the main VFO dial such that the orange arrowhead appears at either end of the expanded tuning scale and then disappears that we actually see our dial frequency change. So what is happening here? Considering that the display only goes down to 10Hz, there are 15 yellow dots that get individually highlighted as we slowly turn the VFO dial, and once we reach the end of the expanded tuning scale our dial frequency changes by 10Hz, this must mean that each yellow dot is equal to 0.625Hz; $9.375\text{Hz}/15 \text{ dots} = 0.625\text{Hz}$ per dot. Another way of looking at this is that there are 16 dots, 15 yellow and then the one center marker; $10\text{Hz}/16=0.625\text{Hz}$. If this is the case, then the expanded tuning scale can be read per Figure 9.

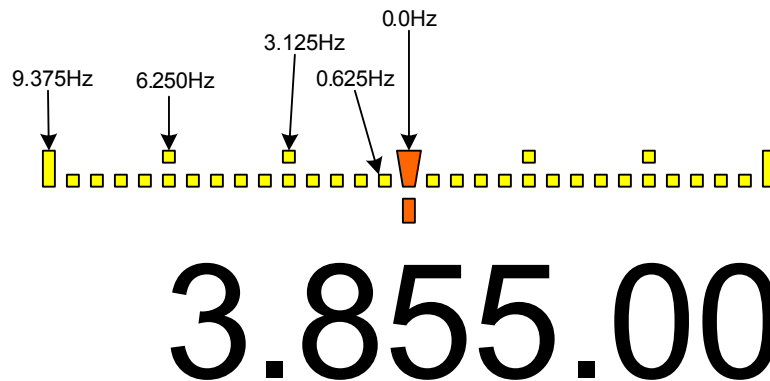


Figure 9: Expanded Tuning Scale

The above figure is indicative of a Mark V that is tuned dead on to the dial frequency displayed. Figure 10 shows a Mark V that it is actually tuned to 3.855.003125Mhz

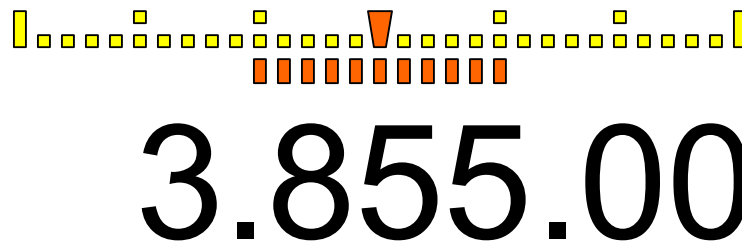


Figure 10: Expanded Tuning Scale as part of Dial Frequency

Now that you know this, you can adjust your Mark V more accurately when attempting to zero beat another station. It should also be mentioned that this ability to accurately tune to any given frequency is imperative if you decide to align your Mark V's TCXO, or tune the Receive Transformers.

5. General Alignments

Now that we have replaced the OEM filters, and have made all of the necessary menu changes, we need to perform some alignments in order to fully realize the benefits of our work thus far. Although, it should be noted that a lot of these alignments can be done to an unmodified Mark V as well.

5.1 TCXO Alignment

One of the more critical alignments that you could make is that of your TCXO. However, one must keep in mind what the acronym TCXO means, Temperature Controlled Crystal Oscillator. What this means is that before we can affect any changes on the TCXO, we need to allow the radio to warm up for at least 2 hours. After a 2 hour warm up period we can be reasonably assured that the TCXO has reached a stable operating temperature, thus its drift is minimal, and it has settled on a frequency somewhere between its upper and lower bound limits.

I apologize to those who are already aware of how a clock is rated and actually operates. However, I feel compelled to provide some background information for those who do not possess that knowledge. Understanding how the clock within our radio works definitely helps understanding why we will adjust it in a particular manner. If you already have a good understanding of this, then skip over to the appropriate alignment sub-paragraph.

All temperature-controlled clocks are rated per their drift over time. Their drift is expressed in terms of +/- Xppm, or in easier terms, plus or minus some level of part per million. So what does this really mean? Well, given a positive movement of 1ppm for a center frequency of 1 million hertz, our total movement from center frequency would be +1hz. A movement of -1ppm with the same center frequency would equate to a movement of -1hz from center frequency. So a +/-1ppm swing with the same center frequency would equate to a total possible movement of 2hz, 1hz in the positive direction and 1hz in the negative direction. The key here is to remember that the clock will drift within the upper and a lower bound during it's warm up period. When first powered up, it could be sitting on some frequency within the lower bound, and drift all the way to some frequency within the upper bound. But once it has reached its stable operating temperature, it will stabilize on a frequency somewhere within the upper and lower bounds.

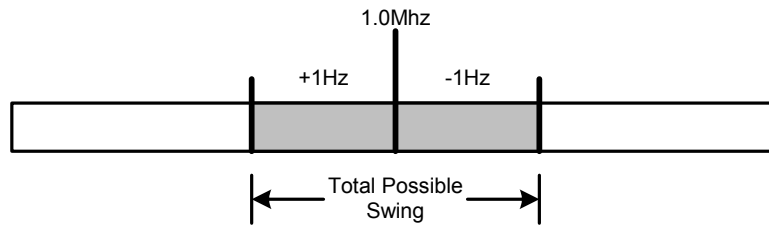


Figure 11: +/-1ppm

So how does this affect us Amateur Radio Operators. Looking at the above figure, how parts per million translates to us is rather easy to figure out given that we operate in the pre-defined Ham bands that are in the million hertz range; 3.50000Mhz, 14.25000Mhz...etc. Given a Mark V equipped with the standard TCXO-4 clock (+/-0.5ppm) operating at a dial frequency of 14.25000Mhz, the positive operating upper bound would equate to +7.125Hz from center frequency. So if we have the standard clock in our radio that has a rating of +/-0.5ppm, and we are operating at a dial frequency of 14.25000Mhz, we could actually be operating between 14.250007125Mhz to 14.249992875Mhz, which is within the upper and lower bounds of our TCXO-4 +/-0.5ppm clock. So let's look at this from another frequency's perspective. Given the same clock rating, but operating at a dial frequency of 3.50000Mhz, our actual operating frequency could be +/-1.75Hz from dial frequency, which is also still within the upper and lower bounds of our +/-0.5ppm clock. For a visual understanding of what this means, look at Figure 12. Both examples are on the same scale, and depict how a +/-0.5ppm clock can affect us dependant upon the frequency we operate on.

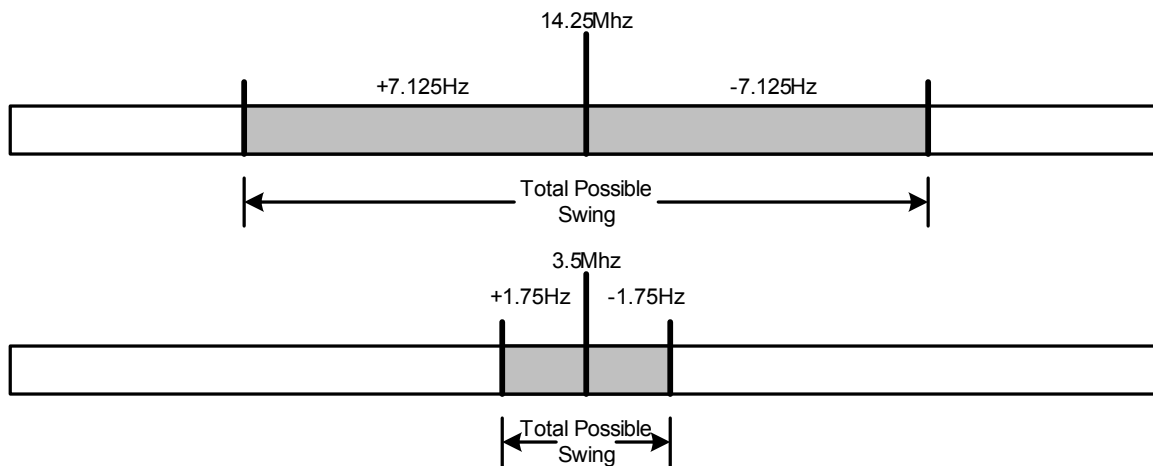


Figure 12: +/-0.5ppm of Drift at 14.25Mhz Vs 3.5Mhz

Now let's take this one more step further. Let us say that we have operator A who is using a Mark V equipped with a TCXO-4 clock that is operating at a +7Hz from a center frequency of 14.25000Mhz. Keep in mind that he/she is well within the +/-0.5ppm tolerance that we have been discussing. Now let's take operator B who is also using the same type of radio and clock and is actually operating at a -7Hz from center frequency of 14.25000Mhz. The difference between the two stations is 14Hz, yet both of their dials reflect 14.25000Mhz. In the ESSB world, given the low audio frequency range that a properly equipped radio can transmit and receive, each operator will be able to detect this difference, and swear that the other is off frequency. However, in reality, both of them are still within their +/-0.5ppm tolerances, and therefore both of them are on frequency per their radio. Makes for an interesting QSO.

So what can be done about this? Well, we could align our TCXO-4 thus ensuring that we are very close to the center frequency; given an analog circuit, you will never be dead on. We could purchase the upgraded clock (TCXO-6) for the Mark V, giving us a tighter tolerance of +/-0.25ppm. Or we can agree that one operator should zero beat the other, and then continue on with the QSO. In reality, most operators', at least those who are aware of how a clock's tolerances can affect our receive/transmit, will purchase the TCXO-6. Given a dial frequency of 14.250000Mhz, the TCXO-6 would narrow the possible operating range down to +/- 3.5625Hz from dial frequency, and +/-0.875Hz for 3.500000Mhz. Again, in reality, an even smaller percentage of those operators will ever align their TCXO-4 or TCXO-6. Opting more for just trusting its alignment straight from the factory. So with this in mind, aligning your TCXO is really a relative action. If you align yours, you will still run into people who have never touched theirs, thus someone has to zero beat the other. This can all be summed up in one statement, particularly when dealing with analog phone transmissions: A clock, whether the OEM that shipped with our radio or the upgraded version, is only there to get us within a ballpark frequency, the rest is up to our ears. So given all of this, you decide whether you want to align your clock or just use your VFO to zero beat. And if you decide to do the alignment, below are two methods that you can use dependant upon available equipment.

***NOTE** – It is imperative and assumed that you have implemented the menu changes as discussed in Paragraph 4.4, and that you are dead on frequency per your expanded tuning display before you attempt aligning your TCXO!!*

5.1.1 Alignment using Test Equipment

This procedure assumes that you have access to either a Frequency Counter, or a Spectrum Analyzer with a built in Frequency Counter, and a time standard such as a Cesium Beam clock. Now I would not expect the typical Amateur Operator to have access to a Cesium Beam clock, much less a Rubidium based clock. But just in case, here is how we can align either the standard or upgraded TCXO.

Set your equipment up as depicted in Figure 13.

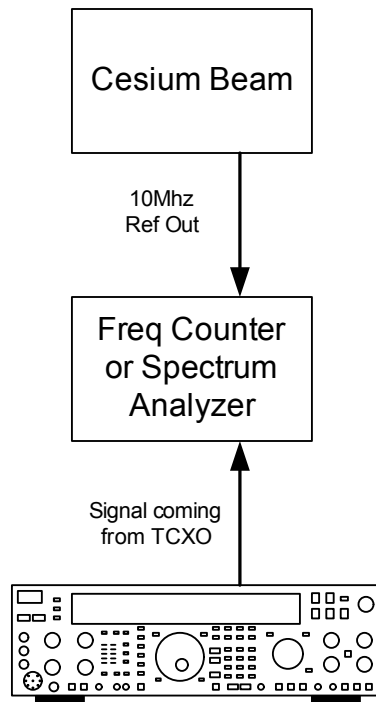


Figure 13:TCXO Alignment Test Equipment Setup

All time standard clocks should have a 10Mhz reference output, and most test equipment should have a 10Mhz reference input. Connect the 10Mhz out to the 10Mhz in using a 50ohm BNC type cable. Next, dependant upon if you are aligning the TCXO-4 or the upgraded TCXO-6, the test point that you need to clip onto in order to see the operating frequency of the clock will be different. Check the documentation that came with the TCXO-6 or the service manual if you have a TCXO-4. In any case, before you even begin the alignment, you definitely need to wait at least 2 hours from the time you powered up both the radio and the test equipment before you do anything else. If you remember, the acronym TCXO stands for Temperature Controlled Crystal Oscillator. In other words you need to wait not only for the radio's clock, but the test equipments clocks to stabilize their operating temperature as well. For grins, and to visually see what I am talking about, go ahead and monitor the TCXO's output during the warm-up period. You will be surprised on how much this stuff can actually drift before finally settling down to a stable frequency.

Once the equipment has been allowed sufficient enough time to warm up, watch the output of the TCXO and insure that it is stable. Regardless if you are aligning the TCXO-4 or the TCXO-6, you are looking for an output center frequency of 10.485760Mhz, or something within +/-5hz (TCXO-4) or +/-2.5Hz (TCXO-6). Using a plastic alignment tool, adjust the trimmer capacitor for your given TCXO such that you achieve the center frequency stated earlier. You should be able to achieve +/- 1Hz. Once you have achieved this, wait at least another 30 minutes and check it again. If it has walked off a little, re-align, wait 30 minutes, and then re-check. If it is rock steady, you are done.

NOTE – It should be mentioned that this test setup can also be used to align the 2nd Local Frequency (46.775Mhz) for the Sub-Receive, the 2nd Local Oscillator (62.240Mhz) on the Local Unit, as well as various other alignments as per the Service Manual.

5.1.2 Alignment using WWV

If you do not have access to the test equipment mentioned in the previous paragraph, do not fret. We still have access to a frequency standard signal, also known as WWV. The WWV stations are timed by a Cesium Beam clock, and therefore can be used as a frequency standard. One of the caveats to using WWV for this alignment is that you need to be able to ensure that you can receive a strong signal from the station. The other caveat is that the WWV station you want to use is the one that resides on 20Mhz.

So why do we want to use the 20Mhz WWV station, and not the 10Mhz or 5Mhz station? Remember the earlier explanation of how a clock can affect us depending upon our operating frequency? With that in mind, then it is fair to assume that at 20Mhz we will have more of an operating spread to work from, thus providing us the ability to more accurately align our TCXO.

You will not be using any test equipment in this procedure other than the radio itself. Again, prior to making any adjustments, you need to allow the radio to warm up for at least 2 hours. Once the Mark V has warmed up, adjust your VFO to the 20Mhz WWV station. This means using the setup in Paragraph 4.4, the only marker that should be lit on the expanded tuning scale when the radio is tuned to 20Mhz, is the center marker. If this is not the case, then you are not tuned properly. Next go to CW mode, you should hear a tone. This tone can be adjusted from 300 – 1050Hz at 50Hz intervals using your Pitch knob on the bottom right of the Mark V. By setting menu 3-5 to “A1 (CW) Pitch” you should see the current pitch setting in the carrier offset display on the bottom of your LED display. By rotating the Pitch Knob, set this to “C-600”, this should set the carrier offset such that you will hear a 600Hz tone when you zero beat a station within the center of your IF pass-band. Next turn on your “Spot” button on the lower right of the Mark V. This engages an internal oscillator that is set to the same pitch that you just set with your Pitch knob. Now there are two separate tones, one of them is generated due to the CW carrier offset that you have setup via the Pitch Knob. The other is the internal oscillator set to the same pitch. If your Mark V’s TCXO is aligned properly this should sound like one tone with no or extremely slow oscillations. If your TCXO is out of alignment then you will hear two pitches that sound like they are oscillating; this literally sounds like wahwahwahwah. If out of alignment, the reason why you will hear this two tone oscillation is because the CW pitch is set to a 600Hz offset from that of the received carrier in the IF pass-band, where as the “Spot” oscillator derives its pitch directly from the pitch setting that was adjusted via the Pitch Knob, ours being exactly 600Hz away from dial frequency. If the TCXO is out of alignment, then the received carrier from WWV will be offset from dial frequency, thus something other than a 600Hz tone will be heard. And with the Spot engaged, you will hear the oscillation between the two tones. For a visual of this, see Figure 14 below.

***NOTE** - It should be noted that the 600Hz tone is the normal operating parameter for the internal Spot oscillator. Any other setting actually deviates the oscillator from the normal operating parameter in order to achieve that tone. You should also either adjust the level of the Spot tone via the side tone level control on the rear panel, or adjust your overall volume level such that both the side tone and Spot tone are the same level. This will help tremendously when making adjustments to the TCXO.*

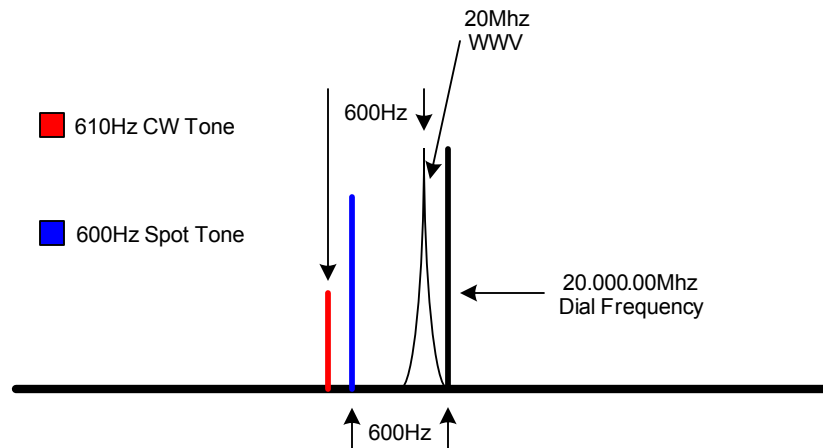


Figure 14: TCXO out of Alignment using CW Method

In this figure the TCXO is out of alignment by exactly 10Hz. The CW audible pitch of 610Hz, as opposed to 600Hz, notes this. Remember we set the CW to have an exact 600Hz offset from our receive carrier. When you turn the Spot on in this scenario, you will be able to audibly distinguish between the 610Hz & 600Hz from the Spot. In this case the oscillation between the two pitches, or wahwah's, will sound pretty rapid. We could cancel this oscillation, creating one separate 600Hz tone, by adjusting our dial frequency 10Hz down. But instead we are going to adjust the TCXO as the WWV station is a frequency standard.

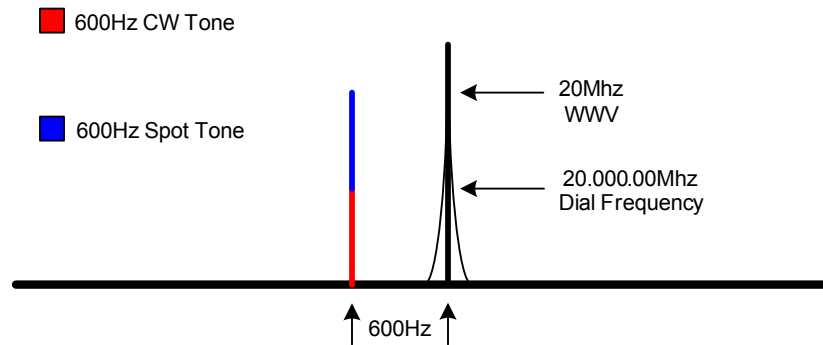


Figure 15: TCXO Perfectly Aligned using CW Method

In this figure the TCXO is aligned perfectly. What will be heard is one solid 600Hz tone with no or extremely slow oscillations; wahwah's. So let's align our TCXO using only the radio and our ears.

NOTE – Be sure that you perform this alignment during the time of day in your area in which the 20Mhz WWV station is the strongest, and will remain strong for the entire period of the alignment. If it should start to fade such that the tone being generated by WWV while in CW mode is hard to distinguish, you can drop down to the 15Mhz WWV station to complete the alignment.

More than likely, when you first set this up, and adjusted your VFO for 20 Mhz, your tone will sound as though it is oscillating; wahwah's. This is because your TCXO is off frequency. The object here is to

adjust the TCXO until you have a steady 600Hz tone with no oscillation. In order to facilitate this adjustment, you can select the 500Hz crystal filter for CW to narrow the receive pass-band, thus eliminating the tick tick tick or voice announcements from the WWV station. To narrow your pass-band even further, by turning EDSP back on, you can select either of the 240/160/60Hz CW DSP filters. This will provide you with an even tighter pass-band, which will really help in determining when you have achieved one solid tone. Again, we are attempting to align the TCXO, so we need to be as accurate as possible. When adjusting the TCXO, make very small incremental adjustments, giving time to listen carefully between each adjustment. Once you have adjusted the TCXO such that you have obtained a single stable 600Hz tone that has extremely slow or no oscillation, walk away and let the TCXO settle down for about 30 minutes; you need to give your ears a rest as well. After 30 minutes, come back and check the 600Hz tone once more. If it is rock steady then you are done. If it is still oscillating a little, then readjust, wait another 30 minutes, and check it again. Clocks really do need time to settle down after they have been adjusted. Simply put, when adjusting a clock you disturb their equilibrium and they need time to find it again.

One last note, you can substitute a Frequency Counter or appropriately equipped Spectrum Analyzer, or even an oscilloscope in place of your ear, and perform the same procedure. Instead of engaging the “Spot” and listening for a solid 600Hz tone, send the radio’s audio directly to the test equipment and adjust the TCXO for a 600Hz reading as per the test equipment’s frequency display. In the case of the oscilloscope you will need to provide a true 600Hz tone into channel 1 as a trigger for channel 2. Insert the radio audio into channel 2, and then align the TCXO until you have a complete match of waveforms on the oscilloscopes’ screen. Of course, all of this is assuming that you trust your test equipment’s own TCXO to be more accurate than your Spot oscillator and WWV.

5.2 IF Interstage Transformer Alignments

Considering that we have just replaced our OEM filters, thus having changed our IF pass-band some what, we should probably tweak our IF Interstage transformer alignments a little to squeeze the most out of the Mark V as we can. Although, I should tell you that after I had replaced the IF filters, made the proper menu changes, I did not tweak the IF transformers right away. I was so excited about the new bandwidth that I got on the air and was able to make contacts and received good reports. So if you are the least bit worried about performing this type of work, you will do fine without ever having to touch their alignments. But if you are the adventurous type, and demand the best possible performance out of your radio, then lets move on to the next paragraph.

5.2.1 Receive IF Interstage Transformer Alignment

Since we only replaced the OEM IF filters, we will concentrate our efforts on the Receive IF transformers that reside on the IF Unit, and negate touching the transformers that reside on the RF Unit. The service manual does a pretty good job of identifying the locations of each of the transformers that we need to touch upon, even points them out in a diagram, but seriously fails in telling you the order in which they need to be aligned. Instead, the manual gives you a shotgun affect as to which transformers need to be aligned, actually listing them in numerical order. The order in which we should align these transformers is definitely not in numerical order. We should align them in the order as they appear in the signal path, starting from the Receive IF input from the RF Unit to the Receive IF output to the AF Unit. Looking at the complete IF Unit schematic in the service manual we can see that the Receive IF enters the IF Unit via J2003. So if we follow that signal path all the way to J2002, where it exits the IF Unit on its way to the AF Unit, we can determine the order in which each transformer needs to be aligned. Here is the order in which they need to be aligned just in case you cannot read the fine print on the schematic:

Receive IF Transformers			
Seq #	Transformer	Seq #	Transformer
1.	T2005	6.	T2009
2.	T2006	7.	T2003
3.	T2007	8.	T2011
4.	T2004	9.	T2012
5.	T2008		

Table 3: Receive IF Transformer Sequence

The other thing that I dislike about the way the service manual tells you to align them is to use the Mark V's S-Meter as your signal level indication meter. There is no other way to put this other than THAT SUCKS. It would take at least a 3db improvement of signal strength to move that S-Meter one bar. How can you make a fine adjustment to an IF Transformer with such a huge bump in signal level required in order to deflect the meter one bar? We definitely need something that is more sensitive than that! What comes to mind that would fit the bill would be a DVM attached to the speaker output jack of the radio. We could then watch the DVM for changes in audio voltage, adjusting the transformers for a peak reading. What would be even better would be an RF millivoltmeter bridged to the output of J2002. We could make our adjustments for a peak reading on it. Any of these ideas would be a whole hell of a lot better than using the S-Meter. But if you have to use the S-Meter, then be sure to turn the RF Gain all the way up, set the AGC for Fast, and turn off the AGC peak hold in the menu system if you have it turned on. It would also be wise to inject just enough signal level into the radio to keep the S-Meter deflection to around an S1 to S2; this seems to be the most sensitive range of the meter.

Now unfortunately we do need a signal source with a steady level. We have used WWV before, but dependant upon propagation, this signal may waiver too much in strength. This will end up causing you great frustration, as you will never get a straight reading when you make your adjustments. So we are definitely going to require the usage of an RF Signal Generator or a second transceiver that has been properly aligned. I am sorry but there is no way around this one if you want to make accurate adjustments.

***NOTE** – If due to the lack of a signal generator, a second transceiver is used, ensure that you use a transceiver that is properly aligned and transmits on frequency. You will need to terminate the second transceiver to a dummy load, or pigtail antenna. Set the transceiver for 14.200 Mhz and transmit a 1-2 watt CW signal, which the Mark V should receive. Alignment of the Mark V at this point is just as depicted below, minus the signal generator.*

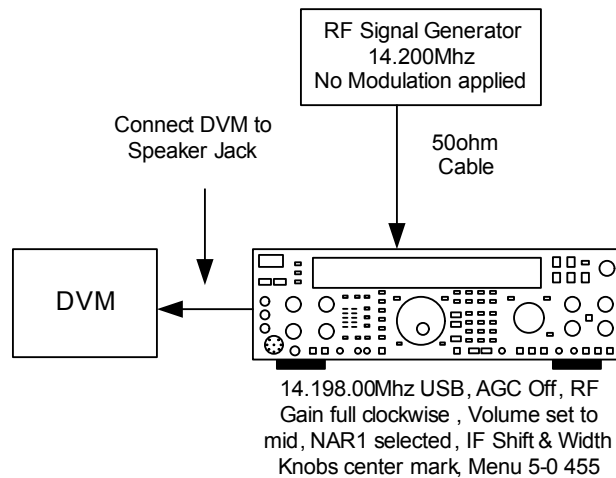


Figure 16: Receive IF Transformer Test Set Up

Once you have secured the test gear, you need to set up your test bed as depicted in Figure 16. The DVM should be connected to the speaker jack of the Mark V, with no termination resistor, and set for an AC reading. The Mark V should be set for USB, 14.198.00 – 14.199.00Mhz, AGC Off, RF Gain fully clockwise or adjusted to keep from overloading the receiver, volume set for mid level, menu 5-0 set for 455, IF Shift & Width set for dead center, NAR1 selected. The RF Signal generator should be set for 14.200Mhz, no modulation, and dead carrier. This will generate either a 1KHz or 2KHz tone out of your radio, dependant upon the VFO setting on the Mark V. It does not matter which VFO setting you use, as the pass-band is so wide, that either of the two settings will result in the same signal level received.

Feed the radio with a +80dBu signal level at 14.200Mhz. Watching your DVM, be sure to adjust the volume such that you keep the voltage reading on the DVM around .075vac RMS. Keep in mind that these transformers may only require a slight adjustment. But then again, you may have to adjust the lug up or down a bit in order to achieve a peak reading on your DVM. Once you are setup and have a good voltage reading on your DVM we can start to adjust the transformers. Using a plastic alignment tool, adjust each transformer in the order provided earlier for a peak reading. When completed, go back and make one more alignment run through the transformers. Only this time knock the signal level down a bit on your RF Signal Generator. This lower RF signal level will help enhance any level movement while adjusting the transformers. On this second go around, our adjustments should be minimal.

NOTE – When making these alignments, it is very important that your IF Shift and Width Knobs are set dead center. If they are not, then affectively, you are aligning your receiver to the location of the band-pass as set by your IF Shift and Width Knobs.

Now that you have done this, I will tell you that I found mine to be off more than I would have expected; more than a half turn on several of the transformers. I do not know if this was due to the IF Filter replacement or just general mis-alignment of the radio. However, I have heard that these radios do ship out of the Japanese factory pretty much aligned, but do not fair well with the whole shipping process from the factory door to your front door. This has been attributed to all of the bump and movement that they go through getting to you, not to mention component values slightly changing over time.

5.2.2 Transmit IF Transformer Alignment

As with the Receive IF Interstage Transformers, the Transmit IF Interstage Transformers probably should be aligned since we have changed out the OEM IF Filters. The good news about this alignment is that we really do not need any test equipment. However, the bad news is that Yaesu, once again, did a poor job in documenting within the service manual the correct order in which these transformers need to be aligned.

Looking at the IF Unit schematic within the service manual we can see that the Transmit IF comes in from the AF Unit at J2009, and leaves the IF Unit on J2010 on its way to the RF Unit. If we trace that signal path, we can determine the order in which to align the transformers pointed out in the service manual. Again, since we only changed out the OEM IF Filters, we will negate aligning the RF Unit transformers called for in the service manual. The specific order we should use is noted below:

Transmit IF Transformers			
Seq #	Transformer	Seq #	Transformer
1.	T2019	4.	T2018
2.	T2013	5.	T2017
3.	T2015	6.	T2014

Table 4: Transmit IF Transformer Sequence

Use the exact method that is pointed out in the service manual. Just negate the alignment of T1001, and T1002 as they reside on the RF Unit. I would perform two alignment passes at these transformers. The second time around the alignment should be minimal if at all needed.

5.3 SSB Carrier Suppression Alignment

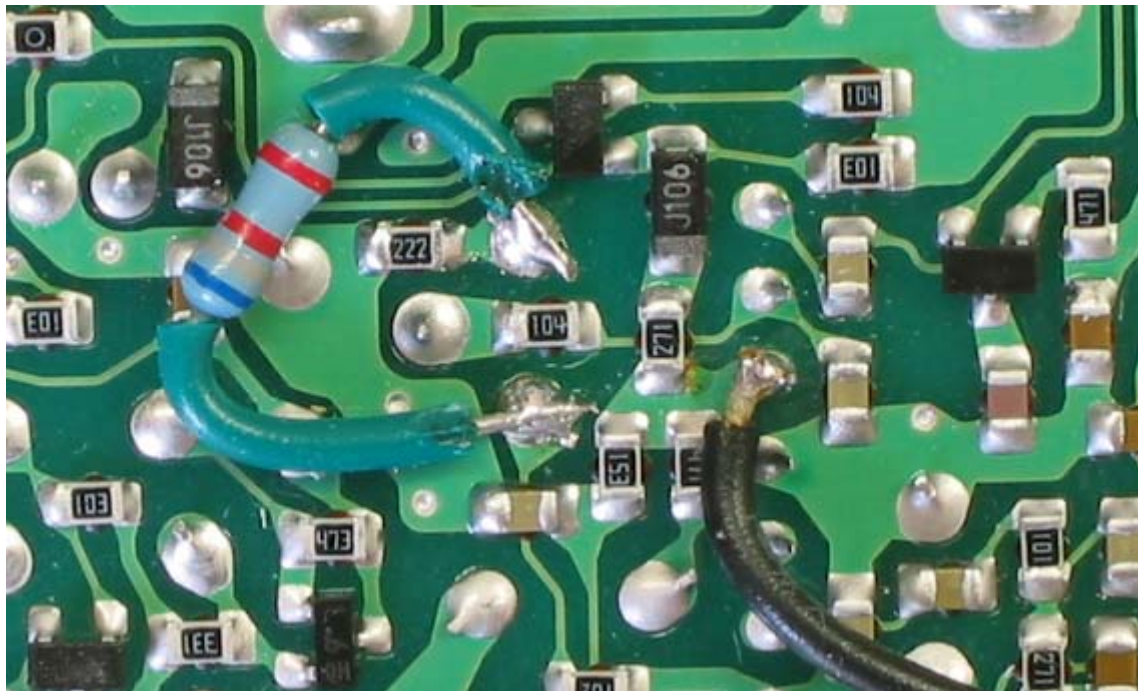
This is a general alignment that should be checked considering that you already have the radio opened up; never skip an opportunity to ensure your radio is in tiptop shape. Directions on how to adjust the SSB Carrier Suppression, officially stated as SSB Modulator Balance by Yaesu, can be found in the Service Manual on page 3-9. It calls for adjusting VR3012, a 10K ohm potentiometer that resides on the AF Unit (See Picture 5) with the radio keyed down in SSB with no modulation applied, while watching the carrier level on a spectrum analyzer or near by receiver. The object is to adjust VR3012 such that you achieve the lowest level of transmitted carrier while operating in SSB. The manual indicates that this should be roughly -50db. If you have ever tried this adjustment on a Mark V, you know that it can be rather difficult, and here is the reason why.

The potentiometer used by Yaesu for this particular circuit is a Rohm, carbon film, 100ohm – 10K Ohm, top adjustable, 210 deg rotational resistor. If the movement of the stator was linear in relation to resistive value, and given a 210 deg range of motion, this would translate to roughly 47ohm's per deg of rotation. Assuming that the value range of the potentiometer was chosen due to the actual target resistance being some where in the middle, this would put the adjusted stator some where around 105 deg. However, in the real world, rotation of a stator in relation to resistance is not always linear in fashion, and values increase more rapidly the closer you are to either the upper or lower bounds of the potentiometer.

So why is this being pointed out? Well I tried performing the Carrier Suppression Adjustment in the hopes of achieving a better result than the factory setting, ~ -50db. This simple adjustment cost me 30 minutes of time and much frustration, as the slightest movement of the stator would result in wild swings

of the carrier level. What was frustrating is that I was able to see fleeting glimpses of a carrier $< -60\text{db}$, but could not actually achieve that value due to the sensitivity of the adjustment. This sensitivity is due to this potentiometer having a non-linear response to rotation of its stator, not to mention at least 47 ohms per 1 degree of adjustment.

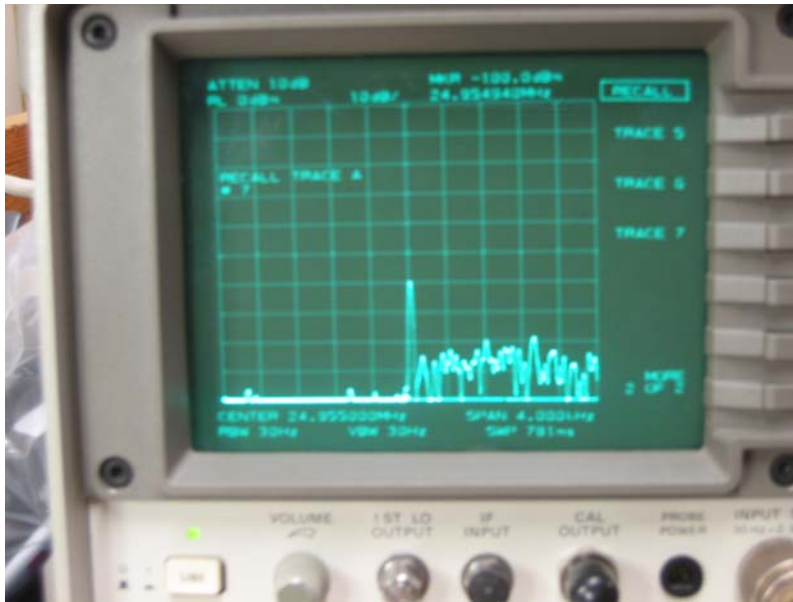
So how do you compensate for an overly sensitive potentiometer in a circuit where the target resistance is assumed to be some where in the middle range of the potentiometer's value? You could search Digi-Key or other parts warehouses for a more suitable replacement, but you will find that there is not a standard potentiometer that would offer a 2K - 7K Ohm range with a 210-degree movement. So the best alternative would be to attach a standard 6.8K Ohm carbon resistor in parallel with VR3012, thus providing an adjustable resistive range of 3.45K to 8.4K Ohm. This modification would keep the assumed target value in the lower middle range while decreasing the ohm value per degree of rotation of the potentiometers stator, overall making it less sensitive and easier to adjust.



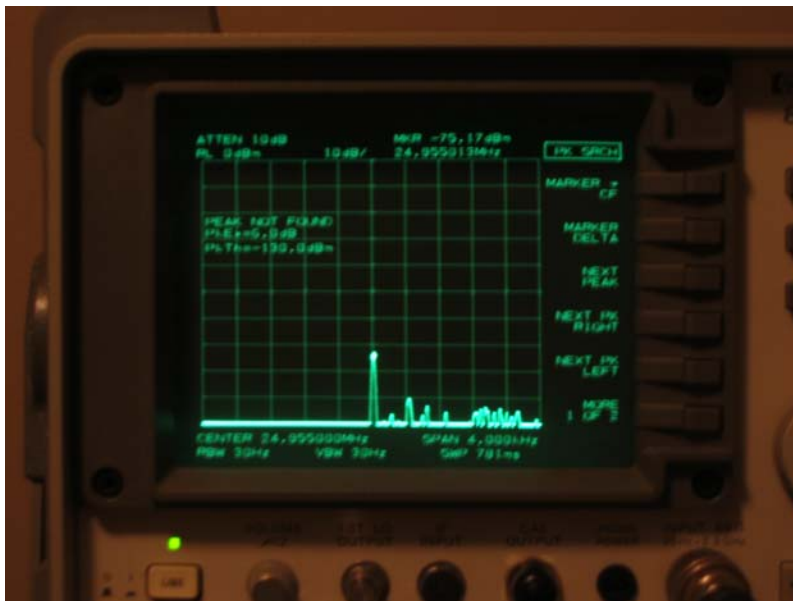
Picture 26: Carrier Balance Resistor Installed

Looking at Picture 26, you will see the resistor already installed on the AF Unit. I used the jacket from a wire as a method to insulate the resistor leads thus ensuring that the leads would not cause a short.

Prior to installing this resistor, I was only able to achieve $\sim 60\text{db}$ carrier suppression, as can be seen from Picture 27 of the spectrum analyzer. After the modification I was able to achieve $\sim 75\text{db}$ of carrier suppression in USB and LSB, Pictures 28 and 29 respectively.



Picture 27: USB Carrier Suppression Prior to Mod



Picture 28: USB Carrier Suppression after Mod



Picture 29: LSB Carrier Suppression after Mod

5.4 Menu 9-2 Transmit IF Alignment

Menu 9-2 controls the Transmit If gain in a band-by-band configuration. The following table decodes each sub-menu system in relation to the amateur band that it controls and shows what my radio's default settings were, and what they are now:

TX IF Gain Translation			
Menu	Band	My Default	New Setting
T IF-018	1.8 Mhz	10	9
T IF-035	3.5 Mhz	8	7
T IF-070	7.0 Mhz	9	8
T IF-100	10.0 Mhz	9	8
T IF-140	14.0 Mhz	9	8
T IF-180	18.0 Mhz	9	8
T IF-210	21.0 Mhz	10	9
T IF-245	24.5 Mhz	10	9
T IF-280	28.0 Mhz	11	10
T IF-290	29.0 Mhz	11	10

Table 5: TX IF Gain Translation

The transmit IF gain electronically controls the final transmit gain out of the IF Unit prior to it being presented to the amplifier section of the Mark V. This is by no means a way of controlling the power output of the Mark V, but is used to set proper IF transmit drive levels. One of the things that we should

be concerned about, with any transmitted SSB signal, is how hot are our 3rd or 5th order transmit IMD products. The Mark V does a decent job of keeping those down in level per their published specifications. But it could do a better job. We can actually help it along by making an adjustment to our Transmit IF Gain setting. Looking at Table 3, you will notice that my new settings are one numeral down from the original factory default settings. In doing this, I decreased my 5th order transmit IMD by ~2-3db, and still maintained my normal transmit power out. I was able to detect this decrease in level by using a spectrum analyzer using peak search, while performing a two-tone test. I even went so far as to take the IF Gain setting down one more notch, thus affecting more of a reduction in my 5th order transmit IMD; an additional 2-3db dependant upon the band. But I did sacrifice some power out on each band. If you want a real clean transmitted signal, then I would suggest turning down your Transmit IF Gain settings by one numeral for each band. In addition, I would use the Class A setting, as it is much cleaner than the standard 200W setting, and drive an amplifier for power out. This will guarantee a cleaner signal than you have ever achieved out of your Mark V to date.

5.5 Menu 9-1 Receive IF Alignment

Menu 9-1 controls the overall Receive IF gain. Adjusting the menu setting either up or down in value has a corresponding affect on the amount of gain that is applied to the Receive IF. Most Mark V's, straight out of the factory, have their value set to 13-14. In reality, this is quite high and does introduce impairments into our receive audio. By adjusting this control down by one or two numbers you will obtain smoother receive audio. Granted this will affect your S-Meter some, but with the right test equipment, and using the Service Manual, you can readjust the S-Meter to take into account the adjustment that you made in menu 9-1.

6. Operational Aspects of Your New Radio

Welcome to your new radio. Most of what you knew about the current operations of the Mark V has now changed. You must keep in mind that instead of working with 2.4 KHz filters, you are now working with 6 KHz filters. This has a dramatic effect on how the radio reacts, and how you, the operator, will control those reactions.

The next several paragraphs go into great detail on what you need to know in order to achieve the greatest potential for ESSB operations, and how those changes affect the operational aspects of your new radio. It is suggested that you keep a copy of this portion of the document very handy until you have mastered all of the operational changes.

6.1 Usage of External Audio Equipment as a High & Low Pass Transmit Filter

Since we have bypassed the low and high pass filters in the Mark V for our transmitted audio, any SSB transmission at this point will be ~5.8KHz wide. In crowded band conditions, this will generate quite a bit of animosity towards you, as you will be stepping on stations next to you. It should also be mentioned that attempting DX contacts using wide band audio will generate quite a bit of confusion, and will probably result in no contact. Basically the DX station will not hear you over that of the other calling stations. This is because the other calling stations are concentrating most of their RF power in a 2.5KHz bandwidth, where as you are spreading your RF power over a huge 5.8KHz of bandwidth in accordance with your new filters cascaded shape. So how do we limit our transmitted bandwidth for those situations that call for pinched up audio? We use external audio equipment configured as low and high pass filters, essentially external DSP filtering.

The Mark V, as any other radio, will transmit the audio bandwidth presented to it, spreading its RF power across that bandwidth dependant upon IF filter shape. So it stands to reason that if we limit the size of the audio bandwidth being presented to the BM, we limit the over all transmitted bandwidth coming out of the radio. We do this by routing the audio through an external equalizer prior to presenting the audio to the BM. The two equalizers that I play around with in my audio chain are the Behringer DSP1124 and the Behringer DEQ2496. There are many others out there that can do the same thing, so do not rush out and buy these specific pieces. It's just that I happen to have these on hand, and use them for my station. However, it should be noted that when choosing an equalizer to use for bandwidth limiting, be sure to pay close attention to the amount of attenuation that it is capable of producing, as well as how sharp it can provision its filters. Ultimately, you are looking for at least -45db of attenuation, and filters sharp enough to be used as brick walls. The DSP1124 is easily capable of producing those filters using one channel. The DEQ2496 will definitely require that you use both the left and right channels to achieve that kind of attenuation. In other words, the DEQ2496 is really not best suited for creating brick walled filters for the purpose of band limiting our audio. If working with audio processing gear is new to you, and or the terms octave and parametric EQ are foreign to you, I highly suggest that you visit some of the more detailed audio websites listed in Appendix A of this document before going any further.

I am not going to going to discuss the menu system for these two specific equalizers, but offer up a means for which to facilitate adjusting any equalizer of your choosing to achieve low & high pass filtering. Looking at Figure 17, you will find a test setup that can be used to adjust your equalizer in real-time, without having to make on the air tests with a buddy. Keep in mind; this setup is only for setting up your low & high pass filters, not for tuning the equalizer to enhance your voice.

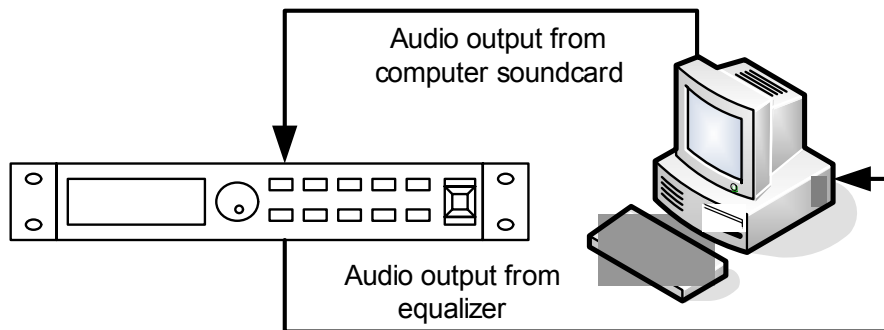


Figure 17: Equalizer Test Set Up

Prior to setting up this test set up, you should down load an additional piece of software that will greatly facilitate you in choosing which frequencies, and their associated octaves to attenuate in order to achieve the filtering you desire. The software was written by the Engineers at Behringer, and is used for remotely provisioning the DSP1100. Even if we do not have a DSP1100 at our disposal, this piece of software is still invaluable to us! Built into this software is the capability to model parametric EQ settings, and view how they will affect our bandwidth. This modeling also takes into account the synergistic affect of having these parametric filters placed close to one another. This means that it is capable of modeling and showing the changes in our bandwidth as we move these parametric filters in close proximity to one another. I do not know about you, but viewing in real time is the cats meow. So go to the below hyper-link and download the software.

http://www.behringer.com/05_support/downloads.cfm?sel_prod=DSP1100P&lang=ENG

When you first startup the software, it will complain about not seeing an input or output device of “MPU-401”, just click on the “OK” button, and you will be taken to the modeling screen of the software. Once there, go to the top left of the screen and click on the “File” tab, and then “New”. This will bring up both a filter input screen and a real-time analyzer screen. If you are familiar with setting up audio processing gear, these screens will be very intuitive. If you are having a hard time understanding what is going on, visit some of the websites in Appendix A of this document.

***NOTE** – You should set your computer’s audio card output to around mid-level. This will ensure that you are not over driving your sound card thus sending out a clean signal from Test Tone Generator. Once you are satisfied with your signal quality coming out of the sound card, then go ahead and test your equalizer settings per the below paragraph.*

When you have finally narrowed down some settings with the modeling program, you will probably want to test them to ensure that they will work in the real world. In order to perform these tests, and subsequent adjustments of your equalizer, you will need to have both SpectraPlus and Test Tone Generator installed and running on your computer, see Paragraph 2.3.1 for hyper-links to the appropriate websites. Feed either the left or right channel input of your equalizer with the audio output of your computers sound card, and take the output of that channel and feed it back into your computers sound card. With Test Tone Generator running, set it up to perform “white noise” generation. Next start SpectraPlus and actuate the “run” button. Be sure that the Test Tone Generator window is in the foreground (active window) and slid off to the side of your screen, otherwise Test Tone Generator will pause its operation. At this point you should see the white noise pattern presented within SpectraPlus. The object here is to adjust the equalizer under test such that the white noise pattern presented to SpectraPlus is limited to the bandwidth of your choosing. Just to double-check your settings, you could run an audio sweep using Test Tone Generator, starting from 10Hz and ending at 7Khz. With SpectraPlus setup for peak hold, you will achieve graphs like those found in Appendix B. I have my equalizer setup for the following bandwidths:

- 2.5Khz – Used for chasing DX (200-2.7Khz, requires usage of both left and right channels, or another EQ in line)
- 2.8Khz – Used to please those older operators listening preferences (100-2.9Khz requires usage of both left and right channels, or another EQ in line)
- 3.5Khz – You did all of this work, you should spread the wings a bit (40-3.5Khz)
- 4Khz – Great for rag chewing with your ESSB buddies (40-4Khz)
- 5Khz – This is the “I don’t give a &^*\$ what you say” setting when confronted by ESSB haters (40-5Khz)
- WAO – Affectionately known as the Wide Ass Open setting (WAO)

As you setup each of your low & high pass filters combinations, write down the settings that you used for future reference, and then electronically save the setting within one of the equalizer’s memories. When you want to change your transmitted bandwidth, it is now just a matter of recalling a preset and activating it.

Another note about setting up your equalizer, I would use the left channel to build your low & high pass filters, and the right channel for actually tailoring your voice, unless of course you have the DEQ2496. All you have to do is take the output of the left channel and feed it to the input of the right channel in order to do this. A word of caution though, be sure that you make individual settings for your voice on a per bandwidth basis, and do watch your level settings between the two channels. How you have the Right Channel EQ'd for your voice within a 3.5Khz band-pass filter setting in the Left Channel will probably not sound as good in a Left Channel 4.5Khz band-pass environment. Most equalizers will let you save both the Left and Right channel settings all in one memory. Again, read some of the material within the suggested audio websites in Appendix A, as well as your audio equipments manual.

***NOTE** -After re-reading the above paragraph, I began to realize that there is probably no way that I will get away with not presenting at least some suggested band-pass filter settings for the DSP-1124, or similar device. So if you look in Appendix B, you will find some suggested settings to get you started.*

6.2 General Hook up of External Audio Equipment

I should at least provide a good starting point on how to interface from your microphone through a simple rack of audio equipment into your radio, and out of your radio into either the computer sound card or external amplifier or both. By now you must realize that in order to get the full potential of your new radio that you can no longer use your standard hand microphone. And even then, doing so will constitute a 5.8Khz transmitted bandwidth with no opportunity to cut it down. I do not wish to get into recommendations of what type and manufacturer of microphone you should use, but only that you will have to use something other than your radio's OEM hand microphone. The problem with any studio microphone is that the audio level output is pretty low, so low, that unless amplified by a MIC pre-amp, it will not be able to drive any other external audio processing gear properly. If you remember from paragraph 6.1, you will need to use an external equalizer in order to provision a low & high pass transmit filter. This also means that you will need to obtain a microphone pre-amp in order to drive that equalizer, and at the same time, it would be nice to obtain a simple mixer as well. So how do you hook all of this equipment up? Well, take a look at Figure 18.

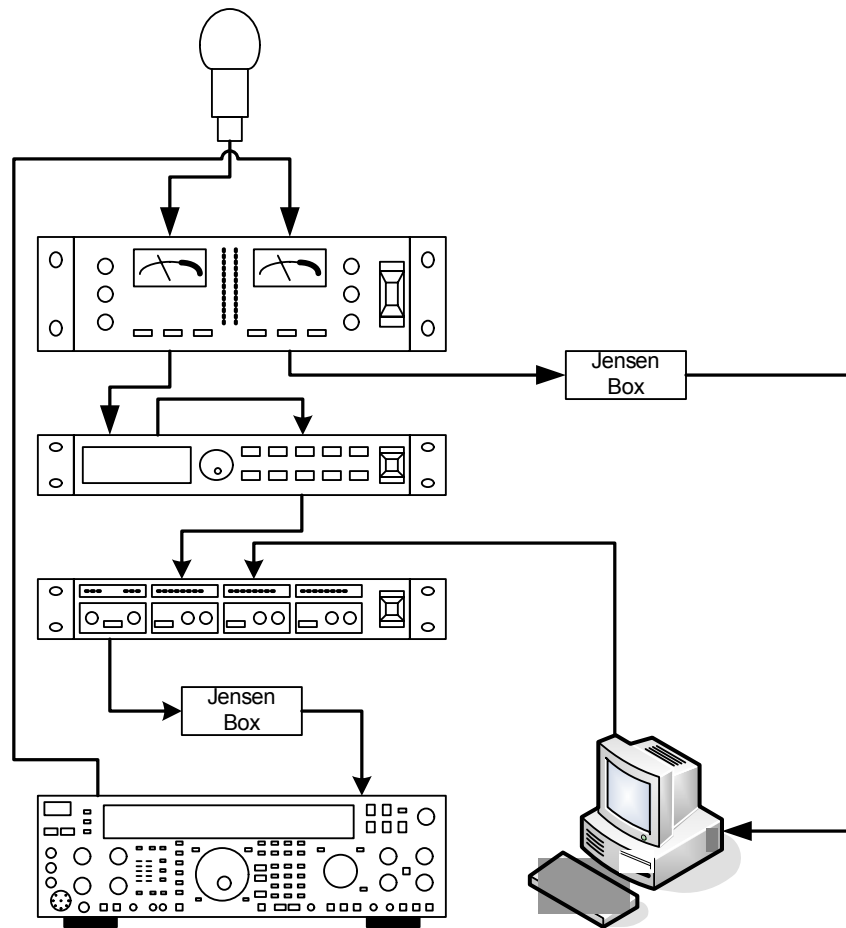


Figure 18: Interfacing a simple Audio Rack

In this figure we see a studio microphone that is routed to a MIC pre-amp. The MIC pre-amp amplifies your voice from the microphone and then sends it on to the equalizer. The equalizer, if setup properly, limits the band-pass of your voice (Left Channel) while at the same time enhances it so that you have a fuller smoother sound (Right Channel). The equalizer then passes your voice on to the mixer, where you can make final adjustments to the overall level being presented to the Balanced Modulator. From there it passes through the Jensen transformer, as discussed in Paragraph 3.5, and then goes straight to the Balanced Modulator via the new RCA Jack in the back of the radio. You will also notice that there is another lead coming out of the computer sound card, going into the mixer. This is so the audio from the computer sound card can be selected for transmission out the radio. There are some very good reasons for this, which is pointed out in Paragraph 6.7.

As was said, this is a very simple audio rack. Believe me when I say that an audio rack can get very extensive, dependant upon the applications and functions that you would desire. To give you an idea, here is the line up for my audio rack:

- Heil PR-40 Studio Microphone

- ART MPA Gold Pre-amp - Left channel for the microphone and Right for receive audio. Original tubes rolled with secret voodoo tubes.
- Bellari 533 – Great Harmonics, Tubes have been rolled, and Op-Amps exchanged
- Behringer SNR2000 – Used exclusively as a noise gate
- Behringer DSP1124 – Used as band-pass limiting filters (Left & Right Channel)
- Behringer DEQ2496 – Left channel used for voice enhancement, Right Channel used for receive audio enhancements
- Behringer EX3200 – Used to enhance the bottom end
- Behringer DSP2024 – Used for fun audio effects
- Bellari 583 – Tube compressor used to put final touch to audio; tubes have been rolled and Op-Amps exchanged
- Behringer MX882 – Mixer used to adjust audio level and route it to the radio

If you think this is extensive, ask Jim (VE7RF) about his line up. I strongly suggest that before you go full blown such as above, or total audiophile such as Jim, I would read the websites within Appendix A, and master the simple rack first.

6.3 What Happened to my VOX?

When we by-passed the Mark V's Microphone input, opting to directly interface into the BM, we also by-passed the circuit that is used to derive the voltage required to actuate the pin diode T/R relay that keys the transmitter when we speak into our microphone. So in other words, in the current configuration, we lost our VOX capabilities. However, there is a way around this. Utilizing our mixer, or some other audio splitting device, we can provide an additional audio feed to our radio, terminating it either on the phone patch jack in the back of the radio, or the microphone connector in the front of the radio. Considering that the VOX circuit is well before the microphone gain circuit in the transmitted audio signal path, the VOX circuit can still generate a control voltage to key the radio without the microphone gain being turned up. And in reading the next paragraph, you will see how we can provide this additional audio feed, and utilize it just in case we want to revert back to transmit EDSP for narrower bandwidths; see Figure 19.

6.4 EDSP or not to EDSP

That is truly a good question. Prior to the modifications, in order to achieve the widest possible band-pass of the Mark V you had to be in EDSP mode. This allowed you to setup both receive and transmit for a 100-3100Hz band-pass. Doing this also provided you some neat features such as Auto-Notch, IDBT, narrow audio DSP filters for the digital extreme user, as well as audio equalization for your voice. It's kind of hard to completely throw those features off to the side, all in the name of ESSB. The plus to the modifications is that you can still use the Mark V setup in transmit EDSP as you did before. You will have to use the microphone or phone patch jack for your transmitted audio. And there is no need to re-adjust your carrier offset. EDSP has it's own defined carrier offset that takes affect the moment you turn that function on. But why would you want to use EDSP again? Let me explain.

Let's look at the Auto-Notch feature first. When you setup the Notch Filter for the Mark V in the menu system for "Auto" while in EDSP mode, you have the best of both worlds when trying to abate yourself from heterodynes and CW QRM'ers of the world. When engaged, the DSP side of the house will go on a hunt and destroy mission, killing up to two signals simultaneously. Then you also have the ability to swing the analog side of the house into the foray if a third signal popped up. Not a bad feature to have, too bad we lost it when we went strictly analog. Or did we? Most of the commercial digital combo

equalizers in usage today by Amateur Operators have what is called a Feed Back Destroyer feature. This feature allows you to setup any number of Parametric EQ's strictly as FBD filters. This means they can be used to go on hunt and destroy missions, killing up to as many heterodynes and CW QRM'ers as there are FBD filters provisioned. Even better, these FBD filters are sharper and have more attenuation per instance than the Mark V could ever hope to achieve. And if you really want to, you could bring the Mark V analog IF Notch into the foray as well; remember Menu 2-9. Two pieces of gear that we have discussed before in this document that are capable of FBD filters are the DSP1124, and the DEQ2496. Since we are using the left channel for transmission, why not use the right channel for tailoring our receive a little, like killing all those pesky high-pitched signals that keep making a nuisance of themselves.

How about those narrow audio DSP filters for you digital extreme users out there? They work great in keeping the adjacent trash out of the way so that you have an excellent copy on that foreign station. It's a pity we lost that ability since the upgrades, right? Wrong. Using an equalizer in the receive path, particularly those with multiple functions built in, you can narrow the receive audio band-pass in a more personalized manner than the Mark V ever could. Personalized filters can be setup for the digital mode that you wish to operate, and those filter skirts are a lot sharper than what you would get using the canned Mark V settings. All you have to do is route the receive audio into an EQ.

Well, we really do not need to reflect on what the proper equalizer can do for enhancing your voice, but how about narrowing your transmission band-pass down for digital work, such that you achieve the ultimate in IMD and signal separation between mark and space. This can be done as well. All it takes is a little tweaking using the test setup as pointed out in Figure 17 to get the right sized transmit band-pass, and off you go. Now don't get me wrong, I make it sound easy, but in reality, it will take some work in selecting the right frequencies and octaves when making an excellent transmit band-pass filter. And it should be noted that you may have to use both the left and right channels in conjunction with each other in order to achieve the filtration you are looking for. But once you are done, the signal reports will be well worth your effort.

So if after reading some of what can be accomplished by placing an equalizer in your receive audio signal path, or using your equalizer more proficiently for digital work, you still want to revert back to EDSP occasionally, then you will need to do a few more things. As was mentioned earlier, you will need to utilize your microphone or phone patch jack. If EDSP is turned on, then the transmit bandwidth will be that which you have setup in Menu 7-7. If EDSP is turned off, then just like the BM, the microphone or phone patch jack will allow a 5.8Khz band-pass to be transmitted.

Now when using EDSP, you can either use your hand microphone or you can still use your audio rack gear. But in using the audio rack gear, you will need to step the audio level back down to a microphone level. This can be done in one of two ways, either you readjust all of your settings in your audio rack gear such that your signal output of the mixer is at least 40-60db down from line level, or you use a step down transformer and leave your audio level settings on your rack alone. The lesser of the two evils would be to use a step down transformer, and this can be accomplished by using another Jensen transformer, the JT-10DB-E. The JT-DB-E is a "direct box" 12:1 transformer, used to convert line level to microphone level.

***NOTE** – Even if you do not modify your Mark V for 6Khz transmitted band-pass, this method of feeding your microphone or phone patch jack with audio processing equipment to enhance your voice will work*

great. Just remember that Menu 4-4 needs to be turned off and Menu 7-7 needs to be set for 100-3100hz all the way around. There are plenty of operators who are doing this very thing today.

Below is a figure depicting how this transformer should be deployed. This is per Jensen, and it works very well.

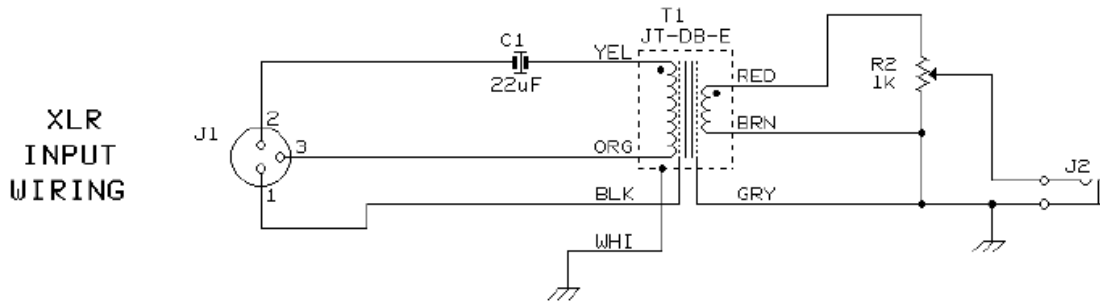
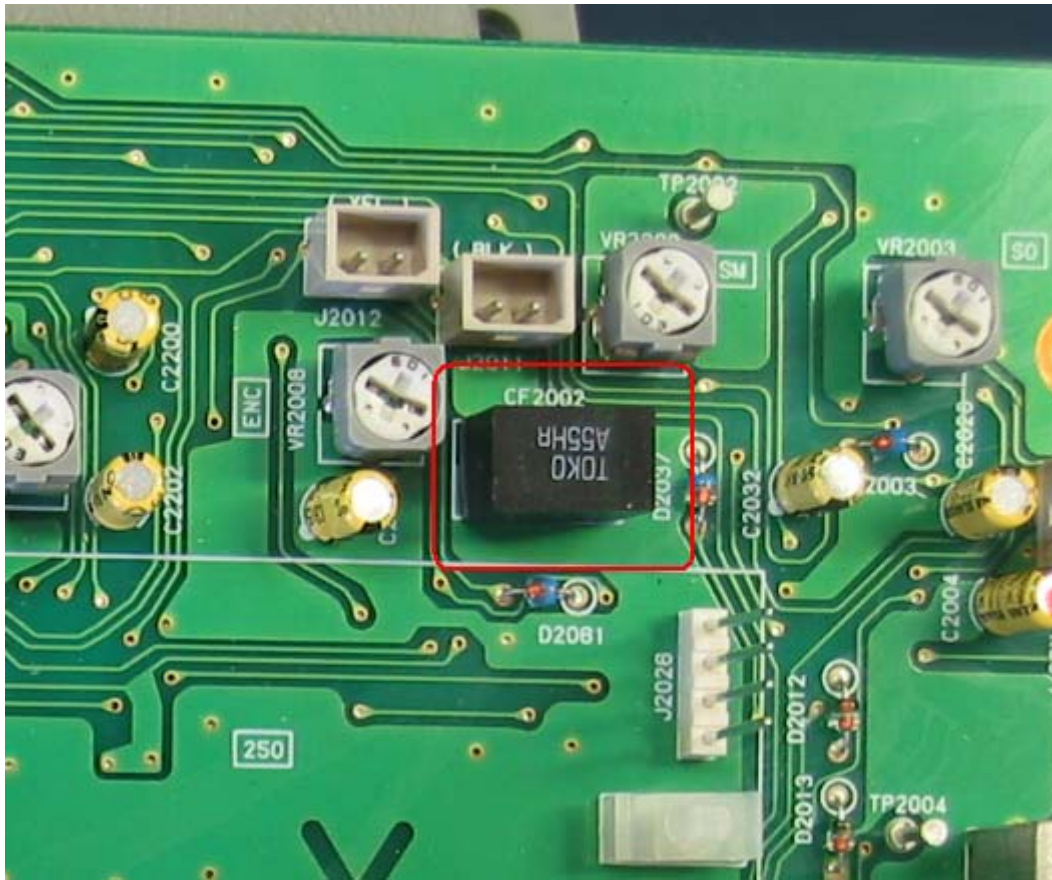


Figure 19: Jensen JT-DB-E

You will notice that the above configuration does call for an audio potentiometer (R2). This can be used to make adjustments to the final audio level being presented to either the phone patch jack or the microphone connector on the front of the radio. The transformer itself provides roughly 25db of attenuation, and coupled with R2, will be able to provide the additional attenuation required to keep your audio equipment at unity gain. I have this built inside of an aluminum project box, and instead of using the connector depicted as J2, I am using an XLR connector. I did this as I am interfacing into the microphone connector, and thus built an XLR to 8-pin DIN cable; pin out per the Mark V Operator's Manual. It should also be noted that the 22uf cap is recommended by Jensen in order to block transient voltages from entering the transformer. Now this capacitor can be substituted with a Panasonic SU 220uf 25Vdc capacitor, or even omitted. I will say that I am using a Panasonic SU 220uf in place of the 22uf capacitor suggested by Jensen.

6.5 AM Operations

Well, we have covered a lot of material for ESSB operations, but what about AM? How do the modifications affect that mode of operation? I can definitely tell you that directly interfacing to the BM does nothing but good for our transmitted audio quality, but does nothing for our ability to transmit enough sibilance in AM to sound superb. The reason for this is that the Mark V has a soldered in Murata 6Khz filter used for AM that straddles zero beat when the radio is placed in AM mode. This means that there is ~3Khz of coverage on the upper and ~3Khz of coverage on the lower sideband from dial frequency. So when transmitting in AM, you are actually only producing ~3Khz of band-pass as decoded by a receiving station. The same goes for receiving an Amplitude Modulated signal as well; that is if you do not have menu 5-0 set for 8.2-455. So how do we increase this, and what will it do in conjunction with the current ESSB modifications already performed on the Mark V?



Picture 30: Murata AM Filter

Increasing the overall transmitted bandwidth for AM will call for another filter change. Looking at Picture 30 of the IF Unit, you will find CF2002 circled in red. This is an 11 x 7mm Murata 455Khz filter, part number CFWS455HT. This filter alone is the limiting factor to our transmitted AM bandwidth. By replacing this filter with a different Murata filter, such as the CFWS455F, we can increase our overall transmitted sibilance to roughly 6Khz; 12Khz total transmitted bandwidth. However, just like the ESSB modifications, we are now transmitting twice the bandwidth we did before. Again, this transmitted bandwidth can be controlled through the usage of external audio equalization equipment. And unlike the ESSB modifications, we do not need to adjust the carrier offset to account for the new bandwidth. So the AM carrier offset, t-A3-cAr, found in menu 9-0 needs to stay at factory default for your radio.

Now it should be said that I have not done this particular modification, but it is well documented and in usage today by K1DEU. You will find a hyper-link to his website in Appendix A of this document.

Now, how will this affect the current ESSB modifications that we just performed? It will definitely add a little to our receive bandwidth as we are now passing the IF through a 12Khz filter, particularly if menu 5-0 is set for 455. But with the good, definitely comes a little headache. When this new Murata filter is selected for SSB receive, it is no longer straddling zero beat, yet passing 12Khz of receive IF for either USB or LSB. Because of this we can expect strong stations within that 12Khz band-pass to affect our AGC. So there is definitely a decision to be made here, one that I have not settled on yet either. However, I am considering using a different Murata Filter, a CWFS455G. This is the same filter used for

FM operations within the Mark V, and has a +/-4.5Khz band-pass. Usage of this one would provide 9Khz of band-pass in SSB if placed in the Receive IF path, as opposed to a full 12Khz.

Now as far as our current configuration, and the ability to transmit AM, we really do not need to perform any menu changes or re-route our audio from our rack gear to our radio. Just setup the Mark V like you have in the past for AM, minus EDSP, and start transmitting. Just be cognizant that you will definitely be transmitting more lows than you ever have before because you are directly interfacing into the BM. So you may want to compensate for that in your EQ. Unless of course you like the idea of rattling your buddies receiving station with some sub-sonic audio.

6.6 On the fly Audio Tailoring

If you are already an ESSB connoisseur then you will know exactly what I am talking about. And if you are not, then here is some of what can happen. Man your bass is way to low, you have too many lows in your audio, I can't understand you because you sound muffled. These are some of the comments that you may receive from stations that are band limited to 2.4 – 2.7Khz. The reason for this is because they are incapable of hearing all of your sibilance, and yes, you are transmitting some lows. The root cause is that you are transmitting more lows than most operators, and you are externally EQ'd for the band-pass that you are transmitting. The symptom is that the complaining station cannot hear enough of the high frequencies to offset the low frequencies. So to them you do sound muddy, or too boomy, or a whole plethora of terms used to describe too much bass. So what do you do in a pinch?

Well, you have about three options here. The first option is to readjust your carrier offset to something more like +1580 for the current QSO. This will move your transmitted band-pass further away from zero beat thus cutting out a lot of the lows that you were transmitting and balance the overall audio spectrum being received by the complaining station. There is a problem with this though; you are still transmitting a 5.8Khz band-pass, which is being band limited by your equalizer. So if you swing that carrier offset such that your 5.8Khz band-pass extends past the attenuation of your EQ for the higher audio frequencies, you will be sending out trash at higher frequencies, up to the point where your filter skirt stops. This may or may not be a problem, and if you use the filter settings in Appendix B, we already anticipated this very issue and built filters to suit. But you do need to be aware of this issue.

Another method that you can use, dependant upon your audio MIC pre-amp, is roll off some of the bass using your MIC pre-amp. Both the Behringer MIC-2200 and the ART MPA Gold, the ones that Jim and I use respectively, have a low cut filter which allows you to roll off the bass quickly. Last but not least would be to access menu 9-0, advance your processor offset up to +.100 or +.200, and then turn your processor on, leaving it at around the 10 O'clock setting. Either of these methods will facilitate rolling off the bass thus providing a more balanced audio spectrum for the complaining station.

Just keep in mind, when talking to a station that is not ESSB capable, you will not sound natural to them, and they will notice it. So if you get these kinds of complaints, do not think that there is something inherently wrong with your equipment. Just re-adjust your carrier or turn on your processor, or switch your EQ to a more pinched up setting, and that will resolve the problem.

6.7 SpectraPlus & Test Tone Generator Tips and Tricks

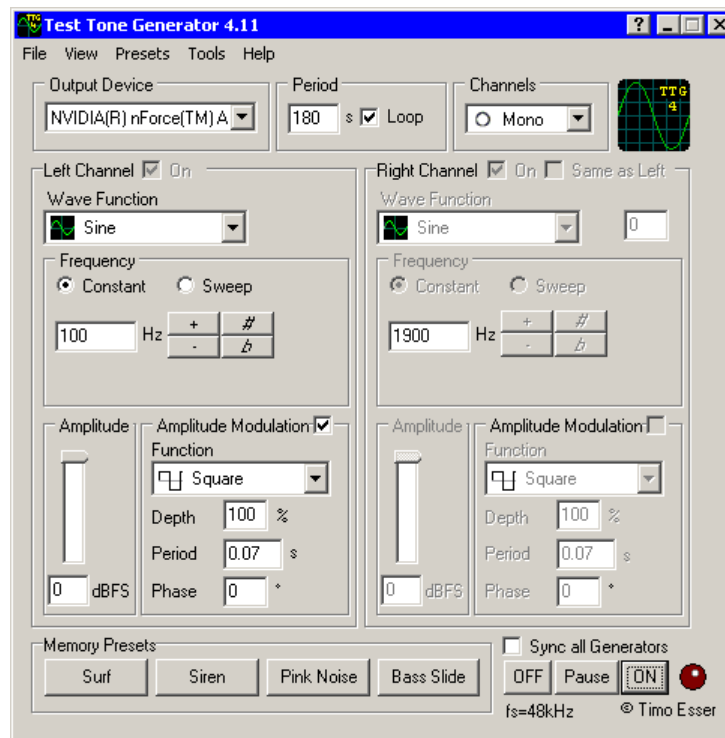
Both SpectraPlus and Test Tone Generator are very powerful tools and well worth the money for the licensed version. Not only can they be used to setup your radio, as was discussed earlier in this document, but they can also be used to enhance your station capabilities.

6.7.1 Pulse Tuning using Test Tone Generator

Ever tried tuning an amplifier using a constant CW signal from your radio? Did you know that when doing this you are attempting to tune your amplifier using full 100% duty cycle for operations in SSB, ~30-50% duty cycle mode? Doesn't sound quite right, does it? When you tune your amplifier using a constant CW signal, you are not only overloading your power supply but you are overdriving your valuable and expensive tubes as well. This is not good by any means, particularly if your tubes cost several hundred dollars. So I tell you what, lets tune your amplifier using a constant CW signal while it's terminated into a dummy load with a peak reading wattmeter in-line. When tuned, I want you to switch over to SSB and speak into the microphone. While doing this I want you to watch and record your plate current peaks. Once you have it recorded, let's tune that amplifier another way, by pulsing the amp.

NOTE – You probably already know this, but I feel compelled to reiterate this piece of the process. You should tune your amplifier for full power out, and then back the exciter down such that you are transmitting at the power level you want. This will help to ensure that your amplifier is operating linearly.

This procedure assumes you have configured your equipment similar to that of Figure 18; your computer audio is being sent to the mixer, thus capable of being transmitted. Open up your Test Tone Generator and set it up like Picture 31.



Picture 31: Pulse Tuning with Test Tone Generator

Once you have Test Tone Generator setup like above, you need to turn on the mixer channel that your computer audio is terminated on, and turn the mixer channel off that your rack gear is terminated on. If you do not do this, then you will be mixing both the MIC input with that of the Test Tone Generator, thus sending the combined audio to your radio. Now click the "ON" button on the tone generator, key your radio into a dummy load, and adjust the mixer channel level until such time that you at least match the

modulation level of that when you normally modulate the radio with your voice. This can be determined by watching your ALC. Once you have it adjusted such that it at least matches, un-key the radio, place the amplifier in-line terminated to a dummy load, and then re-key the radio and adjust your plate and load until you achieve max output per your in-line peak reading watt meter, then advance your load a little until you see the grid dip a little but your RF power remains the same. Now talk into your microphone and tell me what your plate and grid current is.

***NOTE** – You should set your computer’s audio card output to around mid-level. This will ensure that you are not over driving your sound card thus sending out a clean signal from Test Tone Generator. Once you are satisfied with your signal quality coming out of the sound card, then adjust the mixer level to properly drive your Mark V.*

Sure there are other devices that you could purchase out on the open market that can do the same thing. Even the Mark V itself, setup properly in CW mode, could do this on its own. But what I dislike about both of those methods is that we are using a pulsing carrier wave to tune the amp. In other words, we are still not tuning the amp while in SSB, and the CW generated at a rapid pace can be quite audibly annoying to anyone listening near by. Using Test Tone Generator, we are able to tune up the amp while in SSB using a less obtrusive audio tone, and have more flexibility in how we can pulse tune the amp. Besides, I think the shareware fee for this piece of software is cheaper than buying a dedicated pulse tuning device, and the software has a lot more features and functionality than that pulse tuner ever will.

One last comment about this subject, you have just been shown a different way of tuning your amplifier. This is not to say that this is the way you should ultimately tune your amp, but that it is an alternative method of tuning that has worked well for Jim, I, and others.

6.7.2 Tweak your Audio Chain using SpectraPlus and Test Tone Generator

By using these two pieces of software, you can sweep your audio rack from the microphone input, all the way through to the mixer, prior to the audio being injected into the radio. This is beneficial to you because you now have the ability to see the whole audio spectrum picture after either making slight changes to your audio equipment, or how movement of equipment within your chain will affect your audio prior to getting on the air. This technique can also be used to adjust each individual piece of equipment within your rack in order to obtain a flat tonal response, which is great if you are initially setting up a new audio chain. Then exclusively using SpectraPlus to monitor your transmitted audio, you can ensure that your tonal output is flat all the way through the Mark V as well. All of this can be done by using different configurations based upon Figure 18. For example, instead of injecting the computer audio into the mixer, the final piece in your chain, you could inject the audio into your MIC pre-amp, or in the middle of the chain. Ever have anyone tell you that there is some kind of background anomaly in your audio? Using SpectraPlus to monitor your transmitted audio, you can see what the complaining station is talking about and make appropriate adjustments on the fly. SpectraPlus is a software based audio spectrum analyzer, and will show you the offending frequency or frequencies. The biggest point to take away from this is that you have some serious tools at your disposal to help you keep that gear in tiptop shape. The usefulness and possible applications of these tools is only limited by your imagination.

7. Closing

We have finally reached the end of this beast. When Jim and I first started putting this document together, we had no idea that it would grow to this size. But it seemed we kept coming up with new things to add, things that we take for granted in our day-to-day station operations, but may not be

common knowledge. So we just kept adding. Neither Jim nor I are by any means technical experts on the Mark V and we do not have all of the answers for all of the questions that are out there. But we do have a desire to research, test, and learn. With that being said, we view this document as a "Living Document". In other words, look for new revisions from time to time. Even before this revision went public, we had already started a list of things to test, and add to the next revision. The idea here is dot releases, such as 1.1, or 1.2 will signify small corrections and minor changes. A major release, 2.0 or 3.0, will signify completely new hardware modifications. So, if you have tips & tricks that pertain to the Mark V, please send them our way. You never know, it may end up in the document along with proper recognition.

And I guess this is where Jim and I leave you with some parting thoughts. We have given you the keys to the ESSB Community through the Mark V. If you did everything correctly, you should be able to spread your wings and be heard in full fidelity. We only ask a few things from you in accepting our contribution:

1. When spreading the wings, be cognizant of those stations adjacent to you. Do not attempt coming into a 3Khz hole on the 20M band with a 4Khz band-pass or greater. This will only make trouble for you and will provide for a bad impression for the rest of us ESSB connoisseurs.
2. You will get heckled from time to time, that is just the nature of our niche in this hobby. When this happens, do not try and convince the other station that they are wrong. There is a reason why they are heckling you, they do not understand, and they do not want to either. Just be polite with them and drive on. By the same token, if the station is looking for information on why you sound so good, or how you are accomplishing the ESSB transmissions, give it to them, in volumes. The more ESSB connoisseurs the merrier, and the louder our voice will be in the grander scheme of things.
3. While playing in this community, if you do any experimenting, modifying, or anything else that improves the overall quality of your signal or operation, give back to the community. It is real simple, just document it like we did and find someone who is willing to post the information on the NET. Heck, I contacted Mike (WZ5Q) out of the blue, and he was more than happy to post this document within his web site. Or, if it pertains to the Mark V specifically, you can email Jim or I, and it might end up in the next document release.
4. Last but not least, if you hear Jim or I on the air, stop on in and say hello, you have an open invitation. We would love to hear what you have done and what your experiences have been, as well as any suggestions that you may have.

One last comment before we let you go. There are several things within this document that could be viewed as opening the proverbial "can of worms", and we realize that. Just keep in mind that our intention was to only report that which we have actually performed ourselves, or have substantiated proof that works. Do not think that we are trying to convince you to change your operating practices. With that being said, we would like to say, "Take care", and we hope you enjoy your new capabilities!

Later,

Mike & Jim

8. Acknowledgement's

This is where we would like to acknowledge those folks who have helped us in this endeavor, whether directly or indirectly. Acknowledgment goes to the following people for being instrumental in the creation and publishing of this document:

- To my wife, Nadine (W5CUL XYL), for putting up with me and my obsession. Lord knows what goes through your head knowing that I play around with new technology all day at work, and then come home and play some more as a hobby.
- To my wife, Rae-Ann (VE7RF XYL), despite my late night obsession with this hobby, you understand and support my passion for it.
- John Anning (NU9N), for being the original ESSB beacon for many operator's before and after us. Not too mention coining the term that best describes the hobby within the hobby that we all know and love, Enhanced Single Side Band.
- Mike Peak (WZ5Q), for allowing us to post this behemoth of a document on his web site, not to mention taking the time to build the web pages for this modification. Mike, let us know if the hits/downloads and or emails get to be too many.
- Roscoe (VK3KRH), for assisting with the Mark-V Main & Sub Receiver component modifications.
- Greg (WB9DNZ), who really got me (VE7RF) obsessed with what could be potentially done with the MK-V, especially the TX BM and RX product detector's.
- John (K1DEU), who, unbeknownst to him, provided the information via his website for the AM modification.
- Joe (N3JI) and Bill (WB6BNQ) for providing additional sets of eyes in pin pointing typos and general anomalies. Thank you gentleman, we appreciate your attentiveness to detail!!
- And finally, Jim Thomson (VE7RF). Jim, I contacted you out of the blue, you did not know me from Adam. Yet you answered all my questions and confirmed my suspicions regarding the Mark V. Before I knew it, I gained a good friend. Man I love this hobby.

Appendix A: Web Sites with Background Information

ESSB Operators (Great Info)

John Anning, NU9N: <http://www.nu9n.com/home.html>

Mike Peak, WZ5Q: <http://www.wz5q.com/>

Larry Wassman, W3OZ: <http://w3oz.netfirms.com/>

Tyler Stampfli, KA0KA: <http://www.cleanrf.com/ka0ka/>

John Shaw, K1DEU: <http://hamelectronics.com/k1deu/pages/ham/index.htm>

There are plenty more, and I do apologize for not getting all of you on this list. But by going to the above web pages, you will find a plethora of links to other operators web pages, as additional web pages with great info on audio work.

Parts Sources

Jensen Transformer, Inc.: <http://www.jensen-transformers.com/>

Markertek Online: <http://www.markertek.com/>

Digi-Key: <http://www.digikey.com/>

Zzounds: <http://www.zzounds.com/>

Musicians Friend: <http://www.musiciansfriend.com/>

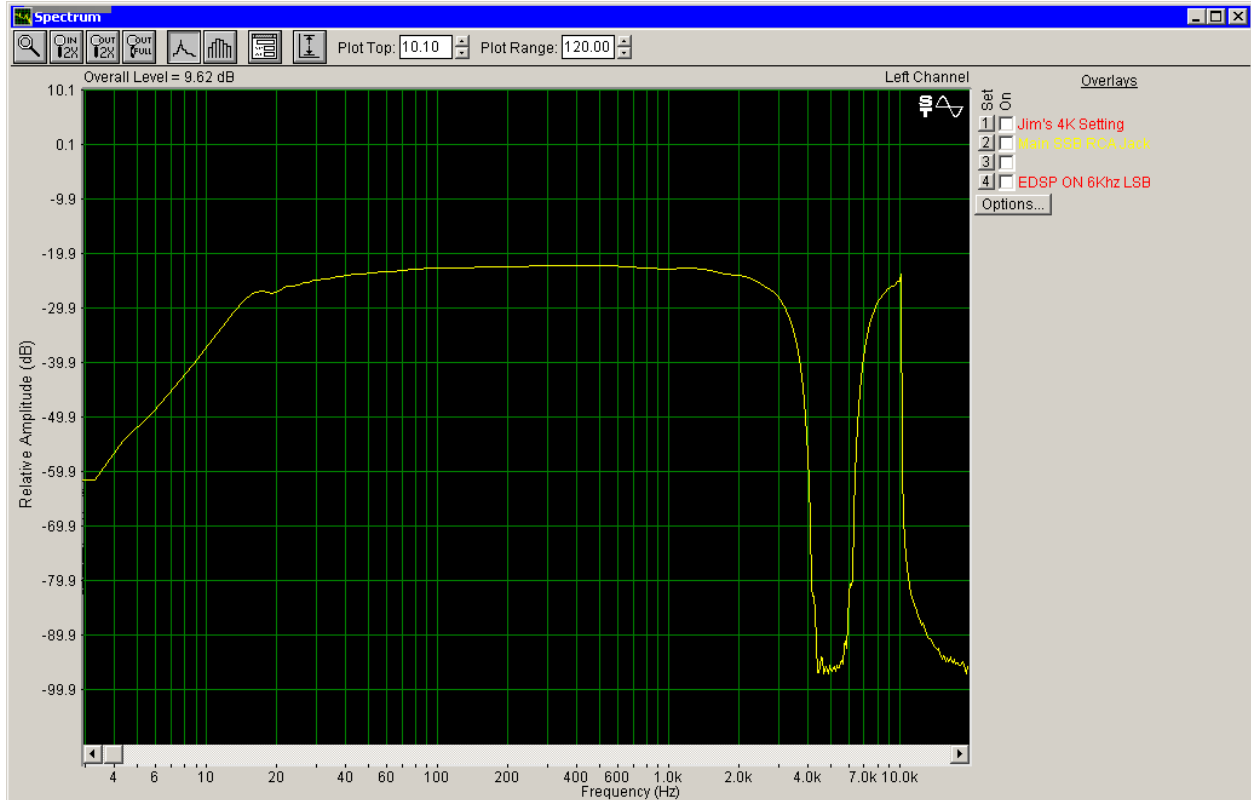
Heil Sound Ltd.: <http://www.heilsound.com/>

Additional Informational Sites

Sound System Interconnections: <http://www.rane.com/note110.html>

EQ by the Octave: <http://www.recordingeq.com/EQ/req0400/OctaveEQ.htm>

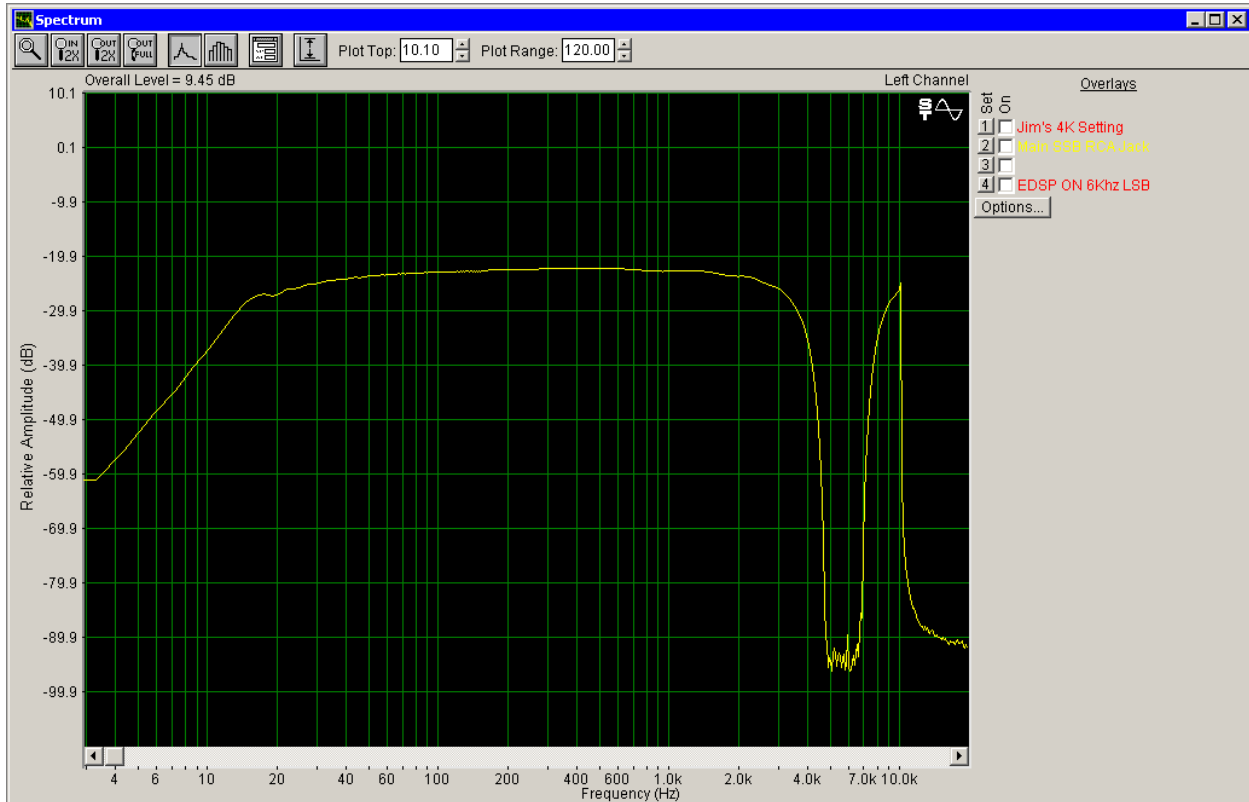
Appendix B: Suggested Filter Settings for the DSP-1124



Picture 32: DSP1124 3.5Khz Chop Filter

DSP1124 3.5Khz Chop Filter				
Filter	Frequency	Fine	Bandwidth	Gain
1	4K	+2	1/60	-48
2	4K	+4	1/60	-48
3	4K	+7	1/60	-48
4	4K	+9	1/60	-48
5	5K	-6	1/60	-48
6	5K	-2	1/60	-48
7	5K	+2	1/60	-48
8	5K	+6	1/60	-48
9	5K	+9	1/60	-48
10	6.3K	-7	1/60	-48
11	6.3K	-2	1/60	-48
12	6.3K	-1	1/60	-48

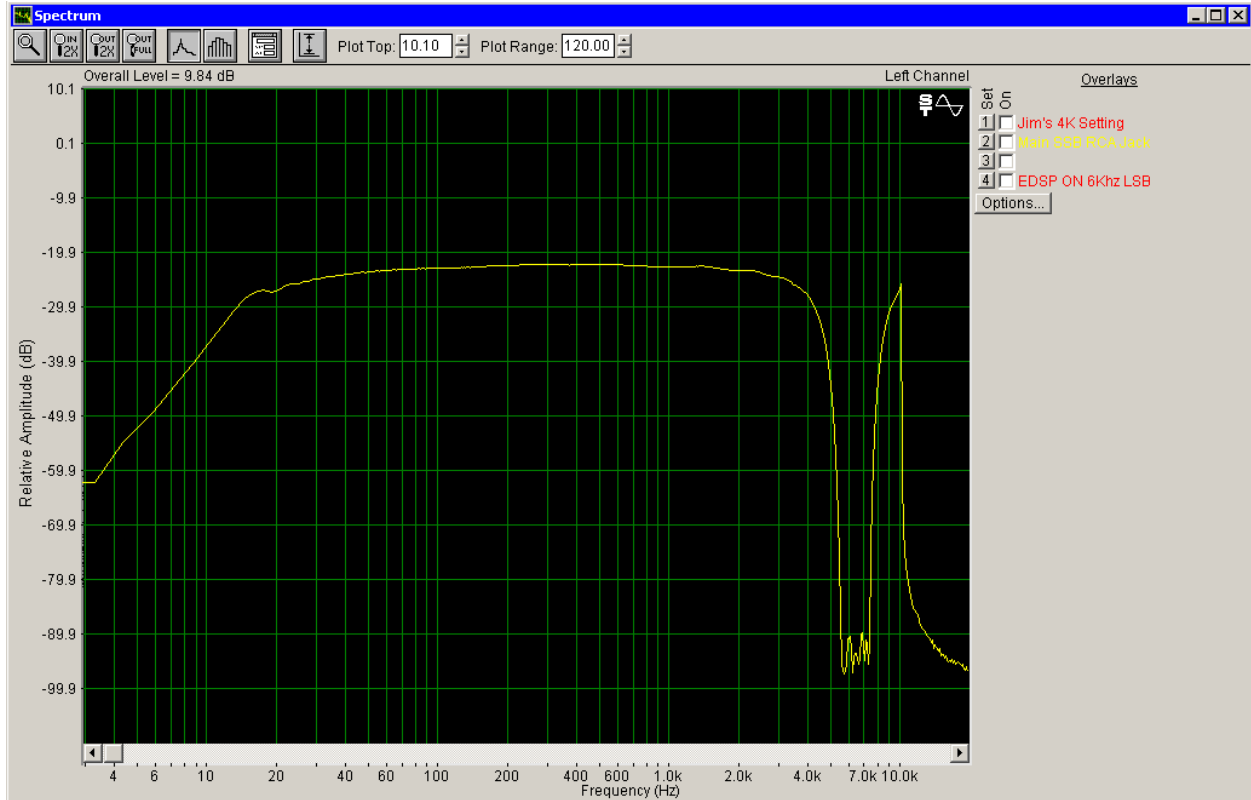
Table 6: DSP1124 3.5Khz Chop Filter



Picture 33: DSP1124 4Khz Chop Filter

DSP1124 4Khz Chop Filter				
Filter	Frequency	Fine	Bandwidth	Gain
1	5K	-6	1/60	-48
2	5K	-3	1/60	-48
3	5K	+0	1/60	-48
4	5K	+4	1/60	-48
5	5K	+8	1/60	-48
6	6.3K	-8	1/60	-48
7	6.3K	-5	1/60	-48
8	6.3K	-4	1/60	-48
9	6.3K	+0	1/60	-48
10	6.3K	+3	1/60	-48
11	6.3K	+6	1/60	-48
12	6.3K	+9	1/60	-48

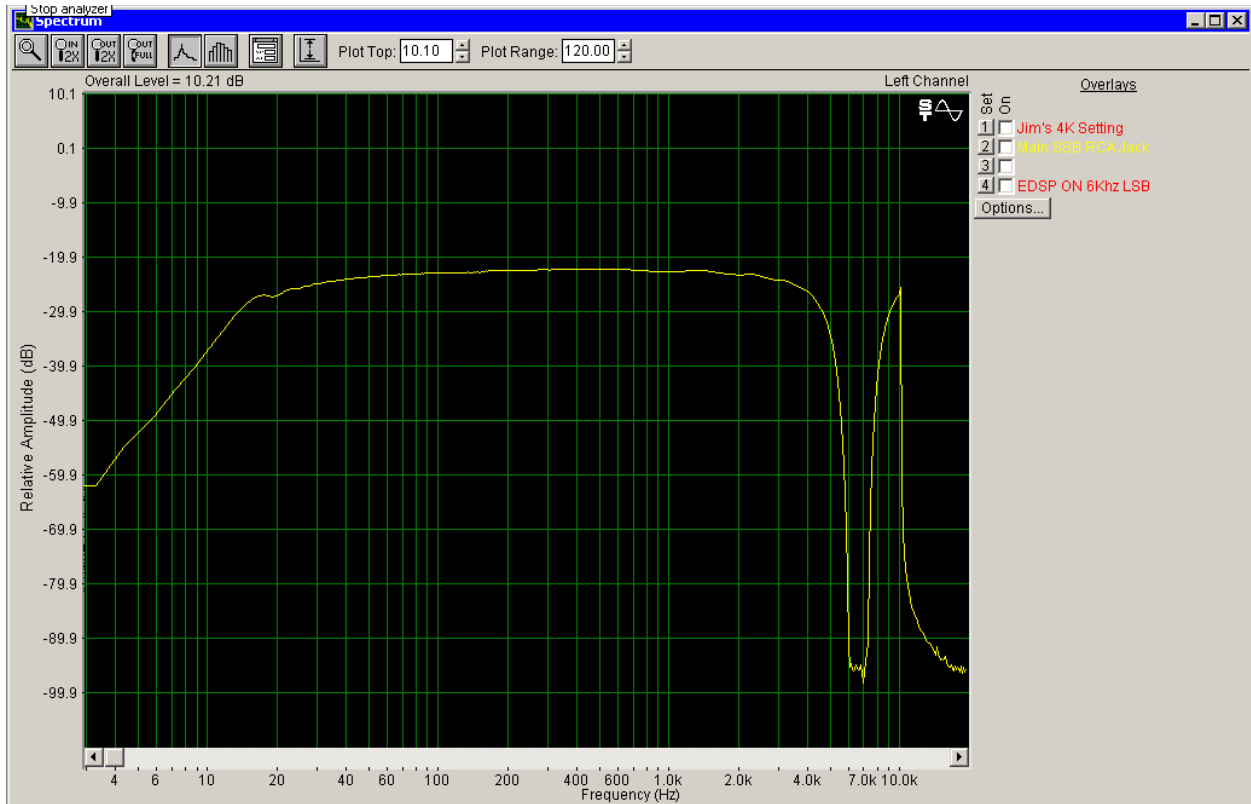
Table 7: DSP1124 4Khz Chop Filter



Picture 34: DSP1124 4.5Khz Chop Filter

DSP1124 4.5Khz Chop Filter				
Filter	Frequency	Fine	Bandwidth	Gain
1	5.0K	+8	1/60	-48
2	5.0K	+9	1/60	-48
3	6.3K	-8	1/60	-48
4	6.3K	-7	1/60	-48
5	6.3K	-1	1/60	-48
6	6.3K	+3	1/60	-48
7	6.3K	+7	1/60	-48
8	6.3K	+10	1/60	-48
9	8.0K	-8	1/60	-48
10			1/60	-48
11			1/60	-48
12			1/60	-48

Table 8: DSP1124 4.5Khz Chop Filter



Picture 35: DSP1124 5KHz Chop Filter

DSP1124 5KHz Chop Filter				
Filter	Frequency	Fine	Bandwidth	Gain
1	6.3K	-4	1/60	-48
2	6.3K	-3	1/60	-48
3	6.3K	-1	1/60	-48
4	6.3K	+1	1/60	-48
5	6.3K	+2	1/60	-48
6	6.3K	+5	1/60	-48
7	6.3K	+8	1/60	-48
8	8.0K	-9	1/60	-48
9			1/60	-48
10			1/60	-48
11			1/60	-48
12			1/60	-48

Table 9: DSP1124 5.0KHz Chop Filter