K7SRG

COR / audio board (card) version 5.4

(for Spectra-Tac version) by Karl Shoemaker

Introduction:

This document is written to include interested people in serious construction of a quality product. Its rather technical however, if you have a basic electronics background with some repeater building experience this should not be an issue. Understanding schematic drawings is required. Allow plenty of time to construct each unit, especially the first one. No free technical support is available however, some printed documents are available on an occasional bases, for a modest cost for P & H. The project is designed for amateur radio (not commercial) and is open for discussing, changes and improvements without notice. Should you feel qualified you are welcome to deviate from the Author's design. Images in this document may be used to illustrate a point only and may have been taken at different stages of research and development therefore, may not show the end “product” in some cases.

Overview:

This board interfaces a receiver to a controller or transmitter, while it performs basic link or repeater functions, except for an IDer. All parts are solid state with no moving parts and no mechanical relays. Emphasis on size, simple design, parts availability and easy modifications limited only to your imagination. Depending on the application you can leave out some parts, while strapping for others, such as COR delay, and polarity on COR and PTT output. There are some extra pads on the board for this purpose. The board's COR input is high impedance therefore, should not affect the squelch circuit of the receiver. The PTT outputs should key any modern transmitter.

"PTT 1" is for every squelch open, for example, controlling audio switching in a patch or to key a remote base or link transmitter. U3 and PTT 2 is left out in this version / application because there’s no tail. The transmitter disable control is an active low input.

History and this Version:

Designed occurred in the early 1980's by the Author. Older methods were used in the circuitry such as a passive potentiometer array for equalization and the LM-386 for audio amplifiers. Some of these "old school" boards were still in service as of 2016. The more recent versions of 5.x were scrutinized by the Author as well. This was based on gathering information and constructive input from users and repeater builders as well. In the 1990's these versions utilized a better, two stage audio equalization, and the quieter, LM-324 for the audio amplifiers and other logic circuits. Multi-turn pots were also used for easy alignment (no backlash problem). Since then the Author further analyzed these features for future SRG projects.

In the latter years-versions of this board were realized and documentation for each version can be found on SRG’s web site. This (5.4) version is for auxiliary (downlink) operation of a remote (satellite receiver driving a downllink transmitter.

Note: In this document, any font in blue indicates a guess and/or not verified at the time of publication.

Acronyms, Definitions, semantics and Theory basics for Telecommunications:

Some of this material may not be popular reading for hobbyist however, is necessary to maintain a complete understanding of the project at hand. “Layman” terms will be used, when practical, to make reading a little more “fun”, at the expense of occasional rough calculations and other “rounded” off math figures. To be very clear on this philosophy, we will start with the basics. Humans wish to communicate since the cave-man days with grunts. A few million years later with smoke signals. A hundred years or so ago with wired telegraph (1800’s) and wireless telegraph (1900’s). In the 20th century voice finally was realized. In the 21st century better sounding, analog voice, then data and digital voice was realized. Only analog communications/transmission for Land Mobile Radio (LMR) will be covered in this document.

Radio systems send intelligence (voice, data, etc.) by modulating the originating transmitter and decoding (detecting) this modulation at the far end receiver back to something usable to be understood. How well this is understood depends greatly on how well the system is set up. Just about anyone can "throw" a system together to make it work, somewhat.

Amateur radio can develop the art of radio and improving operating practices in this area. This can set a good example for others, including the commercial industry, to what some amateur radio systems are capable of doing and to provide public service communications in time of need. This includes the technical side, to produce a high performance repeater and/or link.

A “repeater” is a generic term for user’s signals to be received (input) and retransmitted (output). This greatly increases radio coverage, for a single-site, conventional repeater. Extended (user) coverage can be realized by linking several repeaters together. Further user coverage can be realized with a voter system and simulcasting as well in analog systems.

Most radio systems in the VHF, UHF (and microwave) are line-of-site for the radio paths. On the ground a path has limited range because of obstructions which attenuate signals. From high (remote) sites greatly increase this because most of the obstructions are gone.

A “link” is a one-way transport method for repeater support, such as the remote receivers on a voting system. For example, a repeater’s (input) receiver may need to be “downlinked” to a central control point, such as a voter or connection to the outside world (telephone, internet, etc.). From this control point the system output can be “uplinked” back up to a high transmitter (output) for the users to enjoy wide coverage of such a system. In this case would be a multiple site repeater (system of links, etc.) In conclusion, three factors improve a conventional analog radio system:

- Repeater; to “relay” user signals.
- High location; get away from obstructions.
- Voter system; easy user access, especially with portable-low power subscriber units (users).

A typical (commercial) system uses the audio portion 300Hz~3KHz for repeaters and links. With several links this produces “tinny” and distorted audio. In some cases squelch and signaling circuits produce signals that are annoying and fatiguing to listen to. Because of user tolerance and ignorance this sets a (bad) precedence of what a system is expected to be. This document covering system performance will be somewhat different. The Author’s design and specifications call for a better way, and is practiced in all SRG projects such as this one. For example, “flat” audio, better squelch and other signaling practices are utilized. This keeps a large system nice to listen and operate and may set examples for other groups to improve their systems. It also calls for good technical management.

For one, technician organization and discipline is necessary. Plan on what you want to do for a system design and stick to it. Force yourself to keep good practices. One good practice is to establish level references. Some call these "benchmarks" or "baselines". While old methods used linear (microvolts, watts, etc) units of measure, design of this project and document uses logarithmic units. Once accustomed, it's easier to see the entire picture this way, when designing a system or checking system performance and keeps the guesswork out of troubleshooting a subtle level problem. References can be expressed with a few acronyms.

Test Tone Level   and   Test Level Point:

Test Tone Level (TTL) is referenced to tone that modulates a channel or path 100%. For a testing or aligning a LMR transmitter, receiver or path this would be a 1 KHz (1004 Hz for telephone work) for a FM (frequency modulation) system. Test Level Point (TLP) refers to a measurement point (normally on equipment) in reference to TTL. TLP provides easy reference to any parts of the system for measurement and alignment. 0 dbm is referenced to 1 milliwatt at 600 ohms. A 6-dB drop in (voltage) level would reduce the modulation in half, and so on.

Levels are stated in transmit-receive (Tx-Rx) order. Therefore, an audio (Voice Frequency) "drop" TLP of 0/0 would mean a Tx TLP of 0-dbm, Rx TLP of 0-dbm. For example, a transmitter AF input with a TLP of 0 dbm, with a TTL of 0 dbm tone input, would fully modulate the system. If the far end receiver was set up the same, its output would be a 0-dbm tone as well.

Absolute levels are specific-measured (operating) levels, not to be confused with TTLs. Sometimes operating levels are not at TTL. In this case, a level would be so many db "down" from TTL, or just called "xx down". For example, CTCSS (sub-audible) tones normally are 18 db down. (1/8 deviation from voice, or 18 db down from maximum voice and/or TTL).

To avoid technician confusion two sets of numbers are sometime used in diagrams and on the physical equipment's ports or I/O connections. Non-parenthesis figures are (absolute/actual) fixed operating levels, and as mentioned before, may be at different levels from the TTLs. Figures in parenthesis are the TLPs, which is explained below.

Levels below 0 dbm are negative, while above are positive. Take this into consideration when working with system gains or losses. Normally, the negative levels have a minus in front of the number, while positive (optionally) have a plus sign. This is also true for absolute levels (as opposed to TTLs). This method is used for most any AC frequency (audio or RF). For example, many transmitters run a +42 dbm while most receivers' sensitivity run a -117 dbm for 20 dB quieting.

Other terms:

RF or AF ports at the **T**op **O**f **R**ack are considered "TOR". This is all equipment in/on the station's cabinet or rack. External equipment from TOR is later figured for a system performance (losses or gains). This may be RF lines, a combiner system or tower antenna(s). TOR levels are referred in the order of the transmitter and receiver (Tx and Rx, respectively).

Single digit numbers of "1" and "0" in brackets (" [ ] "), are not to be confused with TLPs. In this case these 1s and 0s identify the logic state of a gate, or other TTL/CMOS I/O driver circuit, and so forth. Another aid to avoid confusion between logic states and a TLP is that the latter normally would have a " + " or " - " before the number (as earlier mentioned). For example, a TLP of -14.8 is the audio input controlled by a logic gate of [1], being a normal logic "high". One last word on the logic state; The brackets indicate a state in normal standby/no activity condition. As a side note, "TTL" mentioned above has nothing to do with "TTL logic", a type of IC series.

Most "TIMM"s and AC voltmeter scales are in "dbm". When measuring across a circuit you may need to have the meter in bridge mode, being medium impedance as not to load down what you are measuring. In such cases a more accurate term of level would be "dBu". Having said this, dbm reading in bridge mode is still understood by most, for a specific (absolute) level measurement using log10 based numbers.

The term "COR" came from the old tube days of "Carrier Operated Relay" whereas, a tube receiver had a point, when its squelch opened, a tube (switch/valve) drew current through a relay's coil, to give some contact closure, to key the associated repeater's transmitter. Repeater stations in the early years were called "Relays" whereas, the station would "relay" a signal rather than "repeat" a signal.

As the solid state technology came in the later 1960's the COR term stayed with repeater operation. In addition, most modern equipment no longer had a mechanical “relay” used. Perhaps a more accurate term would be “Carried Operated Squelch” or “Carrier Operated System” (COS).
Both terms are correct and this gets down to semantics or content of a discussion:

- Modern technology used in the LMR field by amateurs and professionals alike.
- Recent repeater product terminology and it’s manuals.
- To avoid reader confusion; since they may expect the term of "COR".

After careful consideration it was decided to stay with the term "COR". Therefore, this and other SRG documentation will reflect this decision.

"CS" will be reserved to describe "Carrier Squelch" as a receiver's mode of operation, verses "TS", "PL" or "CTCSS" to describe a "Tone Squelch", "Private Line" or "Continuous Tone Coded Squelch System".

"SDI" means Signaling Decode Indicator (or Input). It's also similar to a CTCSS line out of a tone decoder. "HUB" means Hang Up Box. Motorola's uses a "closed loop" for mobiles and base station control. "AND squelch" means it takes both carrier + tone to activate a COR board, transmitter or system. AND squelch is also referred as a variable sensitivity squelch whereas, the squelch setting affects activity threshold. An "OR" squelch does not whereas, it "bypasses" whatever squelch setting, using only tone to keep it active (once the squelch is open on startup reception). More is discussed, later in this document.

Push To Talk:

The term "PTT" came from a button on a radio’s microphone. For this documentation PTT will describe an active going "low" for DC functions, such as transmitter keying ("PTT Input"). It also will describe a receiver's COR line driving a NPN transistor, with the open collector being "Receiver PTT Out", or just "PTT Out". "PTT 1" will describe this function however, with a buffer, such as the output of the cor/af board, which changes state for user signal change of status. This function would be used for audio switching, such as auto-patch audio routing. "PTT 2" will describe a buffered, and “hangtime/tail” output of the cor/af board, to keep a repeater's transmitter keyed up (AKA tail) for normal back-and-forth conversations of the users of such system(s). One or both types of PTTs may be time-out controlled.

PM/FM: (for a transmitter)

Frequency modulation is the common way to send intelligence in the LMR analog world. FM is also referred to "deviation" (of the carrier, at an audio rate). There are two ways to frequency modulate a transmitter, phase modulation (PM), AKA indirect, or (direct or true) FM (frequency modulation). PM is the easiest design with good frequency stability however, lacks audio response. PM has “natural” pre-emphasis which works well for LMR standard. On the other hand, (direct) FM has much better response (flat audio) at the cost of more complex engineering to keep stability. Also, FM needs additional pre-emphasis. With modern synthesized/PLL transmitters this is major consideration. However, later technology-design has allowed direct FM to perform well in LMR systems.

The MI (modulation Index) for a PM signal is always changing, especially for voice traffic. MI is mentioned because FM causes side bands to be created above and below the carrier and takes up bandwidth on a particular frequency, or sometimes called a “channel". Modulation and deviation are the same results when talking about FM. Maximum deviation of 5 KHz means 5 KHz above the center frequency and 5 KHz below the center frequency, making a total bandwidth of 10 KHz possibly including side bands. Radio technologies have different bandwidth standards (for maximum deviation) such as:

- FM radio broadcast of 75 KHz
- TV (analog) aural of 25 KHz
- Legacy cellular of 12.5 KHz
- Legacy commercial/government (LMR) VHF-UHF of 5 KHz (and most amateur).
- Current commercial LMR of 2.5 KHz
- Point-point microwave using (legacy) frequency division multiplexing about 5 MHz, in many cases.

While its good to be aware of these different bandwidth standards only amateur radio standards will be covered in this document. Crowded parts of the U.S. and abroad may use the “narrow band” standard of +- 2.5 KHz. It’s believed the reasoning behind the narrow band is less adjacent channel interference at the cost of lower performance in some cases. The Pacific Northwest VHF bands are still blessed in 2020 with the 5 +- KHz standard and is the standard for SRG projects such as this one.

A quartz crystal is normally used to control the frequency of an oscillator. A variable capacitor across the crystal can fine-adjust the frequency in the form of “warping” it. The fundamental crystal frequency will be converted by multiplying its frequency to obtain the (final) operating frequency. For example, a typical LMR VHF transmitter would be 12 times; or a tripler, driving another doubler, driving a final doubler. (Fc=12 MHz x 3 x 2 x 2 =144 MHz). It’s then amplified to a usable level for transmitting over the air.

Transistors and diodes have a P-N junction inside the case. The former can be used as an amplifier or switch with a potential (voltage) applied to create current flowing in the forward direction (against the schematic diagram arrow).

They also can be used as a variable capacitor. The P-N junction on either device has a “space” in the middle in the form of capacitance called the “depletion zone”. By applying a DC (reverse) voltage across this zone will affect it. This is also called “bias” across the zone. More reverse bias results in more space, thus, causing less capacitance. In a RF circuit this can mean higher frequency, in general.

By applying “intelligence” in the form of audio (AC/voice) across the zone will cause the RF circuit to change in frequency at the same rate, thus, creating frequency modulation. The bias is set up for a fixed value to keep the voice operating in the linear range of this device. This will create good symmetry (waveform) on a frequency modulated RF carrier. This is especially true (no pun) for true/direct FM.

Special diodes are made for this purpose, called a varactor diode or “veri-cap”. They come in various specs, for capacitor ranging 5 ~ 100 pf. Typical is 10 ~ 13 pf for LMR.

Most PM transmitters have the veri-cap diode in series with the crystal causing a phase difference on the fundamental frequency, while most FM transmitters have the diode in parallel to the crystal causing a (direct) frequency change on the fundamental frequency. For FM transmitters, most have the anode to (common) ground.

FM is also used for compensation against frequency drift from temperature changes of an oscillator circuit. In some cases a transmitter uses both PM and FM for audio and compensation, respectively, or two stages of FM, for both reasons as well. Sometimes both circuits are contained (with the crystal) in one module, as in the case of the GE Mastr-II transmitter’s “ICOM”. This way the channel device (element) can be set up (compensated) for each crystal for best performance.

Frequency multiplication also multiples the modulation of the fundamental frequency. Since this arraignment multiples the crystal frequency 12 times it won’t take much capacitance change to obtain 5 KHz modulation (deviation) or temperature/frequency compensation, at the operating frequency.

Flat audio – The long explanation:

As previously discussed, most stock/conventional two-way radios are designed for single path operation, with it's own pre-emphasis, deviation limiting (clipping) and receiver de-emphasis, and "forgiving" squelch operation. Each time a repeated signal occurs some reduction in signal quality happens. For multiple links (long haul) these stock radios can add gross problems, such as excessive distortion, audio frequency response being very poor and very long squelch bursts. All these conditions will cause a system to operate badly and be rather annoying and fatiguing to listen to. Fortunately, these conditions can be corrected.

Some of the problem is human ignorance, interpretation, perception and semantics when discussing audio processing (or not). To fully understand proper audio will take some careful thinking. The other point to keep in mind is the frequency range specification, such as 300 Hz ~ 3 KHz response for a conventional voice circuit, (which some would call “flat”) or 20 Hz ~ 5 KHz (which is more “flat”) or somewhere in between.

Perhaps a better explanation to clear up this argument would be to call the latter “extended flat audio” (EFA). Now, let’s go over some audio processing methods:

There are two types of audio frequency processing when it comes to FM radio equipment; which is conventional (emphasized) and flat (modified or specially designed). One of the standards for FM operation is to improve reception (audio) quality by improving the signal to noise ratio. Consider these two factors:

- Signal; meaning, the intelligence quality of voice or analog data reception.
- Noise, meaning noises from all other sources of this type of communication circuit.

Most of the noise is in the high end of a standard communication channel of 300 Hz ~ 3 KHz; also known as a voice channel. Therefore, by processing the high end of the voice channel can improve audio reception quality. This is normally done by emphasizing (increasing the level) of the high end at the <u>originating source</u> audio by 6 db per octave and de-emphasizing (decreasing the level) of the high end of the <u>far end</u> audio at the same slope.

This is a similar method to “Dolby B” technology used in stereo/hi-fi sound recordings for music listening; except its not companded (compression during recording and expansion during playback). For LMR, the far end listener will experience apparent noise reduction; thus, better S/N ratio. This method is for simplex operation since this processing is done only in the subscriber units. While this may work for a single path, repeaters and multiple links will need further understanding to produce a quality audio path.

<u>**Repeater** stations:</u>

One could use the audio from the speaker of a receiver feeding a mic. input of a transmitter. Since amateur systems can be modified without violation of type acceptance better points can be used. For example, the (flat) DPL (channel element) input is used in the case of Motorola LMR equipment. For the receiver the discriminator output is used. All receiver's discriminators should have great low-end response however, (due to IF filtering restraints) the top end always rolls off too soon. There is also the impedance-loading and level issues to deal with in some receivers. This and other SRG documents address this.

Most amateurs refer to “flat audio” with methods for a <u>single transmitter</u> or a <u>single receiver</u> to obtain quality. The key point is both components of the repeater station have to be the <u>same of one type</u> or the other; you cannot mix types within the same station and expect the (throughput) audio path to be flat. A repeater station with a flat receiver driving a flat transmitter will result in a flat audio path going through that type of repeater. On the other hand, a repeater station with a <u>properly</u> de-emphasized receiver driving a <u>properly</u> emphasized transmitter will also result in a flat path through that type of repeater for a standard voice channel of 300Hz ~ 3KHz. A flat repeater means the path will be transparent and not alter the audio frequency response. While some conventional station curves may have a sufficient for a single path voice transmission, most are not precise enough to be called “flat”; hence, the misunderstanding. The key point to remember is that the term “flat” should refer to path/circuit <u>performance</u> and not the <u>method</u> to obtain this.

<u>One exception</u>

If a repeater is truly flat for subscriber Tx to Rx path (reception) there is one exception for processing within the repeater station for “drop and insert” applications. In the case of flat equipment being used, there is a special situation where pre and de-emphasis is used in addition, to properly interface with non-radio equipment, such as a controller, voice synthesizer or the PTSN (Public Switched Telephone Network), AKA a phone patch. These sources are flat in origination therefore, need emphasizing to properly interface with subscriber (user) radios (compatible audio frequency response curve).

<u>**Deviation limiting** or clipping:</u>

Each time you limit deviation for each link in series will add more distortion. An alternative is passively repeating the audio 1:1. If you do have to limit, only do so at one point, such as the system's controller, user signals or system output transmitter (user receive). Another option would be to set the system limit at 6 KHz and let the system user's transmitters limit at 5 KHz deviation, to avoid audio distortion. Passive mode requires system management and user responsibility with your adjacent “channel” neighbors. This may require some enforcement on the owner's part. There are ways to "punish" or filter over deviated (and modulated) users however, is beyond the scope of this document.

<u>Squelch operation:</u>

FM receivers have large IF gain. At the discriminator there is plenty of noise available during signal absence. This noise is filtered above the standard voice channel near 8-10 KHz, amplified, rectified and DC amplified to usable DC levels. The higher audio frequency range is chosen so normal traffic (voice) won’t affect the squelch operation. This is known as a noise operated squelch, used in LMR-FM analog. A signal into the receiver that is stronger than the noise will "quite" the discriminator audio output, which changes the DC levels in the squelch circuit and turns on the audio amplifier to drive the local speaker for listening. A squelch circuit can also be used to key an associated transmitter; thus, making a repeater.

<u>A twist:</u>

Some FM systems use a sub-audible squelch system, better known as CTCSS (Continuous Tone Coded Squelch System). A carrier operated squelch can work together with a CTCSS to make either an "AND" or "OR" squelch. Companies produce repeater controllers and use this acronym in many cases. Other types of signaling (digital, etc.) can also be used to control a circuit or System. Therefore, the general term used here is “SDI”, for Signaling Decode Indication (or input).

“AND” squelch means it takes both a valid carrier and valid SDI (decode) to activate the squelch. “OR” squelch means a valid SDI (tone in most cases) decode will keep the squelch open regardless of the carrier squelch setting; thus, bypassing the squelch setting. An OR squelch is not desirable for amateur use because of the (annoying) long burst of noise that occurs after the input signal stops. AND squelch is best for amateur to avoid this burst. “OR” squelch, “reverse burst” (squelch tail eliminator) and other theory of operation is discussed in another document on the SRG web site in greater detail.

Stock radio receivers have (carrier) squelch constants (time for squelch to close and mute the audio path) designed for both fixed (base station) and mobile (moving station) signals therefore, are a fairly long (200 msec.) time for squelch closure. This is noticed by a burst of noise at the end of a received transmission. For a single site this is tolerable however, for multiple links (hops) this can quickly add up to something annoying to listen to. It also slows down switching paths, causing user collisions. For links, this problem can be corrected by lowering the R/C constants in the squelch circuits; thus, shortening the squelch burst. However, if they are too low the circuits will be unstable therefore, require some careful selection, which is discussed in other documents concerning link receivers, on the SRG web site.

Links are not intended to receive mobile (moving) signals. Therefore, this squelch modification will be transparent to fixed (links) station use, which should be full quieting, strong signals. Only multiple "clicks" would be heard with this modification. The remote user (input) receivers will still have stock squelch components therefore, will provide for moving (mobile) signal changes, plus, "cover up" the multiple link clicks. The result will sound like a simple, small, single site System.

For flat audio processing there’s a “cor/af board” design (by the Author) to work with most FM receivers. This board is “fixed” with soldered wires (or screws, such as the RF-IF board in the receiver). A “card” is removed simply by pulling it out, such as with the Spectra-Tac shelf. If the cor board is mounted on a card then the entire piece becomes a “card” thus, “cor card” (or module as the OEM manual calls them).

Other definitions, acronyms and other "shortcuts" are for practical reading and document space. For example, names may be truncated only after the **full name** is established. This avoids reader misunderstandings. For example, the parts list shows several manufacturers in truncated form, such as, Mouser Electronics (a major parts supplier) and may be later referred to as "Mouser" or “ME”, etc.

<u>Spokane Repeater Group:</u>

The Author is the founder of SRG, which is a non-profit organization for the development of equipment, operation and enhancement for the benefit of other amateur radio operators doing Public Service (emergency traffic) and other hobby type discussions. http://www.srgclub.org

K7SRG

The Project:

Theory for the Board

Since Amateur stations are not required to have as much splatter control with harmonics, (unlike commercial stations) this should not be a problem. However, you should be aware of any possible bandwidth limitations in your area, since there is a trade-off between bandwidth and system performance. This board was developed in the Pacific Northwest were we are blessed with 20 KHz spacing for repeater pairs. In other parts of the country with narrower spacing make your calculated changes as needed (as discussed earlier with TLPs).

For links, each time you limit deviation for each hop will add more distortion. If you only limit at one point, such as the system's output transmitter will make a much better sounding system. One option would be to set the system limit at 6 KHz and let the system user's transmitters limit at 5 KHz deviation. For "passive mode" option leave VR6 * at maximum (or leave out) and use VR3 to control the drive output, well below the clipping point as described later. This mode requires system management and user responsibility, which may require some enforcement on user's part. A circuit to "punish" over-deviated users is possible however, is beyond the scope of this documentation.

A word about VR3. The 5 Meg Burns pot might be hard to find. You can substitute with the 2M pot with some loss in gain of the second stage. 5 Meg was selected for a highest value. Anything much more would make the amp to go into the differential mode. Without the negative feed back resistance between pins 6 and 7, it's in the differential mode, which is used as a comparator, such as the COR input section. Voltage gain is the ratio of the negative feed-back and input resistors, then covert that to log scale if you need to. With R1 value and VR3 you can control the gain of both stages. Typical figures are in parenthesis on the schematic, (bridged) assuming the input TLP is 0 dbm.

Early versions provided a 455 KHz tunable LC trap for the old receivers. The Micor does not need the trap therefore, the pads are jumpered. The input TLP should be -20 dbm or higher. For lower inputs change R1 value, from the 1 Meg. to around 4.7 Meg. For the Micor uses a 1.5 M resistor. Also, if you don't need a squelch you can leave out Q1.

When the COR is active, U1 input translates polarity (depending on your jumper settings) and drives both the audio squelch and PTT circuits. When active, pin 8 goes low, turning off Q1 and letting the AF input through the two stages of equalization and amplification to the "AF OUT" to drive a transmitter or controller. Pin 8 also drives logic and timers, which drive the open collector PTT outputs, active going low, to key a transmitter or other devices.

U1-pins 12, 13 and 14 are set up as an 'and' gate, which require user activity, but not over activity. This for optional automatic control of a standard repeater per the FCC rules. This section of U1 is controlled by voltage dividers, for three possible conditions; standby, active and timeout. A forth (optional) condition can be realized by a low on “CON-1” possibly, from external control equipment.

In standby, pin 13 is higher than 12 and therefore, pin 14 is a "low". When an active signal causes pin 8 to go "low", pin 13 goes lower than pin 12, causing pin 14 to go "high" and starts the PTT output(s). If the signal stops, the TO timer resets. If the signal stays active too long, (i.e. 3 minutes) U4 times out; its output on pin 3 goes "low". Even though the signal activity caused U1-13 to go lower; now pin 12 is even lower than 13. This condition causes pin 14 to return to a "low" and turns off any PTT outputs. It stays in this status as long as the signal is active. When it stops the timer resets, so the next time activity starts will function any PTT outputs again. Time out range is 0-3:52 with a 100uf cap and 2 Meg trim pot. There’s other value options discussed later in this document.

* VR6 is left out in this version / application.

Setup: (general information first)

Load the board with all the components needed for this version/application. You will need to connect at least ground, power, COR input and AF input for alignment and testing. The other connections can be made on final assembly. Leave yourself enough wire length to work on the board, as you will be experimenting with different component values.

Start with the COR/COS (Carrier Op Squelch) point. Study your receiver's schematic or documentation for the best point and make that connection. For a COR that is low (or lower) on standby, then goes high (or higher), jumper A-D and C-B. This puts the COR buffer to an inverted configuration. For a CORs opposite of that, jumper A-B and C-D. In this case, will be version 5.4 for a Micor receiver driving a down-link transmitter therefore, jumper **A**-**D** and **C**-**B**. There is no “pull-up” resistor on this board because that voltage source is from another card (ACM). That way this cor board won’t inadvertently be activated if another card is pulled for testing. There’s a pull-down resister to insure this won’t happen.

Power up the receiver and board. The green power led should be lit. Remember that it takes about 5 seconds for the board's audio circuits to stabilize on power up. (Since repeater service normally is 24/7 on, this should not be an issue.) Adjust VR1 for proper trigger level when the receiver squelch is open (with noise being heard). Use the yellow led to watch the transition. Give yourself a little margin on this trigger point for stability. For the Micor RUI just set it at **2.50v** on pin 10. Cor voltage from the ACM should be 3.46v during activity. It’s possible this will be slightly different for each chassis.

Inject a clean 1 KHz tone and turn up VR3 to just at clipping point observed on the output with an oscilloscope. Tune the bias at pin 5 with VR2 for best even top and bottom clip on the output. Re-adjust VR3 as needed to fine tune VR2 adjustment. VR2 will be a one-time alignment when done. Pin 3 won’t need adjustment because it will be running well below clipping. You control its gain with R1 value, covered later. Next, inject a test tone into the receiver this board is being set up for. (0dbm0) Adjust VR3 to the proper TLP (level). SRG standard is 0 dbm. That’s 775mV RMS, for math challenged folks ☺

Other Notes for U2:

For a 7810 (U2), the maximum unclipped output of U1 is +10 dbm (bridged). You could operate U1 at higher voltages for example, with a 7812 (U2) which will drive U1 output near +14 dbm (bridged). However, U2 keeps out any small ripple that would be amplified on the system, so it’s best to use the 7810. Since the audio op amp U1 uses a single end supply with voltage dividers for the "+" you need some regulation headroom. Just observe the maximum operating limits of U1.

Flat Audio

Most receivers have high end roll off. This is a conventional method for commercial systems. If you want your system to sound really good (flat) you can extend the system's frequency response. First, plot the receiver's response on a graph, from 10 Hz to 10 KHz. This sounds a little extreme, but this will show how you are progressing. The board has two stages of equalization with amplification to bring the level back up to a usable level, called “EQ-1” and “EQ-2”. EQ-1 will flatten out the upper end; say, above 2 KHz, and EQ-2 will flatten above 4 KHz.

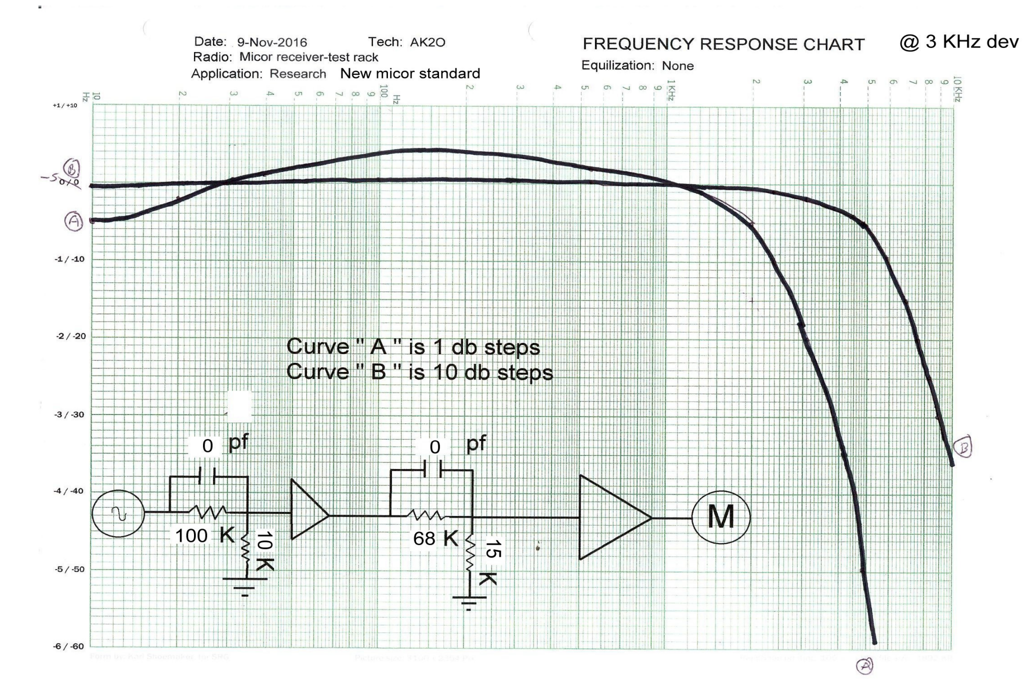
EQ values: (for any Micor versions; mobile, base or Spectra-Tac)

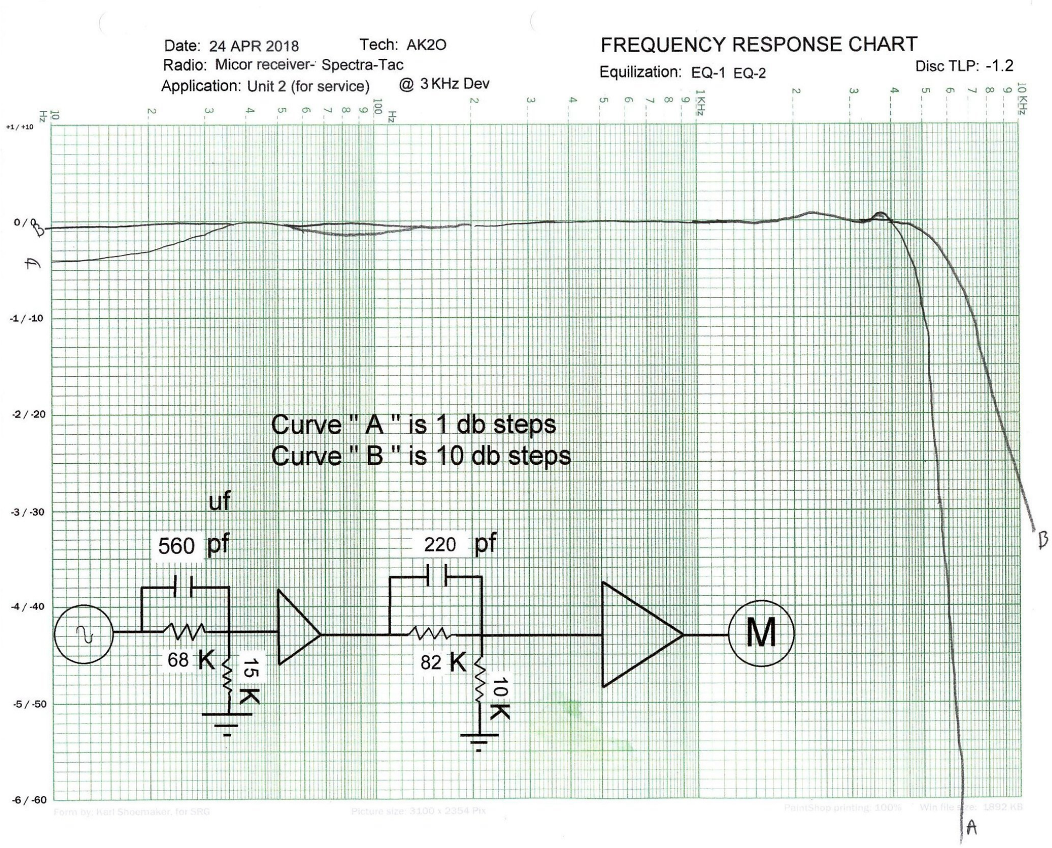
Original values were for EQ-1 series resistor 68K, series cap .0082uf, shunt resister 15K. For EQ-2 series resistor 68K, series cap 390 pf, shunt resister 10K. However, after long RnD the final baseline found to work best is EQ-1 series resistor is 68K, series cap is 560pf and shunt is 15K. For EQ-2 series resistor is 82K, series cap is 220pf and shunt is 10K. However there are a few “stubborn” ones that needed EQ-2 adjusted from 270-470pf cap and usually the shunt to changed to 15K. You will have to test each receiver to verify this. This extends the response out to 5 KHz before significant drop-off. Typically will be 2.5. After you re-plot the receiver's response should show a better (flatter) plot.

For other (non Micor) receivers you may have to try different cap & shunt values from the baseline.
Tip: If you are checking the levels use a medium to high impedance AC meter. Most TIMMS will load down the circuit 2-3 db even in the bridged mode. The HP 3555B meter used in the Lab loads down the

discriminator 2.6 db and the “AF output” 0.2 db. To get around this the (older tube-type) HP-400c meter was used to establish the cor board’s TLPs. It has a very high input impedance feature.

Here’s the receiver’s frequency response plots. The top one being just the receiver with no equalization while the bottom plot is with equalization with the final values just discussed. They were measured with the 3555 meter at the board’s output, as not to affect the loading of the receiver’s discriminator output. Line “A” is 1 db increments while line “B” is 10 db.





These tables reflect the previous discussions on theory and setup: *(note: for micor R1 is 1.5 meg)

Table 1

Input TLP	R1 value (R1)	Remarks
+5	470 K	Or a lower value
0	820 K	"
-5 *	1.5 Meg.	"
-10	2.1 Meg.	"
-15	5.6 Meg.	"
-20	9.3 Meg.	"

Table 2 - - TLPs with the receiver (RF)

Point of Measurement	Level	Remarks (U1 being a 7810)	Noise floor
U1 Input	-1.2		R1 is 1.5 Meg.
Squelch switch	-26.2	Q1 collector	-65.5
U1, pin 2 input	-65.2		
U1, pin 1 output	-2.1	With R1 value of 1.5 M	-58
Junction of 220pf and 10K	-20.8	Second "EQ" stage output	
U1, pin 6 input	-48	Mostly noise	
U1, pin 7 output (max-clipping)	+12.2	VR2 at maximum (clipped)	-40.5
U1, pin 7 output (no clipping)	+10.6	VR2 set just before clipping	Direct (no RF)

Table 3 - - TLPs without the receiver (direct input)

Point of Measurement	Level	Remarks (U1 being a 7810)	
U1 Input	0.0		
Squelch switch	-21.5	Q1 collector	
U1, pin 2 input	-58		
U1, pin 1 output	+2.6	With R1 value of 1.5 M	
Junction of 220pf and 10K	-16	Second "EQ" stage output	
U1, pin 6 input	-58		
U1, pin 7 output	+12	VR2 set just before clip point	

Table 4 - - Typical DC + voltage chart: (with power source at 14.20 v)

COR board Status	U1; pin 13	U1; pin 12	U1; pin 14	Q4 coll. (PTT-1)	U1 pin 8	U4 pin 3
Standby	6.83	4.35	0.007	Off (relaxed)	8.70	8.77
COR activity	2.339	4.35	8.53	On (forced low)	0.008	8.77
COR Timed-Out	2.339	0.039	0.007	Off (relaxed)	0.008	0.072
CON-1 active (low)	n / a	0.607	0.007	Off (relaxed)	n / a	n / a

Notes: AC measurements performed with an AC meter & scope probe, in bridged mode (high impedance). DC measurements performed with a Fluke 77 meter (high resistance) as not to load and affect the circuit measurement and establishment of TLPs.

Tip: If a high impedance AC meter is not available an oscilloscope can be used for linear measurements then covert them back to log10.

“AND” squelch:

Protected paths will have an "AND" type squelch, requiring both carrier and tone (CTCSS) to occur to be active. This will be a “closed loop” to keep the cor line “low”. Both signals (cor & decode) are paralleled making the “AND” squelch. When both carrier and tone is active then it allows the cor line to go “high”.

Use a generic tone decoder, such as the Comm Spec TS-32 using “Out-2” which is a normal collector forced low during standby and relaxes during a decode. The Spectra-Tac’s PLM that is (SRG) modified behaves the same way and is primarily used for this version. To switch to carrier squelch open the loop at the PLM or other decoder used.

Timer:

The time-out is useful for a “stuck mic” issue. When the cor board times out as not to block operation of a shared link path. This way the other receiver downlinks will still continue to work.

As briefly mentioned on page 8 the drawing (and parts list) specs for a 2M pot (VR5) with a 100uf cap. This will give you an adjustment range of about 0 ~ 3.9 minutes. The 2M pot is easy to find. If you don’t need that much range a 68uf cap will give you about 0 ~ 2 ½ minutes. If you can find a 5M pot using the 68uf cap will give you about 0 ~ 6 minutes. Obviously, a 5M pot with a 100uf cap will give you way more time range.

If you have external time-out equipment and don’t need this timer you can disable it by several ways:

- Installing JU2. This will prevent U4, pin 2 from reaching voltage threshold, thus, pin 3 will always stay “high”. The drawing shows Q2 collector has a jumper to ground. Physically, a good spot to do this is a solder bridge between pins 2 and 1 of U4 (green squares on the drawing. This method is good if you plan later, to enable the timer; just remove the bridge with an iron/wick and don’t need to keep track of extra parts to install.

- Leave out U4 and it’s associated components. Where U4 holes are left open, install a jumper from pin 3 to pin 4 (Vcc). This provides the voltage for the divider (two 10K resistors) to keep U1, pin 12, AND gate functioning. This method is good if your decision is permanent.

- Leave out U4 only. This method is good for a future change. Just plug U4 back in its socket. If the orange LED and it’s 1K resistor is installed that should be enough to power the voltage divider to keep U1, pin 12, AND gate functioning. The LED will glow slightly because of this. Instead of a “bug” this may be a feature to indicate this method is being used (without pulling the card to inspect it).

For any method CON-1 is still functional.

Other notes:

Technically speaking, this is a cross-band "repeater" but for a user/beginner to understand it's better referred to an "downlink" (down to the control point).

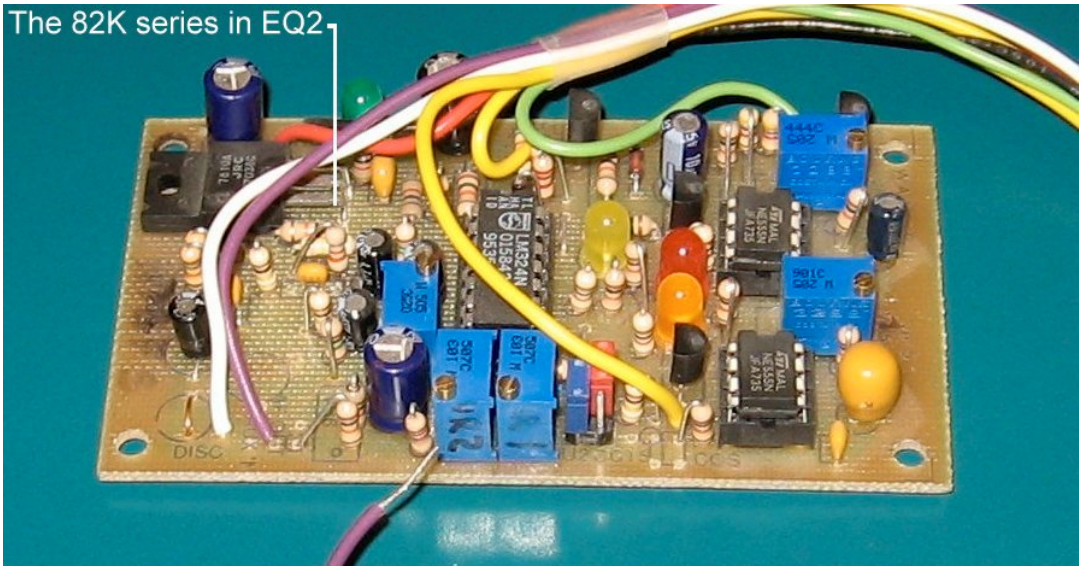
The "AUX AF" is a flat audio input (no EQ) which is inverted from the output and un-squelched. It could be IDer or status tone input . If its not used either ground the input or leave out the 1 Meg resistor to avoid noise being picked up and amplified.

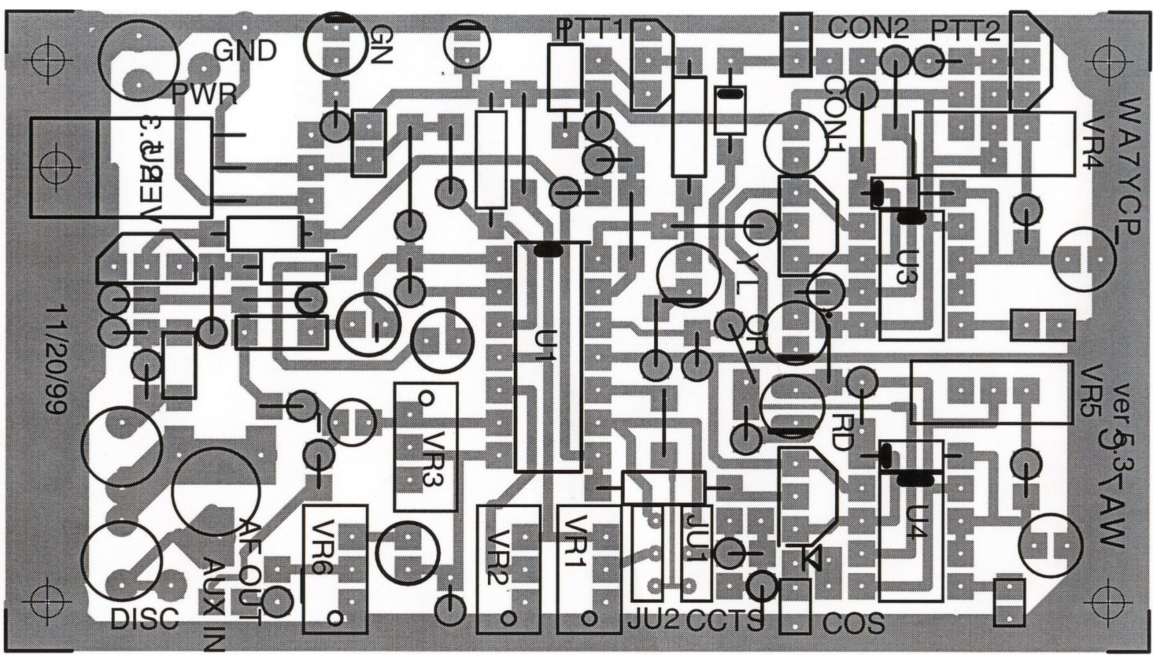
The LED’s for some of the IO is still being researched. This documentation will be updated when this step is completed.

Versions:

Here shows version 5.3 fully loaded and ready for install, along with the board layout (component side). This original version gives you a reference point. It's got both times and controls, etc so is intended for a conventional use, such as a repeater. Full documentation on this version can be found on the SRG web site.

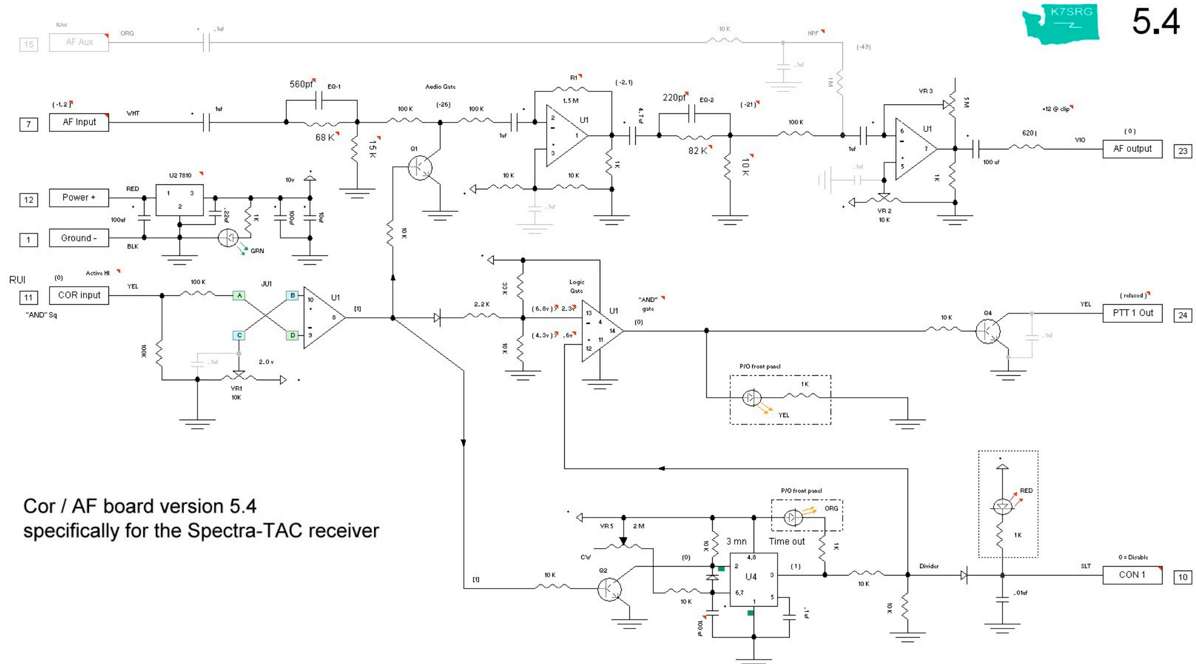
To note: The silkscreen (below) shows the tail and TO (LED) indicators as orange and red, respectively. Since red is a PTT/tail function, the two colors are now swapped on the boards.





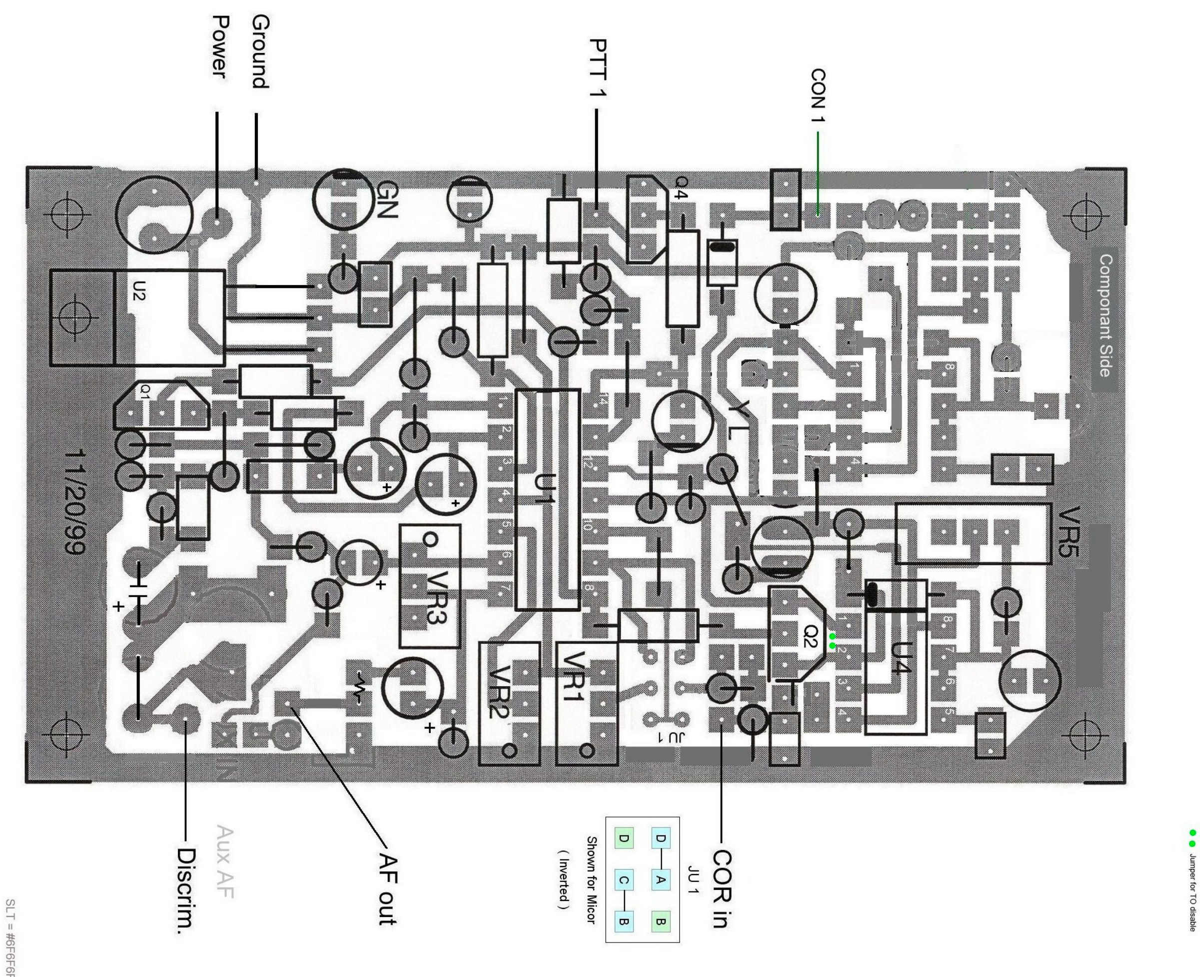
The next page shows version 5.4, which is for this Spectra-Tac project. Also, to note; some the indicators are not mounted on the board but are on the front panel of the cor card.

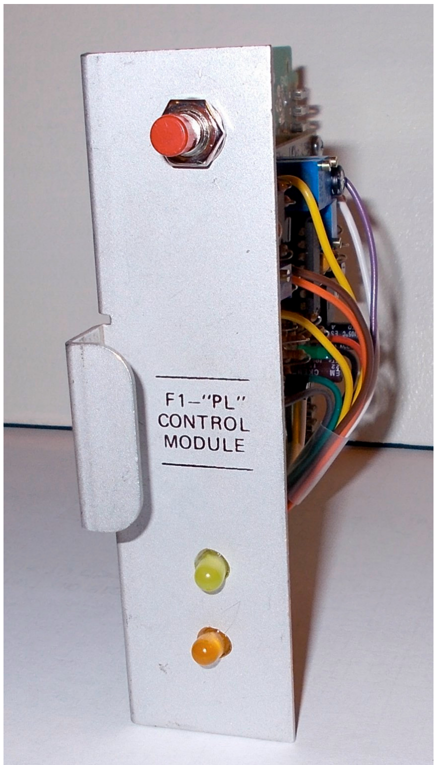
5.4



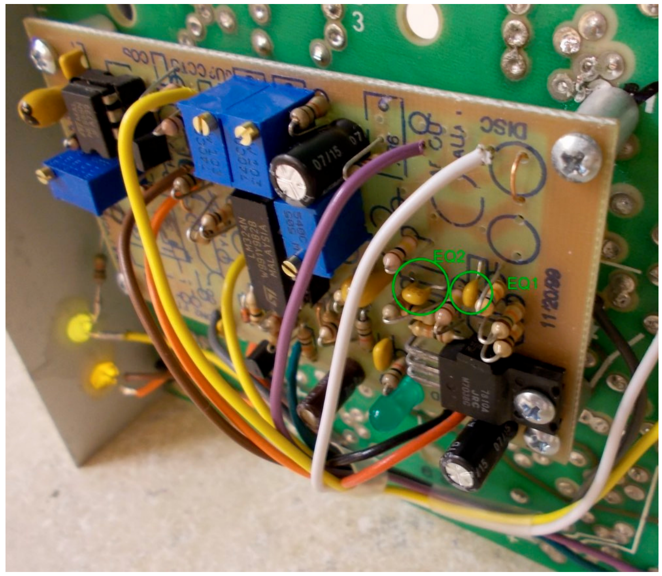
Cor / AF board version 5.4
specifically for the Spectra-TAC receiver

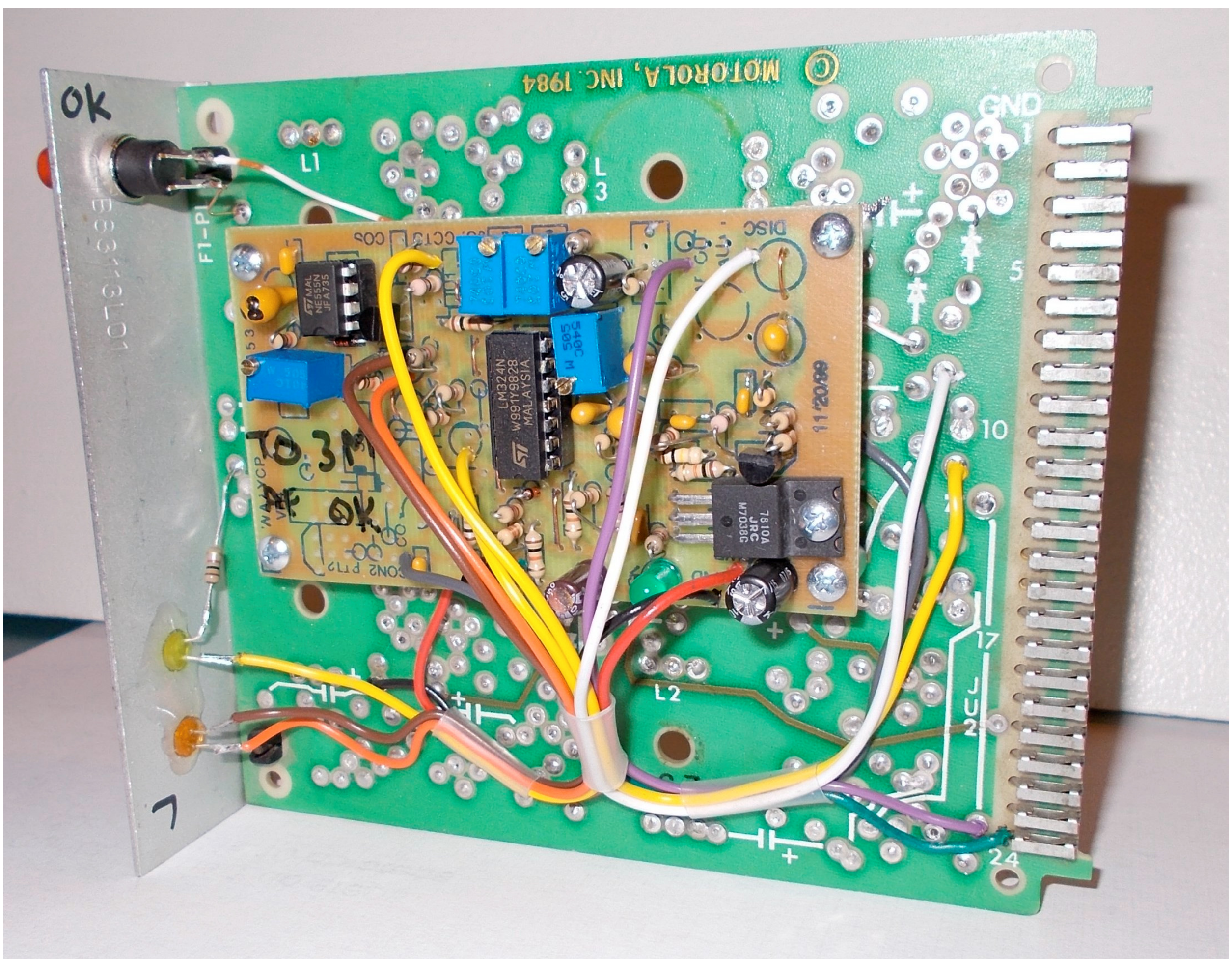
COR / AF board version 5.4 for micor downlink receiver





Here's the finished cor board, installed on the card, aligned, tested and ready for deployment (in service) in the Spectra-tac receiver unit. For maintenance tracking each card is serialized. This particular card needed the monitor button added. Also, the timer and AF settings are indicated to be set.





Parts List: For version 5.3; (part counts are different for version 5.4 and for the 2016 research).

Qyt.	Description	Notes	Part Number	Cost
1	IC, Quad Op Amp, LM-324	U1	511-LM324AN	00.34
1	IC, +8v Regulator, 1.5a 7808	U2	511-L7808CV	00.40
2	IC, Timer, 555	U3,U4	511-NE555N	00.72
5	Transistor, NPN, such as 2N3904	Q1-6	625-2N3904	00.50
2	Resistor, 1 Meg, 1/4w, 5%	One is "R1" value	291-1M	00.07
6	Resistor, 100 K, 1/4w, 5%		291-100K	00.42
2	Resistor, 68K, 1/4w, 5%		291-68K	00.14
1	Resistor, 33K, 1/4w, 5%	(p/o the AND gate)	291-33K	00.07
1	Resistor, 15K, 1/4w, 5%		291-15K	00.07
13	Resistor, 10K, 1/4w, 5%	(one for the AND gate)	291-10K	00.91
1	Resistor, 2.2K, 1/4w, 5%	(one for the AND gate)	291-2.2K	00.07
9	Resistor, 1K, 1/4w, 5%		291-1K	00.63
1	Trim-pot,multi,5 Meg,inline leads	VR 3	Hosfelt #38-184	01.35
2	Trim-pot,multi,2 Meg,inline leads	VR 4,5 (ver 5.3)	594-64W205	04.00
2	Trim-pot,multi,10K,inline leads	VR 1,2	594-64W103	06.00
4	LED's;sub "xx" for:Red:VR,Org:DU,Yel:YY,Grn:MG		592-SLR56xx3	00.72
4	Diode, 1N4148 or 914 type,400piv,150ns		583-FR104	00.40
2	IC socket, 8 DIP, solder tin		571-26404633	00.16
1	IC socket 14 DIP, solder tin		571-26403573	00.08
1	Capacitor, Tantalum, Radial, 100uf/25v		ME	01.75
3	Capacitor, Elect, radial, 100uf/25v		140-XRL25V100	00.21
2	Capacitor, Elect, radial, 10uf/25v		140-XRL25V10	00.10
3	Capacitor, Elect, radial, 1uf/25v		140-XRL25V1.0	00.15
1	Capacitor, Mylar, radial, .0082uf/100v *		140-PF2A822F	00.43
1	Capacitor, Mylar or Disc, 390pf/50v *		140-50P2-391K	00.06
1	Capacitor, Mylar, radial, .22uf	for U2	(vender TBD)	00.18
1	Capacitor, trimmer, 20pf, 225v	455 KHz IF trap	242-3810-23	00.88
2	Choke, radial, 4.7 mh, 9mm dia.	455 KHz IF trap	434-01-472J	02.28
1	Board, COR/Audio, AK2O	FAR Circuits**	ver 5.3	06.00
11	PVC colored wire, "6 long, 22-24 gu.	Various colors		
6	bare wire around 22-24 gu	For board jumpers		
Total Parts Cost (shipping not inc.)		As of March/2000		$29.09

Prices on parts are an estimate. You might find alternative sources such as: Mouser Electronics at (800) 346-6873 or on line at www.mouser.com.

Other Notes: Color of wires: Black, red, white, green, yellow, orange, blue, brown, violet, pink, slate. Allow 2 hours labor for building and 2 more for alignment; common tools and solder equipment.

Unless otherwise specified, DC voltages are positive, in relation to a negative ground system. Resistor values are in ohms, 1/4 w, 10% or better. Chokes in milli-Henries, caps in MicroFarads. IF trap tunable range is 420-800 KHz; use other values of chokes for other IFs i.e. 11.7 MHz, etc.

** The board is designed by Karl Shoemaker, AK2O. To purchase, contact the manufacturer, FAR Circuits at: 18 N. 640 Field Ct.,Dundee,IL,60118, or email at:farcir@ais.net Fred Reimers, KF9GX, is your contact.