COR / audio board

Version 6.5.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. This project is a cor / audio board to interface with the Motorola Mitrek receiver.

Overview and History:

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" 1990 boards are still in service. The more recent versions of 5.x used improved methods of signal processing including higher impedance inputs as not to affect the associated receiver. Other improvement included 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). Version 6.x was developed for the Mitrek, to be mounted inside the radio. Since then the Author further analyzed these features for future SRG projects. Some of these will be covered in this document.

Points of changes for this version:

- More detail is covered concerning DC/cor and tone protection.
- Many past considerations were incorporated in this version.
- This is designed specifically for the Mitrek radio, to be mounted upside down (solder side up) in place of the stock PL deck. However, the board could be use for most any type of FM receiver.
- Optional RFI filtering is used on some of the I/O functions.
- Optional bypass capacitors for filtering DC sensitive areas are used, such as the bias / rail for the audio amps. They can filter out transits that the 100uf caps may miss.
- AGC meter circuitry has been updated.
- This version is engineered differently therefore, the IC pin-out functions are different as well.
- Audio output S/N noise is 49 db, or better.
- This version may require removal of the mobile key/lock system on the radio’s front.

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 originating source audio by 6 db per octave and de-emphasizing (decreasing the level) of the high end of the far end 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.

Repeater stations:

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 single transmitter or a single receiver to obtain quality. The key point is both components of the repeater station have to be the same of one type 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 properly de-emphasized receiver driving a properly 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 performance and not the method to obtain this.

One exception

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).

Deviation limiting or clipping:

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.

Squelch operation:

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.

A twist:

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.

Spokane Repeater Group:

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

The Project:

The board has three inputs. Two of them are for logic signals while the third is for audio. These inputs are processed to interface with a receiver to drive an associated transmitter. The board has about 4 areas of circuitry; logic, control, audio and metering. Each area will be discussed here.

Logic:

The board has inputs for either a cor input, an external signaling decoder indicator (SDI) or both, with any polarity and voltage level change. The schematic drawing includes (optional) pull-up and pull-down resistors you can include, depending on the type of receiver you are working with. The PTT outputs are open collector, normally high resistance, going low on activity. By setting the input buffer jumpers (op-amp/comparator input and bias) this allows just about any arrangement possibility. A tone decoder could be physically on this board if it was small enough, such as the Comm Spec TS32M. Otherwise, such a decoder can be external to the board and receiver.

DC Inputs:

COR from the Mitrek receiver comes from three usable points however, for SRG projects only point "**E**" is used. (For more information on the other points seek documentation on SRG's web site). The cor is a logic "high" during standby and goes low during activity. Therefore, U1 buffer is a non-inverted output on pin 14 by setting JU1 to **A-B** and **C-D.** For other receiver types that have a cor point a logic "low" during standby then goes high (or higher) for activity, U1 buffer is set for set for the opposite (inverting) by setting JU1 to A-D and C-B. For this project the former setting will be discussed. In the event you need a pull-up resistor on the cor input there are pads on the PCB for that option. Same as for the SDI (below).

SDI is external and set up for a logic "low" during standby and goes high during activity. Therefore, U1 buffer is inverted output on pin 1 by setting JU2 to **A-D** and **C-B.** For other types of SDIs that have a standby logic high then goes low (or lower) during activity would have JU2 set for non-inverting with A-B and C-D. For this project the former setting will be discussed.

Repeat audio:

During standby condition U1, pin 14 is a logic high. This causes Q1 to be turned on to squelch any audio into the pre-amp U1's input on pin 6. The audio amp runs on a single-end rail voltage set by the resistive divider voltage on pin 5. This drives the second audio amp, which has an adjustable rail on pin 10. The second amp output on pin 8 drives the System transmitter audio. There are also two stages of top-end audio equalization in between the two amps. More about this is discussed later in this document. JU4 determines the repeat audio being in carrier or "AND" squelch mode; the most common being the latter.

Control buss:

A set of diodes (D1~D8) make up a control "buss" for carrier squelch, tone or both (carrier + tone = AND) for either the audio or PTT paths, or both. This means that all the inputs of this buss have to be a logic low in order for the PTT outputs to be active. This includes the cor, decoder, CON1 or time-out inputs. In a normal standby status only the cor and decoder inputs are a logic high and both normally go to a logic low during activity. Depending on the mode, this can create an "AND" squelch for the PTT outputs only. The repeat audio can be set for carrier squelch or "AND" if desired. For easy reference some of the odd numbered diodes affect carrier, while the even number ones affect the decode configuration.

PTT outputs:

During standby U1, pin 14 puts a logic high on the control buss via D5. Also, U1, pin 1 will put a high on the buss via D6. This prevents PTT output activity (standby status). During <u>both</u> carrier and decoder activity (AND squelch) both will go low, removing the logic highs on the control buss. If the other inputs are also low (CON-1 and time-out) the entire buss will be a logic low causing U3, pin 1 to go high and turn on Q3 for the PTT 1. It also turns on Q4 to start the tail timer on pin 7 output going high to turn on Q5 for the PTT 2 to key the System transmitter. It will stay this way as long as there is activity. When the activity ceases long enough for pin 6 to rise above pin 5 via the RC constant the PTT 2's Q5 will turn off. With a 10uf cap across pin 6 the tail length is adjustable from 1/4 ~ 45 seconds via VR4. For longer tails change the cap to around a 22 or 33 uf value. If you set the timer for a 15-sec tail, using the 10uf cap, the actual resistance will be 791 K ohms. Also, when activity stops PTT 1 immediately turns off. PTT 1 can be used for a link signal or auto-patch audio switching.

Both PTT 1 and PTT 2 are open collectors, turned off (relaxed). During activity they turn on, going to low resistance (about 60 Ω to ground). However, when installed in the Mitrek package, the PTT 1 is pulled high during standby because of the LED in series with it. That's to indicate on the front panel of activity. Note: for "AND" squelch only opening the local squelch (no signaling) will not light that LED. It will light for both carrier and signaling decode.

<u>Time-out:</u>

In this case during standby Q2 is turned on via either D3, D4 or both (for "AND" squelch). This keeps pin 12 of U3 pulled to a logic low. During activity Q2 relaxes, causing the voltage on pin 12 to rise. If activity stops before "time-out" this will reset the timer by pulling pin 12 back to a low and discharging the cap thus, keeping the PTT outputs enabled. This is a normal System condition. However, if the activity continues more than the allowed time (normally 3-minutes) will allow pin 12 to rise higher than the resistive divider voltage on pin 13 to cause pin 14 to go to a logic high. Via D7 this will disable both PTT outputs. This satisfies the FCC part 97 requirement for repeater using "automatic control".

During a System time-out would stay this way until the input activity stopped; either from the carrier (cor), signaling (SDI) or both stops (depending on the mode). When it stops pin 12 of U3 would be pulled back to a low via Q2 being turned back on and cause pin 14 to return to a logic low thus, enabling the PTT outputs to key the System transmitter and indicate (to the users) the timer was reset. The time is adjustable via the RC constant of VR3 and the value for the capacitor across pin 12. This cap value can be selected for your system needs. For example, a 68uf cap provides a 2 ~ 300 second (5 minutes) limit, while a 47uf cap provides a range of 1 ~ 210 seconds (3 ½ minutes) with VR5 set to maximum for the longest time figure. If you set the timer for 3-min, using the 47uf cap, the actual resistance will be 1.664 Meg. ohms. If time-out is done elsewhere you can disable this timer by setting JU3 to **A-B**.

To recap, the two timers on the board are "time-out" (limit of a transmission) timer and "tail" timer (after activity stops). They are adjustable from about 1/2 to 300 and 1/2 to 45 seconds, respectively. The first drives (controls) the second (tail) timer. Take this into consideration when setting up your repeater by adding both times. For example, if you want a 15-second tail with a 3-minute time-out, set the time-out for 2 3/4 minutes and the tail for 15 seconds (165 + 15 = 180 seconds = 3 minutes).

<u>**Three squelch modes** (tone or carrier):</u> (general information)

- Both audio and PTT tone squelch

Connect the receiver's cor point to the input on the board. Connect a CTCSS tone decoder output to the SDI. A good choice is the Comm. Spec's TS-32. It has a choice of active high or low output. Follow the instructions that comes with the decoder. That will affect JU2s setting. This makes both the repeat audio and PTT outputs to an "AND" squelch. Be aware this arrangement will increase delay for an audio signal to get through the repeater. Quick user's first word may be cut off. However, this arrangement does have the most "protection" against RFI sources. Most receiver's cor point will have its own pull-up or down resistor. JU4 is set to **A-C**, **A-B**.

- Audio on CS and PTT tone squelch:

Connect the receiver's cor point to the input on the board. Connect a CTCSS tone decoder output to the SDI as mentioned above. The audio path won't be delayed by tone decode in this arrangement. For the Micor receiver (only) the pull-up resistor would still apply. JU4 is set to **A-E**, **A-B**.

- Both audio and PTT on CS:

Connect the receiver's cor point to the input on the board. Leave the SDI alone; no decoder is needed. The audio path won't be delayed by tone decode in this arrangement, Nether will be the PTT.
This arrangement has no protection against RFI. For the Micor receiver (only) the pull-up resistor would still apply. JU4 is set to **A-E** and **A-D**.

Versions:

6.5 uses CON1 and the time-out is enabled. 6.6 uses CORI and the time-out is disabled. You can also have a combination of either version with these two features. Reference to the diagrams to show this.

For any of the three modes it's good to mention that earlier design called for the cor voltage derived from a pull-up (or down) resistor on the receiver leaving no conditioning on the cor board. The downside is during test and measurement if the cor line was to be removed (for testing only) the cor input buffer may activate on its own due to the high sensitivity of this amp. Therefore, current design calls for any pull-up (or down) to be on the cor board instead. There are spare PCB pads on the board for this purpose.

This chart may help you understand the various modes and arraignments you can have.

Mode Matrix COR Board version 6.5

Arrangements:

AUDIO	PTT	D1	JU4	D3	D4	D5	D6	JU3	Q1
Carrier	Carrier	IN *	A-E,A-D	IN *	DC	IN	OUT	B-C	IN
Tone	Carrier	OUT	A-C,A-D	IN *	DC	IN	OUT	B-C	IN
Carrier	Tone	IN *	A-E,A-B	OUT	IN	OUT	IN	B-C	IN
Tone	Tone	OUT	A-C,A-B	OUT	IN	OUT	IN	B-C	IN
Carrier	AND	IN *	A-E,A-B	IN	IN	IN	IN	B-C	IN
Tone	AND	OUT	A-C,A-B	IN	IN	IN	IN	B-C	IN
AND	Carrier	IN	A-C,A-D	IN	DC	IN	OUT	B-C	IN
AND	Tone	IN	A-C,A-B	OUT	IN	OUT	IN	B-C	IN
AND	AND	IN	A-C,A-B	IN	IN	IN	IN	B-C	IN
Timer not used		N / A	N / A	N / A	N / A	N / A	N / A	A-B	IN
Unsquelched audio		DC	DC						OUT

Jumpers

COR		JU1	
Active	Low	A-B	C-D
Active	High	A-D	C-B

PTT transistors:			
Active	" ON "	" Low "	PTT 1
Active	" ON "	" Low "	PTT 2

SDI		JU2	
Active	Low	A-B	C-D
Active	High	A-D	C-B

Audio	PTT	JU4	
Tone	Tone	A-C	A-B
Carrier	Tone	A-E	A-B
Carrier	Carrier	A-E	A-D

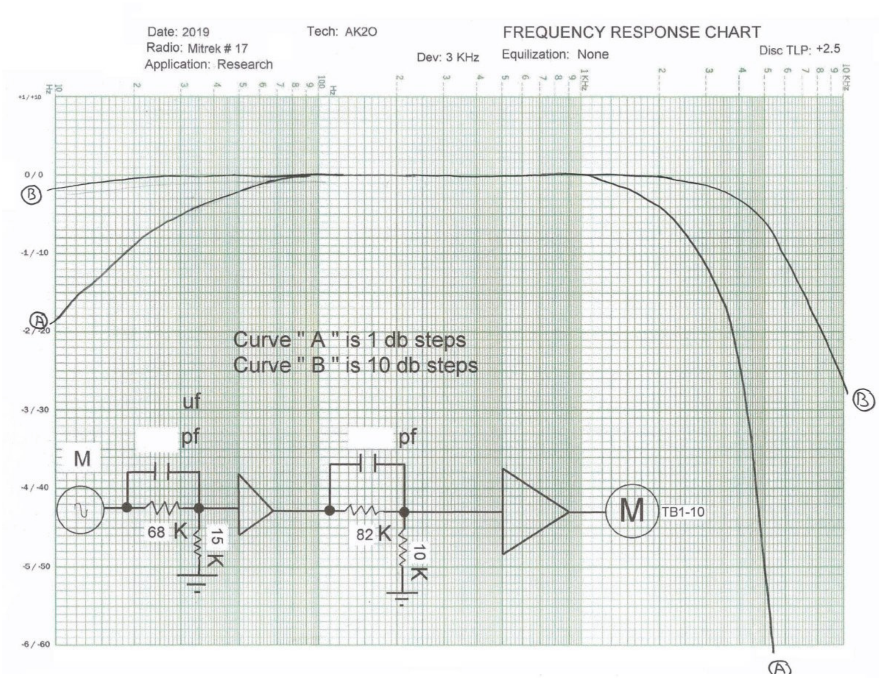
Time-out	JU3
Used	B-C
Disabled	A-B

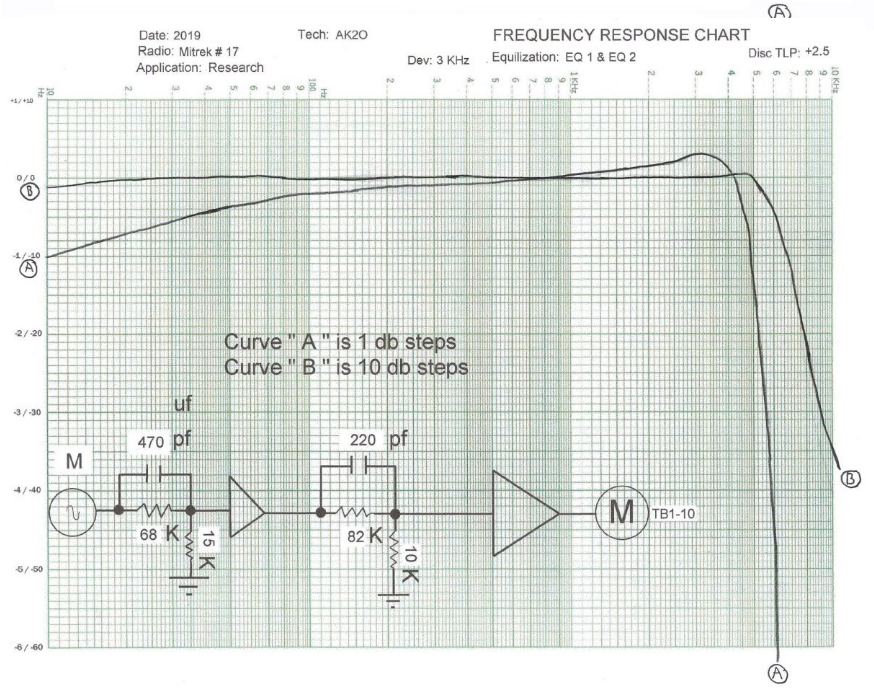
NOTES:
* Can be either "IN" or just jumped with a wire;   "DC" = don't care

Flat Audio:

Most conventional, commercial systems have high-end roll off for audio frequency response. If you want your amateur system to sound really good (flat) you can extend this. 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 top plot is only the receiver's response with no equalization while the bottom one is with 2 stages of equalization. This brings the top-end at 5 KHz around 1 db down as show in the bottom plot. As not to load / affect the circuit these plots were measured at the board's output (Mitrek TB1-10 (AF Out).

The board has two stages of equalization with amplification to bring the level back up to a usable level. The first stage will flatten out the upper end, say above 2 KHz, and the second stage, for above 4 KHz. Typical values are in the first stage are: the series cap is 470 pf, the series resistor is 68K and the shunt resistor is 15K. For the second stage, the series cap is 220 pf, the series resistor is 82K and the shunt resistor is 10K. For your application you may have to deviate from these values and re-plot several times.





Most receiver's discriminators contain DC. A blocking cap in the AF input addresses this.

For correct Mitrek discriminator AF TLP it needs to be verified for proper alignment (per the OEM manual). Afterwards, it should be +2.5 dbm. If so, the typical (bridged) TLPs are shown on the schematic drawing.

If the U2 is a 7810, the typical operational voltages are shown in table 3 on page 13. Some 324 ICs produce lower "and" gate voltages and still work fine, since it's the relative difference between the two input levels that determine the logic output.

Optional filter:

The Mitrek radio is a single-conversion with an IF of 10.7 feeding the detector. However, some older receivers, such as the Motrac and Midland use a second IF of 455 KHz. Because of this lower IF, it's possible some residue IF can appear at the discriminator. If this is the case the optional L-C filter can be used. There are extra pads between the two amplifiers for this purpose. Note: This could have an effect on the response plot therefore, if using it, install it before equalizing the two stages.

Note: for the Mitrek receiver R1 is 820K (per table 1).

Low end:

If you are using the (optional) IF filter there may be a low end drop-off. Sweeping the board only (no receiver) the frequency response is good from 50Hz ~ 5 KHz, however, it was found at 40Hz it's down by .1 db, at 30Hz down by .2 db, at 20Hz down by .4 db and 10Hz down by 1.6 db. If you need the low end more flat change the coupling output cap on U1, pin 7 to a 100 uf. However, if you are using this for high speed packet leave it at 10 uf (as not to disrupt the "eye" pattern).

Setup (in general):

Load the board with all the components needed, depending on your application. You will need to connect at least ground, power, cor input, AF input and output 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 (as earlier discussed). For a cor point that is **high** (or higher) on standby, then goes **low** (or lower), jumper **A-B** and **C-D**. For a cor opposite of that, jumper A-D and C-B (inverted). Note: This is repeated in this paragraph to assist a possible impatient builder.

For alignment only set the board in carrier squelch. That's JU4 to A-E and A-D. This will enable the yellow LED for alignment in the next paragraph. Later, you can change it to whatever mode you wish.

Power up the receiver and board. The green power led should be lit. Remember that it takes about five 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 squelch is active. Use the yellow LED to watch the transition. There is no hysteresis on the input, so you should give yourself a little "margin" with this trigger point, for component/aging variances. For the Mitrek set it (U1, pin 13) for 2 volts (note that earlier specification was 0.70 volt).

Then, inject a clean 1 KHz tone and turn up VR6 to just at clipping point observed on the output with an oscilloscope. Tune the bias/rail at pin 10 with VR5 for best even top and bottom clip on the output. Re-adjust VR6 as needed to fine tune the VR5 adjustment. VR5 will be a one-time alignment when done. To note: pin 5 doesn't need adjustment because it should be running well below the clipping point.

If you need higher output there are some options. Most any 78xx series will work, since the audio op-amps use a single end supply with voltage dividers for the "+" reference. You can operate them at higher voltages. For example, with a 7812 as U2 which will drive U1 output around +12 dbm (bridged). Just watch out for the maximum operating limits of the op amp. Also, higher regulator values reduce ripple protection and makes U2 work harder. The Author settled on the 7810 (+10v) for the best performance for most applications with a maximum output of +10.7. The schematic reflects this design.

IDC:

When properly set, you have the option of using it as a nice "IDC" (deviation limit) because it's linear up to that point, then just flat tops with further increase input. Most conventional IDC circuits use diodes, back to back, which start causing distortion before the actual clipping point. By installing an optional pot in the transmitter with the wiper going to the FM, the IDC mode would work for most any transmitter. Some transmitters already have an IDC pot on its channel element as in the case of the Mitrek radio.

Set the required deviation limit with this external pot, presumably, for 5 KHz. Remember, for this mode U1 will be "working" much harder however, is rated for this type of duty cycle. Note that SRG only uses the IDC at one point of the System (or subsystem in the case of the 20-Wenatchee station). For this project the far end MCP has such a circuit. It's not very relevant however, justifies clarification for the big picture. Note: if you are measuring the receiver's discriminator point use a high impedance probe.

For TLP establishment the input level was 0dbm (@ 1 KHz frequency). Table 2 lists the stage gains and losses. For the Mitrek radio the input TLP will be +2.5 dbm. The schematic drawing reflects this TLP.

Table 1

Input TLP	**R1** value	Remarks
+5	470 K	Or a lower value
0	820 K *	"
-5	1.0 Meg.	"
-10	2.1 Meg.	"
-15	5.6 Meg.	"
-20	9.3 Meg.	"

Table 2 - - **T.L.P.**s (bridged-for R&D & not for Mitrek levels)

Point of Measurement	Level	Remarks (U1 being a 7810)	Noise floor
AF Input	0		-55
EQ-1 output	-20.2	Without EQ cap	
Squelch switch	-34.2	Q1 collector	-66
U1, pin 6 input	-38.2		
U1, pin 7 output	-3.0	*With R1 value of 1M for Mitrek	-47
EQ-2 output	-26.8	Without EQ cap	-63
U1, pin 9 input	-40.5		
U1, pin 8 output	+11.0	VR3 at maximum	-51
U1, pin 8 output	0	Normal, operating TLP	-59

Table 3 - - **Typical + DC voltage chart:**

Condition	**U1; pin 13**	**U1; pin 12**	**U1; pin 14**	**PTT 1 and 2**
Standby	2.00	3.34	8.92	Standby (off)
Carrier active	2.00	0.295	0.26	Active low
Decoder activity only	2.00	N / A	8.92	Off or on
Both carrier and decoder active	2.00	3.34	0.26	Active low
“CON-1 pulled low	2.00	N / A	N / A	Off after tail is done
“CON-2 pulled low	2.00	N / A	N / A	Off
Unit timed-out	2.00			Active low
Standby w/o 100K shunt	Radio’s “E”	3.34	N / A	N / A
Standby w/ 100K shunt	Radio’s “E”	3.17	N / A	N / A
Carrier active (either way)	Radio’s “E”	0.195	N / A	N / A

Note: A minor point that voltage measurements are in reference to the radio’s (black) ground jack. Static resistance (radio off) between that and the cor board’s ground run is very low however, when the radio is on, measuring voltages there will be about 1 mv difference between those two points. Note: table 3 is based on an “AND” squelch mode.

Audio setup and some more Theory:

The receiver's audio output you are using should be -20 dbm or higher. Using table 1 you can match the receiver to the board's TLP for audio input. For most receivers R1 can be 1 Meg ohm. For other TLPs change R1 value per the chart. Being a guideline, these are maximum TLP's; you can run lower levels and/or different R1 values, if desired. Just keep pin 7 output below about +9 dbm (TLP) to prevent clipping. (clips at +10 if U2 is a 7810) You can deviate from the chart with proper tests; as the chart for flat audio section shows for the Mitrek radio's TLP. You can also run TLP's lower than the recommended level of -20, with limited results. Also, if you don't need a squelch you can leave out Q1 or D1 & D2.

This is an optional secondary AF input labeled “Aux in”. It could be used for a local IDer or other tone indication for the “remote” site’s equipment. If its not used the 1M resistor can be left out.

A word about VR6:

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. The 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 the in and outputs 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 feedback and input resistors, then make the logarithmic conversion for a more realistic approach on levels. With R1 value and VR6 you can control the two stage gains.

AGC Theory:

Many commercial receivers either don't have an AGC meter or their "M1" output will not properly drive a meter with a meaningful scale. Most Amateur receivers do have an AGC meter, however, typically are factory set to give an "early" (generous) reading with weak signals, which is a waste of indication. FM receivers quite with signals therefore, you can listen for these changes with the local speaker when checking performance. When the signal gets almost full quieting is when you need a visual (meter) indication to observe signal strength changes. This circuit will favor the latter condition. U3 amplifies the receiver's AGC voltage, then with a strong signal will level out on the meter's full scale. This makes a handy logarithmic chart that can be plotted per RF input changes. U3, pin 10 inputs a fairly wide range of receiver detected/IF amplifier DC meter function. As of 2022 the reference (bias) is fixed. Pin 8 output drives most meters with the VR8 adjustment.

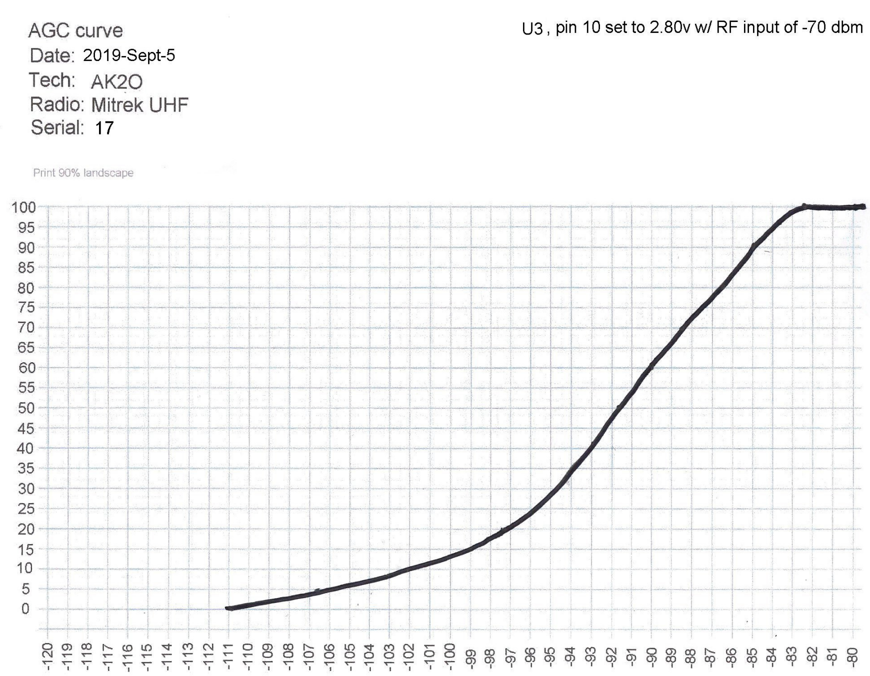
AGC Setup:

This version is specifically for the Mitrek receiver. However, it can be made to work with most any receiver type. As previously mentioned, any of the three amplifier points can be adjusted by changing the resistance values. For the Mitrek receiver standard values produced a fairly good curve:

- The bias, from A+, has a series 10K, with a 4.7 K shunt to ground. The junction has + 3.25 volts. This produces + 2.424 volts at U3, pin 9. This bias (rail-reference, if you will) does not need to be adjusted, therefore, a fixed value is made. (previous versions had a pot.)

- The input appears on the top of VR7, a 2 Meg pot. Adjustment is made with an RF level input. (Some previous versions had fixed resistors in series with a possible shunt during no RF.) * * *

- The output is controlled with VR8. With a strong signal the load (pot + meter) draws about 270 uA. Some meters require a 1K shunt. Refer to the Mitrek conversion for more details on the meter.



The RF filter .01 uf caps on the input and output may not be needed therefore, is optional; there are extra PCB pads for this purpose.

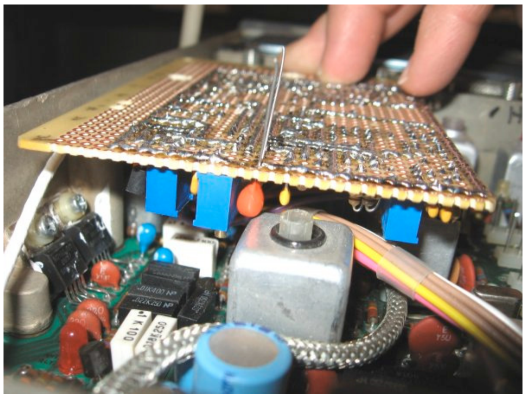
Mitrek serial 17 was used for this plot and is a typical curve for this receiver. In some cases the curve won't be smooth, due to equipment accuracy. The sensitivity for 20 db/q on this one was – 113 at the low end of the band.

For other receivers start by measuring the "M1" or AGC point of the receiver you are setting up. The AGC circuit prefers to "see" DC around 1-2 volts with no RF signal into the receiver.

For the Mitrek is pretty straightforward. Using a RF signal generator input the receiver with an on-frequency CW at –70 dbm level and adjust R7 to produce 2.80v on U3, pin 10. There is a measuring test point on the PCB board for this purpose. Next, adjust VR8 for a full-scale deflection on the panel meter.

Note: for the best curve the meter (load) movement you plan to use should draw between 200 and 500 uA for a full scale indication. The final alignment should produce at least a 25db of RF range.

NOTES: During the R and D, some attention was necessary for clearance around L208 and a few other areas of the receiver section touching the board's VR 8. Therefore, a low profile single-turn trim pot was substituted. Later, the VR 7 and VR 8 were moved on the board to solve this issue. This prototype (version 6.2) proved to be satisfactory. The image doesn't do justification since the board was not held straight for the picture. The final product provided plenty of clearance, including use of the signaling decoder.

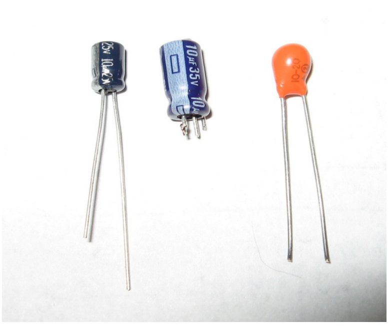


The .1 uf cap across U1, pin 10 appeared to cause some negative feedback to the output on pin 8. If you leave it in the reduced maximum levels are +9.0 at 11.40v +9.2 at 11.74v, +10.5 at 13.45v, +11.6 at 14.71v supply.

The original idea was to filter any ripples. After some research this is not an issue therefore, you can leave the cap out. You will need the higher output for some link units, such as the MX-350 transmitter which needs a +9 for TLP, using it's "DPL" input. S/N is 59 db when running the output TLP of 0 dbm. It's less when running higher TLPs.

Several caps of the same value were considered for their diameter and height for some tight areas. The two electric types are from different vintage, the smaller being more current and advanced, while the third being a tantalum is always a good alternative, plus, they last longer than the former types.

After some engineering the MLCC type was used where the 1uf electrolytic AF coupling caps were used earlier.



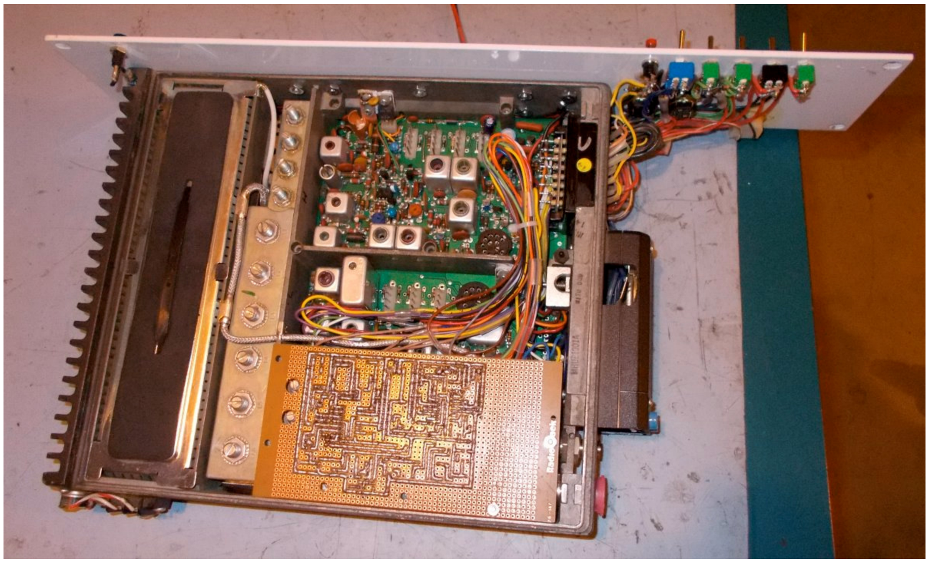
Allow 2 hours labor for building and 1 more for alignment; common tools and solder equipment.

Note: AC (audio levels) measurements are made with a HP 3555 and 400C meters (log scale) in bridged (high impedance) mode. DC measurements are made with a Fluke DVM model "77", including the (alternate) way of checking an AC (audio) level (RMS to log conversion) in one case.

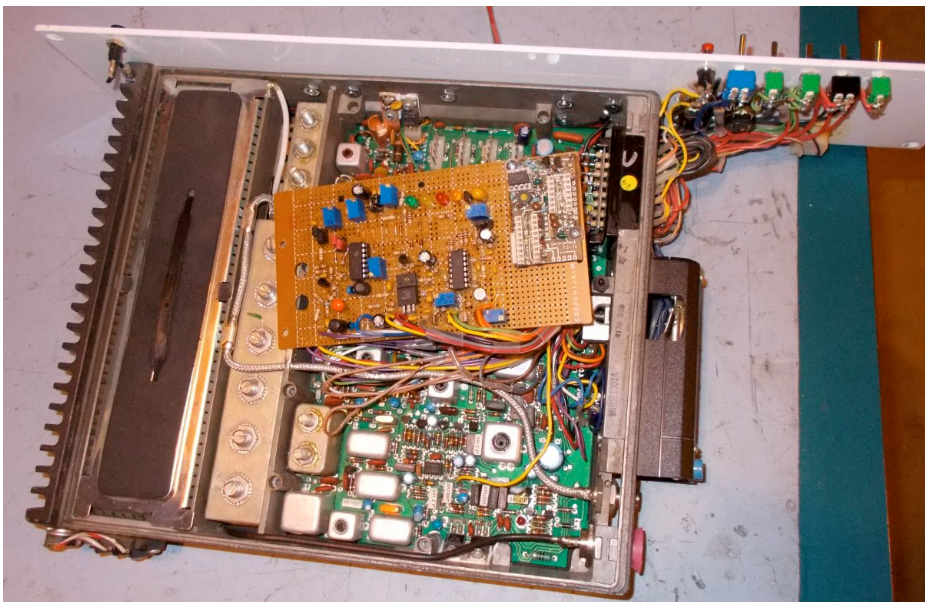
CON1 notes: An application for the CON1 would be for control (disable) of the PTT1 output for a link or auto-patch signal. Otherwise, if you don't need it that line can be used for a CORI, which is a carrier squelch indicator that bypasses any "AND" squelch setting. This is handy to watch on the radio's front panel for the radio's squelch opening from RFI or other sources. This option was mentioned on page 10; version 6.6 of the board.

If you need both you could use an external controller for the CON1. Also, if you are running a single frequency (with JU611 in) those F1 and F2 can be used for this or another feature as well. As a reminder, there is complete documentation on the Mitrek conversion on the SRG web site.

On pages 8 and 9 discussed timer lengths and resistances. This refers to the adjustment pots (VR3 & 4) and the series 10K resistor in series of each one. The resistor is to protect the pots from excessive current (through the transistor) if either one is turned down to minimum.



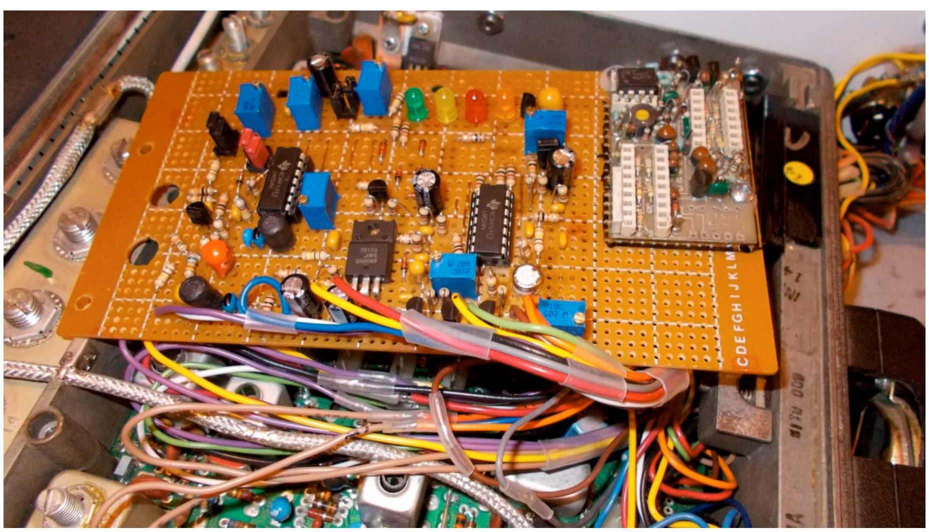
This version of 6.5 prototype was completed in February of 2019 as shown here. It fits nicely in place where the OEM PL deck would be. The mounting screws are not shown in this image. There is room to mount a TS-64 tone decoder on the component side in the lower portion of the board.



The board flips over for easy adjustments to be made.

Several hours were used for board test and measurement.

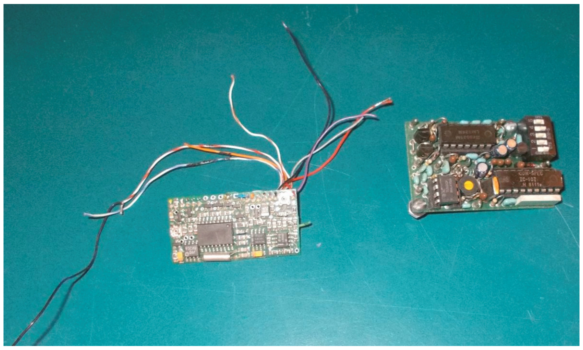
Later, will be the PCB to be manufactured from a vendor.

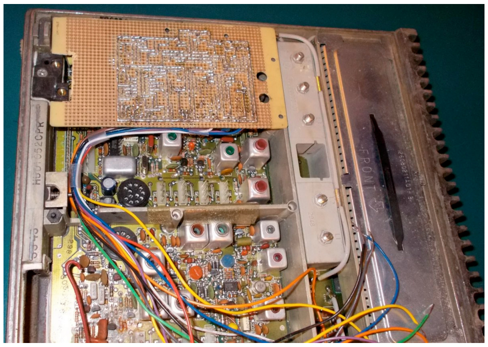


As time permits, the new board will be displayed on this document.

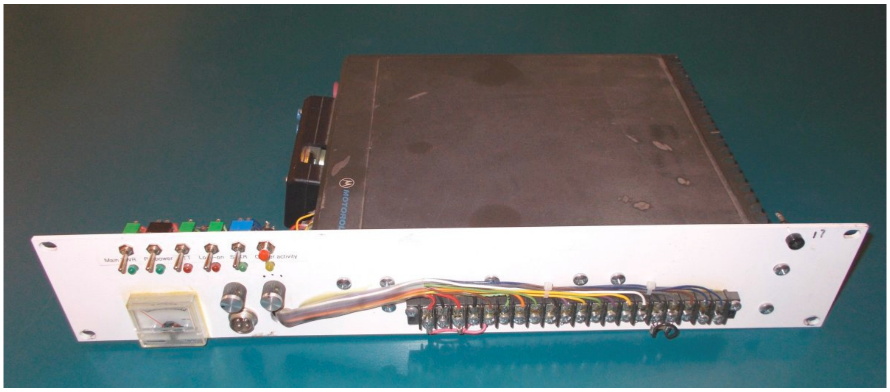
The first page of this document mentioned that the key/lock needs to be removed. As shown here (white arrow) the internal top part interferes with the board's mounting. SRG specs for the Mitrek radio, to be side mounted on a 19" #2 panel therefore, the top cover lock is not used and removed.

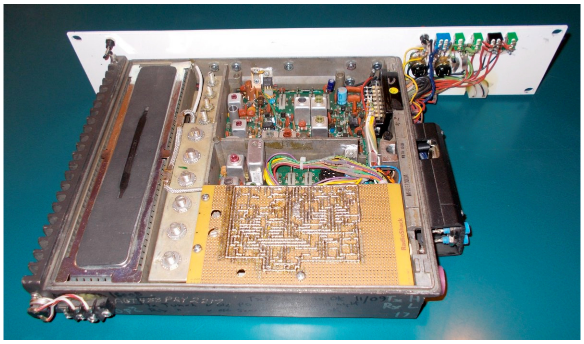
In the event you wish to keep the lock and have tone protection there are some alternatives. For example, you could notch the cor board at the bottom. However, this may prevent the "normal" sized TS-32 board from being mounted although, the vender may have a miniaturized version of this decoder, such as the TS-64WDS. Plus, there may be other brands of decoders that are real small as shown in the left image. If the notch disables the mounting holes you would have to use another method to secure the tone board.

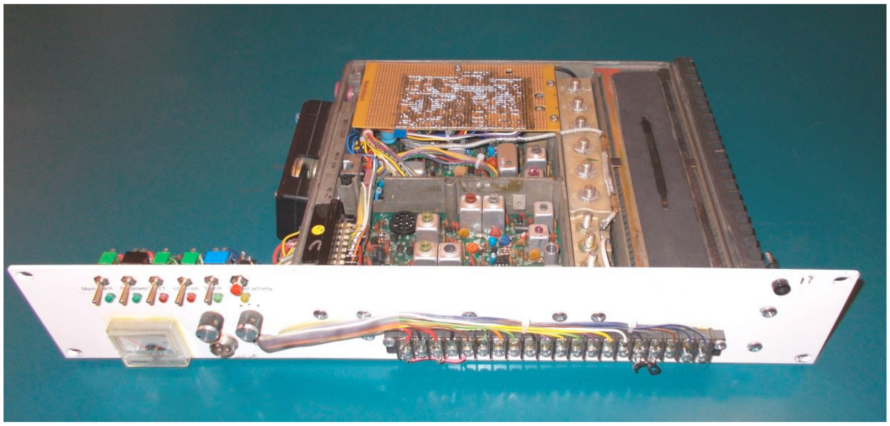


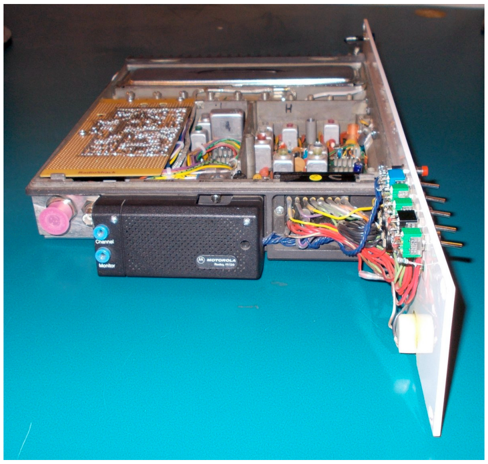


Shown below is the complete radio package. This is self-contained. The local speaker (with test jacks) is a nice touch. The bottom one shows the maintenance cover removed

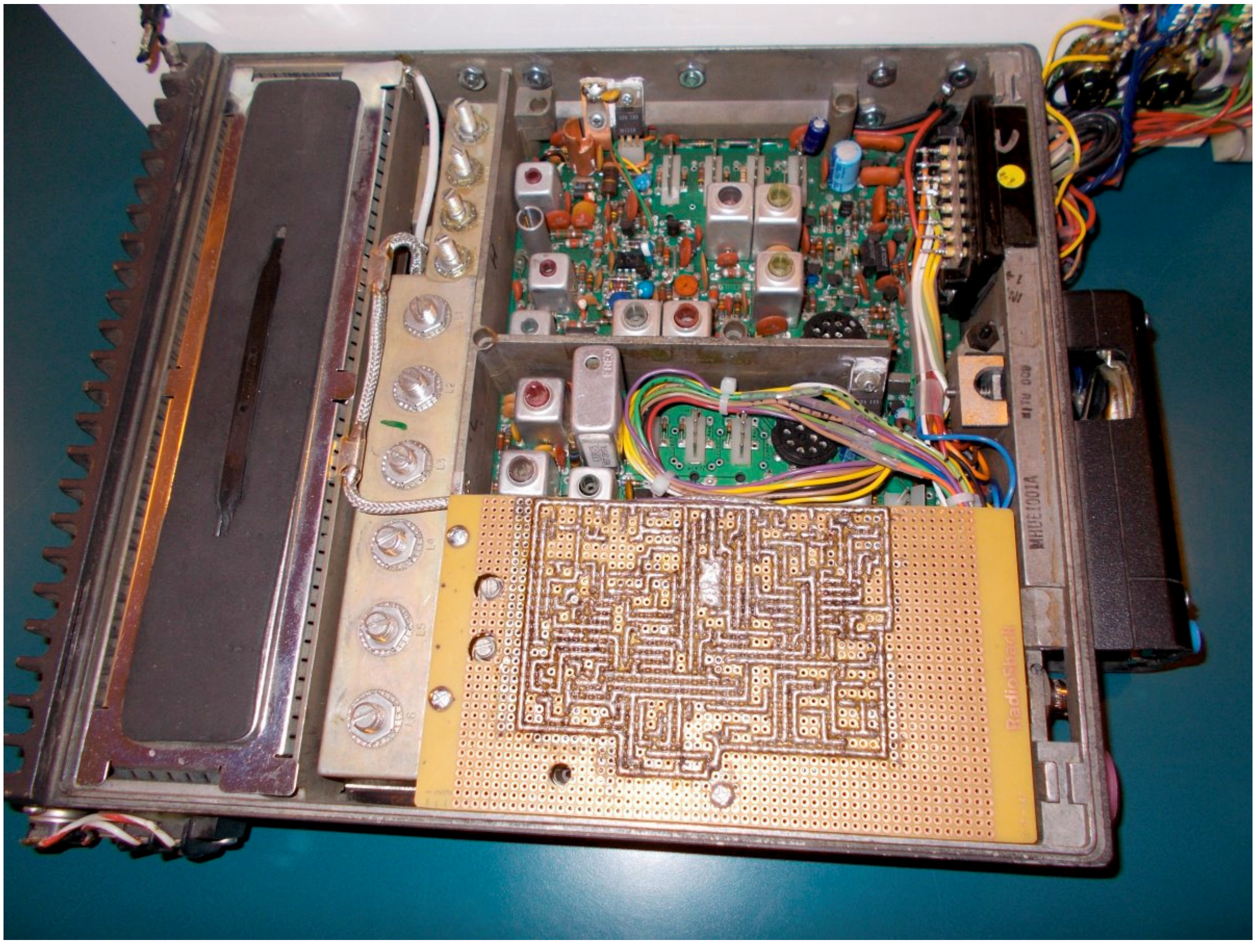


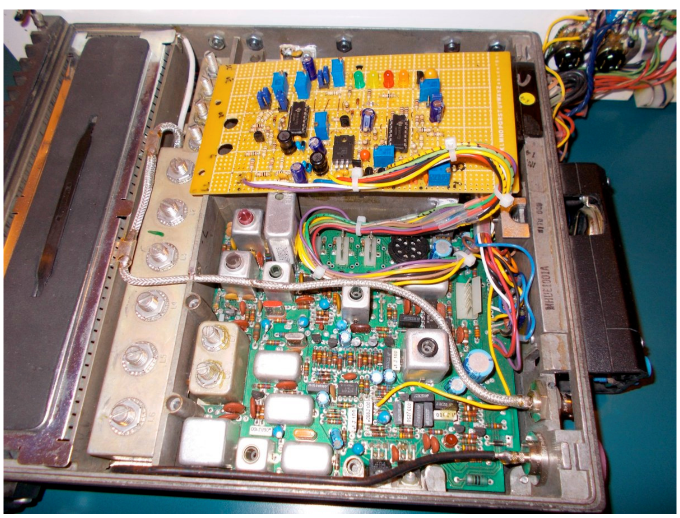


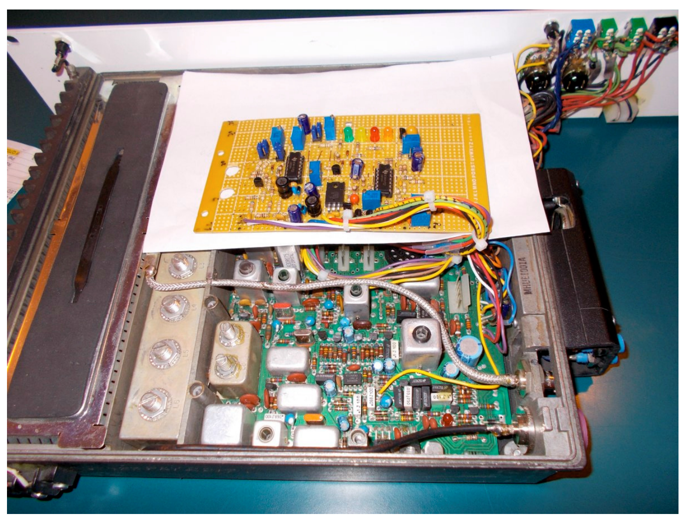




A view of the complete package. The transmitter section is the upper area, the receiver in the middle with the cor board on the bottom of this image.



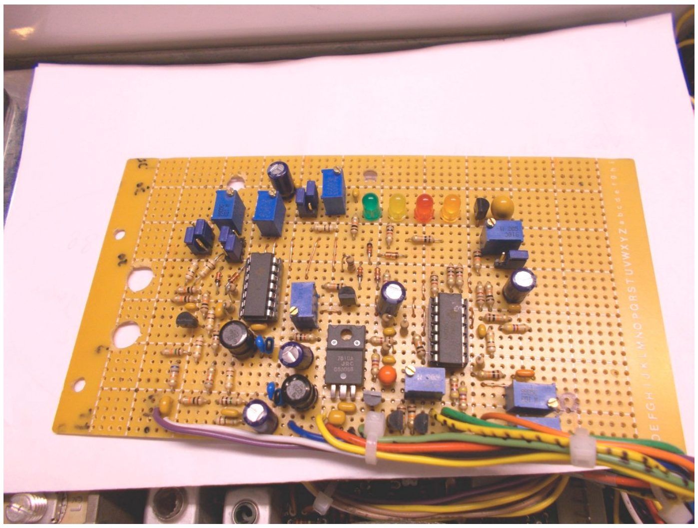


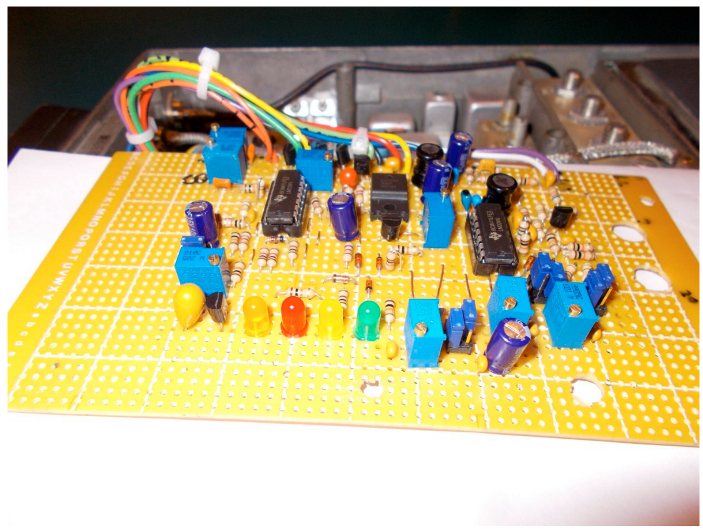


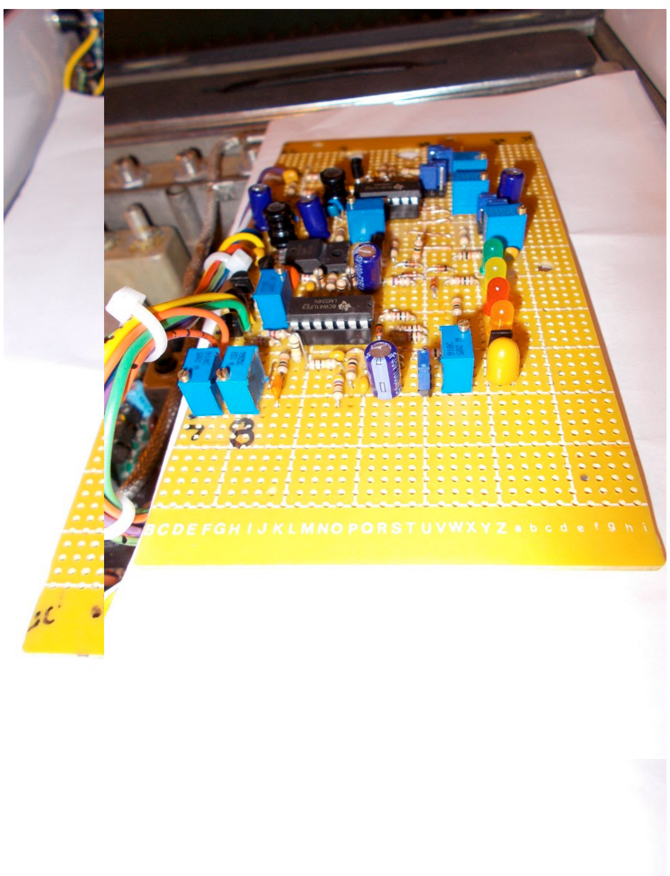
AGC note: Higher versions of this board use a different aligment; Input –70 and adjust VR8 for a meter full scale, then –108 and adjust VR7 for a meter indication of “ 10 “. This produces an even wider range on the curve (plot).

For adjustments remove the three mounting screws that hold the cor board. When the board is powered up you need to insulate its runs with something like a piece of paper as shown in the images here.

Here’s more closer views of the prototype.







Parts List: (may not be complete – still under R&D) Most part numbers are Mouser Electronics source.

Qyt.	Description	Notes	Part Number	Cost
2	IC, Quad Op Amp, LM-324	U1,U3	511-LM324AN	00.34
1	IC, +10v Regulator, 1.5a 7810	U2	511-L7810CV	00.40
5	Transistor, NPN, such as 2N3904	Q1-5	JE:38359	00.50
1	Transistor, PNP, such as 2N3906	Q6	JE:178618	00.50
1	Resistor, 820K 1/4w, 5%	R1	291-1M	00.07
8	Resistor, 100 K, 1/4w, 5%		291-100K	00.42
1	Resistor, 68K, 1/4w, 5%	For EQ-1	291-68K	00.14
1	Resistor, 82K, 1/4w, 5%	For EQ-2	291-82K	00.14
2	Resistor, 1.8K, 1/4w, 5%	For timers	291-1.8K	00.07
1	Resistor, 2.2K, 1/4w, 5%	For Q6	291-2.2K	00.07
3	Resistor, 4.7K, 1/4w, 5%		291-4.7K	00.07
1	Resistor, 15K, 1/4w, 5% *	For EQ-1	291-15K	00.07
12	Resistor, 10K, 1/4w, 5% *	One for EQ-2	291-10K	00.91
8	Resistor, 1K, 1/4w, 5%		291-1K	00.63
1	Trim-pot,multi,5 Meg,inline leads	VR 3	ME 652-3299W-1-505LF	03.39
3	Trim-pot,multi,2 Meg,inline leads	VR 5,6,7	ME 652-3299W-1-205LF	03.39
4	Trim-pot,multi,10K,inline leads	VR 1,2,4,8	ME 652-3299W-1-103LF	03.39
5	LEDs; T 1 ¾, Red, Yellow, Green, Orange and Blue; one each		JE:	
8	Diode, 1N4148 or 914 type,400piv,150ns		583-FR104	00.40
2	IC socket 14 DIP, solder tin		571-26403573	00.08
1	Capacitor, Tantalum, Radial, 47uf/15v	For time-out timer	80-T350K476K020AT	04.16
4	Capacitor, Elect, radial, 100uf/25v	General filtering	140-XRL25V100	00.21
1	Capacitor, Tantalum, radial, 10uf/25v	For tail timer	80-C324C106K3R-TR	01.05
4	Capacitor, Ceramic, radial,1uf/25v	For audio coupling	140-XRL25V1.0	00.15
1	Capacitor, ceramic, radial, 220 *	For EQ2	80-C317C221J1G	
1	Capacitor, ceramic, radio, 470pf *	For EQ1	C317C471J1G5TA	
1	Capacitor, Mylar, radial, .22uf	for U2 filter	DK 399-4289-ND	00.18
2	Capacitor, ceramic, radial, .01uf		ME	
7	Capacitor, ceramic, radial, .1 uf		ME	
1	Board, COR/Audio, K7SRG	FAR Circuits**	ver 6.5	15.00
16	2" bare wire around 22-24 gu	For board jumpers		
3	Header, 2 x 3, .100"	For JU1,2,4	JE: 115035,	
1	Header, 1 x 3, .100"	For JU3	JE: 109576	
7	Shorting bars; .100" for above	(Jameco E)	JE: 19141	

ME is Mouser Electronics; phone (800) 346.6873. JE Jameco Electronics, (800) 831.4242; DK is Digi-Key Electronics (800) 344.4539; AE is All Electronics (888) 826.5432.

Notes: Unless otherwise specified, DC voltages are positive, in relation to a negative ground system. Resistor values are in ohms 1/4 w, 5% or better. Capacitors in MicroFarads, chokes in milli-Henries

* Used for custom selection for each receiver for the equalization (EQ-1 and EQ-2).
The board has five holes; three 31/2mm for the screw mounts and two 8mm for the L7-8 tuning.
Color of wires: Black, brown, red, orange, yellow, green, blue, violet slate and white in AWG 22 stranded.

** The board is designed by Karl Shoemaker, AK2O. As of March 2023 the Author has not found a PCB vender.

*** For Mitrek radio SN 16 has a 100uA AGC meter therefore, required a 1K resistor shunt across the meter using version 6.1 board. (2022)