# COR / audio board

Version 5.3.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. The "PTT 2" is with timer control for normal repeater operation. If you need positive going PTT outputs move Q2 and/or Q4 around so the emitter is the output. There are extra pads on the printed circuit board for this application. (JU3) 'CON 1 or 2' are still available for control in this case.

Transmitter disable controls are active low inputs. You have some choices. "CON 1" simulates a system time out, causing the tail to finish out its time, then drops the transmitter. This affects both PTT 1 and 2. "CON 2" gives immediate transmitter drop out, but does not affect PTT 1, unless the timeout timer expires (if used). Or you can use both "CONs" for control, depending on your needs.

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 are 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. The versions are how the board is used (application) leaving some parts out and adding some parts and jumpers for some of these versions.  Versions run from (this one) 5.3 to 5.7.

The purpose of this project is to promote good communications audio, starting with repeater/systems. Better practices are used for all SRG projects. Some of these will be covered in this document.

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 20$^{th}$ century voice finally was realized. In the 21$^{st}$ 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.

<u>Push To Talk:</u>

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.

**<u>PM/FM:</u>** (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:

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 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 db if you need. With Rx 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.

The audio input has a 455 KHz tunable LC trap. If the receiver has some IF leakage, null it with the trimmer cap while monitoring the output with an oscilloscope. For other IF's, such as 11.2 MHz you can try different LC values and sweep it to find the resonance point. The Micor normally does not need the trap therefore, you can leave these parts out and bypass with a jumper. The input TLP should be -20 dbm or higher. For lower inputs change Rx value, from the 1 Meg. to around 4.7 Meg. 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 is for the FCC requirement of automatic repeater control. This section of U1 is controlled by voltage dividers, for three possible conditions; standby, active and timeout. 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 tail and transmitter key up. If the signal stops, the tail finishes out its time, then drops the transmitter. 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 the tail finishes out its time and drops the transmitter and stays dropped as long as the signal is active. When it stops both timers reset, which finishes out with a tail for the transmitter. This way the others on the system know when the time out has cleared. Since the timeout controls the tail timer, add both timers when figuring the system transmitter time out. Time out range is 0-3:52 with a 100uf cap and tail is 0-30 seconds with a 10uf cap. Parts list shows a 150uf for longer time-out. You can change values to suit your project.

Setup

Load the board with all the components needed, depending on your application. (Choices for different applications are near the end of this document.) 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**. For a CORs opposite of that, jumper A-B and C-D. Power up the receiver and board. The green power led should be lit. Remember that it takes about 10 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. Give yourself a little margin on this trigger point for stability. 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. When properly set, you have the option (discussed below) 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. Pin 3 doesn't need adjustment because it should be running well below clipping. Again, you control its gain with Rx value.

Next, inject a test tone into the receiver this board is being set up for. (0dbm0) Adjust VR3 to just clipping point and then, VR6 for the TLP (test level point) to drive your transmitter, or controller input. This is "IDC" mode. With U2 as a 7808, maximum unclipped output of U1 is about a +9 dbm (bridged). If you need higher output there are some options. You can leave out VR6; experiment with fixed resisters as a pad/voltage divider, or leave them out all together. There are some extra pads on the printed circuit board to do this kind of modification. You can operate U1 at higher voltages, say with a 7812 as U2 which will drive U1 output near +14 dbm (bridged). U2 keeps out any small ripple that would be amplified on the system, so most any 78xx series will work, since the audio op amp U1 uses a single end supply with voltage dividers for the "+" reference. Just watch out for the maximum operating limits of U1.

Flat Audio

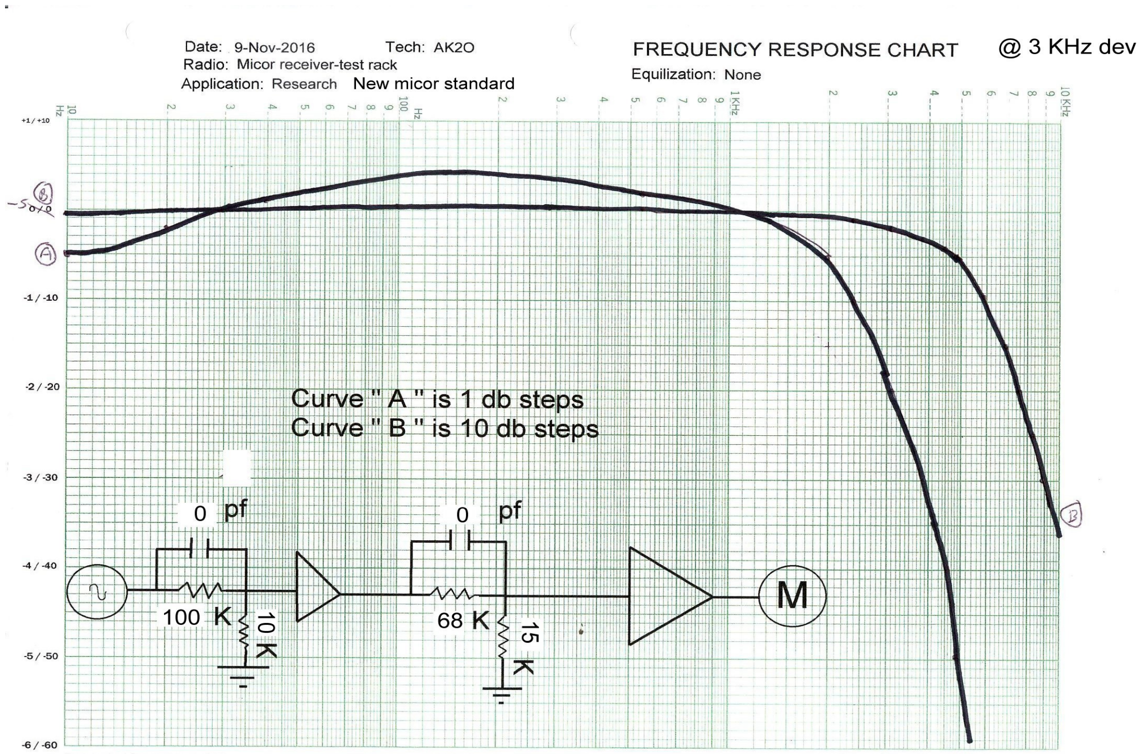
Most receivers have high end roll off. This is a conventional method for commercial systems. If you want your system to sound really good (flat) you can extend the system's frequency response. First, plot the receiver's response on a graph, from 10 Hz to 10 KHz. This sounds a little extreme, but this 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.

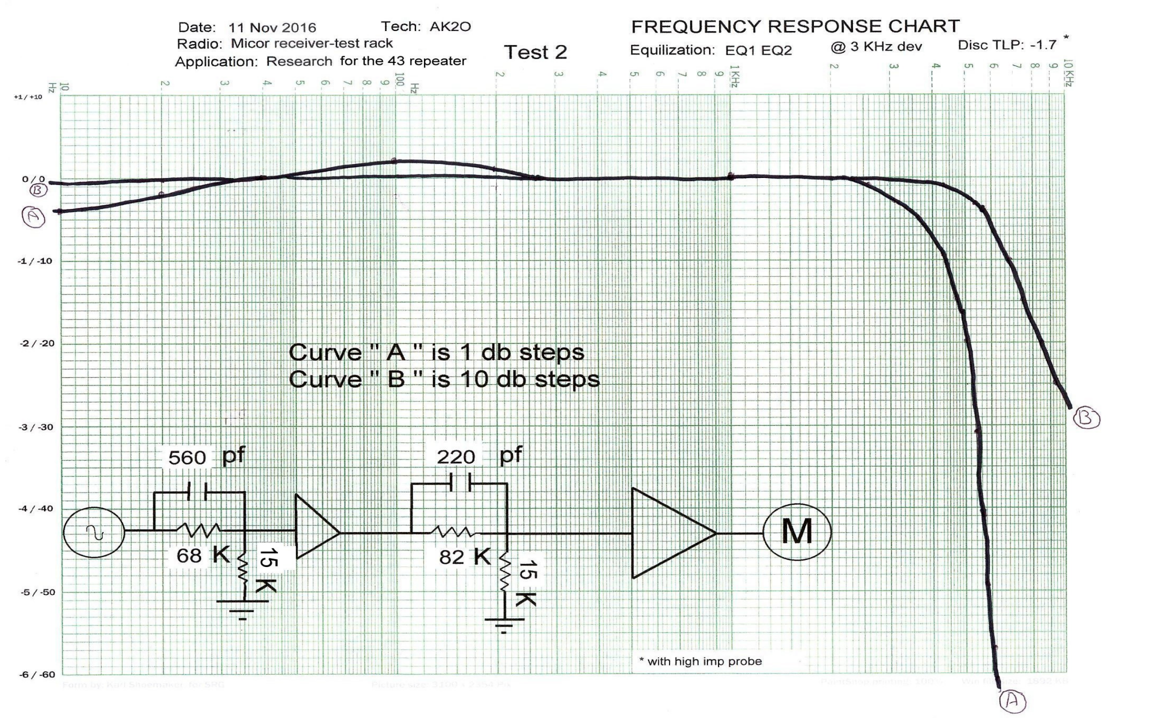
The first stage will flatten out the upper end, say above 2 KHz, and the second stage, for above 4 KHz. Typical values are in the first stage are: the series cap is .0082uf, the series resistor is 68K, the shunt resister is 15K. For the second stage, the series cap is 390 pf, the series resistor is 68K, and the shunt resister is 10K. After you replot the receiver's response should show a better (flatter) plot. With some experimentation with different values you can extend it out to around 6 KHz +- 1 db.

**2016 research:**

For prep on a new repeater (145.43) project the values were re-visited. Theoretically, the receiver frequency response should be the same whether in a mobile, compa or Spectra-Tac configuration. For this test the latter chassis and arraignment was used. A slightly better response was achieved with the first stage series cap of 560pf, the series resistor is 68K with the shunt resistor of 15K. For the second stage was a 220pf cal, with the series resistor of 82K and the shunt resistor of 15. With a couple of resistor changes (from years back) allowed a little more control with the capacitors to extend the top end a little.

Tip: If you are checking the level directly from the receiver's discriminator use a medium to high impedance AC meter. Most TIMMS are 600 ohm; even in the bridged mode will load down your reading another 2-3 db.
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 at the boards output as not to affect the loading of the receiver's discriminator's TLP of –1.7 dbm.



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

Table 1

| Input TLP | **R1** value | 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 - -  **T.L.P.**s (bridged)

| Point of Measurement | Level | Remarks (U1 being a 7808) | Noise floor |
|---|---|---|---|
| U1 Input | -10 | Max input is -10 | Rx is 2.1 Meg. |
| IF trap output | -10 | | |
| Squelch switch | -22 | Q1 collector | -63 |
| U1, pin 2 input | -58 | | |
| U1, pin 1 output | -1.5 | With Rx value of 1 M | -62 |
| Junction of 390pf and 10K | -20 | Second "EQ" stage output | |
| U1, pin 6 input | -59 | Mostly noise | |
| U1, pin 7 output | +7.2 | VR3 and 6 at maximum | -52 |
| | | | |

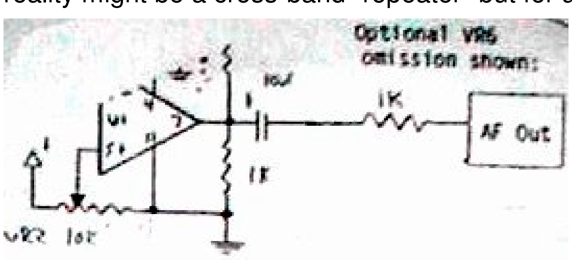
Table 3 - - **Typical DC + voltage chart:**

| **Condition** | **U1; pin 13** | **U1; pin 12** | **U1; pin 14** | **PTT 1 and 2** |
|---|---|---|---|---|
| Standby | 5.24 | 3.41 | "low" | Standby |
| Carrier active | 2.54 | 3.41 | "high" | Active |
| Timed out | 2.54 | 0.03 | "low" | Off |
| "CON-1" pulled low | N/A | 0.6 | "low" | Control off |
| **Micor downlink** | **U1; pin 13** | **U1; pin 12** | **U1; pin 14** | **PTT 1 and 2** |
| Standby | 5.24 | 0 | "low" | Standby |
| Carrier activity | 2.54 | 0 | "low" | Standby |
| Tone activity | 5.24 | 3.0 | "low" | Standby |
| Both carrier and tone | 2.54 | 3.0 | "high" | Active |
| "CON-1" pulled low | N/A | 0.6 | "low" | Off |
| Unit TO | 2.5v | 0.07 | "low" | Off |

Micor downlink voltages are for the direct interface with the PL deck. This arrangement is explain later.

Optional Arrangements and links and notes:

Protected paths will have an "AND" type squelch, requiring both carrier and tone (CTCSS) to occur to be active. For an AND squelch in a repeater or link use a tone decoder with an open collector output, normally low on standby.  A good one to use is the "Communications Specialist" TS-32. Connect the "CTCSS" to the TS-32 'Out-2', with its 'JU-2' cut. Technically speaking, the following arrangements in reality might be a cross-band "repeater" but for a user/beginner to understand it's better referred to an "downlink" or "uplink" (**down** to the control point or **up** to the high site, respectively speaking).

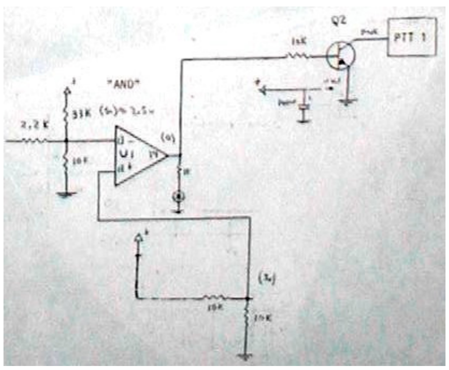



As far as the audio output if you are not using the IDC function, thus passive mode, you could leave out the AF output pot and simply terminate it with a resistor, such as shown here.  The level output is now controlled with the level adjust across pins 6 and 7.

COR input for the Motorola Mitrec only: For "H" method, U1, pin 10 would be the input, and pin 9 would be the bias. You set the bias less than the COR active high voltage. For "L" method, pin 9 would be the input, and pin 10 would be the bias. You set the bias less than the COR standby voltage, but more than the active low voltage. For "E" method, pins 9 and 10 are the same as "L". The difference with "E" is almost a 'logical' change with squelch, rather than a DC analog movement. This method is the most common used, including in the Super Consolette stock configuration.

Unless otherwise noted, **SRG uses "E".**

The "AUX AF" is a flat audio input, 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.

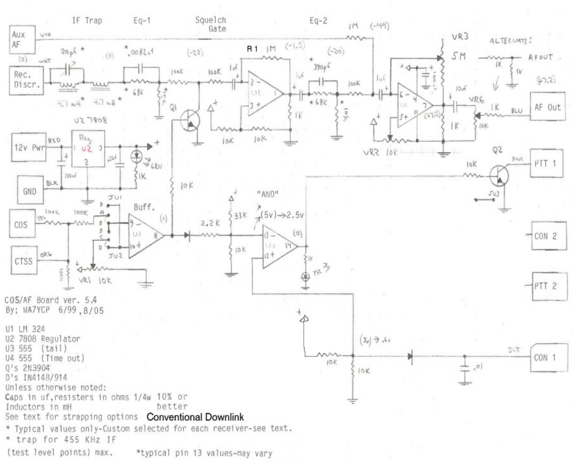



DOWNLINK

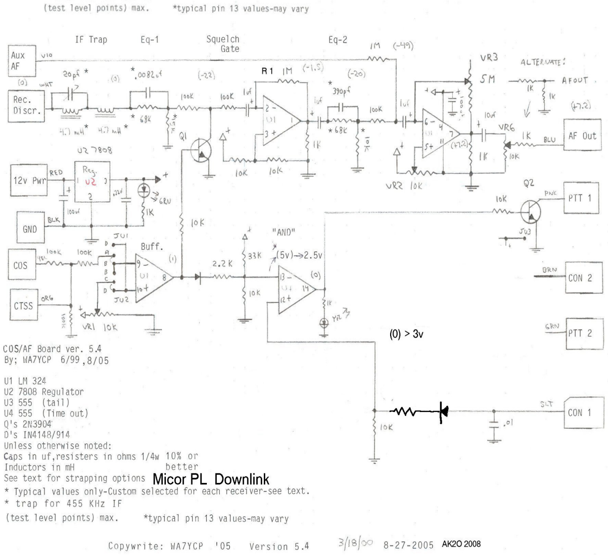
For general down link receiver applications used the PTT-1 output. It’s still timed’ however, you can disable the time-out by either grounding Q3 collector or move it’s base resistor to A+. You can also disable it by leaving U4 out.

For carrier squelch downlinks you don’t need the CTCSS connected to the COR input. In this case both audio and PTT outputs are on carrier.

Some sites have a Tx-Rx package in a single cabinet. Therefore, is not voted. Also, the tone decoding needs to be done at this remote site. Therefore, the PTT needs to be tone protected as well.. The audio path can be either way, so you have some additional choices.




If you wish a **conventional** interface to use with any PL deck (decoder) connect it to CTCSS input. This way also supports micor H.U.B. (closed loop) control. This will make both PTT and audio paths protected.

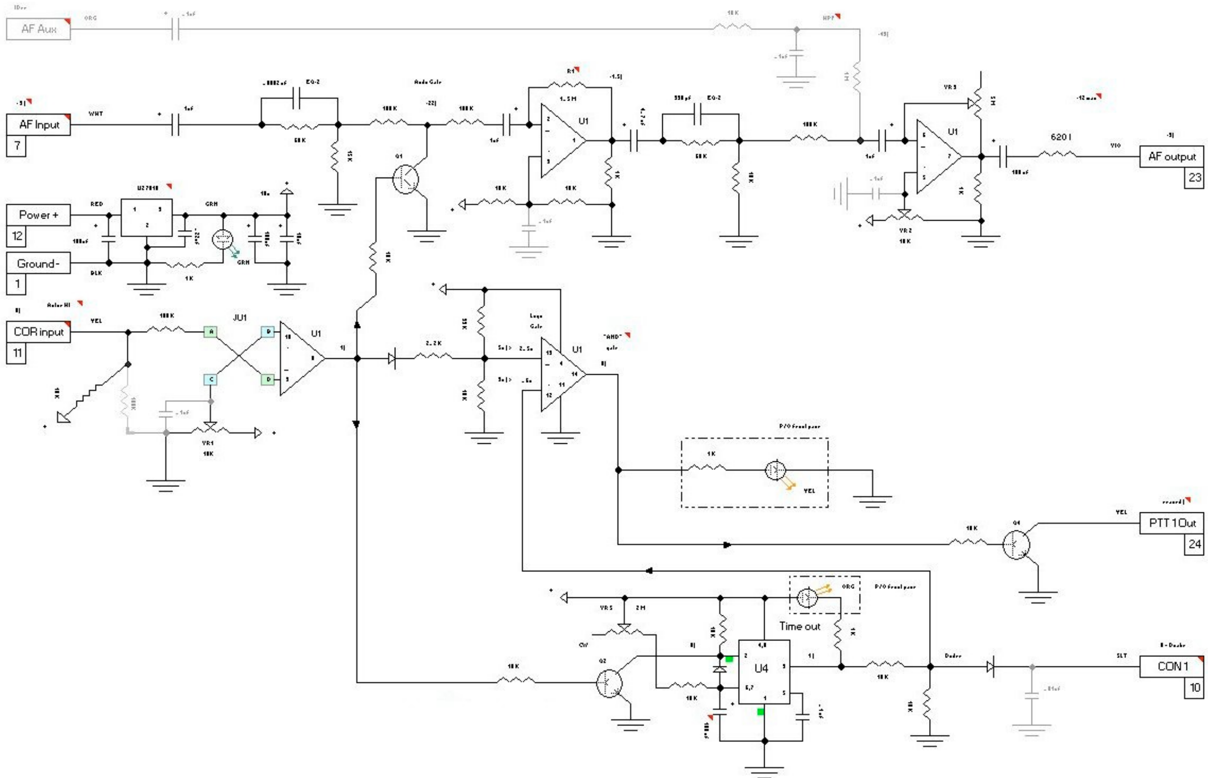


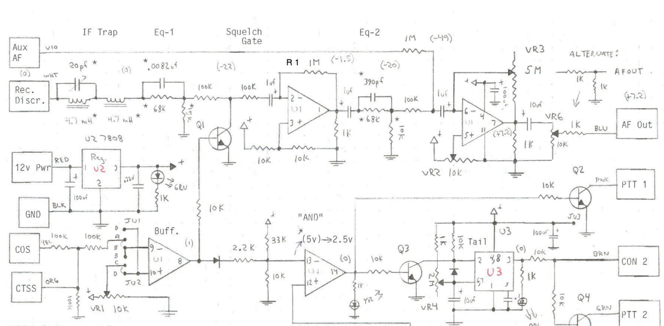

If you wish **direct** interface with the stock Motorola micor receiver and PL deck you will connect the deck's output to an external buffer, then to the COR input as in the case of the Spectra-Tac arraignments (next page).

The stock PL deck's output is a normally "low-no voltage" on standby, going an active "high" on decode however in a forced mode, which can drive a device or input.

As of 2018 the Spectra-Tac PL module is modified for SRG standards, thus an open collector is connected to the cor input, making it an “AND” squelch for both the repeat audio and PTT line.

For the Spectra-Tac version the downlink configuration is show here.  If you do not need the time out time it can be left on the board (intact) but disabling it by adding a jumper (solder bridge) at the green points. See the (separate) drawings for more details.



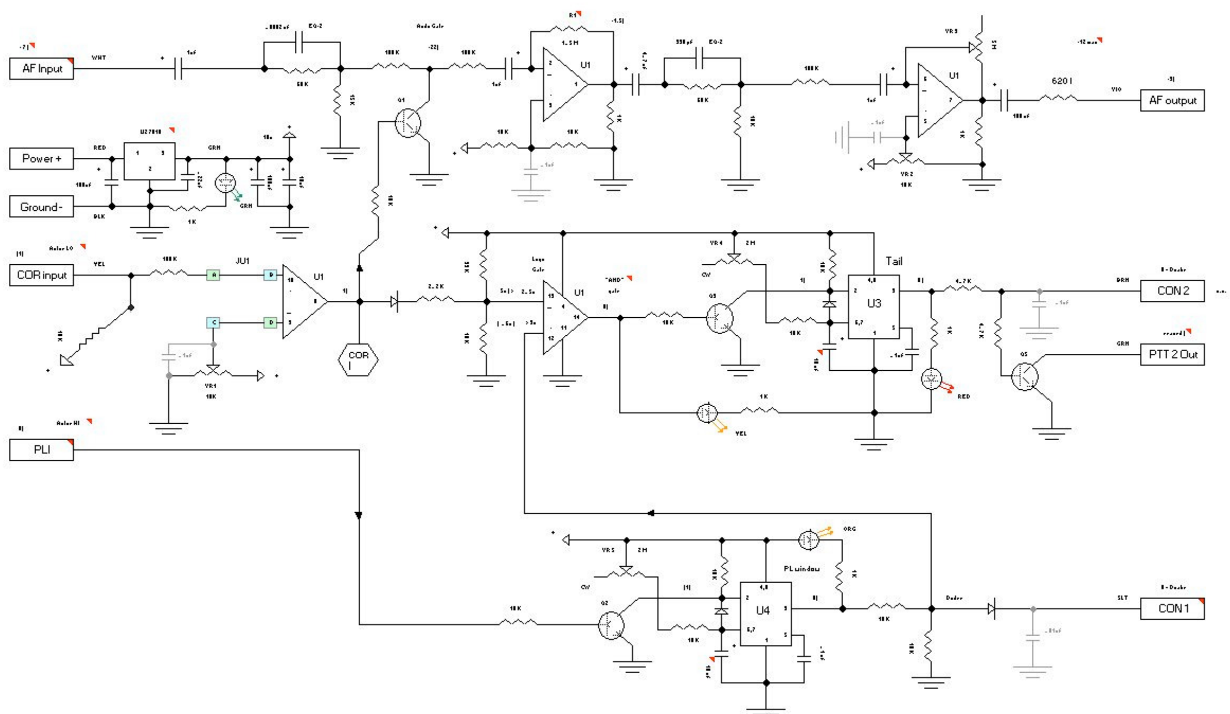

UPLINK

For remote system transmitters you will be using an uplink receiver (from the control point) connected to a system output transmitter. This application is almost the same as the **conventional** downlink strapping, except you will want to keep the U3 "tail" timer to provide a repeater output tail for the users. You will use “PTT-2” for transmitter keying in this case.

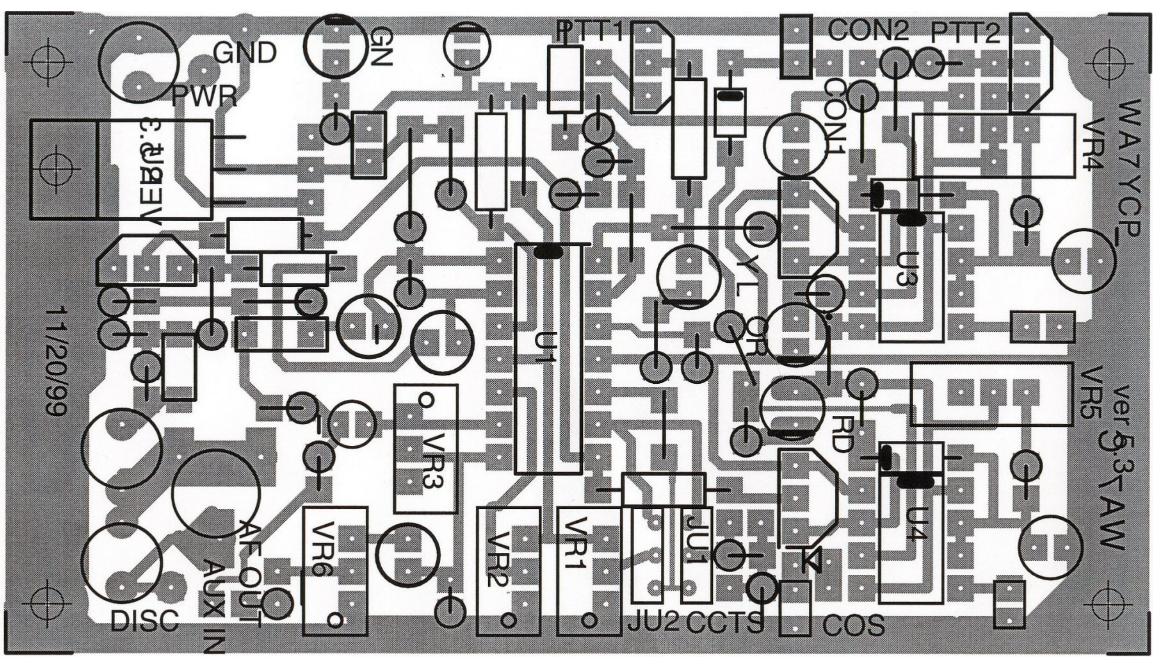
Some prefer the remote transmitter(s) to drop immediately upon link loss of control signal. For this arrangement, use the "PTT-1" to key the system transmitter.

Like the downlink, uplink can be carrier operated for both the audio and PTT paths. Only use the COR connection in this case. For tone protection for both audio and PTT, use the COR input for carrier activity and the tone decoder's output connected to the CTCSS input of the board.

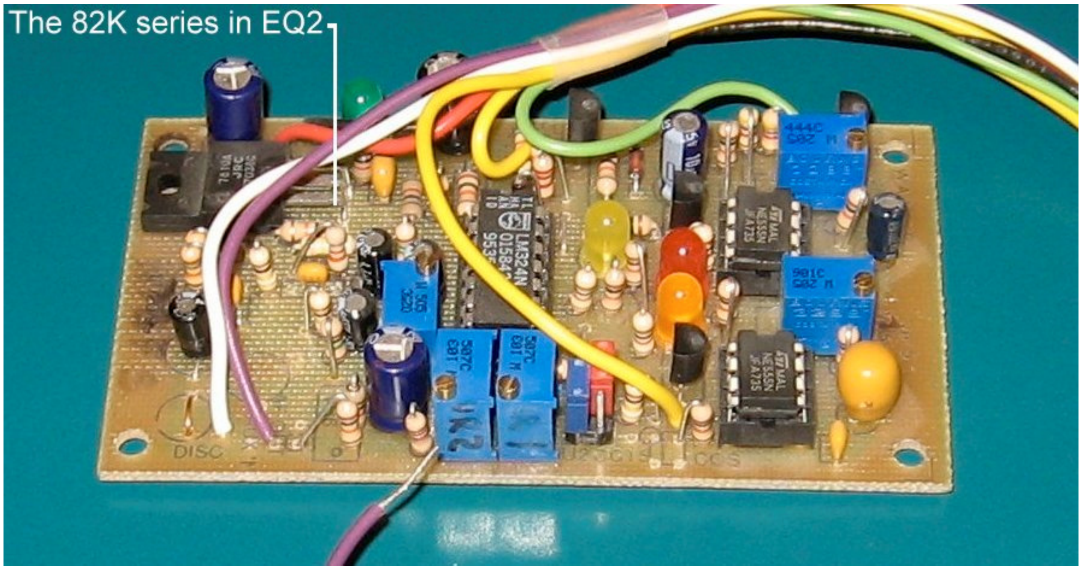
Operating  system on sites presents issues; one being interference from other radios, which gets into carrier operated receivers. A large (linked) system of remotes increases this risk.  Tone protection corrects some of this, however with it's own issues. One issue is tone decode has a delay to let audio through, which can cause missed (first) words and possible user confusion. SRG uses a different arrangement to address this problem. The uplink PTT path is tone protected while the audio path is on carrier squelch (CS). Tone protection allows signaling for a repeater "tail" while user conversations go through the system quickly since there's no tone decode at the remote site for the audio path. Shown here is the arraignment for the board. There's a slight advantage; if the signaling is momentary lost that won't affect the repeat audio; and won't affect the "tail" if the signaling returns before the tail drops out.




Shown here is the fully loaded board with both tail and time-out timers installed.




Notes: 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 board and layout is designed to be universal and easy to modify for your particular needs, limited to your imagination. The standard arrangement (full) schematic is on a separate sheet.




Parts List: (resistors may be difference 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% | | 291-33K | 00.07 |
| 1 | Resistor, 15K, 1/4w, 5% | | 291-15K | 00.07 |
| 13 | Resistor, 10K, 1/4w, 5% | | 291-10K | 00.91 |
| 1 | Resistor, 2.2K, 1/4w, 5% | | 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 | 594-64W205 | 04.00 |
| 3 | Trim-pot,multi,10K,inline leads | VR 1,2,6 | 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, 150uf/15v | | (vender TBD) | 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 | | |

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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 (800) 346.6873 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. 11.2 MHz, etc.

* Used for custom selection for each receiver. Add 1 more 1K resistor if not using VR6.

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



http://www.srgclub.org