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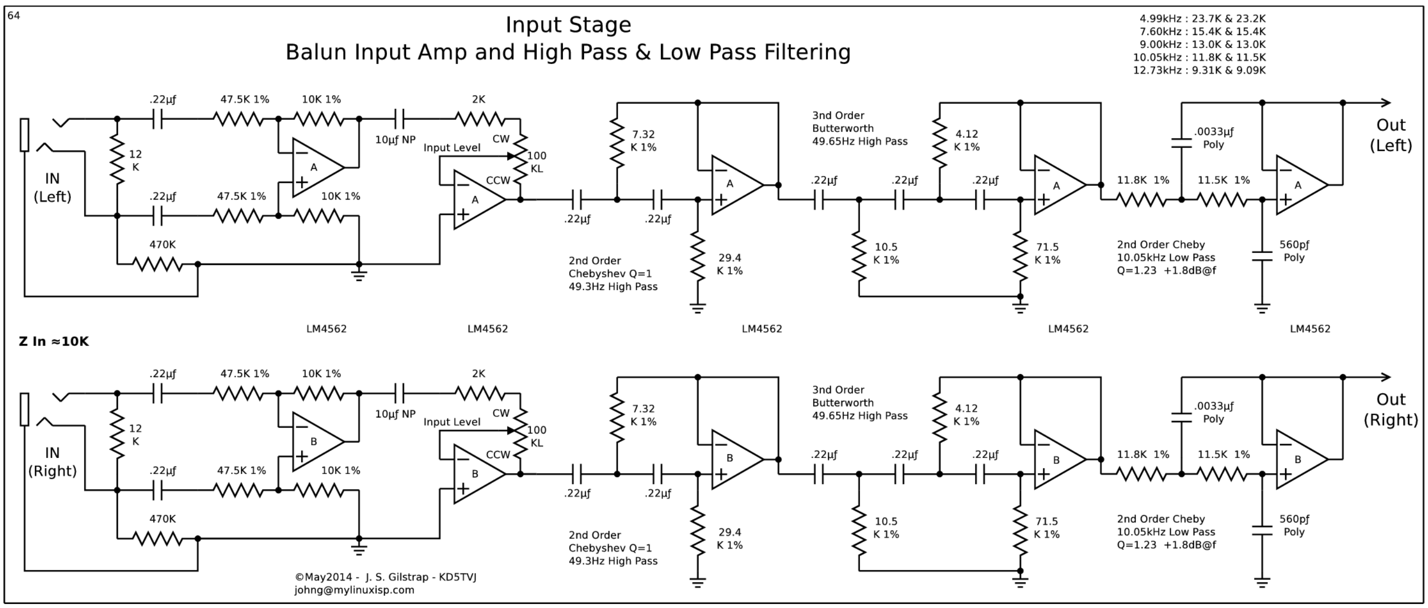
Check out: 100mW TX Mono 1610AM Transmitter article.

C-QUAM® & C-ISB™
AM Stereo Exciter

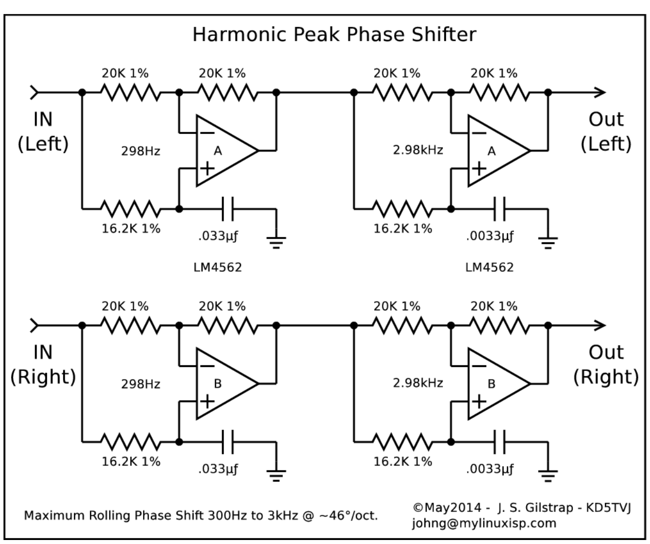
Generating a C-QUAM or C-ISB™ signal requires some pre-processing to produce a quality signal. Bandpass filtering, harmonic phase shifting, pre-emphasis, dynamic limiting, matrixing to name most of them. The following are schematic drawings for the various processes needed to prep the signal for the C-QUAM/C-ISB™ modulator. The modulator has an output section that will produce a composite output or the PM output at TTL & low level 100mV square waves along with L+R audio for the AM component for transmitter interface. The L+R audio also contains a DC component representing the '1+' carrier level providing a DC coupled output. Audio output impedance is 600Ω and is level adjustable to +12½dBm. Linear QuAM & ISB output modes are readily available for composite output by connecting the composite output driver to the output tank of the carrier, I & Q modulators instead of to the output tank of the C-QUAM modulator. For the separate AM & PM components the PM is readily available but the AM component must be recovered from the QuAM output with a pseudo-synchronous detector.

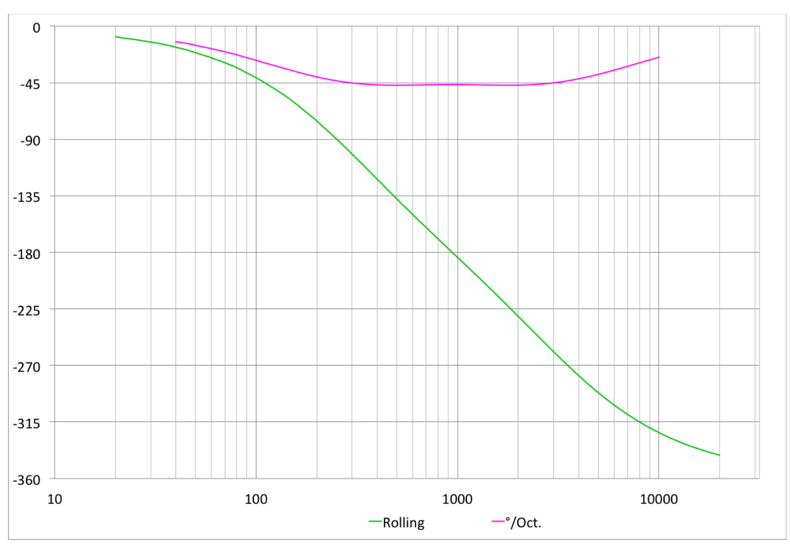
NOTE: When referring to I & Q this is synonymous with [1+]L+R & L─R respectively.

The first image is the input section. For common mode rejection a balun op-amp is used. The input is a ¼” stereo phone jack and in relation to stereo use the tip for Left is (+), the middle ring for Right is (─) and the base ring is ground. If a mono ¼” plug is plugged into it then the contact for the middle ring (─) will be grounded and will convert the input to un-balanced. Next are the 2nd order Chebyshev & 3rd order Butterworth high pass filters at ~50Hz with a combined 30dB/Oct. attenuation and finally the 2nd order Chebyshev low pass filter at ~10kHz (selectable 5, 7⅗, 9, 10, 12¾) with 12dB/Oct. attenuation. A Chebyshev response is used here with a slight boost and in combination with a 3rd order Bessel low pass filter with a similar cutoff frequency later in the processing will simulate a good 5th order Butterworth low pass filter response. For both these filters other component values are specified for different cutoff frequencies.

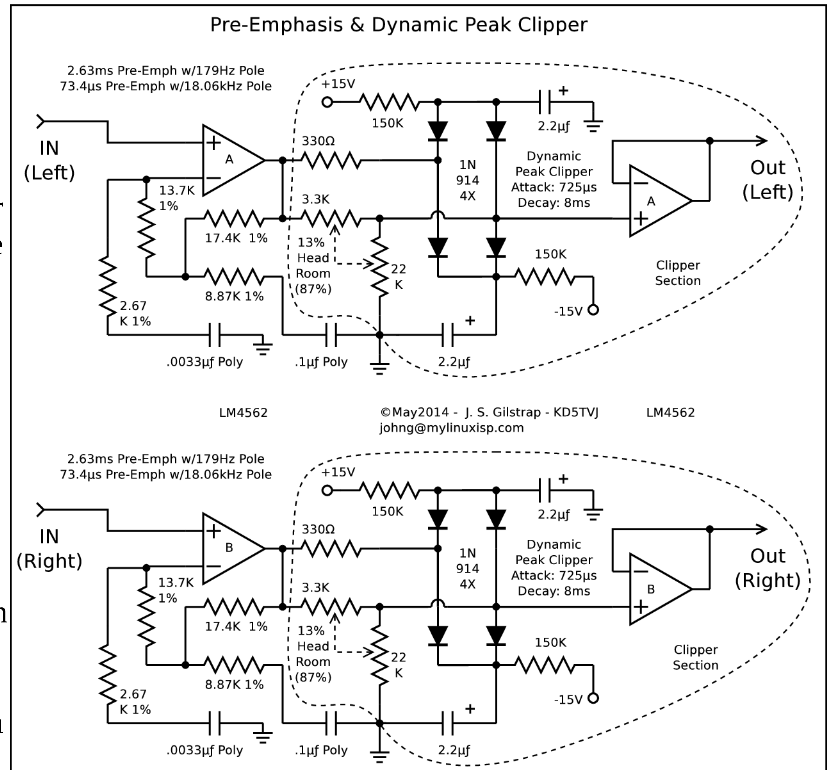


This next drawing is a phase shift network. When an ISB signal is generated phase shift networks similar to these are used to produce a rolling differential phase shift of 90° between L+R & L─R. A beneficial side effect is that this phase distortion which is un-noticeable to the human ear shifts the harmonics off the peaks of the fundamentals reducing peak amplitude but not volume level allowing for increased modulation level. Here the phase shift is the same for both channels and provides a fairly even rolling phase shift across most of the audio passband with a maximum of ~46°/oct. from 300Hz to 3kHz. Depending on programming material and audio bandwidth different frequencies could be used to produce an optimized response. This is not required for C-QUAM but can be beneficial. While C-ISB mode already provides a differential phase shift adding this may also reduce peak modulation levels further.



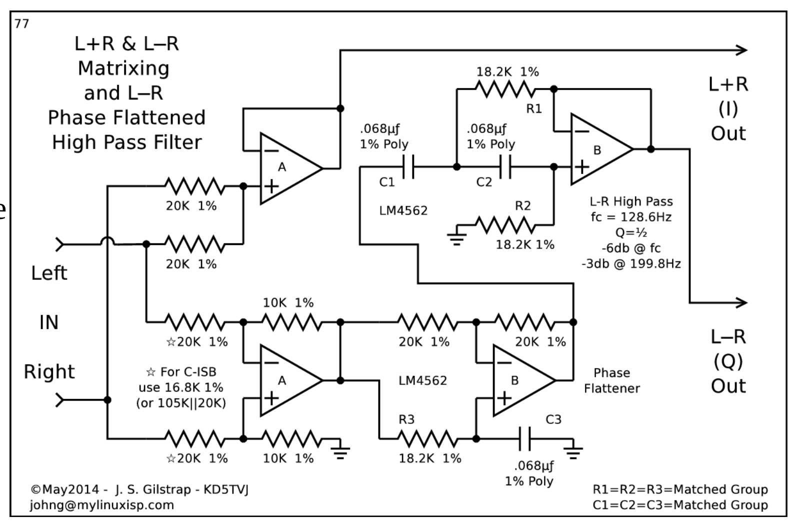


The next image is the pre-emphasis network with a ~75μs response and a pole at 18.06kHz with a dynamic peak clipper afterwards. While the standard pre-emphasis has the pole at 8.7kHz this has been extended higher up to compensate for the gradual roll off of the higher frequencies created by the bandpass tanks in the modulators to approximate the 8.7kHz pole. There is also a bass de-emphasis applied to allow more headroom for mid and high frequency modulation levels. It starts at ~60Hz and levels off at ~180Hz. The bass frequencies are usually the strongest in the signal and rob much of the headroom from the rest of the signal and is the least affected by noise. This ~9dB attenuation at ~60Hz is mild compared to the RIAA curve applied to vinyl records and can be compensated using the bass control on most receivers or the use of bass heavy headphones. To defeat this replace the 0.1μ*f* capacitor with a 1μ*f* NP. The Dynamic peak clipper has an attack time of <1ms and is defined by the 330Ω resistor charging the 2.2μ*f* capacitors. The decay time is set by the 150K resistors in relation to the 2.2μ*f* capacitors. The 3.3K & 22K resistor attenuation network that defines headroom helps to minimize over clipping during initial capacitor charging during initial signal response and can be adjusted by changing the value of the 22K. All three of these parameters can be adjusted for optimal performance. The clipper is placed after the pre-emphasis here but the most ideal place for it may be after all filtering but somewhere just before final Bessel low pass filter.

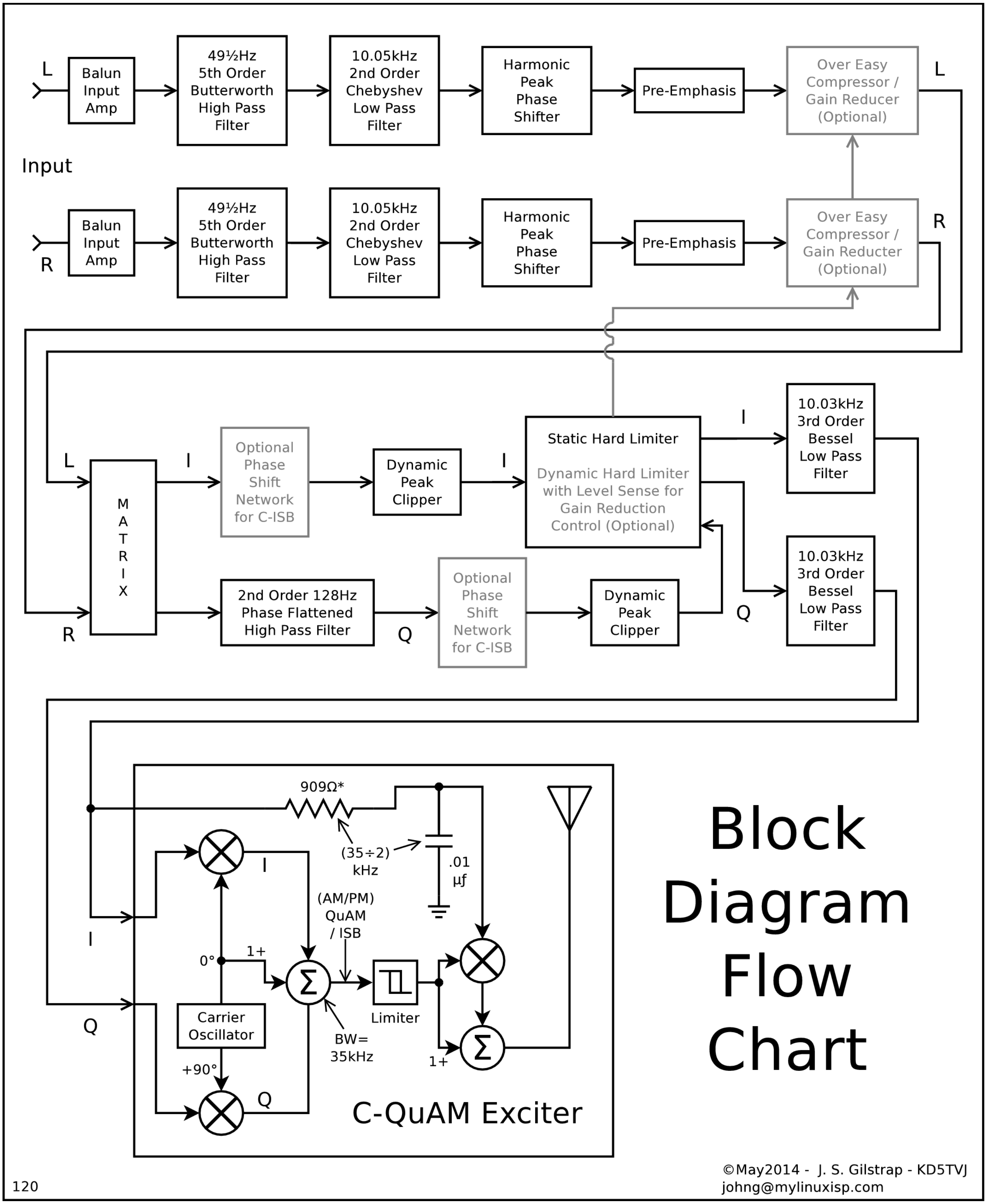


While the clippers here are placed just before matrixing placing them after operating on L+R & L–R rather than L & R will also respond to any peaks that emerged out of the of L & R mixing in both sum and difference and the high pass filterering of L–R. Matrixing along with the phase shift networks used for C-ISB will re-arrange the peaks of the harmonics in relation to the fundamentals and will produce new peaks that didn't exist before. This is normal for all frequency dependent filtering where frequency and/or phase relationships are altered and when signals are mixed during matrixing. As a result these clippers are best placed after all mixing and filtering that does not have a maximally flat delay response and for C-ISB mode this means placing them after the phase shift networks also. Idealy dynamic clipping should be placed after all filtering and before a wideband compressor used for limiting the maximum modulation level going into the modulators.

This next drawing is the matrixing of L & R into L+R & L─R and a high pass filter for L─R. This 2nd order high pass filter with a Q of ½ has a complimentary all pass phase shift network that applies an equal but opposite phase shift to cancel out the phase shift of the high pass filter to produce a 0° phase shift high pass filter response. If C-ISB mode is used then to optimize sideband separation increasing L─R by ~1½dB will help reduce 2nd order harmonics on the opposite sideband during average modulation levels. This can be done by replacing the 20K resistors with 16.8Ks or placing 105Ks in parallel with the 20Ks. The receiver will need to have a 1½dB gain reduction applied also.

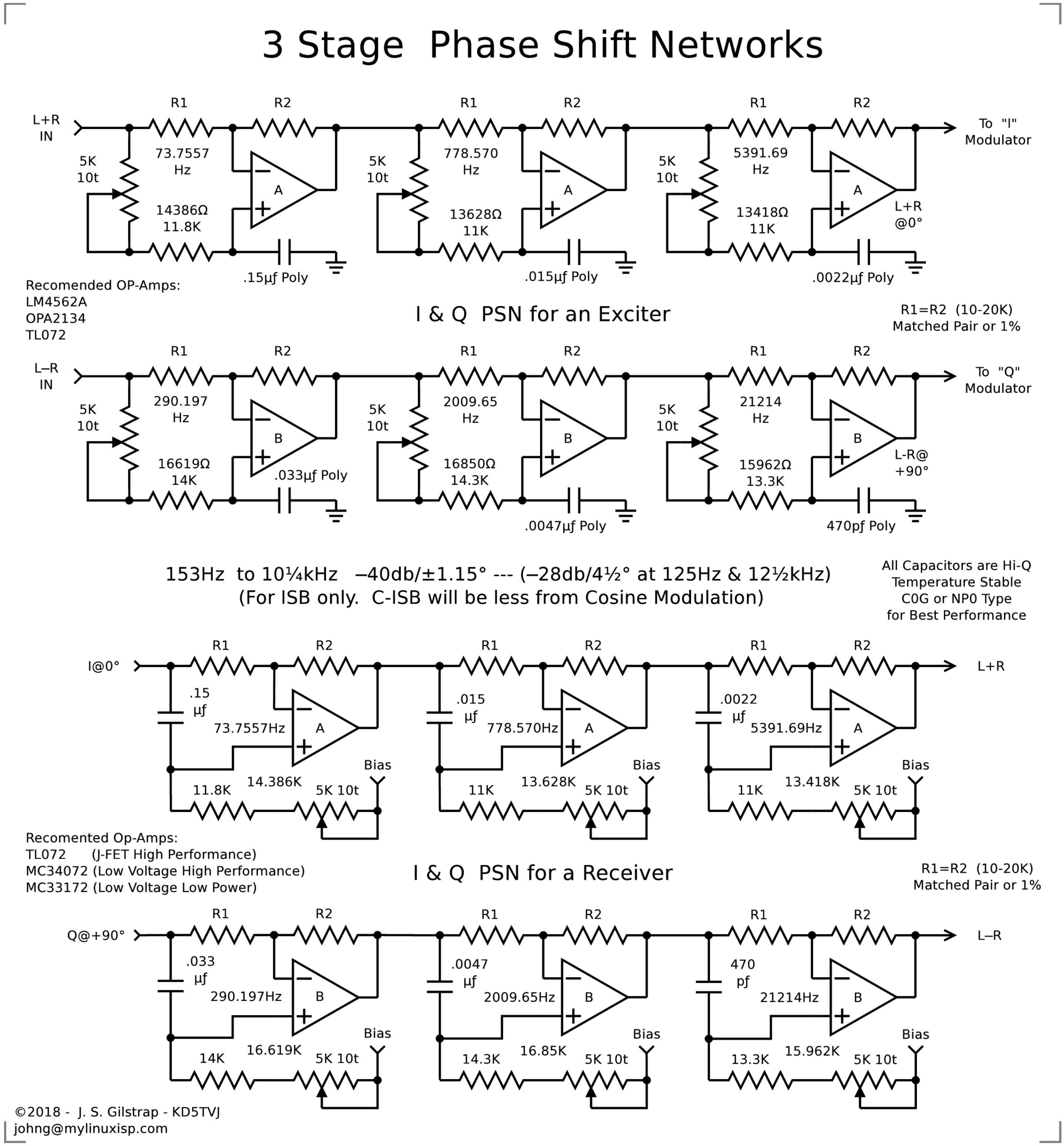


The next image is the block diagram flow chart. It shows all the processes in a sequence. The sequence can vary somewhat depending on several goals and whether it is in C-QUAM or C-ISB™ operation. For C-QUAM the actual gain reduction for the over easy compression is best applied to the separate L & R channels prior to matrixing but sensing for compressor control should be obtained after matrixing. For C-ISB™ while applying the gain reduction before matrixing may offer better single channel control during heavy single channel modulation good results can be obtained when gain reduction is done after matrixing offering more of a stereo expansion effect.



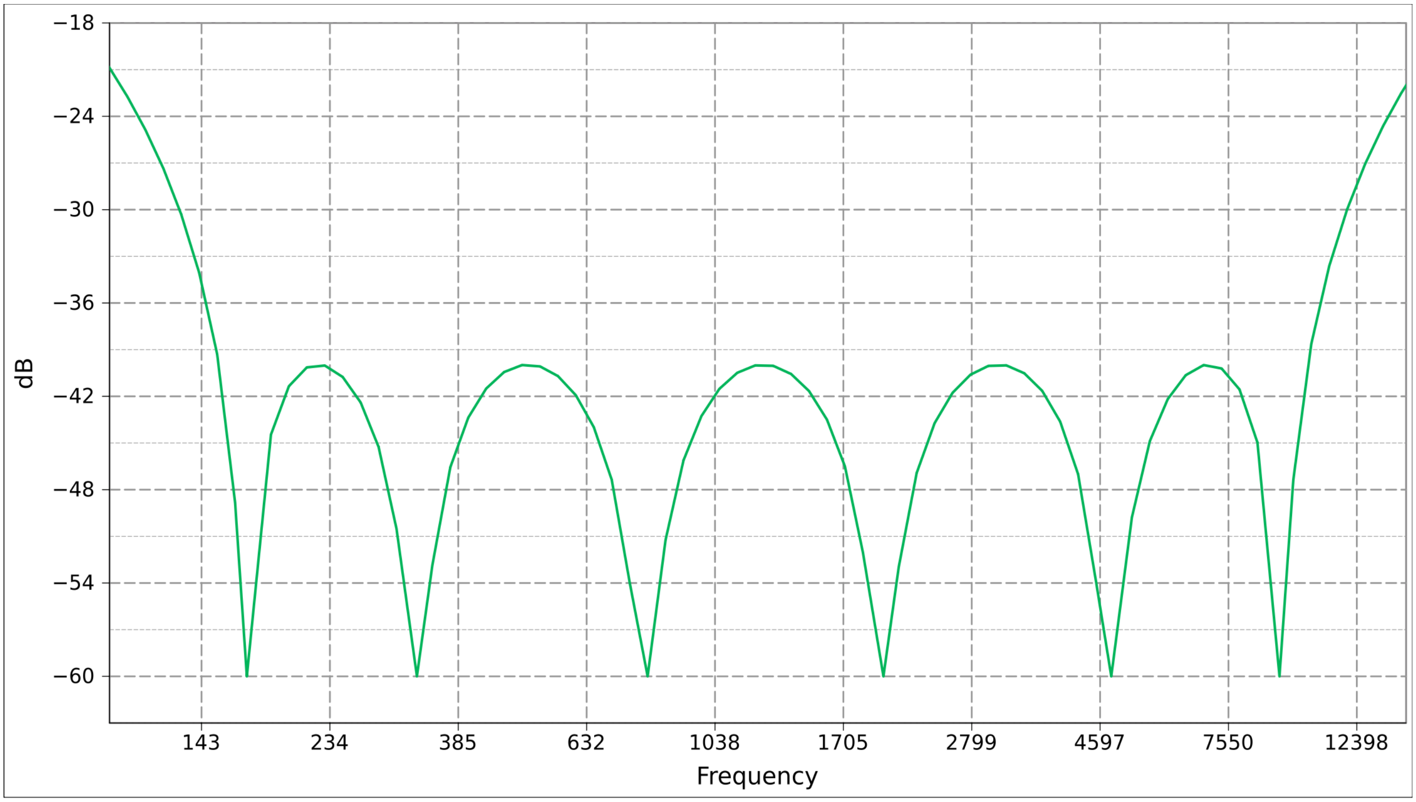
Suggested optimal order after input section. [C-]QUAM: Harmonic Phase Shift Networks, Pre-Emphasis, Gain Reduction / Over Easy Compression, Matrix, L─R High Pass Filter, Dynamic Peak Clipper, Dynamic Hard Limiter with Level Sense for Compressor, Matched Bessel Low Pass Filters. [C-]ISB: Harmonic Phase Shifter, Pre-Emphasis, Matrix, L─R high Pass Filter, [C-]ISB Phase Shift Networks, Dynamic Peak Clipper, Gain Reduction / Over Easy Compression on I & Q Stereo, Static Hard Limiter || Dynamic Hard Limiter with Level Sense for Compressor, Matched Bessel Low Pass Filters.

In the top half of this image is the PSN for [C-]ISB transmission and is placed after the matrix and L─R high pass filter. This is a three stage phase lagging network for transmission.



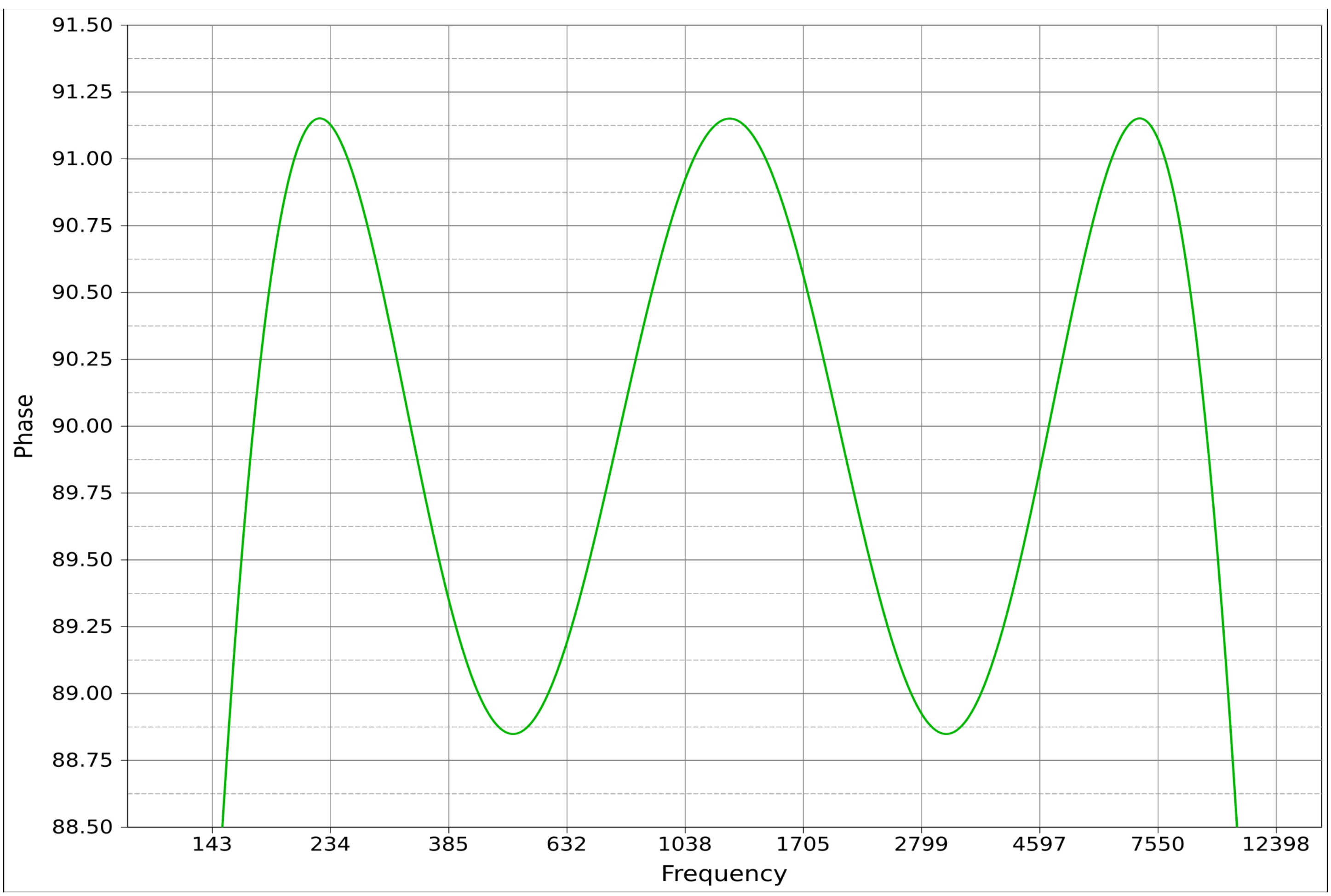
In the bottom half of this image is the PSN for the receiver and is a 3 stage phase leading network using the same frequencies to return the signal's phase to its original state.

This 3 stage PSN has a -40dB opposite sideband suppression minimum from 153Hz to 10¼kHz and at 125Hz & 12.5kHz it is still -28dB for ISB. C-ISB™ Cosine modulation will reduce this. If 2 independent non-stereo channels are used with (No C-) ISB then the 4 stage unit described in 'Discrete QuAM + ISB' could also be used for better sideband suppression but choosing this 3 stage is more economical making it a good standard for all TX/RX.

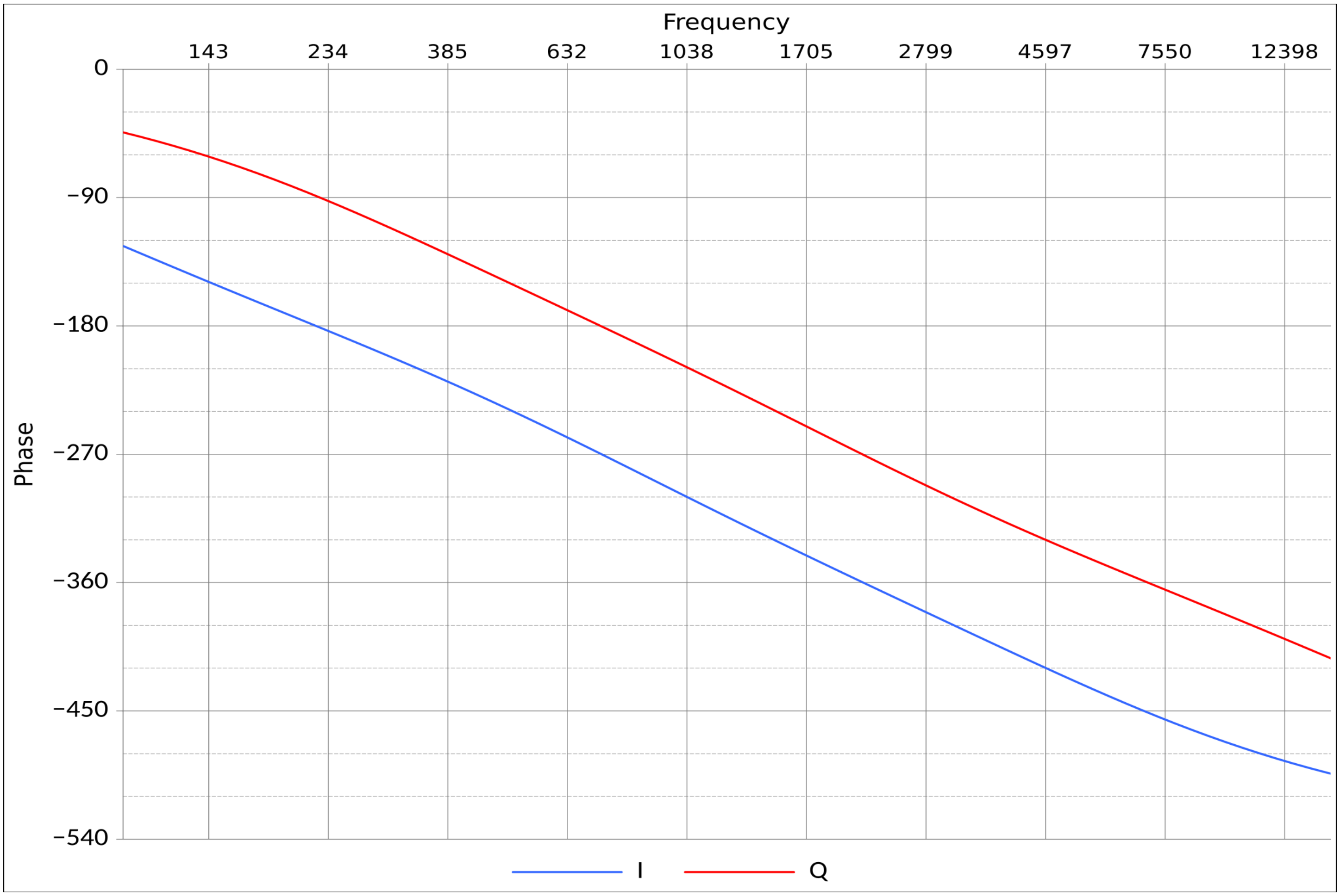


Above is the Opposite Sideband Suppression in dB vs Frequency for ISB.

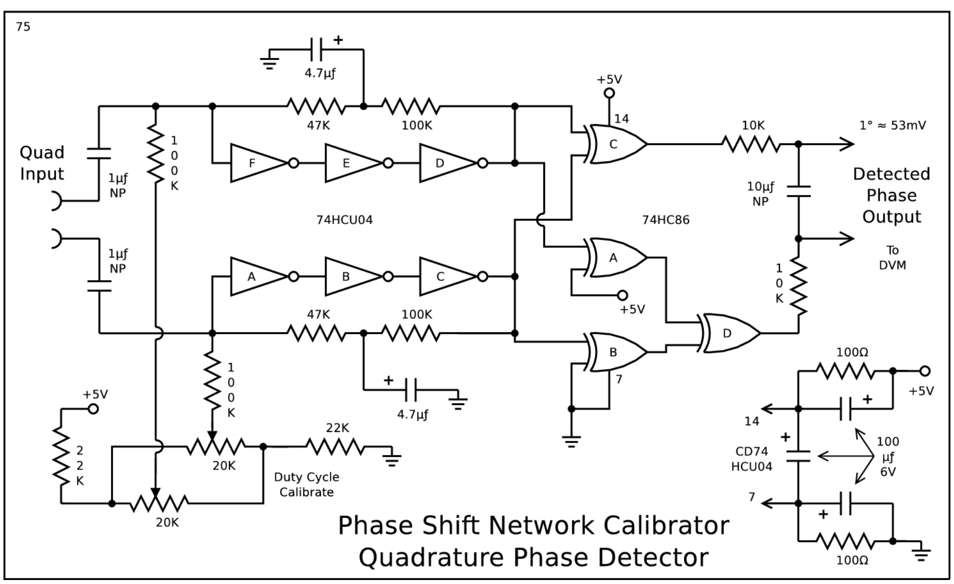
Below is the Differential Phase Shift between I & Q channels and Deviation away from 90°.



Phase Shift for I & Q Vectors providing a maximum ~58°/Oct. Rolling Phase Shift.



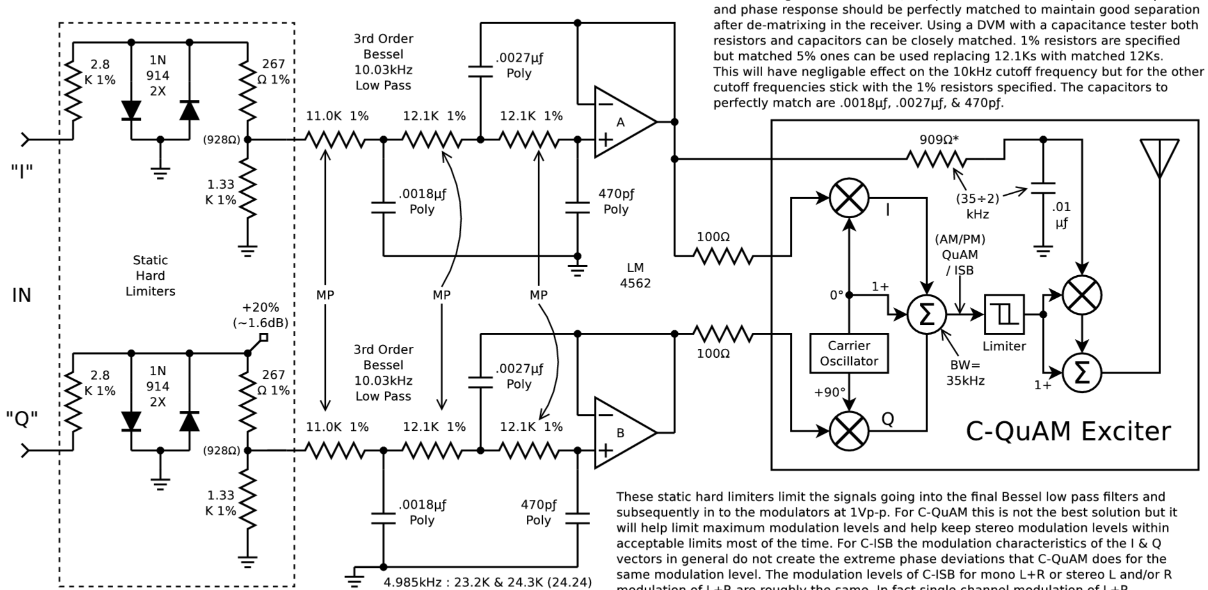
This is the calibration tool to align the phase shift networks for both TX & RX. At its resonance each individual phasor will produce a 90° phase shift between its input and output. This detector will produce a voltage proportional to the deviation away from 90°. Using a DVM each phasor is calibrated when the meter reads 0.000V when the single sinusoidal frequency specified for it is injected into its input. The detector itself will need to be calibrated for 50% duty cycle on the final hex inverter outputs and tested first on a known sinusoidal quadrature signal. A 1Vp-p signal should produce full limiting with a fast rise and fall on the square wave.



As an alternative to applying the +1½dB boost for C-ISB in the matrix circuit it may be more appropriate to apply it after the limiter as shown here, the +20% or ~1⅗dB point. Connect the 11.0K resistor to it instead of where it is now.

Final Limiters & Low Pass Filters Preceeding C-QuAM Modulator

After limiting involving clipping harmonics are generated and if the clipping is not too severe then the vast majority of the harmonics generated will exist outside audio passband and can be removed with a low pass filter without any audible effect to the signal. These filters require a flat delay time in the passband so no overshoot will be generated on the clipped parts of the signal. For this a Bessel filter is used and since this occurs after matrixing the L & R signals into the I & Q components these filter's passband amplitude and phase response should be perfectly matched to maintain good separation after de-matrixing in the receiver. Using a DVM with a capacitance tester both resistors and capacitors can be closely matched. 1% resistors are specified but matched 5% ones can be used replacing 12.1Ks with matched 12Ks. This will have negligable effect on the 10kHz cutoff frequency but for the other cutoff frequencies stick with the 1% resistors specified. The capacitors to perfectly match are .0018µf, .0027µf, & 470pf.



4.985kHz : 23.2K & 24.3K (24.24)
7.44kHz : 15.4K & 16.2K (16.24)
9.08kHz : 12.4K & 13.3K (13.31)
10.03kHz : 11.0K & 12.1K (12.04)
12.37kHz : 8.87K & 9.76K (9.773)

These static hard limiters limit the signals going into the final Bessel low pass filters and subsequently in to the modulators at 1Vp-p. For C-QuAM this is not the best solution but it will help limit maximum modulation levels and help keep stereo modulation levels within acceptable limits most of the time. For C-ISB the modulation characteristics of the I & Q vectors in general do not create the extreme phase deviations that C-QuAM does for the same modulation level. The modulation levels of C-ISB for mono L+R or stereo L and/or R modulation of L+R are roughly the same. In fact single channel modulation of L+R can approach 95% before the same phase deviation is reached for C-QuAM on the same program material where C-QuAM would require a limit of 75% for L+R. Since most program material today is sourced from digital formats coming from a mostly digital process where a practice of using a high level of limiting and normalizing is common this program material usually requires little if any processing to prep it for modulation. With proper input level adjustment the hard limiter here would be adequate for the occasional peak overshoots. Using a simple stereo RMS weighted over easy compressor (soft limiter) would produce better results while the hard limiter would be there to limit the rarer overshoots.

C-QuAM modulation needs more dynamic level control depending on mono, 2 channel stereo or single channel stereo modulation to replace these static hard limiters. Standard limiters and compressors will not provide adequate limiting without greatly reducing overall signal level. The requirements are: -100% of L+R modulation on a mono only signal, -85% modulation of L+R during stereo, and -75% modulation of L+R during L or R single channel stereo modulation. In all cases during stereo modulation phase deviation must be kept within ~±72°. For single channel modulation this occurs when L+R is downward modulated to 75% which leaves the "I" vector at 25% and the "Q" vector at ±75% producing a geometry of ±75/25=±3 or ArcTan(±3)=±71.56°. Matrix processors external to the exciter are generally used for this which take all these parameters into account. Since the geometry that defines these parameters has a DC bias component optimal control is best realized when the matrix processing is intergrated into the exciter that maintains this DC bias throughout the modulation process. The block diagram flow chart shows a general layout of one way this would be intergrated into the exciter.

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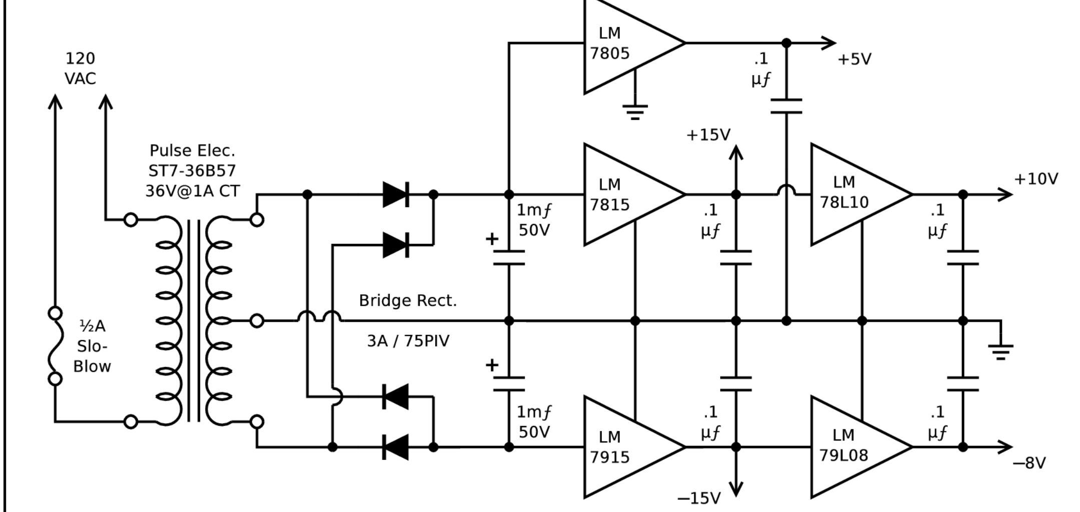
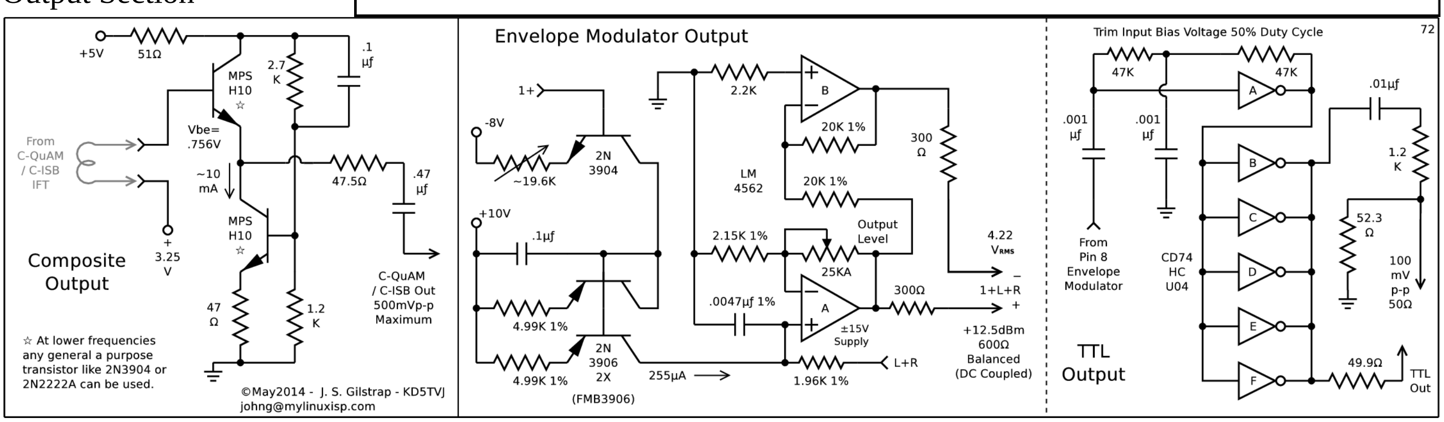
Example Power Supply

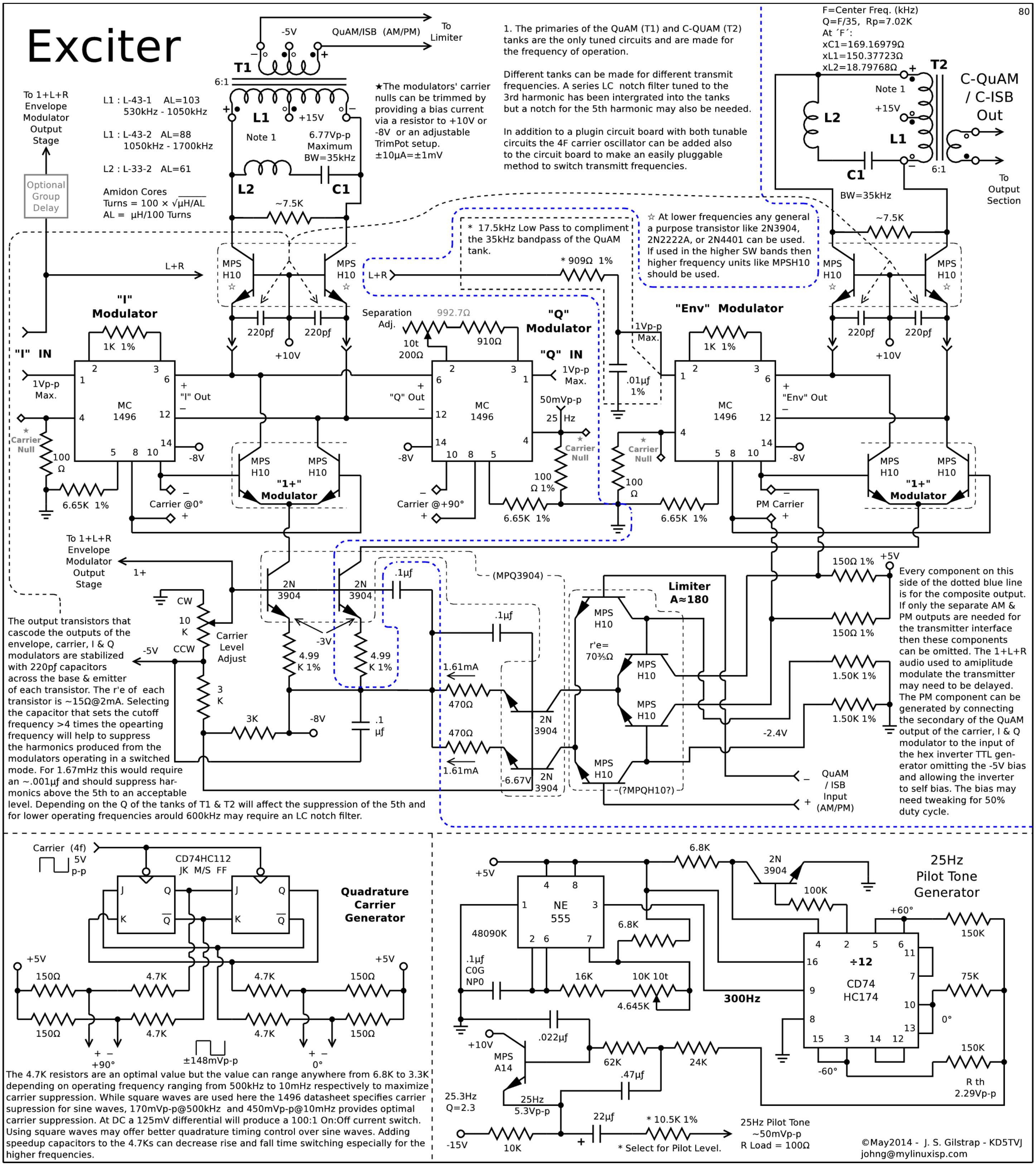
Regulated Power Supply



Output Section



$$L2=1/[(6\pi f)^2 \times C1] \ \ \& \ \ L1=1/[(2\pi f)^2 \times C1] - L2$$

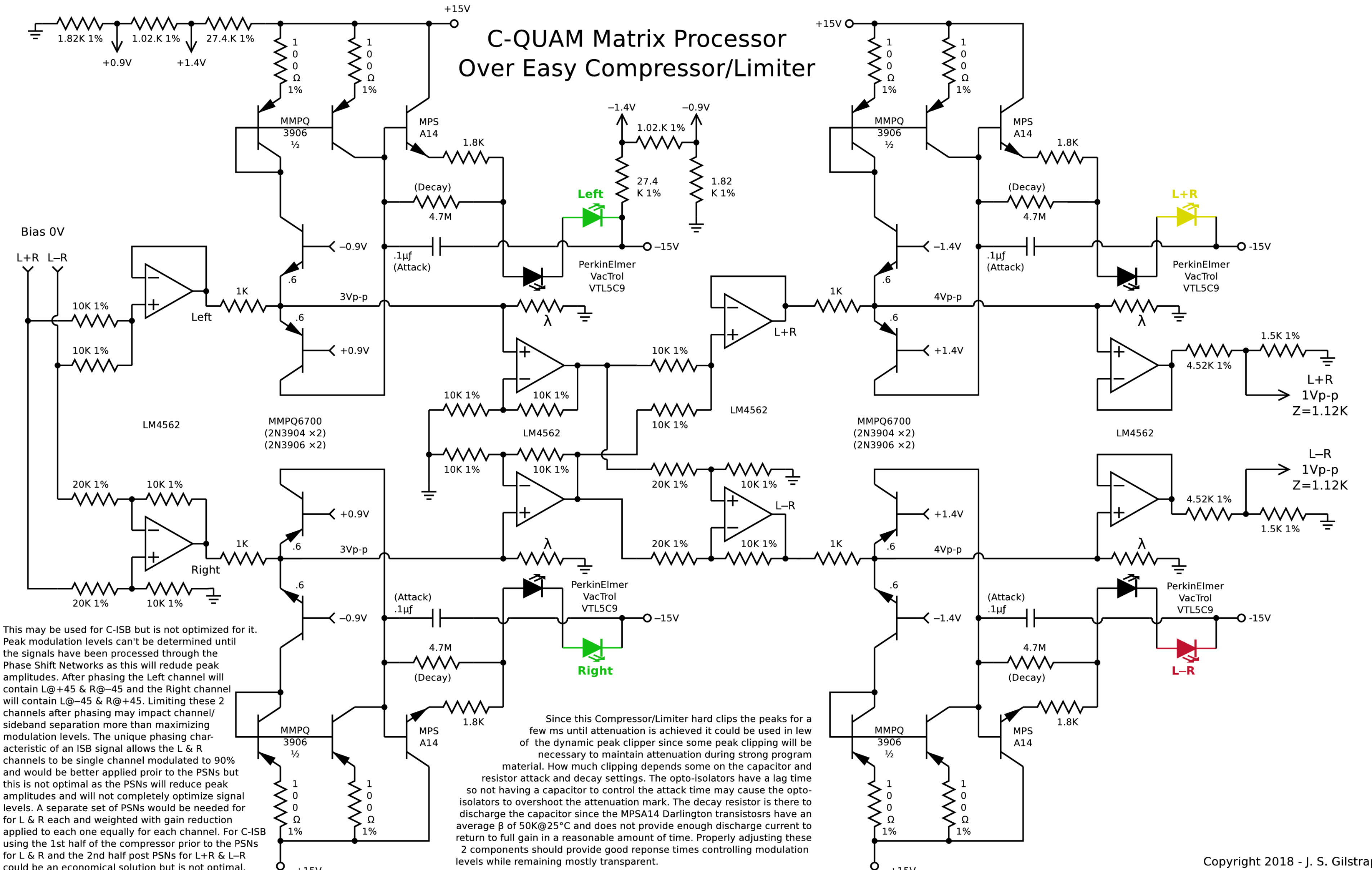


A clock source at 4× the transmit frequency will need to be provided.

For C-ISB conversion of a C-QUAM decoder see: https://www.amstzone.org/QuadMod-4Way_13020.pdf

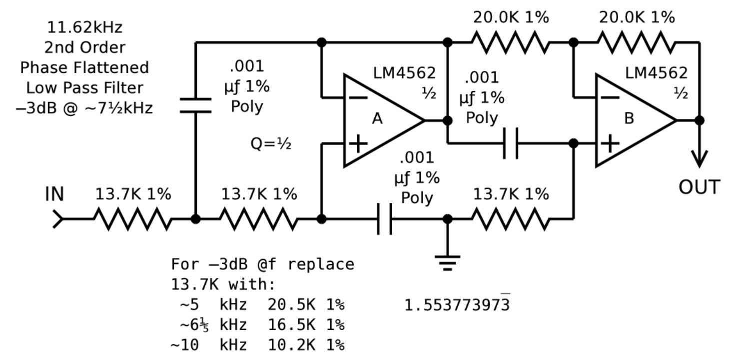
C-QUAM is a registered trademark of Motorola.

C-QUAM Matrix Processor
Over Easy Compressor/Limiter



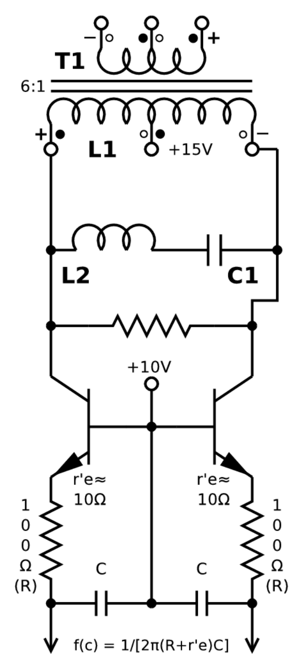
This may be used for C-ISB but is not optimized for it. Peak modulation levels can't be determined until the signals have been processed through the Phase Shift Networks as this will redude peak amplitudes. After phasing the Left channel will contain L@+45 & R@–45 and the Right channel will contain L@–45 & R@+45. Limiting these 2 channels after phasing may impact channel/ sideband separation more than maximizing modulation levels. The unique phasing char- acteristic of an ISB signal allows the L & R channels to be single channel modulated to 90% and would be better applied proir to the PSNs but this is not optimal as the PSNs will reduce peak amplitudes and will not completely optimize signal levels. A separate set of PSNs would be needed for for L & R each and weighted with gain reduction applied to each one equally for each channel. For C-ISB using the 1st half of the compressor prior to the PSNs for L & R and the 2nd half post PSNs for L+R & L–R could be an economical solution but is not optimal.

Since this Compressor/Limiter hard clips the peaks for a few ms until attenuation is achieved it could be used in lew of the dynamic peak clipper since some peak clipping will be necessary to maintain attenuation during strong program material. How much clipping depends some on the capacitor and resistor attack and decay settings. The opto-isolators have a lag time so not having a capacitor to control the attack time may cause the opto- isolators to overshoot the attenuation mark. The decay resistor is there to discharge the capacitor since the MPSA14 Darlington transistosrs have an average β of 50K@25°C and does not provide enough discharge current to return to full gain in a reasonable amount of time. Properly adjusting these 2 components should provide good reponse times controlling modulation levels while remaining mostly transparent.



To the right is a Phase Flattened Low Pass Filter. Use this if the out of band emissions exceed the transmission mask from heavy Stereo (L-R) modulation. It is tuned for a -3dB cutoff at ~7½kHz although its corner frequency is above 11½kHz with a Q of ½. Both C-QUAM & C-ISB are non-linear multiplicative AM/PM modulation systems producing up to 6 times the modulating frequency in harmonics. Reducing the higher frequency content of L-R will reduce the out of band emissions to an acceptable level. The chosen cutoff frequency will depend on the high frequency content of the L-R program material and the amount of reduction of out of band emissions needed. The corner frequency is determined by the 3 resistors chosen in conjunction with the 3 .001µ*f* capacitors using 1/(2πRC) and is 1.553... times higher than the -3dB point. There are 3 other cutoff frequencies specified and the resistors needed. The 3 capacitors and 3 resistors should be matched with each other or 1%. 15.0K resistors will produce a -3dB cutoff at ~6.83kHz. Place this filter in the L-R path, pre or post of the complimentary high pass filter shown on page 4 in the matrix diagram. If using a dynamic limiter it may be more optimal to place it just prior to the Bessel Low Pass Filter on the Q (L-R) channel. This filter is not needed for QuAM or ISB since they are additive modulation systems of 2 AM waves in quadrature producing sidebands ± the carrier reflective of their modulating signals.

Improved Tank
Low Pass Filtering



To the right is a more effective filtering scheme for the output tank circuits of the modulators in the exciter section. There are no emitter resistors and low pass filtering depends solely on the time constant created by r'e and C. During light levels of modulation which creates changes in current through the emitters thus changing r'e, this in turn creates incidental phase modulation, which will mostly cancel out with the balanced push-pull arrangement. Under full modulation cancellation is far from complete and incidental phase modulation may create significant phase distortion especially if the cutoff frequency is not high enough above the transmit frequency. To minimize this emitter degeneration resistors are used that are 10× of r'e reducing total change in resistance to 10% of the change of r'e. A cutoff frequency of 3 to 4 times the transmit frequency should produce a flat enough response in both amplitude and group delay. The table lists suitable capacitors for a range of transmit frequencies. The capacitor values chosen are not critical, just that the cutoff frequency is somewhere between 3 and 4 times the transmit frequency. It might be able to go as low as 2½ times the transmit frequency without any ill effects. Placing 1 or more ferrite beads on each of the resistors' leads will also help to reduce harmonics. A more robust solution is a balanced π filter preceeding the resistors but this is a more complex design. This filtering may have an additional stabilizing effect on overall circuit performance reducing any undesirable parasitic effects.

The exciter section is essentially broadband operating up into the lower portion of the SW band (<10mHz). It is these tank sections that help define the frequency of operation and reduce the out of band emissions.

C pf	-3dB@4f kHz	-3dB@3f kHz
910	390	530
820	440	590
750	480	640
680	530	710
620	580	780
560	640	860
510	710	940
470	770	1020
430	840	1120
390	930	1230
360	1000	1340
330	1100	1460
300	1200	1610
270	1340	1780
240	1500	2010
220	1640	2200
200	1800	2410

C-ISB™ conversion for the MC13028 **AM Stereo** decoder chip.