communications audio processor for reception

This article describes a system aimed at the special needs of Amateur Radio CW and voice signal reception. It's called the Comm Audio processor (CAP) and uses a combination of special voice-frequency filters, binaural synthesis, Tone-Tag modulation, and controlled antiphasic noise.

design considerations

Much attention was given to aspects of human engineering, signal environment, and operation in the CAP design. For example, most ham receivers provide plenty of band spread and a high-quality, calibrated tuning control, so selecting frequency in an add-on unit would be redundant. Accordingly, the CAP has been designed so that once everything is set up for a given voice or CW mode, most attention and operational control remains within your basic receiver. Aside from human engineering aspects, our very complex signal environment — both manmade and natural — contributed to the system design. In this respect the FCC's frequency allocation system heavily influenced all of the hardware because, through their regulation, there are at least three major radio signal environments in the high frequency spectrum: First, clear-channel frequencies are available for many industrial, public-broadcast, and governmental activities. Second, there are channelized frequency blocks, such as CB and many commercial and government allocations. Finally there is our type. We hams are required to fit our signals into frequency *bands* but aren't regimented into *channels* (so a few freedoms do remain!).

Because we have freedom of choice on any frequency within an allocated band, we're able to get five to ten times more useful communications functions in the ham phone bands than can be obtained in FCC channelized allocations of equal bandwidth, and up to fifty times or more in the CW bands. Certainly this signal density is such that QRM is one of the Q codes most popular; but the pressure has resulted, and will continue to result, in transmitter, receiver, antenna, and control design improvements. Clearly, our signal environment demands that we do not transmit on or receive unnecessary frequencies.

In addition to the human engineering and signal environmental aspects, there is the belief on my part that most hams, particularly DXers, are interested in enhancing weak-signal reception when there are strong signals nearby. The key to this is retaining an awareness of the noise floor and maintaining linearity in the signal path throughout an entire receiving system. A strong signal will suppress a weak signal either through the use of agc or by allowing compression of any stage. CAP was designed in consideration of these objectives.

In the system shown in **fig. 1**,* design effort has been directed toward an audio system for the Amateur brand of voice and CW communications. The unit contains the following key features, none of which are found to a refined degree in manufactured receivers:

1. Voice-shaped filter — thirteen poles to improve signal-to-noise ratio and reduce QRM by rejecting a broad spectrum between the first and second voice formants.

2. Binaural synthesis for both voice and CW.

3. Tone-Tag — the system that distinctively modulates any signal tuned to 750 ± 50 Hz, the binaural crossover frequency.

4. Nine-pole, 100-Hz bandwidth, stagger-tuned filter for super-steep skirts with a passive prefilter.

5. A continuously adjustable pink (soft) noise source — the how and why of this is covered in detail.

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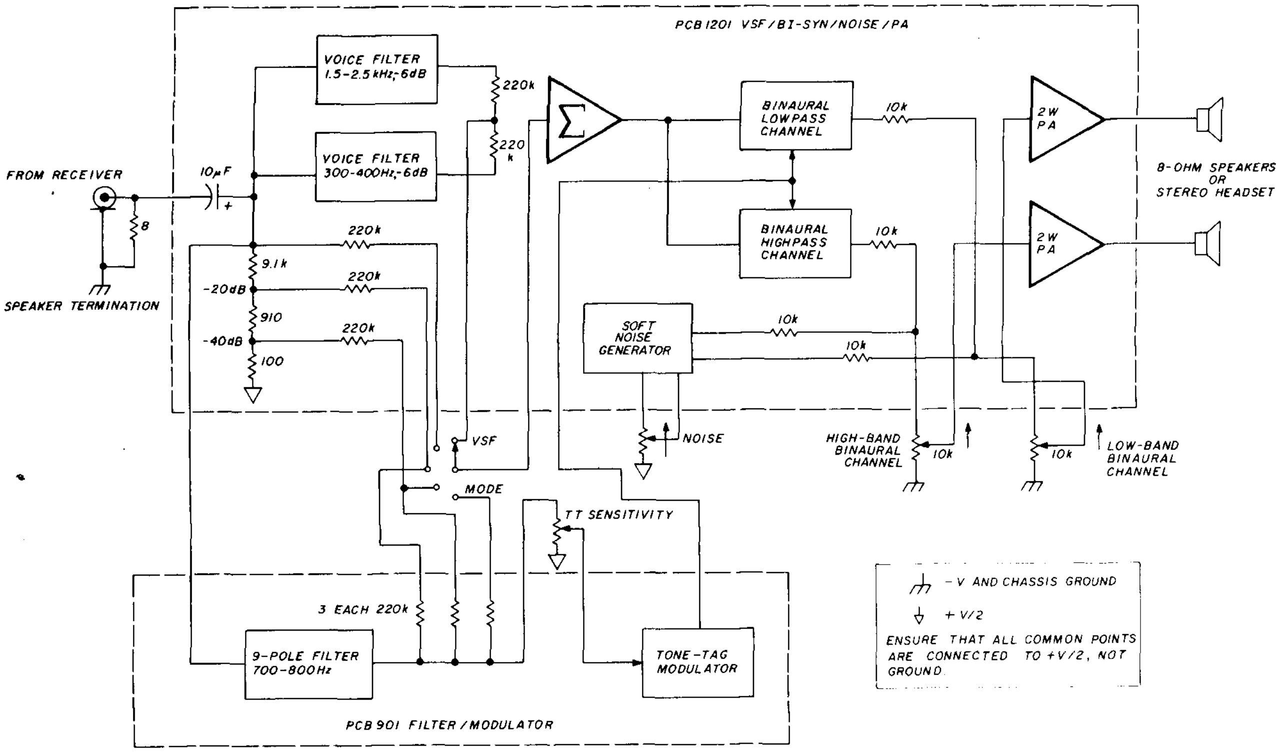


fig. 1. Comm Audio Processor.

Over the years many voice-manipulating systems have been proposed and developed, all aimed at making more effective use of the actual spectral distribution in the human voice. Most recently Dr. Richard W. Harris and J. F. Cleveland developed an excellent technique.[1] In this system, the voice spectrum is transformed following your transmitter's microphone output, as shown in **fig. 2**. The second and third voice formants (contained in the dashed area) are folded by a 3.1-kHz oscillator, mixer, and filters, which results in a band between about 300 and 1600 Hz for transmission.

An inverse system, again using a 3.1-kHz oscillator, mixer, and filters at the received audio end, unfolds (returns) second and third formants to their proper position (dashed). The amount of bandwidth actually involved is not reduced but merely rearranged.

Although frequency tolerances will have to be tightened, this system can nearly double the effective spectrum now used for voice-only, channelized communications. Since it can be applied to services such as CB, police, fire, and many other industrial, commercial, and governmental functions, frequency companding of this type could reduce the pressures to gobble up the ham bands.

The Comm Audio Processor design is based on the reality that those working in ham frequency bands up to 28 MHz are involved in nonchannelized communications, which is also heavily influenced by skip conditions.

We play billiards with the ionosphere, which reduces the effectivity of a frequency compandor. If you can't tell, because of skip, where other stations may be, how do you know where to place your bundles of voice energy?

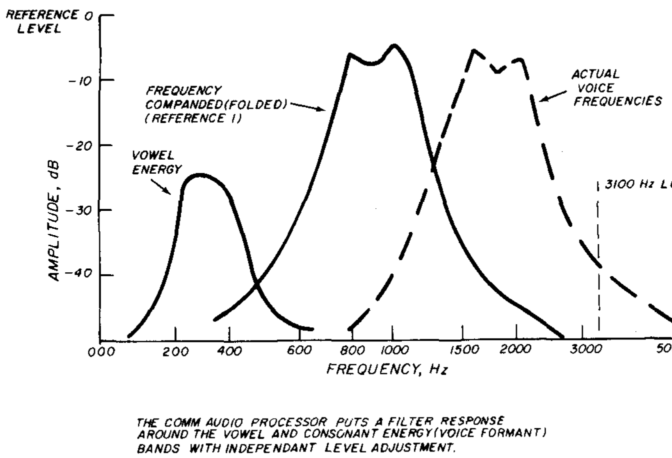


fig. 2. Folded voice-spectrum on approximated energy distribution. The Comm Audio Processor puts a filter response around the vowel and consonant energy (voice formant) bands with independent level adjustment.

*A drilled and etched PC-board set is available for $9.95 postpaid. Send a self-addressed, stamped envelope for a complete product listing to Hildreth Engineering, P.O. Box 6003, Sunnyvale, California 94088. Phone (408) 245-3279.

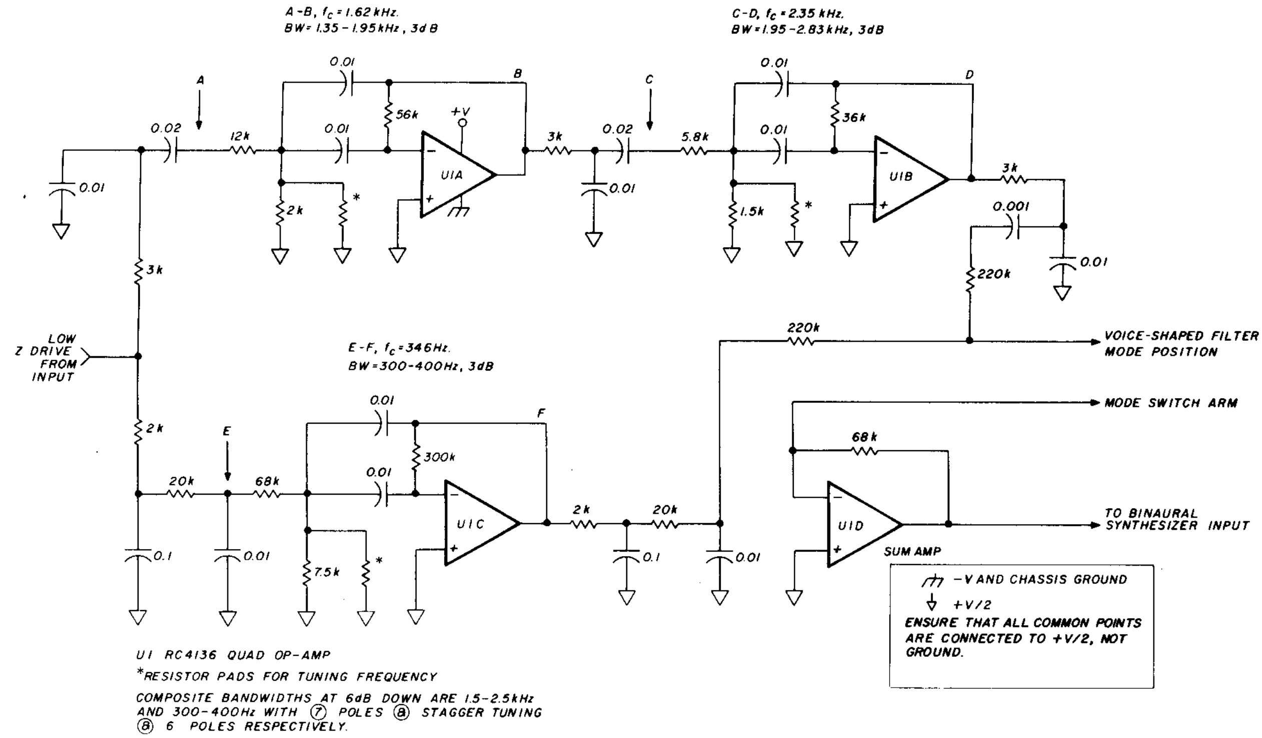


fig. 3. Voice-shaped filter and sum amplifier. Composite bandwidths at 6-dB down are 1.5-2.5 kHz and 300-400 Hz with seven poles and stagger tuning and six poles respectively.

Fig. 3 shows CAP's voice-shaped filter (VSF). In this case, it's not mandatory to roll transmitted voice energy off sharply above 2.5 kHz. It is desirable, however, because 2.5 kHz of bandwidth (or, more accurately, response from 1.5 to 2.5 kHz) will supply the basic needs for voice communications; the human voice does emit unnecessary energy to 10 kHz or so. Even though this energy can be filtered at the receiving end by someone listening to *you*, this energy will appear as lower frequencies — thus unfilterable — for other stations up to 8 or 9 kHz away. The design shown in **fig. 3** can also be used in your microphone circuit. It would make other hams in our crowded bands very happy.

binaural synthesizer

To synthesize a binaural sound environment, the audio output passband from any receiver is divided (with reasonably sharp filters) into two parts. Frequencies below 750 Hz are fed to one speaker and frequencies above 750 Hz are fed to another. The speakers are located as you would place them for stereo listening. Speaker locations and their resulting stereo amplitude potentials are shown in **fig. 4**. A maximum differential of 7 dB (only slightly more than one S unit of signal strength) holds through most of a typical voice communications bandwidth from about 500 Hz to 3 kHz.[2] Below 500 Hz the human stereo potential diminishes; below 300 Hz it's gone.

Since the 7-dB differential occurs because of the "sound shadow" presented by your head, you may leap to the thought of a huge improvement by using stereo headsets. But wait. It won't happen. You can get a better amplitude differential with headsets, but that's not the whole story. Our brain processes a sound wave, or many, in terms of much more than

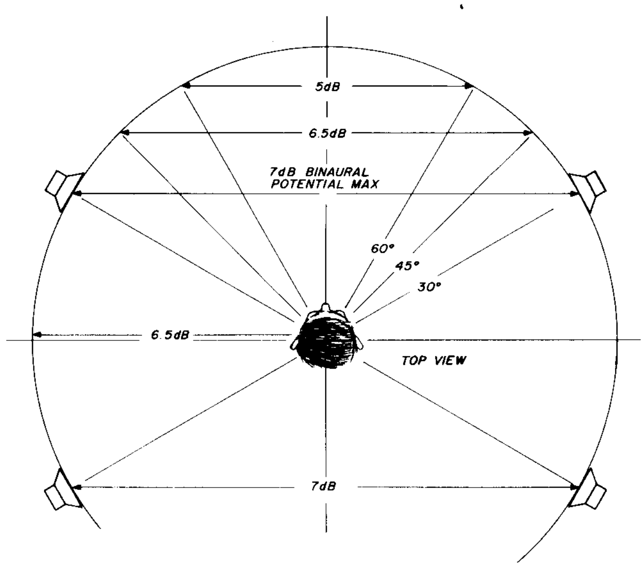


fig. 4. Locating speakers for best binaural amplitude separation. The maximum effective binaural differential based on intensity is obtained when speakers are located forward of, or behind, an imaginary line through your ears by about 30 degrees.

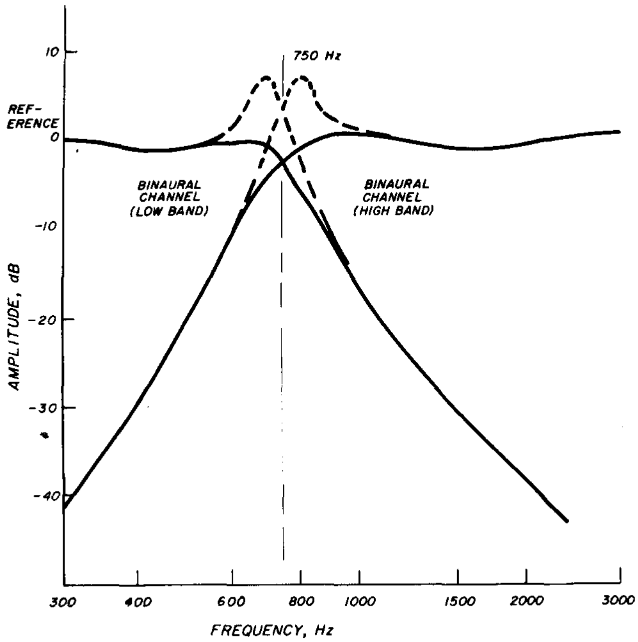


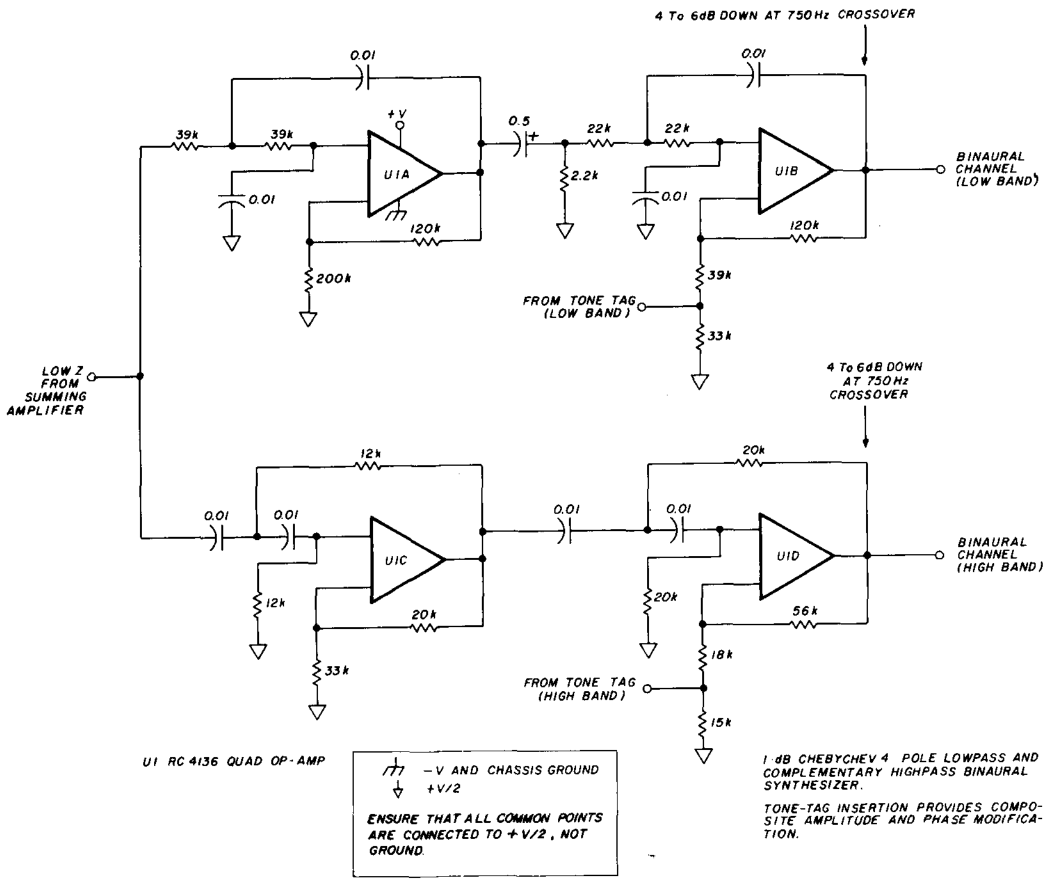
fig. 5 (above). Binaural response. Solid lines show approximate 1-dB Chebychev response. Tone-Tag changes response to dashed lines 100 times/second.

fig. 6 (below). 1-dB Chebychev four-pole lowpass and complementary highpass binaural synthesizer. Tone-Tag insertion provides composite amplitude and phase modulation.

just relative amplitude. There are also time-of-arrival and phase variation phenomena. Headsets wipe these out in our system. When using speakers with the binaural synthesis method used here, sounds originate to the right or left in space, just as in nature. However, when a CW beat note is tuned to the crossover frequency of 750 Hz, the brain gets equal information in terms of amplitude and time-of-arrival from both left and right azimuths, resulting in the impression of "surround sound." Headsets work well, of course, but you just won't get the improvement you may expect.

In the CAP, 1-dB Chebychev four-pole lowpass and complementary highpass filters are used, with a frequency crossover (equal energy from both sides) at 750 Hz. The voltage control voltage source (VCVS) active filter form enables the insertion of Tone-Tag modulation. **Fig. 5** shows a representative response (solid lines). Tone-Tag modulation alternates the response between the solid and dashed lines, but only as driven by a signal tuned to 750 ±50 Hz through a 9-pole filter and at a nominal 100-Hz rate. **Fig. 6** is a complete schematic.

In the first article describing a binaural synthesizer for CW reception[3] a simple cascade of 2-pole filters was used. To improve the crossover slope without adding more poles and to improve Tone-Tag inser-



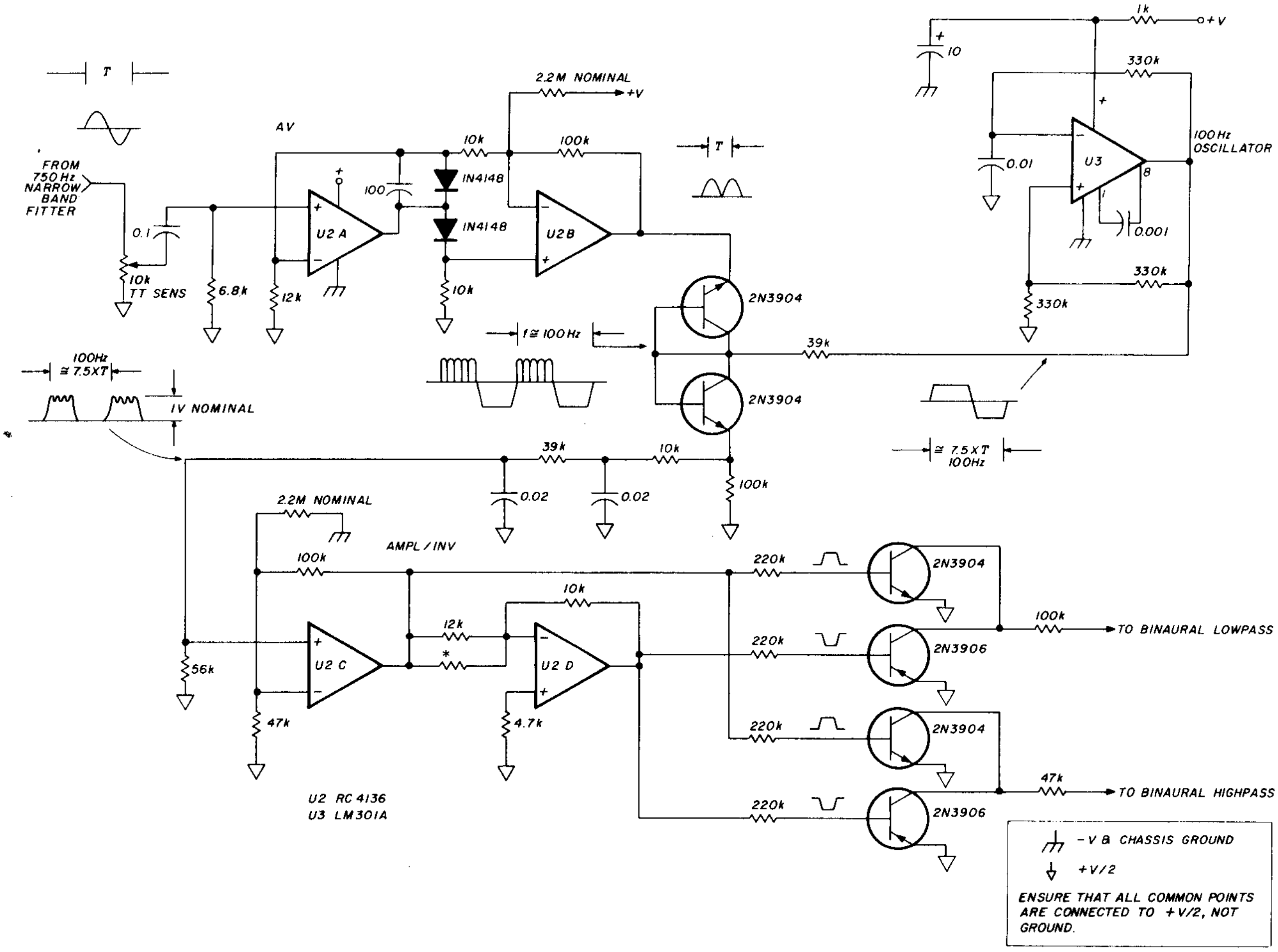


fig. 7. Tone-Tag modulator. Resistor marked with an asterisk at U2D — input is adjusted for minimum tone output from the binaural lowpass when at least 100 mW of 750 Hz signal is at the input above and the signal input line of the binaural section is grounded to signal ground (V/2). Resistor nominal value is 57k.

tion, the design has been changed to the Chebychev type. (Burr-Brown[4] supplies an excellent set of tables for Butterworth, Bessel, and Chebychev designs.) To improve antiphasic response enabled by the binaural system, the low- and high-pass systems are adjusted to be about 4-6 dB down at the 750-Hz crossover frequency.

Tone-Tag

The Tone-Tag system first used (model 1100) was based on simplicity.[5] Now, however, refinement has set in. Our model 1500 changed the modulator from diodes to a balanced bipolar technique to improve weak-signal performance in a noisy background. And CAP adds still further, although minor, improvement. **Fig. 7** shows the heart of our current Tone-Tag modulator design.

The absolute value (AV) circuit — also called a precision rectifier — transforms any incoming signal to a 750 ±50 Hz rectified, unfiltered positive output with a gain of ten. When no signal is present, positive excursions of a nominal 100-Hz oscillator are clamped to a nominal +0.5 volt by a diode-connected transistor working against the low-impedance zero output of the AV op-amp. A second diode-connected transistor, turned around, subtracts most of the +0.5 volt signal and rejects the negative half cycles of the 100-Hz oscillator signal. When an input signal appears, positive excursions appear at the AV output, and the diode-connected transistor releases its clamp on the 100-Hz oscillator, which results in the waveforms shown.

Since more current normally flows in the first diode than in the second, the second diode voltage doesn't quite offset that of the first. The resistor from − of U2B to +V provides a small compensating offset at the U2B output, resulting in a near zero voltage at U2C input.

When no 750-Hz signal is present, U2C provides a nominal gain and a small offset, which sets the modulator transistors to near conduction. U2D inverts U2C output to provide a balanced drive to the com-

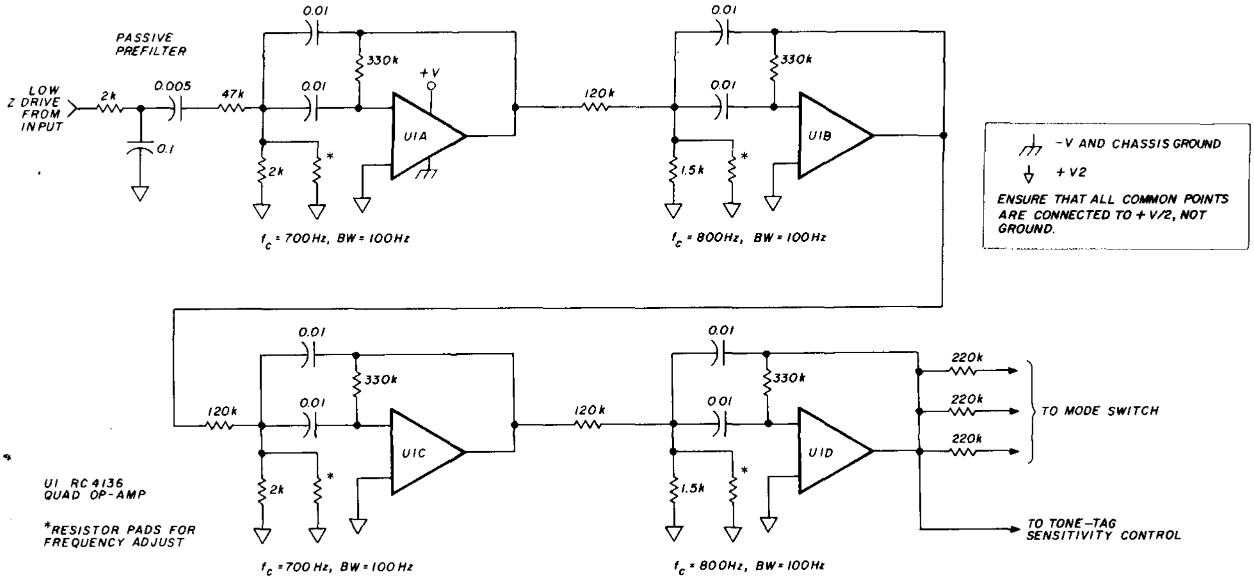


fig. 8. Nine-pole, 100-Hz bandwidth stagger-tuned, 750-Hz filter. Passive prefilter provides pole at 750 Hz with phase-corrective element.

plementary modulator transistor switches. This action is precisely set with the padding resistor marked with an asterisk in **fig. 7**. Resistor-buffered switch outputs are fed to the binaural synthesizer (**fig. 6**).

9-pole narrowband filter

A narrowband filter centered at 750 Hz, the binaural crossover frequency, is required to drive the Tone-Tag AV input circuit. Since a filter of this type is popular in its own right, a very steep-skirted, stagger-tuned device was used. A passive prefilter is used in addition to quality RC4136 op-amps to mitigate the effects of transient intermodulation distortion that may be induced by impulse noise or transients (key clicks). The design also minimizes small-signal compression by strong nearby signals acting on the first, and most susceptible, filter stage. The composite is the 9-pole filter shown in **fig. 8**.

good noise — bad noise

White noise or pink noise (shaped) is often added to communications circuits to mask distracting forms of low-level interference. In most of its applications, this form of noise is soft and fluffy, even pleasant. Pink noise generators are even found in such diverse applications as sleep aids and pain suppressors for some dental operations. When properly used, this is "good" noise.

Then there's that other kind, the abrasive, or "bad," noise that ranges between 10 and 40 dB thick depending on frequency, time of day or year, and your location[6] (**fig. 9**). This noise, available at your antenna terminals, is the sum of many sources. Added to nature's atmospheric electrical disturbances, manmade electrical perturbations mix in to generate the din. Electrical circuits being made or broken create impulse-like radiation that chirps through the spectrum. Auto ignition, power leak . . . the list can go on and on.

Thermal noise — and, to a large extent, galactic noise — can be thought of as a statistical average of a very large number of low-energy electrical perturbations. At the same time, manmade and atmospheric noise comes from a much smaller number of relatively high-energy events. When heard, these noise types influence our receivers and our nervous systems very differently. The sum of the two noise

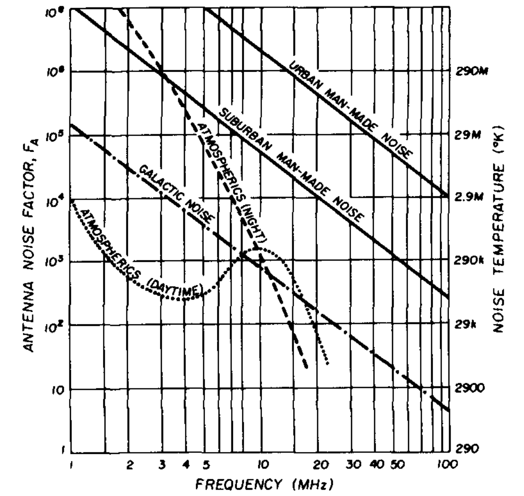


fig. 9. Receiver sensitivity is limited by the external available noise power, which varies with frequency. For a quiet, rural location, galactic noise is the limiting factor down to about 18 MHz, and atmospheric noise dominates below 18 MHz.

types could be described as a foam mattress (thermal noise) perforated by a family of random spikes (impulse noise and signals).

In practice, something approximating "good" noise may be heard by simply removing the antenna from your receiver and listening with an audio bandwidth of 2-3 kHz or more. The other stuff is what you hear when your antenna is reconnected and you tune with high gain to a spot where no signals are present (it's assumed your receiver is not noise-figure limited). Now, with antenna-received noise present, connect your receiver output to one of those ubiquitous

attempt to get most of the benefits of a multipole narrow-band filter without listening "through" it. However to use Tone-Tag under high-sensitivity levels, coincidental to feeding the binaural synthesizer signal input port with the output of a narrowband filter (not the original intent), I found that the filter tinkling sound along with the desired signal was modulated — thus boosted — by Tone-Tag. Although the level was not too high, it was objectionable.

On a monaural basis I decided to try adding soft noise to mask those elements of noise that appear at

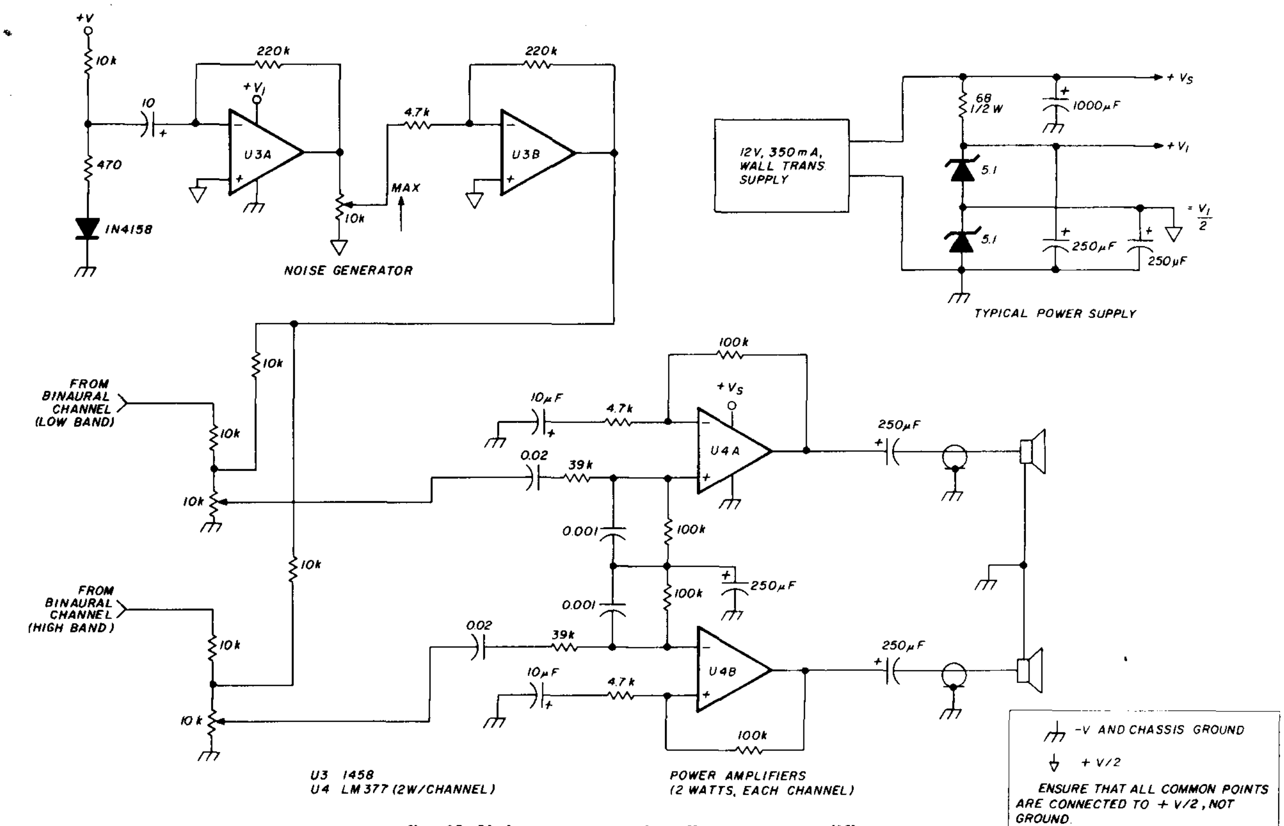


fig. 10. Noise source and audio power amplifiers.

multipole active audio filters. With the rf gain "up," you'll hear that tinkling roar that drives so many of us to look for a better solution, for, even though the little devices do a good job in some respects, they can't filter those noise components that sound like a signal itself. In addition, sporadic, impulse-like noise tweeks the filter to produce sounds approximating what you get if you rapidly strum the strings of a violin while damping them with the other hand. It doesn't ring long, to be sure, but the sounds produced are much too like the code structures in a weak CW signal.

listening experiments

In the Tone-Tag system first published[5], I made an a narrow-band filter output with the desired signal. The resulting signal sounded reasonably smooth; but weak signals suffered slightly under the noise load. The next step was to add the soft noise in-phase to binaural audio channels while the signal plus residual filter noise irritants were fed to the two channels in phase opposition, or approximately so.

It's well known that copying a signal in noise with a binaural antiphasic noise combination has a 15-20 dB advantage over the monaural case,[7] but the question was this: If a weak signal can be clearly heard when added to white noise in this way, will the signal's filtered residual noise components be masked by the white noise? Results were much better than expected!

I've spent many hours listening to compare the readability of a weak signal with or without soft noise added. Clearly it's more pleasing to listen to a signal plus soft noise, and that was expected. But readability, in many cases, was *improved*, which was not expected to the degree I found. Actually, I'd hoped to minimize signal readability degradation with the addition of antiphasic binaural noise. It now appears that when a weak signal and abrasive (impulse) noise pass through a narrowband filter, noise elements are transformed into sounds that are so much like the signal's coded structure that they compete with it. The result is a less readable signal. But with noise added in the right way and amount, readability is actually improved.

In no case during these listening tests could I detect signals with the white noise added that could not be detected without the added noise. As expected there was no basic improvement in signal-to-noise ratio. But a signal could often be copied with the added noise where it could be heard but not copied when the noise was removed. To date I've found no case where the addition of soft noise has made a signal less readable than without it. Clearly, of course, added noise has usually been at a level just necessary to mask the undesirable noise elements coming out of a narrowband filter. Happily, added noise benefits the critical application of the Tone-Tag referred to above as well!

Since the existing binaural synthesizer already supplies the desired antiphasic* condition, summing inphase noise at the power-amplifier input adds very little to the cost in the CAP design — and one more control knob. **Fig. 10** shows the noise generator and how it's added at the junction of the binaural synthesizer and power amplifiers. Although most emphasis has been on the judicious addition of soft noise in CW reception, the feature can also be used to some advantage in the reception of voice signals.

operating with CAP

The main mode control switch enables the selection of the shaped-voice filter, a nominal 2.5-kHz flat bandwidth for either voice or CW or three CW filter positions. Any selected mode is fed into the binaural synthesizer filters. **Fig. 11** illustrates. Independent power output controls are provided for each binaural synthesizer channel, which allows compensation for different speaker or headset efficiencies or for special effects. The two remaining controls include Tone-Tag sensitivity and amplitude control for the soft-noise source.

When the voice-shaped filter is selected, tuning your SSB receiver will be easier — more like tuning an a-m signal than when the usual 2.5-kHz flat bandpass filter is used. When the frequency band from about 400-1500 Hz is deeply rejected, a slight-to-moderately mistuned station will not produce much output energy in this band, thus it more quickly disappears as you tune. In addition, under critical conditions often found on 14 MHz, for example, the binaural low-band from 300-400 Hz may be reduced to or near zero with the independent channel output control. Under this condition, the 300-400 Hz low-band segment is still available at approximately 30 dB down in the high-band binaural channel along with its normal 1.5-2.5 kHz response. The noise bandwidth in this super-sharp condition is only about 1 kHz. Clearly recognizable voice is available although it will be nearly devoid of character. Bringing *up* the binaural low-band control will progressively enable individual signal recognition. Some signal conditions allow boosting the low and reducing highs for 100-Hz bandwidth. One click on the mode control and you have the prevalent 2.5-kHz nominal, flat-voice bandwidth, but in binaural.

The 2.5-kHz binaural position is also useful for CW in general listening. When Tone-Tag is used with this bandwidth, the modulation effect is mild because of competition with a relatively wide noise bandwidth. But it's still adequate to provide excellent selectivity through the significant tone quality difference relative to other signals in the bandwidth. In addition, of course, signals not in the modulated pass-band appear on the right or left specifically, while the tagged signal is heard in "surround sound."

On the next click of the mode control, everything in the 2.5-kHz bandwidth — *except* the 100-Hz band centered at 750 Hz — drops 20 dB. All basic conditions remain the same as above, but the Tone-Tag modulation is now more prevalent, and nontagged signals and noise drop a little more than 3 S units. If this isn't enough, the next position moves the floor down 40 dB — nearly 7 S units.

Now, if you're listening to a signal at about S-5 or so and a signal of about the same strength appears at around 400 or 1100 Hz, for example, you are alerted to the fact that the signal is very strong — nearly 16 dB over S-9 and probably suppressing your S-5 signal at his keying rate (if the signal has key clicks, it will be placing energy in a wide band as well). This alerts you to roll back your i-f (often called rf) gain control to avoid probable compression in your receiver's final down-converter (product detector), assuming your first mixer is not also being overloaded. If overloading is present in the first mixer, insertion of some attenuation between it and your antenna is indicated as well. Actually, however, so much attention has been given to the first mixer over the last

*The term antiphasic describes the case where a signal is fed out of phase to your two ears and noise is supplied in phase. The inverse is also true. The usual case (as in monaural reception) is called homosphasic.

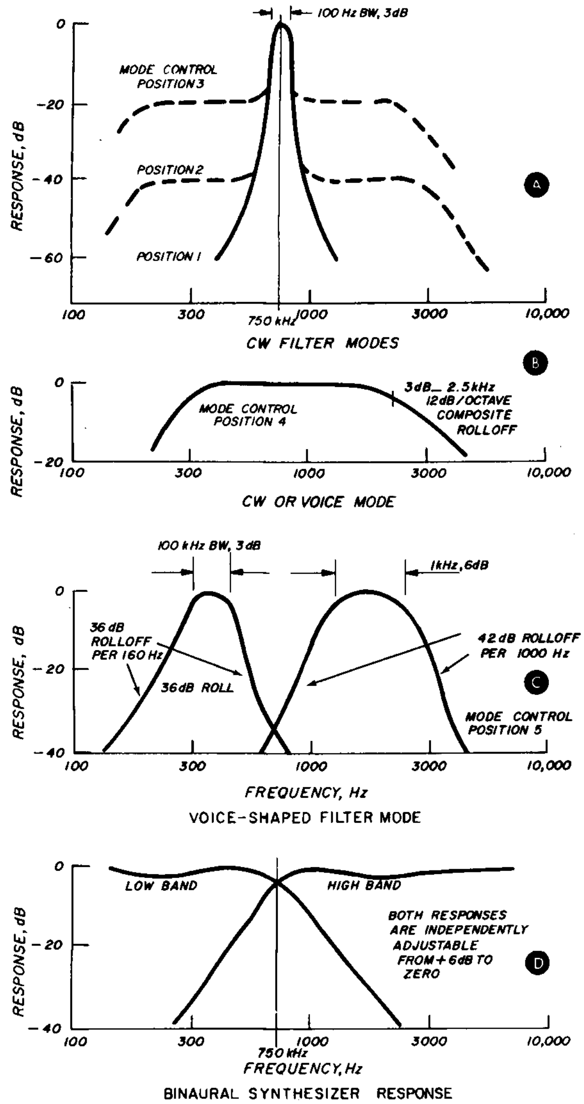


fig. 11. Frequency response. Responses A through C are fed through the response of D to become a binaural function.

two decades that the problem has shifted to the following converter(s) in our current receivers. Corrective action varies somewhat in many receiver designs, but the same general action is taken regardless of whether you are using agc or not.

The last mode select position supplies the basic 9-pole narrowband filter output through the center of the binaural band with no added binaural floor. To make good use of this position usually requires that your receiver have a super-good dynamic range, at least through its product detector.

Tone-Tag may be used or not as you choose in any of the CW modes. In general use, its sensitivity control is advanced to a point of a few degrees above where a given signal is modulated when tuned to the 750-Hz frequency. If no signal is present, increase Tone-Tag sensitivity until background noise is modulated, then back down until the modulated noise is just barely noticed. The best level is also dependent on your receiver audio-output level. Therefore it's possible, after a little experimentation, to generally set the Tone-Tag control on CAP then adjust your receiver gain to the tag level for any signal being received. Tone-Tag's modulator switching uses a nonabrupt design to avoid critical operation and to mitigate the effect of fade generally present on DX signals. In most cases Tone-Tag will make solid those ghostly multi-hop DX signals.

Finally there's the noise control. At the beginning you may nearly wear out this pot. Since the idea of adding noise seems so contradictory, you may constantly test the effect by repeatedly running the noise level up and down when listening to a weak signal. In general, however, a good starting point is found by increasing the soft noise to a point where it's just barely noticeable when the mode control is in the general 2.5-kHz position. The control is left there for all modes. You'll note that the added soft noise power is fixed relative to your receiver gain setting. Through this feature you may vary the ratio of added soft noise to relatively abrasive antenna-derived noise by receiver gain variations.

The binaural function provides a spatial sound environment in all modes without the need for adjustment. In addition, however, with proper physical arrangement and some practice, it can also serve as a tuning aid. For example, in using CAP with a Kenwood TS820S the low-band speaker or headset is placed on the right and the high-band device on the left. In this way, if a signal is heard on the left you know that tuning to the right (clockwise) will move the signal in that direction spatially. This is true with the TS820S because its conversion scheme results in an audio tone that moves from high to low as you turn the knob clockwise. Other receivers will be like this or exactly the reverse. The same may be true when going from band to band (SSB filtering for both voice and CW is assumed). The same tuning directional benefits are also present for SSB voice, although the action is more subtle.

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