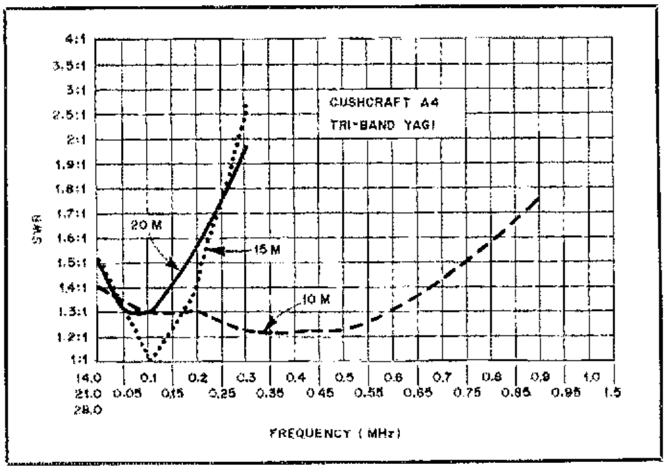
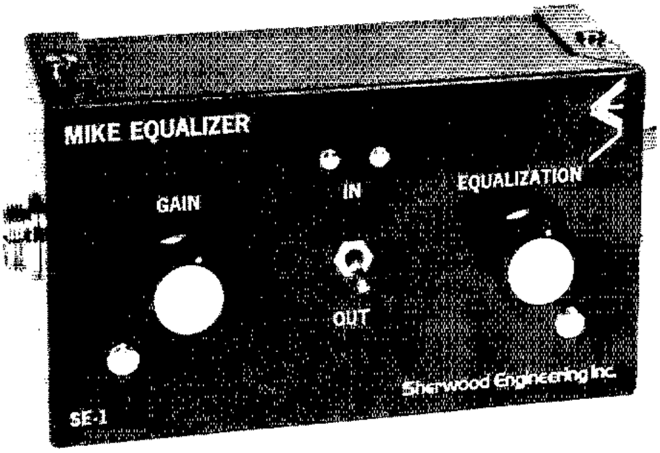


Fig. 2 — SWR curve of the Cushcraft A4 tribander.



jacketed Hardline. Overall transmission-line length is 60 feet. A Bird Thruline wattmeter was connected between the lower end of the feed line and an FT-101ZD signal source. The A4 was adjusted for optimum operation in the cw portions of the three bands, in accordance with the dimensions given in the instruction sheet. The resultant SWR curves are shown in Fig. 2.

$$\text{VSWR} = \frac{1 + \sqrt{\frac{P2}{P1}}}{1 - \sqrt{\frac{P2}{P1}}} \qquad \text{(Eq. 1)}$$

where P1 is the forward power in watts and P2 is the reflected power in watts.

I find it necessary to use a Transmatch to disguise the SWR when operating in the phone bands (which I seldom do). Operation by that means is satisfactory. Those wishing to strike a compromise between phone and cw operation may wish to adjust the antenna for midband resonance. A chart is given in the instruction sheets to provide dimensions for three segments of each band — phone, center and cw.

On-the-air performance has been excellent, with suitable front-to-back and front-to-side ratios evident. I have had no difficulty working DX worldwide, using 100 W and 1000 W of dc input power to the transmitter PA. My batting average for breaking pileups on cw has also been satisfactory.

The A4 has endured several severe wind storms (up to 70 mph) and one ice storm without mishap. Based on my experiences with the antenna during the past year, I would give it a top rating for a small, moderately priced beam antenna. Those wanting four-band operation with the A3 and A4 Cushcraft antennas may purchase the 40-meter conversion kit and install it in the driven element. The A4 price class is $330; A744 (7- or 10-MHz) adapter kit, $90; A4SK stainless-steel hardware kit, $55. Manufacturer: Cushcraft Corporation, P.O. Box 4680, Manchester, NH 03108. — *Doug DeMaw, W1FB*

SHERWOOD ENGINEERING SE-1 MICROPHONE EQUALIZER/ PREPROCESSOR

☐ Recently, amateurs have begun to take greater notice of their ssb signal fidelity. Traditionally, a "speech processor" has been a device that alters the signal peak-to-average amplitude ratio. Such systems invariably alter or degrade the audio characteristics of the voice — usually causing it to sound "bassy" and "rough."

Theoretically, the audio response of a microphone amplifier circuit in a communications phone transmitter should be flat across the voice band (300 to 3000 Hz). Often, this is not the case. The microphone response should be flat across this band, also. Again, frequently this is not the case. As a result, we have come to expect ssb signals to sound "Donald Duck-y."

Recording studios use equalizers to correct for distortion introduced by electrical and acoustical factors. An equalizer has controls that permit the operator to adjust the phase and level of several segments of the audio spectrum. The Sherwood SE-1 represents one attempt to introduce this kind of processing into Amateur Radio transmitters. It won't increase the average power level of your signal, but it may make it easier for operators on the receiving end to understand what you are saying.

The SE-1 is housed in a heavy, steel black box, measuring 3 × 5-1/4 × 2 inches (HWD).[3] Internally, the device is a "black box" also. A single transistor, one IC, a toroid and a few other components are mounted on a glass-epoxy pc board. All markings have been removed from the active devices. Sherwood does not supply a diagram of the circuit, but does provide minimal circuit description. In fact, the only paperwork supplied with the SE-1 are installation and operating instructions on two sides of one sheet of paper. The installation instructions are complete and should be adequate for anyone who is moderately comfortable when using a soldering iron and pliers.

A toggle switch places the circuit in line and applies power (9-V transistor-radio battery). Two controls, GAIN and EQUALIZATION, can be adjusted from the front panel. Adjusting GAIN simply changes the output signal level while changing the EQUALIZATION setting alters the tonal components of the signal. Advancing the control clockwise *appears* to enhance the higher-frequency components. Unfortunately, the exact functioning of the control is not specified in the literature.

I called Bob Sherwood and asked him about the EQUALIZATION control. Bob told me it controls the "tilt" of a processing stage. He went on to tell me what he meant by "tilt." With the control fully counterclockwise, a 2800-Hz tone through the device will have a 2-dB advantage over a 300-Hz tone. As the control is rotated to the fully clockwise position, the advantage increases from 2 to 20 dB. This description matches the subjective appraisal I received from operators on the air.

Bob stated that the microphone preamplifier stage consists of a single FET and a gain control. The high impedance of the FET permits the circuit to be used with any microphone, without danger of the circuit loading the microphone. Beyond that, Bob was reluctant to discuss the circuit.

On-the-air reports indicated that the SE-1, *properly adjusted,* made my ssb signal sound more natural — more like the audio extracted from an fm circuit. I believe it made it easier for other operators to understand what I was saying. If you are concerned about the quality of your signal and if you want to be understood as well as be heard, then you should give the SE-1 serious consideration. The SE-1 is available from Sherwood Engineering Inc., 1268 South Ogden St., Denver, CO 80210. Price class: $100. — *Peter O'Dell, KB1N*

DAIWA AF-606K ACTIVE AUDIO FILTER

☐ Audio filters are in abundant supply these days. They come in many sizes, shapes and price classes, but the Daiwa unit has a distinctive look — akin to that which characterized post WWII military gear. I was impressed with the clean, snappy appearance of the filter when I extracted it from the box. The panel appears to be black anodized aluminum, and the lettering at each control is off-white. A dark gray case provides a two-tone contrast.

But, appearance is not the primary consideration when buying a new piece of equipment for the shack. How does it "play"? That's the question asked by a smart buyer, and rightly so! Bells and whistles (if I may use the vernacular) seem to have a biasing effect on today's purchaser of new apparatus. But some of the fancy gee-gaws being offered could just as easily be omitted in the interest of keeping the unit cost within the reach of the common man or woman. The AF-606K has one feature that might be classed as a frill (more on that later), but it otherwise is a pretty basic audio

[3]mm = in. × 25.4

filter with variable bandwidth and a notch function.

The Daiwa unit does not have a built-in power supply. A low-current, 12-V external power supply is required for operation. *Beware!* The outer ring of the 12-V jack is the positive one. The center pin is for the negative or ground lead of the supply. This is not the U.S. convention, so don't let habit get you in trouble when you hook up your unit.

Installation requires a patch cord from the phone jack of your receiver or transceiver to the input phono connector on the rear of the AF-606K. A built-in speaker permits monitoring the filter output. Alternatively, the operator can attach an external speaker, or may elect to connect headphones by means of a front-panel jack. I found it best to use phones, since the speaker function did not produce room-volume audio at a level that was comfortable during weak-signal reception. Attempts to increase the output level by turning up the receiver audio gain resulted in distortion from overdriving the audio filter (a normal experience with outboard audio filters when the maximum tolerable excitation limit is reached).

Five controls are located on the front panel of the unit. Left to right are NOTCH, PLL, BAND PASS, MODE and POWER. The NOTCH control is variable from approximately 500 to 2500 Hz, with some overrun at each end of the control. I measured the notch depth (at 700 Hz) as 33 dB. I'll discuss the PLL control later on.

The BAND-PASS control is used to peak the audio-filter response for the cw pitch the operator prefers (500 to 700 Hz in my case). A band-pass response is provided in this unit, which yields audio roll-off above and below the desired frequency. The pass band is variable from approximately 400 to 1200 Hz, with some extra range at each end of the control.

MODE selection is accommodated by the near-right control. It enables the user to choose a notch condition, three ssb bandwidths (1.5, 2.0 and 2.5 kHz), three cw widths (80, 110 and 140 Hz) and PLL. The POWER ON-OFF switch is at the far right on the panel. Directly below it is the PHONE jack.

Now comes the "bell" or "whistle," whichever word you may prefer. The PLL function enables the operator to tune in a cw signal and listen to it via a keyed tone that is generated within the AF-606K. In effect, the cw signal from the receiver is detected, then routed to a control circuit, which actuates a tone generator. An LED on the front panel of the audio filter illuminates when the PLL frequency control is set to the pitch of the cw signal, as heard when the filter is turned off. Under this condition the PLL is considered in the "lock" mode. The purpose of the PLL function is to eliminate QRM and band noise. Effectively, all you will hear is a single cw note coming from the audio filter.

Various schemes of this type have been contrived and tested for a number of years. None of them proved to be spectacular. The major limitations are that very weak signals do not trigger the tone oscillator in a reliable manner, which leaves gaps in the cw message. Also, noise pulses will key the tone generator, causing false blips, "stuttering" and incoherence. These problems were noted while testing the Daiwa filter. When a strong cw signal was used to lock the PLL, I noticed the effect of excessive "weighting" on the cw characters. There was also a clicky characteristic to the tone-generated cw note. For the most part, coherence was far superior without the PLL function in use. If the buyer does not wish to have the PLL feature, he or she can purchase the model AF-406K, which is minus the tone decoder.

Filter performance is otherwise excellent. I found no evidence of ringing, and audio output from the AF-606K was very clean within the normal listening range while using headphones. Certainly, the filter did a fine job of "laundering" the receiver output with respect to reducing QRM, annoying heterodynes and receiver wide-band noise. Weak signals were "lifted" nicely out of the noise, providing Q5 copy when copy was not possible without the filter in the line.

This filter and others with similar performance characteristics can spell the difference between success and failure when copying weak cw signals, such as one encounters on 160, 80 and 2 meters, where noise is a universal foe. Sideband operators will find that a good audio filter will reduce adjacent-frequency splatter and rumble. In many instances, the audio filter will give the same effect as a speech processor when it is actuated by the person you are listening to.

Dimensions are 6 × 6 × 2-1/2 inches.[4]

[4]mm = in. × 25.4

Price class is $121. Distributed by MCM Communications, 858 E. Congress Park Dr., Centerville, OH 45459, tel. 513-434-0031. — *Doug DeMaw, W1FB*

HAMLOG/APPLECODER

☐ I have some good news, some bad news and some more good news. The good news is that HAMLOG, a log keeping and maintenance program for the Apple II® computer, does everything it says it will do. The bad news is that there is so much that it will not do, frustratingly so, at times. The other good news is that those of you who feel compelled to use a $47 program and $2000 worth of computer equipment to do what a pen and a box of 5 × 7 index cards can do nearly as well won't be disappointed.[5]

HAMLOG fails to take sufficient advantage of the Apple II system power. Users are constantly restricted by the program limitations. That is, they must adapt to the program needs, rather than the other way around. Those who have seen a demonstration of or used such programs as Visicalc or Visidex know how "user friendly" a program can be. HAMLOG is a casual acquaintance at best, and a relatively expensive one at that.

Using HAMLOG

Initialization. When first using HAMLOG, you must initialize the system with your call sign, various modes that you expect to use, and your station-setup data. The latter demonstrates the limitations of HAMLOG rather quickly. Station-setup data is limited to seven characters. For example, the author suggests that a station consisting of a "standard" transceiver, wire antenna and linear amplifier be designated QRO-W-L. You can have several such station setups, so your log can show which particular configuration you were using when you worked a particular station. The brevity of a seven-character designation does have its advantages when it comes to displaying or printing out log data (this brings up another defiency of HAMLOG, which will be discussed later). Nevertheless, the restriction is too severe. Users, after all, can always choose brevity; it need not be forced upon them.

Log Entry: After initializing the program, you can run it. The program is menu-driven. One item on the main menu is "log entry," which is pretty straightforward, although it is definitely geared toward casual operating and not contesting. You enter one log entry item at a time: call, name, band, etc. However, as you enter the items, the previously entered items "scroll up" and quickly disappear from the screen. By the time you've entered the other station's signal report, the call and name have disappeared, and you cannot see them until you finish entering the entire log entry and select the "display log entry" item from the main menu. This is a major inconvenience. As it takes some time to step through the log entry menu, I found myself frequently writing down the other station's call, name, etc., so I wouldn't forget them in case he turned the QSO back to me before the log entry was completed (initial exchanges can be rather quick). This pretty much defeats the purpose of using a computer.

[5]H. Smith, "A Speedy QSO File," *CQ*, June 1962, p. 62.