Product Review & Short Takes Columns from *QST* Magazine

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Ten-Tec Model 416 Titan II HF Amplifier

Reviewed by Dave Patton, NT1N

When Ten-Tec discontinued production of its Model 425 Titan amplifier in January 1997, it wasn't clear whether or not the well-known American amateur equipment manufacturer would replace it with another legal limit amp. The original Titan employed a pair of 3CX800A7 triodes in a desk-top RF deck (the power supply was in a separate enclosure) to help it secure its place in the high power HF world. That amp easily delivered 1500 W of RF output power on all of the HF bands with just 50 to 70 W of drive power.

The only concern brought up when we reviewed the previous version of the Titan (see "Product Review," *QST*, Apr 1986) was that the input drive power needed to be carefully monitored and controlled so as not to overdrive—and possibly damage—the tubes. This was not surprising, and the manual included just that precaution. With a couple of very popular 200-W output transceivers on the market at that time, the Titan was definitely subject to unintentional abuse. Would the new Titan II require the user to be as cautious?

The Titan II uses a single Svetlana 4CX1600B—a beefy, Russian-made tetrode—to easily deliver 1500 W of RF power output at input drive levels similar to those required by the Titan's pair of 3CX800A7s. Compared with the cost of the pair of 3CX800A7s (around $500 in 1986 and closer to $900 now), the 4CX1600B certainly offers some savings (it's priced in the $375 range) for both Ten-Tec and its customers.

Out of the Boxes and Into Your Station

The Titan II is shipped in two cartons, one containing the transformer—a 46-lb standard EI lamination unit that's custom-built for Ten-Tec—and the other the amplifier itself. Ten-Tec ships the unit with 11 of the 19 screws that secure the top cover removed. This makes the installation process a little easier. The transformer's mounting bolts come in a packing kit along with the remainder of the cabinet screws (and it is these bolts that should be used to install the transformer in the chassis, not the bolts that secure the transformer to its shipping pallet).

Transformer installation is straightforward. Once assembled, this is a big, heavy amplifier—84 lbs—so it does require intelligent handling techniques to safely wrestle it into its final operating position.

For purchasers who send in a copy of their General Class or above amateur license, Ten-Tec will supply a separate 10/12-meter band "input matching board" that replaces a similar circuit board that's factory installed in the unit. The existing board is unplugged and the new board is substituted. A mechanical stop must also be removed from the bandswitch that prevents it from turning into the 10- and 12-meter positions. This operation involves removing the bottom cover of the amplifier to change out the boards, and then replacing it and removing the top cover to gain access to the bandswitch. It's a relatively simple operation, but there are more than 30 screws used to secure the two covers!

At this point, all that's needed to be ready for "fire testing" is to attach a 20 A/240 V plug to the end of the ac line cord. (This amplifier is not designed for operation from 120 V ac.)

Bottom Line

The Titan II amplifier harnesses the capabilities of a single beefy Russian tetrode to effortlessly deliver up to 1500 W of HF RF power.

The instructions in the *Operator's Manual* are easy to follow. Interconnecting my Kenwood TS-930S to the amp was relatively simple—although at first I was a little confused with the section regarding the CW keying hookups. There are two sets of CW connection instructions: one for Ten-Tec transceivers and another for transceivers from the other manufacturers.

For QSK operation with non-Ten-Tec transceivers, the keying device must be connected to the amplifier's phono-type **KEY IN** jack. This is to ensure proper amplifier/exciter sequencing. Cables are installed between the amp's **KEY OUT** jack and the radio's "key" jack, and from the amp's **PTT/VOX** jack to the radio's normally open T/R relay connection points. Most Ten-Tec rigs have a pair of jacks for making the QSK connections to this amplifier—the keying device plugs into the transceiver and the sequencing is handled within the radio.

Some of the current Yaesu transceivers are also set up for this type of full-break-in amp keying. You wire these rigs similarly to the Ten-Tec gear. This allows you to use the radio's built-in keyer and CW memories—again, the sequencing is handled inside these radios. You can't do this with the majority of the other transceivers, though. To operate QSK with those rigs, a straight key, an external CW keyer or the keying line from your station computer must be connected directly to the **KEY IN** phono jack on the amp.

Joe Bottiglieri, AA1GW ♦ Assistant Technical Editor

Table 1
Ten-Tec Titan II, serial number 02C10070

Manufacturer's Claimed Specifications	*Measured in the ARRL Lab*
Frequency Range (US units): 1.8-2, 3.5-4, 7-7.3, 10.1-10.15, 14-14.35, 18.068-18.168, 21-21.45, 24.89-24.99[1], 28-29.7[1] MHz.	As specified.
Power output: 1500 W continuous in SSB, CW, AMTOR/PACTOR (50% duty cycle or less); 1500 W RTTY/SSTV up to 10 minutes (160-15 meters only). 1000 W continuous key-down, all modes and bands.	As specified for SSB and CW.
Driving power required: 80 W (typical).	Typically 60 W (band dependent).
Input SWR: <2:1.	Typically 1.0:1.
Spurious signal and harmonic suppression: meets or exceeds FCC requirements.	43 dB. Meets FCC requirements.
Intermodulation distortion (IMD): Not specified.	See Figure 1.
Primary power requirements: 216-252 V ac.	
Size (HWD): 8.5×17×20 inches; weight, 84 lb.	

[1]As shipped from the factory, operation on 12 and 10 meters is disabled (see text).

As I do not run QSK, I also tried hooking up the keying line from the T/R relay of the radio directly to the **PTT/VOX** jack on the back of the amp—as I've done with my other amplifiers—and operated CW with the amp in the PTT/VOX mode. This worked fine for non-QSK CW operation.

An ALC jack is also provided on the rear panel. ALC connections are not necessary when using this amplifier with most solid-state transceivers, but if you are driving the Titan II with tube-type transceivers or transmitters, you'll probably want to take advantage of this capability.

The front panel of the amp offers a good deal of metering. The first thing I noticed was that the analog meter on the left side of the front panel is unlabeled. The manual revealed that this meter always displays the plate current. The other analog meter has three scales and indicates screen grid current, plate voltage and forward or reflected power. A four-position rotary switch controls the meter's function.

There's also an LED bargraph that indicates the RF output power. This proved to be very handy during tune-up and standard operating.

Three rocker switches labeled **ON/OFF**, **OPERATE/STANDBY** and **QSK/PTT/VOX** are positioned in the lower left portion of the front panel. The power and operate switches have internal lamps that indicate their state.

The bandswitch—located just to the right of the analog meters—is an eight-position rotary switch. There are two positions each for 40 and 160 meters. Thirty meters is tuned in one of the 40-meter positions. Seventeen and 12 meters are tuned in the 15- and 10-meter positions, respectively. The remaining front panel controls are two large vernier dials that control the tuning and loading capacitors.

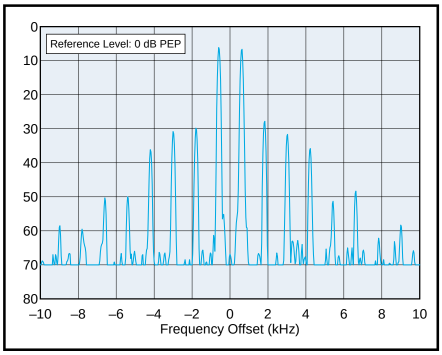

Figure 1—Worst-case spectral display of the Ten-Tec Titan II amplifier during two-tone intermodulation distortion (IMD) testing. The worst-case third-order product is approximately 29 dB below PEP output, and the worst-case fifth-order product is approximately 31 dB down. The amplifier was being operated at 1500 W at 14.020 MHz. The level of the third- and fifth-order IMD products are higher than those we have observed on other recently reviewed amplifiers.

Control Circuits

As with the original Titan, the Titan II is loaded with safety and control circuits to help prevent you from blowing up your beautiful new amp! A three-minute timer is engaged upon power-up. A red **WAIT** LED on the front panel is lit during the warm up period. Other red LEDs on the front panel indicate if there is a dangerous overcurrent condition on either or both of the grid or screen. Overcurrent detection circuitry will automatically trip and take the amp off-line—thereby protecting the tube. Reset is performed with the **OPERATE/STANDBY** rocker switch.

Tuning

During my initial attempt to follow the instructions and tune this amplifier, I felt as if I needed three sets of eyes to keep track of everything. The plate voltage registered 3 kV—exactly as it should—so I switched the meter to read grid current. There are several warnings in the manual not to exceed 55 mA of grid current. With that thought fresh in my mind, I proceeded to follow the directions.

I began the tuning procedure with about 15 W of drive. Nothing seemed to happen with the amp, so I increased the input power to about 25 W. At that point the plate current started to increase. The basic idea is to adjust the **TUNE** capacitor for both maximum plate current and maximum forward power while adjusting the **LOAD** capacitor for minimum screen grid current. It was easy to achieve 1500 W output with about 80 W of drive while coming nowhere near the 55 mA grid current limit.

Moving through the bands and retuning was a snap. At first I was worried that I would have to expend a great deal of effort while retuning—as the knobs seem rather small and the vernier-style controls require a considerable amount of cranking to move the capacitor plates through 180 degrees of rotation. What I soon discovered was that each band has its power peak in control positions that were similar to those of the other bands, so the settings of those controls don't need to be varied over large ranges.

I never neared 55 mA of grid current on any band with only minor adjustments of the input power. Moving across all of 10 meters, I found it quite simple to repeak the tuning by watching the bar graph while leaving the switchable analog meter reading grid current. This operation is very similar to that of the original Titan—which I have enjoyed greatly.

It's a good idea to make a chart of the load and tune settings for each band segment, and there's a log sheet included in the manual for this purpose. The vernier drives turn numbered skirts, so there are no moving pointers to use for adding your own index marks directly on the face of the front panel (a fairly popular marking method on other amps).

Operation

I used the Titan II casually for DXing and in the ARRL 160-Meter Contest. I use a tuner for my 160-meter antenna and it was interesting to watch the amplifier's bargraph power meter while I moved through the band. When I saw the power drop below about 1200 W I knew it was time to adjust the tuner. I usually did not have to readjust the amplifier's tuning once I had lowered the SWR, but I was only operating from 1.8 up to about 1.87 MHz.

The tube is cooled with forced air that exhausts through a chimney on the rear right top of the cabinet. The air intake is underneath, but the manual says that it is okay to operate the amplifier without the bail extended (the bail lifts the amp up a couple of inches). The cooling system is quite loud. The blower really lets you know it is working to keep the tube and cabinet cool. It does a good job—I could barely detect heat during the contest.

This is an excellent, large amplifier. I don't feel a bit squeamish about using it. It is metered well and will put out 1500 W while running cool. The Svetlana 4CX1600B should last many, many years and—if there is a problem—Svetlana warrantees the tube based on its time in service. In the event of a failure within the first 500 hours, the tube will be replaced free. Between 501 and 5000 hours (or 2 years—whichever comes first), the price of a replacement tube will be prorated based on time used. According to Scott Robbins, W4PA, of Ten-Tec, Svetlana claims to have a method for determining length of service by analyzing the tube itself, but this warranty has yet to be exercised. Ten-Tec has always supplied excellent customer service and—with the Titan II—has put itself solidly back into the legal limit amplifier market.

Manufacturer: Ten-Tec, 1185 Dolly Parton Parkway, Sevierville, TN 37862; 865-453-7172, fax 865-428-4483; **sales@tentec.com**; **www.tentec.com**. Price: $2990.

AOR TDF-370 DSP Multi-Media Terminal

Reviewed by Joe Bottiglieri, AA1GW
Assistant Technical Editor

What the heck is *a "multi-media terminal"?* Good question!

The AOR TDF-370 is an Amateur Radio station accessory that defies classification. It can serve as an external audio DSP filter box, but it's much more than just that. It will decode and display RTTY and PSK31 text (and encode these modes when connected to a PC or "dumb terminal"), but it's not quite a multimode TNC. Add in microphone equalization, SSTV interface capabilities, a receive audio recording system and the ability to simulate "stereo" sound on phone and CW signals, and you can begin to appreciate how difficult it would be to pigeonhole this piece of gear in an existing amateur product category. I've come to think of it as a Cuisinart for audio signals.

The Center of Attention

The TDF-370 is packaged in a compact desktop console. It's designed to sit out in the middle of your operating table where you'll have ready access to the top-mounted controls and a clear view of the sloping $^{5}/_{8} \times 2^{3}/_{8}$-inch LCD window. The 2-line by 16-field dot matrix display is capable of portraying any alphanumeric character. A few of the '370's features make use of these same display segments to depict bargraphs and scales.

The controls on the unit include a pair of knobs. One is for volume, the second—**ADJ1**—takes on several different assignments. The keys on its membrane-type keypad are organized into groups. A five-button set controls voice-mode related systems; a three-button set handles CW duties; a two-button set commands digital mode and SSTV operation; a five-button set takes care of input source and level considerations; and a pair are used to work the audio record and playback system. A second button-operated multi-function control—labeled **ADJ2**—is also included.

Just above the keypad is a row of five LED indicators. Three of these serve as a level indicator for adjusting the input signal, the fourth identifies the input source (up to two can be connected), and the fifth—marked "CUE"—lights when the recording feature is active.

Bottom Line

A Cuisinart for the ham shack! The AOR TDF-370 Multi-Media Terminal is an audio signal processor, a data mode controller, a digital recorder and a microphone equalizer all rolled into one.

Stringing the Laces

The TDF-370 works all of its magic at audio, so—much like the arrangement used for TNC or computer sound card interconnection—the unit interfaces to your transceiver via its receive audio output, microphone audio input and PTT lines. Power is supplied by either four internal AA batteries or an external dc source.

Fully integrating the device into your station for transceive operation will require some custom-built cabling, but all of the receive-related capabilities—the DSP filters; the record feature; the "stereo" reception mode; and PSK31 and RTTY decode—can be explored with just a simple connection to your radio's audio output. You can use your rig's external speaker or headphone jack, or a fixed-level receive audio source (usually available from rear-panel "accessory" jacks). Let's take a look at the receive operations first then we'll turn our attention to the setup and operation of the microphone equalizer and digital mode transmit capabilities.

Digital Processing for the Voice Modes

For voice modes, the '370 offers two types of DSP noise reduction, along with bandpass filtering and simulated stereo. The *Instruction Manual* states that the DSP noise-reduction algorithms employed by the brains of this device—a Hitachi SH7034 20 MHz 32-bit microcomputer—are based on the "FFT" (*Fast*

Table 2
AOR TDF-370 DSP Multi-Media Terminal

Manufacturer's Claimed Specifications	*Measured in the ARRL Lab*
Power requirement: 4 AA batteries or 11-14 V external dc, 0.6 A (maximum).	0.36 A. Tested at 13.8 V.
LMS noise reduction: Not specified.	CW, >20 dB; Voice, 30 dB.
LMS notch depth: Not specified.	>35 dB.
Voice passband: 300-2400 Hz.	69-2090 Hz.
Size (HWD): 1.3×4.3×6.2 inches; weight, 12.3 ounces.	

Fourier Transform) and "LMS" (*Least Mean Square*) digital signal processing techniques.

FFT

The TDF-370's "FFT" mode reduces the high-frequency hiss associated with typical band noise and will also help diminish interference from more transient interference such as static crashes. Press the **FTT** button, and the top line of the LCD shows "FTTadaptive." An audio spectrum scope appears on the second line. The **ADJ1** knob is then used to control the amount of noise reduction.

I found this system to be very effective on band noise and pretty good on some consumer-electronics generated hash (computer RFI in my case), but you do have to carefully strike a balance between the amount of noise reduction applied and the resultant degradation in speech intelligibility. If you increase the noise reduction too far, the processed speech will begin to take on a choppy sounding quality. If the level of the interfering noise is particularly high, mid-level noise reduction settings will convert it into a noise floor of rapidly changing low-level tones—a sound that I can only describe as "the music of processing." In many instances, careful adjustment of the audio input level—with the unit's **LEVEL** control and/or **ATT**enuator button—can help enhance the performance of this feature.

LMS

A press of the **LMS** button in the **VOICE** group brings up an "LMS(voice)" message in the display. This feature—as is the case with FFT—works well to reduce band noise, and worked better at handling my flavor of consumer electronics hash. The level of the LMS noise reduction is controlled with the **ADJ1** knob. Again though, if you apply too much noise reduction you'll experience a decrease in voice clarity.

This feature also includes an "Auto Notch" system that does a respectable job of tracking and eliminating constant tones, such as AM carriers and tuning stations. The level of notch attenuation can be varied with the **ADJ2** buttons and is represented on a bargraph scale in the second line of the display. The highest setting invariably worked best for me, but even then the notching performance was not quite as effective as the automatic notch filter that's built into my mid-priced HF transceiver.

Hi Fi

The TDF-370 has the ability to employ its signal processing power to simulate a "stereo" effect on both phone and CW signals. Stereo headphones—or amplified stereo speakers plugged into the '370's **PHONES** jack—are required equipment ("ear bud"-type stereo earphones are included with the unit). The resulting audio is considerably more pleasant to listen to than unprocessed audio—band noise and desired signals sound spatially separated and the noise component is slightly reduced in amplitude and sounds as if it's pushed into the background. The **ADJ1** and **ADJ2** can be used to refine the stereo separation. A **BYPASS** button (that works in all of the unit's audio processing modes) makes it easy to compare the processed and unprocessed signal.

The effect of this feature is pretty dramatic, and was responsible for attracting a steady stream of curious hams to a TDF-370 demonstration set up at the AOR booth at this year's Dayton Hamvention.

BPF

Pushing the **BPF** button in the **VOICE** group brings up a "BPF(voice)" indication in the display and activates a DSP-based audio bandpass filter. The *Instruction Manual* indicates that the **ADJ1** knob serves as a high/low frequency slope control and the **ADJ2** control sets the low cut frequency. In operation though, it seems as if the **ADJ1** behaves more like a high cut control and **ADJ2** acts as a low cut control.

This feature works well for rejecting interference from nearby band activity. You can adjust the passband to favor the lower frequency portion of the desired audio signal to reduce interference from on high, and/or the upper portion to fight off any "alligators" that may be lurking down below.

Digital Processing for CW

For CW connoisseurs, the TDF-370 serves up two varieties of DSP filters and a combination stereo/bandpass filter.

BPF

Tap the **BPF** button in the **CW** group, and "BPF(CW)" appears in the top line of the display. The second line indicates the filter's center frequency and the filter bandwidth. The center frequency is adjustable in 50 Hz steps from 450 to 800 Hz. Filter bandwidths of 300, 200 and 100 Hz are supported. The **ADJ1** knob is used to select the desired center frequency; the **ADJ2** buttons are used to step through the three filter bandwidths.

Typically, you'll want to match the center frequency to your transceiver's CW sidetone offset frequency, tune in a target signal, and then crank down the filter bandwidth. In some instances, however—particularly when the desired signal is reasonably strong—sliding the center frequency up or down slightly can be an effective tool for further reducing interference from nearby signals.

LMS

The CW version of the LMS filter behaves much like the one that's provided for the voice modes. It's very effective on QRN, and—fortunately—the sound of the CW signal isn't significantly changed by the higher level settings of the noise reduction control. The automatic notch filter—for obvious reasons—is not included in the CW mode implementation of this filter.

Stereo CW

The stereo mode for CW operation is combined with the DSP CW bandpass feature described above. The stereo effect on this mode is not as pronounced as that observed with the voice mode version of this feature, but it is noticeable. It is particularly evident when you are tuning across a signal—you'll hear the individual signals you encounter slide from one side of your headphones or speakers to the other as you crank on by.

Digital Recorder

The TDF-370 can record and play back receive audio. An especially interesting aspect of this system is that the captured audio clip will begin 6.4 seconds *before* you hit the record button (the TDF-370 does have to be parked in the record standby mode during that period of time in order to pull this stunt off, though). The system will temporarily hold a total of 102 seconds of audio in up to eight memories.

This feature does have some limitations. If you are going to record multiple message banks, you have to do so during a single recording session. Once you've entered the playback mode, switching back into the record mode—or switching into one of the unit's other features—erases all of the recorded audio. Memories are also lost when the power is shut off.

The TDF-370 is not designed to serve as a contest voice keyer. There are no specific arrangements for replaying recorded audio over the air. The recording system is primarily intended for capturing snippets of audio off the air for immediate analysis.

Digital Mode Receive Capabilities

The '370 can decode PSK31 and RTTY signals, and will display the text directly on its LCD display.

For PSK31, both the BPSK and the QPSK modes are supported. There are two filter bandwidths available: 75 and 220 Hz. In order to capture a signal, you've got to tune very slowly across the PSK31 warbles (a transceiver that provides fine tuning steps is a must). When the system detects a signal and begins to lock onto it, you'll hear a series of tones that are generated internally in the '370. Once this begins to happen, the top line of the display will show the direction and frequency amount—"–020Hz" for example—that you will need to tune in order to zero beat the signal.

Acquiring a knack for tuning in PSK signals using this arrangement takes practice. Those of us who have been spoiled by the put-the-cursor-on-the-signal-and-click sound card PSK31 programs will likely initially experience some frustration. (But hey, at least *we* know what a PSK31 signal sounds like!)

Once the system is properly locked on, copy will appear in the second line of the two-line display. Upper and lower case letters and all of the usual symbols and numbers can be depicted by the dot-matrix display.

One familiar PSK31 characteristic that is not supported—on the built-in display, anyway—is the backspace feature. There's also no way to scroll back through the message contents, so be sure to keep a pencil and paper handy.

For RTTY operation, baud rates of 45, 50 and 75 and shift frequencies of 170, 425 and 850 Hz are available. Tuning in RTTY signals involves listening in stereo headphones or speakers and tuning back and forth across the signal until the RTTY signal can be heard on both the left and right channel. Once you are close to the target signal, you can again use the frequency offset magnitude and direction that's shown in the top line of the display—and the message text—to lock on. Tuning in RTTY signals is relatively easy.

Plug in Your Iron

The receive capabilities of the AOR TDF-370 are pretty neat, but if you want to take full advantage of what this accessory has to offer, you're going to have to melt some solder.

AOR supplies an eight-pin microphone jack and plug and two 3.5-mm stereo plugs for making up cables to connect your transceiver's stock microphone to the unit's **MIC** input, and the '370's **AUX** output to the rig's microphone jack. You'll need to supply a few feet of shielded wire.

The pin-outs for the TDF-370's jacks are provided in the manual. Refer to your transceiver's owner's manual for the microphone connector pin-outs. Pay particular attention to the ground connections. Mike grounds and PTT grounds will need to be kept separate, and this will require making some connections directly between the two eight-pin connectors.

It would be extremely helpful if AOR included specific wiring diagrams for a few of the major transceivers in the documentation that they pack with this device.

The Microphone Equalizer

The Microphone Equalizer mode delivers an eight-band graphic equalizer, a DSP-based background noise reduction feature and a microphone level adjustment. There's even a built-in monitor for evaluating the effect as you change the settings.

The center frequency for each band can be dialed up in the display by using the **ADJ1** knob. The **ADJ2** buttons are then used to vary the individual compensation levels. The manual states that the compensation levels for each band can be varied between −6 and +6 dB. The ARRL Lab measured this range as closer to −10 to +4 dB. The numbers that appear in the display range from –30 to +6 dB in the bands below 1000 Hz and −30 to +12 dB in the bands above. Bottom line: There's plenty of adjustment latitude—use the displayed numbers as a relative level indication only.

The background noise reduction level and microphone gain adjustments are also made with the **ADJ1** and **ADJ2** controls. All of these settings are retained in the TDF-370's memory, so you won't have to reset these values each time you turn the unit on.

Digital Mode Transceive Capabilities

In order to use the AOR TDF-370 for transceiving in PSK31 and RTTY, you'll need to connect it to a "dumb terminal" or a PC running terminal software. AOR supplies a cable that will connect to either a DB9 or DB25 COM port. The manual suggests that you try *Windows HyperTerminal* software, and provides the communications parameters for configuring the port. The transmit audio and PTT connections are made through the cables described above.

Once the TDF-370 is wired up and properly configured, operation is essentially identical to that of a multimode TNC. A "command" mode is used to set up the various operating parameters, and then control key combinations (Ctrl+T for transmit, for example) are used to operate the system. Both transmit and received text will be displayed on the monitor screen, and received text will also appear in the '370's display window.

SSTV?

I was unable to explore the slow scan television transmit and receive features of the TDF-370. Operation on SSTV requires a special computer software package that has not yet been released. AOR reports that the software should be available very soon, and will be downloadable from their Web site.

Information in the manual indicates that this feature will support all of the popular SSTV modes.

Conclusion

With a single audio connection to your radio's audio output, the AOR TDF-370 multi-media terminal delivers a nice variety of receive audio signal processing features, along with stand-alone digital mode reception for two of the most popular digital modes. Build up some adapter cables and position the unit in line between your transceiver's microphone and microphone input jack, and you'll add flexible transmit audio tailoring and DSP-based station background noise reduction. Cable it up to a PC (or "dumb terminal") with the included COM port cable, fire up some terminal software, and you'll enjoy full transceive capabilities on PSK31 and RTTY. SSTV should be coming soon!

Manufacturer: AOR USA Inc, 20655 S Western Ave—Suite 112, Torrance, CA 90501; 310-787-8615, fax 310-787-8619; **www.aorusa.com**. Manufacturer's suggested list price: $329.95. Typical current street price: $325.

PropMan 2000

By Carl Luetzelschwab, K9LA

Rockwell Collins recently released an upgraded version of their HF *Prop*agation Resource *Man*ager software. The new upgrade is called *PropMan 2000*. In a nutshell, *PropMan* identifies and displays the best frequencies for an HF communications link in a user-friendly real-time graphical environment.

My first exposure to the *PropMan* series of propagation prediction software was with Version 3.1 in 1995. This was a *DOS* version that I ran on my old 386 PC. It used *IONCAP* Version PC.25 for its raw data, and presented plots of Signal-to-Noise Ratio (SNR) for the path selected. It was obvious that its heritage was from the military market, as its list of locations (for the receive end and transmit end of the path) was heavily slanted toward military installations and it talked about channels as opposed to frequencies.

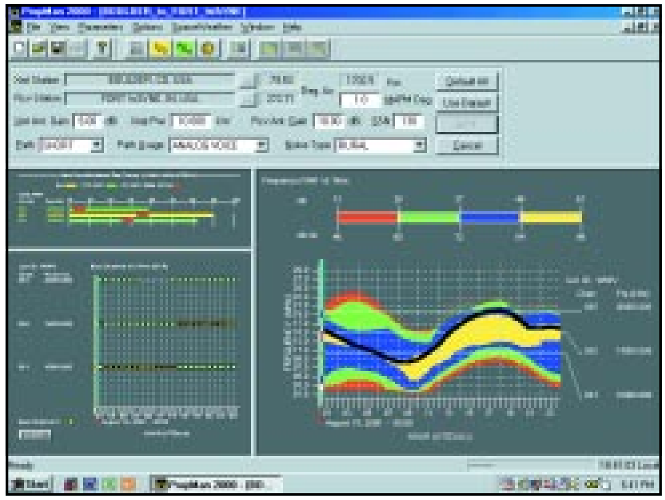

Figure 1—*PropMan 2000* analyzes a path between Boulder, Colorado and Fort Wayne, Indiana.

New and Improved

So what's new in this upgraded version? *PropMan 2000* is now a *Windows* program, so that makes it easy to print the color-coded screens for a hard copy (that was a real task with the *DOS* version). *PropMan 2000* now uses *VOACAP* Version 99.0708W for its underlying propagation predictions and the raw *VOACAP* data can easily be viewed with a couple of clicks of your mouse. And the list of locations has been revised extensively to move it away from its military slant.

PropMan in Action

I set up *PropMan* for a simple path, and took a look at what it had to offer. Once the proper parameters were entered, *PropMan* immediately ran predictions, and the screen looked like Figure 1.

The path parameters are displayed at the top of the screen. The top left plot area displays one of three plots (selected via the colored plot icons at the right end of the toolbar): Best Channel for Selected Time, Best Frequency versus Time, or Channel SNR versus Time. The bottom left plot area is the Best Channel versus Time. The right plot area is Frequency/SNR versus Time. Any of the five plots can be made to fill the entire screen by double clicking on it.

The Best Channel for Selected Time plot shows the predicted SNR on your selected frequencies in real-time format (which means it'd be nice to have your PC set to the correct time!). The Best Frequency versus Time plot is not dependent on transmit power or antenna gains, which suggests it is more of a plot of which frequency is optimum solely in terms of ionization (more on this later). The Channel SNR versus Time plot gives the SNR for each hour on each selected frequency over a 24-hour period. The Best Channel versus Time plot takes the data from the Channel SNR versus Time plot and shows which frequency is best for each hour during a 24-hour period. The Frequency/SNR versus Time plot shows the SNR for all frequencies from 1 to 30 MHz for each hour for a 24-hour period. This last plot also includes the monthly median MUF as a thick black line for each hour.

PropMan also allows automatic input of space weather data from Internet sources, and this feature provides two additional and unique reports: warnings of ionospheric storms, and a table of percent degradations for each hour to be applied to the predicted MUFs. Although the Help menu discusses using Microsoft Internet Explorer for this Internet feature, I had no problems using Netscape. And it is not out of the question nowadays to have *PropMan* continuously obtain space weather updates on a dedicated Internet line.

As you can see from the descriptions of the plots, *PropMan* is heavily slanted toward reporting SNR. SNR (or signal strength) is only one of the outputs of a propagation prediction—the other is mode availability (which *VOACAP* calls "MUFdays"). It appears that the Best Frequency versus Time plot in *PropMan* addresses mode availability, as its values are independent of transmit power and antenna gains and are somewhat less than the median MUF in the Frequency/SNR versus Time plot. This suggests the Best Frequency versus Time plot is somewhat akin to the FOT—the frequency that should be available on 90% of the days of the month.

Conclusion

Overall I found this new version of *PropMan* easy to use in the *Windows* environment. The Help texts and Tutorials are extensive, and should allow you to navigate and use *PropMan* with little trouble. The information presented in all the plots is rather extensive, so some study will be necessary to fully comprehend what you are looking at.

PropMan 2000 is available for $99 US plus tax and shipping and handling. For more information, or for questions regarding *PropMan 2000*, call Rockwell Collins at 319-295-5100 or at 800-321-2223, e-mail **Collins@collins.rockwell.com** or visit their Web site at **www.propman2000.com**.