USE OF THE DSP-10 TRANSCEIVER AND AUDIO PROCESSOR FOR WEAK-SIGNAL EME

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Abstract - **The DSP-10 2-meter transceiver consists of an RF hardware package, an associated DSP processor and a control program that runs on a PC. Alternatively, most of the program functions are available as an audio processor, except for frequency control, without the RF hardware. The programs support specialized weak signal modes. The LTI (Long-Term Integration) mode allows detection and measurement of extremely weak signals. The EME-2 mode performs similar functions as LTI, but includes timing and Doppler corrections for Moon self-echoes. The PUA43 mode applies the principles of LTI to a communications mode using 43-tone FSK. These modes can be used for signals too weak to be heard by the ear. Comparisons between the JT44 mode of the WSJT program and PUA43 demonstrate the advantage of a precise knowledge of time and operating frequency, as is needed by the latter mode.**

INTRODUCTION - Considerable interest has been generated by newer modulation and coding schemes designed for weak signal communication. The PUA43 mode, using the DSP-10 transceiver, was specifically designed to push the limits of weak signal communication by trading time for received signal power. This mode includes provisions for EME contacts. Examples of the performance capability are a 1296 MHz contact in February, 2001, between W7LHL and W7SZ using 5-Watts and TVRO-dish antennas[1] and a contact between myself and W7SLB on 2-meters using 150 Watts and single Yagis. Recently, much interest has been generated by K1JT's JT44 mode that shares many of the characteristics of PUA43, but removes the need for tight frequency and time control, making it quite suitable for use with ordinary transceivers.

As seen in figure 1, the full DSP-10 includes RF hardware[2]. That configuration provides close integration of the frequency control functions. If the frequency control can be handled manually, the RF hardware can be replaced by a commercial transceiver. It is possible to use the DSP-10 programs running in a DSP board, along with the PC control program, as an audio processor. This supports both transmit and receive and gives an easier way for getting started. The weak-signal performance is the same as long as the frequency control is adequate. The DSP board is the EZ-KIT Lite[3] demonstration board available from Analog Devices.

This note summarizes the three weak-signal modes of the DSP-10, and makes some comparisons between PUA43 and JT44, all in the EME context.

PUA43 EME Contacts - For terrestrial PUA43 contacts, it is possible to call CQ and have contacts. But for EME, frequency accuracy requirements almost precludes this . Thus, EME QSO's need to be pre-arranged. Part of the pre-arrangement is finding the latitude and longitude of the station to be worked to allow precise EME Doppler corrections. Then transmit frequency, bandwidth, 14 or 28 character messages, and timing need agreement. This is obviously not suitable for casual contacts, but as will be seen, there are performance benefits from knowing this information fully.

Every minute a full PUA43 sequence is sent. This either consists of 28 characters sent once or 14 characters sent twice, followed by four seconds for CW identification. Transmission and reception are both automatic. Each of the 43 different character symbols is represented by a different tone and during transmission, these are sent out every two seconds. During receive, the "most likely" characters are displayed. Some character is updated every two seconds. In addition, in small characters above each "most likely" character is the "second most likely" character. Additional data is supplied by three colors used to indicate the "quality" of the character reception. This is a measure of the signal-to-noise ratio associated with the particular character. If all characters are white, the probability is high that the message is correct, whereas if they are black, the S/N is low and errors are more likely.

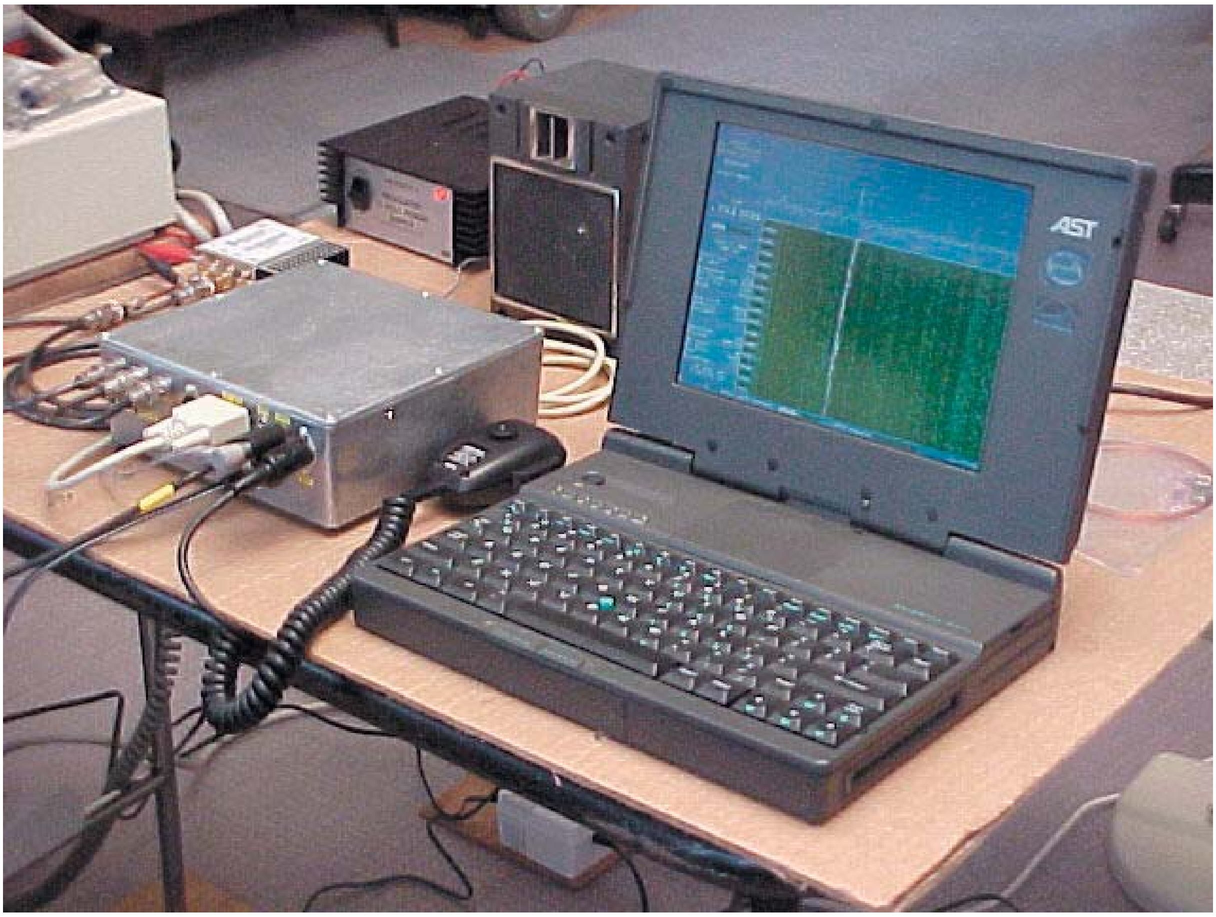

Figure 1 The DSP-10 setup used by W7LHL. The aluminum box to the left of the laptop computer has the DSP board , RF and IF hardware. The laptop computer is a front panel and control device for the station. Displays on the computer show the spectrum of the received stations and the processed results from the weak-signal modes, such as PUA43. (Photo by W7SLB)

Figure 2 is a screen shot of the DSP-10 during reception of PUA43 EME sent by W7LHL and received by W7SZ as in the 1296 MHz QSO noted above. After 17 minutes the copy was 100%. This is shown by the "most likely" received text displayed in large characters at the top of the screen. In this case, the operators decided to send the calls and grid squares twice in a single 28 character message. For this QSO, the S/N in a 4.7 Hz bandwidth was about -5 dB (or -15 dB in a 50 Hz bandwidth) and no indications were visible on the waterfall display.

Normally transmissions continue, either for a predetermined time, or until the "estimated message" is credible. This parallels CW reception, although the amount of machine involvement may be less. Next, the other station transmits, either by pre-arranged schedule, or by other indication. QSO's are complete after there has been an exchange of call signs and some information. Grid squares exchanges make suitable information transfers, but signal reports could be used as well. Acknowledgements in the forms of multiple 'R' characters have generally been used, but are not necessary.

No general tuning is required for PUA43. EME Doppler corrections are fully automatic, applied to the receiver frequency and generally accurate to 1 or 2 Hz at 1296 MHz. This error is proportional to the frequency of operation. Sufficient Doppler-correction accuracy is provided to avoid being a dominating factor, relative to the spectrum broadening associated with Moon reflections.

Automatic switching between transmission and reception is also provided for with the DSP-10 and this can be used with PUA43. Normally periods of 1 to 10 minutes would be used for each end of a QSO.

Synchronization of transmission and reception times requires clocks accurate to about 50 milliseconds[4]. This accuracy also ensures that the Doppler corrections will not suffer from time errors. An internal software clock in the DSP-10 PC program allows one to coordinate time with external standards using an audible "tick." Setting this is a manual operation, but adequate for the job at hand.

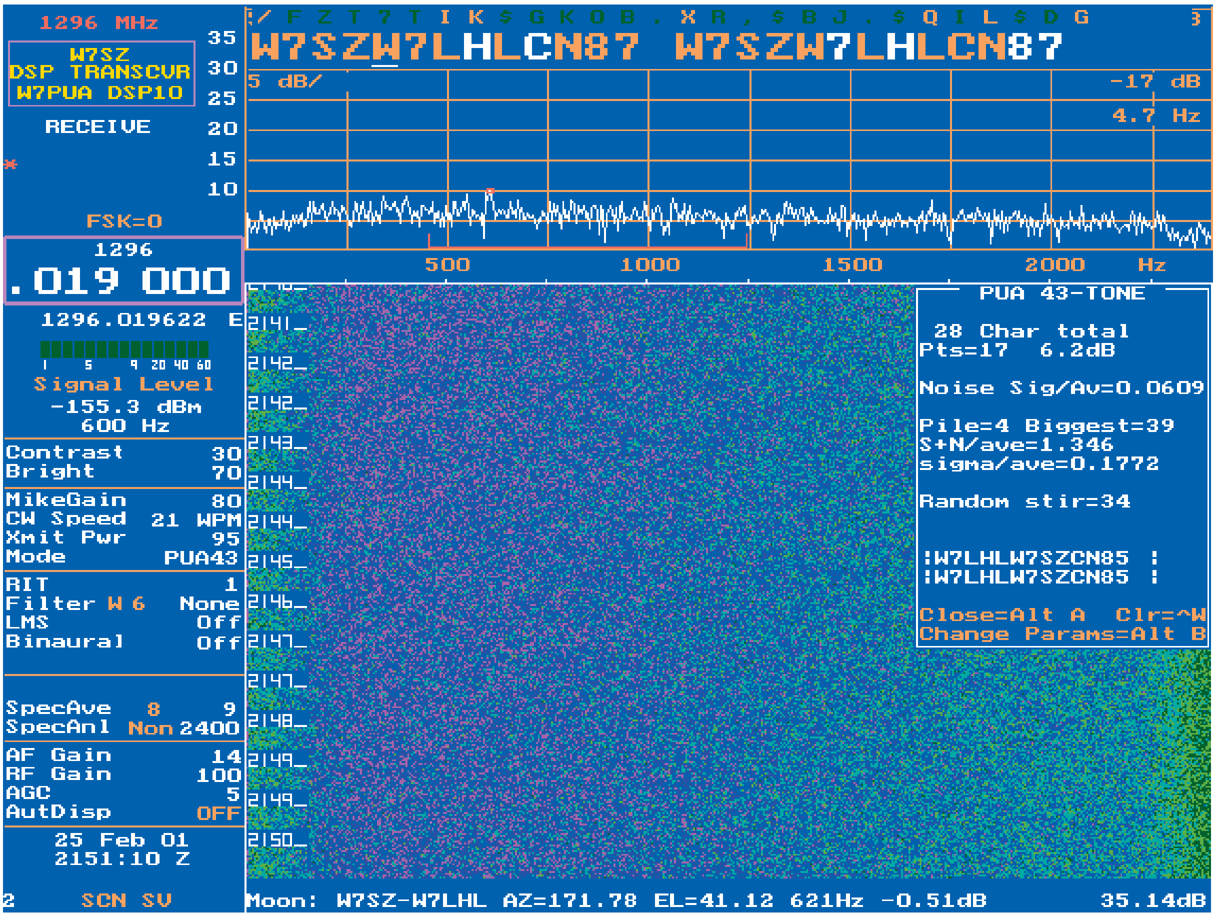

Figure 2 DSP-10 screen at W7SZ while receiving W7LHL with the PUA43 mode by 1296 MHz EME. This shot is 17 minutes into a continuous reception of the repeated message and has achieved 100% copy. Transmitter power was 5 Watts and TVRO dishes were used at both ends of the contact. Down the left side are close-to-conventional controls for the transceiver. There are also controls for determining the display parameters. The DSP-10 operates at 2-meters and 1296 MHz operation shown here uses a transverter. Frequency control of the DSP-10 includes the transverter LO frequencies and makes the transverter operation transparent, including corrections for Doppler shift. At the top of the display are two lines showing the estimated messages for PUA43. The large characters are the most likely and the smaller ones on the top line are the second most likely. This QSO is using 28 character messages, but they consist of repeated 14 character messages.

Before looking at the internal operation of this mode, some data on the S/N performance is useful. The following chart is the general performance that has been measured using non-fading signals. Fading signals, with the same average power, will not change the PUA43 performance, but slow fading, for a constant average power, is beneficial for copying CW. The CW performance shown is what I measured, using a "12 WPM Matched filter[5]," and others may see different results!

S/N in 50 Hz BW	Mode	Performance
0 to 3 dB	CW	Q4 to Q5 Copy by Ear
-12 dB	PUA43	100% Copy 150 sec (2.5 min)
-17 dB	PUA43	100% Copy 1500 sec (25 min)
-22 dB	PUA43	100% Copy 15,000 sec (4 hr)

The PUA43 data is based on multiple measurements in a controlled environment using a 14 Character message. One should also observe that for PUA43, there is no limit to the tradeoff between signal level and time, other than the practical one from the lack of human patience. Based on this data, generalizations are made, such as "PUA43 is able to provide a 10 to 15 dB improvement over CW." The validity of these statements tends to come under question more at lower frequencies where the fade rates are slow and the benefits to CW operation are the greatest. At higher frequencies, such as 1296 MHz, the fading does not allow as much CW copy on the peaks and both modes are left to deal with average power.

Long-Term Integration - All of the weak-signal processing in the DSP-10 is centered around the idea that band-pass filters where both signal and noise is present, will have more power coming out than those with noise only. The word "power" implies a long term average, and indeed a short term estimate of noisy signal power can give poor answers. The longer the averaging time, the more accurate the power estimate. Before seeing how this is applied to communications, it is best to look at a simple power measurement.

A measurement mode, called Long-Term Integration or LTI is available to analyze the signals and noise being received. The data presented is optimized for amateur radio operation, including automatic Doppler corrections for EME work. But the transmit function for this mode is ordinary CW. The easiest signal to analyze is a plain sine wave carrier, and the more time the key is down, the better the measurement will be. Two stations would ordinarily operate in LTI with transmits periods alternating. Provision is available to automate this sequencing.

The received signals are down-converted in hardware to an I-F frequency of about 15 kHz. The remainder of the processing is in the dedicated DSP. For the LTI mode this consists of a single-sideband down conversion to audio followed by a 1024 point discrete Fourier transform[1] (DFT). The total power in each of the 512 DFT outputs is found and is the fundamental quantity for processing. This is equivalent to 512 band-pass filters, followed by true power detectors, all running simultaneously. Each filter is tuned to a different frequency with the frequency spacing determined by the time span of the data going to the DFT. For instance, if the data is collected at 2400 samples per second, the spectral width to be examined is half of that or 1200 Hz. A 1024 point DFT will produce 512 filters spaced every 1200/512=2.34375 Hz.

Interpretation of the data is straightforward if the received signal is on a known center frequency. The power coming from the off-center frequencies can be used to estimate noise. The center frequency power estimates signal-plus-noise. A single measurement of these quantities will not be useful when the signals are weak. However, they lend themselves to averaging, described mathematically as integration. The more the estimates are averaged, the lower the noise becomes. If this process is done with precision, it can be continued for very long periods. Implementations such as is used in the DSP-10 can continue for hours or even days, allowing measurements in the range of 30 dB below audible levels.

Several problems need to be dealt with to achieve this performance. The frequency response of the receiver may not be flat. Measuring the noise both above and below the center frequency can reduce this error. Further reductions come from changing the transmitted signal frequency every minute. For the DSP-10 this is done randomly, with a common table used at each end, to coordinate the changes. A second problem is the presence of coherent birdie signals[2], particularly at lower frequencies, such as 144 or 432 MHz. Again, the random selection of frequency allows the birdies to be made noise-like, since they are highly unlikely to follow the frequency pattern. The power of the birdies reduces the signal-to-noise ratio, but they do not stay on a fixed frequency to totally disrupt the measurement.

Estimation of the actual signal power level is practical, since the signal-plus-noise to noise power ratio is available. This requires knowledge of the system noise temperature, and produces accurate results, even at very low signal levels.

EME-2 - The LTI mode is able to make measurements of very weak signal levels between two stations. EME-2 is a closely related mode that allows a single stations to measure their Moon-bounce capability. The timing is automatic but follows the pattern often used for audible echoes. A 2-second pulse is transmitted, T/R sequencing occurs and then after about 0.6 seconds, the receiver is activated for a 2-second period. Every 5 seconds this pattern is repeated. This would be the audible process except that it can be repeated many times and the powers from the DFT added to reduce the noise in the power measurements. Within a few minutes it is possible to make solid measurements on echoes much too weak to be heard.

An interesting example is shown in the EME-2 spectral plots of figure 3. The power used for this test was only 5 Watts and the antenna was four 2.5 wavelength K1FO Yagis. From my urban backyard, it took just over 3 hours to obtain a well-defined trace as shown. The upper plot (in the top graph[3]) is the average of 2250 2-second pulse returns. The scale for that plot is only 0.05 dB per division, so the S+N/N in a 13 Hz bandwidth is about 0.1 dB. For the estimated noise (Te around 400K) the long-term average signal level is about -178 dBm. To put this into perspective, if the power was raised to 1000 Watts (up 23 dB), Faraday rotation was chosen for a peak (up 3 dB), and a Rayleigh (libration) fading peak was found (often up 6 dB), the signal level would be about -146 dBm. This peak situation, with the high power, would allow CW copy, as one would expect from experience.

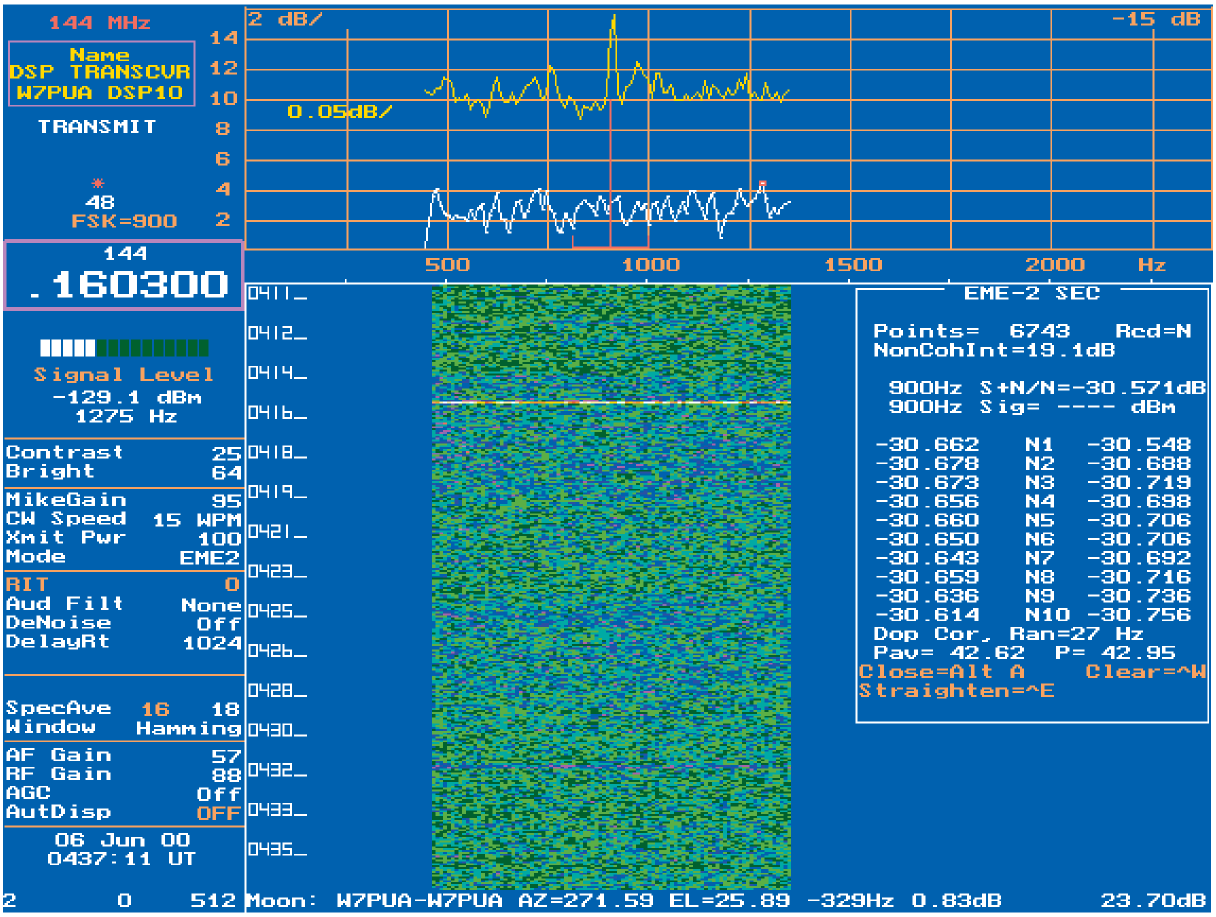

Figure 3 DSP-10 EME-2 mode screen shot of QRP 2-meter self echoes at W7PUA. The lower spectral plot is a single 2-second and does not show any return. Likewise, no signal indications are on the waterfall display. The upper spectral plot is the average of the white traces for just over 3 hours. It shows a distinct peak at 900 Hz, which is the Doppler corrected receiver frequency. See the text for more details of this test.

The amount of time required to achieve a good EME measurement is a strong function of the transmitter power. We illustrated the extreme case with 5 Watts, but every time the power is doubled, the time required for the same accuracy drops by 1/4[4].

The big utility of EME-2 is for evaluating EME performance. In a few minutes it is possible to determine the overall system performance including all equipment, external noise sources and the EME path.

EME-2 also illustrates some limitations on any system that relies on average power. At lower frequencies, such as 2-meters, conventional CW operation benefits from the fading on signals over an EME path. The average power based system does not obtain any of this benefit, with the exception of Faraday rotation. Average power systems can gain back 3 dB by using circular polarization. The remaining fading difference of perhaps 6 dB is not available to long-term power averaging.

In order to allow the 2250 pulse returns to be added power-wise, it is necessary that the frequency of the receiver be Doppler corrected with time. The accuracy needs to be considerably better than the measurement bandwidth, which for the 5-Watt experiment was 13 Hz, but could be as little as 2.3 Hz in the DSP-10. The correction for Doppler in the DSP-10 is adequate to support these measurements.

PUA43 Mode - The LTI mode supports measurements of a second station. EME-2 modifies this to allow single-station measurements of moon-bounce signals. But we still have no ability to communicate information. The PUA43 mode applies these same principles for communicating using multi-tone frequency-shift keying (MFSK).

A short message of 28 or 14 characters is sent repeatedly, once or twice a minute with each character sent for 2 seconds. A separate frequency is assigned to each of 43 characters in the defined alphabet. For each message character position, the receiver looks at all character frequencies and estimates a received message according to the maximum power. For weak signals, there will be errors due to noise. In time, however, the effect of the noise can be reduced, without limit, by averaging. This is the same process used in the LTI and EME-2 measurement modes.

As with the measurement modes, there are potential problems both with receiver response flatness and with birdie interference. A noise-measurement-DFT bin is placed between each of the signal-measurement bins. The two noise-measurement bins surrounding the signal measurement bin are averaged to estimate the noise power. Since these noise measurements occur 14 or 28 times as often as the signal measurements, the noise-estimate accuracy is very good. To further reduce the effects of flatness along with minimizing birdie interference, the frequency associated with any character is changed every minute. This is done by shifting all characters by an amount varying from 0 to 42 frequency spacings. If the frequency is shifted outside the normal band, it is rotated around to the lower end of the band. Of course, the transmitter and receiver must be coordinated. This is done by a random shift table that has 1440 entries, one for each minute of the day.

In total there are 43 DFT bins for signal filters, 45 bins for noise filters and 86 bins that are not used except to increase the separation between signal and noise filter frequencies. The resulting system occupies a band of 174 bins from the 512 DFT outputs. The widest occupied band is for the 9.38 bin spacing where the 43 frequencies are spread across 1575 Hz.

Three different bandwidths are available for PUA43. At lower operating frequencies, the narrow bandwidths can be used. When microwaves are used, the received spectrum is often broadened by the propagation medium, EME or terrestrial. Wider bandwidths must be used for those situations, even though the signal-to noise performance is not as good. The three bandwidths and characteristics are:

BIN SPACING	DFT BANDWIDTH	PUA43 BAND	SIGNAL SPACING	RELATIVE S/N
2.34 Hz	0 to 1200Hz	450 to 844 Hz	9.38 Hz	0 dB
4.69	0 to 2400	450 to 1238	18.75	-1.5
9.38	0 to 4800	450 to 2025	37.5	-3

Doppler corrections for the EME path are applied for PUA43 communication.

JT44 Comparison - The JT44 mode created by Joe Taylor, K1JT, has created a lot of interest in “processed QSOs” since it does not require any hardware in addition to a fast PC and a sound card. The fundamentals of this mode and PUA43 are related. We will try to compare the potential performance of the two modes[5].

PUA43 requires that the frequency and timing be known at both ends of the QSO. JT44 uses a pseudo-random synchronizing pattern to remove this requirement. This creates two differences, the amount of time spent collecting message power and a diminishing probability of synchronizing with lower signal levels. In addition, there are differences in performance due to filter bandwidth and message length, both between the modes and within PUA43.

The time spent processing each character in a message depends on the number of characters sent and the percentage of time spent on this task, as opposed to synchronization. In the case of PUA43, there are two message lengths to consider as is shown here[6].

MODE	CHARACTERS	TIME/CHAR/MIN	REL TIME	REL S/N
PUA43	14	3.84 sec	1.00	0.0 dB Ref
PUA43	28	1.92 sec	0.50	-1.5 dB
JT44	22	1.11 sec	0.29	-2.7 dB

The time per character per minute[7] is a measure of the transmit power being applied. The column for relative time normalizes this to 3.84 for PUA43 with 14 characters. This can then be related to an equivalent S/N, in dB, that would have the same effect as the shorter time periods by S/N=5*log10(t) where t is the relative time.

As long as synchronization is achieved, JT44 can make up the performance difference in the table by increasing the time spent in reception or by increasing the transmitter power, both by the amount indicated in the table. The minimum signal-to-noise ratio for JT44 synchronization has been indicated by K1JT as -12.9 dB in a 50 Hz bandwidth[6]. This corresponds to a PUA43 "100% copy" time of about 4 minutes (14 character message, 2.3 Hz Bandwidth).

The filter bandwidth is selectable in PUA43 as 2.3, 4.7 or 9.4 Hz, with additional width if the selectable windowing functions are used to reduce the interference from strong signals. The intermediate level is similar to the 5.4 Hz bandwidth of JT44. Cutting the bandwidth in half increases the S/N by 1.5 dB since the bandwidth lets in half the noise, but there are also only half the points to average. Thus the factors for bandwidth are:

MODE	BANDWIDTH	REL BW	REL POWER
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PUA43	2.3 Hz	1.0	0.0 dB Ref
PUA43	4.7 Hz	2.0	-1.5 dB
PUA43	9.4 Hz	4.0	-3.0 dB
JT44	5.4 Hz	2.3	-1.8 dB

So long as the signal is strong enough for JT44 to achieve synchronization of time and frequency, the modes can be compared by the amount of transmit power needed to achieve the same effective S/N. This is found by adding the "Rel Power" columns. The biggest difference is for the case of PUA43 using 14 characters and a 2.3 Hz bandwidth. Here PUA43 has about a 4.5 dB advantage (a time factor of 8). For the case of PUA43 using 28 characters and a 4.7 Hz bandwidth, the advantage over JT44 is only 1.5 dB (a time factor of 2).

Conclusions - PUA43 mode of the DSP-10 transceiver allows both terrestrial and EME communication with sub-audible signals. Lower signal levels require more time and this tradeoff can be applied for minutes, hours, or longer. For a transmission time of 25 minutes good copy can be expected at -17 dB S/N in a 50 Hz bandwidth, assuming well behaved noise.

A basic theme of all DSP-10 weak-signal processing is long-term power addition (integration). The I-F filter bandwidths are made narrow to reduce noise, but there are limits to this process due to the coherence of the received signal. To extend the weak signal capability, power addition is used.

PUA43 operating with a 2.3 Hz bandwidth and a 14-character message has the potential to provide equal copy in $1/8^{th}$ the time of JT44. Other combinations of message length and bandwidth will provide smaller differences. In addition, PUA43 has no synchronization requirements and can operate at lower S/N. The price for this improvement is the need for accurate frequency and time control at both stations as well as knowledge of both stations latitude and longitude (for EME).

I would like to thank the DSP-10 user community for their hard work and experimentation. Many people have helped with improving the project along with getting the radios built, tested and integrated with the software. In addition, the work reported here was greatly aided by help from Ernie Manly, W7LHL, Beb Larkin, W7SLB, Larry Liljeqvist, W7SZ and Mike Reed, KD7TS. Comments from Joe Taylor, K1JT were very helpful for the JT44/PUA43 comparisons.

APPENDIX A - GETTING STARTED WITH THE DSP-10

The easiest path to using the weak signal modes of the DSP-10 is through the Audio Processor. This uses the same DSP and control programs, but requires no RF hardware. The EZ-KIT Lite DSP demo board is still needed. This works with almost any PC that can operate with DOS. The programs are available free from the web site in note 2. This can operate with the speaker output and microphone input of any SSB radio. This setup is limited only by the accuracy, set-ability and stability of the frequency control in the radio. Send-receive control can be handled initially by manual means, but to bring this under PC control some additional hardware is needed. EME Doppler corrections can be done automatically as long as the shift is not outside the radio pass-band. For microwave frequencies this can be a problem.

The programs and information for using the audio processor are available from the web site listed in Note 2.

Adding DSP-10 RF hardware makes the full frequency control available, as well as making several operations more convenient. The stability and accuracy of the frequency setting in commercial transceivers (used with the audio processor) is generally not adequate to support the weak signal modes. Software provision is made for using transverters to extend the frequency range of the DSP-10 transceiver.

The articles and web page of note 2 have details of the hardware and its assembly. TAPR has made kits of parts available from time-to-time and they can be reached from the link on the web site. The output power of the DSP-10 is 20+ milliwatts. A companion 8-Watt amplifier can be built and again details come from the referenced web site.

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[1] There is a separate signal path to provide a conventional audio signal for the ears. This is not used at all for the LTI trace.

[2] All of the performance numbers quoted in this note assume that the noise is uncorrelated from sample-to-sample and flat with frequency, called White Noise. Birdies are a form of noise, since they are not the desired signal, but they are correlated and not white. The performance numbers do not at all apply too this general class of correlated noise.

[3] The right side of the display is divided into two parts, one over the other. On the top are two spectral plots, the lower one corresponding to the latest measurement and the top one being the average of many of the lower plots. The scale is different for the two plots. For figure 3, the latest plot has a scale factor of 2 dB per division. The dB scale values for a single plot are all relative. The lower-right display is a waterfall that proceeds from top to bottom with time, while brighter colors represent higher power.

[4] This scaling carries several statistical assumptions which come under question when interfering signals are a major noise source.

[5] It would be interesting to actually measure the performance of various modes, including JT44, PUA43 and CW. This could be done in both a controlled environment and over the air. This may show differences in performance beyond the theoretical potential discussed here.

[6] The numbers in the table apply to random messages, meaning that there is no relationship between the characters in the message.. JT44 has special provision for a repeated two-character sequence, such as "RORORO..." that behaves as if it was a two-character message, rather than 22 characters. This improves the average copy by 5.2 dB. JT44 also integrates the trailing four characters. If these are all the same, there will be an average gain of 3 dB. In addition, any repeated sequence in either JT44 or PUA43 (or CW) will help the operator to deduce the received message from the observed patterns.

[7] To be consistent and allow comparisons, the times assume that one station transmits and the other receives. Both PUA43 and JT44 are normally used with half the time in transmission and half in reception, so in that sense, the time/char/min is only half of that shown here. The relative time and relative S/N remove this consideration.

NOTES:

- Details of several PUA43 EME QSO's are on the web page http://www.proaxis.com/~boblark/wksig1.htm.
- The DSP-10 project was described in a three-part article by Bob Larkin, W7PUA, "The DSP-10: An All-Mode 2-meter Transceiver Using a DSP IF and PC-Controlled Front Panel," QST, Sept., Oct., and Nov. 1999. A large amount of additional information and the computer programs are available from http://www.proaxis.com/~boblark/dsp10.htm.
- The current part number is ADDS-2181-EZLITE. The previous ADDS-21XX-EZLITE works as well. See the DSP-10 web page in the previous note for more details.
- For PUA43 the frequency needs to be known to a fraction of the DFT bandwidth. If a bandwidth of 2.3 Hz is used, it is desirable to know, and control, the frequency to about 0.5 Hz. The 2 second pulses for this mode have inter-symbol interference if the timing is off. Current experience indicates that errors of 50 or even 100 milliseconds are tolerable.
- C. L. MacCluer, W8MQW, "A Matched Filter for EME," Proceedings of Central States Conference, 1995. The 12 WPM dot filter is implemented in the DSP-10 software. For further information on essentially the same approach, see Bill De Carle, VE2IQ, A DSP Version of Coherent-CW (CCW)," QEX, Feb. 1994, pp25-30.
- Joe Taylor, K1JT and Andy Flowers, K0SM, WSJT User's Guide and Reference Manual, Version 2.0, April 14, 2002.

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