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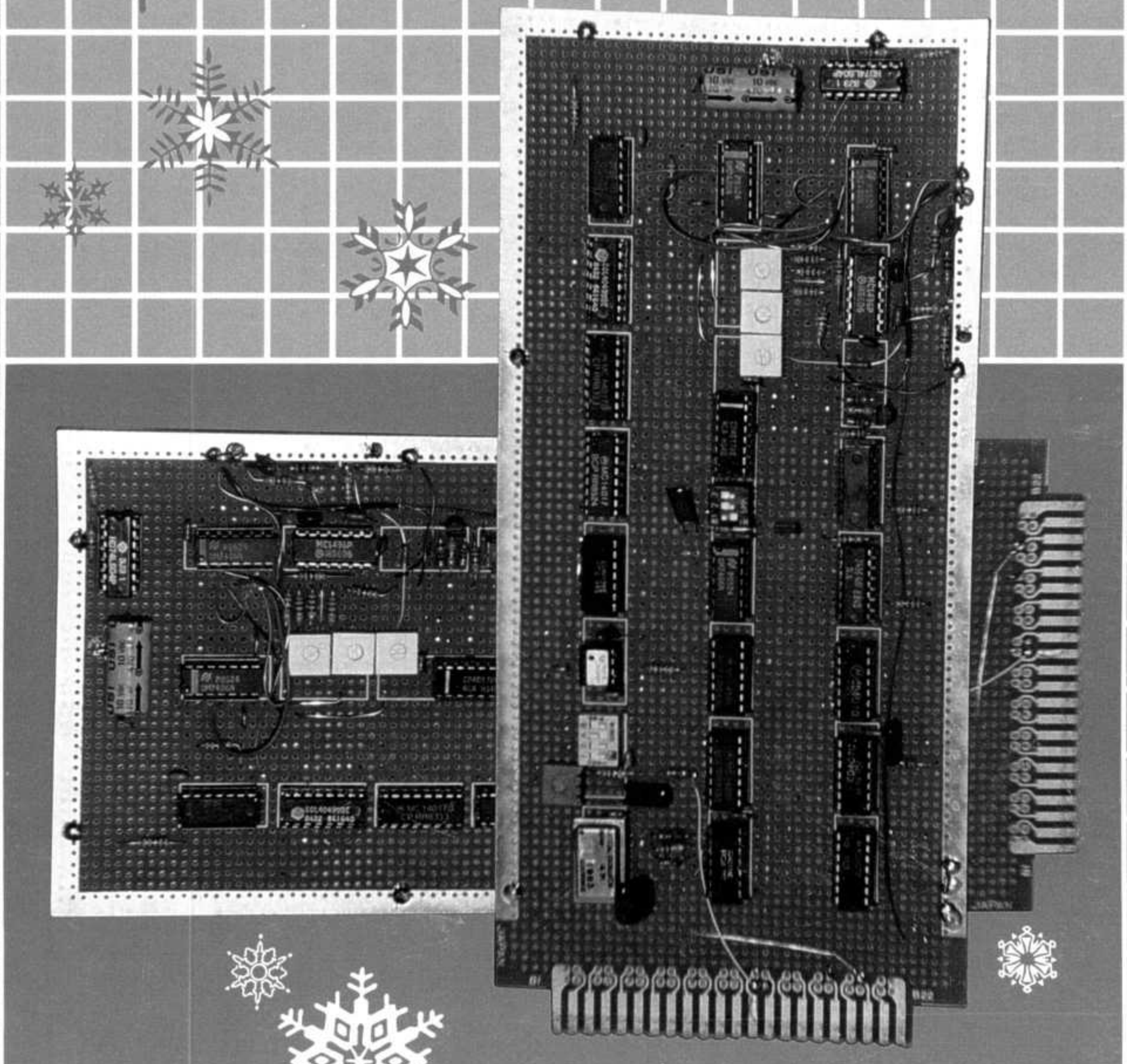

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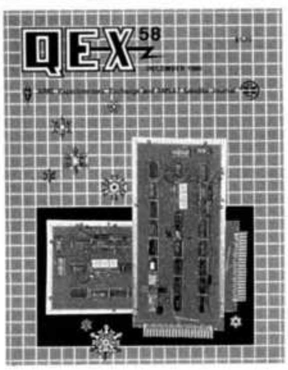

ABOUT THE COVER

On June 1, 1986, the FCC granted Amateur Radio operators authorization to use the spread-spectrum mode for communication. This is one of the experimental direct-sequence spread-spectrum circuit boards. See page 5 for complete circuit information.

Purposes of *QEX*:

1) provide a medium for the exchange of ideas and information between Amateur Radio experimenters

2) document advanced technical work in the Amateur Radio Field

3) support efforts to advance the state of the Amateur Radio art.

All correspondence concerning *QEX* should be addressed to the American Radio Relay League, 225 Main Street, Newington, CT USA 06111. Envelopes containing manuscripts and correspondence for publication in *QEX* should be marked: Editor, *QEX*.

Both theoretical and practical technical articles are welcomed. Manuscripts should be typed and double spaced. Please use the standard ARRL abbreviations found in the 1985 and 1986 ARRL Handbooks and in the January 1984 issue of *QST*. Photos should be glossy, black-and-white positive prints of good definition and contrast, and should be the same size or larger than the size that is to appear in *QEX*.

Any opinions expressed in *QEX* are those of the authors, not necessarily those of the editor or the League. While we attempt to ensure that all articles are technically valid, authors are expected to defend their own material. Products mentioned in the text are included for your information; no endorsement is implied. The information is believed to be correct, but readers are cautioned to verify availability of the product before sending money to the vendor.

Empirically Speaking. . .

Space Symposium in Dallas

There is an electric feeling in the air when a group of high techies meet to talk about their projects. That certainly happened at the AMSAT-sponsored Amateur Radio space symposium held over the November 7-9 weekend in Dallas. Many came to tell others about their current work and that of the past year. Jan King, W3GEY, was there to talk about 1990 and beyond; his subject: Phase 4—a proposed geostationary satellite system.

Jan recently completed an engineering study for Phase 4. He concluded that the next step into space for the Amateur Satellite Service should be satellites that stay put and can be used on a regular schedule. Also, if Phase 4 were to serve only the experimenters and weak-signal VHFers (OSCAR satellites' traditional user base), then funding sources would be insufficient to boost the dollars into the 2 or 3 million dollar range needed. This order of magnitude of funding is needed to pay for the construction and launching of a more-massive spacecraft than previously tried by AMSAT. For a satellite to stay in geostationary orbit, a propulsion system and fuel are needed for station keeping. To have the signal go 22,300 miles and arrive on the ground with good signal strength, the transmitted power needs to be about 10 dB higher than that of OSCAR 10. A larger electrical power source is thus required.

A geostationary orbit and better signal make Phase 4 useful for public-service applications. It is from this realm that Jan expects to find the interest that will generate the additional funding needed. Public-service communications is permitted in the United States under FCC rules for the Amateur Radio Service. We have developed various forms, such as disaster communications, Amateur Radio in the classroom, and routine third-party message relay. According to the ITU Radio Regulations:

> "2733 (2) It is absolutely forbidden for amateur stations to be used for transmitting international communications on behalf of third parties.
>
> "2734 (3) The preceding provisions may be modified by special arrangements between the administrations of the countries concerned."

The United States has third-party traffic agreements with a number of countries. A current list appears on page 51 of the November, 1986 issue of *QST*. Most of the countries in the list are in the Western Hemisphere or IARU Region 2, which has had its share of disasters and knows how valuable Amateur Radio can be in such instances. The other Regions, in general, see Amateur Radio as an experimental and operator-training service, and tend to call upon hams for help only during severe communications outages caused by natural disasters. To say the least, funding sources and their motivations will vary greatly by country. The regional effects will be different for each of the three geostationary satellites needed to provide worldwide coverage.

After Jan explained his Phase 4 concept and how much it would cost, he conducted a survey by show of hands of roughly 60 people who attentively sat through the briefing. Everyone in the audience, with perhaps a few nonrespondents, raised their hands in favor of going for a geostationary satellite system—higher cost included. No one raised their hands in favor of either the status quo or throttling back the satellite effort.

It is one thing for Jan and his team to put together an engineering concept that makes sense. It is quite another thing to translate that into money and action. There is a great deal of work to be done in the next year or so on the system definition. Then, once the parameters are agreed, the various engineering teams need to translate concepts into hardware and software in order to have everything fit together and work well before launch time. If you would like to be part of the Phase 4 engineering team, please write Jan immediately at the AMSAT address given below.

For those who missed the Dallas meeting, but would like the proceedings, they are available at $10 per copy from AMSAT, PO Box 27, Washington, DC 20044.—W4RI

Correspondence

How to Convert A Frequency Counter to A Velocity Measuring Device

Let's refer to the July 1986 Correspondence column—Mr Ennis already has a velocity meter in his frequency counter. A dimensional analysis shows this:

K7FC's frequency counter measures Hertz. Some of us older amateurs remember when it was called cycles per second. But, what is a cycle? It is the rotational distance 2*pi radians, or in this case, the distance traveled as an electrical phase per unit of time elapsed. This is the definition of speed. Specify the direction of travel and the scalar changes into the vector quantity of velocity.

Mr Ennis can easily obtain a direct reading speedometer by arranging for the runners to open or close a switch each time they increment a foot/meter. Trip wires or light beams come to mind.

If such a scheme is not realizable, there are consumer-grade, hand-held radars available which are commonly used in collegiate and professional athletics. K7FC could feed the return signal to his counter and measure the Doppler shift, but there would need to be signal conditioning and scaling.—*Alan Horowitz, KZ1Y, SAIC, Systems Engineer, C3 Systems Div, 255 Hudson Rd, Stow, MA 01775*

Calculate First

I read Jacob Schanker's, W2STM, article on inductance and capacitance magnification with interest (issue no. 56). I've used this technique many times. Keep in mind, however, that currents or voltages are also magnified by this procedure. In transmitting applications, particularly when large values of magnification are used, you may exceed current or voltage ratings of components at even modest power levels. Looking at the capacitance magnification example given ($L = 4.79\ \mu H$ and $C = 1000$ pF), the reactance of the series combination is $-j34$ ohms. This is about equal to a 2600-pF capacitor at 1.8 MHz. The actual reactance of the 1000-pF capacitor is still $-j88$ ohms at 1.8 MHz, however. The voltage appearing across the 1000-pF capacitor will be 2.6 times greater than what would appear across a 2600-pF capacitor for a given amount of current. A similar analysis will show that inductance magnification can result in high currents through an inductor. So, before designing and building that new amplifier or antenna matching network based on magnification principles, it is a good idea to calculate the expected currents or voltages to make sure no components are overstressed!—*Joe Fleagle, WØFY, 320 Green Trails Dr So, Chesterfield, MO 63017*

New Hardware Runs Old Software

A note of encouragement for those who have not had success with the hardware and software published in Thomas Strohl's, KA1VW, article, "Complete RTTY for the Timex," (issue no. 31, Sep 1984). After repeated attempts over a long period of time, my system began to work after a third Z80A® -SIO/Ø was purchased and installed. My first two devices were defective and only produced gibberish on the screen. With the addition of the new interface, W1AW and RTTY bulletins are solid copy on my Timex TS-1000.—*R. H. Knaack, Jr, W7FGQ, 11415-28th SW, Seattle, WA 98146*

Audio Standards For Cellular and ACSSB Systems

After several discussions with Jim Eagleson, WB6JNN, regarding ACSSB, I have a suggestion for setting audio standards. Cellular radios in the US must conform to a set of standards referred to as *AMPS* (Advanced Mobile Phone Service). In the United Kingdom, they have standardized on the Total Access Communications System *(TACS)*. These two systems are nearly identical.

The audio specifications define bandpass, pre and de-emphasis curves and companding performance. It is interesting to note that these standards are quite similar to WB6JNN's optimum ACSSB system. Evidently, what is optimum for cellular radio's FM is also optimum for ACSSB. The AMPS audio processing could also be applied to amateur FM and even HF SSB for significant performance improvement with these systems as well.

The advantages of audio processing standardization to the AMPS specs is twofold:

1) The standards are already in print and have been tested in an existing communications system.

2) Several integrated and hybrid component manufacturers are marketing inexpensive devices to satisfy the large demand for audio processing. For example, Signetics offers the NE570 series of compandors which are widely used by cellular radio manufacturers and ACSSB experimenters. Signetics is also designing a filter, pre and de-emphasis circuits and a dual-tone, multi-frequency circuit all on a CMOS IC. This will result in a two-chip solution for comprehensive FM or ACSSB audio transceive processing. In fact, this two-chip solution allows for full radio telephone duplex operation. Other manufacturers are also offering products conforming to the AMPS spec.

If such a standard is adopted, ACSSB experimenters and manufacturers could all have complementary audio process systems. Furthermore, the hardware solutions are inexpensive, simple, low power and available.—*Robert J. Zavrel, Jr, W7SX, 707 Borello Way, Mountain View, CA 94041*

Bits

Vector Expands Its Line of Prototyping Cards

The Vector Electronic Co has expanded its line of IBM® prototyping cards with the addition of the 4617-3. Designed for use with an IBM PC, AT, the 4617-3 has a pad-per-hole layout pattern with 0.042-inch holes on 0.1-inch centers.

This pattern simplifies the development of specialized circuitry and is particularly efficient for use with designs that require high component densities.

This board, with its plated-through holes, accepts any width DIP IC or Vector wire wrapping terminals. Power and ground buses are located on each side around the edges of the card and terminate to connectors. Connector pads on the board and accompanying bracket accept DB9, 15, 25 or 37-pin I/O connectors. The 4617-3 plugs into any open slot in the IBM AT. For further information, write the Vector Electronic Co, 12460 Gladstone Ave, PO Box 4336, Sylmar, CA 91342-0336; tel 818-365-9661.—*KA1DYZ*

HF Packet: Comments and Tips

By Gregory P. Latta, AA8V
438 Eastland Ave
Akron, OH 44305

Anyone who has operated packet radio on the amateur HF bands is aware that problems exist with this mode of communication. The issue causing the most concern is the number of retries necessary to transmit correct data to a receiving station. It is in excess of what it should be.

Fortunately, this is not an inborn trait of HF packet operation. Rather, the problem is a result of packeteers operating their equipment incorrectly. Here are some suggestions that may help to improve the situation.

Poor Modems

One cause of excessive retries on the HF bands is that many people use the modem that came with their TNC. Most modems built into TNCs are designed for VHF and consist of little more than a simple bandpass filter and a phase-locked loop demodulator. Such a unit works fine for VHF operation, where there may be little noise or fading. However, the modems are unsuitable for HF packet operation at 300 baud. In fact, many perform poorly on RTTY at 45.5 baud as well.

Manufacturers have been slow to realize the modem-TNC crisis. Several companies, however, have begun to respond by producing high-quality modems designed for HF operation. Note that there is nothing new in these modem designs. They are essentially the same proven designs that have been around for years in RTTY service. The modems are modified for the slightly different shift used in packet and a carrier-detect circuit is added. The modems are usually designed with a particular TNC in mind so they are easy to connect to. Because some modems are nothing more than a high-quality RTTY modem, the shift of most RTTY modems can be modified easily. In fact, the difference in the shift (30 Hz) is so slight that the unit may work satisfactorily in its unmodified state. In addition, carrier-detect circuits are simple and easy to add to most TUs.

The moral of this story is that if you plan to operate HF packet, get the proper modem! If you use the TNC's built-in modem, be sure it is designed for HF operation.

Timing Parameters

Improper setting of a timing parameter is another cause of excessive retries on HF packet. Many operators don't understand the importance of proper timing. Setting these timing parameters correctly can literally make or break a packet radio link. Let me illustrate this with some examples.

FRACK

FRACK is a parameter that controls the FRame ACKnowledgement time. It is a little known quirk of the TAPR-1 firmware (Heath, the first AEA TNCs and other TAPR-1 clones have it, too). A TNC begins its timing procedure when transmission of a packet begins. During this period, the TNC waits for receipt of an ACK.

If the FRACK time elapses before an ACK is received, the TNC dumps the packet into the transmit buffer. Once the packet is placed there, it will be transmitted. Even receipt of the ACK one bit later than the FRACK time will not prevent the buffer from sending its data.

HF channels are busy because of varying propagation and lower baud rates (one fourth that of VHF packet). Both factors tend to prolong the time it takes an ACK to get back to the sending station. Because it takes four times longer to send data, FRACK's timing must be increased significantly from the notch it was set to for VHF operation. If this is not done, a situation arises where the ACK time is less than the time it takes to send the packet. Since timing starts concurrently with the packet that is being sent, the TNC expects to receive the ACK before the packet is finished being sent! The result can be an infinite number of retries. This is a common problem with excessively long packets. If you are getting excessive retries for no apparent reason, try increasing FRACK.

TXD

TXD describes the time it takes for the transmitter to key up before data is sent. Here again, this parameter should be increased from its VHF setting; HF rigs generally take longer to stabilize on transmit. Also, because FSK and not AFSK is used to transmit the signal, automatic gain control (AGC) is almost always used when receiving the signal. This helps to keep the signal level constant. Once a signal is received, some AGC circuits require a settling time. If this circuit does not stabilize, nonreceipt of one bit will result in an unnecessary retry.

If you are using AGC on your receiver, and it has several settings, set it to CW, FAST or to the minimum time constant possible. This setting helps the AGC circuit to stabilize faster. When transmitting a packet, assume the other operator's receiver is slow. Set the TXD a bit more than you may think is necessary. It helps in the long run.

Other Parameters

PACLEN

PACLEN is a parameter that is set on the TNC and it means packet length. A PACLEN should not be greater than about 60! The HF bands are subject to noise, QRM and rapid fading. Even with a good modem, a fast AGC, and a good path, it is unreasonable to expect the path to hold up consistently for the time it takes to send a long packet at 300 baud. If you think that sending a long packet and "hoping it will make it" is the answer to successful HF operation, forget it. At 300 baud, the band has only to drop out or fade for 3 ms and a bit will be lost. That means another retry!

MAXFRAME

MAXFRAME is the number of unacknowledged packets that can be enroute at any given time. If too many packets are outstanding, the TNC will not transmit any more data. MAXFRAME is typically 4 on VHF. On HF, it should be 1 or 2. If the path is particularly good, it can be increased to 3 or 4. The trouble with an excessive MAXFRAME setting (especially with excessive PACLEN) is that one station can overtake the frequency for long periods of time. With MAXFRAME kept to 1 or 2, and PACLEN limited to say 60, the length of each packet is kept to a minimum. (Transmitter "on time" is important on HF. With different propagation, more stations may be heard than if you were listening to a local VHF network.)

Final Comments

There is more to getting on HF packet than simply recalibrating the on-board modem, setting Hbaud to 300 and connecting everything to the rig. HF packet can be a successful, reliable mode, but only if it is approached intelligently. Among other things, thoroughly read the instruction manual that came with your TNC. Most manuals have a section dedicated to on-the-air operation. To enjoy the HF packet mode and help eliminate QRM, obtain a good understanding of how the TNC parameters affect operation.

The above recommendations are for normal QSO type contacts. Some of the parameter settings should be modified during file transfers, when using bulletin boards and other specialized operations. However, from on-the-air experience, these suggestions and settings should be correct for the average amateur. See you on the air!

Experimenting With Direct-Sequence Spread Spectrum

By Andre Kesteloot, N4ICK
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Introduction

During the past several years, AMRAD (Amateur Radio Research and Development Corp) has been conducting experimentation with spread spectrum technology.[1,2] As a member of the group, I decided to build a board that would demonstrate some practical aspects of direct-sequence spread spectrum (DSSS). I chose a carrier frequency of 2 MHz so that the phenomena could easily be displayed on any low-frequency (5-MHz bandwidth) oscilloscope likely to be available to the average radio amateur. The equipment, which can transmit and receive data, test square waves and Morse code signals, was demonstrated at the September 1986 AMRAD meeting in Vienna, VA.

Description

Referring to Fig 1, it is fairly easy to conceive of an amateur spread-spectrum communication link if an existing external reference, such as a local TV or radio transmitter, is used.[3] In this experiment, the external reference is a 4-MHz oscillator. A 2-MHz carrier is obtained by dividing the reference frequency by 2; this carrier is also used as clock for a pseudo-random generator. Data is XORed with the output of the pseudo-random generator and the result is XORed with the carrier. This produces a binary phase shift keyed direct sequence spread spectrum (BPSK DSSS) signal. At the receiving site, a pseudo-noise sequence similar to that created at the transmitter, is generated and synchronized to the original one. By mixing this sequence with the reference carrier, it is possible to extract the original data.

In my demonstration model, points T and R were connected to each other with a jumper. In a practical application, point T would be connected to an antenna via an amplifier; point R would similarly be connected to a receiving antenna, AGC amplifier and the like.

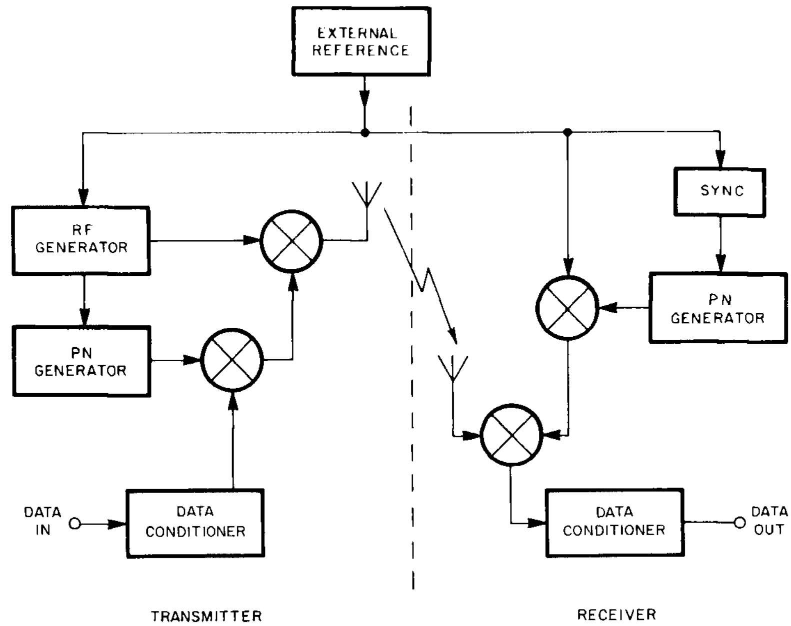


Fig 1—A block diagram of a possible radio amateur's spread-spectrum communications link. The external reference signal can be received from a TV or radio transmitter.

Circuit Operation

Reviewing the transmitter stage in Fig 2, U1 is a 4-MHz DIP oscillator that is used as the reference frequency. U2A, a 4013 D flip-flop divides this frequency by 2 to produce a 2-MHz carrier (referred to as CA). The 2-MHz frequency is further divided by 1,000 (through three consecutive 4017 divide-by-10 stages, U4, U5 and U6) to produce a 2-kHz square wave. This signal is used as a test square wave (Figs 3, 4 and 5) and also serves as clock for another 4013 D flip-flop, U2B.

Under normal conditions, the Morse key is open, the D input of U2B is low and its $\overline{Q}$ output is low. When the key is pressed, the D input goes high. At the next clock pulse $\overline{Q}$ goes to +5 V. When the key is released, the D input goes low again, and at the next clock pulse, $\overline{Q}$ also goes low. Thus, the data (in this case the Morse sequence) is now synchronized to the 2-kHz clock, itself a submultiple of the 2-MHz carrier. This synchronized operation ensures phase-coherent switching. The $\overline{Q}$ output of U2B is our data, or DA.

The 2-MHz carrier (CA) is also used to clock U7, a 74164 7-stage linear-feedback shift register connected as a pseudo-random noise generator (PN), with feedback taps at outputs 1 and 7. Feedback is applied using an XOR stage U8B, while U8A is connected as an inverter. (This is one of the FCC-authorized PN sequences. For a more complete study of this circuit, see note 3.)

The PN sequence is available at pin 2 of U7 and is called the original code or OCO. It is modulo-2 added to the data output of U2B in an XOR stage, U8C, the output of which is OCO × DA. This signal is XORed with the 2-MHz carrier (CA in U8D) to produce the output signal OCO × DA × CA, a BPSK DSSS signal. (In reality, this signal would be applied to an antenna using a power amplifier. On my experimental board, it is simply jumpered to the input of the receiver.)

[1]Notes appear on page 9.

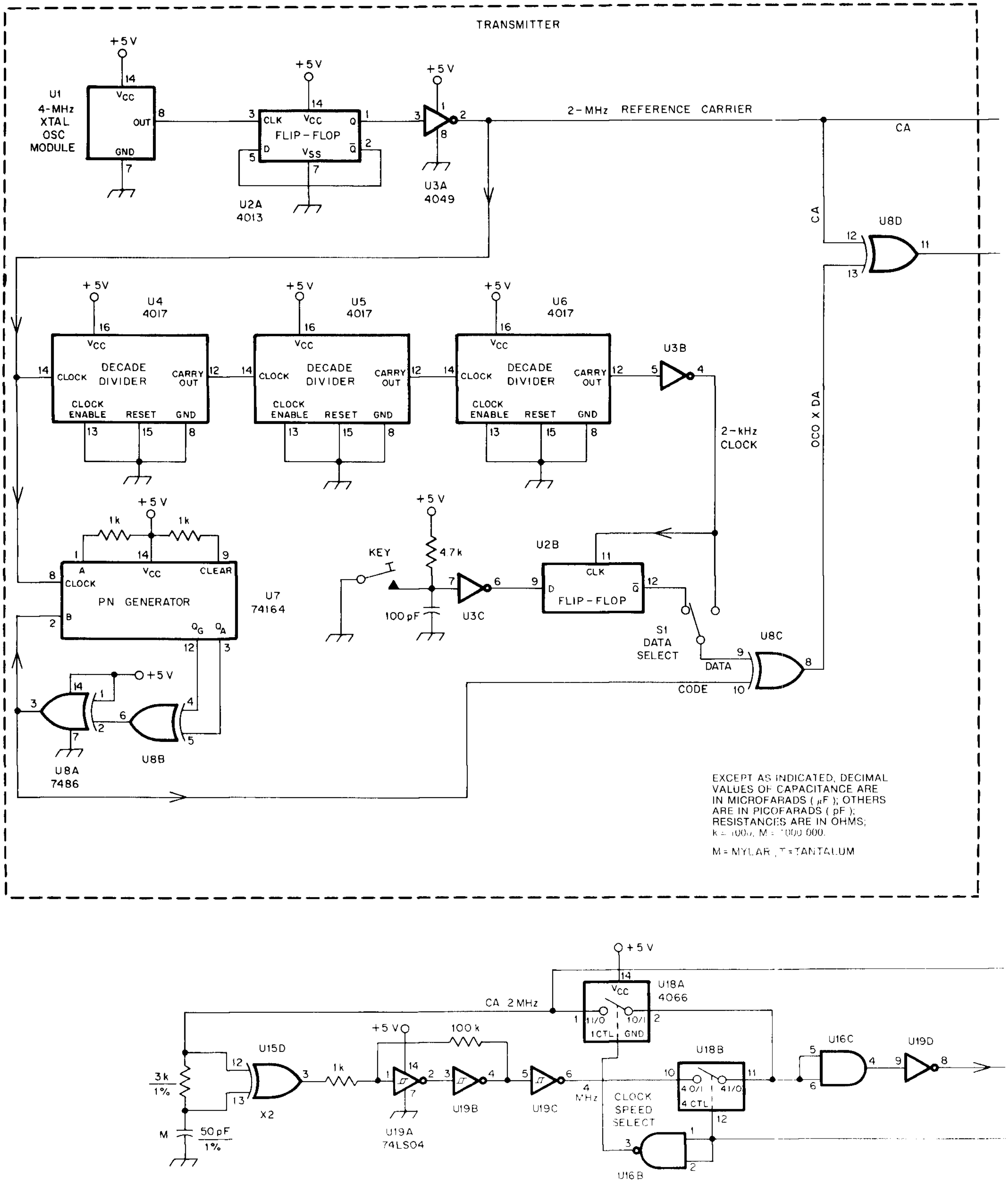
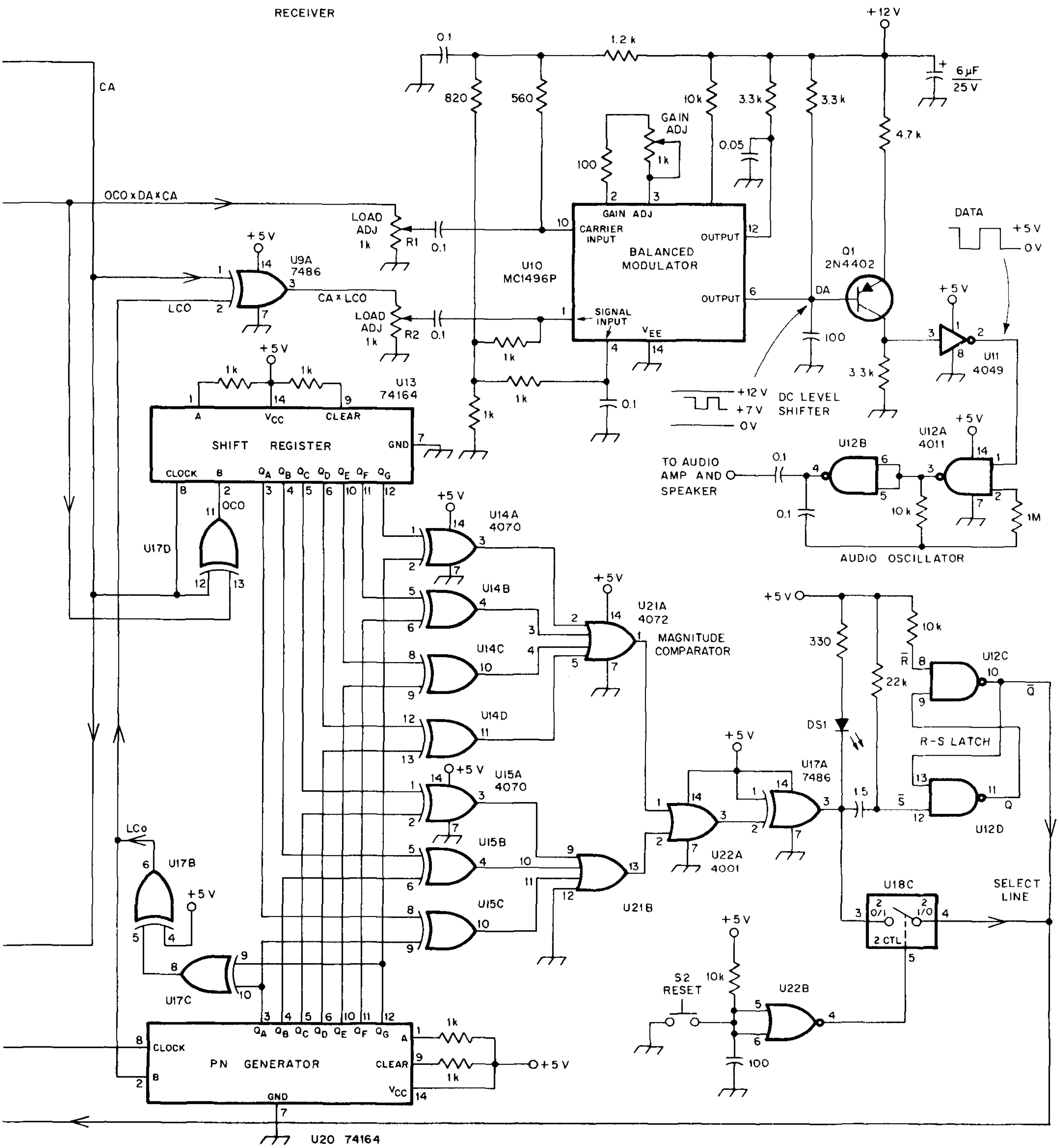


Fig 2—A complete schematic digram of the DSSS transmitter and receiver stages. Operation of the circuit is explained in the text.

Q1—2N4402, PNP transistor.
S1—SPDT switch.
S2—Normally open push-button switch.
U1—4-MHz DIP crystal oscillator.
U2—CD4013BC dual D flip-flop.
U3, U11—CD4049UBC hex inverting buffer.
U4, U5, U6—CD4017BC decade counter/divider with 10 decoded outputs.
U7, U13, U20—74164 8-bit serial-in parallel-out shift register.



U8, U9, U17—7486 quad 2-input exclusive OR gate.
U10—MC1496 balanced demodulator.
U12, U16—CD4011 quad 2-input NAND B series gate.
U14, U15—CD4070BC quad 2-input exclusive OR gate.
U18—CD4066BC quad bilateral switch.
U19—74LS04 hex inverter.
U21—CD4072BC dual 4-input OR gate.
U22—CD4001BC quad 2-input NOR buffered B series gate.

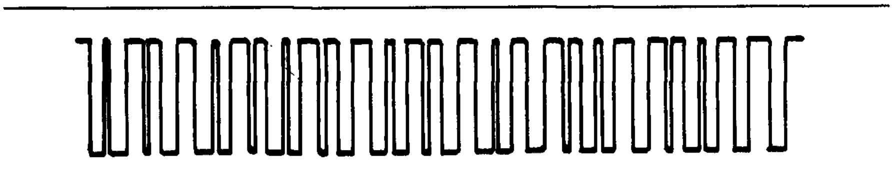


Fig 3—A DSSS carrier is shown at the jumper between the transmitter (output of U8D) and receiver. The scanning rate is 1 µs/div.

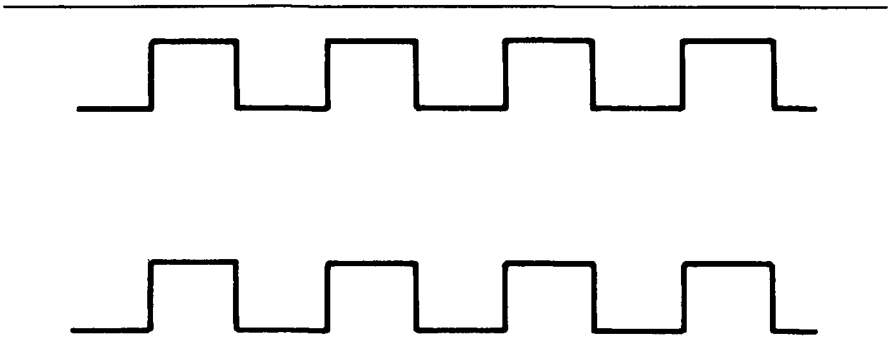


Fig 4—The upper oscilloscope trace shows a 2-kHz test square wave before transmission at the output of U2B. The lower trace is the same signal after decoding at the receiver, at the output of U11A. There is no discernible difference between the two signals. The scanning rate is 0.2 ms/div.

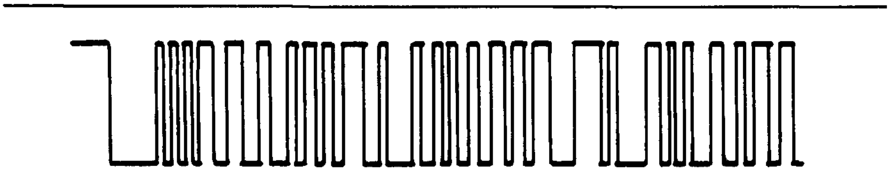


Fig 5—The output of PN generator U7, at pin 2. The scanning rate is 5 µs/div.

Reviewing the receiver stage, we can see that the input signal (OCO × DA × CA) and CA (the carrier derived from the external reference) are both applied to the inputs of an XOR stage, U17D. At the beginning of a transmission, no data is transmitted and the result of XORing (OCO × CA) and CA is thus OCO, the original code sequence. This sequence is serial-loaded into U13, a 7-stage shift register clocked by CA.

Another 74164 shift-register stage, U20, is connected as a PN generator using the same pseudo-noise sequence as the one used in the transmitter. (In Amateur Radio spread spectrum, it is mandatory to announce *in clear* the kind of register used. This enables both operators of the radio link to set their PN generators to the same sequence.) When the receiver is turned on, U20 is clocked at twice the normal 2-MHz rate. This 2 × clock is generated by applying the 2-MHz CA to U15D, a multiply-by-two stage, the output of which is shaped by a Schmitt trigger (U19A, U19B). U18A and U18B (part of a 4066-quad switch) are connected as a SPDT switch. U18B is normally closed, U18A is normally open and the 4-MHz clock signal is applied to the PN generator U20.

Since U8 and U20 use the same feedback taps, they produce the same random sequences. Our goal is to synchronize both signals so that they will be in phase. As was discussed earlier, when no data is transmitted, the output of U13 is OCO, the original code sequence. The words produced by U13 and by U20 are compared, bit for bit, in a magnitude comparator comprised of XOR stages U14A, B, C, D, U15A, B, and C, quad input OR gates U21A and B and NAND gate U16A. As U20 is clocked at twice the rate of U13, its output tries to "catch up" with that of U13. There will be one instant when both output words are equal bit for bit. At that time, all outputs of the seven XOR stages will be low and the output of U16A goes high, driving the output of U17A low. This illuminates the "lock" LED and drives the $\overline{Q}$ output of U12C low. In turn, U18B opens and U18A closes, thereby switching the clock rate of U20 from 4 MHz to 2 MHz. From then on, both U7 and U20 will produce the same PN sequences, will be clocked at the same frequency and will be in phase (OCO = LCO). The edge-triggered RS latch (U12C, D) prevents further changes at the output of the magnitude comparator from influencing the clock speed, something that would otherwise happen whenever data is received.

Loss of sync between OCO and LCO can be simulated by momentarily grounding pin 9 of U7. OCO and LCO will now be out of phase. When this occurs, press the reset button—in the absence of data, or when data is low, the magnitude comparator output is compared to the clock speed selector (by bypassing the RS latch through U18C) to resynchronize LCO to OCO.

If Morse code is sent, for instance, notice that the lock LED extinguishes while the data is being transmitted. This occurs because the code now extracted from the DSSS transmission is no longer in phase with either OCO or LCO. The latter two, however, are still in phase, and as soon as the key is released, the LED lights again.

The carrier (CA) and the LCO are now XORed in U9A, the output of which is CA × LCO. This signal is applied to one of U10's inputs, a MC1496P/1596 balanced demodulator.[4] The other input is fed with the received DSSS signal (OCO × DA × CA). Since OCO = LCO, the output of the demodulator is the data DA. (To adjust the three potentiometers associated with U10, an oscilloscope should be connected to U10, pin 6. Both input potentiometers should be adjusted for no overloading, and the 1-kΩ potentiometer between pins 2 and 3 can usually be left in the minimum resistance position to provide maximum gain for this stage.) As U10 operates from + 12 V, a dc voltage shifter composed of Q1 and U11A is used to shift the data to TTL levels. In my experimental board, this data is then used to gate ON and OFF an audio oscillator (U12A and B) that is connected to a small outboard audio amplifier and speaker.

Conclusions

Using components and test equipment readily available to the average radio amateur, it is possible to experiment with DSSS and gain hands-on experience and a better understanding of the fundamentals of this newly authorized technology. No attempt was made to optimize this demonstration board. The use of some circuitry was dictated by the availability of a particular IC, or because a portion of a quad gate was yet unused. No doubt several gates could have been saved, other more elegant ways of achieving the same results could have been developed and additional circuit simplifications could have been carried out. However, this was never meant to be a finished product, but simply a test bed for ideas. The reader is invited to dream up their own refinements.

If some quad-gate ICs were shared between the transmitter and receiver sections on the prototype board, they were renumbered (for clarity) and shown as separate ICs in Fig 2. (Incidentally, the 555 timer, located at the bottom left of the board [cover photo], is an adjustable clock generator only used in some early experiments. It is not shown on the schematic.) The three square potentiometers near the center of the board are associated with demodulator U10. The reset switch is mounted on the rear of the board and is not shown in the photograph.

To build a radio link for 440 MHz, XOR stages U8C, U8D, U9A and U17D would have to be replaced with doubly-balanced mixers such as the SBL-1 from Micro-Circuits, or the M53T manufactured by Magnum Microwave.[5,6] You'll need a reference carrier higher than 440 MHz, and for that purpose a local UHF-TV transmitter carrier comes to mind. At both amateur sites, a free-running oscillator could be used at approximately the reference frequency and a phase-lock loop (PLL) to lock this oscillator to the external reference TV transmitter would be necessary. This arrangement is equivalent to the CA on my demonstration board.

In the experiment described earlier, no preamble signal was sent for synchronization purposes. Sync was acquired before data was transmitted. On 440 MHz, it may be advisable to send a short preamble to synchronize the divider chain from which the 440-MHz carrier will be derived.

Finally, to transmit audio sine waves rather than square waves, it should be relatively easy, after raising the low clock rate from 2 kHz to 20 kHz, to apply that new clock and the audio sine waves to a conventional 555 pulse-width modulator, for example. To raise the clock rate, lower the divider ratio from 1/1,000 to 1/100 by eliminating one of the 4017 divide-by-ten stages.

Further Reading

Radio amateurs who reference readily available literature on spread spectrum will find ample mathematical treatment, plenty of block diagrams and a remarkable paucity of detailed schematic diagrams or specific information on hardware. Chapter 21 of *The ARRL 1986 Handbook for the Radio Amateur*[7] offers a good theoretical background on the subject. The articles by Scholtz[8] and Pickholtz et al[9] should also be referenced. Details of prototype realization are available in a paper by Van Der Gracht[10] and to a lesser extent in an article by Maskara and Das.[11] Finally, anyone contemplating involvement in spread spectrum should have access to the fundamental book by Dixon.[12]

Acknowledgements

I am grateful to AMRAD members Hal Feinstein, WB3KDU, and Mike O'Dell, N4NLN, for their suggestions and encouragements.

Notes

[1]H. Feinstein, "Spread Spectrum, Frequency Hopping, Direct Sequence and You," *QST*, Jun 1986, p 42.

[2]A. Kesteloot, "Practical Spread Spectrum: A Simple Clock Synchronization Scheme," *QEX*, Oct 1986, p 4.

[3]See note 2.

[4]R. Hejhall, "MC1596 Balanced Modulator, Application Note AN-531," Motorola Semiconductor Products, Inc, Phoenix, AZ 1982.

[5]Micro-Circuits Co, Inc, RR 1, PO Box 518, New Buffalo, MI 49117, tel 616-469-2727.

[6]Magnum Microwave Corp, 365 Ravendale Dr, Mtn View, CA 94043, tel 415-968-9281.

[7]M. Wilson, ed., *The ARRL 1986 Handbook for the Radio Amateur*, Newington: ARRL, 1985.

[8]R. Scholtz, "The Spread Spectrum Concept," *IEEE Transactions on Communications*, Vol Com-25, No. 8, Aug 1977.

[9]R. Pickholtz, D. Schilling and L. Milstein, "Theory of Spread-Spectrum Communications—A Tutorial," *IEEE Transactions on Communications*, Vol Com-30, pp 855-884, May 1982.

[10]P. Van Der Gracht and R. Donaldson, "Communication Using Pseudo-Noise Modulation on Electric Power Distribution Circuits," *IEEE Transactions on Communications*, Vol Com-33, No. 9, Sep 1985.

[11]S. L. Maskara and J. Das, "Concatenated Sequences for Spread Spectrum Systems," *IEEE Transactions on Aerospace and Electronic Systems*, Vol AES-17, No. 3, May 1981.

[12]R. Dixon, *Spread Spectrum Systems*, New York; Wiley, 1976.

Bits

Communication Equipment Protection

With a host of communication equipment using solid-state devices, voltage-line surges and interruptions can cause havoc with computer files and other data transmission systems. Many manufacturers have taken a serious look at this problem and each have marketed their own protective devices. The following protective products are only a few of the many that are available.

Modem and Portable Computer Protection

Electronic Specialists, Inc has expanded their computer protection line to include the Modem/Power/Static Pac. This device is a complete protection package for the traveling portable computer. It has combined broadband ac power filtering, extended range spike suppression, modem RF filtering, modem spike suppression and a static discharge plate. Power is available from a conventional 3-prong outlet and a CEE-22 universal portable computer power connector. A 6-foot power cord is provided and modem connection is through standard modular RJ-11 connectors. Model MPS-2(22) sells for $184.95.

Kleen Line Modem protection is available for 4- and 8-pin telephone modular connectors. The system (Model PDS-11/SUP) uses modern semiconductor, metal oxide varistor and gas discharge tube suppression techniques to suppress damaging telephone and power line spikes. Model PDS-11/SUP sells for $109.05.

A 40-page color catalog describing communication equipment protection is available from Electronic Specialists, Inc, 171 S Main St, Natick, MA 01760; tel 617-655-1532. The catalog describes numerous standard, off-the-shelf products, custom designs and hot-line problem solving. Ask for catalog 861; it is available free of charge.

New Conditioned Uninterruptible Power System is a Line-Saver®

Kalglo Electronics Co has added a new standby uninterruptible power system to its existing line of power conditioning equipment. Designated the Line-Saver, Model LS500, the unit is applicable to home or small business computer market situations. It is engineered to give trouble free standby back-up power available in 117/234 V, 60/50 Hz, with 500 W capacity.

The unit uses pulse width modulation (PWM) technology to regulate the RMS ac output voltage for greater efficiency to various load conditions. The PWM ac output increases battery efficiency which in turn increases battery back-up time. The unit is also furnished with an internal 24 V sealed rechargeable battery, two Spike-Spiker® voltage surge protected and EMI/RFI filtered ac outlets, audible and visual power failure warning system, test mode indicator and switch, replaceable external fuses, compact in size with external 24 V dc battery connectors to allow for mobility and extended hold-up time.

The Line-Saver weighs 27 lbs and has a footpoint of 0.64 square feet. The unit comes with a 6-foot detachable 3-prong heavy-duty grounded cord with a CEE-22 connector. Suggested retail price is $795.00. For further information contact Kalglo Electronics, Co, Inc, 6584 Ruch Rd, E Allen Twp, Bethlehem, PA 18017; tel 215-837-0700.—*KA1DYZ*

A Two-Tone Generator With Very Low Distortion

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Translated by George W. Allen, N1BEP†

A two-tone audio signal is commonly used to test the linearity of an SSB transmitter. The amplitude of each signal should be identical, and the signals and their mixing circuitry should be set to produce the least distortion possible. With this in mind, two signals that are not harmonically related should be chosen. This article describes a two-tone generator that produces a 900- and 2100-Hz signal combined so that all harmonics are greater than 60 dB below the level of the desired signal. (The measurements were performed in the laboratory using a spectrum analyzer.)

Circuit Description

The two-tone generator shown in Fig 1 is essentially composed of two classical Wein-bridge oscillators A1 and A2 (see Fig 2), which oscillate at 900 and 1200 Hz, respectively. The nonlinear network of the two diodes assures that enough gain is present to maintain oscillation at a low distortion level. The 10-kΩ resistor in parallel with the diodes serves to reduce crossover distortion. The oscillator circuit has a high output impedance. Any change in the load changes the active point of the diodes, which in turn changes the output amplitude. This phenomenon can be used to control the generator. More on this later.

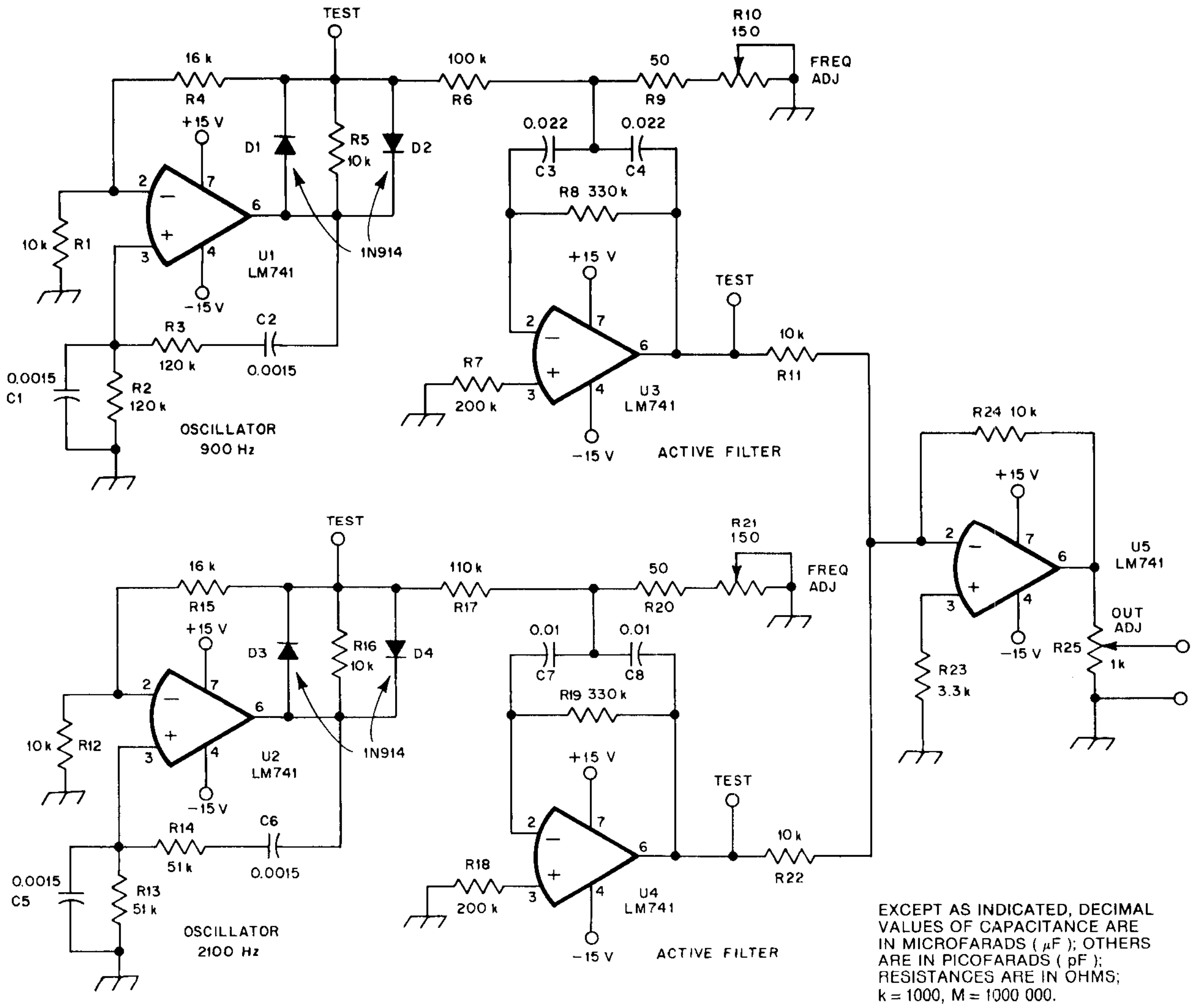

Fig 1—The schematic diagram of the two-tone audio generator.

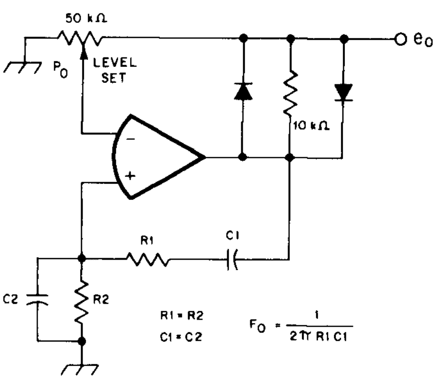

Fig 2—The Wein-bridge oscillator. Two of these oscillators are used in the generator to produce the 900- and 1200-Hz signal.

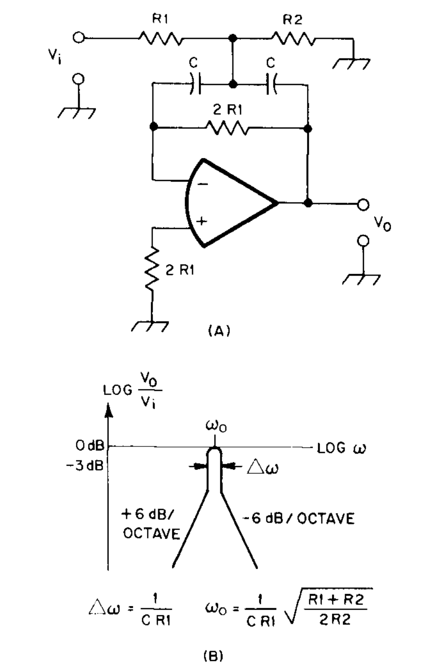

Fig 3—A shows a typical band-pass filter. R2 adjusts the center frequency. B is the filter's peaked response curve. Frequency shifts are made with the bandwidth and frequency held constant.

The unloaded frequency of oscillation will be stable if R1 = R2 and C1 = C2 at

$$F_o = \frac{1}{2\pi R1C1} \quad \text{(Eq 1)}$$

The potentiometer (P_0) is adjusted to where oscillation begins. At that setting, the inverted input to the op amp is at a potential approximately equal to eo/3. When oscillation begins, the diodes start to conduct and their impedance reduces the ratio of negative feedback. Adjusting P_0 varies the output amplitude. Note that in this arrangement, the more the amplitude increases, the more the distortion decreases. Moreover, the action of the paired diodes further reduces the distortion. In my circuit, I have chosen fixed resistor values for P_0 that were determined experimentally.

Reducing Distortion

The simplest and most efficient way to reduce the distortion of the sinusoidal signal is with an active filter. Amplifiers A3 and A4 are set respectively to the frequency of each oscillator to satisfy this condition. Each filter is a band-pass filter with a peaked response curve of that shown in Fig 3. The filter's selectivity is such that an output which is nearly perfectly sinusoidal (Figs 4 and 5) can be obtained. The detailed theory of these filters are discussed elsewhere.[1]

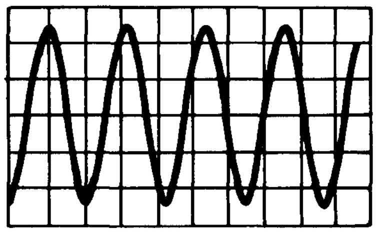

Fig 4—A 900-Hz signal is produced by the active filter. The horizontal scale is 0.5 ms/div and the vertical scale is 1 V/div.

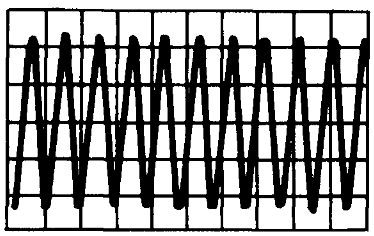

Fig 5—A 2100-Hz signal produced by the active filter. The horizontal scale is 0.5 ms/div, while the vertical scale reads 1 V/div.

With the heart of our generator realized, the two sinusoids must be adjusted in the summing amplifier A5. The inverse input to A5 appears to be at virtual ground; this means that the output of each filter sees a resistance of 10 kΩ with respect to ground. By connecting these components to virtual ground, we are ensured that these filters and their associated oscillators do not interact.

Adjustment

Three adjustments must be performed. P1 and P2 set each filter to the frequency of the associated oscillator. They also set the output amplitude of each filter. As we have seen, changes in the oscillator's load also change the amplitude and the frequency of the signal, but this is not critical. Instead, set P1 and P2 to obtain identical outputs from each filter at pin 6. In my case, I set the levels to four volts P-P. These adjustments are very sensitive and the use of an oscilloscope will be helpful. Those who do not have access to a scope can use an electronic voltmeter to set the amplitudes. Do not exceed four volts P-P or output distortion may result. If everything is correctly adjusted, the output signal should look like that shown in Fig 6. P3 sets the output level.

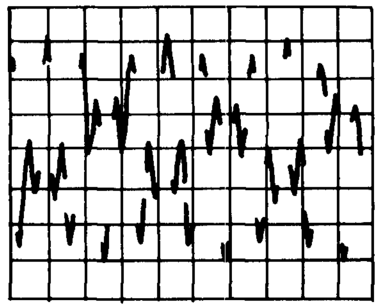

Fig 6—The oscilloscope trace shows a 900- and 2100-Hz signal, both with equal amplitudes.

Construction and Choice of Components

My generator was constructed on a PC board; no special precautions were taken. All components must be 1% tolerance, particularly the capacitors in each filter. The capacitors used should either be polycarbonate, mica, Mylar or polystyrene. Their quality is important for good stability and having a two-tone generator that performs well.

Finally, which ±15 volt power supply is used for the generator is a decision left to you. Many bipolar power supplies that are able to adapt itself to the generator exists. (Don't go below ±15 volts.)

This generator will assist those operators who like to fuss over the adjustment of their linear amplifiers. And, Amateur Radio operators who have access to an oscilloscope or a spectrum analyzer can easily test the two-tone audio generator's performance.

Notes

[1] M. Wilson, ed, *The 1986 ARRL Handbook*, (Newington: ARRL, 1985), p 25-28.

[2] Burr Brown, *Operational Amplifiers, Design and Applications*

[3] National Semiconductor, "Easily Tuned Sine Wave Oscillators," *App Note LB-16*.

[4] Sescosem: *Application Manual for Linear Integrated Circuits*, vol 1.

†G. Pelissier, "A Two-Tone Generator With Very Low Distortion," *Radio-Ref*, Dec 1985, p 772.

Xerox 820-1 Compendium—Part 5

By Dave Borden, K8MMO
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Talking to Fred

In our previous encounter with frequency hopping, the Xerox 820-1 board was hooked to an ICOM IC-2AT handheld transceiver to frequency hop rather slowly (10-100 hops/second).[32] This month we will lay the groundwork for high-speed frequency hopping on HF. Note that the new FCC rules do not allow this, but our AMRAD Special Temporary Authorization (STA) does.[33] Before we can hop cleanly, it is necessary to have a Fred. What is a Fred?

The Fred Williams Synthesizer

In 1984, *QST* published an article on a remarkable synthesizer that produced a clean signal from 1 Hz to 7 MHz, in 1-Hz steps.[34] The synthesizer used a TRW D/A converter and hard-to-get parts. It was designed and built by Fred Williams, an employee of the TRW LSI Products Div. AMRAD members were able to obtain the parts, an EPROM and the schematic to build one. In a subsequent article, *QST* published information on a hand-set frequency controller to allow the synthesizer to be programmed easily.[35] At the same time, a revised schematic was available; it used less expensive parts that were more obtainable. Also, A & A Engineering (2521 W LaPalma Ave, Unit K, Anaheim, CA 92801; tel 714-952-2114) produced kits for both the synthesizer and controller. AMRAD ordered one kit each and Elton (Sandy) Sanders, WB5MMB, constructed the units. AMRAD named the project and synthesizer "Fred" in honor of its designer.

Xerox 820 Talks to Fred

Sandy brought the completed units to my laboratory to demonstrate that they worked. Once convinced that both units were working properly, Sandy constructed a cable to run between the Xerox 820 parallel port and Fred. The details of that cable appear in Fig 7. Note that Fred expects serial programming. The three lines that run to Fred from the computer are responsible for changing the unit to the desired frequency.

- Frequency Load (Transfer Line, "Change Freq Now")
- Transfer Clock (Clock transitions for digit valid)
- Frequency Control Data (Desired Freq—24 bits)

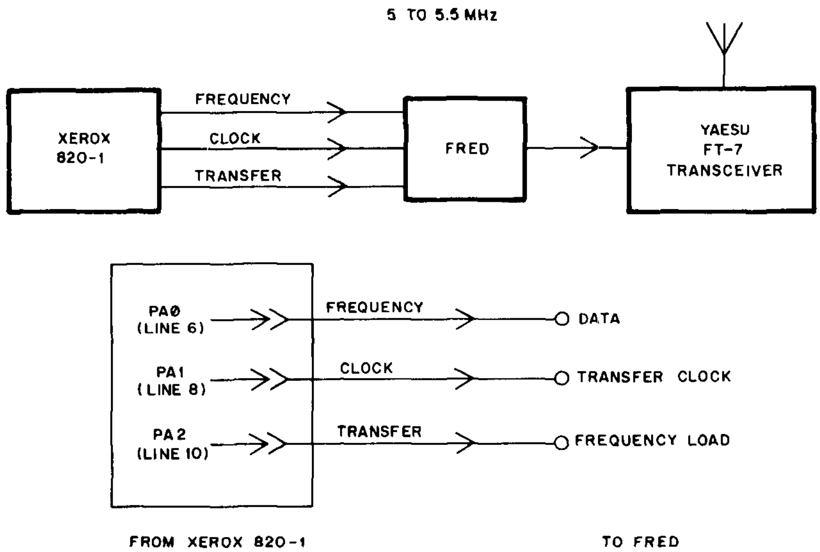

Fig 7—A Xerox-to-Fred-to-Yaesu interface allows the operator to have computer control of the FT-7 transceiver. Refer to the lower drawing: On the frequency line, send the frequency as 24-binary bits, with the most significant bit (MSB) first. Toggle the clock line for each binary digit sent. Lower the transfer line at the beginning of the transfer; raise the line toward the end.

Hal Feinstein, WB3KDU, and I programmed Fred to do several tasks. We first hooked Fred's output to a Yaesu FT-7 HF transceiver. The FT-7 wants a VFO frequency of 5.0-5.5 MHz, just perfect for Fred. The FT-7's socket that accepts the external VFO resides on the back of the rig. The socket looks like a 5-pin DIN, but it is not. The coaxial cable from Fred was soldered to the FT-7 and the power to the internal VFO was cut. Using the handset controller that Sandy built from the kit, we had frequency control of the FT-7 and could move up and down the band. We quickly unplugged the controller and plugged in the Xerox 820. The program in Table 1 controls Fred; let's start in a simple manner. Enter the 24 binary digits (1 or 0). When the 24th digit is typed in, Fred will change to the new frequency and the FT-7 will hear whatever is on frequency.

PACANSWR

The basic complaint about packet bulletin board systems (PBBS) is that users interact with other local PBBS computers on 145.01 MHz. This causes problems because the computers are channel hogs. Computers and human typists do not share the same frequency well; the computer always wins. Therefore, I offer some suggestions: (1) Marry the AMRAD frequency agile ICOM-2AT transceiver with the packet radio TNC code from Phil Karn, KA9Q, and produce the Frequency Agile Bulletin Board; (2) The PACANSWR machine.

For years, people have used telephone answering machines. Many people dislike them, but they help to screen calls. Thanks to Ward Christensen's MODEM program, many people now leave their TNCs in the capture mode (capture all traffic on disk during a 24-hour period). If your TNC is set for the monitor mode, you will receive all traffic on disk, too. If you disable the monitor mode, then traffic sent to you exclusively is received only. The problem? Callers who know you have a capture mode will leave a message, others won't.

What is needed is a packet answering machine. This device informs the caller that they have reached my station, but that I am busy and they can leave a message.

Let's take the easy way out. Disable the

32Notes appear on page 15.

Table 1

Frequency Set Program in BASIC

```
10 REM HANDSET BY XEROX 820 KEYBOARD FRED OUTPUT
20 REM ARRAY OF 24 BINARY DIGITS FOR FREQUENCY
22 DIM D(24)
25 PRINT "BEGINNING FREQUENCY SET OF FRED"
26 REM BEGIN BY INITIALIZING PARALLEL PORT
27 REM SET PORT A TO BIT MODE
28 OUT 9,207
29 REM SET PORT A TO ALL OUTPUT BITS
30 OUT 9,0
31 REM DISABLE INTERRUPTS ON PORT A
32 OUT 9,64
33 REM CLEAR ALL LINES FOR ACTION, RAISE TRANSFER
34 OUT 8,4
35 REM THE BIT FREQ SET LOOP, QUIT WITH CTRL-C
36 FOR I = 1 TO 24
37 PRINT "THE CURRENT BINARY DIGIT IS-",I
40 INPUT D(I)
50 IF D(I) = 1 THEN GOSUB 1000
60 IF D(I) = 0 THEN GOSUB 2000
70 NEXT I
75 REM THE FREQ IS NOW SET, SO DO IT
80 OUT 8,4
85 REM DO IT AGAIN UNTIL CTRL-C SAYS QUIT
90 GOTO 35
1000 REM SEND A ONE
1005 REM SET A ONE, TOGGLE CLOCK, SET XFER LOW
1010 OUT 8,1
1020 OUT 8,3
1030 OUT 8,0
1035 RETURN
2000 REM SEND A ZERO
2005 REM SEND A ZERO, TOGGLE CLOCK, SET XFER LOW
2010 OUT 8,0
2020 OUT 8,2
2025 OUT 8,0
2030 RETURN
```

monitor mode on the packet board so it will only respond to connect requests. I did this and captured one hour's worth of monitor mode and examined it. I was receiving the WB2MNF PBBS being relayed through WB2RVX0, WB4APR6 and WB4JFI5 as well as normal nightly traffic on the busy 145.01-MHz channel. The result was 16 kbytes of disk. By disabling the monitor mode, the TNC is discriminative and will ignore beacons, PBBSs and so on. It also makes the code simpler.

Let's look for *CNCT* and *DISC* in our code. On connection, the packet caller is informed that our PACANSWR program is ready to capture traffic.

The hardware is simple and does not require a schematic. Run a four-conductor wire from your computer serial port DB25 female connector to the Vancouver/Ashby DB25 female connector. This wire need only contain a hookup for pin 2 (Frame ground), pin 2 and 3 (receiver and send) and pin 7 on the Vancouver side. The Macintosh® computer uses RS-422, which has a DB9 connector instead of a DB25. Wire the chassis ground to DB9, pin 1. Signal ground goes to pin 3, transmitted data goes to TXD, pin 5 and receiver data goes to RXD, pin 9. Do not bother with handshaking. The Macintosh does not handle it correctly unless a serial driver is installed.

I coded Table 2 in Microsoft® BASIC 2.0 for the Macintosh. It is easier to understand than the C language. Notice the BASIC used is not like one you've seen before. There are no line numbers and this style of BASIC resembles FORTRAN. There are nice syntax structures included like the IF-THEN-ELSE statements. If are familiar with BASIC, this code should be easy to follow.

That's the program. I keep it simple. The Macintosh has an 8530 serial port that is handled in BASIC by setting up a

Table 2

PACANSWR in Microsoft® Basic 2.0

```
REM
REM --- PACKET ANSWER MACHINE PROGRAM
REM
PACANSWR:
     GOSUB INIT
     OPEN "0",#4, "CAPTURETEXT"
REM
REM N IS THE NUMBER OF ASTERISK RECEIVED AS IN
REM *CNCT* OR *DISC*
REM M IS A FLAG, 1 = CONNECTION AND 0=NO CONNECTION
REM
     N = 0
     M = 0
LOOP:
     P$ = INPUT$(LOC(1),1)
     PRINT P$;
     PRINT #4, P$;
     IF P$ = "*" THEN N = N + 1
     IF N = 2 THEN GOSUB CNCTDISC
     IF LOC(1) > 0 THEN LOOP
     K$ = INKEY$
     IF K$ = "@" THEN GOTO QUIT
     IF K$ = "" THEN LOOP
     PRINT #1, K$
     GOTO LOOP
QUIT:
     CLOSE #4
     END
REM
REM INITIALIZATION SUBROUTINE
REM MONACO MOSPACED NON-PROPORTIONAL FONT
REM 9 POINT ALLOWS 80 CHARACTERS PER LINE
REM XOR TEXT WITH WHAT IS ON THE SCREEN
REM
INIT:
     TEXTFONT 4
     TEXTSIZE 9
     TEXTMODE 1
REM SETUP COMMUNICATIONS 8530 PORT
    OPEN "COM1:300,N,8,1" AS 1 LEN=2000
    RETURN
REM
REM CNCTDISC SUBROUTINE
REM DETERMINE WHICH IS GOING ON
REM
CNCTDISC:
     N = 0
     IF M = 0 THEN GOSUB CNCTMSG:RETURN
     IF M = 1 THEN GOSUB DISCMSG
     RETURN
REM
REM IF CONNECTION ESTABLISHED, INFORM CALLER
REM
CNCTMSG:
     M = 1
     PRINT#1,"YOU HAVE CONNECTED WITH K8MMO, DAVE"
     PRINT#1,"BUT HE IS UNAVAILABLE, SO PLEASE"
     PRINT#1,"LEAVE YOUR MESSAGE AT THE BEEP AND"
     PRINT#1,"HE WILL GET BACK TO YOU...THANKS"
     PRINT#1,"DAVID'S COMPUTER...SKINNY MAC"
     PRINT#1,"               BEEP"
     RETURN
REM
REM DISCONNECTION, RESET FLAG AND ANNOUNCE TO CRT
REM
DISCMSG:
     M = 0
     PRINT "DISCONNECTION"
     RETURN
```

Continued on page 15.

By Bill Olson, W3HQT
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Measuring Antennas

Hams have always loved experimenting with antennas, and VHFers are no exception. Since most VHF and UHF antennas are relatively small, they are especially attractive to the workshop builder and tinkerer. There are many excellent "cookbook" designs in the amateur literature that, if duplicated exactly, will perform as predicted. Unfortunately many of us don't always follow the recipe and try to improve the design or modify it to use the materials at hand. In either case, it's a good idea to be able to test the antenna. It's no fun lugging a new array to the top of your hundred foot tower, only to find a high back-to-front ratio or twelve lobes, all equal in amplitude, coming off the front! This month's column is by no means a complete treatise on antenna gain measurement; it will, however, help a few readers get started in a very interesting field.

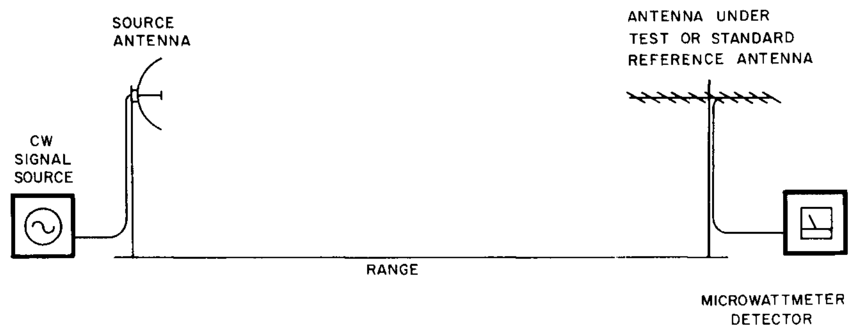

Fig 1—Simple antenna-gain measuring setup.

The Measurements

Assuming the antenna we want to measure is matched to 50 ohms, the important measurements are 1) absolute forward gain with respect to a dipole or other reference radiator, and 2) pattern (response in all directions, related to the main lobe). Qualitatively speaking, at least for directional antennas such as dishes and Yagis, we would like the antenna to have as much gain as possible off the front and as little as possible everywhere else. If these measurements can be made, even roughly, on the ground, there won't be any surprises when the antenna gets up on the tower.

Gearing Up

Any antenna-measuring setup has four basic components: 1) a range; 2) a signal source; 3) a detector; and 4) a standard-gain reference antenna. See Fig 1. RF at the measurement frequency is fed into the source antenna, which is aimed at the antenna under test. Then, a reference level is established. This is accomplished by using a reference antenna with known gain. Any subsequent measurements of unknown antennas are compared to this known reference level. Simple eh? Theoretically, yes. In practice, no. There are many ways to get false readings... read on.

The Range

It would be nice if we could all set up an accurate range to do antenna measurements! An ideal antenna range would be in outer space, I guess, but about the best we can do is a large flat area with no trees or buildings nearby. There should be nothing around that will reflect signals into the test antenna from the wrong direction. The worst things to have near an antenna range are large metal structures such as cars, buildings, towers and so on. These will really throw off readings in an unpredictable manner, especially when measuring sidelobes and antenna patterns.

Of course, we do have a reflector that we must live with—the earth. If the source antenna is placed close to the ground, say within a few wavelengths, the source signal will appear to originate from a point below the source antenna. This is because the reflected signal from the ground combines in phase with the direct signal. This is okay and actually has the effect of increasing the gain of the source antenna by a couple of decibels. If the source antenna is many wavelengths above the ground, the reflected signal is probably very low compared to the direct one. In this case, the signal appears to originate directly from the source antenna.

The other important feature of an antenna range is that it must be long enough to put the source in the far field of the test antenna. The far field (or Fraunhofer region) starts at a distance where the wave originating from the test antenna appears to be planar—a requirement for accurate antenna measurement. The far field starts at a distance of $2D^2/\lambda$. D is the major dimension of the test antenna (length of a Yagi or diameter of a dish) and is *always* longer than is convenient or possible. Shorter fields are usable, but they tend to compress the higher gain readings.

Source and Detector

The RF source and detector must be compatible to provide a linear response over the measurement range. There are a number of source/detector combinations that will provide reasonable accuracy for amateur measurements. With the vast amount of good test equipment on the surplus market, you should have no problem coming up with the right gear. The equipment described here is available at flea markets at affordable prices.

A simple setup uses a *clean* CW transmitter and a high-gain antenna with a clean pattern at the source end. The detector can be a microwattmeter such as the HP431 or HP435. These wattmeters use a bolometer sensor and will measure power directly in decibels in a very linear fashion. Since bolometer detectors are not as sensitive as diode detectors, you will need a good amount of ERP at the source. A couple of watts of RF and a relatively high-gain source antenna should do the trick.

Other detectors are more sensitive, but these require a 1000-Hz modulated source. The HP415 SWR detector is such an instrument. It uses a diode detector mount and is calibrated directly in decibels (at least some of the models are).

No matter which setup you use, be sure to check the linearity of the system. The easiest way is to insert an attenuator of known value in the line from the test antenna to the detector and see how much the detector meter reading changes. For example, if you insert a 6-dB pad, the meter reading should decrease by 6 dB.

Another simple detector is a receiver tuned to the test frequency. You can read the AGC voltage with a VTVM to get rough measurements, but there are too many places for nonlinearities here. I don't recommend this system for real accuracy.

A more sophisticated system, used at many antenna-gain measuring contests, is shown in Fig 2. It consists of an HP416B ratiometer as a detector and a 1-kHz modulated source. The '416, as the name implies, compares the signal coming into two separate ports and reads the difference in decibels. In this system, a *reference-port antenna* is left connected to one port at all times. All measurements are compared to the signal level from the reference antenna. This system has the desirable feature of taking out inaccuracies caused by variations (over time) in the output power from the source. The ratiometer uses a square law (diode) detector, so be sure to check for linearity when using this system.

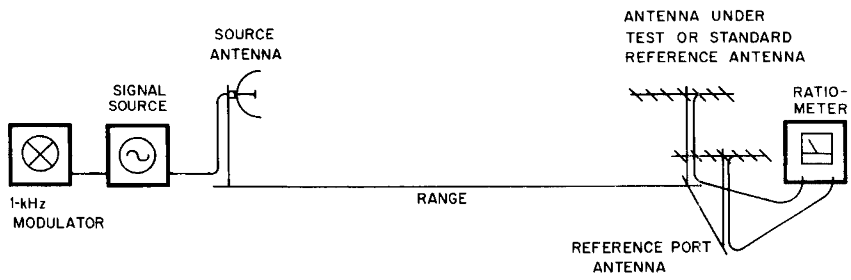

Fig 2—This improved antenna-gain measuring setup uses a ratiometer to factor out variations in source output power.

Standard Reference Antenna

The final critical part of the antenna measuring system is a standard-gain reference antenna with which all unknown antennas are to be compared. This is usually an antenna with moderate and *very* predictable gain. The EIA standard-gain antenna, a pair of dipoles over a groundplane is one such antenna. Various horn antennas with predictable performance can also be built.

If you don't have access to one of these standard antennas, an antenna of known performance (for example, the RIW-13 432-MHz antenna you measured at last year's VHF conference and which has been hanging up in the garage since) is probably as good as anything. Although your absolute numbers may be off a decibel or two, you really are only looking for system improvement. Relative readings are fine. Do not try to use a dipole as a standard reference. It will respond to signals from all over, including reflections from the test equipment. The higher the gain of the standard reference antenna, the better.

Making Measurements

Once you've gathered and assembled your test range, making the measurements is fun. Turn on the source and point the source antenna toward the detector end of the range. Walk back to the detector end and probe the field with the standard-gain antenna. Look for the maximum reading on the detector. You may have to move a bit or climb a ladder to get the peak.

Next, record the reading on your detector meter. The meter reading should be in decibels, probably referenced to a milliwatt (dBm). This is your reference level. Attach the test antenna to the same feed line in place of the standard-antenna and peak for maximum signal. If you know the gain of the standard, the gain of the antenna you're testing can be calculated. If, for example, your standard antenna reads 3.5 dB on the meter and your test antenna reads 6.5, your test antenna has 3-dB more gain than the standard. So, if the standard has 7 dB gain, your test antenna has about 10-dB gain.

If your measurement system has enough dynamic range, you can rotate the antenna 360 degrees and record the total pattern. At a minimum, you can record the sidelobe level and front-to-back ratio. If the gain is what you expected and the pattern is clean with sidelobes down 16 or 17 dB below the main lobe, put the antenna up. If not, maybe you better go back and read the instructions or try again!

I receive about one card or letter each month from the readers. I am assured that there are more who actually read the column, but it would be nice to hear from some of you. Any suggestions for future columns? I'll be glad to oblige if I can. In any case keep that card or letter coming!

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2000-character buffer. When a character comes to the computer from the Vancouver board, an interrupt occurs and the character is placed in the 2000-character buffer at the next available place. Using the LOC statement, INPUT$(LOC(1),1), one character is input to us from the 2000-character buffer. If LOC(1) is 1, this means at least one character is in the buffer. When the buffer is cleared, LOC(1) is a zero and we can check the keyboard.

This program is based on the terminal program supplied with the BASIC on the Macintosh. If you own a Macintosh, give it a try. If each packeteer had a program like this, most of the bulletin boards could be eliminated.

Disk Drives for the Xerox 820-1

Canon MDD2 double-sided, double-density drives were purchased from B.G. Micro (PO Box 280298, Dallas, TX 75228; tel 214-271-5546). The drives are 2/3 height, which means that two occupy more space than one Shugart SA450. They make fine drives for the Xerox computer. After two of these drives came into my possession, Mel Seyle, WA3KZR, discovered that the following switch settings work fine with Xerox 820s.

SW1—1 on, 2 off, 3 off, 4 off
SW2—1 on, 2 off, 3 on, 4 off, 5 off, 6 on, 7 off
SW3—1 off, 2 on, 3 on, 4 on, 5 on, 6 off

These settings allow the Xerox to shut off the drives after about 20 seconds of nonuse. The drives appear to work excellent and its low price encourages other ideas concerning packet and spread-spectrum software.

[Part 6 concludes the Xerox 820-1 Compendium; a composite video board for the 820-1 is discussed.]

Notes

[32]D. Borden, "Xerox 820-1 Compendium—Part 4," *QEX*, Sep 1986, p 10.

[33]H. Feinstein, "Spread Spectrum: Frequency Hopping, Direct Sequence and You," Jun 1986 *QST*, p 42.

[34]F. Williams, "A Digital Frequency Synthesizer," *QST*, Apr 1984, p 24.

[35]F. Williams, "A Microprocessor Controller for the Digital Frequency Synthesizer," *QST*, Feb 1985, p 14.