By Doug Smith, KF6DX

Digital Voice: An Update and Forecast

Having read the background information in my article last month,[1] you may be asking, "That's fine, but what are hams doing with digital voice now? What can we expect in the future?" Well, I am back to tell you what I have learned about that since writing the last article.

Things are definitely warming up on the digital voice front. Hams have fielded working systems and interest is growing. There is still a lot of work to be done, but the use of digital voice technology in Amateur Radio is rapidly expanding.

Amateur Digital Voice Systems

G4GUO and Friends

As reported last time, Tucson Amateur Packet Radio (TAPR) is producing a digital voice coder/decoder ("vocoder"). Charles Brain, G4GUO, and Andy Talbot, G4JNT, began work on that design in 1998.[2] They sorted through the coding algorithms available and decided on an advanced multi-band excitation (AMBE) vocoder from Digital Voice Systems, Inc (DVSI). AMBE vocoders are available in chip form and the algorithm, coded for several popular DSP platforms, is available for licensing from the manufacturer.[3] As explained previously, AMBE vocoders attain good voice quality at low bit rates by using advanced parametric speech-coding methods.

Charles and Andy chose a Microchip 17C44JW PIC microcontroller for their design to do the data handling and control. The circuit also has a Motorola MC14LC5480P µ-law coder. µ-law extends the dynamic range of the system (see previous article). Using the forward error-correction (FEC) facility of the AMBE chip, Charles and Andy operated the system using 2400 bits/s for voice; an additional 1200 bits/s were necessary for the FEC, producing a final bit rate of 3600 bits/s.

3600 bits/s is a fast rate for HF and a 36-tone PSK modem was used for initial testing. The system requires no feedback from the listener. It can therefore be operated with one talker and many listeners. It is capable of full-duplex operation.

By March 1999, Charles and Andy had made their first digital voice contact on the 40-m band over a 70-km path. They report that when signal-to-noise ratio (SNR) was 25 dB or better, it sounded like a telephone conversation—no background noise whatever, except for the "comfort noise" inserted by the vocoder itself. Lower SNRs produced degradation in some proportion.

At the time of this writing, the TAPR kit is in its beta test phase and does not include a modem. Modems are readily available, though, for VHF-and-above work that will sustain data rates up to 9600 bits/s. The vocoder may be programmed to operate at various slower rates for experimentation. Contact TAPR for more information.[4]

Work continues at G4GUO on high-speed HF modems having sophisticated error-correction schemes. We can expect

[1]Notes appear on page 41.

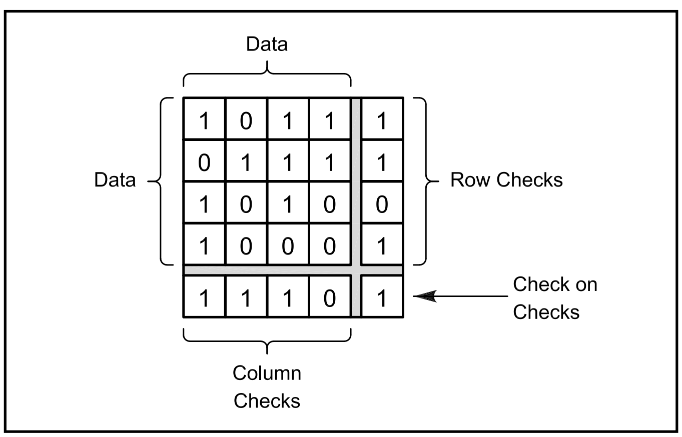

Figure 1—An example of iterated coding.

Figure 2—505RC remote-control setup.

Charles' work to benefit data communications as well as digital phone. We are finding that many digital voice schemes tolerate up to about 1% data loss without seriously affecting performance.

The KF6DX Remote Control

Also in 1998, I designed a remote-control system for the Kachina 505DSP transceiver called 505RC that employed digital voice techniques. I arranged the system for operation over either telephone lines or dedicated data radios.

I chose a continuously variable slope-delta (CVSD) vocoder, the MX-COM MX609. CVSD was discussed in the previous article and also in *QEX*.[5] It is a waveform vocoder that aims at exact reproduction of the input signal; AMBE is a parametric vocoder that focuses on reproducing the correct time spectrum of its input. AMBE operates well at bit rates of less than one-quarter those of CVSD, but CVSD is very simple and inexpensive. The MX609 does not require any programming by a microprocessor and the chip needs only 5 V dc, audio and serial data in and out, and a crystal to run. CVSD also does not require synchronism between talker and listener other than reasonable clock accuracy.

I wanted to be able to send command and control data along with the voice data over the control link, as well as some telemetry and feedback to and from the transceiver. I arranged to time-multiplex the command, control and telemetry data with the voice data, producing a single serial bit stream. I added synchronization bits to allow the demultiplexer to sort out the parts of the data stream. I used an error-correction scheme called *iterated coding*.[6]

Iterated coding is a fairly simple block code that can detect and correct a single bit error in the encoded block. Checksum bits are computed for all rows and columns in the block, modulo-2. See Figure 1. A check bit is also computed for the row and column containing the other check bits—a sort of check on the checks. For a block of M bits, it requires $2M^{1/2}+1$ additional bits to be sent. An input block of 49 bits, for example, results in a coded block of 64 bits.

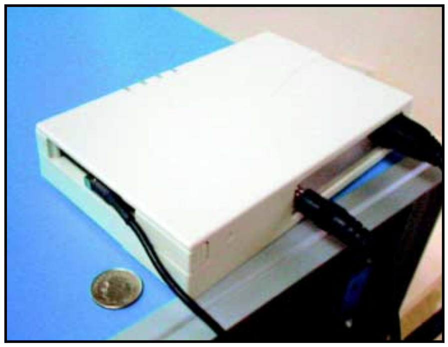

Figure 3—Lucent EC/S high-speed data radio.

The serial bit streams in and out of the control unit are passed to the data radios at standard EIA-232 rates up to 38.4 kbits/s. At that highest rate, about 26.4 kbits/s are used for digital voice and 12.0 kbps for telecommand data and coding overhead. The audio quality of CVSD at 26.4 kbits/s is reasonably good, attaining a mean opinion score (MOS) of almost four out of a possible five. I made the audio bandwidth proportional to the data rate; it is about 2.7 kHz at the highest rate. In addition, the system supports three discrete serial command/control channels, also at EIA-232 standard rates. It uses one channel to control the transceiver (using software running on a PC) and the other two may be used to control an antenna rotor control and a RTTY modem, for example. Five ancillary open-collector outputs are also provided. Those are good for turning power on and off to a solid-state amplifier or for operating a digital antenna switch. Figure 2 shows one of the remote-control units. A complete system requires two units: one at the control point and one at the transceiver site.

I used data radios originally intended for Part-15 use on the 900-MHz industrial, scientific and medical (ISM) band. I finished with a pair of 2.4-GHz ISM radios from Lucent (model EC/S) having serial input and output (see Figure 3). They produce about 35 mW of transmit power and are capable of passing 11 Mbits/s of data in half-duplex—overkill! But they are reasonably inexpensive and they work well over a 5-km path with 24-dB-gain, grid-dish antennas when forced to 1 Mbit/s pseudo-full-duplex. Full-duplex operation is simulated by switching the radios rapidly between transmit and receive. The switching times of the Lucent radios are on the order of a microsecond. It is neat to operate remotely—my system works well enough that it is hard to tell that the radio isn't in front of you.

The WK6F Remote

Ken Beals is WK6F and he, too, chose CVSD for his remote-control system.[7] He began work on his design while at Cal State, Chico in 1994, but the results were not published until 1999. The vocoder chip chosen is the Motorola MC3418. He uses 10-GHz Gunnplexer transceivers for the control link and separate channels for control and voice data. A 4-5 MHz subcarrier is modulated with the digitized voice data and the control channel supports up to 115 kbits/s. Both channels are full-duplex.

Ken did a beautiful job (see Figure 4) and he predicts the system would work well over a 40-mile line-of-sight path with decent antennas. I was unaware of his work until I jumped on the *QEX* bandwagon. Check out his article (see Note 7).

APCO 25

APCO 25 is a standard that provides digital voice and messaging to the public-service community. The system incorporates AMBE vocoders at VHF and above. Both APCO and ARRL understand there may be a need for interoperability using those rigs during emergencies and at least one group of amateurs, the Motorola Amateur Radio Club of North Texas (MARC), has been using the technology in the Fort Worth, Texas area since August 2001. They have installed a Motorola Quantar repeater at their facility that is compatible with the APCO 25 standard.

Harold Reasoner, K5SXK, reports that the Fort Worth chapter of the Texas VHF-FM Society gained access to APCO 25 mobiles and hand-helds for testing purposes through its relationship with MARC. The Quantar repeater operates in both digital voice and traditional analog voice modes. When asked to rate the

Figure 4—WK6F's remote-control units.

Figure 5—Motorola APCO 25 mobile and hand-held transceivers.

voice quality of the APCO 25 system on a scale from zero to five, Harold said: "It's near a five when you're in range. When traditional analog modes are getting noisy, the APCO 25 radios remain virtually noise-free." Testing by users shows that the coverage area is consistently greater when operating in digital mode than in analog mode, although quality tends to fall off rapidly at the extremes of the coverage area.

The APCO 25 mobile rigs cover portions of the two-meter amateur band, plus public-service frequencies in the range 148-174 MHz. They put out 75-80 W on 2 meters (see Figure 5). Occupied bandwidth as configured in digital voice mode is virtually the same as that of a normal, 5-kHz-deviation analog signal. The equipment can also operate on 12.5-kHz channels. Motorola programmed the frequencies and certain digital group codes into the units. The group codes allow selective reception of messages intended only for a particular group. According to Harold, Motorola has recently announced a voice-over-IP (VoIP) option for APCO 25 systems operating in the 800-MHz, 821-MHz and the newly allocated 700-MHz public-safety bands. The implication is that the units can be tied into IP networks or through the Internet.

Although APCO 25 radios are more costly than regular amateur rigs, the standard may catch on with more hams as they and public-safety officials work together to meet increasing demands.

Alinco's Digital Voice System

Several months ago, Alinco announced a digital voice option for some of their VHF and UHF transceivers.[8] Their DJ-596 dual-band hand-held (see Figure 6) and DR-135, -235 and -435 mobiles may be fitted with digital voice units. Models EJ-40U and EJ-43U use—you guessed it—CVSD and Gaussian minimum-shift keying (GMSK) modems employing the V.32 modulation standard. CVSD audio is transmitted at 14 kbits/s.

Alinco spokesman Jeff Reinhardt, AA6JR, describes the system as "purely experimental" and "a transitional step." That may mean Alinco has something even greater in mind for the future.

Other Systems from Japan

Others in Japan are right there with digital voice technology, too. Last year at Ham Fair 2001, three organizations displayed prototype digital transceivers (see Figure 7). Both ICOM and Kenwood demonstrated 23-cm (1.2 GHz) digital transceivers. The ICOM unit is designed to operate at 8 kbits/s in digital voice mode and at 128 kbits/s in data mode. It also includes a regular, analog FM phone mode. For digital voice, the rig uses a G723.1 vocoder (code-book-excited linear-prediction coding, or CELP) and it even sports a 10Base-T network interface. Both digital modes utilize a GMSK modem. Digital voice sensitivity is listed as only 6 dB worse than in analog FM mode. How that sensitivity was determined is not known. Maximum transmit power is 10 W.

Kenwood also showed a prototype 23-cm digital transceiver, operating digital voice using AMBE at 2.4 kbits/s and a GMSK modem. Specifications and other details were not available at the time of this writing.

Also shown were 23-cm and 3-cm (10-GHz) digital terminal equipment, including a digipeater. It looks as if those units are intended for use for high-speed networking applications, perhaps using TCP/IP. Further details were not available. For an English translation of a short *CQ Ham Radio* article on that part of the show, visit **www.arrl.org/tis/info/digivoice.html**.

Digital Audio Broadcasting and IBOC

In mid-2001, the International Telecommunication Union (ITU), an arm of the United Nations, approved certain systems as standards for digital audio broadcasting.[9] One of these systems allows the simultaneous transmission of both a standard AM signal and a digital audio signal. Such in-band, on-channel (IBOC) systems are thus compatible with existing analog AM receivers and also supply an enhanced digital audio signal.

The appearance of digital audio on international broadcasting channels will soon give rise to a new crop of digital short-wave receivers. Only a few stations are experimenting with those systems now; but it is expected that soon, many more will join in. Some hams believe we can learn something from the technology, too.

Michael Schulhof, K1OKI, reports that one of the early developers of technology for digital audio broadcasting is Thales Corporation (formerly Thomson CSF), a French company. "Their approach to HF digital audio and their participation in developing the ITU standards have been vigorous from the beginning," Michael said.

According to Schulhof, the Thales system has already been tested in an occupied bandwidth of 3 kHz, which makes it a likely candidate for Amateur Radio trials. A subset of MPEG-4 AAC (advanced audio coding, a form of parametric vocoder) is used. Like other digital audio broadcasting systems, it uses orthogonal frequency-division multiplexing (OFDM) modulation. OFDM is another multiple-sub-carrier method that is getting attention among digital TV designers. The Thales scheme includes error correction and can handle either monaural or stereo sound in its preferred embodiment.

Amateur Radio transatlantic trials are in the planning stages at the time of this writing. Schulhof, the former Chairman and CEO of Sony Corporation of America, also points out that the Thales system satisfies a requirement for fast signal acquisition as the receiver is tuned. He added, "Hams will eventually be seeing it show up in manufacturers' specifications."

Michael Schulhof holds a PhD in physics from Brandeis University and was instrumental in introducing to the public many of the digital services we now take for granted. He has been continuously involved in Amateur Radio since 1958. He currently runs a private investment firm in New York.

Voice Quality Evaluation

In its recent report to the ARRL Technology Task Force, the ARRL Digital Voice Working Group (DVWG) recommended some standards for voice-quality evaluation of digital voice systems.[10] Those standards are based on the subjective judgments of listeners. The term subjective means that questions are asked of the listeners and voice quality is rated based on their answers.

The ITU is working with KPN Research of The Netherlands and British Telecom to refine a standard for the objective measurement of voice quality.[11] The term objective means that physical measurements are taken of the original and decoded signals and a complex numeric analysis is used to determine voice quality. Researchers are designing their algorithms carefully so that the results correspond closely to the kind of subjective evaluation proposed by the DVWG.

When evaluations must be made continually over short time frames, say every five minutes, objective measurement wins over subjective by a long way. It is quicker to make some physical measurements than to ask a bunch of listeners how something sounds. Objective measurements are inherently repeatable and can be done by those having the necessary test equipment and computing power. Objectivity is difficult to achieve, though, when you are considering what someone hears or does not hear. Much work remains to be done; but I am confident that as far as human-hearing traits can be identified, they can also be formulated.

What Else Does the Future Hold?

It is not easy to predict the future; but within the realm of digital voice, we see some very interesting possibilities on the horizon. Consider the following ideas as examples and not as an exhaustive list.

Much work in the coming years will focus on improving the quality and robustness of digital voice communications. HF is an especially difficult medium to tame and Amateur Radio experimenters will continue to work on high-speed data transmission and digital voice through it, alongside digital broadcasters. That is a reassuring prospect, since we may find the results valuable the next time someone asks, “What have you done with the spectrum lately?” That same thought applies equally well to the rest of our allocations.

Unlike broadcasters, though, amateurs can consider the possibility of transmitting digital voice at a slow rate, then speeding it back up at the receiver. That opens the door to narrower bandwidths that allow greater distances to be covered. It would not surprise me to see Earth-Moon-Earth (EME) voice contacts become commonplace that way—as long as you are willing to wait! Additionally, transmissions may be sent many times to achieve a large measure of FEC, accomplishing the same thing (long time integration). I wonder what that will do to voice contests and distance records.

Hams and other users may be willing to accept less than perfect voice quality in return for other capabilities and services. The embedding of coded identifiers in digital voice transmissions suggests some very exciting possibilities. For example, those codes could be used to identify source and destination addresses for messages, extending store-and-forward capabilities to users. In fact, TCP/IP and other packet schemes may be attractive for digital voice on certain bands. Equipment is out there now for wireless networking systems. We could be using it to occupy the 33-cm, 13-cm and 5-cm bands and to exploit our privileges there before commercial interests overrun them.

Embedded codes could also provide feedback about propagation conditions. For example, a spread-spectrum user could arrange to reduce his transmit power to the minimum based on feedback from the listener, in accordance with the new FCC rules regarding that mode. During a CQ call, those same types of codes might indicate the caller’s areas of interest or that the call is directed at a particular country or group.

Through multiplexing techniques that are currently widespread in cellular telephone systems, more than one QSO could be supported simultaneously through digital repeaters or “digipeaters.” The same thing applies to satellite operation. Code-division multiple-access (CDMA) and time-division multiple-access (TDMA) are proven technologies that may go well with digital voice over Amateur Radio.

Figure 6—Alinco DJ-596 transceiver.

Figure 7—Prototype digital radios shown recently in Japan.

A movement is afoot to tie Amateur Radio networks together with the global Internet. That is already providing unparalleled robustness and redundancy to critical communications systems. We can bolster our public-service value and enhance our enjoyment by continuing to expand and enhance such cross-connections. Would it not be neat to operate through your repeater in San Diego and work a handheld station in New Zealand? Or anywhere your embedded codes indicate you want?

Finally, detection and correction of multipath distortion on digital links is an area ripe for experimentation. Amateur Radio is already in the thick of it.[12] Who says we’re not on top of the technology, eh?

Acknowledgment

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Notes

[1]D. Smith, KF6DX, “Digital Voice: The Next New Mode?” *QST*, Jan 2002.

[2]C. Brain, G4GUO, and A. Talbot, G4JNT, “Practical HF Digital Voice,” *QEX*, May/Jun 2000.

[3]Digital Voice Systems, Inc, 234 Littleton Rd, Westford, MA 01886, tel 978-392-0002, **www.dvsinc.com**.

[4]Tuscon Amateur Packet Radio, 8987-309 Tanque Verde Rd, Tucson, AZ 85749-9399, 940-383-0000, **www.tapr.org**.

[5]D. Smith, “Signals, Samples and Stuff, Part 4,” *QEX*, Sep/Oct 1998.

[6]H. P. Westman, ed., *Reference Data for Radio Engineers*, 6th ed., Howard W. Sams & Co, Indianapolis, IN, 1975.

[7]K. Beals, WK6F, “A 10-GHz Remote-Control System for HF Transceivers,” *QEX*, Mar/Apr 1999.

[8]See **www.alinco.com** for more information.

[9]“Empirically Speaking,” *QEX*, Jul/Aug 2001.

[10]For more information, visit **www.arrl.org/announce/board.html** and look for “Committee Reports.”

[11]P. Denisowski, “How Does It Sound?” *IEEE Spectrum*, Feb 2001.

[12]See articles by N2MJI and KF6DX in *20th Digital Communications Conference*, 2001, ARRL/TAPR.

Doug Smith, KF6DX, is Chair of the ARRL Digital Voice Working Group. Since 1998, he has edited QEX: Forum for Communications Experimenters for ARRL. He is the author of the DSP chapter in the current edition of the ARRL Handbook for Radio Amateurs (ARRL, ISBN 0-87259-189-1, ARRL order number 1891) and of Digital Signal Processing Technology: Essentials of the Communications Revolution (ARRL, ISBN 0-87259-819-5, ARRL order number 8195). Doug may be contacted via e-mail at **kf6dx@arrl.org.**