D-STAR, Part 2 of 3:
Design Considerations

D'PRS™

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D-STAR, Part 2: Design Considerations

Come learn what JARL put into their proposed new VHF communication standard.

By John Gibbs, KC7YXD

In the first segment of this series, we considered the attributes we would like to see in any new VHF/UHF Amateur Radio system. In this second segment, we discuss the technical issues involved with selecting the parameters of an Amateur Radio digital system to meet those attributes.

Perhaps the easiest way to organize the design considerations for a digital radio system is to use the OSI Model. The OSI Model is officially known as The Basic Reference Model for Open Systems Interconnection. We start the description with the bottom level of the OSI model, the physical layer. Then we will work our way up to the top levels, which are open to amateur experimentation and application development.

Physical Layer

Transceiver Frequency

We hams have a large spectrum allocation at 1.2 GHz (60 MHz wide in the US) that is little used today. In fact, we should use this spectrum or risk losing it to commercial interests. Also, if we want to develop a system to send high-speed data, we will need a wide-bandwidth signal and there is little available spectrum at 70 cm. If it is desired to use the D-STAR protocol at lower frequencies, the high-speed data mode could be dropped. In fact, a prototype portable 2-m HT using only the digital voice mode has been developed and was shown this spring at IWCE.

Like the previous FM system, the system design logically calls for half-duplex operation for digital voice and simplex for high-speed data.

Repeater Link Frequency

Since it is desired to have multiple contacts and high-speed data packets on the repeater link, a wide bandwidth is required. Therefore the repeater backbone must be at microwave frequencies and the amateur band at 10 GHz is a logical choice. Today's equipment generates usable 10 GHz power, is affordable and the 10 MHz of bandwidth is practical. The bi-directional, asynchronous nature of multiple contacts demands a full-duplex repeater link.

Given the high-speed data requirements of the system and the desire to use digital voice to reduce transceiver spectrum requirements, the repeater link should be digital. With the asynchronous nature of the system, packet mode is a natural choice. However, because a packet system does not guarantee real-time communication, voice should be given priority over data to minimize the possibility of voice disruptions.

Modulation

An ideal modulation system should generate a signal that has a narrow spectrum, with low side lobes so that it does not interfere with other nearby users. On our crowded ham bands, this is becoming more of an issue daily.

There are several commonly used modulations for digital data. The digital modulation scheme chosen will significantly affect the performance of a communication system. Generally we want to maximize the data rate within

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the constraints of acceptable level of latency, available bandwidth, acceptable error rate, product costs and operating environment (that is mobile, portable, fixed-link). In particular, mobile and portable operation causes variable multipath fading and fast phase shifts that can wreak havoc with digital radios.

Less spectrally efficient modulations generally have better operation characteristics in poorer SNR conditions. Also, they are more forgiving of frequency-offset errors between the transmitter and receiver and frequency and phase response error on the channel—an important consideration if costs are to be kept down in a UHF system.

4 FSK—FSK, MSK and GMSK are very attractive because they are constant amplitude modulation. This means that the power amplifier can be class C, which offers low cost and excellent power efficiency.

FSK has of course been used in amateur systems for years, dating back at least to the introduction of RTTY. Newer variations on FSK use more frequencies than just mark and space. For instance, the new weak signal mode, JT44 uses very slow FSK (about 5 Hz data rate) with 44 different frequencies each corresponding to a character.[1] But at the higher data rates needed for a VHF/UHF digital voice system, four FSK frequencies offers an attractive option for improved FSK performance.

GMSK—Among FSK, MSK and GMSK, GMSK offers the best spectral efficiency with only a slight degradation in the BER compared to FSK and MSK. These advantages have made GMSK one of the most popular digital modulations worldwide. Other more complex modulations like QPSK require a more expensive linear power amplifier that also typically requires more current, which is critical in portable operation.

GMSK low-pass-filters the data stream with a filter that approximates a Gaussian time and frequency response. A Gaussian filter is used because of its desirable properties in both the time and frequency domains. This filtering reduces the high-frequency content of the modulation and therefore narrows the frequency spectrum of the modulated signal while widening the data response minimally. However, as you continue to narrow the filter, the spectrum continues to narrow and the time response of the filter lengthens. This causes the peak amplitude to decrease and the adjacent data tails of the time response to interfere with the decoding of the desired symbol, a phenomenon called inter-symbol interference (ISI).

A typical compromise between ISI and bandwidth used by many systems is for the bandwidth/data rate ratio to be equal to 0.5. This yields almost no degradation due to ISI compared to MSK and yet dramatically reduces the spectral occupancy of GMSK compared to MSK.

QPSK—In theory, quadrature phase-shift modulation could have a constant amplitude format. However, the rapid switching of the input data causes a QPSK signal to have large sidebands that destroy its spectral efficiency. Therefore in practice, raised cosine filters are used on input data to reduce these sidebands. To preserve the wave shape induced by these filters requires the use of a more expensive and less power-efficient linear amplifier. If a class-C amplifier were used with QPSK, the sidebands that were removed by the cosine filter would be regenerated.

In the presence of additive white Gaussian noise (AWGN), QPSK requires about 3 dB less signal-to-noise than does FSK. However, in real channels, with multipath and poorly synchronized receivers, the 3-dB advantage quickly disappears.

Data Link Layer

Time Division Multiple Access (TDMA)

TDMA is one of the two commonly used multiplexing standards for cellular phones. The cell tower site acts as the master clock and assigns a time slot to each of several cell phones that are assigned the same frequency. For proper operation, it is critical that each phone transmit and receive exactly in its assigned time slot. This is not attractive for amateur simplex operations because operations are as two or more equals, and there is no master to determine the clock and assign time slots.

Any Amateur Radio system has to work without a centralized frequency reference and master clock. In addition, the radios must be able to acquire signals that are somewhat off frequency and acquire timing without the need for a separately transmitted clock signal. These requirements may make an amateur system less spectrally efficient than a centrally-controlled system like the cell phone, but they are more in keeping with the spirit of Amateur Radio, particularly the capability to operate when the infrastructure is destroyed.

Code Division Multiple Access (CDMA)

CDMA (also known as spread spectrum) is also used for multiplexing cellular phones. In CDMA, several cell phones share the same frequency and transmit simultaneously. Each phone on a frequency is modulated with a code sequence that spreads the spectrum in a unique way. If the receiver is synchronized and has the same code sequence, then the signal is restored. Otherwise, the signals from other phones become part of the background noise.

An important limitation on the system is that undesired signals are not completely rejected. Depending on the length of the codes used and the attendant difficulty in synchronizing, perhaps 20-30 dB of so-called processing gain can be attained. Therefore, a strong nearby CDMA signal can overpower a more distant signal. This classic problem with spread-spectrum communications is called "the near-far problem."

In a cell-phone system, this problem is addressed by power control. Since all the nearby phones are communicating with the same nearby cell site, the cell site remotely controls the power level of each phone to minimize the possibility of interference. However, in Amateur Radio, particularly with multiple simplex contacts, this is not a solution.

Frequency Domain Multiplexing (FDM)

TDMA, CDMA and other modern multiplexing schemes require coordination between the units that is incompatible with the basic goals of the Amateur Radio Service. One of the major justifications for our service in the US is emergency service. TDMA and CDMA require an infrastructure to provide the coordination. This infrastructure would quite possibly be destroyed in an emergency. So, the best solution for Amateur Radio is what we have traditionally used, FDM.

Network Layer

In the network layer, the binary data stream is divided into discrete packets of finite length. In addition, error checking is performed by cyclic redundancy check (CRC) at this level. If an error is detected, it is corrected by the retransmission of packets.

Transport Layer

In the transport layer, we multiplex and split all the data streams we need to send and receive. In an Amateur Radio system, we would typically need to include repeater control data; source, destination and routing information (that is, call signs of both operators and repeaters used); and what is called the payload, which is the voice or data to be sent.

[1]Notes appear on page 28.

Presentation Layer

Codec

As mentioned in the first segment of this series, simple PCM encoding of voice results in a 64,000 bits/s data stream. Codecs have been developed to compress voice with good quality down to 2400 bits/s and lower. These codecs develop their extreme data compression by modeling short segments of the human voice and only transmitting the reduced information needed to describe the voice model.

One of the major difficulties in designing a digital voice radio is in testing the voice quality. High-compression codecs are designed to work with a human voice; traditional tests like frequency response and harmonic distortion with sinusoidal tones do not generate meaningful results. Consequently, a subjective method of testing called *mean opinion score* (MOS) has been developed. MOS is estimated by a test with a group of normal listeners who are asked their opinion on a five-point scale (1 = bad, 5 = excellent) and the results are averaged together.[2]

A very important factor in conducting MOS tests is the acoustical environment. Since the codec is designed to highly compress the information in a human voice, it is easy to imagine that the presence of other signals and noise can severely affect the performance of the system. An excellent test for Amateur Radio is performance in an automotive environment including engine noise and wind noise from an open window.

A final issue in codec selection is the MOS performance in the presence of the channel impairments we commonly find in VHF/UHF communication paths. As Digital Voice Systems point out on their Web site, "Vocoders…designed for extremely low bit-error rates, such as those encountered in land-line communications, often experience serious degradation when applied to the much higher bit-error rates found in wireless communications. Consequently, it is important to consider robustness to channel degradations during the vocoder-algorithm design process."

Scrambling

Bit synchronization and accurate level slicing in the receiver require frequent transitions in the data (no long

Table 1—D-STAR Transmission Characteristics

Mode	*Transmission Speed*	*Bandwidth*
Backbone	10 Mbps or less	10.5 MHz
Data	128 kbps or less	130 kHz
Digital Voice		
ITU	8 kbps	9 kHz
AMBE	2.4 kbps	5 kHz

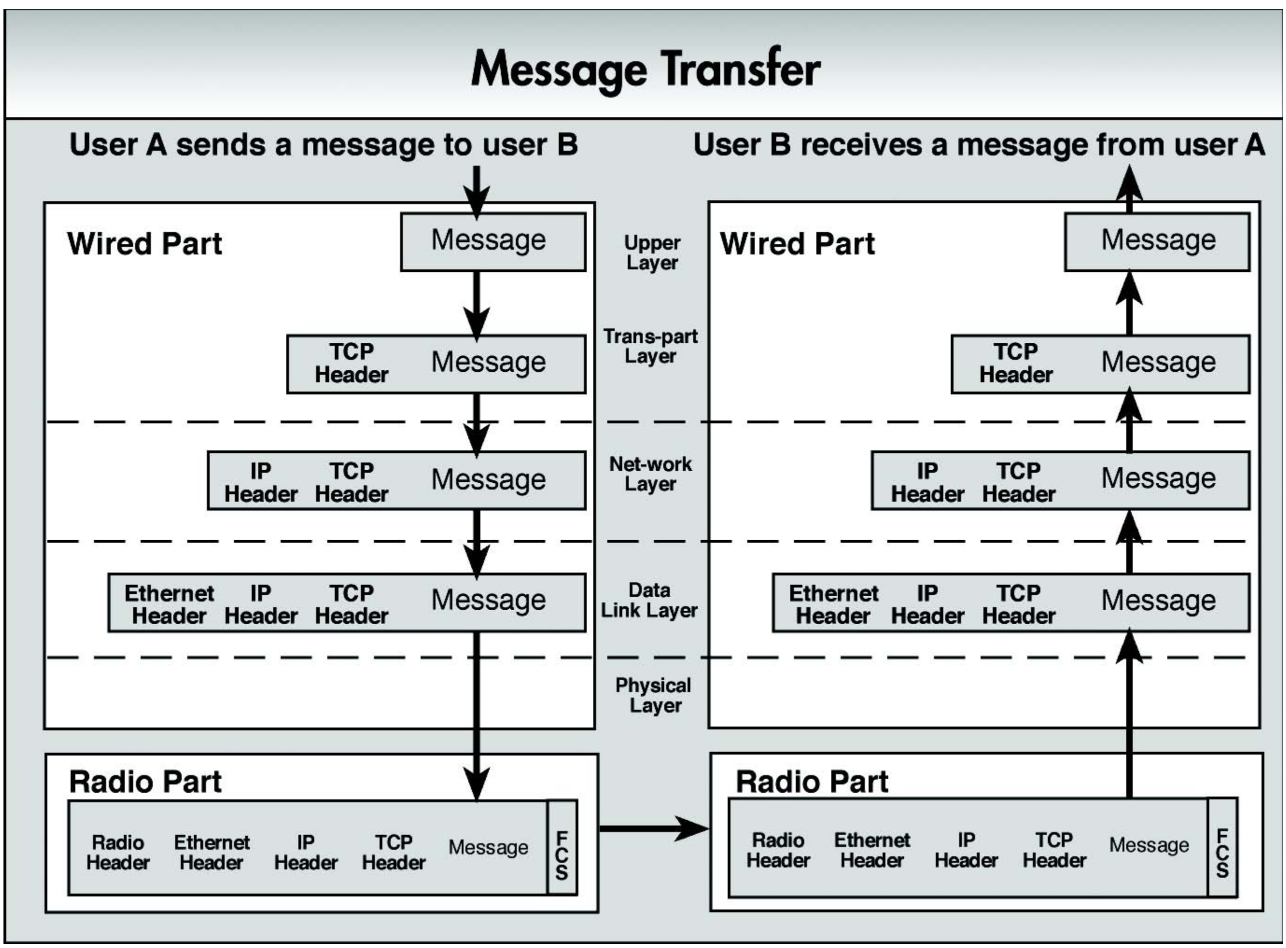


Fig 1—Header additions with TCP/IP protocol.

strings of ones or zeros). To ensure that condition, most digital radio systems use a device called a scrambler to randomize the input data stream.

"Scrambler" is an unfortunate term for people who are familiar with analog radios; it is common to interpret this as encryption. The FCC of course forbids encryption for amateurs. Scrambling is not an attempt to hide the message content, however; it is a fixed and published method known by all potential receivers for converting the input data stream into a data stream with short strings of ones or zeros.

Scrambling is typically done with a shift register and exclusive OR gates. CCITT recommendation V.26 recommends this procedure.

Application Layer

This is the layer where hams can begin to customize the system and add their own applications. In addition, this is the level where the system design allows for user control entry and data entry, both data from the IEEE 10BaseT Ethernet and analog audio from the microphone.

D-STAR Proposed Standards

As stated in the first part of this article, D-STAR is not a finalized standard at the time of this writing. However, the field trials are finished and standard publication begins here. Table 1 shows the system as it stands at this writing.

To describe the proposed D-STAR standard, we will start at the input side of the transceiver and work our way out to the antenna, working our way down the OSI model. We will then see how the standard defines the repeater operation and the links between repeaters. First we will consider the high-speed data mode and then the digital voice.

High-Speed Data

The standard interface for high-speed data into and from the D-Star system is IEEE802.3 (10BaseT Ethernet.) In Fig 1 you can see how the D-STAR transmitter adds a radio header extension to the Ethernet message just as the Ethernet protocol added a header to the Internet Protocol, TCP/IP. Since this radio header is stripped off in the receiver, the radio link appears to be a "wireless Ethernet cable." Therefore, it is possible using existing software (such as browsers) to communicate the same images, text and voice as is handled by Ethernet, including links to the Internet, without modification.

Data Multiplexing

Fig 2 shows the details of a communication packet from the radio part in Fig 1. Each packet consists of a radio header and the Ethernet packet described above followed by an error-checking frame. The radio header is worthwhile to study in some depth as it shows many of the D-STAR system capabilities.

Each frame of the radio header is identical in both the high-speed data mode and in the digital voice mode. If the standard is approved as proposed, it will contain the following information:

The first two fields are common to most digital radios, the bit sync and the frame sync. These preambles are designed to allow the receive modem to establish timing and level lock as quickly as possible.

The flag field describes the content of the data field.

- Bit 7 Data or voice communication flag.
- Bit 6 Repeater or simplex flag.
- Bit 5 Communication-interruption flag.
- Bit 4 Control signal, data or voice signal flag.
- Bit 3 Emergency/normal signal flag.
- Bits 2-0 Transmission-control bits (see below).

The ID field can hold four call signs:

1. The local repeater you are accessing (optional).
2. The linked distant repeater the called party is using (optional).
3. The station you are calling (can be CQ).
4. Your own call sign.

The PFCS field is a check word for the header. Some of these bits require a little more explanation if we are to understand the operation of the system. Notice that when bit 3 of the flag field is set, you are asking for an emergency break-in. (On many FM repeaters you would say "break" today.) For

Data Mode Message Transfer Details

Radio Header								Data					FCS
			ID						MAC Header				
Bit Sync	Frame Sync	Flag	Destination Repeater Callsign	Departure Repeater Callsign	Companion Callsign	Own Callsign	PFCS	ELen	SA	DA	Type	Data Frame	CRC
64bit	15bit	1byte	8byte	8byte	8byte	8byte	2byte	2byte	6byte	6byte	2byte	46-1500byte	4byte

Fig 2—Proposed bit pattern in high-speed digital mode.

Table 2—Call-Sign Combinations and the System Function

(Uses Evergreen Intertie Call sign examples[3])

Called Station	*Departure Repeater*	*Destination Repeater*	*Own Station*	*Function*
CQCQCQ	KB7WUK	K7NWS	KC7YXD	CQ Portland
N7ABC	KB7WUK	K7NWS	KC7YXD	Call N7ABC in Portland
N7ABC	K7NWS	K7NWS	KC7YXD	Call N7ABC on local repeater
N7ABC	DIRECT	DIRECT	KC7YXD	Simplex

instance if you want to report an accident, pushing the emergency key on the radio will set bit 3 and all D-STAR radios within range will open squelch and their volume to be set high.

The flag field bits 0, 1 and 2 are used for transmission control. They implement functions like ACK, ARQ and repeater control.

One of these repeater control functions is repeater lockout. Repeater lockout is used mainly to block illegal stations. A D-STAR repeater can hold a black list of call signs that have consistently violated repeater and/or FCC rules. If a blacklisted station calls the repeater, the repeater does not repeat the message but instead calls the offending station back with the lockout bit set. The offending station's radio will then display a message indicating that it is blocked from the repeater. So now, it is not necessary to shut down the repeater for everyone when one individual is misusing the repeater.

Another important field for understanding the capabilities of the system is the ID field. Understanding the ID field is important because it shows the great flexibility available in the system calling capabilities. The first thing to notice is that the D-STAR protocol automatically IDs at every transmission. This easily meets the FCC ID requirements for ID at start, end and every 10 minutes of transmission.

Next, to understand how the four ID fields work, Table 2 illustrates the contents of each field if KC7YXD were to transmit on a fictional D-STAR Evergreen Inter-tie system. It is not necessary to always fill in all the ID fields. If you respond to a CQ or a call directed at you, your D-STAR transceiver will automatically fill in the fields for you.

Digital Voice

Codec—Those of you who had a chance to see the D-STAR presentation at last year's Dayton Hamvention or at the DCC in Denver last fall may recall that the Digital Voice mode occupied 8 kHz of bandwidth using the ITU G.723.1 Codec standard. At that time, the two codecs were undergoing field trials; today the JARL has selected AMBE as the standard. The two standards under consideration were the ITU standard G723.1 and a Digital Voice Systems proprietary codec that uses the AMBE algorithm.

ITU G.723.1 uses an ACELP (algebraic code-excited linear prediction) algorithm that generates a 5.3-kbps data stream. With an algorithm delay of 37 ms, the total wireless-communication-throughput delay is a little over 100 ms, quite reasonable for half-duplex communications.

AMBE stands for advanced multi-band excitation. AMBE can use different levels of compression to trade off voice quality and bit rate. Tests show that at the 2.4-kbps data rate, the voice quality was at least as good as the higher-data-rate ITU G.723.1 over real radio links. The algorithmic delay is only slightly longer than G723.1 (44 ms), so the factor-of-two improvement in data rate (and spectral efficiency) comes with no noticeable latency increase.

The data-rate reduction from the AMBE codec is particularly significant because of worldwide pressure from regulatory agencies to reduce the occupied bandwidth of voice communications sufficiently to allow 6.25-kHz signal spacing. When using a modulation scheme sufficiently robust to give reliable communication in mobile and portable applications, only the AMBE data rate meets this signal-spacing requirement.

The decision between codecs is complicated by the fact that G.723.1 is an open public standard codec whereas AMBE is the patented intellectual property of Digital Voice Systems. Unlike many companies, however, the present owner of this technology supports the Amateur Radio community and is willing to sell these parts in small quantities.

Table 3—AMBE Vocoder-Based Systems

Inmarsat
Thuraya
Iridium
APCO Project 25 (IMBE)
G4GUO & G4JNT HF Digital Voice System

The JARL is not alone in deciding on AMBE for its high voice quality and very low bit rate. Table 3 shows several digital systems that have standardized on this codec technology. For instance, the Telecommunications Industry Association (TIA) selected DSVI's codec technology over CELP and other codecs for the APCO Project 25 North American land-mobile radio-communication system. This is particularly significant because at least two Amateur Radio groups are evaluating Project 25 radios as an alternative digital radio standard for amateur usage.

Fig 3 illustrates the bit pattern used in the digital voice mode when the AMBE codec is used. As mentioned before, the radio header is identical to the high-speed digital mode radio header and so will not be discussed here.

The most interesting part of Fig 3 is that the digital-voice data frames are interleaved with data frames. These frames are currently reserved by the D-STAR standard with no dedicated usage by the system overhead. This means that the system is capable of supporting a 2400-bits/s data stream from a user application while the user is talking on the system! Notice that the D-STAR system itself provides no error detection for this data, so it would be up to the user's application to provide error detection and error correction. This and other overhead would decrease the end-to-end data rate slightly; but if radios are built to exploit this capability, hams could potentially add many interesting features to the D-STAR system.

What is not shown in Fig 3 is that the frame and sync fields are repeated often so that the errors between the transmitter and receiver clocks can be

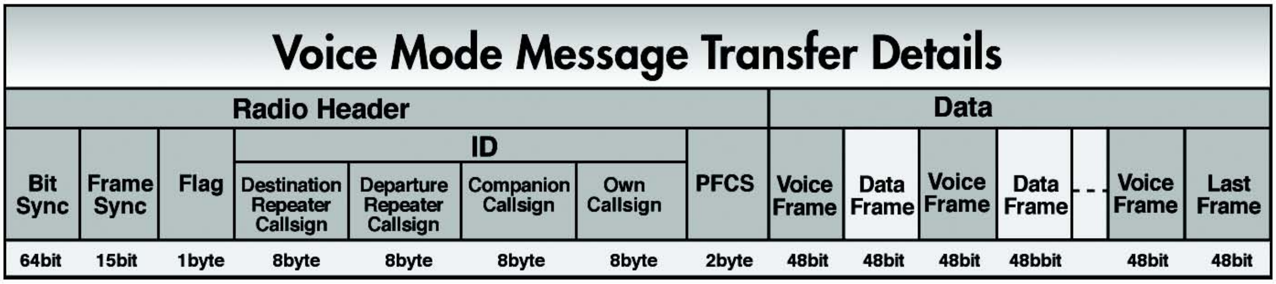


Fig 3—Bit pattern in digital-voice mode.

corrected without requiring a master clock signal. It also means that another amateur can tune into the middle of a contact and listen to the conversation without waiting for the sync frames of the radio header at the next over.

Modulation

Several modulation methods were investigated during the development of the D-STAR standard. Modulations tested included GMSK, FSK, 4-FSK, MSK and QPSK. GMSK has been selected for the backbone line between repeaters. The standard for the portable and mobile transceivers may include more than one modulation format.

Gaussian minimum-shift keying (GMSK) and quadrature phase-shift keying (QPSK) are the two finalists. A third, 4-FSK, has been recently proposed as an alternate standard and is now under investigation. The reason for the delay is that selecting the best modulation for D-STAR real world applications is not a trivial exercise. In real mobile communications systems, the link between a moving node and a base station will be subject to multipath, which results in Rayleigh fading. This will have a significant effect on the resultant BER performance, possibly increasing the required C/N for a specific BER by as much as 10 dB.

QPSK is commonly used in fixed-link communication systems. Under ideal conditions, QPSK would give better performance than GMSK or 4-FSK and its higher spectral efficiency is obviously attractive. However, QPSK's higher spectral efficiency also leads to higher susceptibility to transmission impairments such as multipath and phase hits. Yet, the biggest disadvantage of QPSK is the need for extremely linear power amplifiers to avoid spectral growth—what we Amateurs call splatter.

The front-runner at the time this article is written is GMSK. In its favor, GMSK is a well-proven technology, and probably the most commonly used digital modulation in the world for portable applications (see Table 4). GMSK has two basic advantages. First, it is more robust than QPSK to common transmission impairments. Second, GMSK, as a form of FSK, has constant amplitude and can therefore use very efficient class-C power amplifiers. Third, GMSK is not as sensitive to frequency errors between the transmitter and receiver. Because no master frequency reference is available in the D-STAR system, tuning errors on a 1.2-GHz signal can be substantial, particularly with the extremes of temperature found in portable operation. The alternatives are to suffer the expense of a precision frequency reference in all the radios or adopt a modulation method like GMSK that is more tolerant of frequency errors.

Table 4—GMSK is used in Systems Worldwide

GSM cell phone
DECT
Cellular Digital Packet Data (CDPD)
Mobiltex

However, GMSK is not so spectrally efficient as QPSK. For instance, at 128 kbps, GMSK with a BT product of ½ occupies a bandwidth of 135 kHz. For the same data rate, QPSK requires only 83 kHz.

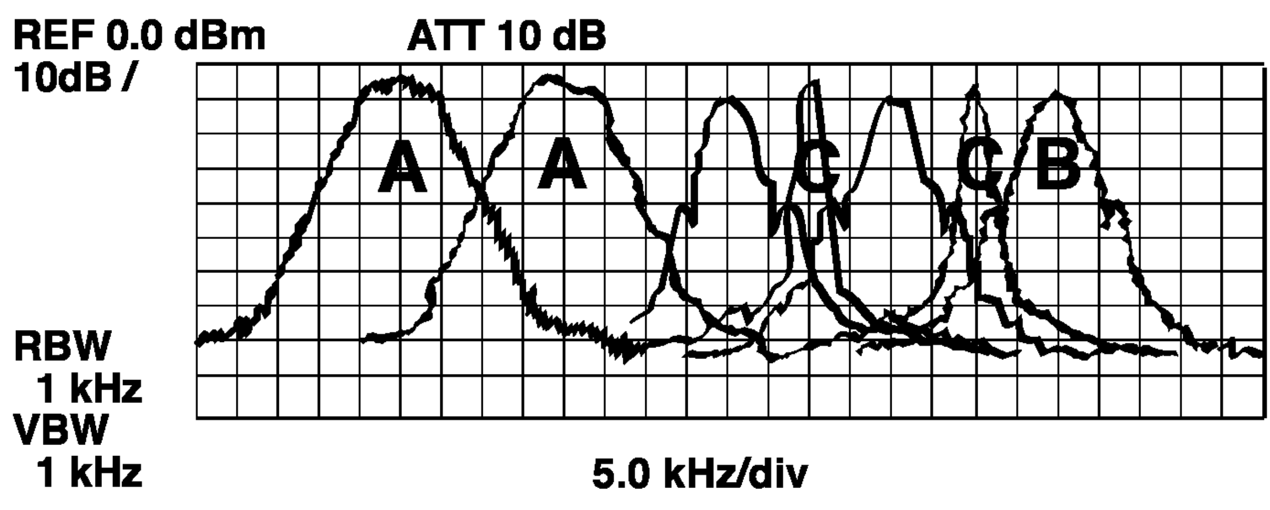
The best solution is probably to use a codec with the AMBE algorithm described earlier that reduces the data rate as far as possible and then use GMSK for more robust communications, but the tests still continue.

Fig 4 is a somewhat busy graph that dramatically shows the difference in occupied bandwidth. The existing FM system bandwidth can be determined by Carson's Rule to be about 16 kHz. While not all combinations of modulation and codec algorithms are shown, you can clearly see that you can fit many more digital voice contacts into the same spectrum.

Repeater

The digital voice mode is half-duplex with a 20-MHz offset between transmit and receive frequencies. High-speed data is simplex.

As shown in Fig 5, much of the repeater site function is to provide a



A: Conventional FM
B: GMSK and ITU G.723.1
C: GMSK and 2.4 kpbs AMBE
Unmarked: QPSK and ITU G.723.1

Fig 4—Occupied bandwidth of digital radio.

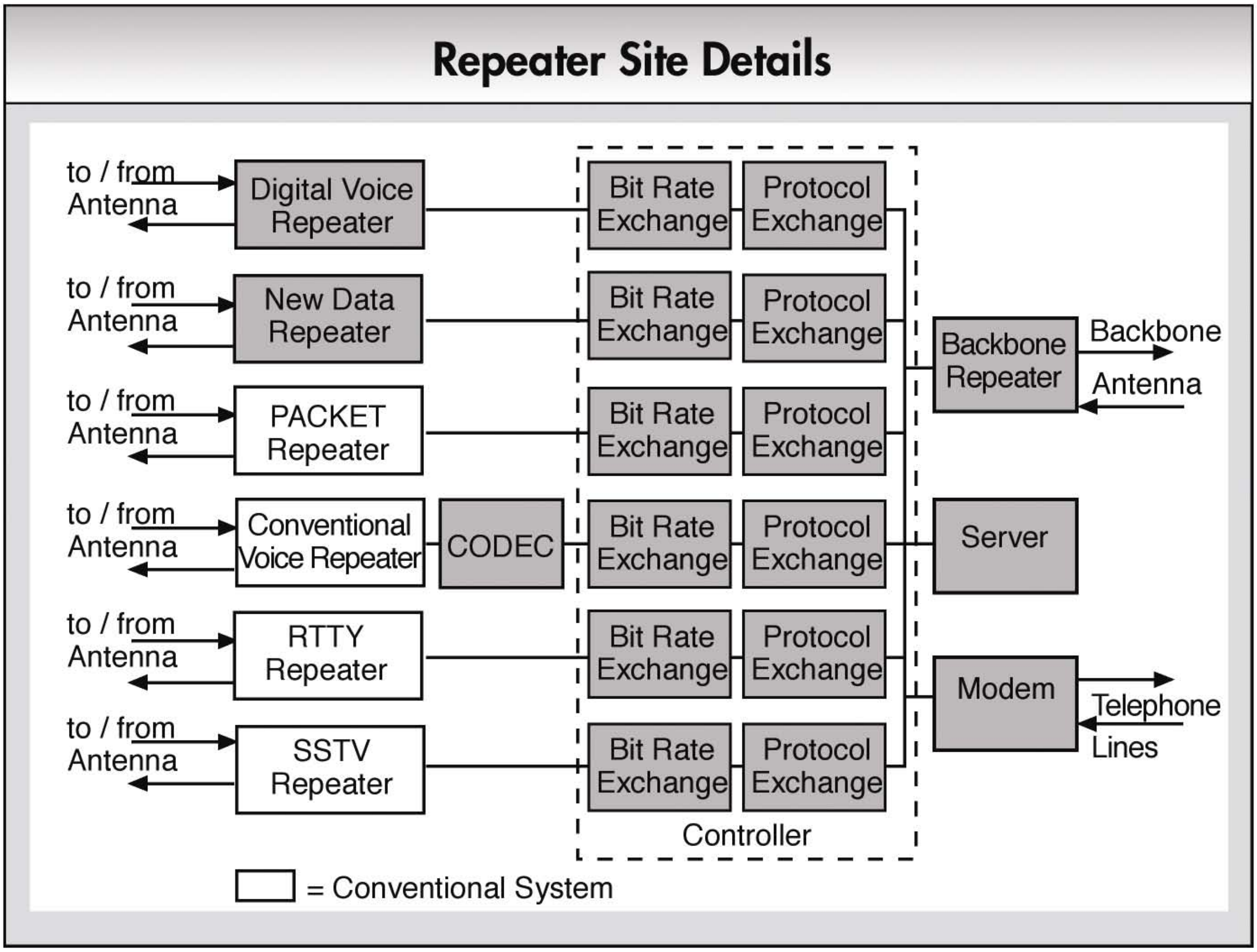


Fig 5—Integrated site with analog and digital radio repeaters, high-speed backbone and Internet connection.

gateway to other repeaters, both at other sites and to repeaters on different modes. The repeater also provides an interface to the repeater backbone link to the Internet if desired. The system is designed to support remote control of the repeater over radio and/or landline links.

Repeaters could be linked via the Internet instead of the backbone, but because of bandwidth limitations, much of the high-speed multiple contact capability would be lost.

Repeater Call Sign Protection

To protect repeaters from co-channel interference, CTCSS tones are used to prevent interfering signals from triggering the repeater. In the D-STAR system, the digital header contains the call sign of the repeater to be accessed. If the repeater does not see its call sign in the message, the repeater is not opened.

The Backbone Repeater Link

One of the major advantages of the D-STAR system is the full-duplex 10-MHz-bandwidth backbone link between repeaters. This wide bandwidth allows multiple voice and data contacts to occur simultaneously on the link. An analysis of the frequency of use of data and voice communications demonstrated that a 10-Mbit/s full-duplex link would support the needs of up to 12 linked repeaters.

The high-speed data and digital-voice data streams from multiple repeaters are multiplexed into a single data stream according to the asynchronous transfer mode (ATM) standard. This 10-Mbit/s data stream is GMSK modulated onto a 10-GHz carrier, resulting in an approximately 10-MHz wide signal.

The ATM cell is made up of a short 53-byte packet that consists of a 5-byte header and a 48-byte payload. The ATM cell is sent to the required destination according to the preset list that is set by the ATM switch set at each repeater site. Because the priority level can be designated in the header, voice signals arrive in real time. This avoids the delays that happen with VoIP on the existing Internet Protocol.

Backbone field tests have been carried out with a 36-dB-gain parabolic antenna and a 1-W transmitter. Heavy rains in southern Japan of more than 12 inches per hour limit the practical distance that the repeaters can be separated. It was found that taking into consideration these extreme weather conditions, the maximum range for uninterrupted communications is about 12 miles. Obviously, the fog and rainfall at the location and the acceptable probability of communication interruption dramatically affect this number.

Notes

[1]"JT44: New Digital Mode for Weak Signals," (World Above 50 MHz) *QST*, June 2002, pp 81-82.

[2]D. Smith, KF6DX, "Digital Voice: The Next New Mode?" *QST*, January 2002. For a discussion of MOS, see the sidebar, "How Do I Sound?" on pp 29.

[3]K7NWS is a Seattle, Washington, repeater and KB7WUK is a Portland, Oregon, repeater on the Evergreen Intertie. These examples assume an identical system to the Evergreen Intertie but based on D-STAR. □□