Digital Voice Communication

By Andy Talbot, G4JNT *

THIS EXPERIMENT started after a phone conversation with Charles Brain, G4GUO, where we had been discussing various data modes used by the military and other commercial users on the HF bands. One of the modes that has been in use for several years is digitised voice, transmitted in a bandwidth comparable with normal analogue voice communications and making use of existing transmitters and receivers. I must have said something like "it would be nice to try this on the amateur bands", because after the phone call finished Charles then went away and had a long think.

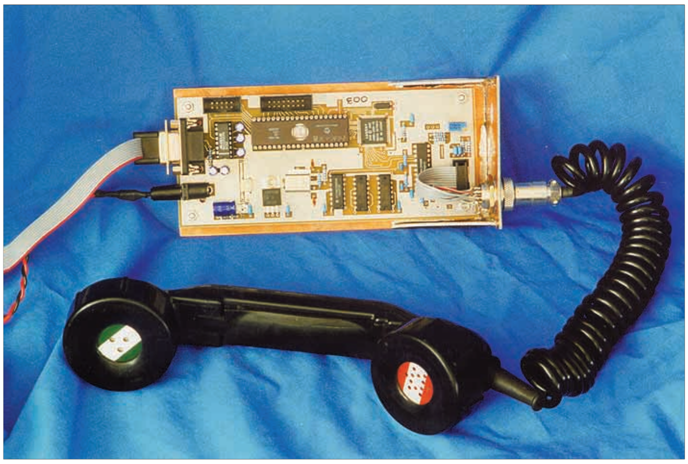
DIGITAL TECHNIQUES

MEANWHILE, WE need to digress a little to cover digital communication techniques and the problems given by the HF communications environment in order to fully appreciate why one type of datacomms technique is employed over any other.

When properly implemented, digital communications can show considerable advantages over their analogue counterparts with regard to quality and robustness over noisy transmission media - compare the quality of CD music recordings with the old vinyl or tape system and the new digital telephone network with the old system. But there are several major issues to be resolved before the conversion is made.

To digitise an analogue signal such as voice, it first has to be sampled, ie turned into a series of numerical values. Theory dictates that the sampling has to be at a rate at least twice that of the highest frequency component present - the Nyquist Criteria - and there must be no components present at more than half the sampling rate otherwise these will appear as spurious components at other frequencies causing distortion. This is known as aliassing, and the high frequency components need to be removed by conventional filtering before digitisation. For a voice signal as transmitted using telephone or SSB, the frequency range of 300-3300Hz is usually taken and dictates a sampling rate of at least 6.6kHz. In practice, to ease the anti-aliassing filtering, a sampling rate of 8000 is often adopted.

Since the analogue signal can have an infinite number of instantaneous amplitude levels, these cannot be represented exactly and it is necessary to choose a suitable number of levels to be used to represent the signal (instead of levels, it is more convenient to think of the number of binary bits needed to give the quantising needed, eg 8 bits gives $2^8 = 256$ absolute levels, 16 bits per sample gives $2^{16} = 65536$ levels). The effect of the random instantaneous error at each sampling point is to add a noise component to the signal - referred to as Quantising Noise. There is a simple rule of thumb that can be applied here: The best Signal-to-Noise ratio (S/N) that can be achieved is given simply by S/N (dB) = 6n - 1.75, where 'n' is the number of bits of quantisation and '1.75' is a fiddle factor (and sometimes takes slightly different values in various text books), but S/N is approximately 6n. If a figure of 40dB is taken as 'good communications quality' then 8 bit quantisation, allowing a bit less than 48dB S/N, would be adequate. This is the system adopted on the public telephone network, although in slightly modified form.



The AMBE module.

We can see that for 8000 samples per second, sampling at 8 bits per sample, a total of 8 x 8000 = 64000 bits per second (b/s) is generated [1]. The digital telephone network has a wide enough bandwidth with optical fibres and microwave links to be able to pass 64kb/s directly, but the radio communications link does not have this luxury! At HF we want to be able to pass digital voice over a bandwidth comparable with SSB, ie 3kHz; at VHF, if NBFM is taken for the standard channel width, we can increase this figure to 12kHz, but in order to preserve the enhanced voice quality that good S/N FM can give, more quantising levels should really be used.

Although it is theoretically possible to transmit 64kb/s in a 3000Hz bandwidth, the S/N ratio required to do so with a sufficiently low error rate is very high - around 64dB according to Shannon's Information Theorem - so other techniques have to be adopted to transmit digitised voice signals. Ideally a data rate comparable with the RF bandwidth is wanted for optimum transmission in signal-to-noise ratios that would be just acceptable for poor speech quality, ie around 3000 bits per second for 10-15dB S/N in 3kHz.

VOICE ENCODING

THE TECHNIQUE adopted is to encode the voice to reduce the number of bits per second needed for transmission. There has been a considerable amount of research into various techniques for doing this over the last 10 or so years and some very effective compression schemes are now available. The techniques are too complex to cover in any detail here, but usually involve modelling the human voice tract and coding the various elements, such as voiced and unvoiced sounds. Various schemes such as Code Excited Linear Prediction (CELP) and various other 'ELPs' have come to the fore for voice. As an example, GSM mobile phones use a technique that allows transmission at 13000 bits per second. Whatever technique is used for voice encoding (vocoding), there is usually a trade off to be had between data rate generated and the quality of the resulting speech. Some of the early systems had a very synthetic sounding Dalek-like result, but modern variants provide very much better toll-quality [2] speech. GSM at 13000b/s is an example that has been around for several years now.

Previously, G4GUO had written some voice encoding software for a DSP system based on published algorithms using these techniques, but it was not too successful and

* *15 Noble Road, Hedge End, Sothmpton SO30 0PH. E-mail: actalbot@dera.gov.uk*

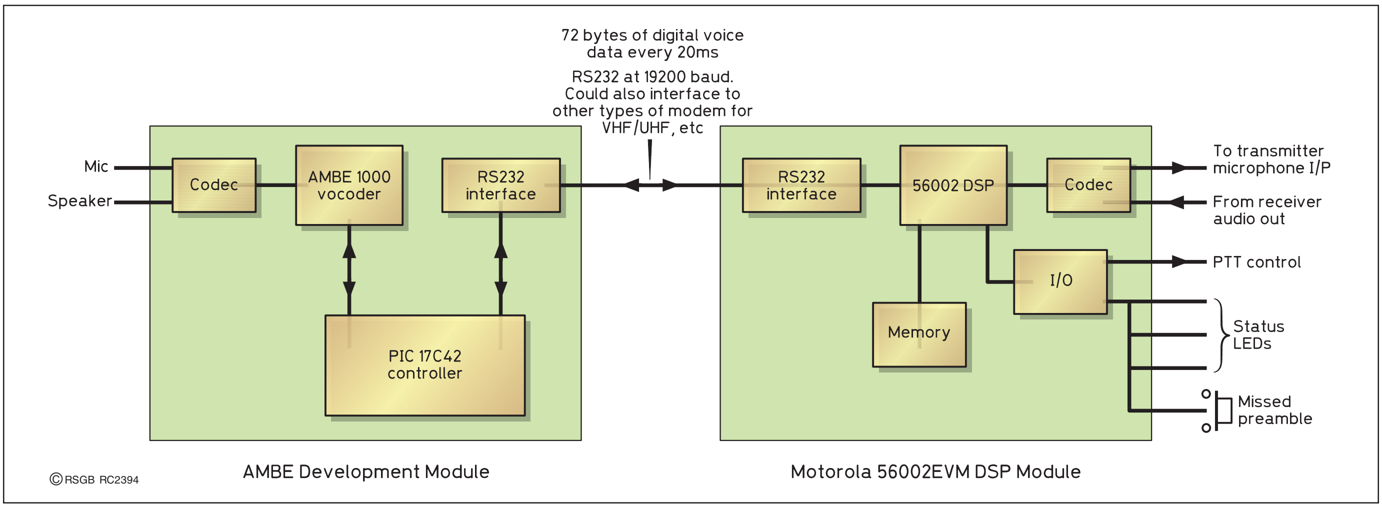


Fig 1: Block diagram of the Digital Voice hardware.

as many of these algorithms are patented using them could run into all sorts of problems or prove expensive in licensing fees! Furthermore, CELP uses a large codebook, too big to fit on any of the low cost DSP development cards.

A comprehensive search of the literature on vocoding techniques threw up yet another new system called Advanced MultiBand Excitation (AMBE) that appeared to offer major improvements over earlier systems. This moves away from the concept of modelling the voice tract, and instead models the spectrum of the signal every 20ms. Not many technical details appear to be available to date as it is still a commercial system, but the results of test programmes show the technique to be better than any of the ELPs, and has thus been adopted for at least one of the new satellite-based mobile phone systems. Much more importantly for us, a single chip solution is available for converting from microphone input to encoded digits, so rather than try to write vocoding DSP software based on published algorithms, it was decided to just buy a chip to do the job.

The AMBE1000™ chip by Digital Voice Systems Inc, (available from Lucent Technologies) implements the whole process and provides the user with an extensive trade off of output data for different levels of link quality, as well as Forward Error Correction (FEC). A range of output data rates from 2400 bits per second all the way up to 9600b/s in 50b/s increments is available, along with a range of user selectable FEC options. Forward Error Correction is essential at HF to overcome burst and CW interference and the effects of multipath - the manufacturers have implemented FEC within the chip that is optimised to the needs of the AMBE technique and in fact claim that "Any other separately implemented FEC technique cannot be as good as our one". Eventually the data rate adopted was 2400b/s of voice data plus 1200b/s of FEC, giving a total of 3600b/s to be transmitted over the RF link. The IC outputs samples every 20ms and can be regarded as a real time system in this sense. Any 20ms samples that get lost just create glitches in the speech that cause minimal disturbance and can often go unnoticed. In fact, during early phases of the experiment, the FEC got left out and tests carried on for some time over our 40m test link before we noticed no error correction was being applied.

VOCODER MODULE

IN USE, the AMBE chip has to be programmed at turn-on to set the operating conditions. The easiest way to do this was to include an on-board PIC microprocessor. G4GUO produced a PCB which includes the AMBE chip, a Coder-Decoder (CODEC) and PIC17C42 controller with other peripheral components. A photograph of this is shown left.

The digitised output samples, at a net overall rate of 3600 bits per second, are sent via an RS232 interface to the modem in packets of nine bytes at a time for each 20ms frame. The data rate used for this part of the link is 19200 baud. If you do the sums on this, it is possible to see there is a lot of spare capacity here for users who want to use the development board for their own purposes. An example of this would be for the inclusion of data and control signals. The PCB was produced as a development tool and thus the controller chip chosen is capable of considerably more processing capability than that needed to just control the AMBE and convert to RS232 format. In fact there is enough processing capacity in the PIC to include encryption / decryption software, but it is left to other users to make use of this capacity! All input/output ports for the PIC are available on connectors.

The AMBE chip is a 100 pin device using the surface mount TQFP package with a lead out spacing of 0.65 mm. Mounting this device caused a few headaches, but use of a good quality magnifier and careful soldering followed by careful inspection resulted in success first time. All the other devices on this board are either conventional surface mount or through hole mounted. **Fig 1** shows a block diagram of the system

RF MODEM CONSIDERATIONS

SO FAR WE HAVE digitised the voice signal and added redundant information to correct potential transmission errors resulting in an overall, but packetised, rate of 3600 bits per second. We now need to turn this data stream into a continuous modulated audio waveform, suitable for feeding to the input of an SSB transmitter for upconversion to the final RF signal for transmission.

There are a number of different ways of modulating an RF waveform for data transmission. Probably the most popular system amongst amateurs is Frequency Shift Keying (FSK), where one audio tone is used for a logic '1' and another tone for logic '0'. In terms of bandwidth utilisation FSK is not very efficient - either a frequency shift significantly wider than the data rate has to be used for reliable copy in noisy conditions, or as the shift is narrowed to preserve bandwidth so errors occur more frequently. The two systems where this trade off is evident is in amateur and commercial RTTY, where 170Hz frequency shift or wider is used for 50-75 b/s data. The converse is HF packet radio, where 200Hz shift is used for 300 b/s with its very poor reliability in noisy conditions. One advantage of FSK is its simplicity of implementation based on filters and phase locked loops, hence its popularity in the past, but with the increased use of Digital Signal Processing (DSP) everywhere, this simplicity is no longer a real issue.

A much better solution is Phase Shift Keying (PSK), where instead of changing the transmission frequency for binary 1s or 0s, the phase is reversed, or effectively the signal is inverted, between 0 and 1 states. It is possible to show theoretically that there is at least a 3dB improvement in S/N versus error rate performance given an 'ideal' demodulator for each modulation, and with simple FSK modems very much better than this is possible in practice. PSK to replace keyboard-to-keyboard RTTY has appeared on the amateur bands recently in the form of PSK31 (see the article in December 1998 and January 1999 *RadCom* by Peter Martinez, G3PLX). For very nearly the same data rate as RTTY, the bandwidth needed has shrunk from around 200Hz to 20Hz, with a corresponding increase in reliability and error rate. By using four phase states 90° apart instead of two, it is even possible to encode two bits at a time without increasing the bandwidth, but this does incur a 3dB penalty as the transmission power is now shared between twice as many bits in a given time. This technique, Quaternary Phase Shift Keying or QPSK, is available in PSK31, where it is included as an option for adding the extra data needed for Forward Error Correction in noisy environments. A properly filtered PSK signal has a bandwidth that is can approach half the baud rate - in fact PSK31 is optimised to do just this - but if not implemented correctly with waveform control and filtering, the bandwidth of the transmitted signal can easily spread alarmingly in a manner analogous to CW key clicks.

Fig 2 shows how raw QPSK modulation by a rectangular data signal would look, showing how the data bits are coded two at a time. This signal as shown in the diagram is unfiltered and would have a very wide frequency bandwidth due to the sharp phase transitions.

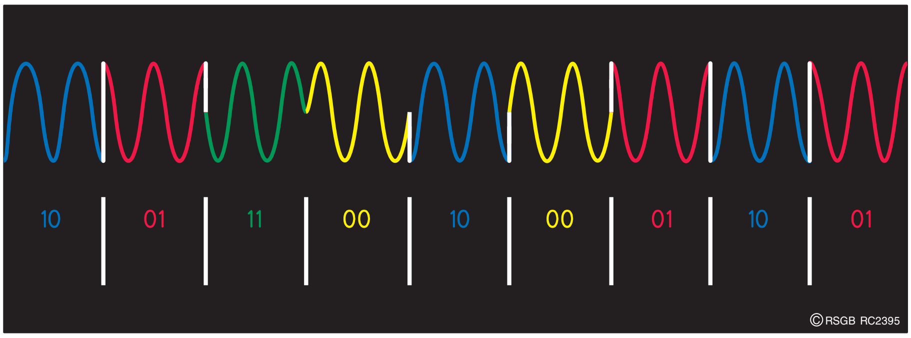


Fig 2: Illustration of an unfiltered QPSK signal, showing how pairs of bits can be coded onto a carrier.

Here we also need to introduce the concept of the Baud rate as opposed to bits per second, which is the rate of information change on the final transmission, or the symbol rate. For BPSK and FSK systems the Baud rate is the same as the bit rate, ie one bit per baud, and the terms are often used interchangeably. Where two bits are transmitted per symbol, as for QPSK, the Baud rate is half that for the given bit rate. It is possible to extend the transmitted bit rate higher by simply increasing the Baud rate to suit the data to be transmitted. For the 3600 b/s needed for the digitised voice experiments, either binary PSK at 3600 baud or QPSK at 1800 baud would be adequate - the QPSK signal at potentially 1800Hz bandwidth could even be transmitted unmodified over SSB radios. However, whilst this technique is ideal for UHF or 'clean' VHF links, there are particular characteristics on a typical HF transmission path that make simple high Baud rate signals very prone to errors and frequently unusable.

REFERENCES

[1] As an aside, consider CD music recording. A sampling rate of 44100Hz is chosen to allow a 20kHz maximum audio frequency; 16 bit quantisation is used to give a dynamic range in greater than 90dB. With two independent channels for stereo, this results in a data rate in excess of 1.4Mb/s.

[2] The term 'toll-quality' comes from America, and means a quality of telephone transmission that people are prepared to pay for. ♦

To be continued...

RadCom Helplines for members

● AJ Spayne, RS176038, is looking for a wave change switch for an **AR88D**. Also, a service manual and circuit diagram for the **Sony CRF-1** communications receiver. All costs covered. AJ Spayne. Apartado 167, Carcavelos 2776-902, Portugal.

● Dana Jensen collects KW equipment ("virtually unsold and unheard of in North America"), and is looking for a **KW2000E** and **KW600**. E-mail: danaj@home.com

● Jerry, GM4CAZ, would like to know if the **Amateur Radio Mobile Society** still exists. Tel: 0973 937258. E-mail: jclefever @aol.com

● Charles, G4RTV, requires a copy of the operating manual for the **Yaesu FC-902** ATU. All costs covered. G4RTV, QTHR. Tel: 020 8402 2108.

● Richard, G4PRI, requires the loan of the instruction book for the **Katsumi EK-150** electronic squeeze key. All costs covered. G4PRI, QTHR. Tel: 01895 270772.

● Harry, G3MFW, is looking for a manual or the circuit of the **Sony-Tektronix 326** portable oscilloscope, and the 18V power unit for his **Toshiba T2200SX** notebook computer. Costs gladly covered. G3MFW, QTHR. Tel: 01726 73608.

● SP Shackleford, G2HAX, is looking for information on and the circuit diagram of the **Heathkit HM-102** SWR meter. All costs covered. G2HAX, QTHR. Tel: 0118 941 0235.

● Don, G3WDY, would like to borrow a manual for the **Cossor 343** Ganging Oscillator. All expenses paid, and originals speedily returned! G3WDY, QTHR. Tel: 020 8653 4738.

● Sam, G3HVI, is trying to locate a valve type **815** or **QV04-20**, for use in a restored RCA AM transmitter. G3HVI, QTHR. Tel: 01782 393349.

● Ron, G0GHX, is looking for the source of an RF hybrid module **OM631**, as originally fitted in the **Advanced Communications** 6-way TV/FM distribution amplifier. G0GHX, QTHR. Tel: 01202 880194.

● Don, G0ACK, is looking for instructions, information and the circuit diagram of the **Tradiper TE-15** Grid Dip Oscillator. G0ACK, QTHR. Tel: 020 8845 9575.

● Roy, G3JNM, requires a copy of the instructions for assembly and adjustment of the **Hy-Gain 12AVQ** vertical antenna. G3JNM, QTHR. Tel: 01204 843999.

● Alan, GM4IIR, is looking for advice on how to feed a **132ft long wire** antenna. GM4IIR, QTHR.

● Arnold, G8AHE, would like technical help with an **IBM 8513** video monitor. G8AHE, QTHR. Tel: 0121 458 2406.

● Bill, GW3DGT, is looking for a copy of the instruction manual for the **Sharp EL-1615** electronic printing calculator. Also, technical data on the circuitry. W Barrett, 'Stevina', Ludchurch, Narberth, Dyfed, SA67 8JF. Tel: 01834 831369.

● Anthony, M1AVE, is looking for the circuit diagram and alignment details for the **Revco RS2000** AM/FM scanner. Also, any information on the **Anglian 1000** transceiver, especially on the interconnecting cable to the power supply. Tel: 01908 373114.

● VC Whitchurch, G4HSA, would like a copy of the service manual and circuit diagram of the **Philips PM3240** oscilloscope. Also, operating instructions, as in the lid, for the **Wayne Kerr CT492** (mil type) LCR bridge. All costs covered. G4HSA, QTHR. Tel: 01761 414169.

● Ron, G4MNB, is looking for a copy of the manual for the **Yaesu FRG-7** receiver, plus any modifications. All costs reimbursed. G4MNB, QTHR.

Digital Voice Communication

*Final part, by Andy Talbot, G4JNT **

IN PART ONE digital techniques were discussed, and this led on to the subjects of voice encoding and modem considerations. The problem is that conditions on the HF bands make simple high bit rate modems prone to errors and frequenctly unusable.

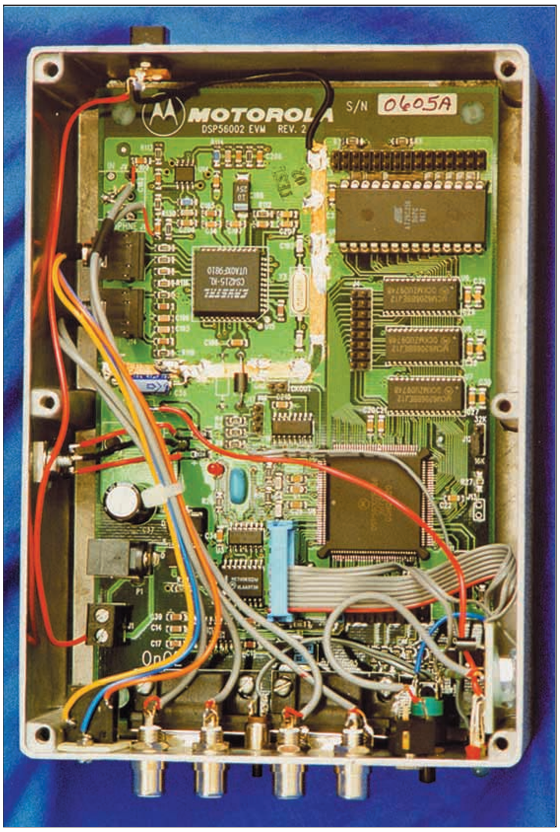
SKYWAVE

THE GREATEST PROBLEM is multipath. Skywave signals frequently arrive after several ionospheric hops, with the same instantaneous element of signal arriving at different times after travelling different distances. For a signal such as SSB voice, these two or more signals will cause alternate cancellation and reinforcement, giving the characteristic multipath fading as a notch passes through the audio passband. Differences in arrival time of typically up to 5ms are often observed, and in poor propagation conditions can reach a lot more than this. The effect on digital signals can be far more catastrophic than it is for speech, because as a particular 'bit' of information arrives at several different points in time, it can easily land on top of another bit arriving via an alternative path. This mixing up of received information causes *intersymbol interference* and is the major cause of bit errors on what might otherwise appear to be a good link with a strong signal. As a way round this, if we can arrange for one symbol of information (ignore what that might mean for now) to be sent in such a way that when mixed with itself delayed by up to, say, 5ms, no corruption is encountered, then the error rate due to intersymbol interference can be reduced or even eliminated. One method is to reduce the baud rate to such an extent that the 5ms multipath period is insignificant - a figure of 20ms is often taken in practice, resulting in 50 baud signals. It is no coincidence that the data rate adopted for RTTY signals for many years has been in the 45 to 75 baud region!

To reduce our 3600 bits per second to 50 baud signalling means trying to compress 72 bits into one symbol. While there are some direct techniques of doing this, such as Quadrature Amplitude Modulation these are prone to other types of errors and inefficiencies, and another system is needed that is more resistant to in-band interference. That technique is as follows:

Instead of using a single carrier modulated with a complex multilevel waveform, we use a large number of multiple carriers, each one modulated with a simple waveform. If there are N carriers, each one independently modulated with 50 baud QPSK then it is possible to transmit data at 2.N.50 bits per second. The spacing between each carrier has to be consistent with the baud rate and a carrier spacing equal to at least the symbol rate is required. If we do a few calculations, it soon becomes evident that many solutions are possible for 3600 B/s in a voice bandwidth [2].

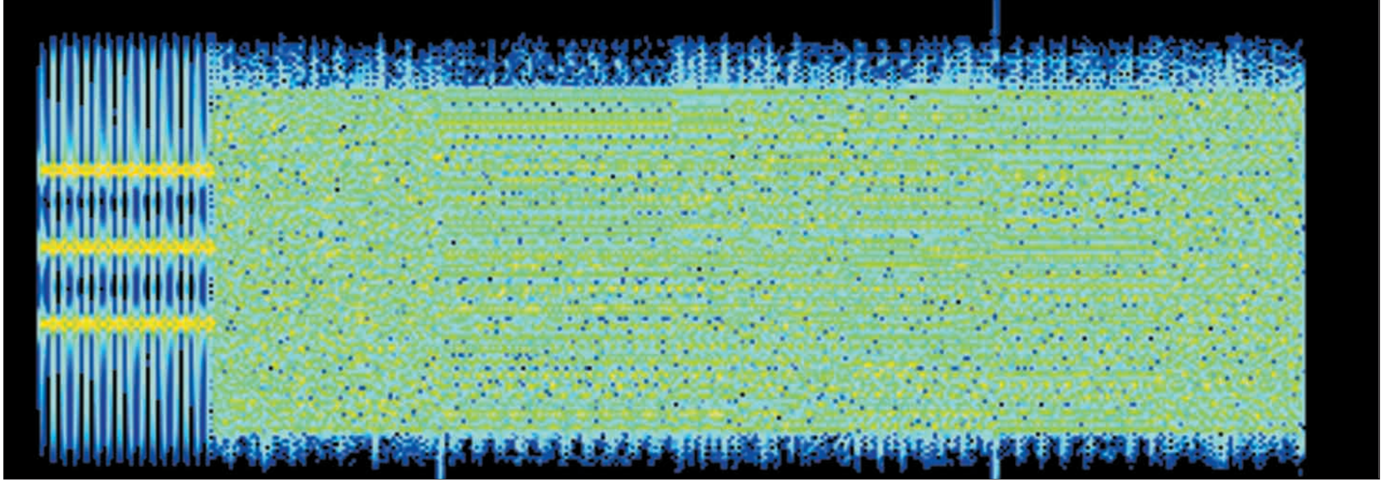
An existing military system in widespread use makes use of 39 parallel tones for digitised voice as well as data communication. We initially wondered if this standard could be adopted (the waveform is in the public domain) but came to the conclusion that the narrower SSB filters characteristic in amateur grade equipment would not pass the signal



The EVM board, mounted in a diecast box.

*15 Noble Road, Hedge End, Sothmpton SO30 0PH.
E-mail: actalbot@dera.gov.uk

Fig 4: Spectrum of the Digital Voice signal.



without excessive loss of the band edge tones. Parallel tone modems are generally regarded as being more suitable for digital voice at HF than serial tone ones because of their failure mechanism in the type of interference present. Whereas serial tone modems result in long bursts of errors - which could lead to a whole section of speech being knocked out - parallel tone designs give shorter errors merely resulting in a few lost syllables.

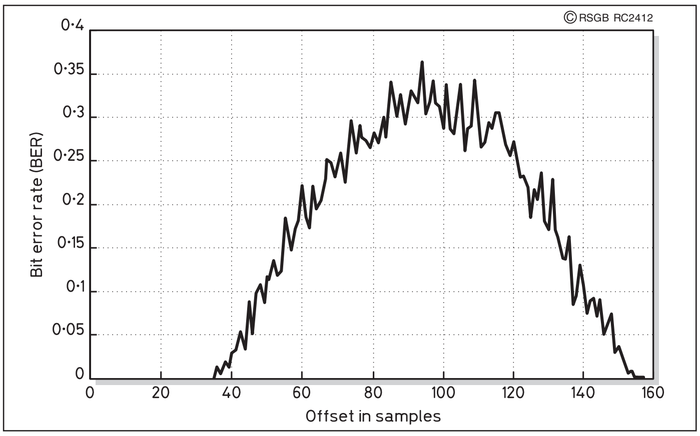


Fig 5: Sync Offset Error vs Bit Error Rate.

After mathematical simulations and on-air testing, the combination finally chosen uses 36 carriers each carrying QPSK modulated at 50 symbols/second. The 50 baud period fits in very well with the data output from the AMBE vocoder, which outputs its data in bursts every 20ms allowing one burst of data to be spread across all tones equally - 72 bits per baud - and allowing easier synchronisation between the two subsystems. The carrier spacing is set at 62.5Hz giving a total signal bandwidth of 36 x 62.5 = 2250Hz, which fits comfortably inside the typical amateur SSB filter width of 2.7 - 2.8kHz. The 62.5Hz spacing adopted here is not as narrow as could be used - 50Hz is feasible for 50 baud signals - but does allow the luxury of a 4ms guard interval in the decoding process (the difference between the reciprocal of the spacing 16ms, and the baud period) giving even further resistance to multipath interference. In fact, without the guard period, the modem has no tolerance to multipath interference of equal signal strength, so the overhead in bandwidth caused by the use of a guard period results in a reduced error rate.

Another advantage to the parallel tone approach is that with the Forward Error Correction bits added, it is possible to lose a few of the tones and still be able to send valid data. This means that the 2.3kHz wide digital voice signal can actually sit on top of several CW signals, each potentially stronger than our wanted signal, and be oblivious to them.

To allow the modem software to lock up to the received waveform it is necessary for the receiving side to achieve coherent frequency and timing lock with the signal received. To enable this, each transmission is preceded by a 700ms preamble made up of three tones of 1000, 1500 and 2000Hz, transmitted simultaneously. Each tone is BPSK modulated with a known pseudo random data sequence that the receive modem can lock to and generate its timing reference. Only one modulated tone is actually needed for this lockup function, but to allow for the case of the preamble being corrupted by interference, three tones carrying the same data are transmitted. The whole preamble sequence is generated automatically when the PTT button is pressed and there is a resultant delay before voice transmission starts, but unless your own signal is being monitored simultaneously, the effect will only be noticed as a delay in transmit receive switch over.

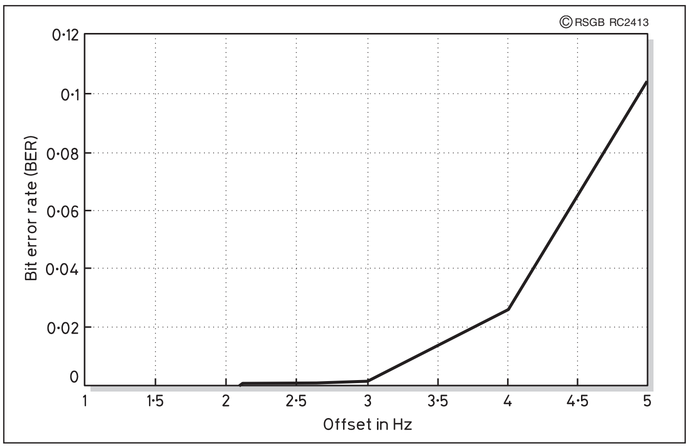


Fig 6: Offset Frequency Error vs Bit Error Rate.

The picture of the signal spectrum is shown in **Fig 4**, which shows in spectrogram form the three tone preamble followed by a period of data transmission. It is not possible to make out all of the individual 36 tones as the modulation sidebands from these overlap when viewed in this form, but the general 'brick wall' shape of the spectrum can be discerned.

TUNING & TIMING ACCURACY

IT IS OBVIOUS that any frequency error between transmitter and receiver must be reduced to well below the value for the tone spacing and that the start time of each symbol period needs to be known precisely in order for its phase during the symbol period to be measured. The receiver tuning needs to be within 62Hz of the correct frequency point, for residual errors below this figure the modem automatically corrects and it is the function of the preamble acquisition phase to do this. In noisy conditions or where interference is present, the receive modem may not be able to make best use of the preamble information and timing/frequency errors can occur. From mathematical simulations during the development phase of this project, values for the Bit Error Rate versus symbol timing error and tuning frequency error were calculated and **Fig 5** shows how bit error rate varies with the regenerated timing error, expressed as an offset in number of samples of the 8000Hz sampling frequency. **Fig 6** shows the corresponding Bit Error Rate for residual tuning error.

It can be seen that the preamble acquisition phase is the most important part of the whole transmission cycle after the PTT is pressed, although some timing information can be generated based on the data stream this is not nearly as effective as using that available from the preamble. Should the preamble be missed the whole transmission could become garbled rubbish if there is no other way of acquiring timing and frequency information. Fortunately, the crystal oscillator used to generate the sampling rate is sufficiently stable to be able to use the previous timing information, and the final system provides the ability to use existing timing by pressing a button.

IMPLEMENTATION

THE AMBE VOCODER board has already been described as a stand alone unit in its own right. The parallel tone encoder is built around a low cost DSP evaluation kit, the Motorola 56002EVM, which is a single PCB module that contains the DSP chip, a CODEC for interfacing to the audio feed to the transceiver, an RS232 interface, computer interface plus programming tools. A photograph of the EVM module mounted in a diecast box is shown opposite. Thus the complete digital voice system comprises just the two PCBs plus transceiver. If a flash ROM is installed for non-volatile memory storage then, once it has been programmed with the digital voice modem software, the 56002EVM can be used as a standalone module without any need to subsequently load the software from a host PC. To complete the hardware system, the transmitters PTT line is controlled by an output from the EVM so that seamless operation is possible.

OPERATION IN PRACTICE

THE PATH between G4GUO and myself is 70km and 7MHz usually works reliably between us at most times of day. It is a high angle path and thus suffers from quite severe multipath during much of the day, providing a good test of the modem's capabilities.

At G4GUO's end an IC-706

transceiver was employed, while I initially used a home-built transmitter in conjunction with an RA1792 receiver, later to be replaced by an IC-746. Tuning error was not a problem, as we could both set frequency to well within 1Hz accuracy. Some of the initial tests were made with earlier variants of the Digital Voice system, such as 24 parallel tones, 36 tones with no FEC (by mistake!) and single tone preamble. These all worked to some extent, but were susceptible to QRM and multipath fading. Once the final configuration (as described) was adopted, it became very obvious how much the modem performance had improved. Operation was usually tested on a random basis, with the rigs left tuned to the 7MHz frequency and calls made spontaneously. A feature built into the modem means that a LED is illuminated once a preamble tone is detected. Even if left unattended, it was usually possible to see that one station or the other had called and a return call usually resulted in a QSO.

The only times that the link could not be made was when either the path was too weak even for comfortable SSB copy, or when strong impulsive type QRM caused the preamble sequence not to be recognised. In a number of cases, in the presence of strong co-channel CW signals, the FEC could cope with CW interference knocking out some of the parallel tones. The real-time nature of the link also showed itself. Cases of burst interference resulted in the loss of a few syllables of speech, or the occasional 'gargle', but very rarely was intelligibility lost. Occasionally the modem would not lock to the preamble, but copy could usually be re-started by pressing the button to use existing timing information.

The most effective and impressive demonstration was one evening in April when a QSO lasted for an hour and a half as the sun set. Copy started out as perfect, with no lost preambles or garbled messages. The multipath became worse as dusk arrived, so copy worsened slightly, but it wasn't until nearly dark when the link had almost faded out completely that the DV link became unusable.

Once some boards have been made up in the US, we hope to be able to try some transatlantic tests.

EQUIPMENT AVAILABILITY

THE AMBE PCB card was produced by G4GUO in small numbers, most of which have now gone to users. TAPR in the US have arranged bulk manufacture and assembly of the difficult to mount SMT chips. It is intended that a developers kit will be made available.

The 56002EVM module is available from a number of UK and US suppliers, including Farnell / Newark who can supply ex-stock. The necessary software will be available via the web [3].

NEXT STAGES

SO FAR DIGITAL voice operation has only concentrated on HF. The next stage is to develop a modem suited to VHF/UHF. While the same parallel tone waveform could be used, it is wasteful and non-optimum for these frequencies. Multipath at V/UHF has a different characteristic pattern, generally of much shorter duration, and signal fades are also much faster, particularly when operating mobile. In this situation is it probably better to return to a single tone waveform operating at a higher baud rate and correct for multipath and fading by equalising the signal and interleaving (spreading the bits over a longer time duration to straddle the fade and then allowing the FEC to correct the errors).

The parallel tone modem has shown its robustness in the presence of QRM and multipath interference. 3600 b/s is faster than any amateur modem in use today, although it has to be noted that this is the uncorrected error rate (the FEC data was generated by the vocoder). For data modes, a different type of error correction could be added, such as interleaving plus FEC. It is quite possible to foresee an error corrected rate of 2400 b/s being achievable with fallback modes in the case of a poor link. This is still higher than any amateur mode presently available (except for Clover 2K and possibly some other privately developed ones) and could be worth further development for packet or data transfer use at HF.

CONCLUSIONS

THE DIGITAL VOICE experiment set out to try a new technique on the amateur bands adapted from the commercial and professional world, and provide a practical demonstration to prove theory, calculations and simulations. The system adopted has allowed a multipurpose vocoder development tool to be developed which has the hardware in place for a much more sophisticated voice coding system in its own right, as well as a robust HF modem capable of further development as a high speed data transfer medium.

All of the simulation and development work was done by G4GUO, who is now hard at work developing other modem waveforms and data protocols. I will remain on hand to be the other end of the RF testing link and throw in useful/useless comments from time to time.

REFERENCES

[2] IT IS NO coincidence that this low baud rate parallel tone approach has been adopted for Digital TV transmission, where 2048 parallel tones are employed in an 8MHz bandwidth multipath on the UHF TV frequencies is typically a few microseconds in duration and the individual baud rate for each tone is consistent with this. The technique is further refined to minimise bandwidth by using the minimum carrier spacing and ensuring that sidelobes from one modulated carrier do not interfere with adjacent ones. The system is referred to as Coded Orthogonal Frequency Division Multiplexing (COFDM). A similar coding method with 1536 tones of 1kHz spacing is used for the terrestrial Digital Audio Broadcasting network. Parallel tone modems are one of the candidate technologies for HF Digital broadcasting and there is a lot of professional interest in parallel tone technology.

[3] G4GUO's web page is: http://www.users.dircon.co.uk/~chbrain/

This site also contains some .WAV files of typical Digital Voice signals, and links to other related sites. ♦

RadCom Helplines FOR MEMBERS

● John, G3YJD, is looking for modifications for the **Heathkit SW717** receiver, especially to improve SSB reception. He would also like to hear from owners of the **JRC JST-245** transceiver. G3YJD, QTHR. Tel: 01908 379250.

● SV1AGK is looking for a copy of the service manual for the **Icom IC-751** transceiver. SV1AGK, 5, Psaron Str, 190 16 Artemis, Greece. Tel: 0294 85 285.

● The **Bracknel Repeater Group** is looking for a voluntary, long-term 'project manager', to look after the replacement of GB3BN with more modern equipment at a new location. G3NCN, QTHR. E-mail: john.ellerton@tesco.net

● Paul Goodhall is writing a history of the **Oxford and District Amateur Radio Society**, G5LO, and would like to hear from anyone who has ever had contact with it, anyone who has an old photo of members, a press cutting of it, etc. Paul Goodhall, 2 Manor Place, Holywell, Oxford OX1 3UN. Tel: 01865 248629 (3-5pm only).

● Kevin, G4CJT, is looking for circuit diagrams/manuals for the **Belcom Liner 2** transceiver, **Marconi TF1066B** signal generator, **Marconi TF1065** output test set, and Marconi TF791D deviation meter. All costs covered. G4CJT, QTHR. Tel: 01483 268604.

● Roger, G7NEG, is looking for a copy of the circuit diagram of the 2m linear/preamp made by **Polar Electronic Developments** (of Liverpool). All costs reimbursed. G7NEG, QTHR. Tel: 01604 767533.

● G3EFK recently purchased a **KW1000** linear amplifier. The equipment is OK, but he has lost the name and address of the vendor. G3EFK, QTHR (nr Dorchester).

● Eric, G1WCQ, is looking for information on the **Triumph Adler Imperial 510 Typewriter**. G1WCQ, QTHR. Tel: 01772 686708.