A Practical Approach to Implementing H.F Digital Voice in the Amateur Service

Charles Brain G4GUO
7 Elverlands Close
Ferring
West Sussex
BN12 5PL
ENGLAND
email chbrain@dircon.co.uk

Abstract

This paper describes a practical approach to building a working digital voice system suitable for NVIS operation on the H.F Amateur bands.

Introduction

This whole project began due to a comment a friend Andy G4JNT made over the telephone he said that it would be fun to transmit real time digital speech on the amateur bands. Now there was a challenge,! As he was located some 70 km away over a fairly obstructed path it would have to be on H.F, even more of a challenge!

Choosing the Vocoder

A number of candidate systems were studied. The Vocoder had to operate at a low data rate, be inexpensive, be standalone and relatively available. The systems looked at were, LPC-10e, MELP, AMBE and various CELP systems.

I experimented with LPC-10e and even managed to implement a version of it on a Motorola 56002EVM, the speech was understandable but I never did manage to get it to track the pitch correctly. Having listened to a commercial implementation of LPC-10e I decided that it did not have acceptable' speech quality anyway.

I then went on to find an implementation of MELP (the new DOD standard) on the Internet, I managed to get the code to compile and added some Win95 sound handling routines. The speech quality was much better, but it consumed about 90% of the CPU resources on my P133 machine. Also after contacting the patent holders I was told that they were not at all happy with what I was doing.

I then looked at CELP based systems, these require large codebooks and clever search algorithms, something I thought was beyond the ROM capability of the Motorola EVM and my programming skills !

So finally I settled on the AMBE vocoder chip manufactured by DVS INC, this chip is relatively cheap, has very good sound quality, is scaleable between 2400 bps and 9600 bps and the: manufacturer would sell me some!

Choosing the Modem

After a literature search I came to the conclusion that the modem would have to use parallel tone technology. This was because it was easy to implement, was well proven, would run on my EVM and was more suitable for digital voice transmission than serial tone modems. Serial tone modems tend to produce long bursts of errors when the equaliser fails rather than the more random errors produced by a parallel tone modem. Speech is unlike computer data, in that the occasional error does not significantly affect the intelligibility.

Designing the modem

Amateur radio equipment has very poor filtering compared to military equipment. The filters tend to be quite narrow and have poor group delay characteristics. This means the modem has to use a narrower bandwidth than the equivalent military one would. This ruled out the MIL-STD 188- 110A 39 tone modem.

In the end I decided on a 36 tone modem, with a baudrate to match the 20ms frame length of the AMBE vocoder chip. This provided a raw data rate of 3600 bps and enough time for a 4ms guard period. The guard period was required to give the modem multipath tolerance. The data was modulated using DQPSK which meant that each tone carried 2 bits of data during each baud interval. Unlike military modems my modem has no Doppler correction tone and no slow sync on data facility. So far both of these facilities have proven unnecessary. The modem remains in lock for long periods of time (well beyond my ability to carry on a monologue) .

I then did some MATLAB computer simulations that showed that the modems had to be within 5Hz of the correct frequency to work properly.

To achieve initial timing and frequency offset correction the modem used three BPSK modulated preamble tones. It differentially decodes them using a delay of one baud interval it then integrates the received symbol over that time, from this it deduces the timing epoch. Then by looking at the energy in the FFT bins either side of the preamble tones it calculates the frequency error and make a correction by translating the received signal in frequency using a complex mixer. The reason for three tones is to provide some frequency diversity as on air testing showed a single tone can get lost during deep fades.

Each symbol consists of 160 samples with a sample rate of 8ks/s. The 36 tones were created by using a 128 point complex FFT, the guard period is added by taking the last 32 samples from the output of the FFT and adding them to the beginning of the FFT samples to form a total of 160 samples. These 32 samples form the 4ms guard period. The data is differentially coded and mapped to the output phases using Gray coding before transmission.

After the preamble has been sent the modem sends a reference vector, i.e. transmits a known phase on each of the 36 tones, it follows this by a sync sequence. When the receiving modem detects the sync sequence it ceases hunting for the preamble and starts passing (hopefully) valid data to the vocoder board.

When the operator releases the PTT the modem detects the loss of voice data and transmits an EOM (End Of Message) sequence embedded in the data stream, this message is in fact the SOM (Start of Message) sequence inverted. Transmit / receive control of the modem is triggered by the presence / absence of data from the vocoder, there is at present no protocol between them.

One problem with parallel tone modems is that they tend to have very high peak to mean ratios. To combat this my modem used different initial phases on each of the tones and. also applied clipping and filtering of the output signal. This allowed the transmitter to be driven quite heavily before errors began to start to appear in the received signal. The simplest way to set the audio level up was to increase the drive level until ALC action occurs then back it off.

The modem is capable of full duplex operation, it does not require a feedback channel so can be used in broadcast operation i.e. one sender and many listeners.

The modem also incorporates a CW-ID feature to comply with U.K regulations, so the old meets the new. The CW callsign has to be hand coded into the DSP S/W but can be switched off. It is not sent at the end of each over but after a programmable period.

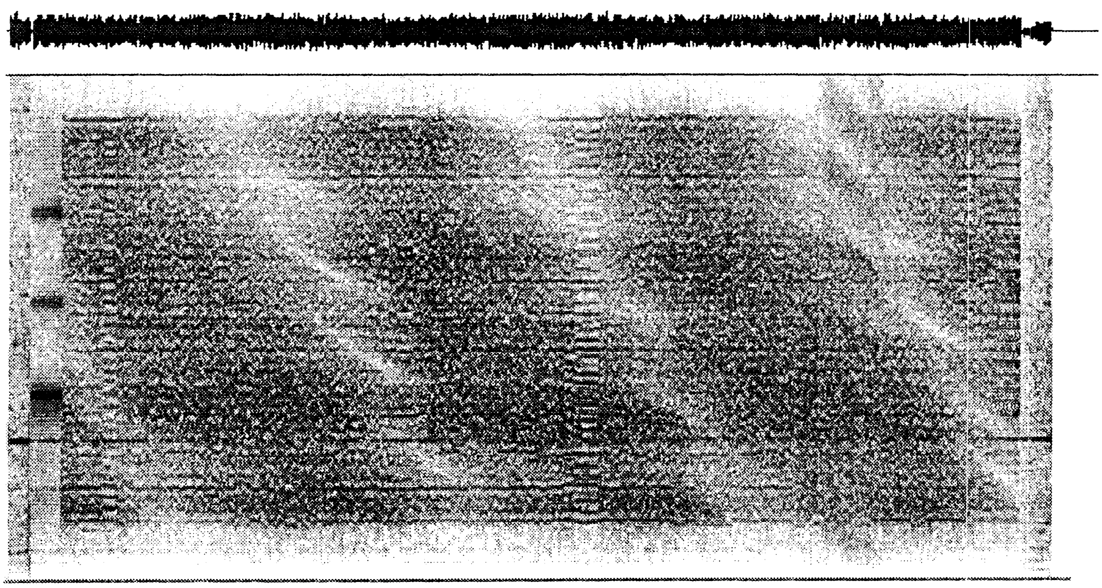

Fig 1 Off air spectrogram of M36 modem

Fig 1 shows a compressed spectrogram of an off air transmission. The three preamble tones can be clearly seen along with the selective fading (diagonal stripes), as can the carrier of an AM broadcast station in the background and a burst of interference at the end. The distinct vertical stripes were in fact pauses in the speech.

The use of FEC

The modem has no inherent FEC embedded in it, instead it uses the FEC in the AMBE vocoder chip itself. The vocoder tailors the FEC to match the significance of the bits in the data stream, so it can

On air testing

The system has been tested over a 70km path using frequencies in the 40m 'band. Andy and I made our first successful contact at the first attempt on the 27th of March 1999. This is not a weak signal mode and requires about 25 db S/N to function. However when working it makes H.F sound like a telephone conversation. There is no background noise, total silence, except for the comfort noise inserted during gaps in the speech by the vocoder itself. The system can tolerate strong CW interference and also the multipath induced selective fading found on H.F. SSB interference is more troublesome as it affects more than one of the tones. If RTTY/CW interference gets too bad it is even possible to switch a DSP notch filter in circuit, there is enough power in the FEC to cope with the missing tones, however the notch filter must be switched out during the preamble phase.

The weakest part of the modem is the preamble phase, to help solve this I added the ability to save the frequency offset correction and timing epoch after each successful preamble synchronisation. If for some reason the receiving modem misses the start of the transmission it is then possible to press a button on the front panel and revert to the last set of sync information. In a one to one QSO this works most times.

Another change that was made to the modem was to allow the different tones to be given different amplitudes to compensate for the amplitude response of the transceiver. The group delay in the transceiver will however reduce the modems tolerance to multipath.

With the new generation of IF DSP radios this will not a problem as their filter characteristics are much more suited to this kind of operation.

As well as H.F testing I have also used in on 2m both on SSB and FM, and there is no reason it would not work via a repeater as there is no ARQ (but I have not tried it).

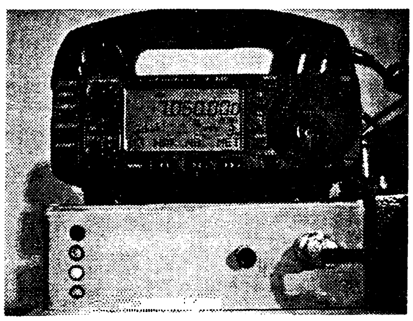

Fig 3 Current Digital Voice Station

Conclusion

It is now possible for the home constructor to build for about $300, a portable, working digital voice system for H.F, with near toll quality audio. This system can equally be used to experiment with digital speech using different DSP modems on different frequencies.

For further information and a full technical description plus some sound files surf along to my website http://www.chbrain.dircon.co.uk/dvhf.html