

# HF Digital Voice Transmission using an OFDM Modem with Space-Time Coding

Matt Ettus, N2MJI

matt@ettus.com

<http://ettus.com>

August 7th, 2001

## Abstract

The High-frequency (HF) radio channel presents a unique challenge to the modem designer. It is characterized by large delay-spreads, rapid fading, and impulse noise. Those modems which have been successful in this environment have relied heavily on time-diversity (through coding and interleaving), or very low bit rates. However, long interleaver delays are not tolerable in a two-way voice contact, and so some other means of improving reliability is necessary if digital voice traffic is to be accommodated.

This paper discusses the design of a digital voice HF modem which uses orthogonal frequency division multiplexing (OFDM), some times referred to as a “parallel-tone” modem. While OFDM provides some frequency diversity, there is no inherent time diversity. In order to make up for this, spatial diversity is used, both on the transmit and on the receive sides.

## 1 Introduction

This project was both inspired and greatly influenced by the work of Charles Brain, G4GUO[1]. The goal of

this project is to design a modem for the amateur HF bands which is capable of transmitting digital voice of a quality equal to, or greater than that of standard SSB. The aim is to minimize the amount of special hardware necessary, and so standard PCs and sound cards are used for signal processing and acquisition. A basic system will only need a standard HF radio with good frequency accuracy. A system capable of diversity transmission or reception would require a second antenna and a second radio system.

The modem uses OFDM modulation with DQPSK(differential quadrature phase shift keying). The occupied bandwidth is the full 2.7 kHz of the amateur HF voice channel. It sends a net of 2400 bits per second (4600 with coding), and will be able to take advantage of transmit diversity, receive diversity, or both, if available. A 2400bps vocoder is used to compress the voice down to the data rate of the modem.

## 2 Vocoder

The first obstacle to the transmission of digital voice over HF is the very low bit rates which can be sent. It was decided that 2400 bits per second was the highest rate that could be expected to work in moderate conditions. Unfortunately, this severely constrains the choice of vocoder, as most low-rate vocoders are proprietary.

The first, and most obvious choice for a vocoder is the LPC-10e standard, used by the U.S. military, among others. It is not patent encumbered, and public domain implementations are available. It was developed in the early 1980's, and unfortunately, its quality is not great. It is quite usable, however.

In the last few years, several 2400 bps vocoders were created, mainly to compete to be the official replacement for LPC-10e (with the goal of better sound quality than 4800 bps CELP). Among those are AMBE, from DVS Inc, used in [1], and the winning entry, MELP2400. MELP, as the winner, is heavily documented[5], and source code is available. It is not "free" in that it is owned by a corporation which licenses it<sup>1</sup>. It is not certain in what way this would impact its use in amateur radio, but for experimentation purposes that is not a huge concern. There is always the option of falling back to LPC-10e if there are significant entanglements.

## 3 Modem Design

### 3.1 Modulation

In OFDM, many carriers are modulated at a low rate. Conventionally, this would be done with many os-

<sup>1</sup>ASPI, Inc. claims unspecified "Intellectual Property" over MELP. Use at your own risk.

cillators, but in modern systems the inverse fourier transform is used. The magnitudes and phases of each carrier are written into the appropriate bin, and the IFFT creates the corresponding time-domain signal which is transmitted. On the receiving side, a fourier transform is performed, and the magnitudes and phases of the transmitted carriers are recovered. Precise alignment of frequency and timing for the fourier transform window is required, or orthogonality is lost, and self-interference occurs.

One advantage of OFDM is that since the symbols are modulated at a low rate, the sidebands do not extend far beyond the edges of the signal. Because of this, pulse shaping (i.e. root raised cosine filtering) is usually not used. Also, since the individual tones are spaced at the minimum orthogonal width, greater spectrum efficiency can be obtained. In the limit of many tones, BPSK gives one bit per second per hertz, QPSK achieves two, 8PSK three, etc.

In channels with multipath, the last part of the previous symbol will interfere with the beginning of the next one. In order to combat this, guard times were originally used between symbols. It was later shown that by cyclically extending the symbol, and ignoring the beginning (where energy from both symbols is present), orthogonality could be maintained.<sup>2</sup> This is known as the cyclic prefix (CP). It is formed by copying the last few samples to the beginning of the transmitted symbol.

In this modem, we will be sending 2400 bits per second, encoded at rate 12/23. Thus, we have to send 4600 bits per second. We will be using DQPSK, which gives us 2 bits per carrier per symbol. The carriers are spaced at the inverse of the symbol rate,

<sup>2</sup>Note that this does not remove the effects of multipath. Each frequency bin will undergo independent flat fading. They just won't interfere with each other.

which we choose to be equal to the frame rate of the vocoder, 22.5 ms. Each frame contains 54 bits. We use a cyclic prefix of 2.5 ms, leaving 20ms for the useful symbol time. The tones need to be spaced to be orthogonal over 20ms (i.e. 50 Hz), not 22.5ms (44.44 Hz), which costs us some spectrum efficiency.

One of the main disadvantages of OFDM modulation is the very high peak to average power ratio (PAR). Since most power amplifiers are limited by peak power (and FCC regulations limit peak power), a bigger amplifier is necessary for the same average power as single-carrier systems. This power backoff also causes low power efficiency, since the amplifiers must be linear. Some methods for reducing the PAR of OFDM have been proposed, but the one currently implemented uses random initial phases for the first symbol.

### 3.2 Coding

In order to provide protection against noise, forward error correction (FEC) is used. The Golay (23,12) code was selected because a free implementation is available, and it is quite simple. The fact that the rate is slightly less than 1/2 (for each 12 bits in, 23 coded bits are generated) is also quite helpful to us, since it allows us to fit within standard radio passbands without having to move to a higher-order modulation like 8PSK. More advanced coding, utilizing convolutional codes and soft-decisions is under consideration.

### 3.3 Synchronization

Proper synchronization, both in symbol timing and frequency, are very important for successful OFDM reception. Small timing errors cause phase rotations

proportional to frequency, and a loss of orthogonality, resulting in intercarrier interference (ICI). Larger timing errors result in intersymbol interference (ISI). Both cause significant bit error rate (BER) degradation. Frequency errors of just a few percent of the intercarrier spacing cause significant ICI because of the loss of orthogonality. In our modem, the intercarrier spacing is 50 Hz, so relying purely on the accuracy of the frequency reference is not practical.

We use a maximum likelihood joint estimator for both time and frequency offset which was proposed in [7]. This estimator takes advantage of the cyclic prefix (which would otherwise be wasted energy). By computing the correlation between samples separated by the frame size, the CP is detected. The relative phase of the correlation at that point gives the frequency offset.

This estimator is capable of coarse acquisition of timing, as well as fine tracking. For frequency, it is capable of fine tracking, to within better than 1% of the intercarrier spacing. Some other means of tracking frequency to within +/- one half of the intercarrier spacing is necessary. For now, we rely upon the frequency accuracy of the transmitter and receiver to get close enough for the ML estimator to work. This requires better than 1 ppm accuracy on 20 meters.

### 3.4 Spatial Diversity

Because the HF channel is time-varying, most HF modems use extensive interleaving. This ensures that long sequences of bits which are lost to deep fades will be spread out, allowing the error correction coding (FEC) to work. This does not work well with voice communications because of the long delays, sometimes as much as 10 seconds. Instead of time diversity, we must find another way of mitigating the

effects of fading – spatial diversity.

Spatial diversity takes advantage of what every operator intuitively knows – that when one spot undergoes a fade, moving the antenna can help. Instead of moving the antenna, however, we have multiple antennas, the signals from which we can combine coherently. Little has been published on the subject of coherence distance<sup>3</sup> on HF channels, but a spacing of 2 wavelengths should be sufficient. For those without that much space, polarization diversity has been shown to provide nearly as much gain.

For receive diversity, maximal-ratio receiver combining (MRRC) will be used. In MRRC, the received signals are added together after multiplication by the complex conjugate of the path gain. Thus, each signal is weighted by its received strength. This can be used with an arbitrary number of antennas. Of course, an accurate estimate of the [complex] path gain is necessary. The first two symbols at the beginning of each transmission can be used to get an initial estimate, which will need to be continually updated after subsequent symbols are received.

Transmit diversity is more complex. In order for the receiver to be able to separate the signals from each of the transmitters, they must be encoded differently. The method used in this project was first proposed in [2], and was applied to OFDM in [3]. It is limited to two transmit antennas. Basically, at time (or frequency in the OFDM case)  $n$ , transmitter  $T_0$  sends signal  $s_0$ , and  $T_1$  sends  $s_1$ . At time (or frequency)  $n+1$ ,  $T_0$  sends  $-s_1^*$ , and  $T_1$  sends  $s_0^*$ . Two slot are used to send two symbols, thus the same data rate is maintained.

Assuming  $h_x$  is the [estimated] path gain from antenna  $x$  to the receiver, and  $r_x$  is the received sig-

<sup>3</sup>Coherence distance is the minimum spacing between antennas which provides essentially uncorrelated fading.

nal at time (or frequency)  $x$ , then the receiver need only perform the following computations to decode the signal:

$$s_0 = h_0^* r_0 + h_1 r_1^*$$

$$s_1 = h_1^* r_0 - h_0 r_1^*$$

The math gets a little more complicated with multiple receive antennas, but the concept is the same. Either the transmitter, the receiver, or both could utilize diversity in this system. A useful result of this is that if only one participant has a dual-antenna setup, both sides can still take advantage of the 2-way diversity. Of course, if both have dual-antenna setups, then 4-way diversity is possible, with a corresponding increase in diversity (there are now 4 paths between transmitters and receivers).

## 4 Future Work

Currently, most of the basic work is completed. OFDM transmission and reception work, as do the time and fine-frequency synchronization. An algorithm for estimating coarse frequency error will be necessary if high-accuracy frequency references won't be used in the transceivers.

In order to implement the receive processing which is required for the transmit and receive diversity schemes, a channel estimator will be necessary. Several options for this are being investigated, but it is unclear how the very short training sequences will affect performance.

Hardware to implement the diversity scheme will also be required. For the prototypes, T2 transmitters and R2 receivers [6] will be used, due to their simplicity, low cost, high dynamic range, and flat passband.

The various methods of systematically reducing PAR and linearity requirements will be investigated.

A coding scheme which has more coding gain than Golay codes, and which can use soft decision information will likely be added. It may be necessary to switch from DPSK to 8PSK with trellis-coded modulation if typical radio passbands are not clean enough. In this case, an outer Reed-Solomon code could easily be added, possibly making up for the loss of power efficiency which comes with 8PSK.

[7] J.-J. van de Beek, M. Sandell, and P. O. Börjesson, "ML estimation of time and frequency offset in OFDM systems," *IEEE Transactions on Signal Processing*, vol. 45, no. 7, pp. 1800-1805, July 1997.

## References

- [1] Charles Brain, "A practical approach to implementing H.F. digital voice in the amateur service," *18th ARRL and TAPR Digital Communications Conference*, pp. 17-21, September 1999. See also <http://www.chbrain.dircon.co.uk/dvhf.html>
- [2] Sivash M. Alamouti, "A simple transmitter diversity scheme for wireless communications," *IEEE Journal on Selected Areas in Communications*, vol. 17, no. 3, pp. 451-460, March 1999.
- [3] K. F. Lee and D. B. Williams, "A space-frequency block-coded OFDM transmitter diversity technique," *IEEE GLOBECOM 2000*, San Francisco, CA, November 2000.
- [4] Martin C. Gill, "Coded-waveform design for high speed data transfer over high frequency radio channels," Ph. D. Dissertation, University of South Australia, February 1998.
- [5] See <http://www.plh.af.mil/ddvpc/melp.htm> and <http://www.aspi.com/products/speech/melp.html>
- [6] Rick Campbell, "High-performance, single-signal direct-conversion receivers," *QST*, January 1993.