Frequency Division Multiplex Digital Voice (FDMDV)

1. Introduction

FDMDV is a digital voice mode intended for transmission and reception over high-frequency (HF) radio. It uses a frequency division multiplex (FDM) modem with 15 carriers and no forward error correction (FEC). An open source, low-bit-rate coder-decoder (CODEC) provides voice quality audio without the listener fatigue caused by noise and interference normally associated with analog single sideband (SSB) voice. Setup and operation of the Windows®-compatible program was developed to make operation straightforward. An HF transceiver, personal computer and two sound cards are required. Path simulation and on-the-air HF testing have shown that decoding voice is possible at a signal-to-noise ratio of 3 dB.

FDMDV is based on ideas by Peter Martinez, G3PLX, and written in C for Windows XP by Francesca Lanza, HB9TLK.

2. Waveform and Emission Designator

The waveform consists of 14 quadrature phase-shift keying (QPSK) carriers with 75-Hz spacing between centers and a necessary bandwidth of 1.125 kHz. A binary phase-shift keying (BPSK) carrier is centered between the 14 QPSK carriers, making the total of 15 carriers. The BPSK carrier has twice the power of the 14 QPSK carriers and provides an auto-tune function and frame indication. The QPSK carriers operate at 50 bauds each. Combined, they provide the voice CODEC data. Transmit auto-level-control (ALC) compresses the waveform to increase the average power while reducing the peaks (crest factor reduction).

The International Telecommunication Union (ITU) emission designator is 1K20J2E.

3. Data Rate

50 Bd per carrier X 28+1 = 1450 bit/s
1400 bit/s CODEC rate + 50 bit/s text rate

4. Frequency Tolerance and Correction

FDMDV provides accurate tuning within ±5 Hz to synchronize transmit and receive signals. Both manual and automatic tuning functions meet this requirement using the mouse pointer and left/right mouse clicks on the waveform. Automatic frequency control (AFC) is capable of correcting frequency errors up to 50 Hz/minute.

5. Frame Structure

Open source LPC frame: 25 Hz, 40 ms, 54 bits.
Text frame: 25 Hz, 40 ms, 2 bits.

Data frame: 50 Hz, 20 ms, 14 carriers X 2 bits per symbol = 28 bits.

Voice and Text to data frame mapping:

A BPSK carrier with no phase change indicates the first data frame. It carries the first 28 bits of LPC frame data. The BPSK carrier phase inversion (PI) phase change indicates the second data frame. The second data frame then contains the last 26 bits of LPC data and 2 bits of text data intended for call sign identification. The text data is encoded with a 2-bit Varicode and may be extended to 80 characters.

Text -Varicode // Each character is separated by two zeros.

// The bits are sent most significant bit first, always 2 bits as a unit.

// 03
0x0003, // ASCII = ' ' 11
0x0001, // ASCII = '/' 01
0x0002, // ASCII = '#' 10 (Start of message)

// 09
0x000F, // ASCII = '1' 11 11
0x0007, // ASCII = '2' 01 11
0x000B, // ASCII = '3' 10 11
0x000D, // ASCII = '4' 11 01
0x0005, // ASCII = '5' 01 01
0x0009, // ASCII = '6' 10 01
0x000E, // ASCII = '7' 11 10
0x0006, // ASCII = '8' 01 10
0x000A, // ASCII = '9' 10 10

// 27
0x003F, // ASCII = 'a' 11 11 11
0x001F, // ASCII = 'b' 01 11 11
0x002F, // ASCII = 'c' 10 11 11
0x0037, // ASCII = 'd' 11 01 11
0x0017, // ASCII = 'e' 01 01 11
0x0027, // ASCII = 'f' 10 01 11
0x003B, // ASCII = 'g' 11 10 11
0x001B, // ASCII = 'h' 01 10 11
0x002B, // ASCII = 'i' 10 10 11
0x003D, // ASCII = 'j' 11 11 01
0x001D, // ASCII = 'k' 01 11 01
0x002D, // ASCII = 'l' 10 11 01
0x0035, // ASCII = 'm' 11 01 01
0x0015, // ASCII = 'n' 01 01 01
0x0025, // ASCII = 'o' 10 01 01
0x0039, // ASCII = 'p' 11 10 01

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0x0019, // ASCII = 'q' 01 10 01
0x0029, // ASCII = 'r' 10 10 01
0x003E, // ASCII = 's' 11 11 10
0x001E, // ASCII = 't' 01 11 10
0x002E, // ASCII = 'u' 10 11 10
0x0036, // ASCII = 'v' 11 01 10
0x0016, // ASCII = 'w' 01 01 10
0x0026, // ASCII = 'x' 10 01 10
0x003A, // ASCII = 'y' 11 10 10
0x001A, // ASCII = 'z' 01 10 10
0x002A, // ASCII = '0' 10 10 10
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6. Modem

The modem is based on a raised-cosine tone filter response having the property of zero inter-symbol interference (ISI) and zero adjacent tone interference with no side lobes. Half the channel filter is in the transmit (TX) side and the other half in the receive (RX) side. It may be called ñroot raised cosineò because the channel filters at each end have the net response of a square-root of the complete filter. (G3PLX modem description)

7. Soundcard

48-kHz sampling rate for both sound cards and sound adaptors available from many different manufacturers. The second sound card may be a USB headset such as used with VoIP applications to provide voice input and output.

8. FDMDV References

KØPFX Quick Start Guide and Operation Document, http://n1su.com/fdmdv/

9. Acknowledgement

This description was prepared by Mel Whitten, KØPFX.
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