Doc. #26A

Digital Voice Update

Doug Smith, KF6DX, ARRL Digital Voice Working Group

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Experimentation with digital voice in Amateur Radio continues to develop along at least two fronts: Thales' SkyWave, a proprietary system that uses a sound card and a PC; and stand-alone systems that use the AMBE-2020 voice codec.

The Thales software is still in its alpha testing phases but progress has been made on distribution arrangements. Tentative agreement has been reached to make copies available to Digital Voice Working Group members for beta testing. The distribution would possibly include source code for the user interface. The modem and voice codec are proprietary and source code would not be available. Thales and *QEX* published an article with some details of the system early this year.[1]

Plans are being discussed to release the software in September 2003. Licensing and intellectual-property issues remain.

Our friends at AOR Japan have generated a document that details the protocols and formats used in their ARD9800 digital voice system. That document has been passed to Dennis Silage, K3DS, at Temple University, where a team of students is pursuing a compatible system. Each team is trying to maintain mutual compatibility and also backward compatibility with the original AMBE-based system of member Charles Brain, G4GUO. The specification is attached as an Appendix.

The AMBE-2020 and Thales systems cannot intercommunicate and no nonproprietary digital voice-coding scheme has been uncovered by our Group. The future of digital voice may lie mainly in multimedia modes-- that is, in simultaneous voice, data and video. The Group will be interested to see the reports of the *ad hoc* ARRL committee assigned to study high-speed HF data modes and that of the High-Speed Multimedia Working Group.

Regulatory issues regarding simultaneous digital voice, data and video on HF have not been adequately addressed. Regulations in countries such as Japan, the UK and the USA differ by a wide margin and in many cases, they are ambiguous or speak nothing to the issue. Digital voice *per se* is unaffected by that; but without simultaneous data, video or file-transfer capabilities, further development will be held back in my view.

Respectfully,

Doug Smith, KF6DX

Reference:

1. Cédric Demeure and Pierre-André Laurent, "International Digital Audio Broadcasting Standards: Voice Coding and Amateur Radio Applications," *QEX*, Jan/Feb 2003.

Appendix: ARD9800 Protocol

De AOR Japan:

1. Digital Voice Format

1-1 General Description

The voice waveform is sampled at 8 kbps and processed through an AMBE vocoder. It compresses the voice data stream into 72 bits for every 20-msec symbol. This is equivalent to a 3600 bps rate. At that rate, 2400 bps is plain voice information and 1200 bps is parity information for error correction as FEC. This format is determined by the AMBE vocoder chip.

The 72 bits feed the OFDM data frame. In this OFDM scheme, voice information is carried by 36 orthogonal tones. Each carrier is modulated by DQPSK. The OFDM packet structure is very simple. Before the voice packet, it has a three-tone header. This preamble makes it possible to capture the correct symbol timing and to compensate for any frequency offset caused by SSB tuning.

After a one-second header burst, we have a 72-bit reference vector symbol. Each carrier is modulated by DPSK; so the receiver must have the initial phase information as a reference. Followed by the reference comes a 4-symbol voice packet start marker. Receivers detect the initial timing of each voice packet by matching unique 4x72 bits pseudo-noise (PN) codes. When the unit detects PTT has been pushed, the MODEM generates 72-bit voice packets from vocoder.

After the MODEM detects that PTT has been released, it must send an 8-symbol closing code. The receiver can terminate voice packet decoding by detecting this end marker. This is also a unique PN code but it is just an inverted copy of the start marker code.

In each 20-msec frame, a 4-msec guard interval is inserted that protects against degradation from multi-path fading. The guard interval is just added to the start of each 16-msec voice frame.

Fig 1 Voice Packet

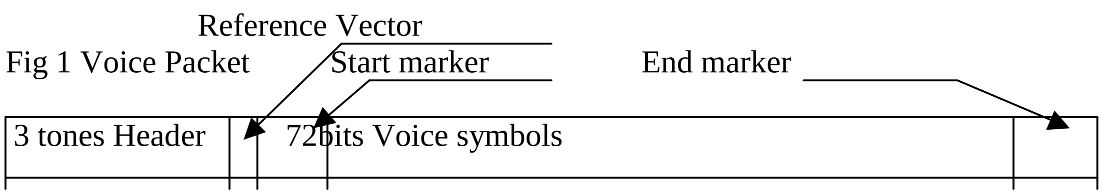
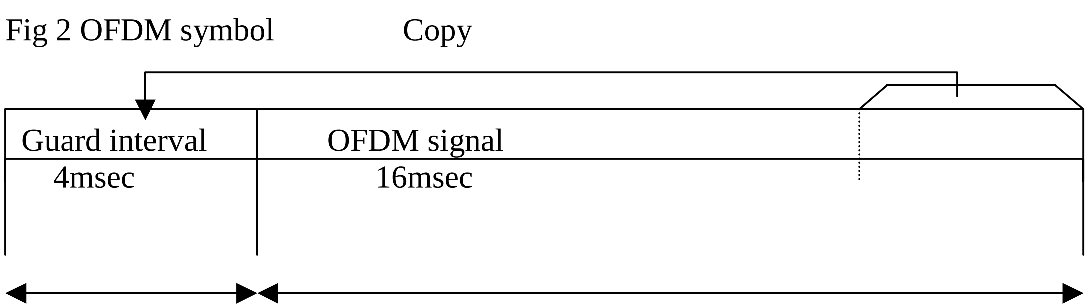


Fig 2 OFDM symbol



1-2 AMBE(Advanced Multi-Band Excitation)

AMBE was developed by DVSI. If you want to implement this protocol into your system, you need to use the AMBE chip supplied by DVSI or you have to make a license agreement with DVSI. DVSI have not released detailed information about their AMBE frame.

There are many possible types of output format from the AMBE chip. In this protocol, a 3600-bps data frame has to be selected. This includes 2400 bps plain voice data and 1200 bps parity data. Error control is a part of AMBE. For the AMBE2020 chip, the parameters you have to set are as follow:

Fig 3 AMBE2020 parameters

1	.$13EC	Frame head marker
2	0000000000000000_2 ********------------- ------------*------------ -----------------------*-	(Depends on your system) Power control: normal operation Lost Frame is used former data CNI Disable
3	$1030	Rate information 1 2400/1200
4	$4000	Block Coding
5	$0000	Rate information 2
6	$0000	Rate information 3 Block Code
7	$0048	Rate Information 4

AMBE compresses 8-kbps-sampled digital voice into a 72 bits/20 msec data stream. The parity data are included in 72-bit frame. The OFDM modulator has to synchronize its symbol timing with the 20-msec voice frame of AMBE chip.

1-3 Three Tones Header

HF communication has been done by SSB in ham radio for a long time now. With analog voice, mistuning causes incorrect pitch in a receiver; but a little bit is acceptable. In the OFDM scheme, error rate is very sensitive to frequency offset. Without automatic compensation, it would be impossible to make digital communications by OFDM through an HF channel.

An OFDM receiver has to be locked to the symbol timing of the transmitter. Synchronization by detection of the convolution peak of the guard interval is possible; but it is too weak to get the correct phase. On the other hand, autocorrelation of the header provides fast lockup.

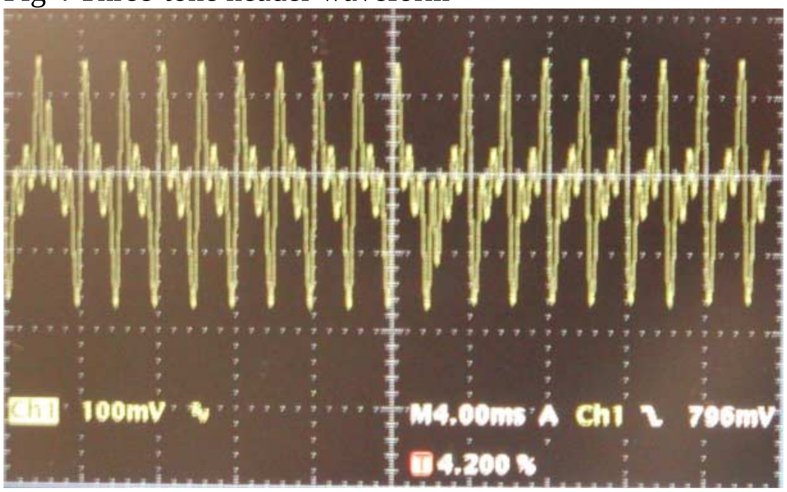
For that reason, the digital voice packet has a three-tone preamble. The original idea was the G4GUO modem. From a compensation point of view, a single tone is enough to send as preamble; but it is not enough for frequency-selective fading. Even if a receiver were unable to detect one of three tones, it can detect and compensate the signal using the other two tones. Consider frequencies as follow:

Tone 1 62.5Hz x 8 = 500 Hz
Tone 2 62.5Hz x 16 = 1000 Hz
Tone 3 62.5Hz x 24 = 1500 Hz

In this OFDM mode, carrier spacing is 62.5 Hz so that header can also be generated by the same signal processing as another OFDM voice frame. This means the three tones are also orthogonal (mutually exclusive).

The header frame has the 20-msec symbol structure shown in Fig 2. The phase of each tone is inverted every 20-msec symbol. This is equivalent to a 1,0,1,0… data stream modulated by BPSK on each tone. Modulated tones are added as in Fig 4's waveform.

Fig 4 Three-tone header waveform



1-4 OFDM carriers and IFFT

This OFDM symbol has to be synchronized with 20-msec frame of the vocoder. The digital voice sampling rate is 8 kHz. OFDM modulation is managed by a 128-point inverse fast Fourier transform (IFFT). That means that 128 samples of voice data are feed to IFFT processing. A 4-msec guard interval is inserted into each OFDM symbol. Basic parameters of the IFFT are as follow:

8 kHz = 125 usec.	digital voice sampling rate
125 usec x 128 = 16 msec	IFFT output
125 usec x 32= 4 msec	guard interval
125 usec x 160= 20 msec	total OFDM symbol
8 kHz /128 = 62.5 Hz	carrier spacing

The output signal has to be real so that 64 carriers can be used for this modem. An HF radio can send signals from around 300 Hz to 2.6 kHz. Only 36 carriers are selected for this modem.

62.5 Hz x 5 = 312.5 Hz	lowest carrier
62.5 Hz x (5+35) = 2,500 Hz	highest carrier

1-5 OFDM modulation

36 carriers are modulated by the differential QPSK. At first, each 72-bit AMBE frame is formatted into 36 symbols/2-bits pair.

Fig 5 Formatting 36 symbols

MSB 72bits AMBE frame LSB

A1	A2	A3	A4	A5	A6	A71 A72		

36symbols for 36 carriers

A1	A3	A5	A71	U axis (Real part)
A2	A4	A6	A72	V axis (Imaginary part)

Each symbol is Gray coded.

Fig 6 Gray code

A2n-1 (U)	0	0	1	1
A2 n (V)	0	1	0	1

Gray code	Uj	0	0	1	1
	Vj	0	1	1	0

Symbols are modulated by the differential QPSK.

Fig 7 DQPSK modulation

Uj	0	0	1	1
Vj	0	1	1	0
Phase Change	0	+90	-90	+180

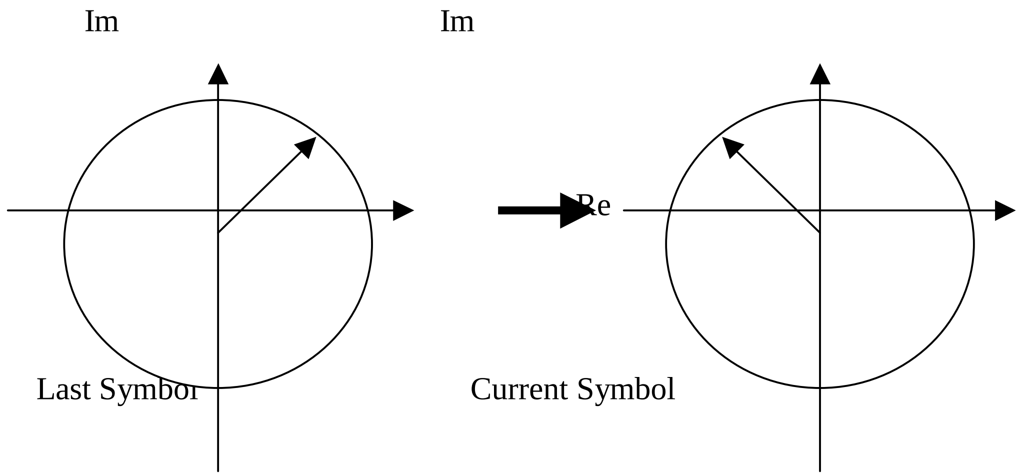
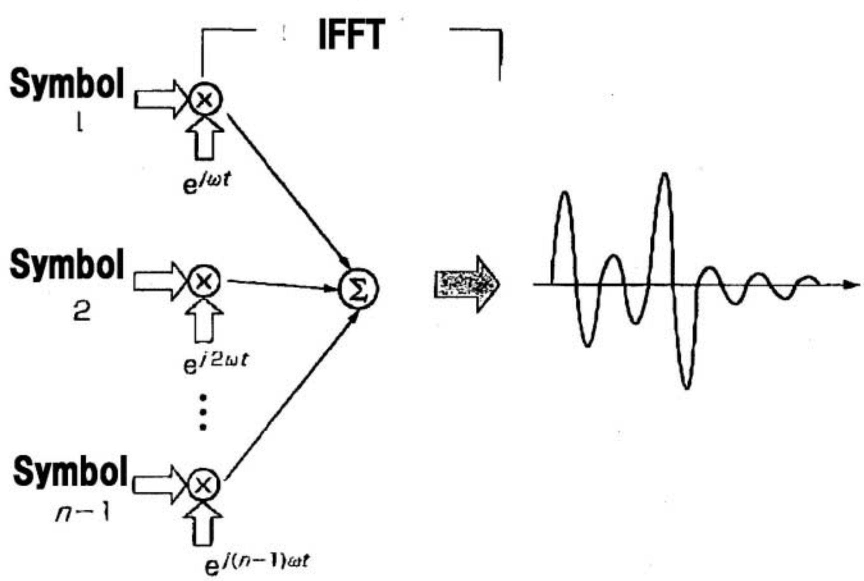


Fig 8 IFFT



1-6 Reference Vector

In this OFDM scheme, each carrier is modulated by differential QPSK so that receiver must know the last phase position to calculate the current phase change. Just after the three-tone header, the receiver cannot have the most-recent phase information of the transmitter. So each OFDM packet has the reference vector at the end of header. It would be acceptable to set any vector as the reference; but for convenience, an example is shown as follows:

MSB LSB

.$F5,$5F,$5F,$F5,$F5,$5F,$5F,$F5,$5F

1-7 Start Marker Code

The OFDM signal is pseudo-random colored noise so it is hard to identify the signal as OFDM or just noise at the receiver end. After the reference vector, four symbols are inserted into each OFDM voice packet. 4x72 bits are a unique PN code. They are modulated by the normal 36 tones of OFDM instead of the 72-bit AMBE voice frame.

Fig 8 Start Marker Code

MSB LSB

$65	$10	$F8	$12	$19	$71	$5A	$D1	$06	Symbol1
$26	$BB	$C1	$FB	$79	$A3	$A9	$4B	$BE	Symbol2
$60	$00	$65	$10	$F8	$12	$19	$71	$5A	Symbol3
$D1	$06	$26	$BB	$C1	$FB	$79	$A3	$A9	Symbol4

1-8 End Marker Code

To close each OFDM voice packet, the end marker code is added at the end of each voice frame. Eight symbols are inserted into the voice packet after the modem detects PTT is released. The code is a unique PN code but it is just an inverted copy of the start maker code.

Fig 9 Start Marker Code

MSB LSB

$9A	$EF	$07	$ED	$E6	$8E	$A5	$2E	$F9	Symbol1
$D9	$44	$3E	$04	$86	$5C	$56	$B4	$41	Symbol2
$9F	$FF	$9A	$EF	$07	$ED	$E6	$8E	$A5	Symbol3
$2E	$F9	$D9	$44	$3E	$04	$86	$5C	$56	Symbol4
$9A	$EF	$07	$ED	$E6	$8E	$A5	$2E	$F9	Symbol5
$D9	$44	$3E	$04	$86	$5C	$56	$B4	$41	Symbol6
$9F	$FF	$9A	$EF	$07	$ED	$E6	$8E	$A5	Symbol7
$2E	$F9	$D9	$44	$3E	$04	$86	$5C	$56	Symbol8

Annex A Crest Factor

This is not a part of the digital voice format but you have to pay attention to the peak-to-average (crest factor) of OFDM transmissions. OFDM has a flat spectrum so that it must generate a very high crest factor. This may cause serious problems for the final amplifier of transmitter.

Crest-factor management is recommended in your system. There are a lot of ways to fix this problem. The easiest way is just to clip peaks over a certain level. In many cases, this would work well.

This standard has not described the absolute phase of header burst tones, but it is very important to decide initial phase in BPSK modulation. These phases strongly affect the crest factor of the header burst signal.

Annex B Compatibility

This digital voice coding is based on Charles Brain, G4GUO's work. He is a pioneer of practical OFDM voice modems in HF radio. (http://www.chbrain.dircon.co.uk/).

Unfortunately, I have never had a chance to test compatibility between G4GUO's modem and mine. One thing I would like to note is that his modem board used the AMBE1000 vocoder. It also uses a 3600-bps rate, but I am not sure whether it is the same format as the AMBE2020 or not, because DVSI has not released detailed information the formats.