

Vocoder Redux: the AMBE-2020

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Abstract

A Digital Voice Systems AMBE-2020 voice coder (vocoder) has been implemented as a functional replacement for the obsolete AMBE-1000. The effort, in part, was a capstone senior design project in Electrical and Computer Engineering at Temple University, which fosters advanced technical topics in undergraduate education using amateur radio. The implementation is open-source and complies with the data communication protocol of the original G4GUO AMBE-1000 vocoder design. This AMBE-2020 vocoder is offered as a reasonable standard for amateur radio digital voice experimentation. Further documentation for the AMBE-2020 vocoder is available at www.temple.edu/k3tu.

Aborted *Beta* Testing

Charles Brain G4GUO and Andy Talbot G4JNT awaken the amateur radio technical community with their voice coder (vocoder) implementation that introduced high-quality digital voice communication on HF within a 3.3 KHz bandwidth^{1,2}. A nominal 3 KHz voice signal sampled at 8000 samples/sec with only 8 bits of resolution, the parameters of a pulse code modulation (PCM) telephone circuit at 64 Kb/sec, would require at least a +64 dB SNR for a 3 KHz transmission bandwidth³. A vocoder was chosen to lower the data communication rate to 3.6 Kb/sec, but, and more importantly, the minimum SNR required then is only +15.5 dB³. They investigated several candidate vocoders, including LPC-10e, MELP and CLEP, before settling on the Digital Voice Systems (DVS, www.dvsinc.com) Advanced Multi-Band Excitation (AMBE) vocoder.

The selected DVS AMBE-1000 vocoder was based on a Lucent DSP16 microprocessor. The vocoder provided a large ensemble of possible data rates and forward error correction (FEC) options. A voice data rate of 2.4 Kb/sec with 1.2 Kb/sec of forward error correction (FEC) was adopted for their tests. An auxiliary microprocessor, a Microchip PIC 17C44JW, was used to initialize the AMBE-1000 vocoder and provide process control. A Motorola MC14LC5480 8-bit, μ -Law serial coder/decoder (codec) digitizes the analog voice input signal and converts binary PCM data back to an analog voice output signal.

The G4GUO/TAPR Vocoder was produced and offered for *beta* testing by TAPR (Figure 1). Several experimenters obtained the device and a TAPR Vocoder Development List was organized. Unfortunately, by early in 2002 the DVS AMBE-1000 was discontinued (“*not recommended for new designs*”) and replaced by the interoperable but not compatible DVS AMBE-2000/2020. Thus the technological momentum for amateur radio digital voice experimentation was seemingly thwarted, as is so often the case, by *obsolescence*.

Scoring the Vocoder

The DVS AMBE-1000/2000/2020 vocoder utilizes a class of Multi-Band Excitation (MBE) algorithms initially developed by MIT in the 1980s. Rather than model the vocal tract itself with linear prediction, the MBE algorithm codes small temporal segments of the voice signal (AMBE uses 20 msec segments)

into several frequency bands (*Multi-Band*) from which pitch and other spectral parameters are determined and coded.

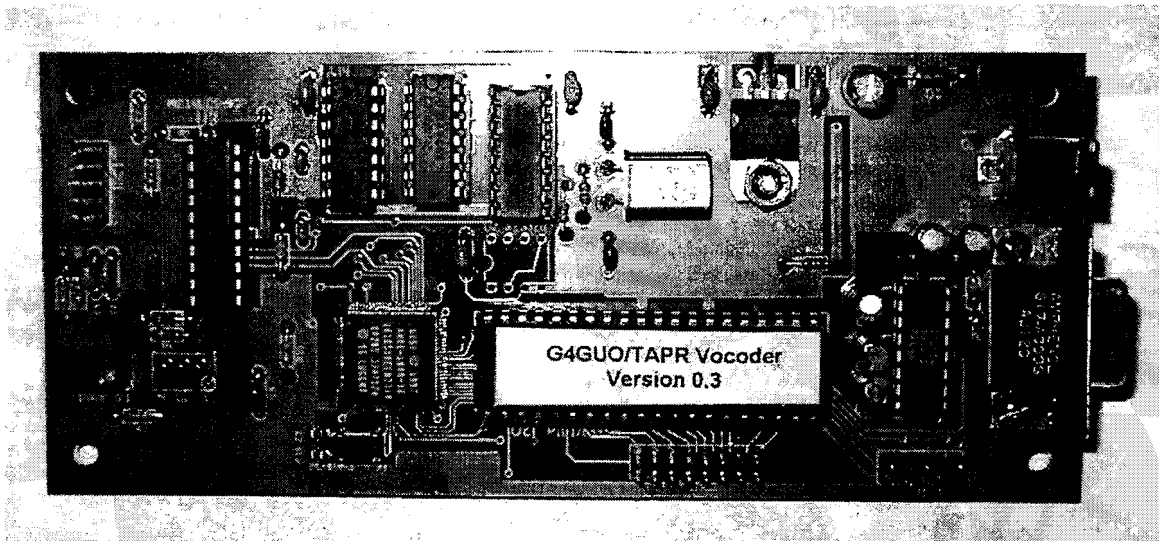


Figure 1 – G4GUO/TAPR AMBE-1000 *Beta* Vocoder

A subjective Mean Opinion Score (MOS) is used in speech processing research to assess the performance of a vocoder. Independent commercial and academic laboratories have confirmed that the 4.8 Kb/sec AMBE vocoder subjectively outperforms other vocoder standards at their significantly higher bit rates: G.729 at 8 Kb/sec; G.726 at 32 Kb/sec; and GSM at 13 Kb/sec. AMBE even subjectively outperforms the FS-1016 vocoder at the same 4.8 Kb/sec. These better MOS metrics obtained at lower bit rates have improved the performance of mobile satellite and radio systems, including the APCO Project 25.

New Design, New Challenges

The DVS AMBE-2000/2020 is based on the TI TMS320-LC541-66 DSP microprocessor. DVS provides the half-duplex AMBE-2020 at half the cost of the full duplex AMBE-2000, although there are no hardware changes involved. The half-duplex device was selected for this design, even though this precludes full digital vocoder loop-back testing.

The AMBE-2020 operates only on +3.3 VDC at a clock rate of 16.384 MHz. The analog front-end selected was the Analog Devices (www.analog.com) AD73311L low power CMOS 16-bit linear device. DVS recommends a linear 16-bit device for maximum voice quality. The new AMBE-2020 vocoder seemingly exhibits a marked improvement over the voice quality performance of the G4GUO/TAPR AMBE-1000 vocoder, which used an 8-bit, μ -Law codec.

The functional organization of the AMBE-2000/2020 differs from that of the AMBE-1000 in a variety of interesting ways, but most significantly only a serial data communication channel is available. The G4GUO/TAPR AMBE-1000 vocoder utilized a bi-directional 8-bit parallel port. The AMBE-1000 featured programmable *digital* input and output gain controls. The new AMBE-2020 vocoder design also utilizes a programmable *analog* input and output gain control in the AD73311L device.

The auxiliary microprocessor is a Ubicom (formerly Scenix, www.ubicom.com) SX28A, which is a nearly one clock per instruction cycle processor. The SX-Key from Parallax (www.parallax.com) provides a convenient, low cost, 50 MHz/50 MIPS development system. The PIC 17C4X microprocessor is a 33 MHz/8.25 MIPS device, which, although possibly adequate, could have somewhat impeded the redesign effort.

Although Charles Brain G4GUO provided the AMBE-1000 PIC assembly language source as a reference, the functional differences in the new analog front-end and AMBE vocoder hardware configuration by intentional design meant that new Ubicom SX assembly language routines would be required. The hardware would have to be prototype assembled and tested.

Who would volunteer to *do* all that work?

Enter the Undergraduates

The capstone senior design project course is designed to provide a real-world experience for undergraduate students in Electrical and Computer Engineering (ECE). The Temple University Amateur Radio Club K3TU (TUARC, www.temple.edu/k3tu) was the host facility for the vocoder project. TUARC is not only an amateur radio student organization, but is supported by the ECE Department (www.temple.edu/ece) as an active part of the undergraduate curriculum in DSP and digital data communications.

The 2002-2003 ECE senior design project course requires a team approach and extended for the two semesters of the senior year. A group of three ECE undergraduates were recruited for the vocoder project: Yazime Allen, Melinda Gleiter, and Jodi Moore. They were charged with the task of comprehending the mound of technical literature on the components and formulating a plan for the design.

The plan that they pursued, logically, was to configure the AD73311L analog front-end with the auxiliary microprocessor and verify its performance with a digital loop-back test; configure the *basic* operation of the AMBE-2020 vocoder and establish output data communication; buffer the output data (the vocoder is half-duplex); and transmit the buffered data back to the vocoder input to complete the voice circuit. All of this was required to be accomplished within the confines of a six semester hour course in the throes of a 34 semester hour ECE senior year!

Although not all the tasks were completed by the April 2003 ECE senior design project course deadline, the AMBE-2020 vocoder design was extended and is now being *alpha* tested (Figures 2 and 3).

Analog Front-end

The Analog Devices AD73311L is a low power +3.3 and +5 VDC CMOS 16-bit analog front-end. The encoder channel consists of an input configuration block, a programmable gain amplifier (PGA), and a sigma-delta ADC. A digital filter, part of the over sampled sigma-delta ADC, provides significant noise reduction (+76 dB SNR). The input configuration block can provide for differential, single-ended, or analog loop-back from the decoder. The PGA can be programmed with 3-bits for discrete gains from 0 to +38 dB.

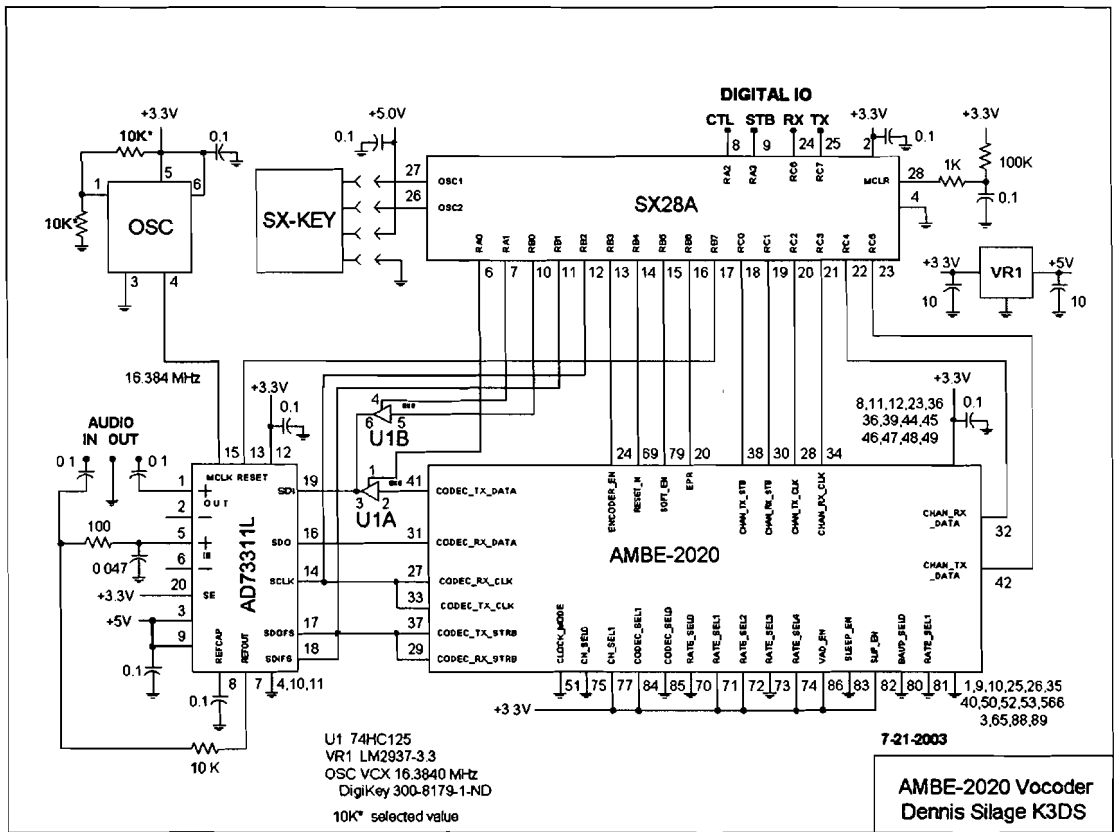


Figure 2 – K3DS AMBE-200 Alpha Vocoder Schematic

The decoder channel consists of a digital interpolator, digital sigma-delta modulator, 1-bit DAC, an analog switched capacitor filter and PGA and exhibits a low noise performance (77 dB SNR). The PGA can be programmed with 3-bits for discrete gains from +6 to -15 dB. The AD7331L has an internal band gap reference for its ADC and DAC.

The AD7331L has six internal 8-bit control registers to organize all of the complex encoder and decoder functions and a bi-directional synchronous serial port (SPORT) for communication. SPORT has a control mode, data mode and a mixed control and data mode. The AD7331L analog front-end must be properly configured on power-up through the SPORT before operation. Unfortunately, the AMBE-200 does not provide a digital pass-through channel, so the auxiliary microprocessor controls the initialization of the device through 74HC125 tristate buffers.

AMBE Vocoder

The DVS AMBE-200 is a half-duplex voice encoder and decoder⁴. The encoder receives a sampled data stream of 16-bit voice data at 8 Ks/sec (128 Kb/sec) and outputs a stream of data at 19.2 Kb/sec, consisting of 24 16-bit words (384 bits) every 20 msec. The decoder receives a stream of data at 19.2 Kb/sec and synthesizes 16-bit voice data at 8 Ks/sec. Although the vocoder data streams can be unformatted, formatted data is employed, where each 20 msec temporal segment is preceded by a known data structure.

Fortunately, the formatted data overhead from the encoder is removed by the auxiliary microprocessor and *not* transmitted. The actual transmit channel data rate is 3.6 Kb/sec here. The auxiliary microprocessor reconstitutes the 19.2 Kb/sec formatted data stream during receive from the 3.6 Kb/sec data channel for synthesis by the decoder.

The first 12 16-bit words consist of a synchronization word header, status and control information that is removed on transmit and replicated on receive by the auxiliary microprocessor. The remaining 12 16-bit words represented the encoded data bit field. These 192 bits every 20 msec are not fully populated when the data rate is less than 9.6 Kb/sec. At a data rate of 3.6 Kb/sec only the first 72 bits are used, although all 384 bits must be transferred out of the encoder.

The DVS AMBE provides active clock skew compensation (*slip*), which is enabled (*SLIP_EN*), since the passive formatted data stream is asynchronous to the clock of the analog front-end (*SCLK*). Slip control allows the vocoder to compensate for drift between the frame (determined by the auxiliary microprocessor) and the sample (determine by the oscillator *OSC*) of approximately 6000 parts per million (ppm). This feature is being testing at baseband with a second *alpha* vocoder prototype.

The DVS AMBE-2020 operates on +3.3 VDC at 65 mW. The device is packaged in a 100-pin tiny quad flat pack (TQFP) plastic mount.

Setting the Parameters

Many of the functions of the DVS AMBE-2020 can be accessed through both hardware and software interfaces. The hardware reset pin (*RESET_N*) reads the channel interface (*CH_SELn*), codec selection (*CODEC_SELn*), and coding rate selection (*RATE_SELn*) pins. These are set for the passive formatted mode, an AD73311 (but *not* the AD73311L) analog front-end, and a 2.4 Kb/sec voice data but a 1.6 Kb/sec FEC rate. This is a 4 Kb/sec data communication rate, but can be reset in operation by the auxiliary microprocessor to 2.4 Kb/sec and 1.2 Kb/sec with 5 16-bit control words. Similarly, the auxiliary microprocessor properly configures the AD73311L analog front-end.

Auxiliary Microprocessor: The Little Chip That Can

The Ubicom 50 MHz SX28A microprocessor has 2 KB of Flash memory and 136 bytes of bank switched SRAM. All of the instructions are *single clock cycle* (20 nsec) except branch instructions (60 nsec). Interrupt latency is 60 nsec and can occur on programmable, maskable, positive or negative edge triggered IO bits. The microprocessor has an 8 bit real-time clock/counter with a programmable 8-bit prescalar. The SX28A can operate on +3.3 VDC at 50 MHz with a power dissipation of 275 mW. A 75 MHz device is also available.

The auxiliary microprocessor provides digital output signals for the vocoder encoder/decoder enable (*ENCODER_EN*), the reset (*RESET_N*), the Viterbi soft decision encoding enable (*SOFT_EN*), the channel transmit and receive frame synch pulses (*CHAN_TX_STB* and *CHAN_RX_STB*), the channel transmit and receive clocks (*CHAN_TX_CLK* and *CHAN_RX_CLK*), the channel receive data stream (*CHAN_RX_DATA*), and the digital IO strobe (*STB*), control (*CTL*) and transmit data (*TX*).

The auxiliary microprocessor responds to digital input signals for the analog front-end serial clock and framing signal output (*SDOFS* and *SCLK*), the vocoder encode packet ready (*EPR*) and the channel transmit data stream (*CHAN_RX_DATA*), and the digital IO receive data (*RX*).

The *alpha* prototype K3DS AMBE-2020 vocoder is shown in Figure 3 with the KK7P DSPx module configured as a modem. The *alpha* prototype vocoder utilizes a Parallax SX-Key assembly language programming environment, which facilitates watch registers, single instruction stepping and interactive debugging. The Ubicom SX assembly language routines are event interrupt driven and provide deterministic clock and control signals. In the final design, the master clock signal produced by the SX-Key would be derived by a discrete crystal and internal oscillator. Note that the SX-Key requires +5 VDC for normal operation.

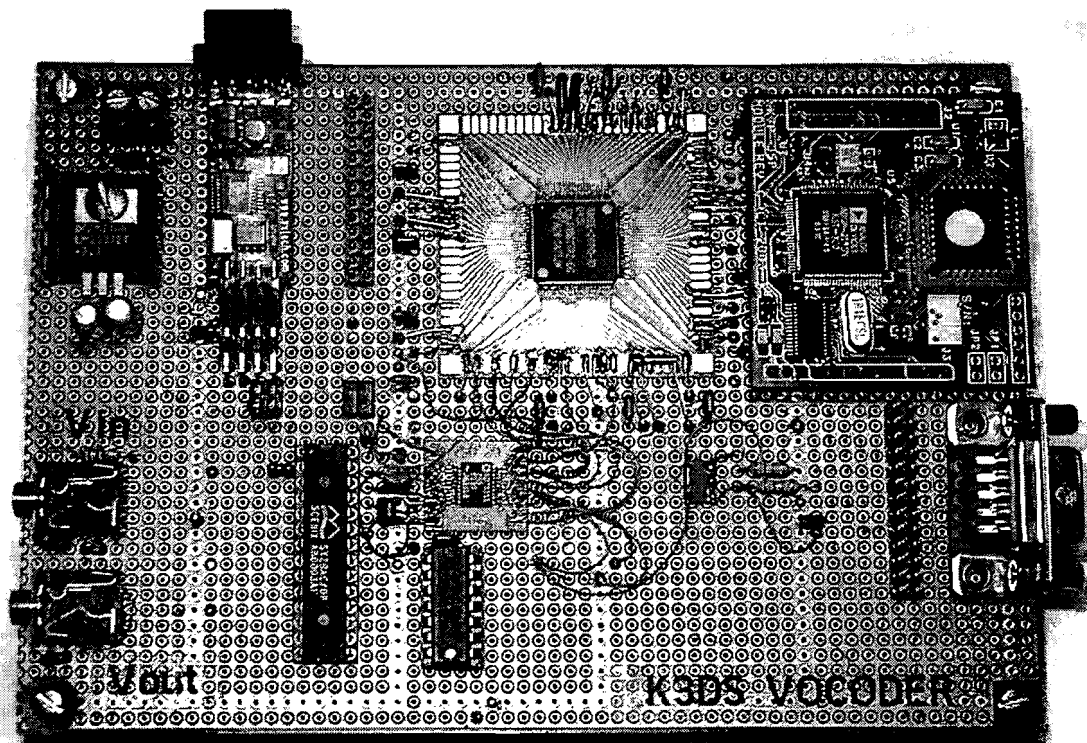


Figure 3 – K3DS AMBE-2020 *Alpha* Vocoder with the KK7P DSPx

The Ubicom SX assembly language source code for the auxiliary microprocessor will be released as a *beta* prototype with the complete hardware rendered as a printed circuit board. An assembled-BASIC compiler for the SX28A is also now available for further experimentation (www.sxwiz.com). In the final design the voltage controlled, nominally 16.384 MHz oscillator (*OSC*) will be replaced by a fixed oscillator.

Contemporary Developments

A parallel development of an AMBE-2020 vocoder was undertaken by Yoshi Nishimura JA6UHL at AOR Data Communication Equipment (www.aorja.com), which has resulted in a commercial product. The ARD9800 Fast Modem for both digital voice and digital images employs a 36-tone orthogonal frequency division multiplexing (OFDM) modem, similar to the G4GUO design². The ARD9800 could become a *de facto* amateur radio standard for digital voice *if* we chose to go in this direction. The JA6UHL digital voice data protocol and specification for the OFDM modem are available.

However, the further development of an open-source amateur radio vocoder and modem could certainly insure compatibility with a published commercial standard and probably go well beyond them in performance. Other past innovations, such as Clover™ and GTOR™, were not open source and became technical end-points for amateur radio development.

The recent availability of the KK7P DSPx module (www.tapr.org/tapr/html/Fdsp.html), utilizing an 80 MHz/80 MIP Analog Devices ADSP-2185N DSP *microcomputer*, can provide a standard platform for further development of an OFDM modem for the AMBE-2020 vocoder. The DSPx is used as an audio DSP adapter for the popular Elecraft K2 transceiver. TAPR has produced the KDSP10 adapter that converts the DSPx into an equivalent of the popular ADSP-2181 EZ Kit Lite. The digital IO port of the AMBE-2020 vocoder *alpha* prototype has been modified to accommodate the DSPx (Figure 3).

So Many Control Bits, So Many Possibilities

An open source AMBE-2020 vocoder is offered as a reasonable standard for amateur radio digital voice and modem experimentation. The 12 16-bit words that constitute the status and control information fields of the AMBE-2020 data stream provide many possibilities for an *evolving* amateur standard for a digital voice transport layer protocol beyond that of a *fixed* voice data rate of 2.4 Kb/sec with 1.2 Kb/sec FEC:

- Lost Frame Indicator bit causes the decoder to construct a 20 msec voice frame from the previous frame, which would mask the effects of short periods of data loss.
- Comfort Noise Insertion bit causes the decoder to output a frame of slight background noise, rather than complete silence, during discontinuous transmission.
- Dynamic Bit Error Rate (BER) information is provided by the decoder, which could be used to adjust the voice data and FEC rate.
- Rate Selection Bits are used to vary the encoder and decoder voice data and FEC rates.
- Viterbi decoder for rates greater than or equal to $\frac{1}{4}$ with up to 4 bits of soft decision decoding.
- DTMF tones can be generated by the encoder and detected by the decoder.

Acknowledgement

The initial design efforts of Yazime Allen, Melinda Gleiter, and Jodi Moore during their 2002-2003 ECE senior design project at Temple University is appreciated. Conversations with Doug Smith KF6DX and Charles Brain G4GUO helped to spur the redesign of the AMBE vocoder. Chip Morgan N3IW has provided and continues to provide technical support. Jim Biddle W3DCL has organized MOS testing of the AMBE-2020 vocoder and loop-back digital communication channel testing of the AMBE vocoder and modem on 222 MHz.

Bibliography

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