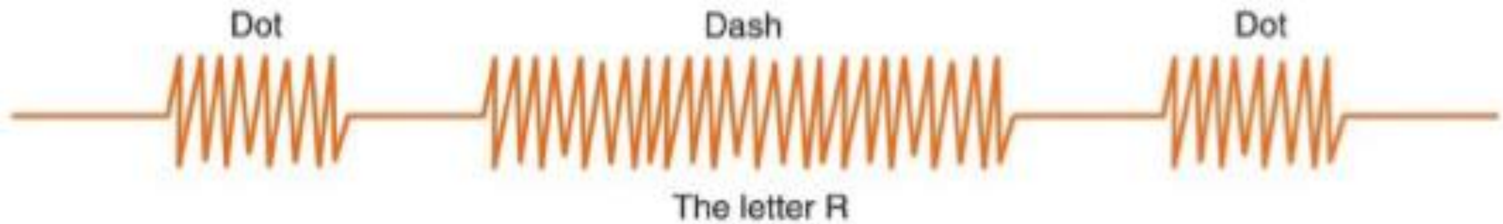
What's All This Digital Voice Stuff?

Dennis Silage K3DS

What's All This Digital Voice Stuff?

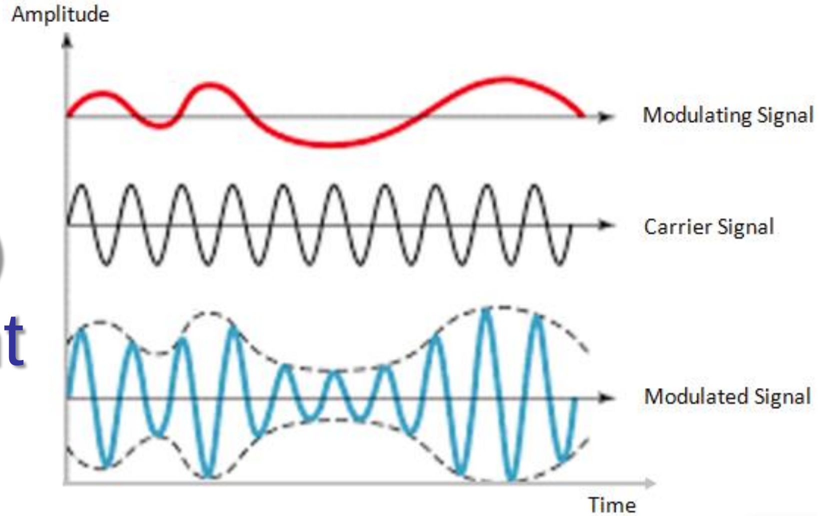
Amateur Radio communication began with a *digital* signal: a Morse coded ON/OFF keying of the RF carrier (CW).



The spark gap transmitter and receiver was rather primitive.

What's All This Digital Voice Stuff?

However, Amateur Radio soon adopted the *analog* amplitude modulation (AM) then used for entertainment radio broadcasts.

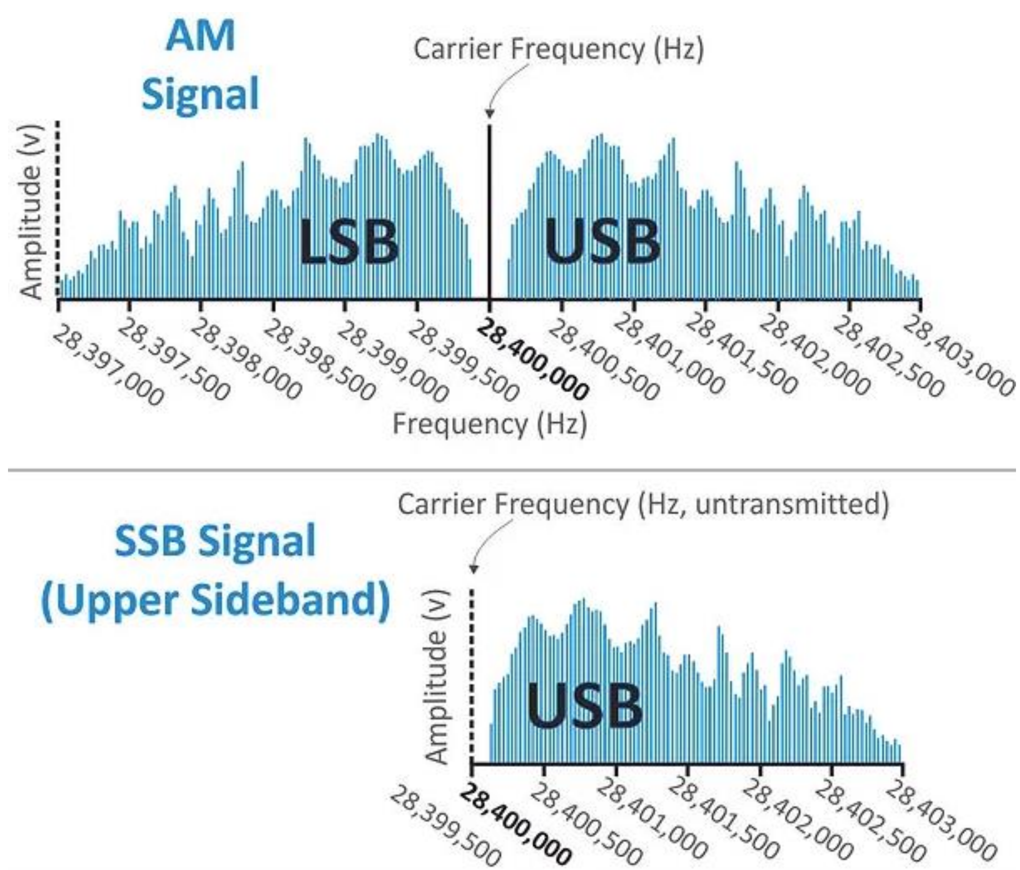


The AM transmitter and receiver was more complicated and expensive.

What's All This Digital Voice Stuff?

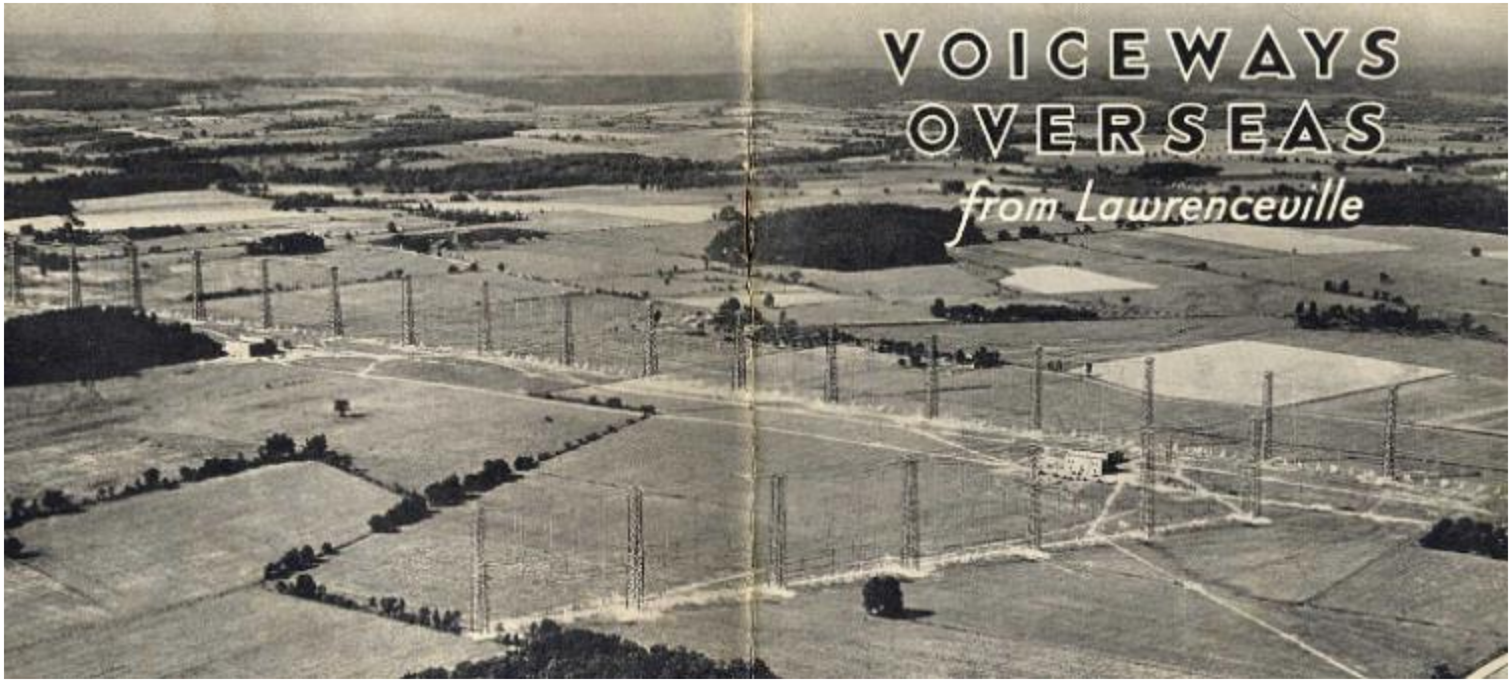
Analog single sideband amplitude modulation (SSB-AM) was also developed early. The first U.S. patent application for SSB modulation was filed in 1915 by John Carson.

The U.S. Navy experimented with SSB before WW I.



What's All This Digital Voice Stuff?

SSB first entered commercial service in 1927 on the transatlantic radiotelephone circuit between New York and London with the "antenna farm" in Lawrenceville NJ.



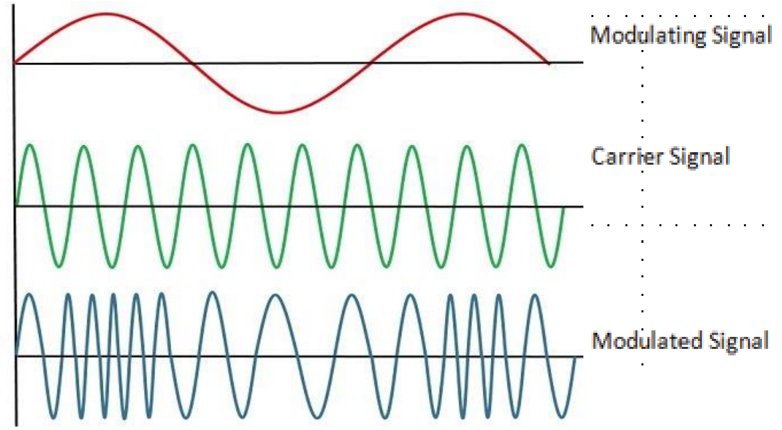
What's All This Digital Voice Stuff?

Amateur radio operators began serious experimentation with *analog* SSB after WW II. The Strategic Air Command established SSB as the radio standard for its aircraft in 1957 due to General Curtis LeMay W5EZV

Collins KWM-I HF SSB transceiver (1957)

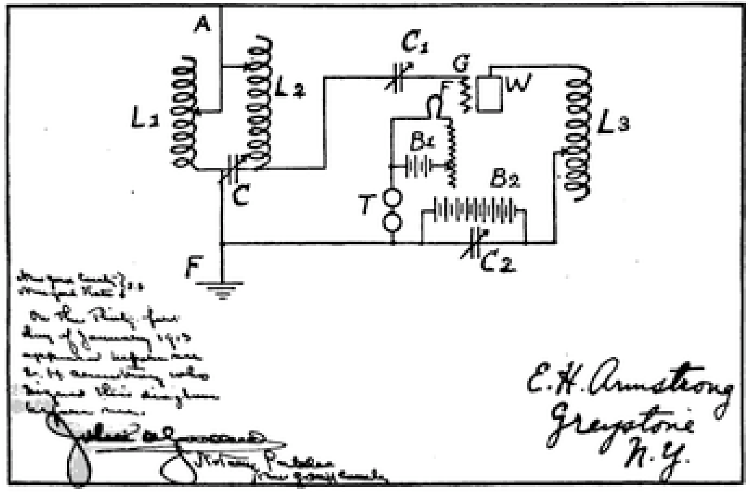
What’s All This Digital Voice Stuff?

Amateur Radio then adopted *analog* frequency modulation (FM) invented by Edwin Armstrong in 1936. FM was used primarily on the emerging VHF and UHF bands.



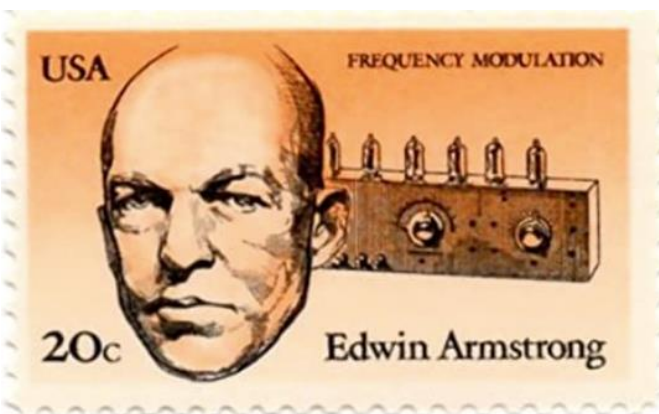
What's All This Digital Voice Stuff?

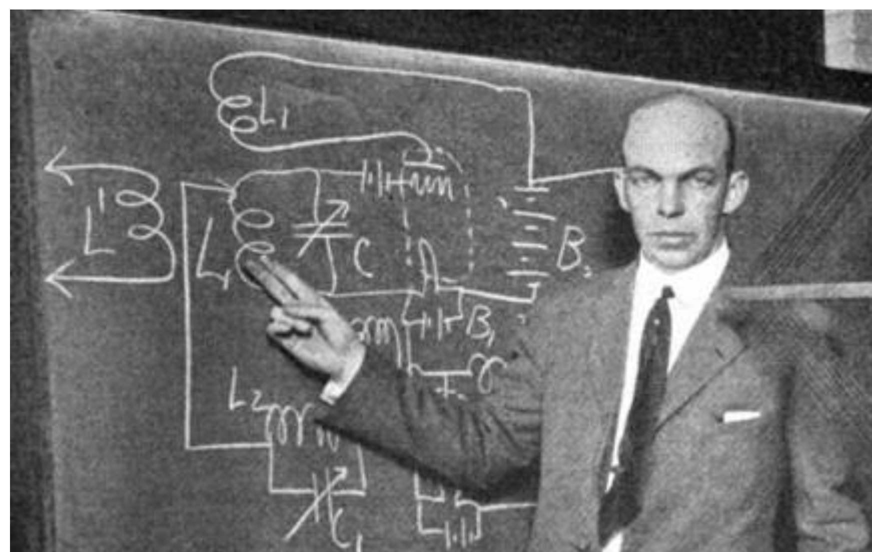
Edwin Armstrong (1890–1954) was an American electrical engineer who developed not only FM but also the regenerative and superheterodyne AM receivers.



What's All This Digital Voice Stuff?

Edwin Armstrong held 42 patents and received numerous awards, including the first Medal of Honor awarded by the Institute of Radio Engineers.





What's All This Digital Voice Stuff?

Although Amateur Radio CW, RTTY and later audio frequency shifted keying (AFSK) packet radio and WSJT are *digital* text modes, they are *not digital voice* modes.

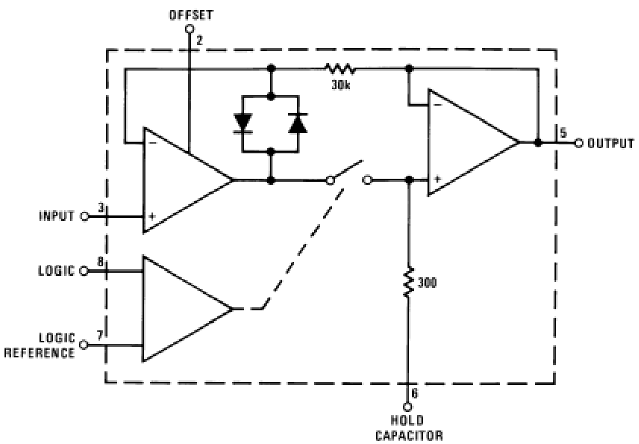
What's All This Digital Voice Stuff?

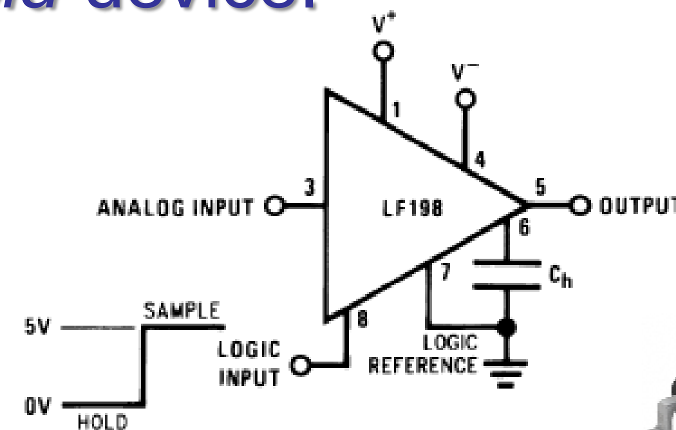
A digital voice mode requires the *sampling* and *quantization* of an *analog* voice signal producing *digital* data followed by *compression* to limit the transmission bandwidth.

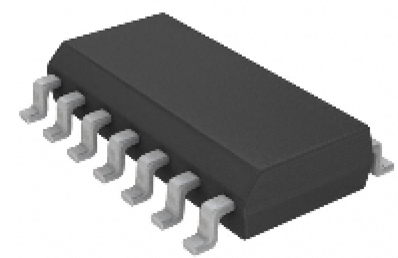
A voice signal displays a large range of amplitudes and pauses in normal speech.

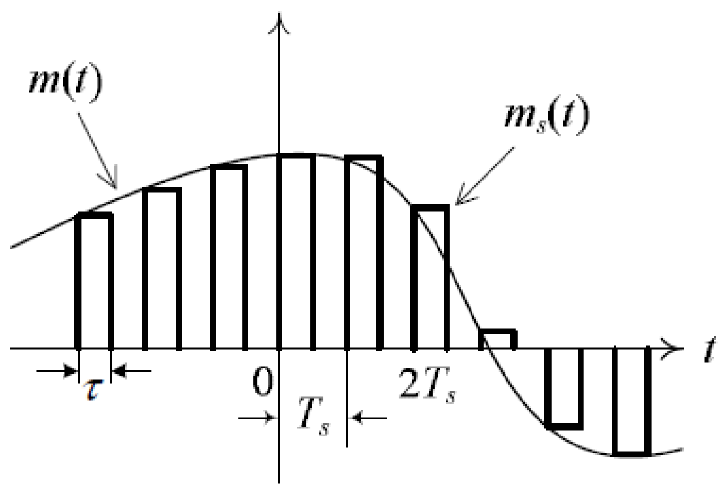
What's All This Digital Voice Stuff?

Sampling is the process when periodic samples of an analog signal such a voice is measured by a *sample-and-hold* device.



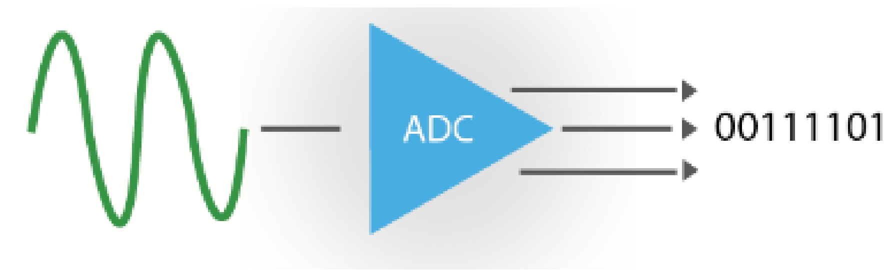


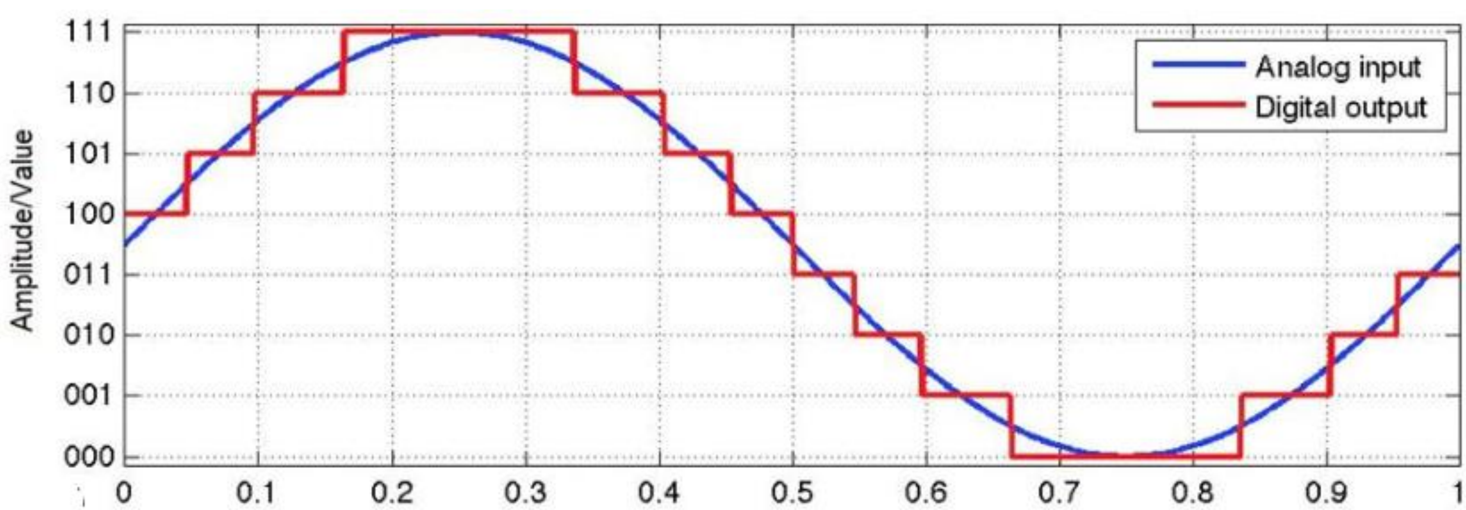




What's All This Digital Voice Stuff?

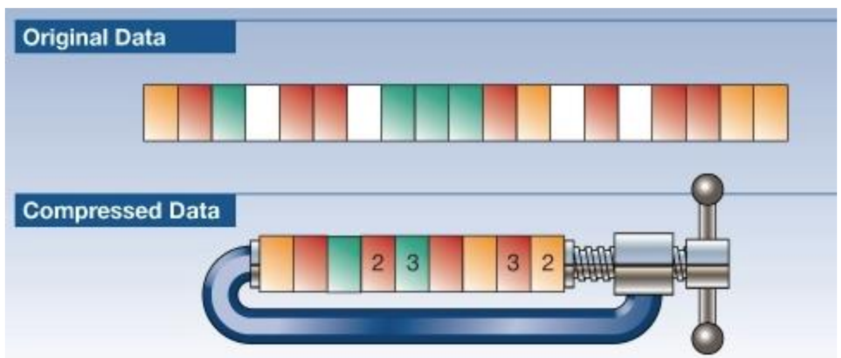
Quantization the conversion of the periodic samples to a digital value that can be processed using an *analog-to-digital* converter.

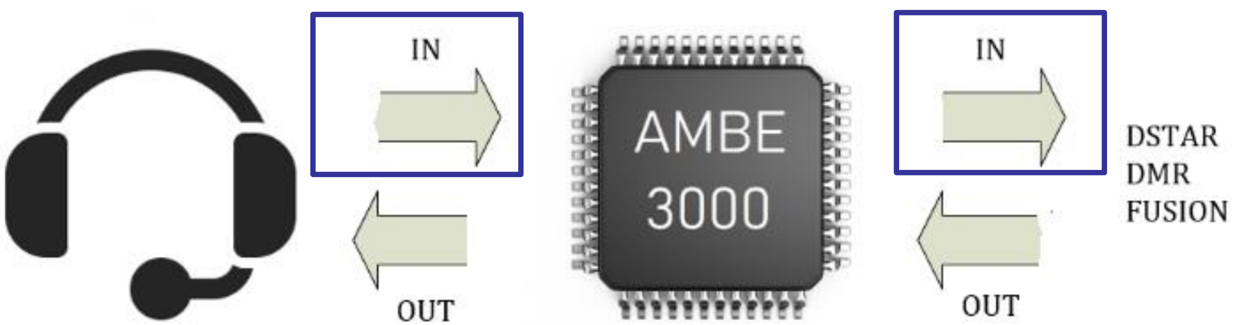




What's All This Digital Voice Stuff?

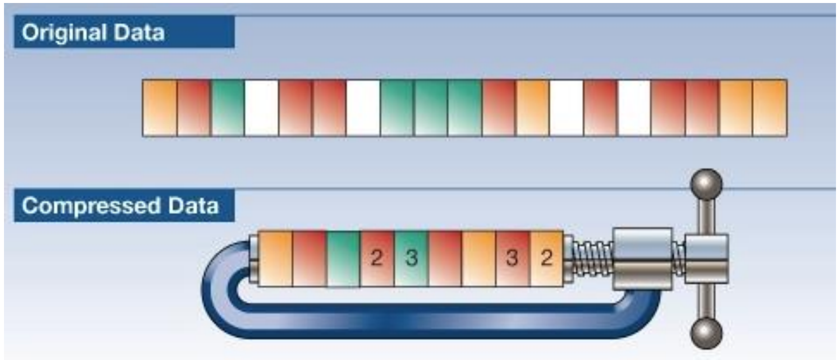
Compression processes the original data rate of the digital values in bits per sec (b/sec) of the voice signal and reduces that number to effectively limit the RF transmission bandwidth.

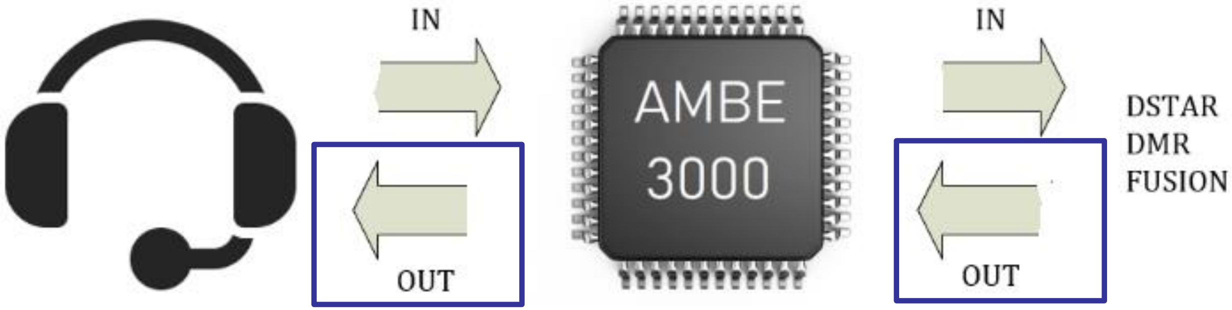




What's All This Digital Voice Stuff?

Decompression attempts to restore the original data rate of the compressed digital values in bits per sec (b/sec) of the voice signal after RF reception.

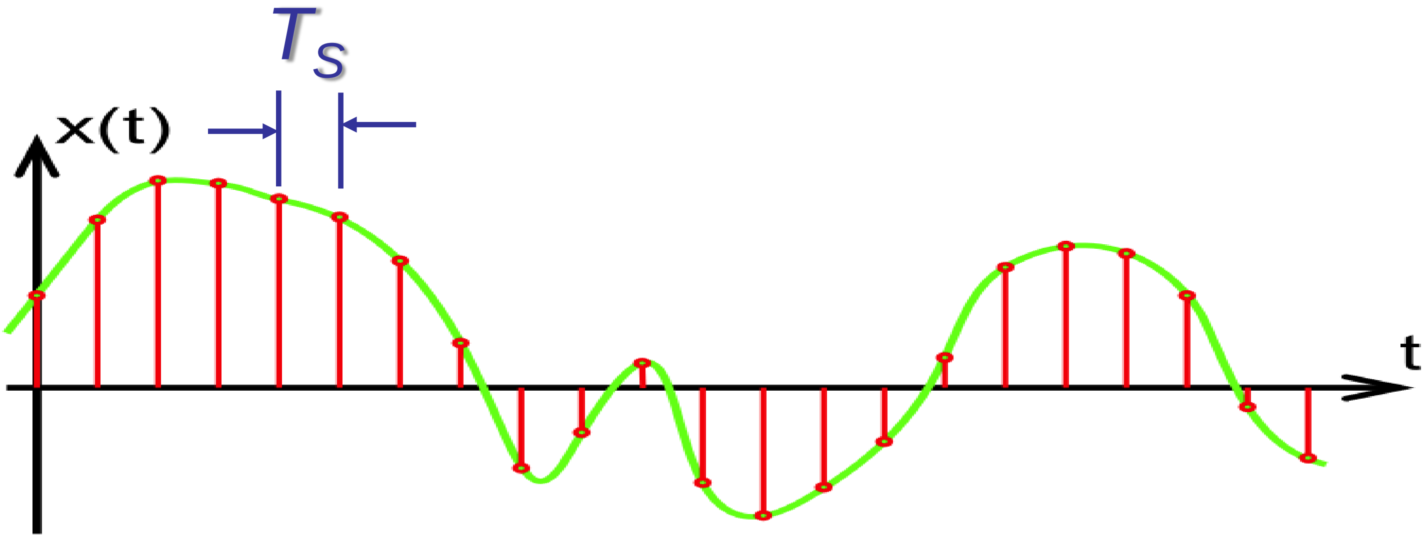




What's All This Digital Voice Stuff?

The analog voice signal is *sampled* (a value is recorded) at a fixed interval (the sampling period T_S).

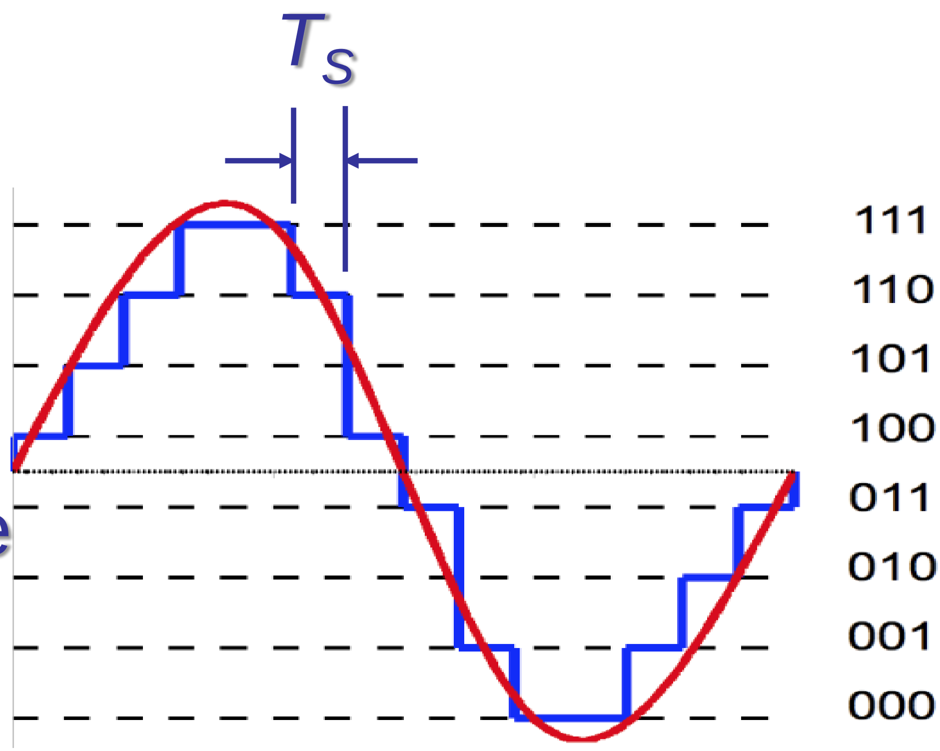
For a *communications quality* voice signal (up to 3 kHz) T_S = 0.125 msec (or a rate of 8 kHz).



What's All This Digital Voice Stuff?

After the analog voice signal is sampled, the recorded values are *quantized* (converted to an approximate digital value) by an analog-to-digital converted (ADC).

The example here is an analog sinusoidal signal quantized to only 3 bits of resolution or 8 possible values at the sampling period T_S.

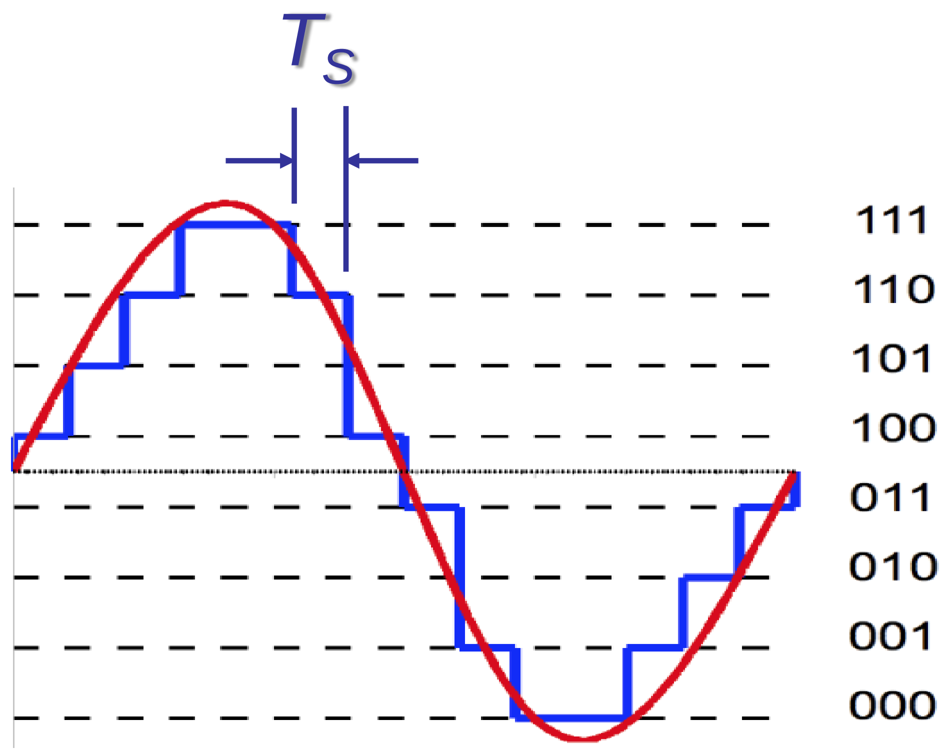


What's All This Digital Voice Stuff?

Quantization accuracy is affected by the tradeoff of the sampling period (T_S) and the number of bits (n) in each sample.

The data rate (r_b) is:

$r_b = n / T_S$ b/sec



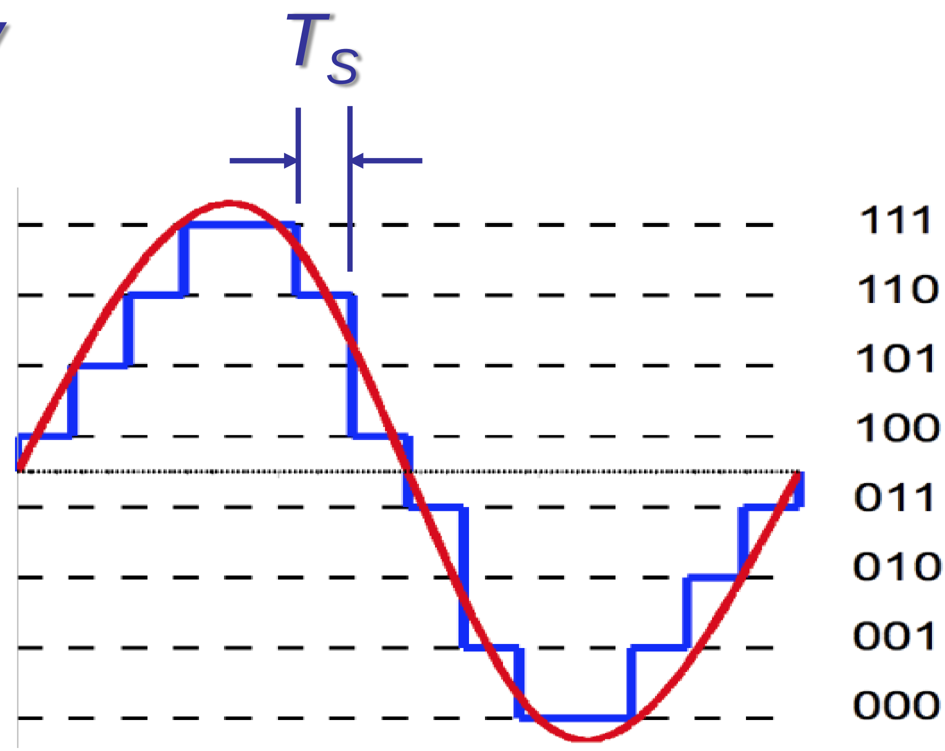
What's All This Digital Voice Stuff?

The 3 bit example here is encoded as follows:

111 = 7	101 = 5	011 = 3	001 = 1
110 = 6	100 = 4	010 = 2	000 = 0

This is straight *binary encoding*.

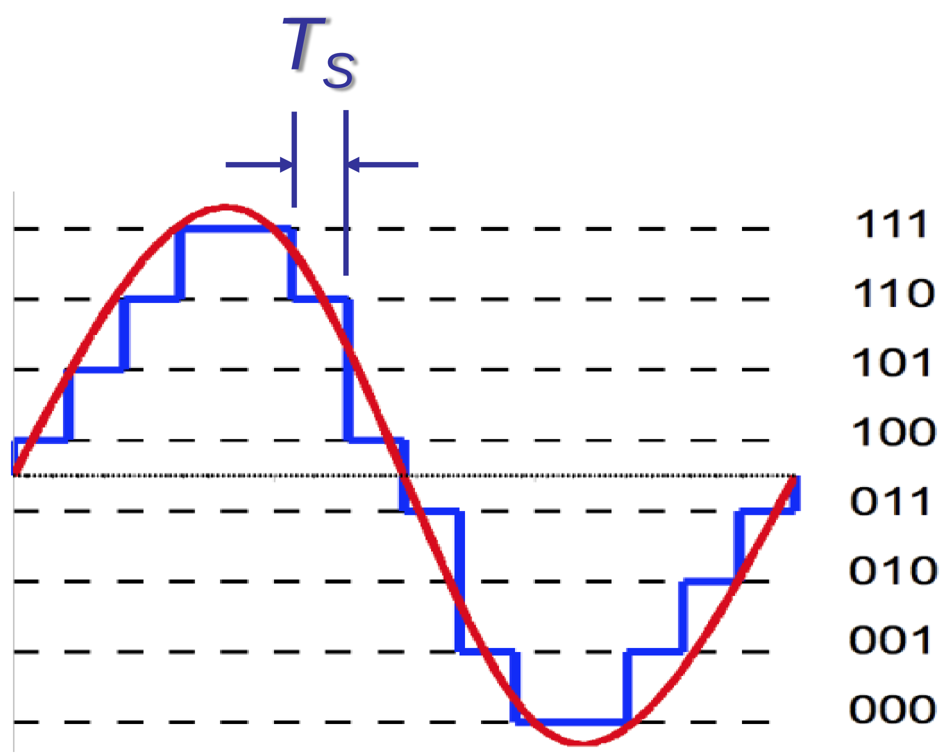
The data rate for even this crude example is $3 \times 1/T_S = 3 \times 8000 =$ 24 000 b/sec



What's All This Digital Voice Stuff?

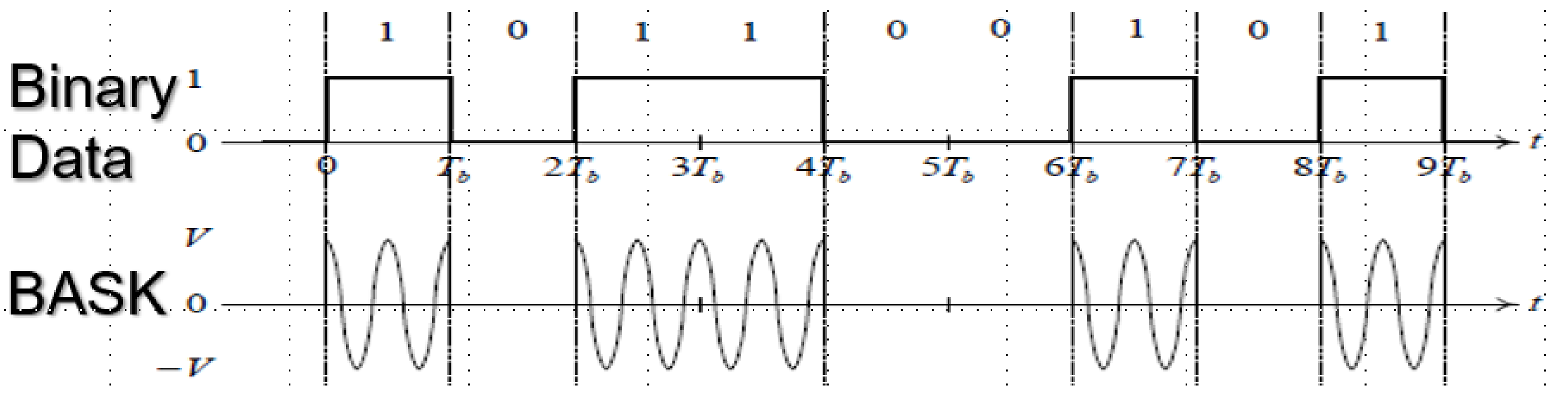
For a reasonable sounding voice, 8 bits of resolution is used (as in telephony) resulting in $8 \times 1/T_S = 8 \times 8000 = 64\ 000$ b/sec = 64 kb/sec.

Simple modulation techniques for binary digital data include amplitude, phase and frequency shift keying (BASK, BPSK and BFSK).

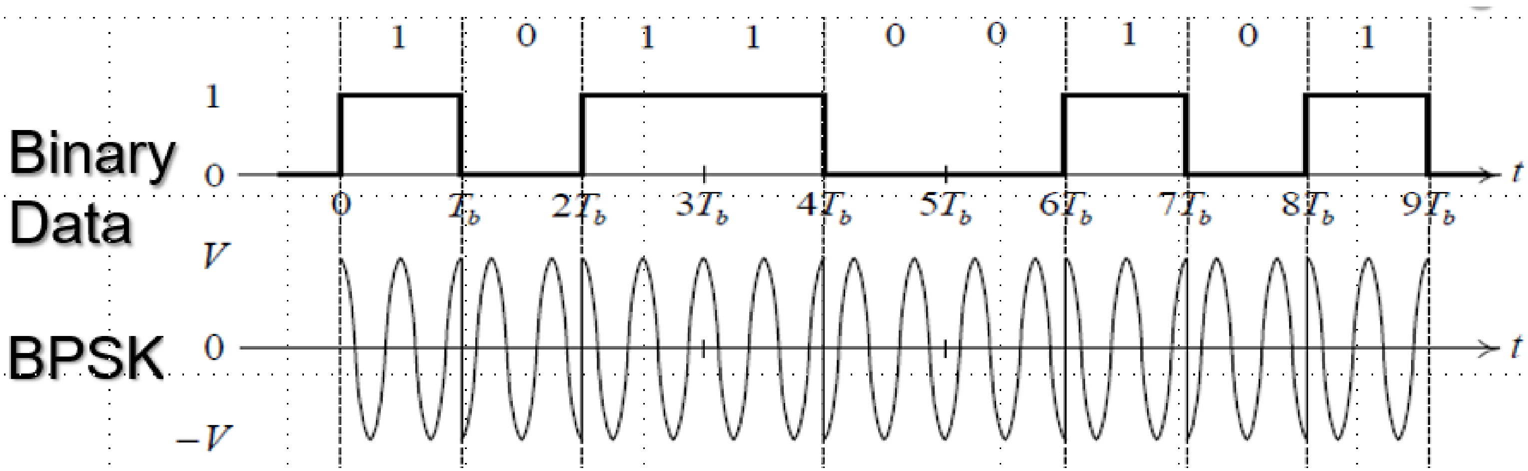


What's All This Digital Voice Stuff?

BASK – Binary Amplitude Shift Keying

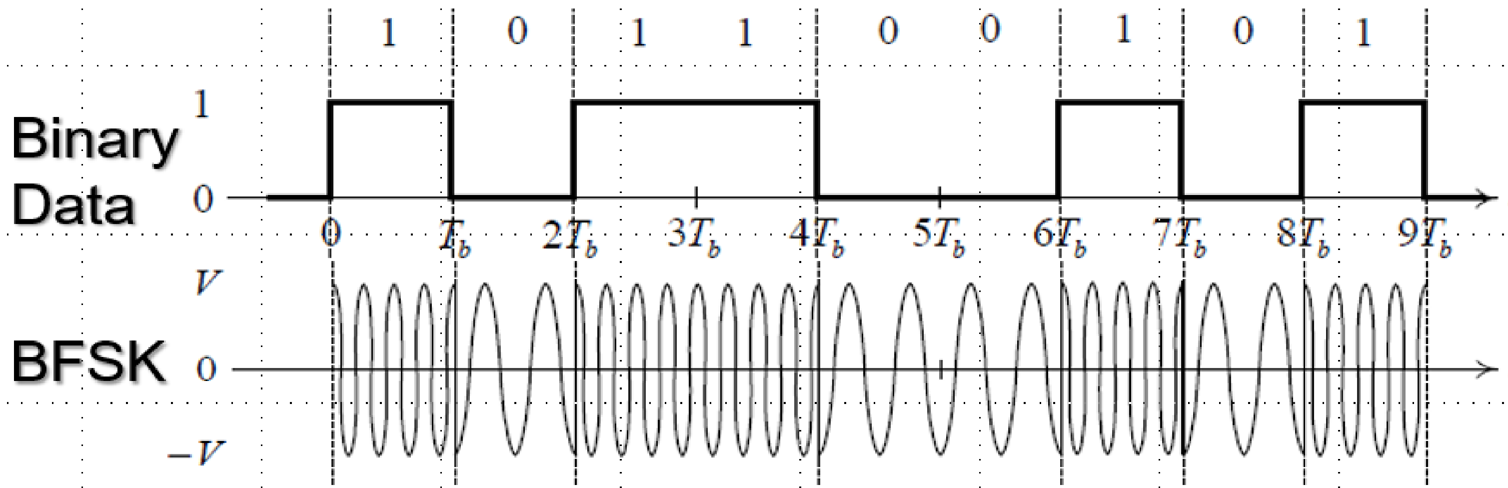


BPSK – Binary Phase Shift Keying



What's All This Digital Voice Stuff?

BFSK – Binary Frequency Shift Keying



BASK and BPSK require a transmission bandwidth of *four times* the data rate, while BFSK requires a transmission bandwidth of *four times* the data rate *plus twice* the frequency deviation (for 95% power).

What's All This Digital Voice Stuff?

A transmission bandwidth of four times the data rate of 64 kb/sec or 256 kHz for BASK and BPSK or greater for BFSK is unacceptable for Amateur Radio use and *data compression* is required.

If the original 64 kb/sec could be substantially reduced by data compression, then the transmission bandwidth would be also reduced appropriately.

What's All This Digital Voice Stuff?

Data compression can be accomplished by a specialized processor implementing the Advanced Multi-Band Excitation (AMBE) algorithm.

The first AMBE processor was developed by DVSI in 1997 (AMBE-1000) and used to produce a *vocoder* (voice encoder) to compress the quantized voice signal from 64 kb/sec to as low as 2.4 kb/sec.

DIGITAL VOICE SYSTEMS, INC.
The Speech Compression Specialists

What's All This Digital Voice Stuff?

AMBE is a *codebook-based* vocoder that compresses the 64 kb/sec data to bitrates of between 2.4 and 9.6 kbit/s at a sampling rate of 8 kHz in 20 msec *frames*.

A 20 msec frame of data is correlated to a preconfigured codebook of possible values by the processor to reduce the data rate.

DIGITAL VOICE SYSTEMS, INC.
The Speech Compression Specialists

What's All This Digital Voice Stuff?

AMBE is based on the original Multi-Band Excitation Vocoder (1987) and improved by DVSI.

AMBE is used by the Inmarsat and Iridium satellite telephony systems and the APCO Project 25 public safety systems.

Multi-Band Excitation Vocoder

RLE Technical Report No. 524

March 1987

Daniel W. Griffin

Research Laboratory of Electronics
Massachusetts Institute of Technology
Cambridge, MA 02139 USA

What's All This Digital Voice Stuff?

AMBE is used in the ICOM D-Star, DMR and Yaesu System Fusion II Amateur Radio digital voice systems.

AMBE has met some criticism from the amateur radio community because its patent and licensing runs counter to the openness of amateur radio.

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The Speech Compression Specialists

What's All This Digital Voice Stuff?

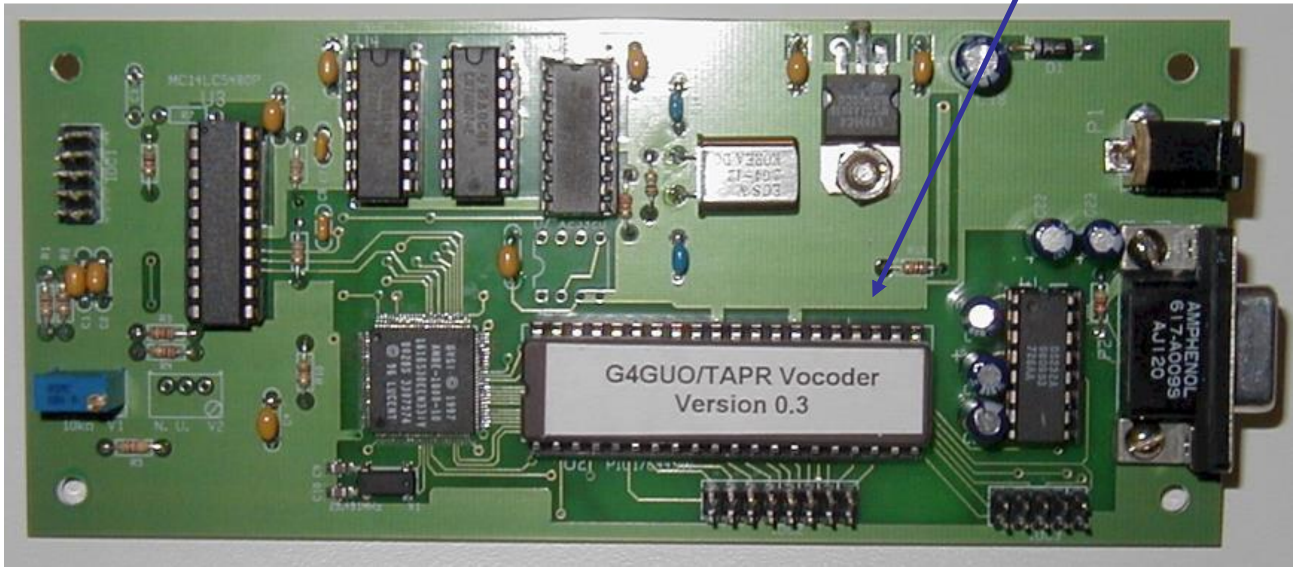
However, the configuration of the AMBE processor is open-source, the devices are readily available and many Amateur Radio devices are now available using the device.

ZUMspot AMBE Server

DVMEGA DVstick

What's All This Digital Voice Stuff?

The Amateur Radio research organization TAPR produced a vocoder using the AMBE-1000 device in 1999.



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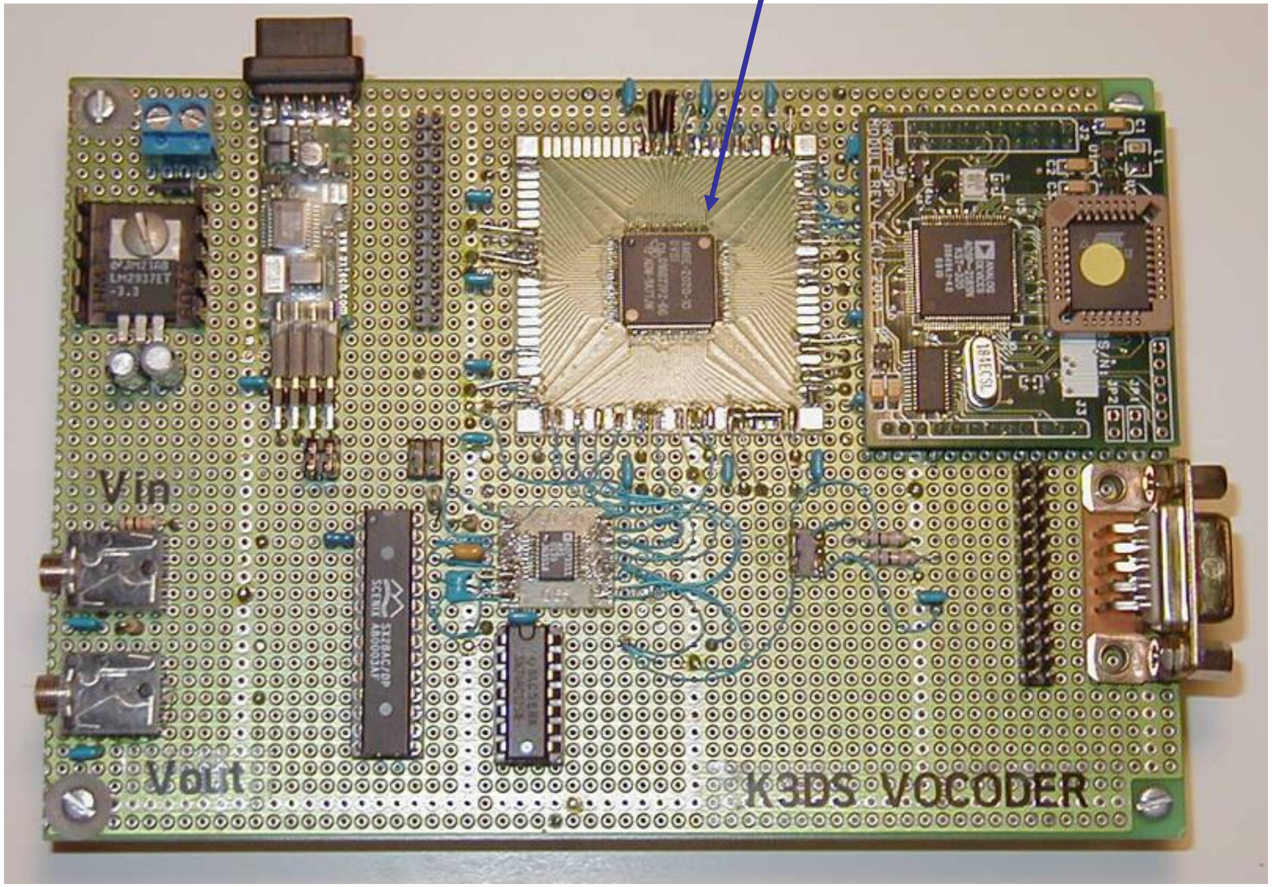
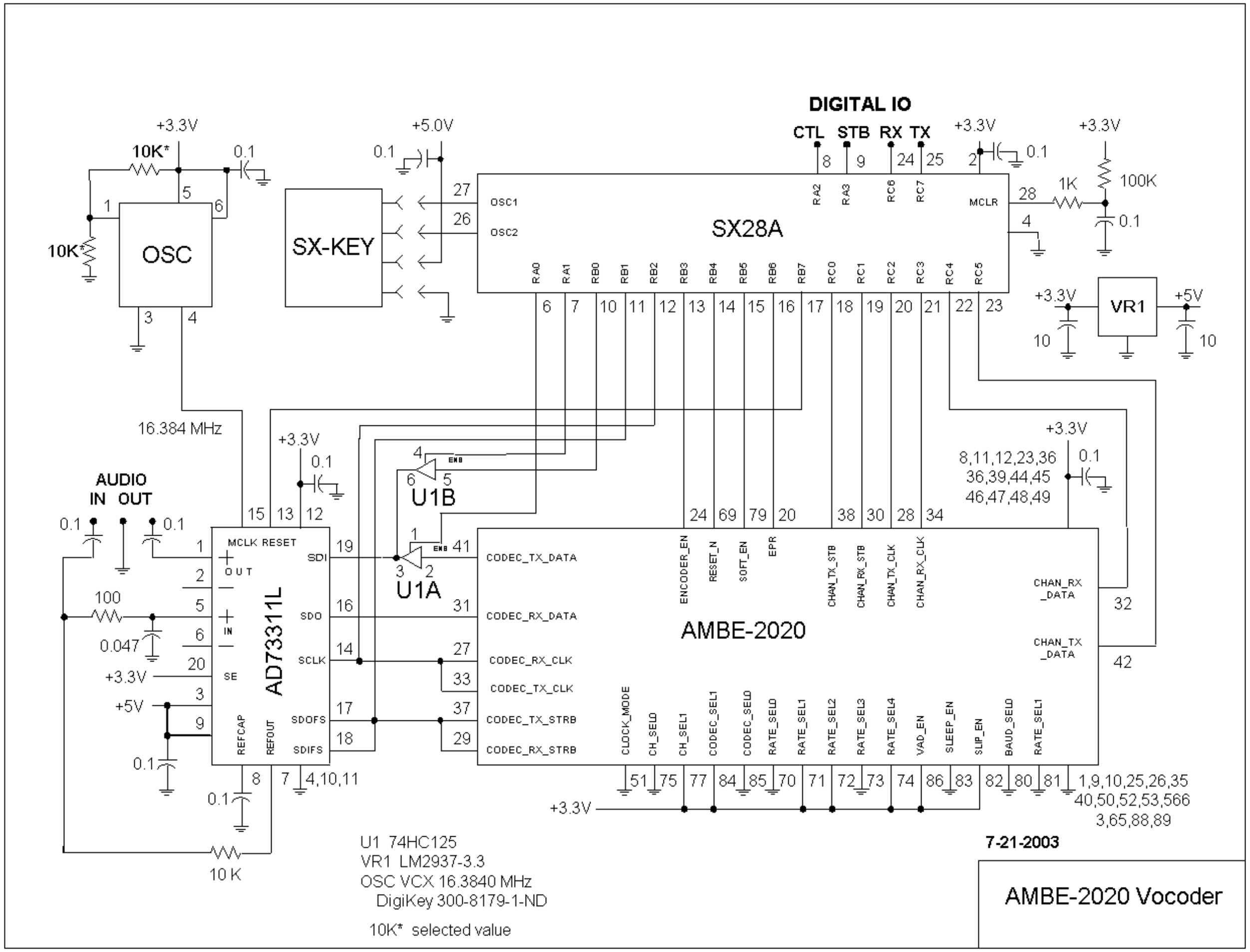
What's All This Digital Voice Stuff?

TUARC K3TU produced a vocoder using the second-generation AMBE-2020 in 2003.



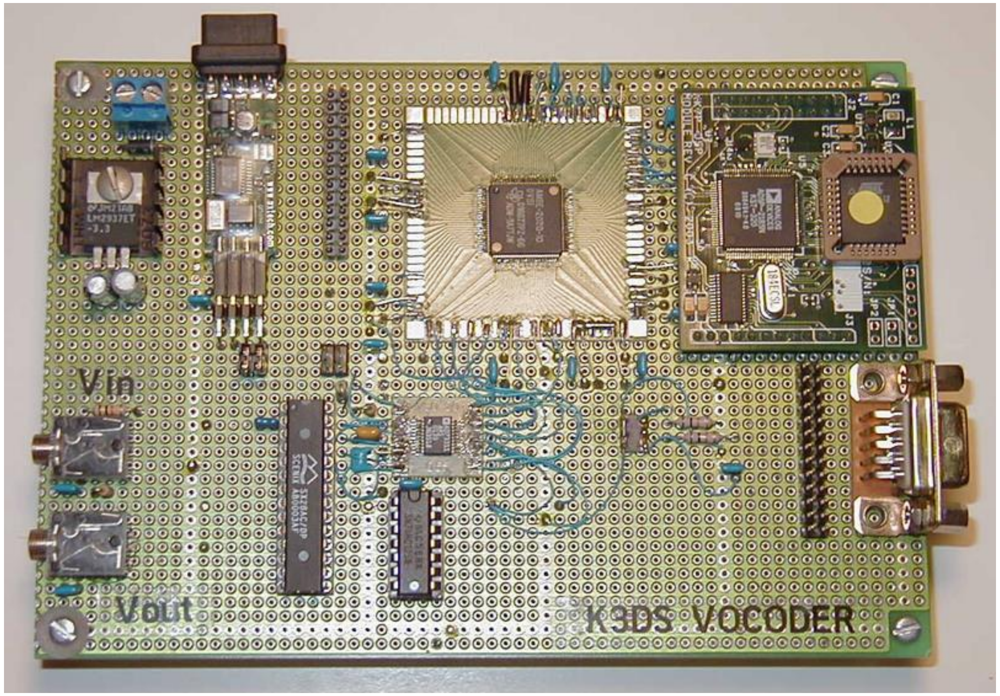
What's All This Digital Voice Stuff?

The AMBE-2020 vocoder was presented at the 2004 TAPR Digital Communications Conference.



What's All This Digital Voice Stuff?

The AMBE-2020 was available *over-the-counter* ($22 in 2003). The TUARC ECE Senior Design project was configured for ICOM D-Star and worked!

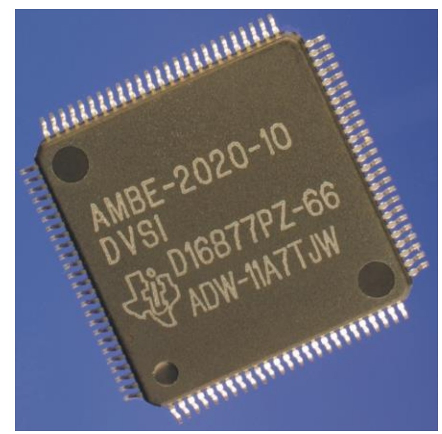


What's All This Digital Voice Stuff?

Amateur Radio digital voice techniques all utilize the now third and fourth generation AMBE processing algorithm.

What's All This Digital Voice Stuff?

The AMBE-2020 vocoder was used in the first Amateur Radio digital voice system: ICOM D-Star in 2004. The first product was the IC-2200H 2 m analog FM and digital voice transceiver.



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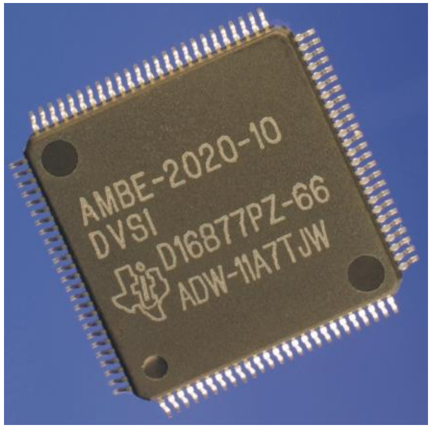
What’s All This Digital Voice Stuff?

The D-STAR digital voice protocol has the voice signal sampled and quantized to 64 kb/sec but compressed and encoded as 3.6 kb/sec data using AMBE.



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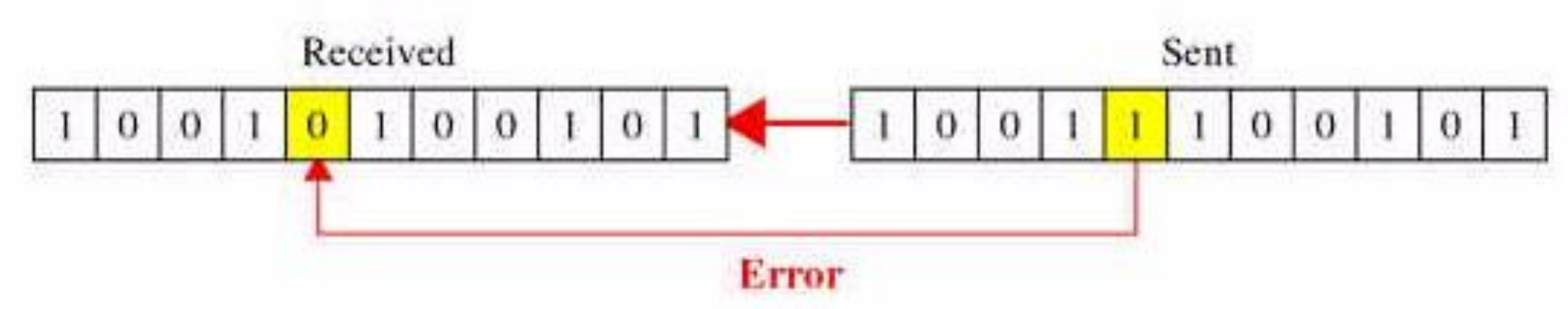
What's All This Digital Voice Stuff?

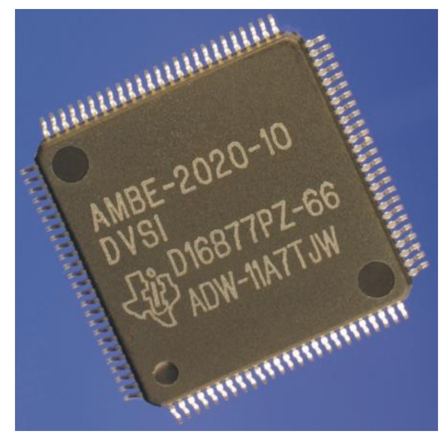
However, for robust data transmission 1.2 kb/sec of *forward error correction* (FEC) is added resulting in a bit rate of 3.6 + 1.2 kb/sec = 4.8 kb/sec for the 2 m, 70 cm and 23 cm bands.

FEC can detect errors in the data transmission and to some extent fix the errors on reception.



What's All This Digital Voice Stuff?

However, if the D-STAR digital voice protocol at 4.8 kb/sec used BASK or BPSK the transmission bandwidth would be an unacceptable 19.2 kHz or using BFSK would be even greater.



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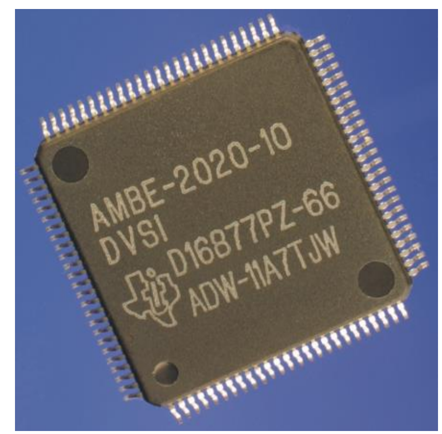
What's All This Digital Voice Stuff?

But D-STAR digital voice protocol at 4.8 kb/sec used an advanced modulation technique (Gaussian Minimum Shift Keying or GMSK) to lower the transmission bandwidth to only 6 kHz, lower than the 19.2 kHz expected.



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What’s All This Digital Voice Stuff?

Digital Mobile Radio (DMR) is an open standard for digital voice defined by the European Telecommunications Standards Institute (ETSI) first proposed in 2005.

DMR was designed for reduced transmission bandwidth and more users on a particular frequency by data compression and time division multiple access (TDMA).

DMR
DIGITAL MOBILE RADIO

What's All This Digital Voice Stuff?

DMR also uses 8 bits of resolution sampled at 0.125 ms (8 kHz) again resulting in 64 kb/sec.

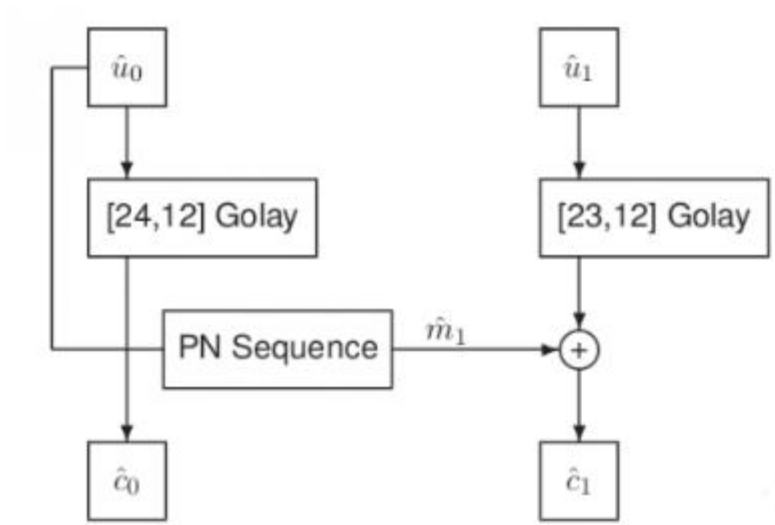
The 64 kb/sec is compressed and encoded as 2.45 kb/sec data using AMBE with a complex FEC protocol resulting in a data rate of 3.6 kb/sec.

DMR
DIGITAL MOBILE RADIO

What's All This Digital Voice Stuff?

The DMR FEC protocol is described in detail but is very complex in operation.

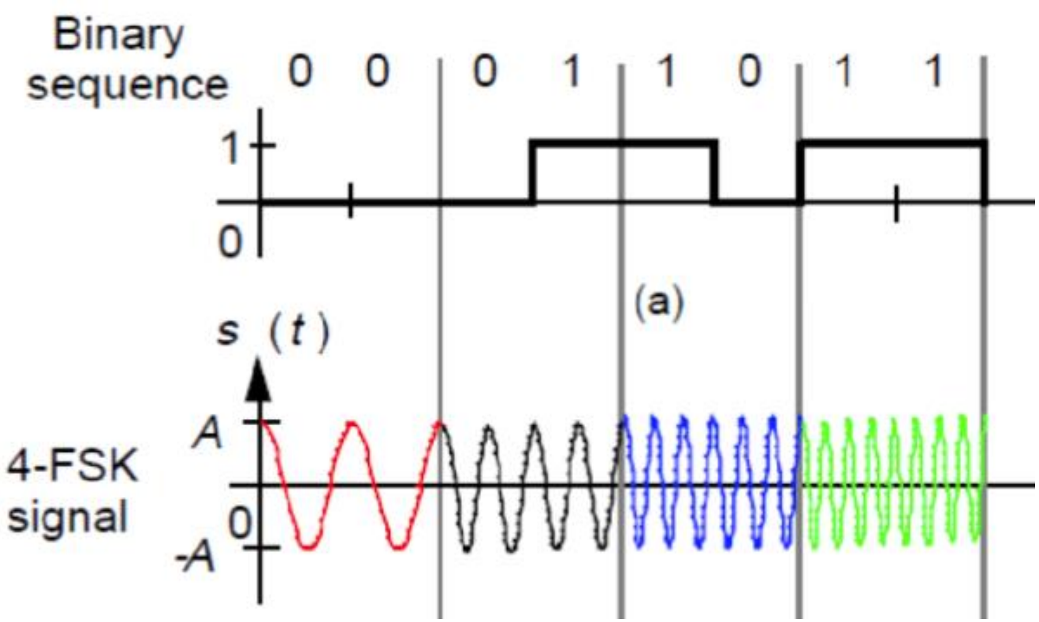
DMR FEC:



DMR
DIGITAL MOBILE RADIO

What's All This Digital Voice Stuff?

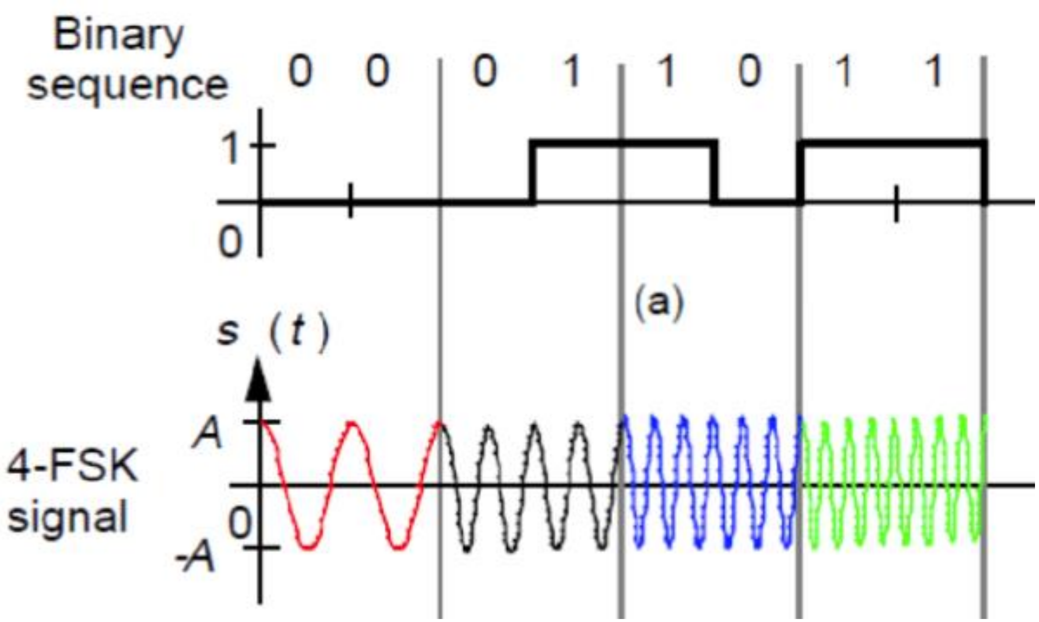
DMR uses four frequency shift keying (4-FSK) where two bits at a time (*dibit*) are sent as one of four frequencies with respect to the carrier.



What's All This Digital Voice Stuff?

The dibits are offset as +1944 Hz, +648 Hz, -648 Hz and -1944 Hz from the carrier frequency.

Data compression results in a data rate of 3.6 kb/sec but here are two such channels available.



DMR
DIGITAL MOBILE RADIO

What's All This Digital Voice Stuff?

DMR has a two 30 msec slot, time division mode which allows two users to use the same frequency for transmission.

With two channels and 4-FSK modulation the overall transmission bandwidth is 12.5 kHz.

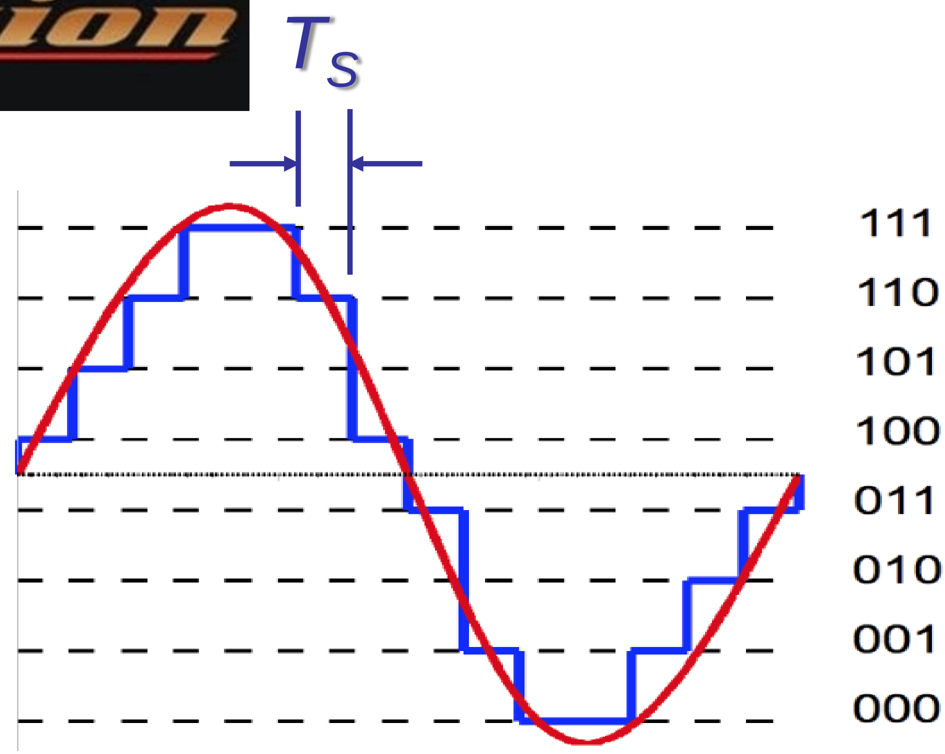
DMR
DIGITAL MOBILE RADIO

What's All This Digital Voice Stuff?

In 2015 Yaesu introduced System Fusion II (a successor to an earlier System Fusion) to implement digital voice. The first product was the analog FM and digital voice 2 m and 70 cm FTM-100DR and FTM-400XDR.

What's All This Digital Voice Stuff?

Yaesu System Fusion II also uses 8 bits of resolution sampled at 0.125 ms (8 kHz) again resulting in 64 kb/sec.



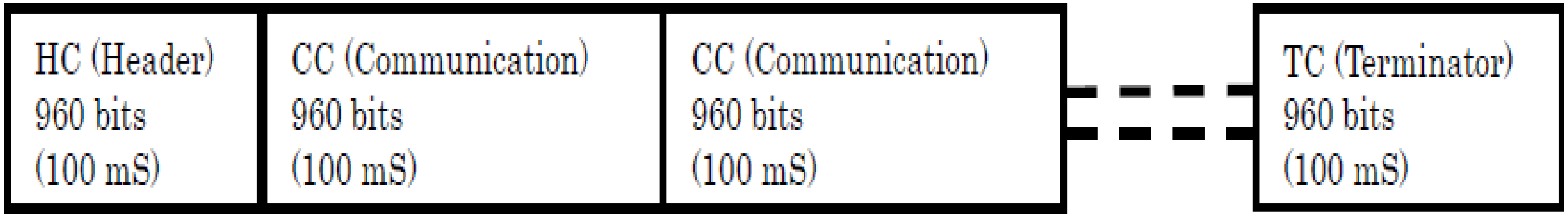
What's All This Digital Voice Stuff?

The System Fusion II voice signal sampled and quantized to 64 kb/sec is compressed and encoded as 4.4 kb/sec data for high quality voice using AMBE.

Again, for robust data transmission 2.8 kb/sec of *forward error correction* (FEC) is added resulting in a bit rate of 4.4 + 2.8 kb/sec = 7.2 kb/sec.

What's All This Digital Voice Stuff?

System Fusion II digital voice is a complex protocol with several modes of operation. There are frame headers and terminators and additional information that raises the data rate to 9.6 kb/sec.



What's All This Digital Voice Stuff?

Yaesu has provided a complete description of the protocol.

Frame composition:

HC (Header)	CC (Communication CH)				...	TC (Terminator)
	FN=0	FN=1	...	FN=7 (maximum)		
← 100 msec (960 bit) →	← 100 msec (960 bit) →	← 100 msec (960 bit) →		← 100 msec (960 bit) →		← 100 msec (960 bit) →

Header and terminator composition:

FS	FICH	DCH-1(0)	DCH-2(0)	DCH-1(1)	DCH-2(1)	DCH-1(2)	DCH-2(2)	DCH-1(3)	DCH-2(3)	DCH-1(4)	DCH-2(4)
40	200	72	72	72	72	72	72	72	72	72	72

What's All This Digital Voice Stuff?

If the System Fusion II digital voice protocol at 7.2 kb/sec used BASK or BPSK the transmission bandwidth would be a very unacceptable 28.8 kHz or using BFSK would be even greater.

System Fusion II

What's All This Digital Voice Stuff?

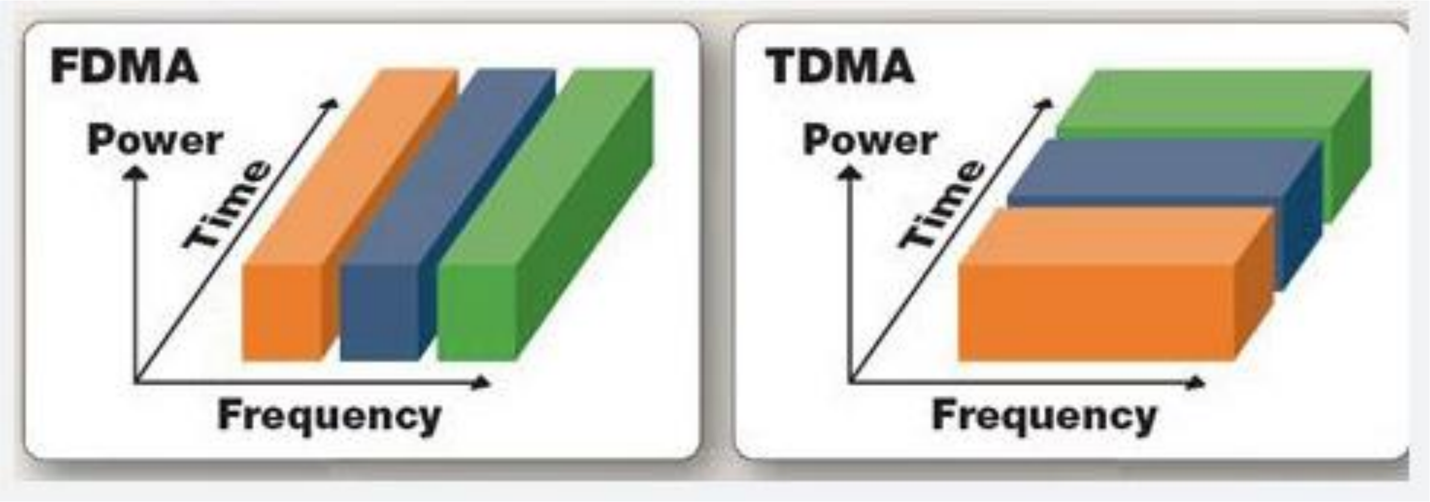
But the System Fusion II digital voice protocol at 7.2 kb/sec uses C4FM (Continuous Four Level Frequency Modulation) for a transmission bandwidth of 12.5 kHz, lower than the 28.8 kHz expected.

C4FM is a special case of 4-FSK as in DMR with intentionally smoother transitions with each symbol level.

What's All This Digital Voice Stuff?

C4FM is used in conjunction with Frequency Division Multiple Access (FDMA).

DMR uses 4-FSK but is impaired because of the use of TDMA during the time slot transitions.



What's All This Digital Voice Stuff?

C4FM and FDMA is the same mode that is used in APCO P25 Phase 1 public service transceivers, but it is not compatible System Fusion II at the level of the protocol.

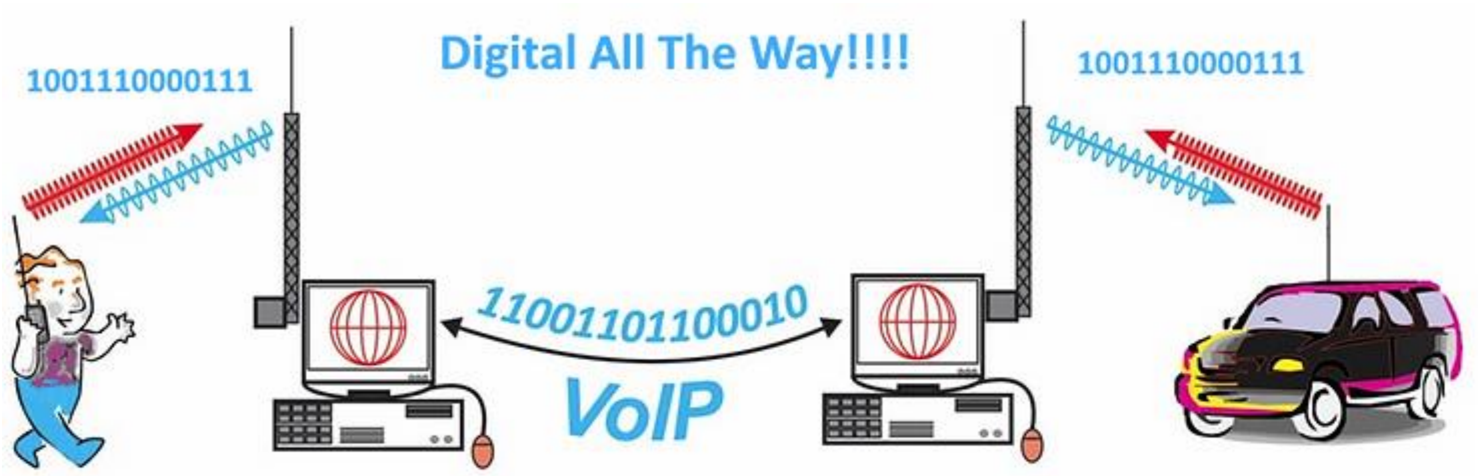
C4FM also uses four frequency shift keying where two bits at a time (*dibit*) are sent as one of four frequencies with respect to the carrier.

Dibit	Symbol	Frequency Deviations(Wide)	Frequency Deviations(Narrow)
00	+1	+900 Hz	+450 Hz
01	+3	+2700 Hz	+1350 Hz
10	-1	-900 Hz	-450 Hz
11	-3	-2700 Hz	-1350 Hz

What's All This Digital Voice Stuff?

Finally, *what good is digital voice*?

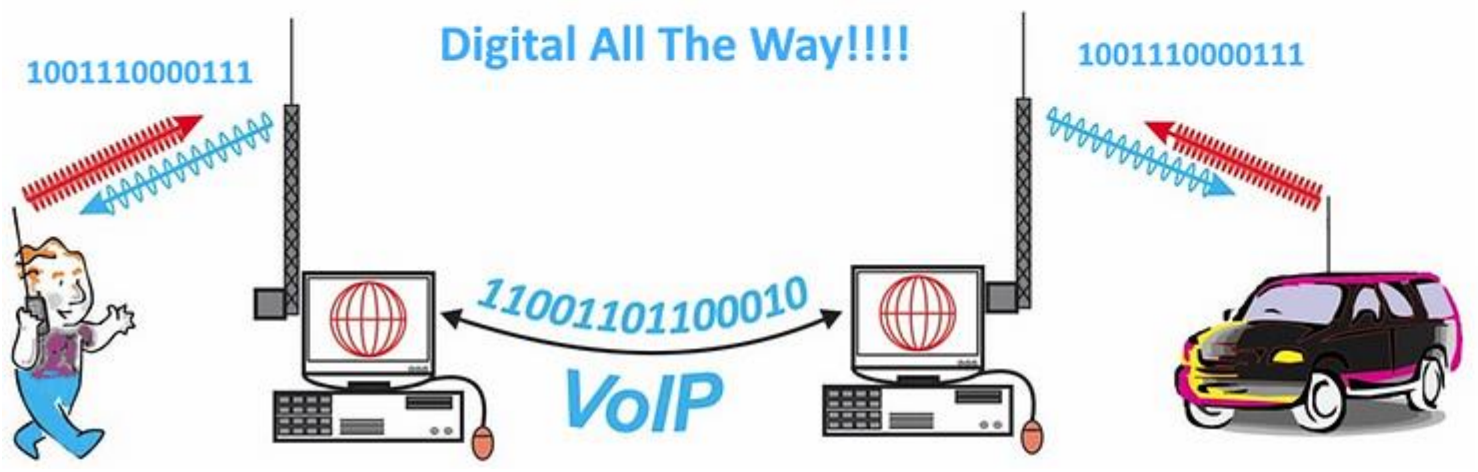
Linking repeaters has been done somewhat primitively in Amateur Radio on analog FM with *Echolink* for decades. But routing and access has been limited.



What's All This Digital Voice Stuff?

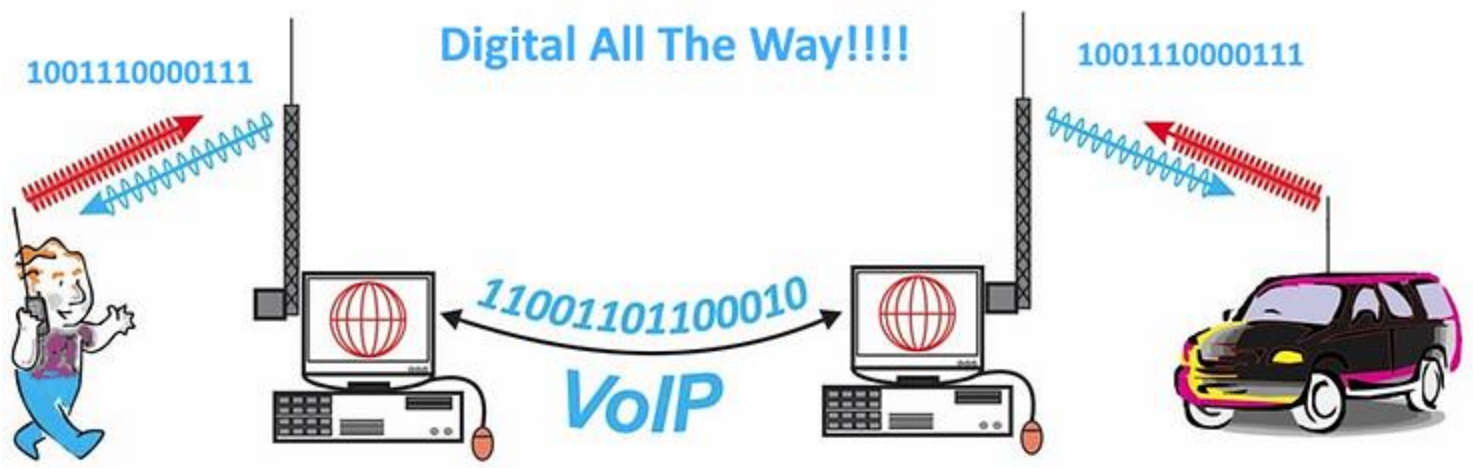
The digital voice systems are much more amenable to such routing and access.

The MARC 444.050 MHz Darby and 445.675 MHz Paoli repeaters are both using Yaesu System Fusion II.



What's All This Digital Voice Stuff?

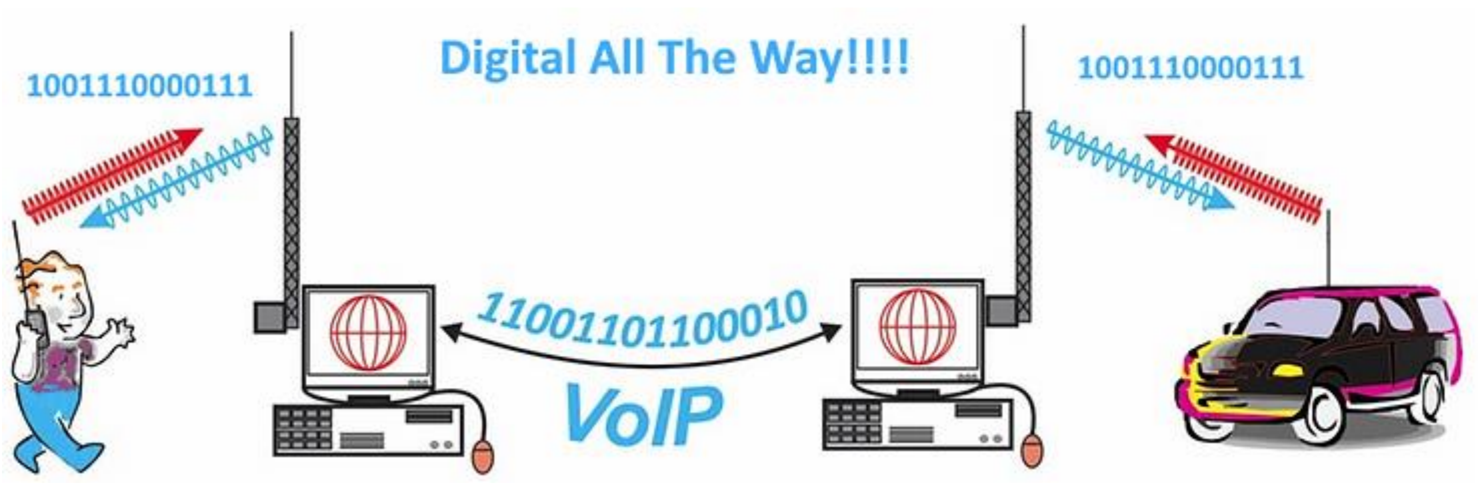
The MARC 445.675 MHz Paoli repeater is currently node-connected by Steve K3ZFT to the Keystone Wide Digital Net.



What's All This Digital Voice Stuff?

The MARC 444.050 MHz Darby repeater will be node-connected after repair and may link Paoli and Darby for increased coverage.

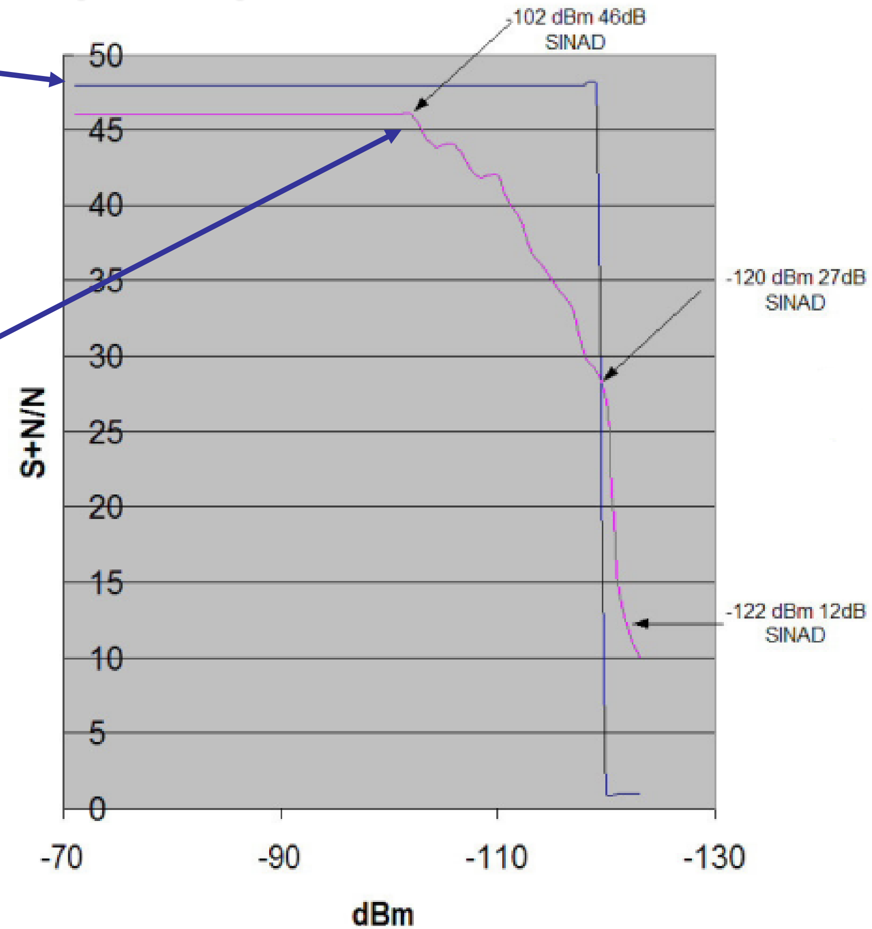
But is digital voice a better performer than analog FM?



What's All This Digital Voice Stuff?

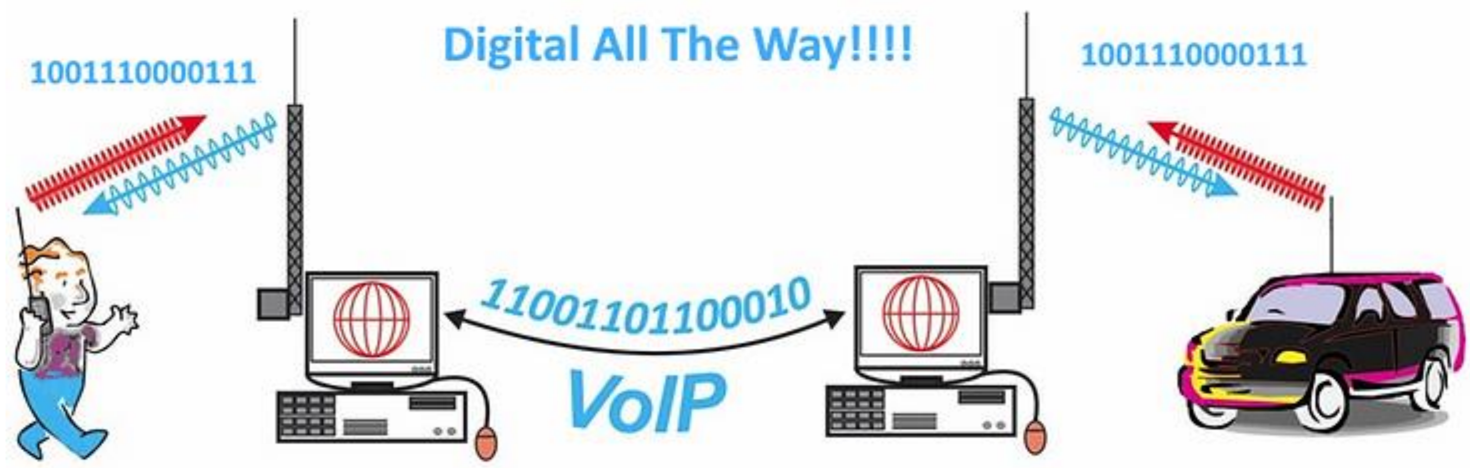
Because of FEC and digital modulation, the various digital voice techniques perform at a slightly better level and then *fall off* precipitously.

Analog FM performs poorer and degrades slowly.



What's All This Digital Voice Stuff?

Digital voice implementation and applications, along with APRS, WSJT, AMSAT and AREDN mesh networks, demonstrate to the FCC that Amateur Radio remains technically vibrant and ensures our continued relevance.



What's All This Digital Voice Stuff?

Dennis Silage K3DS