BASIC CONCEPTS OF DIGITAL VOICE IN AMATEUR RADIO

Specific to D-Star, TDMA DMR, NXDN, and P25 Phase 1 digital

Orange County Communications Auxiliary

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DIGITAL VOICE USAGE IN CENTRAL FLORIDA

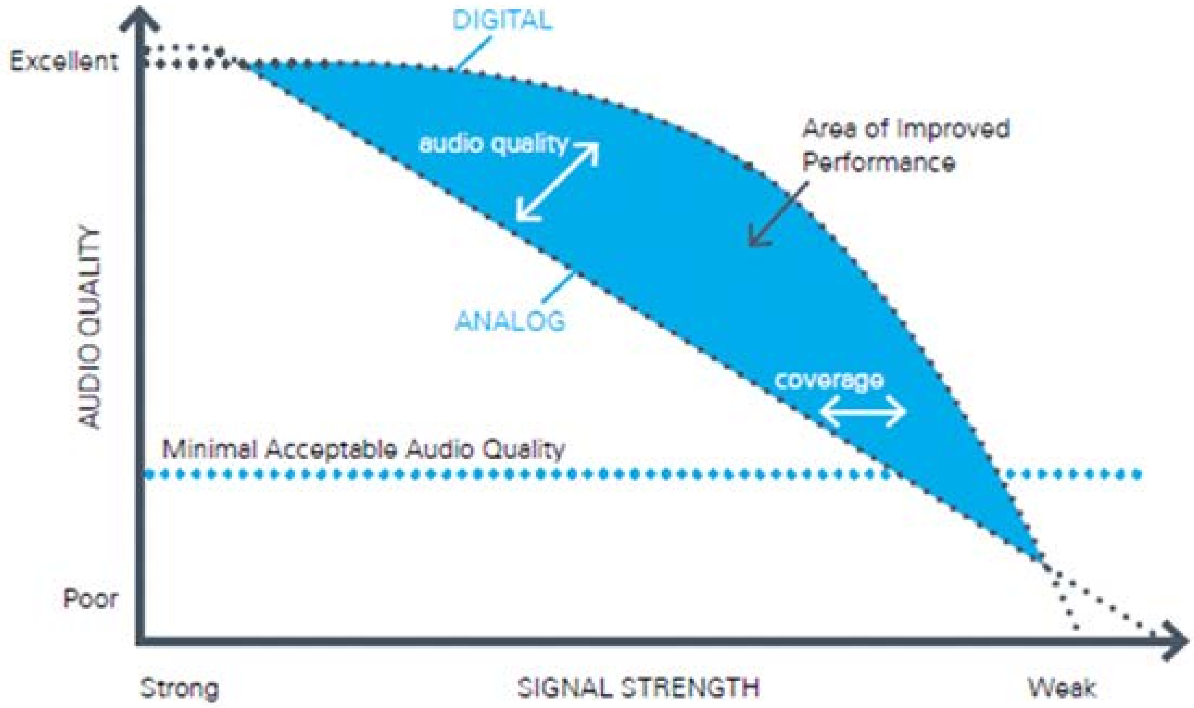
- The Central Part of the Florida Peninsula is one of a few areas of the U.S. where four of the most common Digital Voice technologies used in Amateur Radio are in use and can be tested. It is not so much of a battle, as it is a test bed and development environment.
- Of course, D-Star has been prevalent here since the early implementations of that technology.   We also have TDMA DMR, NXDN, and P25 Digital technologies currently in use in our little corner of the nation.

WHY USE DIGITAL VOICE?

- There are some basic reasons to use digital signals instead of analog, noise being the number one. Since analog signals can assume any value, noise is interpreted as being part of the original signal. For example, when you listened to an old LP record, you heard noise because the needle was an analog device and it did not know the difference between the music originally recorded and the noise inserted by dust or cracks.
- Digital systems on the other hand, can only understand two numbers, zero and one. Anything different from this is discarded. That's why you won't hear any unwanted noise when listening to an audio CD, even if you played it thousands of times before.
- Another advantage of digital systems over analog is the data compression capability. Since the digital counterpart of an analog signal is just a bunch of numbers, these numbers can be compressed, just like you could compress a Word Processing file using WinZip to shrink down the file size for example. The compression can be done to save storage space or transmit bandwidth.

WHY USE DIGITAL VOICE?

- Another reason to use digital, is because the audio retains better quality than analog does, as signal strength decreases, until the farthest extent of signal range.



ANALOG TO DIGITAL CONVERSION

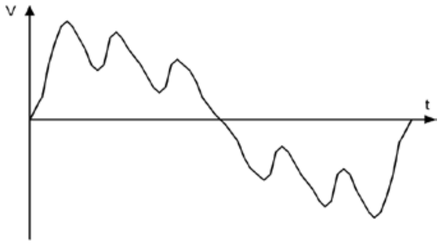
- Signals in the real world are analog: light, sound, you name it. So, real-world signals must be converted into digital, using a circuit called an ADC (Analog-to-Digital Converter), before they can be manipulated by digital equipment.
- When you scan a picture with a page scanner, what the scanner is doing is an analog-to-digital conversion: it is taking the analog information provided by the picture (light) and converting into digital.
- When you record your voice, or use a VoIP solution on your computer, you are using an analog-to-digital converter to convert your voice, which is analog, into digital information.

DIGITAL TO ANALOG CONVERSION

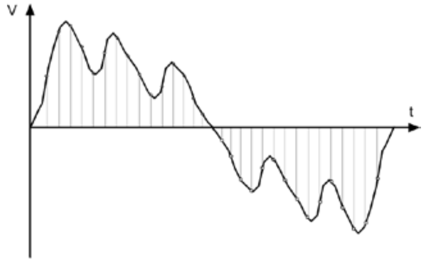
- Whenever we need the analog signal back, the opposite conversion – digital-to-analog, which is done by a circuit called a DAC, Digital-to-Analog Converter – is needed. When you play an audio CD, what the CD player is doing is reading digital information stored on the disc and converting it back to analog so you can hear the music. When you are talking on the phone, a digital-to-analog conversion is also taking place (at the central office switch, if you use an analog line, or at your home, if you use a digital line like ISDN or DSL), so you can hear what the other party is saying.
- As mentioned earlier, the real world audio signals that we humans can work with, need to be Analog.

ANALOG TO DIGITAL CONVERSION

- For our explanations, consider the analog signal found in Figure 1. Let's assume that it is an audio signal, since this the most popular applications for analog-to-digital and digital-to-analog conversions.



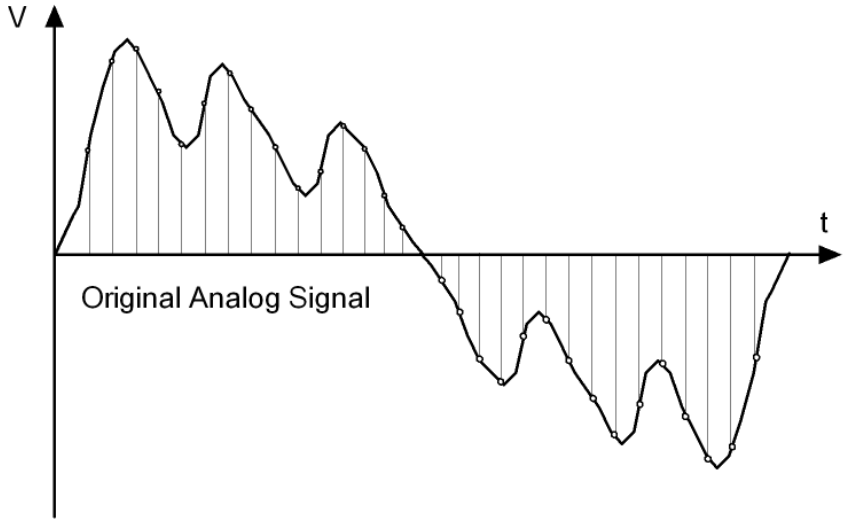
- What the ADC circuit does is to take samples from the analog signal from time to time. Each sample will be converted into a number, based on its voltage level. In Figure 2 you see an example of some sampling points on our analog signal.

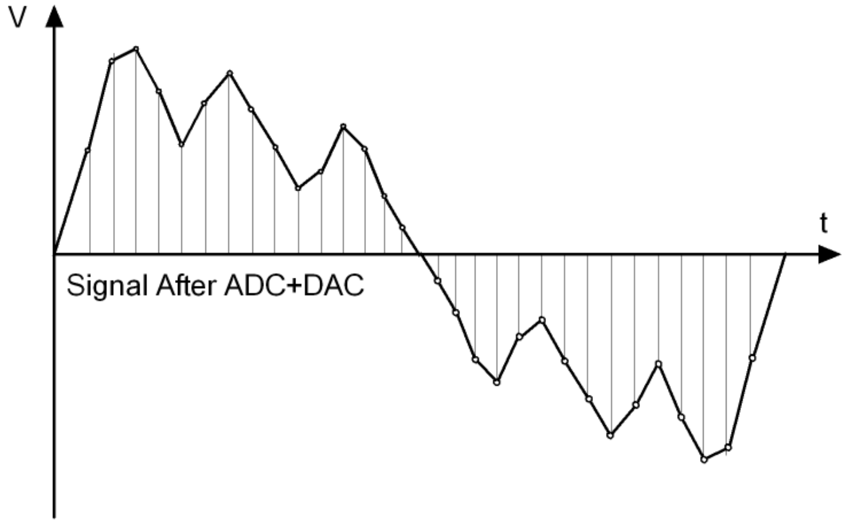


ANALOG TO DIGITAL CONVERSION

- The frequency on which the sampling will occur is called sampling rate. If a sampling rate of 22,050 Hz is used, for example, this means that in one second 22,050 points will be sampled. Thus, the distance of each sampling point will be of 1 / 22,050 second (45.35 µs, in this case). If a sampling rate of 44,100 Hz is used, it means that 44,100 points will be captured per second. In this case the distance of each point will be of 1 / 44,100 second or 22.675 µs. And so on.
- During the digital-to-analog conversion, the numbers will be converted again into voltages. If you think about it for a while, you will see that the waveform resulted from the digital-to-analog conversion won't be perfect, as it won't have all the points from the original analog signal, just some of them. In other words, the digital-to-analog converter will connect all the points captured by the analog-to-digital converter, any values that existed originally between these points will be suppressed.
- You can see an example in Figure 3, where we show how the signal would be after being converted to digital and back to analog. As you can see, the original waveform is more "rounded".

ANALOG TO DIGITAL CONVERSION





FORWARD ERROR CORRECTION

- Forward error correction (FEC) is a digital signal processing technique used to enhance data reliability. It does this by introducing redundant data, called error correcting code, prior to data transmission or storage. FEC provides the receiver with the ability to correct errors without a reverse channel to request the retransmission of data.
- FEC adds redundancy to transmitted information using a predetermined algorithm. The redundant bits are complex functions of the original information bits. Bits are sent multiple times, because an error may appear in any of the samples transmitted. FEC codes generally detect the last set of bits to determine the decoding of a small handful of bits.
- With FEC, each character is sent two or three times, and the receiver checks instances of each character. It is accepted only if conformity occurs in both instances. If conformity is satisfied for an instance, the character conforming to the protocol is accepted.

http://www.techopedia.com/definition/824/forward-error-correction-fec

WHAT IS A VOCODER?

- A voice Encoder / Decoder, or Vocoder, when used in VHF and UHF digital voice radio applications, usually incorporates the functions that convert and process the verbal information (data) back and forth between analog and digital as needed.
- The bulk of the digital processing needed for this type of communications;
  Sampling and analog and digital conversions, FEC, and additional Digital Signal Processing (DSP) is typically accomplished within the vocoder unit.

DVSI VOCODERS

- All of the radio systems addressed in this class use the MBE (Multi-Band Excitation) series vocoders from Digital Voice Systems, Inc. (DVSI). These vocoders are well suited to the low bit rate transmission limitations of most two way radio voice technologies, as well as mitigating the effects of unwanted background noises.
- The DVSI vocoders are proprietary and there is a license fee for their use. This cost is generally a small fraction of the overall cost of the radio system.
- There are other Vocoders available and some are not proprietary and even considered open source.

DVSI VOCODERS

- IMBE is used for P25 Phase 1 Digital Voice.
  Effective Sample rate is 8Khz.
  Data rate can vary between 2.4 Kbps and 9.6Kbps, depending on how it is implemented.
- AMBE-2020 is used for D-Star.
  Effective Sample rate is 8Khz.
  Data rate can vary between 2Kbps and 9.6Kbps, depending on how it is implemented.
- AMBE+2 is used for TDMA DMR technologies and also NXDN.
  Effective Sample rate is 8Khz.
  Data rate can vary between 2Kbps and 9.6Kbps, depending on how it is implemented.

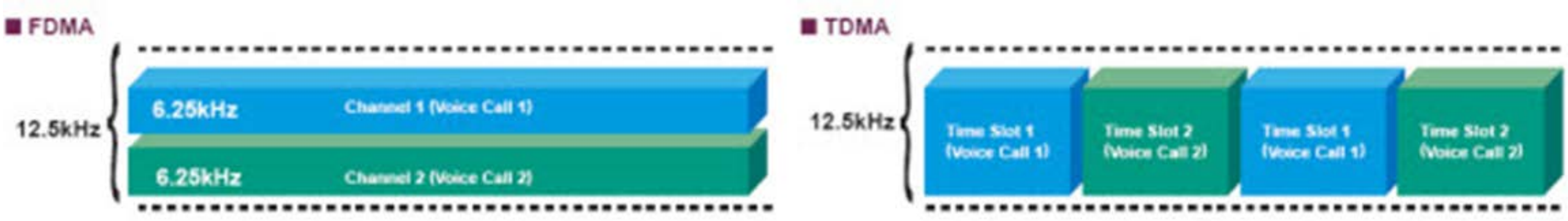
SPECTRAL EFFICIENCY

- Oh boy, here come the religious arguments…
- The technologies discussed here today use transmission techniques that are generally categorized as FDMA (Frequency Divided Multiple Access) and TDMA (Time Divided Multiple Access).
- D-Star, P25 Phase 1 digital, and NXDN are all considered FDMA technologies.
- TDMA /DMR technologies use TDMA transmission techniques.

SPECTRAL EFFICIENCY

- Without getting too technical, the basic difference between FDMA (Frequency Divided Multiple Access) and TDMA (Time Divided Multiple Access) is the definition of a channel and how it is used (accessed). In FDMA a particular bandwidth (E.g. 6.25kHz) at a particular frequency (E.g. 443.200MHz) is used to define a channel. Basically, the way channels have been allocated for decades. In TDMA, the same principle applies regarding bandwidth and frequency, but the signal is divided into time slots that allow the channel to have 'extra' capacity in the same bandwidth E.g. Two 6.25kHz 'equivalent' channels in a 12.5kHz channel. See the diagram on the next slide for a graphical explanation.

SPECTRAL EFFICIENCY



- Until recently, TDMA was more spectrum efficient at wider channel spacing's like 25kHz, for example two or three users could access the same bandwidth as one FDMA channel user. However in the case of the narrowband 6.25kHz FDMA technologies like D-Star and NXDN, these technologies and 2-slot 12.5kHz TDMA technology achieve the same result as far as spectrum efficiency is concerned.

MODULATION / SIGNALING TYPES

- Ready for more acronyms?
- D-Star uses Gaussian Minimum Shift Keying (GMSK).
- P25 Phase 1 Digital, NXDN, and the TDMA DMR technologies all use Continuous 4 Level FSK (C4FM).

MODULATION / SIGNALING TYPES

- GMSK
  *Continuous-phase frequency-shift keying modulation scheme
  *Digital data stream is first shaped with a Gaussian filter before being applied to a frequency modulator
  *Gaussian filter needed to reduce sideband power and out-of-band interference between signal carriers in adjacent frequency channels
  *Requires more complex channel equalization algorithms such as an adaptive equalizer at the receiver
  *High spectral efficiency, but needs a higher power level in order to reliably transmit the same amount of data.
  *Most notably used in the Global System for Mobile Communications (GSM)

MODULATION / SIGNALING TYPES

- C4FM
  *4-carrier modulation format
  *Carrier shifted frequency at a particular rate (time) to a particular location around a center frequency
  *Allows for each of the 4 "states" to represent a binary number.
  *Each state is a "DiBit" or "Symbol" which contains two bits of information.

MODULATION COMPARISONS

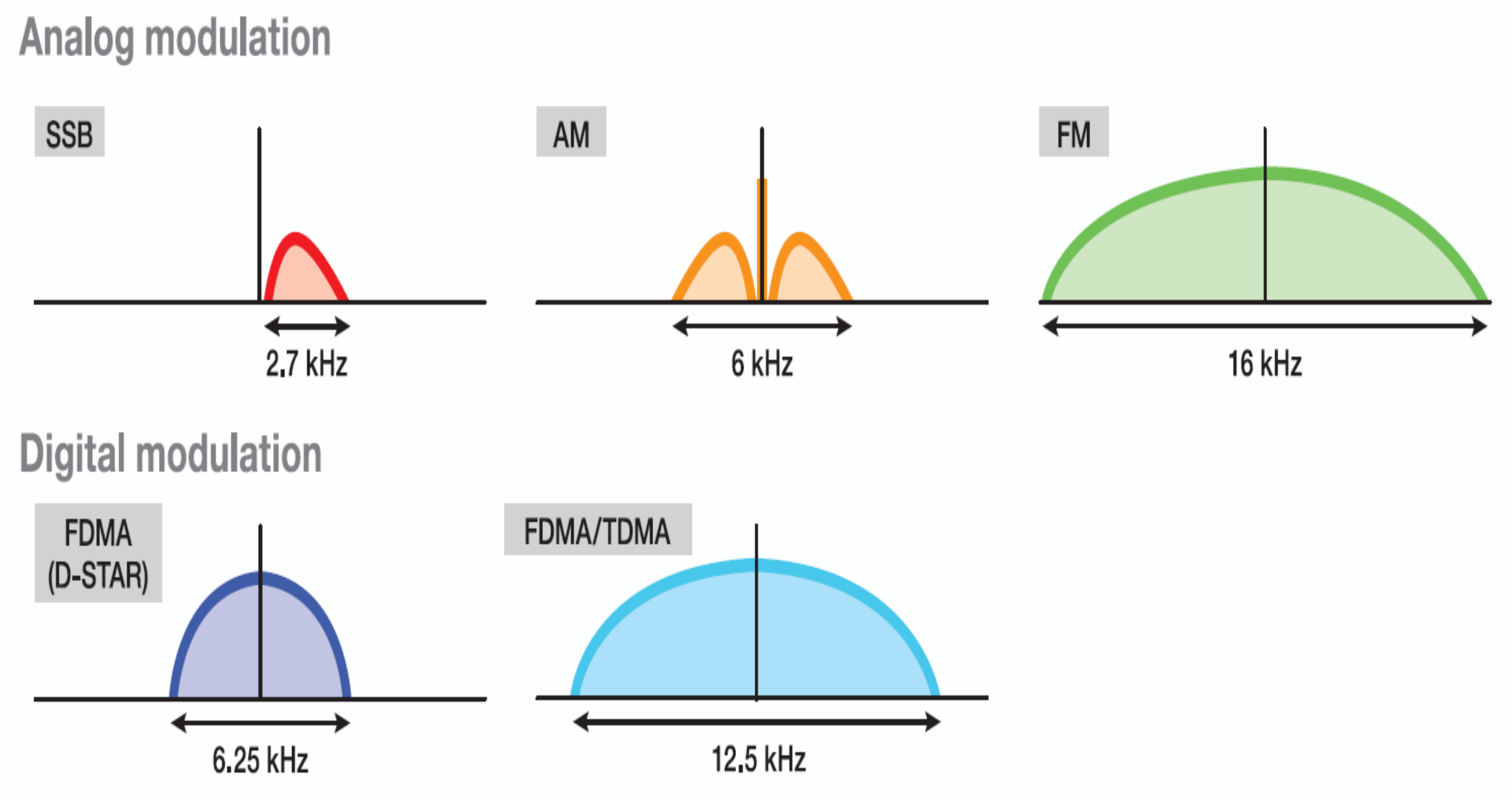
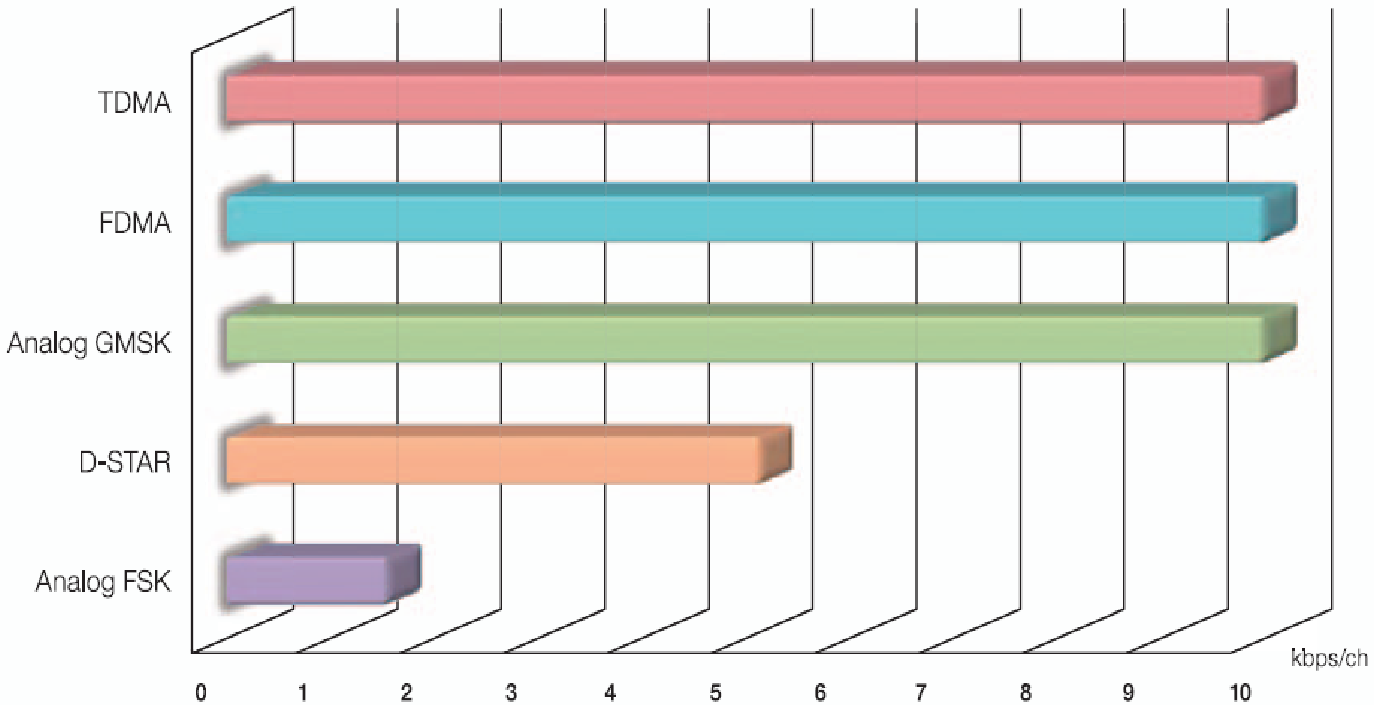


Fig. 8 Comparison of the occupied bandwidth

MODULATION COMPARISONS



TDMA	FDMA	Analog GMSK	D-STAR	Analog FSK
9.6 kbps/ch	9.6 kbps/ch	9.6 kbps/ch	4.8 kbps/ch	1.2 kbps/ch

Fig. 7 Comparison of the data transferring speed

FLORIDA DIGITAL AMATEUR RADIO

According to FRC website Florida has:

- 52 D-Star repeaters
- 25 P25 repeaters
- 16 DMR repeaters
- 6 NXDN repeaters
- http://florida-repeaters.org/dblist.htm

QUESTIONS…

Take a break and then we will have the specific presentations for D-Star, P25 Digital, TDMA DMR, and NXDN.