

“Surveillance Transmitter of the Future”

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Abstract

Officers and agents who put their life on the line every day in undercover situations wage the war on crime and terrorism. In many cases, these law enforcement professionals wear a covert audio transmitter (body wire or concealed transmitter) in order to provide backup team members with information on the situation.

The surveillance transmitters of the future will be based on the capabilities of a digital radio.

This will add more capability, security, and intelligence to the communications needs, compared to analog radio. Digital radio was developed for the purpose of data transmission using R-F; however it is not restricted to only data and has been adapted to voice through the use of audio compression or voice coding as it is more commonly referred to.

The Bodywire transmitter will have to operate seamlessly with the existing systems in order to prevent the need for a custom digital system. The modes of operation will mimic the Project 25 digital/analog modes. This will create flexibility for the surveillance operation and allow interoperability with the newer radio systems. This would allow the Bodywire to not only work with new digital systems but the older analog systems as well.

This paper will discuss the current project being funded through ONDCP/SPAWAR. It will also explain the features and advantages of this state of the art digital surveillance transmitter and the capabilities it will lend to surveillance operations.

The Roadmap to a Digital Surveillance System

The first question that must be asked is why Digital Radio or a Digital Bodywire? Digital Radio is already established in the communications market, as the standard for future communications needs. The new communications method of digital radio has several advantages over Analog Radio.

1. The use of data communications over the voice communications link.
2. Efficient use of the spectrum, mandated by NTIA, FCC, and other Federal Branches of Government.

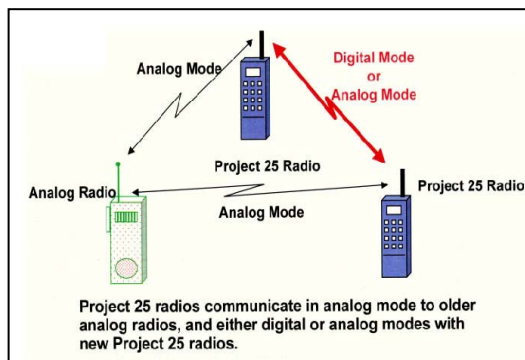
3. Error detection and correction capability increasing the reliability of the system.
4. Networking with an in place system such as Project 25
5. Increased secure communications.
6. Implementations of digital encryption for highly secure transmissions.
7. Analog-based components are eventually going away.

Digital Radio inherently will add more capability to the communications needs of the surveillance groups as compared to the existing analog radio systems.

Through the efforts of Project 25, and APCO, the makeover from analog to

digital communications networks is underway.

State and Local Law Enforcement Agencies have deployed these networks in large metropolitan cities. The Federal Government is currently using similar systems to update their capabilities. The equipment deployed currently has four modes of operation analog (wide) 25 KHZ, analog (narrow) 12.5 KHZ, Digital, and



Digital encrypted.

Older surveillance transmitters will be able to operate with Project 25 receivers tuned to wideband analog. Under the NTIA mandate to go to narrowband, newer analog surveillance transmitters will operate with Project 25 systems tuned to narrowband analog. The obvious gap is that there are no currently available surveillance transmitters, capable of inter-operating in either of the digital modes. The issue of audio clarity, not voice, and relative performance is of primary concern to most agencies involved in surveillance. Will the digital audio work as well as the analog equivalent that we have used for many years? How will body coupled antennas effect the digitally transmitted waveform? Will the digital system give us the same range performance?

Sufficient thought and analysis must be given to the overall transmitter design and architecture. An analytical approach is needed in order to create flexibility, a design

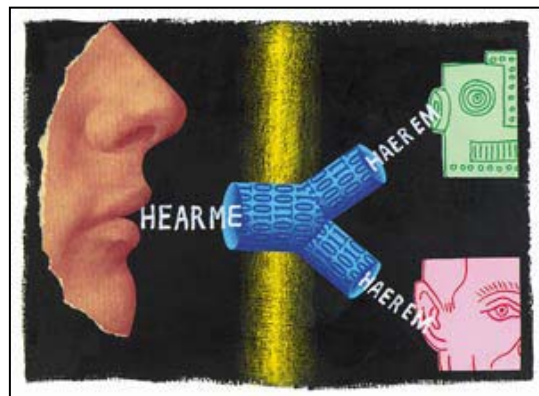
that meets surveillance needs, and to preclude immediate obsolescence of the investment.

The key to an effective digital transmission system, used for surveillance, will be in the analog to digital conversion techniques used to bandwidth limit the transmission. This bandwidth limiting is necessary in order to transfer the audio or voice signal, which is wideband, through the narrower digital communications link. Through this process the audio will be changed and at the same time the audio quality must be retained.

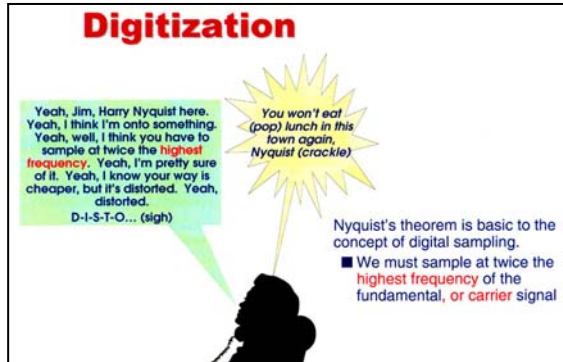
The modes of operation will mimic the project 25 systems in order to create flexibility for the surveillance operation. This would allow the Bodywire to not only work with new digital systems but the older analog systems as well.

Understanding Analog to Digital Conversion of Audio

The change to a digital system requires some use of compression or coding of the analog or voice signal, prior to digitization. In the world of audio communications everything starts with a device called a microphone, which is inherently analog in nature. This microphone, a wideband analog device, must be turned into a digital signal through analog to digital conversion.



The typical communications radio system is primarily concerned with transmitting voice, 300Hz to 3,000Hz, through the communications network. The surveillance system is also concerned with similar frequency bands with the exception of some audio response above 3,000Hz for purposes of intelligibility.



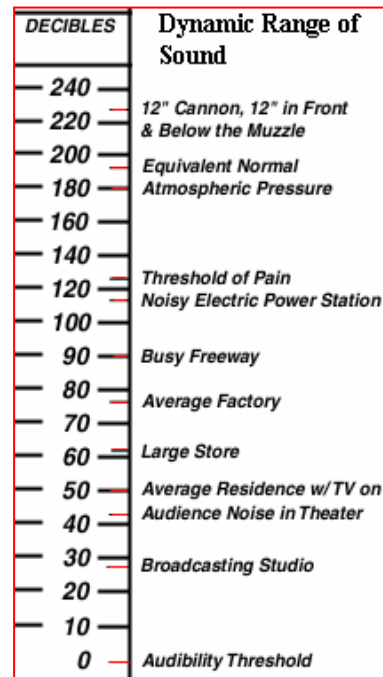
When we digitize the audio we want it, after converting it back to analog, to reflect the original sound. To do this we must sample the waveform often enough to capture all of the important changes and fluctuations of the audio signal. If we sample less often than is necessary, we will get a corrupted and incorrect representation of the audio signal.

The minimum sampling rate, or lowest sampling frequency, for any waveform is, according to Nyquist and Shannon, equal to twice the bandwidth of the signal. For our communications audio signal, with a bandwidth of 4 kHz maximum, we see that we should sample a rate which is at least 8 kHz, or 8000 samples a second. If we also assume that the resolution is 8 bits, the overall data rate is 64kbits per second. This data rate is well beyond the bandwidth currently available for standard audio communications. This means for digital communications the data rate needs to be reduced by factors of more than 10 using audio compression techniques.

The surveillance system will have to recognize other sounds such as doors closing, packages being handled, and various other background noises not just voice. Most voice coding techniques will create an audio signal sufficient for voice communications; however it may be marginal for surveillance applications due to its method of converting the audio to a digital word. The coding in vocoders is optimized to recognize and reproduce voice not other sounds. This is primarily how the signal processing takes a data stream that would otherwise be about 64kbits and reduces it to a 4 kBits per second data rate.

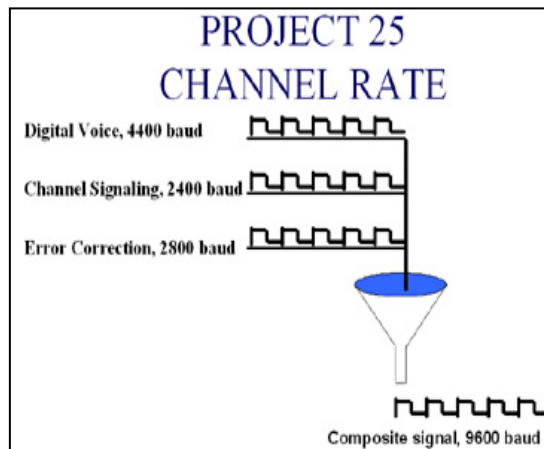
"The primary purpose of this voice coding and compression of analog signals is to reduce the analog to digital bandwidth for transmission over a digital network".

The dynamic range and the quality of the transmitted audio are a function of the frequency response and the ultimate signal to noise ratio of the system.



In a digital system the audio is converted from an analog signal to a digital word, error correction is added, and the complete digital packet modulates the transmitter. The quality of the audio is a function of the audio processing from the analog world to the digital format.

In terms of the communication system the digital system employs methods of error correction to improve the quality of the transmission and to reduce the effects of multipath. This overhead required for error correction and digital redundancy adds to the amount of data that must be transmitted in a given amount of time. The typical data rate increase, due to error coding, is a factor of two. The error correction impact in essence doubles the total data transmission rate from the compressed audio rate hence the expression of $\frac{1}{2}$ rate coding. In the Project 25 system the audio rate of 4.8kBits per second is doubled to a 9.6kBit per second rate of data transmission.



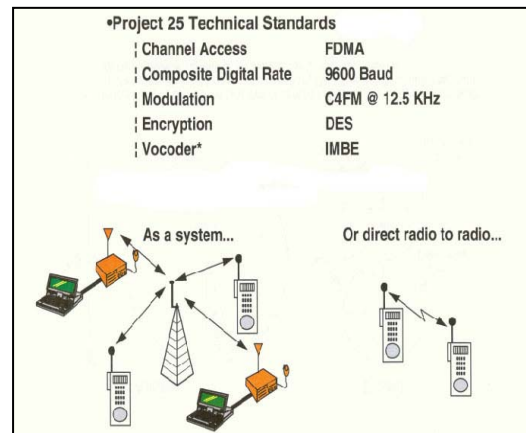
All of this processing puts continual pressure on reducing the data rate of the digitized audio. It is important to remember that one of the reasons to go digital is to improve the use of the transmitted spectrum. This only means one thing; limit the

bandwidth of the transmission. An increase in the transmitted data rate will only increase the required bandwidth needed for transmission.

The available bandwidth issue ultimately gets us to the pivotal question: How much information bandwidth do I have to transmit my audio signal? All of the digital systems are different in their use of precious frequency spectrum and the way in which they handle the audio conversion varies. Most vocoders operate in the 3 to 5kBit per second area for audio compression with reasonable audio performance

This gets us to the next most important question. Is there an open and published standard that everyone can use?

There is one open standard, using digital techniques, that has been developed for Law Enforcement Communications that has been successful, "Project 25". This standard has allowed the development of different types of equipment by many manufacturers.



The Project 25 standard is based on being both digital and backward analog compatible with existing analog equipment. This standard provides for devices that can be used with both analog and digital systems.

P25 Vocoder

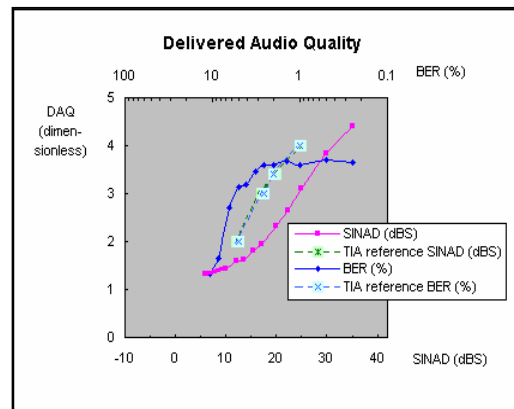
In mobile communications system, the design and subjective test of speech coders has been extremely difficult. Without low data rate speech coding, digital modulation schemes offer very little in the way of spectral efficiency for voice traffic. To make speech coding practical implementations must consume little power and provide tolerable, if not excellent speech quality. The goal of all speech-coding systems is to transmit speech with the highest possible quality using the least possible bandwidth.

Vocoders achieve a very efficiency in transmission bit rate and are in general more complex. They are based on using prior knowledge about the signal to be coded, and for this reason they are in general signal specific. Vocoders are a class of speech coding systems that analyze the voice signal at the transmitter, transmit parameters derived from the analysis, and then synthesize the voice at the receiver using those parameters. All Vocoder systems attempt to model the speech generation process as a dynamic system and try to quantify certain physical constraints of the buzzes and hisses.

The most prevalent Vocoder, of recent times, is the Multiband Excitation coder. A form of this Vocoder called the IMBE, Improved Multiband Excitation Vocoder. This is the vocoder used on "Project 25". The improved and Multiband Excitation Coders are based on the MBE analysis-synthesis model.

Multiband Excitation coding is a frequency domain coder that better models the excitation through various innovations. Many speech segments are not purely voiced or unvoiced. A single voiced/unvoiced decision is not completely accurate in those cases. The MBE excitation is mixed,

allowing both harmonic and random components in a single frame of speech. The manner in which the MBE Vocoder represents the vocal tract frequency information can be thought of as a channel vocoder that has all channels centered at the harmonics of the input signal. The MBE model allows a separate voiced/unvoiced decision for each frequency band in each frame of the input signal. This allows a more true representation of the input signal over the voiced/unvoiced decision based vocoders.

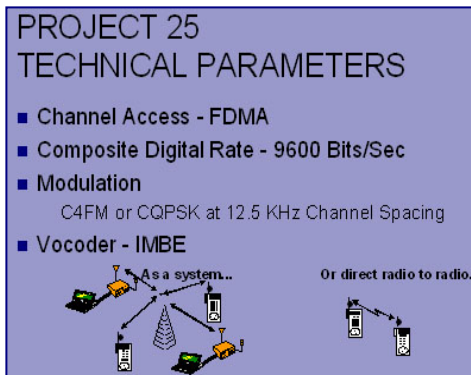


Design Approach to a Digital Bodywire Compliant with Project 25

The SPAWAR/ONDCP transmitter project involved the following major design efforts;

System Design and Definition

During this phase of the project DTC reviewed in detail the Project 25 published standards. The review of the standard allowed for a block diagram level system design. The P25 standards are focused on a complete Project 25 implementation of a transceiver capable of operating in a trunked system.



The bodywire will be a simplex design involving just the transmit side of the transceiver. It was important during this phase to determine any unnecessary signaling and features that could be eliminated in the bodywire design. This will allow for reduced size and lower power consumption.

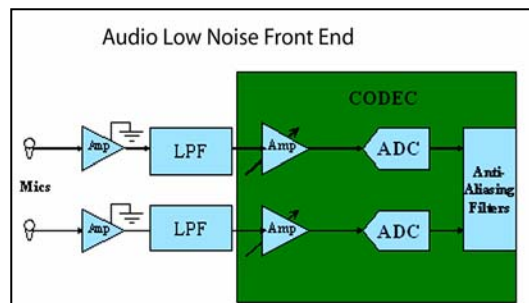
It was also necessary to interface with DVSI in order to determine the DSP requirements of the vocoder software.

The vocoder is intellectual property used in all Project 25 compliant devices and requires a licensing agreement between DTC and DVSI. This agreement gave DTC access to and the rights to use the IMBE vocoder software in the bodywire design. The integration of the vocoder with other signaling is what ultimately determined the size and power of the DSP used in the design.

The system design resulted in an overall power budget along with a preliminary specification. The specification included overall size targets, power output, and general operational requirements.

The vocoder has been tested for audio quality in a typical communications environment. This is an open task that will be undertaken in this project. Surveillance is a much different environment than close talking communications.

A considerable amount of effort has been spent optimizing the audio performance, analog front end, of the design.



This typically starts with the front-end signaling, low noise audio design, and dynamic range scaling.

Schematic Design

This phase of the project took the block diagram system design and built upon it to add the detail necessary to create a schematic representation of the bodywire. The process started with designing and analyzing the audio input requirements. This step also included the evaluation of audio processing necessary to optimize the design for surveillance applications.

There will be two boards in the design. One board will be for all signal processing and the other board will be used for R-F signal generation.

The signal processing board will use a new low current DSP. The software design will have multiple levels of audio processing necessary to enhance the operation of the audio to vocoder signal conversion.

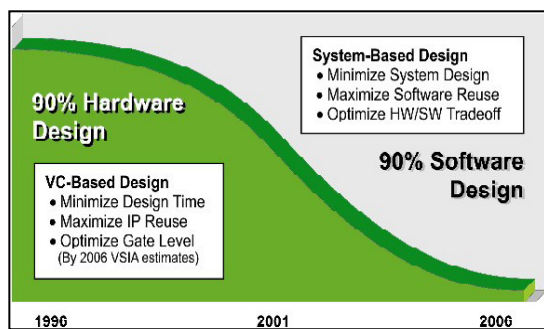
To date the prototype DSP board is up and running. The R-F board is in layout and will be complete by the end of June 2003.

We will initially interface the DSP board to an existing transmitter to continue system and R-F testing. The prototype will be delivered for testing ahead of schedule.

There will be additional audio performance and qualification testing done to evaluate the audio processing that was added to improve the vocoder performance. The audio improvements were an integral part of the design. The processing will be evaluated with surveillance type situations. The focus will be to optimize the audio performance of the vocoder.

Summary of the Design

The designs of the future will be based on software reconfigurable platforms. With the complexity of digital transmission the proliferation of hardware-based designs will become outdated.



The software-based radio is now a reality and it will be applied to surveillance devices.



The capabilities of this new digital transmitter design go far beyond a simple

transmitter. Subsequent development activities will add increased functionality and enhanced operational capabilities to **"the surveillance transmitter of the future", which is here today.**

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[2]Anna Watson, Measuring Perceived Quality of Speech, ACM Multimedia Electronic Proceedings, 98

[3]DVSI, Voice Coding Overview, 1999

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[6]Theodore S. Rappaport, Wireless Communications Principles and Practice, Prentice hall 96