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Underwater Acoustic Data Communications
for
Autonomous Platform
Command, Control and Communications

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1. Introduction

The purpose of this study is to provide an analysis and assessment of the state-of-the-art of underwater acoustic (UWA) data communications technology with application to multiple vehicle operation in support of shallow-water mine countermeasures (MCM) operations. This document is intended to provide system architects with a reference to what the current state of acoustic communications technology can provide and to support the execution of system architecture trades. Included are discussions and analyses that may be used to predict the performance, strengths, and limitations of different approaches to acoustic communications in facilitating the overall MCM system approach.

UWA communications is one technology that provides autonomous undersea vehicles (AUVs) with the potential to effectively perform shallow-water missions that would not otherwise be possible with present MCM assets [Bovio99]. Conversely, the bandwidth limitation imposed by the environment and existing transducers, in turn, limits the MCM concepts of operations when multiple AUV assets are considered.

Not only does the underwater environment constrain the data rates achievable but it also limits the maximum ranges achievable. Acoustic attenuation in the ocean is frequency-dependent with higher frequencies propagating shorter distances. The requirement to provide connectivity to submerged Naval assets over very long ranges is unlikely to be met directly with a single acoustic link due to the complex effects of acoustic propagation and limited propagation ranges. One solution is to create an UWA network of communications nodes. Another solution to this requirement is to use a buoy to convert acoustic signals to and from radio frequency (RF) signals. This solution provides real-time two-way communications to distant surface ships, aircraft and/or satellites that can act as relays to integrate undersea communications into the RF-based communications network.

The next section provides some insight into some of the advantages derived from using acoustic communications with AUVs. The third section addresses the complexity of the UWA environment and the resulting effects on the communications signal. Section 4 presents a detailed analysis of communications technologies for the UWA environment including standard and novel modulation/demodulation approaches, multi-user communications, and non-overt signaling. This section also describes selected experimental results. Figures summarizing achieved data rates, ranges, and bandwidths as functions of one another are presented and discussed in Section 5. This section also contains a survey of UWA modem implementations.

Section 6 presents a list of measures-of-effectiveness and an approach for using these measures for system comparison. The seventh section contains an outlook toward the future of UWA data communications. It contains subsections on an acoustic/RF communications node concept, on UWA communications design rules-of-thumb for designers of an overall MCM concept, and on technologies that are likely to provide breakthroughs in UWA communication. The acoustic communications requirements for two example MCM system designs are detailed in Section 8. Acoustic communications parameter trade-offs and system design for application to the AUV MCM mission are presented in Section 9.

The final two sections contain a listing of the acronyms and a bibliography of the material used in the preparation of this study, respectively.

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2. Acoustic Communications as a Discriminator

The conclusion that multiple vehicles must be deployed in order to accomplish thorough, timely, cost-effective underwater mine reconnaissance using autonomous undersea vehicles is quite widespread. If the vehicles are not to be recovered, a means of communicating mission results back to user must be provided. It is generally acknowledged that tethered operation (by fiber or copper) is not a workable solution for a number of reasons. This leaves radio and acoustic communications.

Beyond the communication of mission product data to the user, acoustic communications may be employed alone or in conjunction with radio communications to enhance the mission effectiveness of a multi-AUV based mine reconnaissance system.

Acoustic data interconnection enhances the robustness of a multi-platform system. Acoustic communications can provide the command and control link from the operator to the vehicle(s). Using such a link, an operator can modify vehicle tasking to suit changing conditions. As the vehicle does not need to break task to receive acoustic messages, only acoustic communications can be used asynchronously to modify vehicle tasking.

If vehicle status is transmitted, the system can respond to changing conditions, such as the loss of a vehicle or the failure of its sensor(s). Depending on the sophistication of the vehicles, this retasking could take place without operator intervention.

Depending on the operating area and the technical sophistication of the adversary, acoustic communications can be considered completely clandestine. This trait provides flexibility to the system architect because acoustic communications technology spans a wide range of capabilities and costs.

Ultimately, the details of what UWA communications can add in terms of enhanced overall MCM system performance and/or in terms of providing new MCM system capabilities can best be measured by how the MCM system-level measures-of-effectiveness are improved. Within the framework of the MCM system itself, the different signaling technologies can be compared and the communications system design trades made.

3. Environment Overview

It has been proposed that MCM capabilities can be significantly improved by incorporating timely measures of the shallow-water environment into the MCM operation [Tubridy98]. Appropriately instrumented AUVs can provide in-situ measures of parameters such water depth, currents, water column properties, sound speed, and propagation conditions back to the MCM commander via a system based, at least in part, on UWA communications. The environment not only affects the performance of the MCM sensors but also affects UWA communications performance.

From the perspective of communications, the characteristics of the UWA channel drive the design of any system. The UWA environment presents a bandlimited communications channel that causes temporal and frequency dispersion of the signal [Quazi82, Baggeroer84, Catipovic90c]. The Doppler spreading is caused by relative motion of the platform, surface, and the ocean volume. The multipath is most-often caused by reflections of the acoustic energy off the surface and bottom. The environment also imposes absorption and geometric spreading losses as a function of frequency and high levels of ambient noise especially in shallow water.

3.1. Channel Variability

Any environment has unique deterministic properties that are defined by the bathymetry and the geo-acoustic parameters of the bottom, and the sound speed profile. Long-term variability of these properties observed at a given location can result from seasonal or large-scale oceanographic phenomena.

The non-stationary ocean environment causes the stochastic variability of acoustic signals. Very short-term variability (VSTV) in the fine- and micro-structure can cause phase and amplitude fluctuations within the duration of a signal. Short-term variability (STV) in the internal wave pattern perturbs the local sound speed profile and fluctuates the coherence of the acoustic field between signals received at the same location. VSTV affects signal-related statistics between receivers and the STV serves to create different propagation paths.

Variability can also be introduced by the tactics/dynamics of the transmitters and receivers. Changes in geometry over time caused by forward motion, depth excursions, etc. result in variations in the acoustic propagation paths between signals and potentially within the signal duration. For example, the delay between a direct path and a bottom-bounced arrival in an isovelocity channel is given by

$$\Delta_t = \frac{x}{c}\left[\frac{1}{\cos\alpha} - 1\right]$$

where $\alpha = \tan^{-1}(2y/x)$, c is the speed of sound, x is the distance between the source and receiver, and y is the source and receiver elevation over the bottom. For small y/x ratios, the interpath time delay varies as the square of the elevation.

3.2. Multipath

The UWA channel, particularly in shallow water, is also characterized by rapidly fluctuating multipath [Catipovic90c]. Sound is refracted by inhomogeneities in the water and reflected by boundaries. The inhomogeneities and surface are constantly moving. Only the boundary represented by the ocean bottom is considered to be fixed, especially if the source and receiver are moving. The net result of this motion is that sound transits from transmitter to receiver over a multiplicity of paths, each path changing over time. Depending on the exact transit time for a path, it may combine constructively or destructively at the receiver with another path or paths. This can result in deep, rapidly changing, and severely localized fading. In severe shallow-water environments, the time scale for these channel fluctuations can be on the order of tenths of seconds.

Another effect of multipath arrivals is that of intersymbol interference (ISI). ISI occurs in multipath channels energy from different transmitted symbols arrives at the receiver simultaneously. The act of recovering data from a transmitted signal is that of deciding which symbol was actually sent. This estimation is made difficult if multiple, overlapping, delayed versions of a sequence of symbols is received.

3.3. Conventional Methods for Dealing with Intersymbol Interference

ISI severely challenges any data communications system's ability to accurately recover the transmitted data. Three straightforward methods can be employed to counteract its effect [Baggeroer84]. These methods, however, limit data rate performance.

The first is to sustain each symbol for a time much greater than the duration of the ISI. Thus, the receiver can observe the signal during the interval when only a single symbol is present. The problem with this method is that since the symbol duration is artificially prolonged, it severely limits the achievable data rate [Widmer93].

The second method is to shorten the duration of the entire message to a time less than the difference between successive arrivals of the signal [Howe92, Edelson98]. This allows the receiver to capture the entire message prior to the onset of any ISI. The problem with this method is that the required signal bandwidth is proportional to the symbol rate. Shortening the symbols in order to pack more into the available time requires additional bandwidth that may not be available.

The third method of dealing with multipath is to use beamforming to receive only a single arrival. The problem with this method is that the available signal energy is distributed over all the multipath arrivals. To isolate and use only one is to throw away a good portion of the available signal resulting in degraded signal to noise at the receiver.

4. Communications Overview

Acoustic telemetry from underwater submersibles and sensors has been researched and tested since it was recognized that the ocean was able to support signal transmission. Because of the attenuation properties of the ocean, the transmission of information between a source and receiver underwater has been achieved most reliably by the use of sound. Sound is taken here to include frequencies both well above and well below the range of human hearing. Many of the advances in underwater acoustic (UWA) communications have come from the Navy needs for communications with submarines, and other government interests.

While an information bearing signal on a wire may be transduced directly into pressure fluctuations proportional to the instantaneous signal voltage, the characteristics of the underwater acoustic environment, as well as the equipment used to transmit and receive sound underwater, make this impractical. The underwater acoustic environment and equipment limitations are analogous to those encountered in using radio signals to convey information above the waterline. The limitations give rise to the same need to perform frequency translation in order to transmit information. The process of frequency translation is a reversible one where the spectrum of the information bearing signal is shifted to a frequency range more conducive to transmission.

As outlined in the previous section, due to absorption losses, low frequencies propagate farther underwater than high frequencies. So frequency translation may be employed to ensure adequate link range. Translation also may be employed to limit link range as a means of providing privacy.

However, acoustic transmitters operate effectively only when their dimensions are of the order of magnitude of the wavelength of the signal transmitted. Thus practical physical considerations place a lower limit on the frequency range employed. The practical limitation on bandwidth for acoustic transmitters is one to two octaves. For a given transmit power, information theory dictates that channel information capacity increases with bandwidth. To increase the link information rate, the operating frequencies must be increased.

Finally, frequency translation may be employed to avoid frequency bands already occupied by other signals such as noise or other natural and manmade signals. This may be done so as not to interfere with other users.

Frequency translation is accomplished by multiplying the information-bearing signal with an auxiliary sinusoidal signal known as a carrier signal. This *modulation* of a *carrier* signal may be performed in a number of different ways. Common forms of modulation include amplitude modulation, pulse position modulation, pulse code modulation, angle modulation, and hybrids of those methods.

4.1. Amplitude Modulated Signaling

In amplitude modulation (AM), the amplitude of the information signal is used to vary the transmitted level of the carrier signal. Through multiplication, the information-bearing signal $m(t)$ is impressed on the amplitude of the carrier signal. Thus, the signal that is transmitted is of the form

$$y(t) = m(t) \times A\cos(\omega_c t)$$

Here the signal $m(t)$ is the signal to be transmitted, and $y(t)$ is the modulated signal that is transmitted through the water. Demodulation consists of removing the carrier signal from the received waveform.

A form of AM useful for digital communications is amplitude shift keying (ASK) modulation or pulse amplitude modulation (PAM). Given digital data for transmission, a signal is generated by representing a '1' bit by the presence of a tone and a '0' bit by the absence of a tone. Alternatively, a '1' bit corresponds to the acoustic transducer being on (a tone is transmitted) and a '0' bit corresponds to the acoustic transducer being off (a tone is not transmitted). For this reason, ASK is also known as On-Off Keying (OOK).

4.1.1. Selected Experimental Results

The best known example of the use of amplitude modulation for underwater communication is probably the US Navy's AN/WQC-2 (Gertrude). The AN/WQC-2 system is a single sideband (SSB) analog voice transmission system that employs existing platform mid-frequency (MF) sonars for transmission and reception [Quazi82]. Three fundamental limitations result from its being an analog voice system.

First, analog systems tend not to be suited to exchanging large amounts of data, tactical or otherwise. Being an analog system also limits the amount of signal processing that may be employed to aid intelligibility at the receiver. In particular, multipath is not cancelled, so the signal becomes less intelligible as multipath levels increase. That the AN/WQC-2 is usable at all is due to the fact that there is a man in the loop who can understand speech in a crowded, noisy environment.

Second, SSB voice transmission makes very poor use of available transmitter power. The average amplitude, and hence power, of a voice signal is a small fraction of its peak amplitude. The transmitted power of an AN/WQC-2 signal is often as much as 30dB below the peak available sonar transmit power. This peak sonar transmit power can be achieved only for constant amplitude signals.

Third, analog voice systems are hard to make secure. Voice *scramblers* are difficult to implement, particularly for the underwater acoustic environment. Digital methods of encryption are not feasible, as there is no obvious way to adapt this system for the transfer of digital data.

Figure 4.1 shows a block diagram of single-side-band amplitude modulation processing

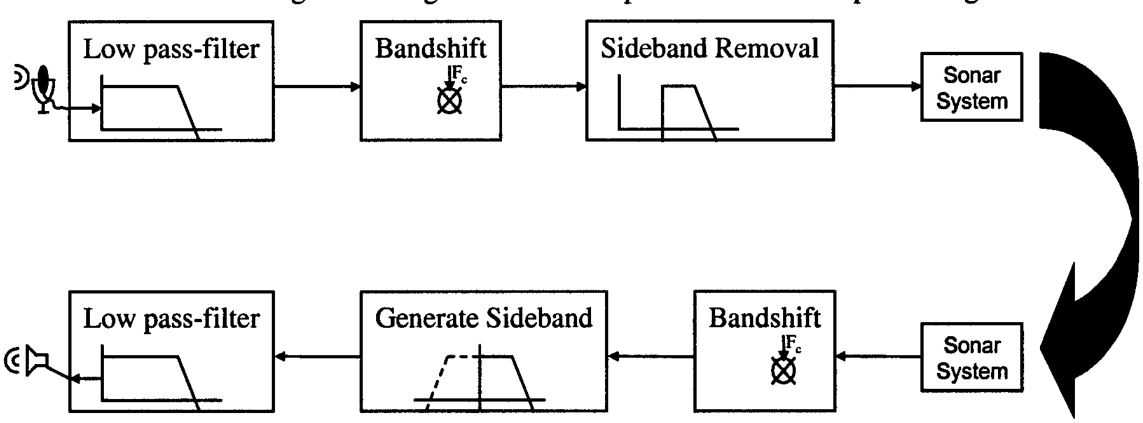


Figure 4.1 - Block Diagram of Single-Side-Band Amplitude Modulation

Other instances of AM systems for the underwater acoustic channel have been documented. ASK modulation was commonly used in the 1970s and early 1980s. ASK was used for clean paths and where reverberation was low. During a 1976 experiment in a lake by [Andrews76], a data rate of 600 bits per second (bps) was achieved. In the mid 1980s, some work was done with video telemetry in a vertical channel via amplitude modulation [Collins83, Galloway85]. In this work, a narrow (1°) transmit beam with a center frequency of 215kHz and bandwidth of 12kHz was successfully used to relay video signals through a 300m vertical column of water.

4.2. *Pulse Period Modulation*

In pulse position modulation (PPM), the temporal positions of transmitted pulses are varied in accordance with some characteristic of the information signal. In analog PPM, a short transmitted pulse is delayed with respect to the sampling start by an amount that is linearly proportional to the amplitude of the information signal. Digital PPM (DPPM) exhibits greater bandwidth efficiency than its analog counterpart and is less complex to implement. However, DPPM is more sensitive than analog PPM to synchronization errors.

4.2.1. Selected Experimental Results

An early UWA communication system that employs the use of time delays to transmit information is the Navy's AN/WQC-6 (ProbeAlert). The AN/WQC-6 system is, in essence, an acoustic Morse code signaling scheme using continuous wave (CW) transmission. This system allows the transfer of small amounts of digital information by transmitting a known waveform with various time delays. The received signal is correlated against delayed versions of itself, so that the delay between transmissions can be measured. These delays are then converted to digital information.

This method overcomes the average versus peak power problem of the AM-based AN/WQC-2. Given that it is a data transmission, it can easily be made secure. The AN/WQC-6 does not, however, have very high data rates (the actual rate is classified). The system is generally used with a set of preformatted messages in order to confine the transmitted data to as few bits as possible. This method of employment severely restricts the flexibility of the system. For transfer of sensor data as would be done with an Unmanned Underwater Vehicle (UUV) data link, its data rate is completely inadequate.

[Smith93] reports that Sonardyne developed a PPM operating in a frequency range of 8 to 50kHz and a communications range of up to 5km. In addition, [Woodward96] reports on a DPPM system designed to establish voice communications between a pair of divers or divers and a surface ship. This system involves transmitting 3 bit frames of 1.25ms each. Within each frame are 8 time slots of 125μs each, plus a 250μs guard band. The three bits are encoded by transmitting a signal during only one of the 8 time slots in each frame. The receiver attempted to decode the data by determining which time slot had the most received energy. Synchronization pulses are occasionally added between frames. There have been no reports of testing this system outside of a laboratory. The system seems unlikely to work reliably except under benign conditions because the simple receiver design does not adequately address the problems associated with multipath.

An enhanced version of PPM, referred to as sequence position modulation (SPM), is introduced for acoustic communications in [Sanchez99] and discussed in [Preisig00]. SPM uses signals phase modulated by maximum-length sequences, or m-sequences, that have wide bandwidth as well as good auto-correlation and cross-correlation properties. The data is encoded in the relative delay between the transmission of different sequences. For example, two bits of information can be communicated via a transmitter that sends two sequences with any of four delays between them. If the receiver can resolve the arrival time of a sequence to within one half of the smallest of the possible delays, the signal can be successfully demodulated. The system was tested in a time-varying surf-zone environment using 6 m-sequences. Performance depends on the ability to resolve arrival times and the stability of the arrival time over the duration of the sequence transmissions.

4.3. *Pulse Code Modulation*

Pulse code modulation (PCM), differential PCM (DPCM), and adaptive DPCM (ADPCM) are techniques for converting analog waveforms to digital signals, and are frequently used for speech. When performing speech conversion, PCM typically results in a data rate of 64kbps, while ADPCM typically results in 16, 24, 32, or 40bps (according to the standards of the International Telecommunications Union standards).

It would be possible to transmit digital data by converting it to an analog waveform by one of these techniques, then transmitting the resulting analog waveform, and converting back to digital on the receiving side. There would typically be referred to as amplitude modulation, which is essentially an analog transmission technique.

PCM refers to representing analog waveforms in discrete amplitude steps, typically with a fixed sample rate, as would be generated by an analog to digital converter. DPCM performs the same conversion, but instead of storing the full amplitude value for each conversion, it just stores the difference between successive samples, allowing the use of fewer bits for each sample. ADPCM is DPCM where the number of bits used to represent each sample is allowed to vary.

The digital data generated by converting analog waveforms to digital data streams via one of these techniques could be transmitted via any of the time-delay, non-coherent, or coherent communications techniques described in this report. However, in this context, PCM, DPCM, and ADPCM are not modulation techniques that can be used directly for acoustic communications.

4.4. Angle Modulated Signaling

In angle modulation, the amplitude of the information signal is used to vary the transmitted frequency (frequency modulation) or phase (phase modulation) of the carrier signal.

4.4.1. Frequency Modulation

Frequency modulation (FM) refers to altering the frequency of the modulated signal proportionally to the amplitude of the information signal. For radio frequency (RF) communications, FM is more useful than phase modulation partly for its ease of generation and decoding. Compared to AM (another common RF method), FM provides improved signal reception while requiring less radiated power. On the downside, FM requires more bandwidth (up to 20 times as much) and a more complicated receiver and transmitter than AM. For FM, the frequency of the carrier signal is changed in proportion to the information signal $m(t)$. A transmitted signal is of the form

$$y(t) = A\cos\left(\omega_c t + k\int m(t)dt\right).$$

Here the signal $m(t)$ is the signal to be transmitted, and $y(t)$ is the modulated signal that is transmitted through the water. Demodulation consists of computing the slope of the phase of the received signal via a differentiator and then applying an envelope detector to the output of the differentiator to recover the information signal.

Conceptually, an implementation of frequency modulation in the underwater environment might resemble the block diagram in Figure 4.2.

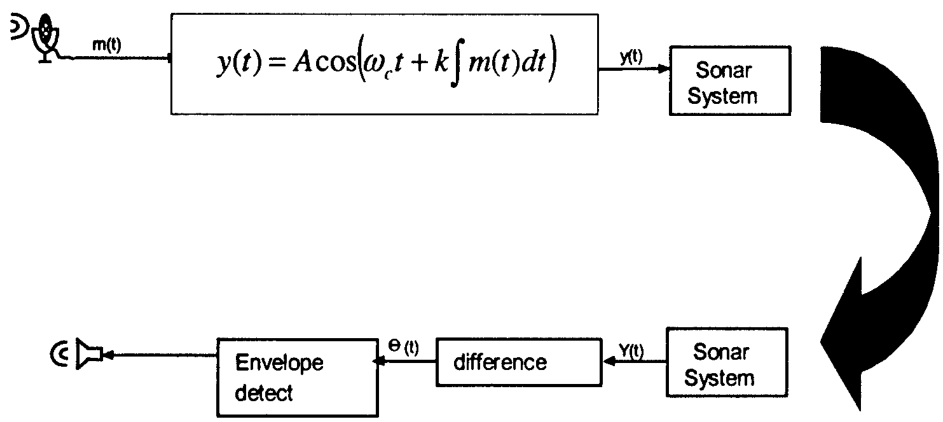


Figure 4.2 - Block Diagram of Frequency Modulated System

However, the use of FM for UWA analog communications does not appear in the literature.

4.4.2. Frequency Shift Keying

On the other hand, communications systems employing other forms of angle modulation have been designed and implemented for the underwater acoustic environment. One of the simpler methods of angle modulation for digital signals is frequency shift keying (FSK) modulation. In FSK, a '1' bit is represented by a tone at frequency f_1, while a '0' bit is represented by a tone at frequency f_2 [Catipovic97]. If two are used, two bits, b_1b_2, can be sent simultaneously, with f_1 used to represent bit b_1 and f_2 used to represent bit b_2 (tone present for a '1', tone absent for a '0'). This is the simplest form of multiple FSK (MFSK) modulation.

If 4, 8, 16, 32, … frequency bands are used, then a system can transmit in one band at a time, effectively relaying $\log_2(N)$ bits per symbol period (N is the number of bands). That is, if 8 bands exist, then a tone can be transmitted in one of the 8 bands at a time, indicating the transfer of a value between 1 and 8, and effectively transferring 3 bits of information. This is another form of MFSK modulation. Some systems employ a frequency hopping algorithm, which is used to select the next set of frequencies in which to transmit – in essence a time-varying set of frequency bands that are alternated to minimize ISI. Some systems reserve one frequency band for a Doppler pilot tone. In these cases, the designer wishes to space frequency bands so close together that they could become ambiguous in high Doppler situations, so a pilot tone is continuously broadcast in one frequency band, and is used to resolve the ambiguity.

4.4.3. Phase Shift Keying

Phase shift keying (PSK) is a digital phase modulation in which the phase of the carrier signal is discretely varied in relation to a reference phase in accordance with the data bits being transmitted. In PSK systems designed so that the carrier can assume only two different phase angles (binary PSK, or BPSK), each change of phase carries one bit of information, i.e., the bit rate equals the modulation rate. If the number of recognizable phase angles is increased to 4 (quadrature PSK, or QPSK), then 2 bits of information can be encoded into each signal element, or symbol; likewise, 8 phase angles can encode 3 bits in each symbol.

For an M-PSK system, with $M=2^N$ recognizable phase angles (i.e. N bits are encoded into each symbol), each transmitted symbol is of the form

$$y(t) = g(t)\cos(\omega_c t + \theta_m), \quad m \in \{1, \ldots, M\}, \quad 0 \le t \le T \,,$$

where $g(t)$ is the signal pulse shape, $\theta_m = (2\pi/M)(m-1)$ is the assigned phase of the carrier signal, and T is a time interval known as the symbol interval.

4.4.3.1. Differential Phase Shift Keying

Differential phase shift keying (DPSK) is a variant of PSK modulation in which the phase of the carrier signal is discretely varied in relation to the phase of the immediately preceding signal element. The same set of phase values is used to transmit the data bits, but these values are now associated with the differences between the phases in two adjacent symbols rather than the actual phases themselves. The advantage of DPSK over PSK is that the receiver does not require a perfect reference phase. However, differentially coherent detection suffers from a performance loss as compared to coherent detection.

4.4.4. Quadrature Amplitude Modulation

Quadrature amplitude modulation (QAM) improves upon amplitude modulation and phase modulation by combining the two. QAM uses two carrier signals out of phase by 90° and amplitude modulated by separate signals corresponding to two data sources. Each transmitted symbol is of the form

$$y(t) = A_{mc}\, g(t)\cos(\omega_c t) - A_{ms}\, g(t)\sin(\omega_c t),$$

where A_{mc} and A_{ms} are the information-bearing signal amplitudes of the quadrature carriers. In this formulation, the phase modulation aspect of QAM is not readily apparent. However, $y(t)$ can equivalently be written as

$$y(t) = V_m\, g(t)\cos(\omega_c t + \phi_m),$$

where $V_m = \sqrt{A_{mc}^2 + A_{ms}^2}$ and $\phi_m = \tan^{-1}(A_{ms}/A_{mc})$. This expression more clearly indicates the combination of phase and amplitude modulations involved in QAM.

4.5. Non-Coherent Communications Processing

The limitations of UWA communication systems is a combination of the effects of multipath propagation and temporal channel variability that cause intersymbol interference (ISI) and strong phase fluctuations of the signal [Catapovic90c, Stojanovic96a]. These limiting factors led to non-coherent detection and low signaling rates for system design in the 1980's. Non-coherent demodulation works well with lower signal-to-noise ratios (SNRs), uses less processing power, and usually costs lower to implement. Even though there are non-coherent methods like FSK modulation that are reliable, non-coherent systems make inefficient use of bandwidth underwater. Therefore, the major drawback with non-coherent modulation is that throughput is generally limited.

Non-coherent communications systems generally operate by transmitting a tone or set of tones for a short period of time (the tone or set of tones is called a symbol, and the period of time is a symbol period). The symbol period is generally selected to be at least as long as the expected channel multipath duration; if the symbol period is much longer than the multipath duration then decoding the symbols becomes easier.

When multipath is present, a tone that is transmitted for T seconds will be received for T+M seconds, where the additional M seconds accounts for the multipath. If T is much longer than M, then the effect is generally negligible, but if M is on the same order as T, then ISI is present, and the effect is time-smearing which can cause incorrect symbol decisions to be made. It is possible to reduce the effect of ISI through the use of an equalizer. Since the multipath conditions in the UWA channel tend not to be very stable, the filtering which makes up an equalizer is generally adaptive, and can become quite complicated. While equalizers have been used with non-coherent signaling, they are generally very simple or not present.

4.5.1. Reliability Issues

Non-coherent communications systems reliability depends on a few key issues:

- Sonar equation parameters
- Throughput versus multipath extent
- Frequency selective fading
- Temporal fading
- Spatial fading

Each of these topics is discussed here separately. The specifics of this section apply to non-coherent systems but the conceptual framework applies to coherent systems as well.

4.5.1.1. Sonar Equation Parameters

The non-coherent receiver must determine which frequencies have been transmitted. As received SNR decreases, the probability of a decision error increases. The exact received SNR required for reliable operation of a receiver depends on the details of the design, the acceptable false alarm rate, and the environment. Typical received SNRs are approximately 7 to 12dB.

The one-way sonar equation, which is appropriate for communications, is

$$SNR_{receive} = P_{transmit} - L - N + DI_{transmit} + DI_{receive} + G,$$

where $SNR_{receive}$ is the received SNR, $P_{transmit}$ is the transmit power level, L is the one-way propagation loss, N is the ambient noise level in the band over which the signal must be detected, $DI_{transmit}$ is the transmitters directivity gain, $DI_{receive}$ is the receiver's directivity gain, and G is any other processing gain that the system may achieve. This equation can be used to estimate the received SNR, required transmit power level, or maximum acceptable propagation loss, if all other parameters are known.

Propagation losses can be modeled as spherical spreading plus absorption or can be looked up in tables of experimental results. Details of the spherical spreading plus absorption model can be found in any underwater acoustics textbook, but it is important to remember that this model assumes a constant speed of sound, which is rarely observed in nature. In general, the speed of sound varies with depth, causing some depths to be well ensonified. A number of experimental measurements are described in the report "Range Performance Summary Plots, Volume I, World and Representative SSPs", (available from Naval Undersea Warfare Center Division in Newport (NUWCDIVNPT)). This volume plots propagation losses versus range and expresses the percentage of environments in which these losses may be observed. In general, the measured propagation losses exceed the spherical spreading plus absorption model by a considerable margin.

The ambient noise level can be estimated from Wentz curves, or from experimental results if they are available. The effect of bandwidth on the system noise level is:

$$N_{bw} = N_{1Hz} + 10\log_{10}(BW)$$

where N_{bw} is the noise in the system's detection bandwidth, N_{1Hz} is the Wentz curve value for noise in a 1Hz bandwidth, and BW is the system's detection bandwidth.

Table 4.A shows some examples of the sonar equation.

System Description	$SNR_{receive}$	$P_{transmit}$	L	N	$DI_{transmit}$	$DI_{receive}$	G
3500Hz center frequency 1 second symbol period, giving 1Hz detection band.	10	200 dB	137 (approx. 110 km in 50% of environments	53	0	0	0
As above, but with 10 msec symbol period; 100Hz detection bandwidth	10	200 dB	117 (approx. 69 km in 50% of environments	73	0	0	0
As above, but transmitter array has 10dB directivity gain.	10	200 dB	127 (approx. 91 km in 50% of environments	73	10	0	0

Table 4.A - Examples of Sonar Equation

4.5.1.2. Throughput vs. Multipath Extent

Non-coherent receivers are generally designed to operate in an environment where the multipath extent is significantly less than the symbol period. This can be summarized as

$$T >> T_{m,}$$

where T_m is the multipath extent and T is the symbol period. If the multipath extent begins to approach a symbol period, then an equalizer must be designed to cancel this multipath. The inclusion of an equalizer vastly increases the complexity of the non-coherent receiver, and negates much of its appeal. The method used for measuring the multipath extent T_m is controversial [Rice97, Yang98b]. In particular, in the case of sparse multipath, it is not clear whether T_m should include all of the multipath or just the main arrivals.

Consideration of the multipath extent can dominate the design of non-coherent systems since the extent in the UWA channel can vary widely (from practically none to over 1s). However, as the symbol period increases, the bandwidth required for each tone decreases, and the limiting factors become Doppler shift and Doppler spread. Doppler shift is a predictable and correctable time dilation or contraction (or frequency shift for narrow-band signals), caused by the movement of the source and/or the receiver. Doppler spread is a random phenomenon, caused by changes in the speed of the water movement, turbulence in the water, changes in the acoustic propagation speed in the water, and interactions with the surface. Doppler spread is difficult to compensate and places a lower limit on the bandwidth of each tone for MFSK non-coherent systems.

Recent work [Green98b] has begun to violate the $T >> T_m$ relationship. In order to improve throughput in non-coherent systems, research is being conducted for the regime where $T < T_m$ and an equalizer is used to minimize multipath effects.

4.5.1.3. Frequency selective fading

In many channels, some tones will experience more propagation loss than others. This condition could be time-varying or fixed by the channel geometry. For the time-varying cases, it is generally accepted that frequencies separated by more than

$$\Delta f_c \approx 1/T_{m,}$$

where T_m is the channels multipath spread, will have uncorrelated fading [Proakis91b]. That is, as long as two frequencies are separated by at least Δf_c, it is unlikely that fading will be observed simultaneously at both frequencies.

Because of the frequency selective nature of fading, many designers choose either to add an error control code to add redundancy to the signal or to repeat transmitting information, utilizing frequencies that are spread by at least Δf_c.

4.5.1.4. Temporal fading

Temporal fading is subject to similar rules as frequency selective fading. In this case, the governing relationship is:

$$\Delta t_c \approx 1/B_d$$

where B_d is the Doppler spread, and Δt_c is the coherence time [Proakis91a]. Designers frequently choose to repeat data, or interleave error correcting codes with a time spread of at least Δt_c.

4.5.1.5. Spatial fading

Fading effects can also be minimized through the use of several input channels. In order to be effective against fading, the receivers should be separated by more than the spatial coherence length, so that the input data is slightly correlated. The goal, then is to space the receivers at about

$$D \approx 2c/B_d$$

where c is the speed of sound.

If the sensors are placed too closely together, then the signals can be highly correlated. Although this does not help to overcome spatial fading, it can be used to boost SNR through beamforming.

4.5.2. Limitations

Several limitations on simple non-coherent receiver design have been discussed in the previous sections.

First, the symbol period should be much greater than the temporal multipath extent, $T >> T_m$. Second, the frequency spacing for adjacent symbols is limited by the temporal extent of the symbol and by the uncorrectable Doppler spread, $F > (1/T + B_d)$, where F is the frequency spacing for adjacent tones in the MFSK system. The number of tones available to the MFSK system is then limited to

$$N = W/F < W/(1/T + B_d) << W/(1/T_m + B_d)$$

where W is the bandwidth available for transmit. For an FSK system that transmits one tone at a time, the data rate of the system is then limited to

$$R = \log_2(N)/T << \log_2(W/(1/T_m + B_d))\,/T_m.$$

As an example, if the maximum expected multipath extent is 1 second, the Doppler spread is 5Hz, and the system's transmit bandwidth is 3000Hz, then

$$R = \log_2(N)/T << \log_2(W/(1/T_m + B_d))\,/T_m = 9\text{bps}.$$

For an MFSK system that interprets each tone as a bit that is either on or off, the data rate of the system is limited to

$$R = N/T << W/(1 + T_m B_d)$$

So, if the multipath extent is 1 second, and the Doppler spread is 5Hz, and the systems transmit bandwidth is 3000Hz, then

$$R << W/(1 + T_m B_d) = 500\text{bps}.$$

Of course, practical limitations such as the need for error correcting codes, fade resistance, synchronization, and Doppler compensation will significantly reduce throughput. In cases where the multipath extent and/or the Doppler spread are known to be limited, the throughput of a non-coherent system could be significantly improved.

4.5.2.1. Diversity Processing

One way to enhance the performance of a communications system is to employ some form of diversity processing. If many frequency bands are available for use, the same data may be transmitted in multiple bands simultaneously, in case one band had a frequency dependent fade at the transmission time. This is known as frequency diversity. Time diversity is implemented by transmitting the same signal at separate times to combat time-selective fading. The use of multiple receivers to combat spatially-selective fading is referred to as spatial diversity. Some systems split the available frequency band into two or more banks, switching banks every symbol period to minimize ISI effects.

4.5.3. Selected Experimental Results

Acoustic telemetry systems in the 1970s and early 1980s were often based on FSK modulation. FSK was used when high reliability was needed and when reverberant acoustic channels existed. This is because FSK systems are robust to time and frequency spreading of the channel. Furthermore, the FSK modulation of digitally encoded data enabled the use of explicit error-correction techniques to increase reliability of transmissions and permitted some level of compensation for the channel reverberation both in time (multipath) and frequency (Doppler spreading) [Kilfoyle99, Garrood82].

A wide range ranges of data rates have been achieved using FSK modulation. Low data rates of less than 100bps were achieved by employing simple coding techniques to improve reliability. The use of FSK in combination with a parametric sonar that generated very narrow beams enabled getting the highest data rates at several kilobits per second with high signal-to-noise ratios. The limitations of using this technique were caused by the power requirements of the parametric sonars [Baggeroer84].

As processor technology improved, technologies that increased demodulation speeds were implemented. These included the use of frequency sweeps instead of tones as well as high-MFSK modulation. With the use of MFSK modulation in the early 1980s, systems operated as low as 40bps for data rates with very low error probability, less than 10^{-6} [Garrod81, Jarvis84]. MFSK systems did achieve data rates of 1200bps for short-range reverberant channels [Baggeroer81, Catipovic84]. Using linear acoustics that employed phase shift encoding, data rates reached 20kbps [Mackelburg81, Kearney84].

[Widmer93] describes a low-frequency, long-range incoherent MFSK communications system that was tested in 1991. This system used waveform blocks that span a bandwidth of 160Hz. Within each block, tones were transmitted at the low and high frequencies to measure and correct for Doppler. The block was then divided into 8 frequency bands corresponding to 8 bits of data, with guard bands separating the data bands. A '1' was represented by the presence of a tone, while a '0' was represented by the absence of a tone. At the transmitter, a block of data was encoded using a 3/4 convolutional encoder (which generated 4 encoded bits for every 3 data bits). After encoding, the data was interleaved to reduce the effects of errors on adjacent bits, and then the signal was modulated. At the receiver, a transmitted signal was automatically detected using criteria based on frequency coverage and duration. The beginning of a transmission was found by searching for the maximum energy over the duration of the communication, and the signal was corrected for Doppler. The received signal was deinterleaved, and a Viterbi decoder was used to recover the data bits.

Testing of this system occurred in both deep and shallow water at ranges of over 185km. For communications in the deep ocean with a surface duct, a bit error rate of 1.3% was observed. In the deep ocean with a convergence zone, 88% of the messages were received without error. These error rates were measured for data rates of up to 40bps. In shallow water, a message success rate of 95% was measured for the same data rates.

Table 4.B provides a summary of various non-coherent implementations found in literature.

Paper	Range m	Center Freq Hz	Data Rate bps	Bandwidth Hz	Mod Type	Depth	
Scally84	280	50,000	1,200		FSK	200m	to surface 1000m
Catapovic84	37,020	50,000	400	10,000	MFSK	800m	
Catipovic89a	700	25,000	5,000		MFSK	20m	
Catipovic89a	700	25,000	1,200		MFSK	20m	
Catipovic89a	3,700	25,000	1,200		MFSK	3000m	
Catipovic90b	750	25,000	10,000	20,000	128-FSK	20m	
Estes90	50	46,000	1,400	10,000	FSK		
Catipovic90c	10,000	25,000	3,750	10,000	128-FSK		
Catipovic91	700	25,000	10,000	20,000	128-FSK	20m	
Smith91	1,000	35,000	1,200		8-MFSK		
Smith91	4,000	35,000	50		8-MFSK		
Merriam93	8,000	16,000	1,200	8,000	MFSK	4,000m	
Coates93	250	600,000	200,000		FSK	15m	
Smith93	6,500	6,000	20		MFSK	27m	BPS estimated
Chappell94	100	17,500	2,400		MFSK		
Ayela94	2,000	53,000	200		FSK		
Pietryka95	10,000	11,000	4,800		MFSK	610m	
Asakawa96		40,500	250		FSK		
Kuchpil97	4,200	10,000	16		FSK	220m	
vonderHeydt98	2,000	18,500	50		FSK	3500m	
Wei-Qing98	4,000	17,500	1,200		MFSK	40m	
Porta98	0	10,500	100	5,120	MFSK		
Freitag98b	3,702	25,000	75	10,000	FSK	shallow	

Table 4.B - Non-Coherent Implementations

4.6. Coherent Communications Processing

In order to recover the information from some received signals, estimation of the transmitted carrier phase angle is required. Demodulators that perform this phase estimation, explicitly or not, are referred to as coherent demodulators.

UWA communication systems that use coherent demodulation are capable of achieving raw data throughputs that are an order of magnitude higher than the non-coherent detection methods [Catapovic90c]. Bandwidth-efficient phase-coherent communications have proven to be a feasible way of achieving high-speed data transmission through most underwater channels, including the severely time-spread horizontal shallow water channels [Stojanovic94a, Stojanovic93b, Stojanovic95c]. Some of the advantages of coherent demodulation are the efficient use of bandwidth and energy, resulting in a potentially a higher data rate. The disadvantages are increased processing complexity, cost, and higher SNR required to maintain a given error rate [Porta98].

4.6.1. Channel Equalization

The optimal demodulator for decoding digital signals received over channels that introduce ISI (see Section 3) in a background of additive Gaussian noise is the maximum-likelihood sequence estimator (MLSE) [Proakis95]. The MLSE receiver for decoding phase coherent modulated signals is unfortunately too impractical for use in underwater acoustic channels due to the symbol extent of the ISI. For channels of this type, signal processing techniques have been developed to compensate for the ISI caused by the multipath. These techniques are termed equalizers and can either be linear or nonlinear in their structure.

A linear equalizer consists of a finite impulse response filter with coefficients that can be updated over time. Linear equalizers are only used when the channel is not overly distorted. For many fading channels, particularly the shallow-water acoustic channel, spectral nulls can appear in the frequency response due to the time-varying multipath. Linear equalizers tend to provide higher gain in the regions near these nulls to compensate for the distortion, thereby amplifying the additive noise [Proakis91a].

Nonlinear equalization methods have been designed for linear channels that exhibit severe ISI-induced distortion. The most popular of these nonlinear techniques for UWA communications has been the decision-feedback equalizer (DFE). The underlying concept behind the DFE is that, after a symbol has been detected, its ISI contribution to future symbols can be estimated and removed. The DFE can be thought of as equalizing a channel in two steps: first, a feedforward section (linear filter) with coefficients $\{a[k]\}$ shapes the overall response appropriately, and then the feedback section with coefficients $\{b[k]\}$ uses previously recovered symbols (termed symbol decisions) to cancel postcursor ISI.

The output of a (nonsparse) DFE, $\hat{d}[n]$, which is an estimate of the current symbol to be recovered, can be expressed as

$$\hat{d}[n] = \sum_{k=-L_1}^{L_2} a[k]z[n-k] + \sum_{k=1}^{M} b[k]\tilde{d}[n-k] \quad ,$$

where $z[n]$ represents the baseband received data and $\tilde{d}[n]$ the symbol decisions. For a multiple input receiver, *e.g.* a receive array, there will be one feedforward filter $a_i[n]$ for each array element i. For a fractionally-spaced equalizer, the received data $z[n]$ will arrive at some integer f times the rate of the decisions $\hat{d}[n]$ and $\tilde{d}[n]$ [Proakis95]. In the equation for $\hat{d}[n]$, $L = L_1 + L_2$ contiguous feedforward taps and M contiguous feedback taps are used. The goal of a sparse equalizer is to set as many of the values of $a[k]$ and $b[k]$ as possible to zero, while still correctly decoding the data or achieving some prescribed mean squared error (MSE).

4.6.1.1. Evolution of the current ACOMMS ATD equalizer

The seminal work on the use of a DFE for the coherent processing of PSK signals collected in horizontal channels at sea was first reported in the early 1990s [Stojanovic93b, Stojanovic94a]. This original receiver algorithm design accounts for the combined time-varying multipath and the severe phase fluctuations exhibited by the horizontal underwater acoustic (UWA) channel. Adaptive channel tracking was addressed by a fractionally-spaced DFE that used a recursive least squares (RLS) algorithm in combination with a second-order phase-locked loop (PLL).

The inclusion of the PLL for phase synchronization within the DFE structure is primarily meant to deal with the rapid and/or large phase fluctuations caused by the channel. The receiver algorithm in [Stojanovic93b] and [Stojanovic94a] incorporates one PLL per channel at the output of the feedforward filter (FFF) to allow for phase synchronization to occur after equalization. Without the PLL, the basic DFE is only capable of correcting for constant phase offset and some slow variability in the phase by continuously varying the phase of the FFFs. The interaction between the DFE update algorithm and the PLL complicates the behavior of both.

This multichannel receiver structure had limited success across deep and shallow environments [Stojanovic94a] because it exhibited slow convergence and instabilities under certain conditions. These performance limitations were caused by the location of the PLLs and their potential to cause the RLS adaptation algorithm to diverge. The performance shortfall was handled by placing a PLL before each FFF rather than after [Johnson97].

More recently, investigations at the Woods Hole Oceanographic Institution (WHOI) have shown that a single PLL can be more robust when dealing with channels that require large feedforward sections [Freitag97a]. The robustness is derived from a basic decoupling of the PLL operation from the RLS operation (or the operation of a least mean squares algorithm) by applying the phase correction to the combined outputs of the feedforward and feedback filters. The single-PLL receiver has provided robust performance across a variety of channels [Freitag97b] and is currently the structure of record on the Acoustic Communications Advanced Technology Demonstration (ACOMMS ATD).

4.6.1.2. ACOMMS ATD Coherent Processing

4.6.1.2.1. Signal Preprocessor

Several signal preprocessing functions are performed on the post-detected input signal prior to equalization using the DFE. The signal is frame synchronized to more precisely estimate the turn-on time of the signal in each input channel. The synchronization is performed by matched filtering to the frequency-modulated probe pulse. The magnitude of the matched filter output is used as an estimate of the channel impulse response.

Next the signal is processed to compensate for gross Doppler shift due to transmit/receive clock mismatch as well as interplatform relative motion. As there is no explicit symbol timing recovery built into the equalizer, it is necessary to compensate for any Doppler induced signal compression or dilation.

Doppler is estimated by picking the peak of a wide-band ambiguity function calculated over the duration of the training symbols [Johnson97]. This is performed by postulating a Doppler induced carrier offset and corresponding compaction/dilation of the training sequence, retuning a reference sequence (the training data) by that shift and correlating the frequency shifted sequence with the received signals. The set of correlator outputs forms an ambiguity surface.

The peak of this surface is found for each input channel. The location of the peak in frequency is used as the Doppler estimate. The signal is retuned by the mean Doppler carrier offset. Then, if the Doppler shift exceeds a programmable threshold, the signal is resampled to compensate for apparent Doppler induced signal compression or dilation. Resampling, if performed, is done to each input signal using a sliding interpolation filter. The *slide* of the filter is calculated to offset the Doppler.

After retuning and, possibly, resampling, the equalizer is initialized. An order estimation routine is performed to determine the size and relative locations of significant multipath energy so that filter support can be placed at these locations. The filter tap placement algorithm places feedforward taps based on the distribution of energy in the estimate of the impulse response and feedback taps based on thresholding.

4.6.1.2.2. Data Equalizer

The data equalization algorithm compensates for the multipath-induced intersymbol interference (ISI) by performing joint channel equalization and data estimation using a process based on decision feedback equalization. By using multiple signal inputs (*receiver diversity*), the effects of selective fading can be ameliorated. The decision feedback equalizer (DFE) consists of a feedforward filter, a phase-locked loop, a decision block, and a feedback filter [Proakis95]. Due to the time-varying nature of the channel, the coefficients of the feedforward and feedback filters, and the phase-locked loop must be adapted during the course of reception. Either a recursive least squares (RLS) or a least mean squares (LMS) adaptation algorithm is used to perform this filter adaptation [Haykin96]. Figure 4.3 depicts the modified structure of the DFE that has been implemented on the BAE SYSTEMS VME modem for data equalization of QPSK and "medium-rate" BPSK packets. The rate refers to the coding applied to the data and is described in a section that follows on error control coding/decoding for the ACOMMS ATD.

In addition to inducing distinct arrivals, the underwater acoustic channel induces Doppler spreading on any given arrival. The feedforward filter serves to *despread* the energy contained within these arrivals by filtering the received signal with an inverse of the estimated channel.

The decision block estimates the value of the received symbol based on the phase compensated output of the feedforward and feedback filters. The feedback filter is used to remove ISI from the output of the feedforward filter due to past symbols. The difference between the observed symbol and the estimated symbol is used as an error to update the filter tap weights.

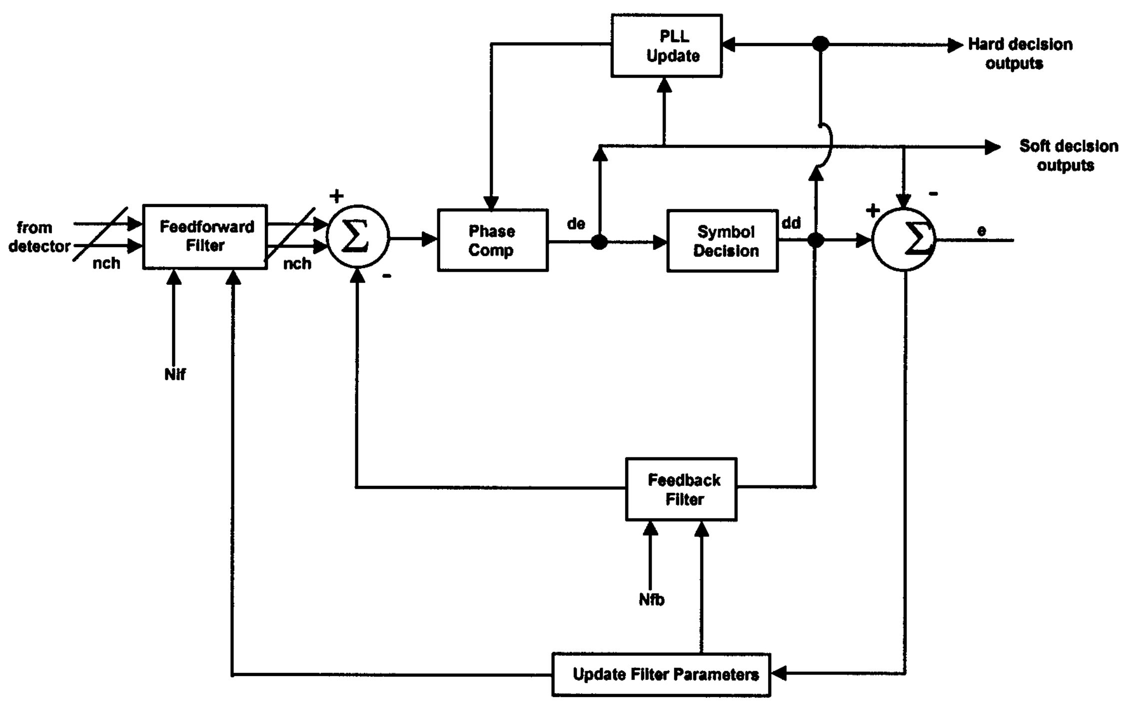


Figure 4.3 - Decision Feedback Equalizer Block Diagram

"Low-rate" BPSK packets have also been transmitted and processed in quasi-real time and off-line as part of the ACOMMS ATD. They have been included in order to exploit the performance improvements afforded by lower rate codes in combination with DFE-embedded error control coding (ECC). A modified equalization structure developed at WHOI is used that performs soft-decision decoding of the received symbols within the equalizer. After the training phase of the receiver is complete, the data are processed in sections which are the same size as the block code length, freezing filter updates during a first pass, then performing the soft-decision decoding step, and then finally updating the filter coefficients with reliable feedback decisions. The equalization proceeds through the packet in this manner so that the final output of the equalizer is the data with the inner block code already decoded.

4.6.1.2.3. Carrier Phase Synchronization

Another component of the joint channel equalization and data estimation process is that of carrier phase synchronization. Doppler frequency shift is equal to relative phase shift per unit time. In the underwater acoustic channel, the received carrier phase offset varies with time and arises from three sources - a constant phase offset between transmitter and receiver clocks, a difference in clock frequency between transmitter and receiver, and the propagation time of the signal from transmitter to receiver. The propagation time is affected by propagation distance and by sound speed along the signal path. These, in turn, vary with platform motion and with changes in the transmission medium. In order to achieve the bandwidth and power efficiency of coherent communications systems, phase synchronization is performed jointly with channel equalization. This results in proper operation of the equalizer even in the presence of the severe phase fluctuations encountered in the underwater acoustic channel. The decision block measures the distance from the expected location of the symbols to the measured location. Phase offsets result in a rotation of the received signal constellation about the origin. The resulting offset of the received symbol must be minimized along with other error sources.

4.6.1.2.4. Error Control Decoding

Error control coding (ECC) takes advantage of redundancy introduced into the input symbol stream at the transmitter to detect and/or correct channel errors. The parameters of the code determine the type of errors that can be detected/corrected. The approach implemented for QPSK and medium-rate BPSK on the ACOMMS ATD modem is designed to correct both random errors as well as burst errors of some nominal length.

The input and output to the encoders and decoders are arrays of symbols. Two types of block encoder/decoder pairs are employed. The notation (x,y) indicates the number of encoded symbols x for each y input symbols. The interleaver is parameterized by I, the interleaver block size. In the current implementation, a heavily coded inner code is used to correct short burst errors. An interleaver and highly coded outer code cleans up the residual errors. The combination of these two codes allows for the correction of a wide combination of random and burst bit errors.

The output from the second coder then passes through a bit interleaver. This interleaver takes in a defined number of blocks and rearranges the bits in a manner that will increase the resilience to channel burst errors. Finally the data undergoes a pseudo-random bit-flipping process referred to as whitening. This process removes any correlation in the data stream and ensures that the transmitted symbols occur with equal probability.

Decoding of the data stream is done in reverse of the encoding. First, the pseudo-random bit-flipping and then the bit interleaving are undone. The inner code and outer code decoders then attempt to correct any errors that occurred during transmission and produce an output bit stream.

The low-rate BPSK packets are encoded differently. A very heavily coded inner code is deciphered within the equalizer, while an outer code is able to correct the occasional code-word errors in the inner code.

Applying coherent demodulation to appropriately modulated waveforms potentially allows for greatly increased bandwidth efficiency. This burst rate efficiency for the ACOMMS ATD, specified in bits per second per Hertz of bandwidth (bits/s/Hz), is 0.5 and 1 bits/s/Hz for BPSK and QPSK, respectively. The realized throughput is not this high as a number of transmitted bits must be allocated to protocol maintenance and error detection and correction.

4.6.1.3. Sparse Equalization

The complexity of the DFE is directly related to the number of coefficients used in the feedforward and feedback filters. ISI can be span tens of symbols, requiring large filters that must be adapted at the symbol rate. Updating these filters can be prohibitive in certain underwater channels, and so it is desirable to find a way to reduce the amount of computation necessary to operate a DFE-based receiver. In [Johnson94, Kocic94a, Kocic94b, Kocic95], methods for reducing the number of filter coefficients were introduced. The basic idea is to search the channel response for intervals that contain a significant portion of the received signal energy. Then, the feedback filter coefficients corresponding to sections of the response with little or no signal energy are discarded. This effectively makes the channel sparse. The remaining coefficients are updated regularly, resulting in a lower complexity receiver that performs nearly as well as a full complexity receiver.

4.6.1.4. Reduced-Complexity Receiver Using Spatial Diversity

A reduced-complexity receiver for underwater acoustic channels jointly applies a spatial processor and a temporal processor to received signals to reduce the processing load of the receiver [Stojanovic93a, Stojanovic93b, Stojanovic95c]. The spatial processor component involves an optimal spatial diversity combiner. The temporal processor component consists of an equalizer. The two components are determined adaptively, creating a suboptimal, lower complexity receiver that achieves nearly the same performance as a full complexity receiver (i.e. an adaptive decision feedback equalizer). Other benefits include improved algorithm stability and a reduction in noise enhancement.

Conceptually, beamforming is typically viewed as steering nulls to cancel unwanted interference, whereas diversity combining attempts to make use of repetitive signal arrivals. Both are important techniques for mitigating effects caused by multipath propagation. Stojanovic, et al. establish an equivalence in performance between a pure K-channel diversity combiner that has no knowledge of the spatial signal distribution and a fixed beamformer followed by a P-channel equalizer that makes use of the angles of signal arrivals ($P < K$) [Stojanovic95c]. The first approach is the K-channel fully adaptive equalizer that has been well-studied. The second approach, as stated, does not appear feasible, as the angles of arrival will change with time due to the relative motions of transmitter and receiver. However, this approach can be modified by introducing an "angle-locked loop" to track the angles of arrival. Such a loop may be extremely sensitive to the choice of initial angle estimates and angle tracking constants.

The proposed algorithm combines these two approaches, through the use of an unconstrained K x P adaptive beamformer followed by a P-channel adaptive equalizer. Essentially, the two components share the task of mitigating the intersymbol interference (ISI). This is accomplished by joint adaptation of the beamformer and equalizer. The use of the adaptive beamformer requires no explicit assumptions about the underlying spatial signal distribution, avoiding the need for explicit angle tracking. Furthermore, there is less noise enhancement as compared to the K-channel equalizer since the adaptive filter is smaller.

Computationally, the savings are clear, especially when there are several channels to be processed. Using a length-N equalizer, the fully adaptive K-channel equalizer requires K x N taps. The beamforming approach requires $K \times P + P \times N$ taps. The number of computations is greatly reduced when N is large and K is several times greater than P.

Despite this reduction in complexity (as compared to the K-channel equalizer), there may not be any performance degradation, as long as there is an underlying spatial signal distribution permitting the decomposition into a beamformer and an equalizer. This is because the multipath structure of the received signals is not independent among the array sensors. An additional benefit of the reduction in complexity is that with fewer adaptive parameters, simpler update algorithms with improved stability can be used, like standard RLS.

The choice of equalizer is left to the user. Stojanovic, et al. suggest the use of a multichannel decision feedback equalizer for the underwater acoustic environment. This yields the following algorithm:

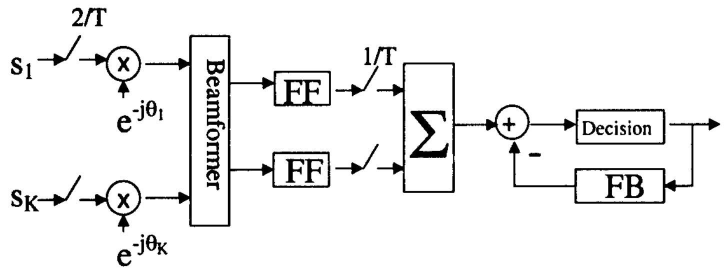


Figure 4.4 - Multichannel Decision Feedback Equalizer

4.6.2. Selected Experimental Results

One of the earliest works reported on coherent communications is found in [Birdsall84]. It describes an energy efficient, very long range communication system (1 bit per minute, 3000km range) developed for the Ocean Acoustic Telemetry System. The results indicate that using low-power transmission of very long symbols maximizes efficiency. The system was first used to test Gold codes as a set of symbols to transmit, but the results were poor, possibly due to non-uniform noise levels. Then m-sequences were transmitted with a 224Hz center frequency. The signals were repeated multiple times in order to boost received energy.

In the early 1990s, phase coherent methods with differential and purely coherent detection were first implemented. [Proakis94] describes deep and shallow water experiments using a single channel DFE that accomplished adaptive equalization jointly with carrier phase synchronization. The measured impulse responses presented in [Proakis94] span no more than 20 symbols. A successfully demodulated QPSK packet is shown that was transmitted at 666 bits per second (bps) at a range on the order of 2000km (3 convergence zones) in deep water. In shallow water, 8PSK was transmitted over a distance of approximately 90km and the results of two packets are presented. No errors were detected on a packet using a transmission rate of 600bps, but a bit error rate of 10^{-2} was measured for a packet with a transmission rate of 1500bps. These experiments established the promise of PSK-based phase coherent communications.

A 1991 experiment in a deep-water channel off the coast of California established the superiority of a multichannel receiver that jointly performs carrier phase recovery, multichannel combining, and fractionally-spaced decision feedback equalization [Stojanovic93b]. In this experiment, a 12-element vertical receiver array spanning a depth from 500m to 1500m was used to receive signals transmitted over ranges of 75-260km. The measured multipath spread was on the order of 50ms. QPSK, 8QAM, and 8PSK were transmitted at burst rates up to 1000 symbols per second (i.e. 2000, 3000, and 3000bps respectively) at a carrier frequency of 1kHz. The multichannel receiver showed a 3 to 5dB improvement over a single channel receiver, along with some error-free receptions that eliminated the errors observed in the single channel case. Another test at the same location involved a 32-element receiver array [Stojanovic94a] spanning a depth from 375-1750m. Similar results were obtained.

A multichannel receiver was also tested in shallow water in 1991 at Buzzards Bay [Stojanovic94a]. QPSK was transmitted at a power of 183dB re μPa with a carrier frequency of 15kHz. Signals were received over a range of 2-8km in 17m-deep water. The receiver incorporated one directional hydrophone and two omnidirectional hydrophones. Successful packet transmissions were achieved at burst rates reaching 20kbps.

The experiments described by [Coates93] used descendants of the Birmingham Acoustic Signaling Systems (BASS) telemetry link originally developed for the Woods Hole Oceanographic Institute (WHOI) to perform DPSK communications experiments at carrier frequencies that ranged from 5kHz to 600kHz. A summary of these results is given in Table 4.C.

Carrier Frequency (kHz)	Modulation	Bandwidth (kHz)	Data Rate (kbits/sec)	Max. Range (m)	Depth (m)	Bit Error Rate
600	2-DPSK	10	10	250	6	$\sim 10^{-2}$
600	4-DPSK	10	20	250	6	$\sim 10^{-2}$
50	2-DPSK	10	10	900	15	$>10^{-4}$
50	2-DPSK	10	10	2200	15	$>10^{-3}$
50	4-DPSK	10	20	900	15	$>10^{-4}$
50	4-DPSK	10	20	2200	15	$>10^{-3}$
5	2-DPSK	1	1	10,000	80	$>10^{-5}$ $<10^{-2}$

Table 4.C - BASS Summarized Results

[Galvin94] describes two pier-to-pier BASS tests across a very shallow river inlet with range to depth ratios of 70:1 and 150:1 that are summarized in Table 4.C. The system used transmit (+/-3 degrees vertical x +/-20 degrees horizontal) and receive (+/-4 degrees vertical by +/-60 degrees horizontal) beamforming to minimize multipath. Over a 2.2km path, data was exchanged at burst rates of 10kbps (2-DPSK) and 20kbps (4-DPSK), where the integrity of the link varied throughout the test. Some good text was received, while some data was completely corrupted. The 2.2km channel was considered reverberation limited, with signal to reverberation ratios as low as 10dB. It was considered impossible for the equalizer to track these changes. At 900m range, 10kbps and 20kbps exchanges also occurred, with low bit error rate (BER) data occurring somewhat more reliably.

The results of BASS experiments characterizing the impulse response of a channel at 5kHz in the Mediterranean Sea are described in [Galvin96] and [Zheng96] and shown in the last row of Table 4.C. A real-time M-ary differential phase shift keying (MDPSK) communication system utilizing parametric transduction was constructed. The primary frequency was at 50kHz with a 5kHz difference frequency. Amplitude fluctuations were shown to have a Ricean distribution from pulse to pulse, and a 0.25Hz period of oscillation. Phase fluctuations were Gaussian in nature, with a similar period. The experiments were conducted in a high SNR environment, so the amplitude fluctuations were attributed to fading, not to noise. The amplitude and phase variations had 1.24dB and 1.76 radian standard deviations, respectively. Although the phase fluctuations are large, the oscillations occurred at a very low frequency (0.25Hz), so they were not too troublesome. Burst data rates were achieved at 1, 2, or 3kbps for 2-DPSK, 4-DPSK, and 8-DPSK respectively in shallow water with a range-to-depth ratio greater than 100. The 2-DPSK bit error rate varied from 10^{-5} at 30dB to 10^{-2} at 10dB. The BER range was caused by a combination of bad weather conditions and ship noise within the experimental area.

The reduced complexity receiver using spatial diversity was introduced in [Stojanovic93a], and was tested on experimental data gathered in May 1992 by WHOI on the New England Continental Shelf

[Stojanovic93b, Stojanovic93c, Stojanovic95c]. In shallow water (50m water depth), a 20-element vertical receiver array was deployed spanning depths from 15 to 35m. Signals were transmitted at ranges of 30 to 120km at a power of 193dB re μPa and a carrier frequency of 1kHz. The measured multipath spread was about 100ms. QPSK and 8PSK modulations were used at burst rates up to 1000 symbols per second (i.e. 2 and 3kbps). The spatial processor combined the 20 receiver input channels into 3 channels for adaptive equalization. A processing gain of 2dB was obtained over the best case same-size, full-complexity receiver (i.e. choosing the best 3 channels rather than adaptively combining the 20 inputs into 3 outputs). Also, no loss in performance was observed when compared with the full-complexity receiver operating on all 20 channels.

A description of an acoustic communications system developed for Marine Utility System (MARIUS) AUV is discussed in great detail in [Barroso94]. Selected bandwidth efficient PSK modulation and a digital equalization structure were used to minimize the problems created by multipath propagation. In June 1992, a test message was transmitted at 120 baud at approximately 1km range. A phase-locked loop (PLL) was used around the decision block of a DFE. A data-driven timing loop adjusted the exact time that samples were taken for digitization, even though the feed-forward section was fractionally spaced. A fast transversal filter (FTF) implementation was used for the tap adaption block. To maintain FTF stability, two requirements were identified: 1) detect numeric instabilities and restart the filter a few samples back, prior to the instability taking hold and 2) use of extended precision 48-bit numbers rather than the fixed point 24-bits supported by the hardware.

Reduction in complexity achieved by sparse equalization was demonstrated on data collected in an Arctic experiment in the Beaufort Sea in March 1994 [Johnson95]. The ocean depth was several kilometers and the receiver was tested at a range of 3.7km. Sparse multipath components were found in these data that fell within 32ms windows. QPSK was transmitted at a carrier frequency of 15kHz at a burst rate of 5000bps. Some error-free decoding was attained despite a reduction in the number of filter taps by a factor of 7 after sparsing. Two other methods of reducing computation were also demonstrated during this experiment. One method called for a reduction in the frequency of equalizer updates. Instead of updating the DFE at the symbol rate, the DFE was adapted only when the mean squared error (MSE) exceeded a threshold. Performance was not affected by this reduction in complexity. The other method for reducing the computational load involved the use of a lower-complexity update algorithm. The authors implemented an update algorithm using a least means squares (LMS) algorithm for the feedback taps and recursive weighted least squares (RWLS) for the feedforward taps that achieved both the desired performance and a reduction in computations on the order of 75% compared to standard RWLS.

Results of several tests using FSK, BPSK, and QPSK modulation in deep and shallow water are described in [Carvalho95]. This work focuses on instrumenting test ranges so that submarines can participate more fully and relay their position to the test coordinator. The experiments achieved BER between 1% and 10%, with some experiments going as high as 49%. Communication ranges were 1.83 to 9.1km with SNRs between 11 and 40dB. There were no apparent trends in the data to indicate that high SNR-performance outweighs that of low SNR or that shallow-water data suffers a performance shortfall relative to deep-water data. There was failure of the FSK signaling that was attributed to its inability to discriminate against multipath interference without added diversity in time, space, or frequency.

[Carvalho95] also describes a coherent PSK equalizer designed by NUWC and WHOI. This equalizer incorporates adaptive differential feedback with phase correction (earlier work by this author used differential PSK coding). Using the coherent equalizer, the BER dropped to 10^{-3} or 10^{-4} at SNRs of 25dB and higher. With the addition of ECC, a BER of 2×10^{-5} at SNRs of greater than 35dB are reported for a data rate of 4600bps and an information rate of 2300bps. This was a dockside test in Narragansett Bay over a range of 366m.

A counterintuitive result is presented in [Stojanovic96c]. Improved performance of an adaptive equalizer was made possible through the use of *higher* signaling rates. This was demonstrated for a shallow water channel at a range of 2km, transmitting at a 15kHz carrier frequency. Communications signals were successfully equalized at a transmission rates up to 20kbps.

Early work in placing an acoustic modem on an UUV for the Autonomous Minehunting and Mapping Technology (AMMT) program are described in [Freitag96]. During trials in the winter and spring of 1996 off Fort Lauderdale, Florida, the acoustic modem uploaded regular status messages from an UUV to a surface vessel once every two minutes and periodically transmitted images from an on-board camera. The modem successfully operated at ranges over 2km with vehicle depths up to 300m. Raw burst data rates of the uplink were up to 10kbps, with the system operating consistently at 5kbps. The downlink achieved burst data rates of 2.5kbps. The configuration for the system served as the initial form of the ACOMMS ATD equalization structure with a sparse equalizer, soft-decision decoder, QPSK modulation, block encoding, and intra-packet interleaving.

Improvements due to better Doppler tracking and correction were tested by WHOI and reported in 1997 [Johnson97]. The authors studied two different scenarios: 1) Doppler shift experienced when an AUV passes a stationary vessel and 2) Doppler shift for two moving vessels. During each of these tests, the Doppler was successfully tracked for data rates of up to 10kbps using QPSK modulation. The superior tracking allowed for a reduction in receiver complexity on the order of a factor of 50.

Doppler tracking and correction were also incorporated into a study described in a 1997 WHOI report [Freitag97a]. This study specifically focused on the long-range shallow water channel. At a close range of 6.5km, error free receptions were achieved for QPSK data transmitted at a rate of 2.5kbps at a carrier frequency of 2.25kHz. At a longer range of 45km, QPSK was transmitted with similar parameters and similar success. A wedge channel (deep water to shallow water propagation) with a range of 11 miles was also tested for this report. Using the same parameters, transmissions were not completely error-free, although error correcting codes compensated for some of the errors present.

[Chang-Hong98] and [Wei-Qing98] describe tests of an MPSK UWA communication system of an AUV with design nearly identical to that originally presented in [Stojanovic93b] and [Stojanovic94a]. The modem used a 13-element Barker probe, a short gap between the probe and the packet, training symbols, an then data symbols. LMS, RLS and FOLMS (fast self-optimized LMS) algorithms were analyzed to adjust the equalizer weights. FOLMS is an LMS variant where the step-size parameter is adapted. For the phase correction block (one phase correction in each feed-forward section), both the second-order digital PLL (DPLL) and a FOLMSPE (fast self-optimized LMS phase estimator) algorithm were used. Experimental results showed that FOLMS and FOLMSPE together worked better than FOLMS and DPLL or SFTF (stabilized fast transverse filter) and DPLL (based on output SNR and BER (10^{-3} and 10^{-2} respectively)) for the very simple channels that were encountered (1 or 2 feedforward (FF) taps, 11-15 feedback (FB) taps, 3 input channels). Data received at ranges from 100 to 4000m with a transmission rate of 10kbps were successfully processed and the BER was on the order of 10^{-5}.

In [Freitag98b], medium and high bit rate coherent communications from an AUV to a support research vessel (RV) are reported. The high data-rate communications implemented PSK, while FSK was used as a back-up communications scheme for lower data rates and noisy conditions. The modem transmitted at a raw burst rate of 10kbps, reduced to 6.7kbps with the addition of error-correction coding. The reception ranges were from one to two kilometers. Data were sent from the AUV to the RV twice per minute in one kilobyte packets. The down-link data from RV to the AUV operated at a 3kHz carrier frequency and 2kbps to ranges greater than 3km.

As discussed previously, the error rates for the DFE can be improved through the use of error control coding (ECC). Such coding works by introducing redundancy into the signal being transmitted, with the primary downside being that the bit transmission rate is effectively decreased along with the error rate.

[Subramaniam98] describes a trellis coded modulation (TCM) scheme that was implemented through simulation for the vertical channel and for shallow-water medium range channels. A rate 2/3 convolutional code was used to encode bits and then was "transmitted" using 8PSK.

A similar scheme has been investigated as part of the ACOMMS ATD. Processed data collected on and just off the New England Continental Shelf in 1997 was reported in [Kilfoyle98a]. Here signals were coded with a concatenated outer block code and a TCM inner code. The received signal was processed using a DFE and decoded using a Viterbi decoder with finite, truncated decision delay embedded in the DFE. QPSK with no coding, 8PSK with a rate 2/3 code (truncated to length 3), and 16-QAM with a rate ¾ (truncated to length 7) were compared. Carrier frequencies of 3.5 and 25kHz were used with symbol rates of 1.25 and 5kHz, respectively (bit rates depend on the type of modulation and the coding employed). An average coding gain of 2 to 3dB was reported for the 8PSK TCM packets while a coding gain of 2 to 5dB was reported for the 16-QAM TCM packets. Coding gains did not attain the theoretical limits possibly due, in part, to impulsive biological noise sources.

Communications in a shallow-water underwater environment has been demonstrated as part of the EC MAST II project ROBLINKS (long-range shallow-water ROBust acoustic communications LINKS) by [vanGijzen00]. The experiments took place in the Dutch coastal waters of the North Sea in April-May 1999. The transmitter was located on a moving vessel while the receiver was fixed on a platform. The average water depth was 18m, and the signaling range varied between 1 and 10km. The source depth was 9m and the average power was 190dB over a frequency range of 1-14kHz. The receiver consisted of a vertical array with 20 hydrophones with a 13m aperture. A bit rate of 4kbps was observed using BPSK modulation with the moving vessel traveling at 4kts away from the source.

A communications sea trial performed in the wake of a surface vessel is described in [Eggen00]. The dominant interference source is due to propeller cavitation. Receiver structures studied for this problem include a decision feedback equalizer and a Viterbi receiver. Ranges between the transmitter and receiver varied from 300m to 3km. The water depth varied from 15m to a few hundred meters. The bandwidth used by the PSK signaling was 2kHz. Bit rates of 1.5-5kbps were observed over the specified ranges. A performance gain was observed for the Viterbi receiver structure in the presence of ship noise.

The use of a coherent path beamformer (CPB) and recursive least squares (RLS) adaptive beamformer, both in combination with an RLS time filter to process communications signals in the underwater channel was studied by [LeBlanc00a]. The CPB forms a beam adaptively in the direction of a collection of coherent signals representing the strongest path while forming nulls in the direction of uncorrelated signals (assumed to be interference). The RLS adaptive beamformer is an application of the RLS algorithm to beamforming. These approaches were tested near Tampa, FL in a water depth of 7m. The transmitter was mounted on an AUV 2m above the seafloor and generated a source level of 175dB at a center frequency of 50kHz and a bandwidth of 20kHz. A 64-element receiver array was located at a range of 200m. The CPB beamformer/RLS time filter combination was found to be more robust and had less stringent SNR levels required for successful equalization as compared to the RLS beamformer/RLS time filter combination. Error-free transmission was observed at bit rates.

The CPB/RLS approach was enhanced by [Beaujean00] to include angle beam diversity and was termed the multiple beam adaptive decoder. In addition to forming the beam corresponding to the strongest path, this method also involves forming beams along secondary paths that contain useful information. The single and multiple CPB/RLS approaches were tested in 14m deep water off of the coast of Fort Lauderdale, FL. The distance between source and receiver varied between 3000 and 3500m. The test involved a real-time implementation for communications in a marginally overspread channel operating between 16-32kHz. BPSK and QPSK modulation was used to encode data. With the single CPB method, rates up to 8000bps were processed reliably. The multiple CPB/RLS combination enabled reliable communications at rates up to 32000bps. The limitation on performance was determined to be the 14kHz bandwidth of the source transducer.

Table 4.D provides a summary of various coherent implementations found in literature.

Paper	Range m	Center Freq Hz	Data Rate bps	Bandwidth Hz	Mod Type	Depth	
Hakizimana87	13	5,000	1,250	1,250	BPSK	lake	
Coates93	2,200	50,000	20,000		4-DPSK	10m	
Stojanovic93b	88,848	1,000	2,000		QPSK	shallow	
Stojanovic93b	203,610	1,000	1,000		8QAM	deep	
Catipovic94	460	15,000	5,000		QPSK	20m	
Howe94	926	40,000	9,975		DPSK	100m	
Herold94	5,550	15,000	5,000		QPSK	12m	
Johnson94, Johnson95	5,000	15,000	5,000		QPSK	63m	
Proakis94	88,848	1,600	1,500		8PSK	shallow	
Proakis94	203,610	1,600	666		QPSK	deep	
Barroso94	1,000	53,000	500		PSK	deep	
Ayela94	4,000	12,000	2,400		PSK	shallow	
Brady94	2,000	15,000	1,000	6,000	BPSK	2000m	
Gray95	33,318		333		QPSK	shallow	
Gray95	55,530		333		QPSK	shallow	
Loubet95	63,000	1,000	500		PSK	400m	
Stojanovic95c	88,848	1,000	1,500		8PSK	shallow	
Neasham96	2,000	50,000	40,000	30,000	2/4/8 PSK	100 m	
Coates96	130	300,000	20,000	50,000	param. BWF; 2, 4 DPSK	25m	
Stojanovic96a, 96c	1,851	15,000	40,000	10,000		shallow	
Singh96	2,000	15,000	5,000		QPSK	shallow	
Galvin96	1,700	50,000	1,000	20,000	DPSK	700m	
Kojima96	78	55,000	16,000	10,000	QPSK	shallow	
Zvonar96	750	15,000	2,000		BPSK	18m	
Thompson96, Sharif97a, 97b	50,000	1,700	212		BPSK	100-300m	
Hou97	50	50,000	10,000		BPSK	lake	
Kojima97	800	96,000	32,000	20,000	QPSK	deep	
Sharif97b	2,000	50,000	20,000		QPSK		
Jones97	1,000	50,000	30,000	10,000	8-DPSK		vert. comms
Zvonar97		15,000	1,000		BPSK	2500m	
Freitag97b	3,000	3,000	2,000		QPSK	deep	AUV->surface 1000r
Freitag97b	40,000	5,000	2,000		QPSK	deep	
Freitag97b	40,000	2,250	2,500		QPSK	deep	
Freitag97b	30,000	2,250	2,500		QPSK	shallow	
Freitag97b	45,000	2,250	2,500		QPSK	shallow	
Loubet97	79,593	1,500	36	375	SS	shallow	
Loubet97	9,255	2,000	81	500	SS	shallow	
Loubet97	37,020	1,666	24	520	SS	shallow	
Al-Kurd98a_b	1,000	3,500	2,000		QPSK		
Wei-Qing98	4,000	17,500	5,000		QPSK	40m	
Caimi98c	366	50,000	10,000	20,000	QPSK	15m	
Albonico98	900	62,000	33,000		2, 4, 8-PSK		
Freitag98b	3,702	25,000	10,000	10,000	PSK		
Boulanger98	20,000	1,666		520	BPSK		
Boulanger98	50,000	1,666		520	BPSK		
Blackmon99, Jarvis Oceans 97	4,500		1,800		PSK		
Pointer99	1,500	15,000	41,000		COFDM		
Pointer99	1,500	15,000	41,000		16QAM	150m	
Pointer99	3,000	15,000	20,000		QPSK	150m	

Table 4.D - Coherent Implementations

4.6.3. ACOMMS ATD Results

The primary objective of the ACOMMS ATD was to develop and demonstrate emerging undersea acoustic coherent communication technologies at operationally useful ranges and data rates. The secondary objective of the ACOMMS ATD was to develop a fleet-compatible advanced acoustic communication capability that can easily be transitioned into planned Fleet Commercial-Off-The-Shelf (COTS) upgrade programs. In summary, the ACOMMS ATD has demonstrated the potential to provide a reliable, robust, moderate data rate acoustic communications capability for tactical use amongst submarines, surface combatants, unmanned undersea vehicles (UUVs), and other platforms.

The current ACOMMS ATD algorithms are originated from the algorithms developed in the early 1990s by the Woods Hole Oceanographic Institution (WHOI) and Northeastern University (NEU) [Stojanovic93b, Stojanovic94a]. During the three-year ACOMMS ATD period, a total of four at-sea demonstrations were conducted. A key attribute of the testing is that the channel complexity grew with each demonstration. The evolution of the coherent processing algorithm from signal preprocessing through to ECC was based on the experience gained in previous demonstrations and other sea-tests performed. A quick-look at the four demonstrations is shown in Table 4.E. All four at-sea demonstrations successfully met or exceeded the exit criteria. The actual results are classified.

Test Date	Location	Vessels	Exit Criteria
20 June – 2 July 1997 (FY97)	Shallow and deep waters in Narragansett Bay Operating Areas	RV to/from RV	MF, HF 14.8km @ 2.4kbps, 3.9km @ 10bps
9-14 December 1998 (FY98)	Shallow and deep waters in Narragansett Bay Operating Areas	SSN to/from RV	MF 55.6km @ 2.4kbps
11-16 April 1999 (FY99)	Shallow and deep waters in Southern California (off San Diego)	DDG to/from SSN	MF 64.8km @ 2.4kbps
15-25 May 1999 (FY99)	Shallow and deep waters in Hawaii Operating Area	SSN to/from RV	HF 4.5km @ 10kbps

Table 4.E - Quick-Look of ACOMMS ATD At-Sea Demonstrations

The first at-sea test was conducted in shallow and deep water in the Narragansett Bay Operating Areas (NBOA) off the coast of New England in early summer 1997. The shelf water depths ranged from 122-183m, the shelf break ranged from 183-457m and the deep-water depths ranged from 762-1829m. Transmission frequencies and bandwidths were selected to approximate the tactical frequencies of interest while optimizing the predicted performance of the available test equipment. Research vessels (RVs) were used for the transmit platform and for the receive platform.

During the test, data was processed in-situ using a PC modem system developed by WHOI. The primary purpose of this processing was to collect the data in real-time and process packets at pseudo real-time to gauge the difficulty of the channels and the expected algorithm performance. For the majority of the test, the algorithm was initialized with static parameter settings set *a-priori*. Off-line processing includes the efforts applied to equalization of the data post-test in a laboratory setting. The analyses were conducted by WHOI, BAE SYSTEMS, and the Naval Undersea Warfare Center Division in Newport (NUWCDIVNPT) from re-digitized data.

The major obstacle to robust in-situ performance was the multipath complexity of ducted arrivals. Much of the gain observed in the off-line results stemmed from proper initialization and selection of efficient equalizer realizations to facilitate the large degrees of freedom required for these extended multipath environments.

The data contain transmissions that propagated along the shelf, up the shelf break (183-457m water depth), and beyond the shelf break in deep water (762-1829m water depth). All of the packets contain 6784 symbols (512 training) of mid-frequency QPSK data. The transmit platform towed the projector array at speeds between 0 and 7 knots while the receive platform drifted.

All signals were received on a 16-element vertical array with approximately 1-foot separation between elements. The array was placed at depths of 60 and 200m in the shallow and deep-water segments, respectively. A 13-chip Barker sequence was transmitted a fixed time prior to each QPSK packet to enable packet detection and synchronization. Typical channel responses and their corresponding packet numbers from the shallow and deep sets are shown in Figures 4.5 and 4.6, wedge and deep, respectively. These responses are derived from matched-filtering the probe signals on the first element of the receive array. Within these figures are responses of varying complexity and length, some sparse and others continuous. Note that 0.4s along the horizontal axes of these figures equates to 500 symbols in time.

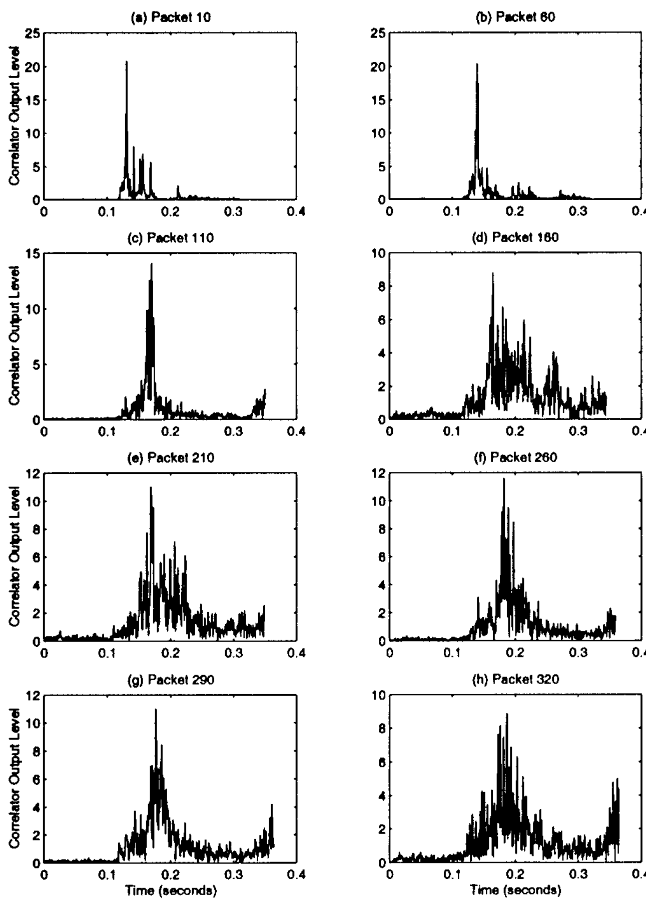


Figure 4.5 - NBOA97 Shelf-Break (Wedge) Channel Responses

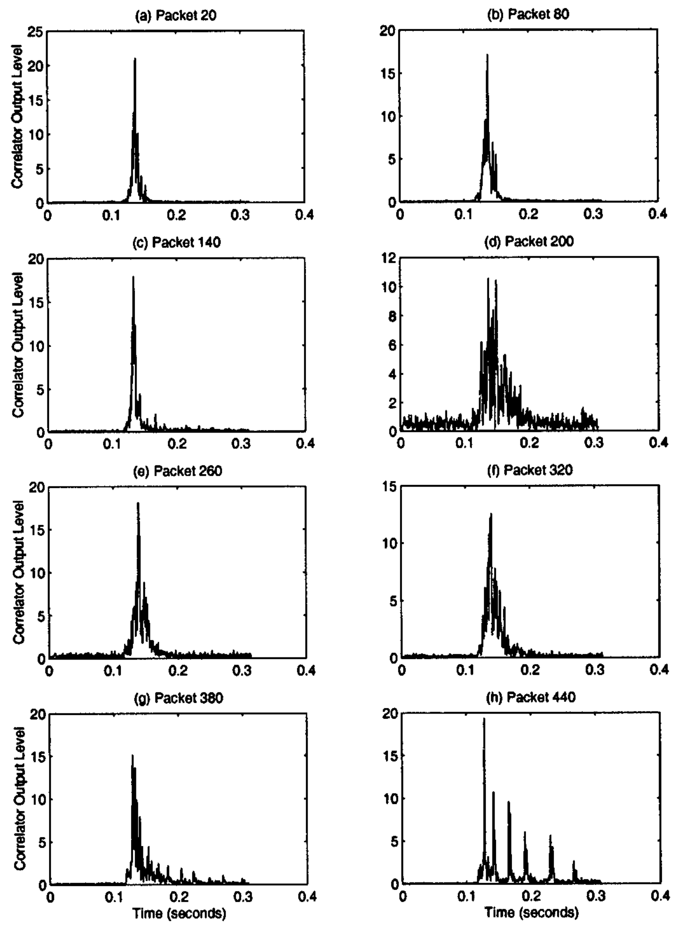


Figure 4.6 - NBOA97 Deep-Water Channel Responses

The in-situ performance summary shown in Table 4.F was derived using a subset of eight channels for both the HF and MF arrays. The performance degradation due to processing a sub-array versus a full array is considered negligible as the ranges of interest had high SNR. For each event, the number of packets detected and the percentage of packets decoded with less than 10^{-2} BER (uncoded), referred to as availability, is given in terms of the operating ranges and input SNR.

Event	No. of Packets	Availability (BER < 10e^{-2}) In-Situ	Off Line	Gain	SNR* @ Max Range (dB @ nmi (km)) In-Situ	Off Line	Gain
Wedge	481	66%	96.5%	30.5%	8dB @ 24.4 (45)	8dB @ 24.4 (45)	None
Rough Topology	644	85%	96.6%	10.6%	30dB @ 13.0 (24)	30dB @ 13.0 (24)	None
Flat Bottom	345	80%	99.4%	19.4%	25dB @ 10.9 (20)	25dB @ 10.9 (20)	None
Shoal	316	99%	100%	1%	10dB @ 22.8 (42)	10dB @ 22.8 (42)	None
DDG Emulation	362	46%	98.3%	52.3%	20dB @ 11.9 (22)	20dB @ 11.9 (22)	None
HF-Shelf-Demo			90.3%	14.3%			
Long Range (0-3nmi)	398	76%			25dB @ 3.1 (6)		
Short Range (0-1.5nmi)	485		91.3%	15.3%		40dB @ 1.6 (3)	N/A
Long Range (1.5-3nmi)	359		88.9%	12.9%		25dB @ 3.5 (7)	0.4 nmi (1 km)
HF-Deep-Demo			72.7%	10.2%			
Short Range (0-1.5nmi)	117	62%			40dB @ 1.6 (3)		
Short Range (0-1.5nmi)	186		98.4%	36.4%		40dB @ 1.6 (3)	None
Long Range (1.5-3nmi)	363		59.5%	N/A		20dB @ 3.5 (7)	20 dB / 1.9 nmi (4 km)
MF-Deep-Demo			73.4%	6.4%			
Med Range (0-10nmi)	409	67%			5dB @ 10.9 (20)		
Short Range (1-5nmi)	313		90.3%	23.3%		20dB @ 5.4 (10)	N/A
Med Range (5-10nmi)	190		45.2%	-21.8%		5dB @ 10.9 (20)	None
MF-Deep-CZ							
Long Range (19-22nmi)	195	67%			10dB @ 21.7 (40)		
Long Range (19-22nmi)	131		78.6%	11.6%		10dB @ 21.7 (40)	None
* SNR measured in the waveform bandwidth of the omni-directional hydrophone level.							

Table 4.F - In-Situ vs. Off-Line Performance

The second ACOMMS ATD demonstration was conducted in shallow and deep water in the NBOA off the coast of New England in December 1998. The shallow-water depths ranged from 183-457m and the deep-water depths ranged from 762-1829m. These locations were selected to cover a variety of propagation environments including the difficult shelf area, where transmissions up or down the wedge created by the shelf break were expected to be the most difficult to decode based on experience from the previous year. The transmission frequency used for this demonstration was the MF carrier center frequency of the SSN. The RV and SSN were used to both transmit and receive MF PSK signals.

This RV to SSN test yielded a challenging set of transmissions recorded under a variety of propagation conditions at the MF frequency regime. During each experiment, multi-channel data was recorded continuously to provide a direct record of all received waveforms. The major obstacle to robust in-situ performance was again the multipath complexity.

An improved probe was transmitted a fixed time prior to each packet instead of the 13-bit Barker sequence used the year before to better enable packet detection and synchronization. Typical channel responses from the shallow and deep sets are shown in Figure 4.7. The deep-water responses run down the left column of the figure while the shallow-water responses are in the right column. These responses are derived from matched-filtering the probe signals on one of the receive channels. Within these figures are responses (1) of varying complexity and length, some sparse and others continuous, and (2) of greater complexity than those encountered during NBOA97.

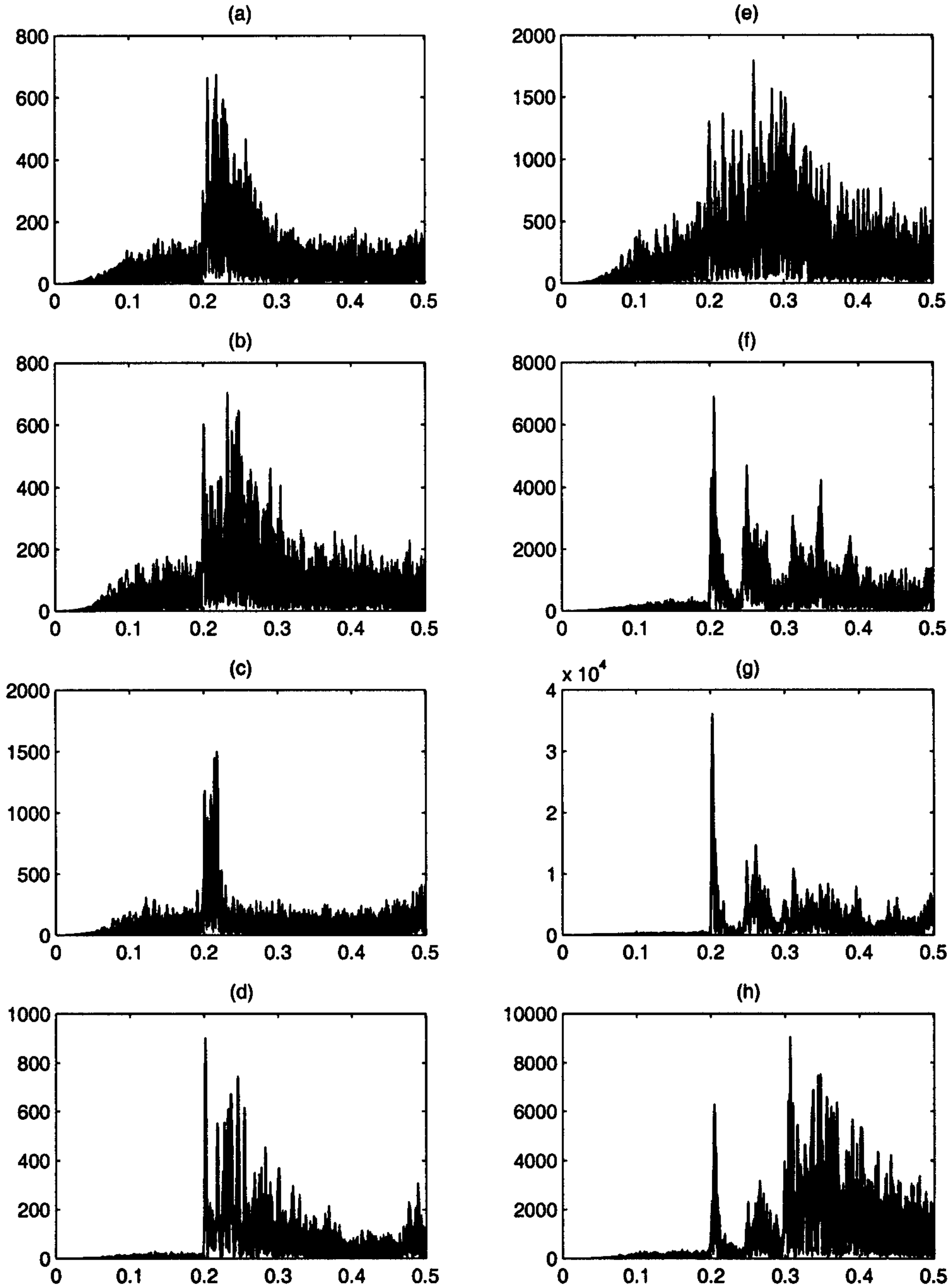


Figure 4.7 - NBOA98 Deep (a-d) and Shallow (e-h) Water Channel Responses

During the test, data was processed in-situ using a VME modem system developed by BAE SYSTEMS and using off-line Matlab code that faithfully represents the real-time VME modem processing. Results of this in-situ processing were used not only to help setting initialization parameters, but also to provide for the real-time transfer of information between the two platforms.

The off-line processing performed by WHOI, BAE SYSTEMS, and NUWCDIVNPT showed that QPSK, BPSK, and low-rate BPSK modulation types all performed well in deep water. Only BPSK and low-rate BPSK were reliable in shallow water.

Event	Rate	Ranges (nmi)
Bravo	Medium	3-.8 – 11.9
	Low	
Charlie	Medium	30.8 – 32.8
	Low	
	Medium	16.2 – 28.1
	Low	
Foxtrot	Medium	8.1 – 18.4
	Low	
Golf	Medium	4.9 – 10.8
	Low	

Table 4.G - Off-Line BPSK Performance for FY98 – MF

Different inner coding schemes were used with the two BPSK signaling types in order to explore possible performance improvements using lower rate, methods. The low rate code had three times as much coding as the mid-rate code. In order to take advantage of the gain available from these codes, a modified equalization structure was used that performed soft-decision decoding of the received symbols within the equalizer.

The first of two at-sea demonstration for the final year of the ACOMMS ATD was a MF test between a submarine and a surface ship, performed in shallow and deep water off the coast of Southern California in April 1999. This test was conducted as part of the Fleet Battle Experiment (FBE) ECHO and Limited Objective Experiment 99-3. The shallow-water site was located approximately 75km west of San Diego and the deep-water site approximately 225km southwest of San Diego. The shallow-water site depths ranged from 366-1097m and the deep-water depths were greater than 1829m. The transmission frequency used for this demonstration was the MF carrier center frequency of the SSN.

Low, medium, and standard rate links were established out to at least 65km in both deep and shallow water. A continuous communications link between the SSN and the DDG was maintained throughout the ranges of interest. Significant amounts of text, image, voice, and JMCIS (Joint Maritime Command Information System) messages were successfully decoded in real-time and/or processed off-line.

Throughout the course of the demonstration, data was processed in real-time on each platform using a VME modem system for the medium-rate BPSK and standard QPSK packets. The low-rate BPSK packets were processed in quasi-real time using an off-line system. Fully robust in-situ performance was at times hindered by greater than expected Doppler, the lack of an automated tap placement algorithm on the VME modem, noisy channels, and/or complex multipath.

Once again, the improved probe was transmitted a fixed time prior to each packet to enable packet detection and synchronization. Typical channel responses from the shallow and deep sets are shown in Figure 4.8. The deep-water responses run down the left column of the figure while the shallow-water responses are in the right column. These responses are derived from matched-filtering the probe signals on one of the receive channels. Within these figures are (1) responses of varying complexity and length, some sparse and others continuous, (2) deep-water responses with distinct main arrivals, and (3) some shallow-water responses of greater complexity than those encountered during NBOA97.

The post-test analysis of the data conducted by WHOI, BAE SYSTEMS, and NUWCDIVNPT served to mitigate most of the issues affecting in-situ performance and demonstrated the results achievable with automated parameter and algorithm selection techniques.

The low-rate BPSK packets were processed off-line both in-situ and in the lab using not only the embedded ECC as described previously but also (i) with a fully-automated filter tap placement algorithm and (ii) with Doppler tracking throughout the entire packet. The tap placement algorithm placed feedforward taps based on the distribution of energy in the estimate of the impulse response and feedback taps based on thresholding. The embedded ECC, automated filter tap placement, and Doppler tracking algorithms were not yet implemented on the VME modem.

The medium-rate BPSK and standard QPSK packets were processed in-situ using the VME modem. The VME modem did not have the embedded ECC implemented and also required user intervention to manipulate the number of feedforward taps. In addition, the VME modem did not track Doppler throughout the packet, making only a single estimate at the beginning of the packet.

These packets were processed off-line in the lab using a fully automated filter tap placement algorithm that differs from the one used to process the low-rate BPSK packets. This technique is based on taking advantage of the interaction between the expected minimum mean-squared error optimal feedforward and feedback filters, rather than attempting to optimize over each separately [Lopez99]. One limitation of this automated tap placement approach is that the feedforward support is centered on the peak of the impulse response estimate, thereby not directly addressing the overall energy distribution. Again, as with the VME implementation, Doppler tracking was not used to process these packets off-line. It is expected that with the inclusion of Doppler tracking certain performance gains will be achieved for these packets.

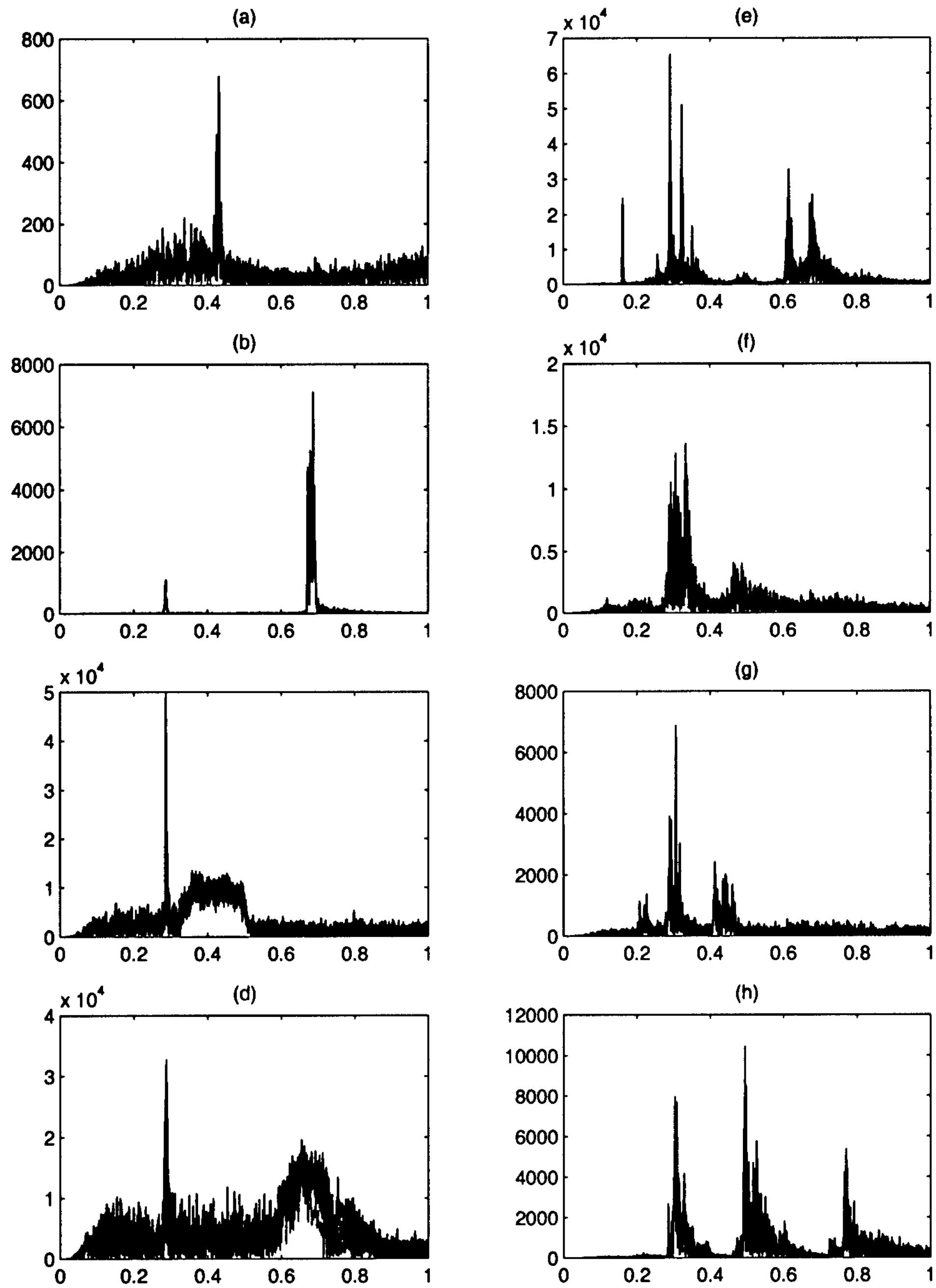


Figure 4.8 - FBE Echo Deep (a-d) and Shallow (e-h) Water Channel Responses

The final at-sea demonstration of the ACOMMS ATD, as well as the second at-sea demonstration for 1999, was performed in shallow and deep water off the coast of Lanai, Molokai, and Kahoolawee Islands in the Hawaii Operating Area in May 1999. The shallow-water site was located 28km south of Lanai and the deep-water site 93km south of Oahu. The shallow-water site depths ranged from 183-732m and the deep-water depth was approximately 4572m. The transmission frequency used for this demonstration was the center frequency of the SSN's prototype HF array. The RV and SSN were both used to transmit and receive HF ACOMMS signals.

Low, medium, and standard rate links were established out to at least 4.6km in both deep-and shallow-water. A link between the submarine and the RV was maintained throughout the ranges of interest. Significant amounts of text, image, and voice messages were successfully decoded in real-time and/or processed off-line.

As in the FY99 MF demonstration, throughout the course of the FY99 HF demonstration, data were processed in real-time on each platform using a VME modem system for the medium-rate BPSK and standard QPSK packets. The low-rate BPSK packets were processed in quasi-real time using an off-line system. Again greater than expected Doppler, the lack of an automated tap placement algorithm on the VME modem, noisy channels, and/or complex multipath affected performance.

Once again, a linear frequency-modulated probe was transmitted a fixed time prior to each packet to enable packet detection and synchronization. Typical channel responses from the shallow and deep sets are shown in Figure 4.9. The deep-water responses run down the left column of the figure while the shallow-water responses are in the right column. These responses are derived from matched-filtering the probe signals on one of the receive channels. Within these figures are responses of varying complexity and length. The embedded ECC, automated filter tap placement, and Doppler tracking algorithms were not yet implemented on the VME modem nor in the off-line processing for processing of the medium-rate BPSK and standard QPSK HF data.

For both at-sea demonstrations during FY99, the low-rate BPSK packets provided continuous connectivity across environment and platform. This robust fallback waveform worked over all combinations of platform, Doppler, and environmental scenarios. It still provides a substantial gain in data rate over existing systems. The medium-rate BPSK packets provided consistently good connectivity in the deep water while the standard QPSK demonstrated the least amount of performance. The highest data rate (i.e. standard) signaling was possible under certain environmental conditions such as deep water or specific ranges where the multipath complexity and SNR were not issues.

The consistent performance and performance shortfall evident throughout the data sets indicates that quantitative bounds on the mean-squared error exist for shallow water and deep water, respectively. A majority of the performance bounds measured relate primarily to multipath complexity given the data is not SNR limited (except for certain instances in deep water) and the performance is invariant to differential Doppler.

The medium-rate and standard packets were processed identically. The difference between these two and the low-rate packets is not only in the power of the coding but also in the demodulation and pre-processing algorithms.

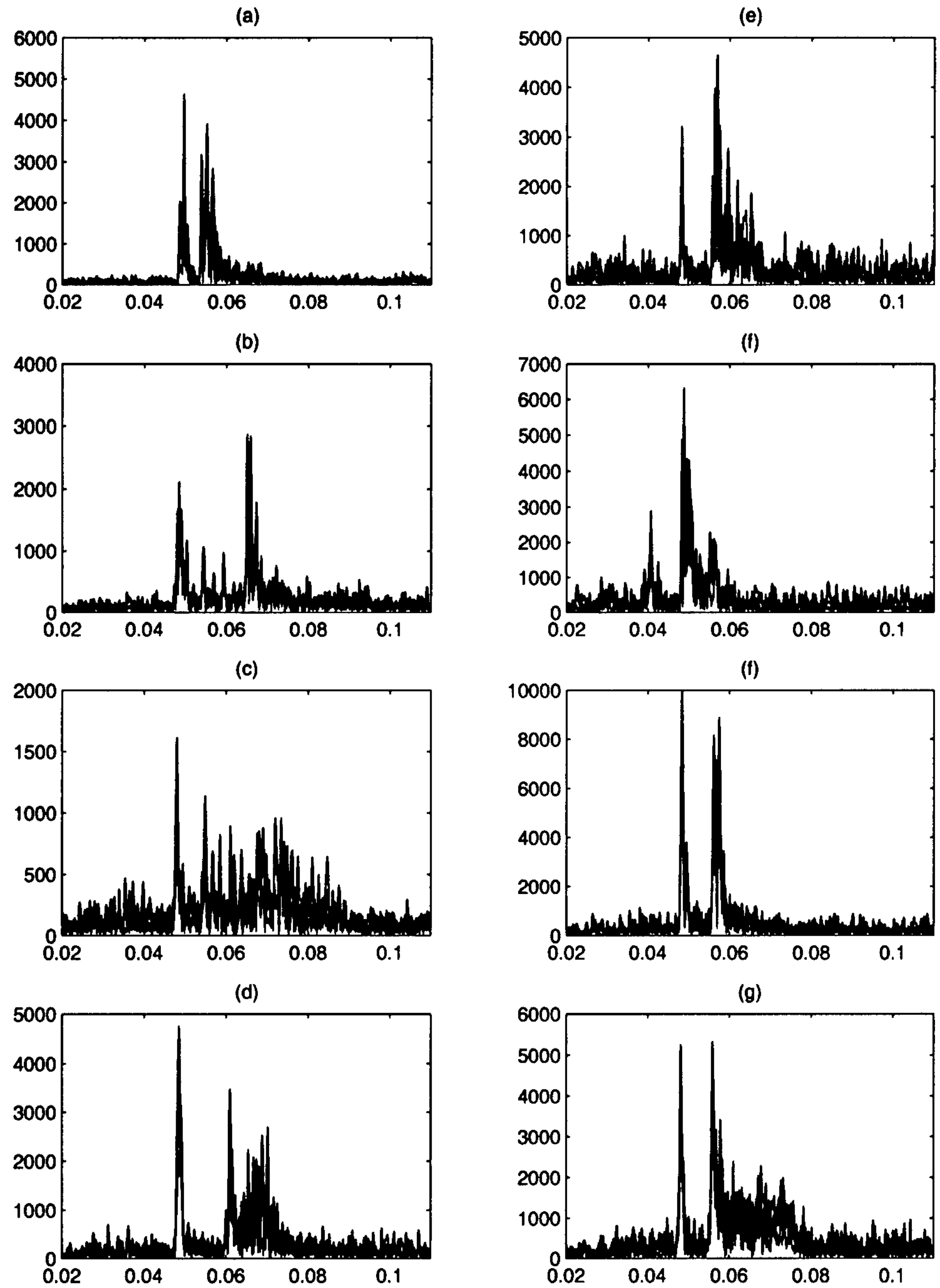


Figure 4.9 - HI'99 Deep (a-d) and Shallow (e-h) Water Channel Responses

4.7. *Alternative Modulation/Demodulation Approaches*

4.7.1. Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing (OFDM) is a technique for achieving bandwidth efficient frequency-division multiplexing with either a coherent or non-coherent receiver. OFDM allows the symbol period to be extended, in order to minimize the effects of inter-symbol interference.

OFDM requires that the transmit band be separated into N subbands, with carriers spaced at precisely $1/NT$ Hz, where $1/T$ is the bandwidth of the entire transmit band. If the subbands are spaced correctly at $1/NT$ Hz, and the symbol's time-domain windows are rectangular, then the frequency response of each subband has a sinc function shape, which has a null at every other subband's carrier frequency. This feature makes the bands orthogonal to one another and allows the bands to be placed very closely together.

If a single-carrier system transmits one symbol every T seconds, then the equivalent OFDM system will transmit N symbols in N subbands every NT seconds. Unfortunately, the orthogonality of the subbands of the OFDM system is violated at symbol boundaries, so that ISI is present at these boundaries. This can be overcome by introducing a cyclic prefix, which is a temporal guard band of a particular set of well-studied sequences attached to the beginning of each symbol. All ISI is encountered during this prefix. If the cyclic prefix has length Δ, then the efficiency of the OFDM system is $N/(N+\Delta)$ when compared to the single carrier system. The product ΔT is typically chosen to be equal to the longest expected multipath extent.

In the case of underwater acoustics, the multipath extent may be a second or more ($\Delta T = 1$). If the single carrier symbol rate is 1000 symbols per second, then T=1ms, so Δ=1000. In order to maintain 90% efficiency, the OFDM system will require $N > 9000$, which means that symbol periods will be 9 seconds long.

Because the OFDM system uses such narrow frequency bands, it may suffer from frequency selective fading. This can be corrected by the use of error correcting codes that span across all the symbols transmitted during a single time interval. Since single carrier systems which are subject to large amounts of ISI require error correcting codes over several symbol periods to combat errors induced by ISI, this requirement for error correcting codes probably does not present any additional overhead. Theoretical analyses of coded OFDM, or COFDM, are presented in [Davies98a] and [Davies98b]. While the promise of COFDM to combat multipath is described, difficulties arise due to the severity of acoustic Doppler in the underwater channel. The authors conclude that spatial processing is necessary to reduce the effects of Doppler.

4.7.1.1. Multi-frequency orthogonal chirp keying

[LeBlanc96] has developed an acoustic modem that relies on encoding a signal in the frequency domain using many narrow-band phase or amplitude-encoded chirp pulses. Each chirp pulse is uniquely situated between other chirp pulses within the frequency band. Each bit to be transmitted is translated into a combination of two adjacent chirp spectra that are assigned based on the binary message to be sent and the packet number. If a bit is a 1, the second spectrum is set to be all zeros, while if a bit is a 0, the first spectrum is set to be all zeros. The received bits are recovered by comparing the magnitudes of the two spectra. Extra pairs of chirp spectra are transmitted for labeling and parity checking.

A modem containing a single TI320c31 DSP and employing this scheme was tested at sea. Each chirp pair was allocated a bandwidth of 150Hz at center frequencies ranging from 8 to 16.4kHz.

32 packets of 28 chirps were tested with both phase and amplitude modulation. MFSK chirp pulses were transmitted at a rate of 2000bps at ranges of 100m, 500m, and 1000m. Performance was affected by the wind at a range of 100m, though not at farther distances. At 1000m, performance degraded at SNRs near 10dB. DPSK chirp pulses were transmitted at a rate of 4000bps, but more errors were detected.

4.7.2. Equalization via System Identification

Recently, researchers at the Harbor Branch Oceanographic Institution in Florida suggested an alternative approach based on system identification [Caimi98a, Caimi98b]. The channel impulse response is explicitly estimated and the transmitted symbols are then recovered from the received signal via deconvolution. The overall approach is termed equalization via system identification or EQSID. EQSID is conceptually different from the DFE approach and could potentially provide improved performance in the underwater environment. Specifically, Caimi, et al. claim that EQSID will require fewer coefficients than the DFE, is not subject to accumulated decision errors, and avoids the difficulties associated with deep spectral nulls in the frequency response of the underwater acoustic channel.

EQSID divides the problem of equalization into two blocks: channel estimation and deconvolution. To characterize a channel, Caimi relies on an adaptive channel model. The model is adapted in response to an error criterion based on the difference between the received transmission and an estimated received signal as computed using the model. During training, the estimated received signal is obtained via the convolution of the training symbols with the computed channel response as specified by the model. In decision-directed mode, the estimated channel response is convolved with actual decisions to produce the estimated received signal.

While Caimi acknowledges that nonlinear or complicated physical-based models can be used with this approach, he suggests the use of a linear moving average (MA) model to estimate the channel. Using the model, the length L estimated channel impulse response $\hat{\mathbf{h}}(n)$ is computed. To update the model, an estimate of the received signal $\hat{y}(n)$ is computed by convolving $\hat{\mathbf{h}}(n)$ with the vector of data symbols $\mathbf{s}(n) = \left[\hat{d}(n-L+1) \ \ldots \ \hat{d}(n)\right]^T$, where $\hat{d}(n)$ is the symbol decision at time n. The update is accomplished using either the recursive least square (RLS) algorithm or the least mean squares (LMS) algorithm to minimize the error $e(n) = |y(n) - \hat{y}(n)|$, the magnitude of the difference between the actual received signal $y(n)$ and the estimated received signal at time n.

A deconvolution block follows the channel estimation block. If the channel response is minimum phase, the transfer function $G(z)$ of the deconvolution block is the inverse of the transfer function of the channel $H(z)$, i.e. $G(z) = 1/H(z)$. For non-minimum phase channel responses, the deconvolution transfer function takes on a more complicated form. While deconvolution can be attempted in either the frequency domain or the time domain for a time-invariant channel, time domain processing allows for symbol-by-symbol updates of the time-varying underwater channel. If $G(z)$ is properly chosen, the deconvolution block should produce the symbols that were transmitted. Because the deconvolution cannot be done perfectly, the output of the deconvolution block is processed to make symbol decisions.

The following is a block diagram of the EQSID method in decision directed mode:

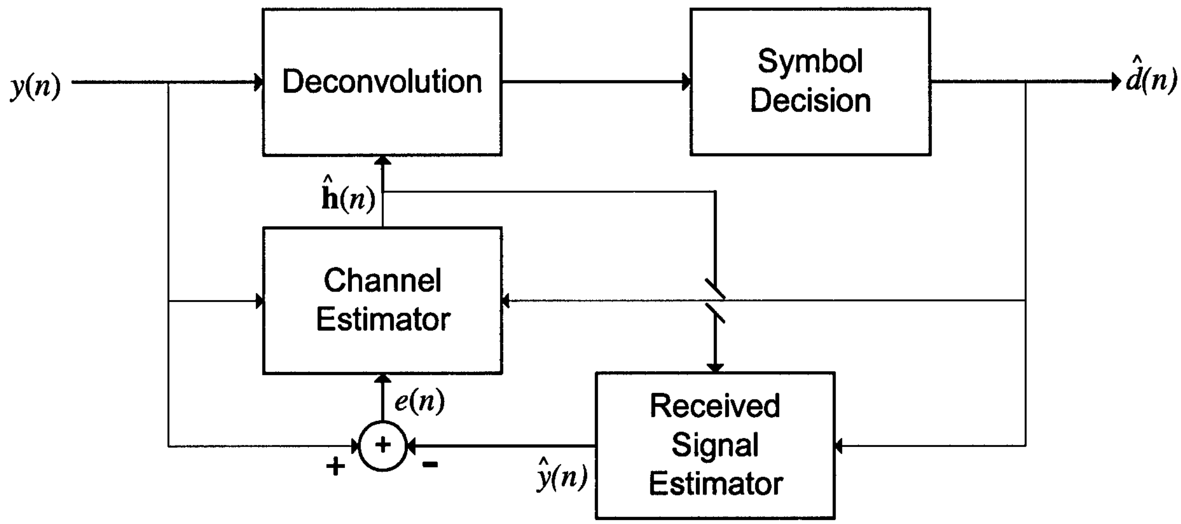


Figure 4.10 - Block Diagram of the EQSID Method in Decision Directed Mode

An experiment demonstrated the suitability of this approach to the underwater channel. Signals were transmitted through shallow water at ranges of 200 to 400m at a carrier frequency of 50kHz using a bandwidth of 20kHz. Rates of 10kbps to 20kbps were tested with error-free receptions achieved at 200m and an error rate of less than 10^{-2} at 400m.

A different type of system identification approach was recently suggested in [Stojanovic99]. Instead of a deconvolution block, this method uses a decision feedback equalizer. The channel estimator is used to compute the parameters for the DFE. The channel estimate is optimally sparsed by selecting out only those sections of the estimate with significant energy. The sparsed estimate is then used to cancel out post-cursor ISI prior to equalization. This is used in conjunction with spatial diversity processing to reduce the complexity of the algorithm. This was tested on data from the Continental Shelf off the coast of New England in 1997 at depths from 100 to 200m. A vertical 8-element array with hydrophones spaced 0.027m apart was used to receive QPSK over a range of 3km. Error-free receptions are shown for data transmitted at a rate of 10kbps using a carrier frequency of 25kHz.

4.7.3. Phase-Encoded Frequency-Hopped Signaling

Phase-coherent digital communications systems require complicated receivers to overcome the intersymbol interference caused by the time-varying multipath propagation of shallow water acoustic channels. However, many of these channels feature "sparse" impulse responses [Freitag96] in that the signal arrives in distinct clusters separated by time periods in which little signal energy arrives at the receiver. If the phase-coherent message is divided into multiple "mini-packets" such that the mini-packet duration is shorter than the time span between the distinct multipath clusters, then the receiver configuration can be simplified considerably [Howe92] as long equalizers and/or dedicated interference suppressors are no longer required.

The short packets may be frequency hopped so that a given frequency is vacated until subsequent multipath arrivals have died out. Each mini-packet is decoded using a maximum likelihood decoder. For highly dispersive channels the system reduces to a non-coherent frequency-hopped modulation technique [Gillespie97, Edelson98].

The motivation for the frequency-hopped modulation technique is multifold. Primarily, the goal is to define a technique that can be supported by a low complexity receiver and that can provide medium data rates relative to the system bandwidth if so desired. Additional design goals are to be detection resistant by operating at low signal-to-noise ratios (SNR) without channel probes and/or training data, to provide the flexibility to communicate across a wide variety of environments, to support network applications, and to mitigate the effects of channel- and frequency-selective fading.

The relatively short packet duration suggested above is amenable to frequency hopping and the transmitter dwell time is governed by the minimum time span between clusters. The use of differential frequency hopping allows several "next hops" chosen from a larger hop set. The one taken is determined by the data being transmitted. For a given data symbol, X_L, and previous hop frequency, F_{L-1}, the frequency of the next hop is defined as

$$F_L = G(F_{L-1}, X_L)$$

where G is known only to the transmitter and to the intended receiver. This frequency-encoding scheme lends itself well to a network application, where each node in a "neighborhood" will be assigned a unique initial frequency and hop-encoding scheme. This aids in waveform conflict resolution and in identifying the transmit node by the receiver.

The frequency-hopped signaling method has two operational modes. The features of the available transmission channel and/or the demands of the mission determine the exercised mode. The concept is not to adaptively alternate between the two modes, but to offer the flexibility to operate over a broad range of environments and throughput requirements.

For channels having a resolvable multipath structure, this method would burst a phase-modulated mini-packet with duration such that multipath interference is avoided. At the conclusion of the burst, the next packet would be transmitted in a different, independent frequency sub-band (as determined by G), thereby preventing intersymbol interference. This phase-encoded modulation and demodulation exploits the channel capacity often offered by the presence of resolvable first multipath arrival via direct-path, ducted, or specular propagation. For dispersive channels not supporting the coherent "mini-packet" bursts, the idea is to revert to a non-coherent frequency-hopped spread-spectrum (FHSS) signal with data-driven hopping patterns.

This multi-mode frequency-hopped signaling concept provides a self-consistent structure to exploit geographic variability. In addition, the coherent mini-packet mode provides a means to increase data rates which, in the case of fixed message sizes, may provide for a decrease in the exposure time of the system. The differential frequency hopped encoding scheme works to ensure that false alarms die out quickly and that detections from other nodes using different encoding algorithms in an acoustic network also die out.

4.7.3.1. Coherent Mode

The coherent realization of this modulation technique is predicated on the existence of a clear gap between two multipath arrivals in the channel impulse response. This gap allows the use of a simplified channel equalizer. To mitigate the inter-symbol interference caused by additional multipath arrivals, an entire mini-packet of data is transmitted before the next multipath arrival can begin to corrupt packet symbols at the receiver. Each packet is encoded by a phase shift keying system, with information symbols contained completely within the chip. The chip length is driven by the multipath/channel impulse response. The minimum hop distance becomes a function of the mini-packet symbol rate and expected Doppler spread of the channel. The chip rate is selected to ensure that the channel has cleared within a particular frequency band prior to the next chip being sent. The individual chips are now spread in frequency and have data driven content.

With multiple receive hydrophones, both temporal and spatial diversity can be exploited. When the mini-packets are time-aligned, the micro-multipath structure for each multipath component will be different. Consequently, the phase structure imposed by the channel on each multipath component will differ. Assuming that there are N diversity data vectors of equal SNR denoted $\mathbf{x}_1,\ldots,\mathbf{x}_N$ and K codewords denoted $\mathbf{c}_1,\ldots,\mathbf{c}_K$, the maximum likelihood receiver

$$\hat{k} = \underset{k}{\operatorname{argmax}} \sum_n |\mathbf{c}_k^H \mathbf{x}_n|^2$$

accounts for this variability in phase structure.

4.7.3.2. Non-Coherent Mode

In the event that it is known *a priori* that the dispersive channel cannot support the coherent mini-packet bursts, this signaling technique reverts to a non-coherent frequency-hopped spread-spectrum (FHSS) signal with a differential frequency hopping modulation pattern whereby all the information is contained in the hopping sequence (in frequency). The potential detection and intercept resistant (LPI/LPD) properties of this waveform are discussed in Section 4.9.

An advantage of this non-coherent method is that hop-sequence dropouts can be recreated by "backtracking" through the hop tree using knowledge of the symbols decoded before and after the dropout sequence. This is analogous to the transition path of a knight on a chessboard, which is determined by the constraints placed on an individual move and can be deduced even if every move is not observed. Similarly, the function G constrains the signal hop sequence, allowing missed hops to be recreated.

4.7.3.3. Experimental Results

This clearing property of certain underwater acoustic channels was first utilized for communications on tests that were conducted over a 13m deep, 100m range shallow-water channel and utilized a 2-DPSK waveform [Howe92]. A single hydrophone was used as a receiver while the transmitted signal was a 10kbps burst centered at 50kHz. Between two and five bits were contained in each mini-packet, with a new burst transmitted every 3ms, to achieve an maximum data rate of 1.6kbps. Bit error rates were small until the number of bits per mini-packet reached five, at which point the burst was corrupted by multipath. The system failed to function at 250m because the multipath arrived with a delay of less than one bit (in time).

"Proof-of-concept" frequency-hopped mini-packet data were collected as part of an at-sea experiment in the Narragansett Bay Operating Area of the Continental Shelf in June 1997 [Edelson98]. A 16-hydrophone vertical receive array was positioned at 40 06.0N, 70 58.8W at a depth of 69.5 to 73 meters in one-meter seas. An 8-11kHz UQC projector was positioned at a range of approximately 1.8nmi from the receiver and at a depth of 3 meters. The simple hop sequence of [8.5 9.25 10.0 8.75 9.5 10.25 9.0 9.75 10.5] (in kHz) was used for demonstration purposes.

Under these conditions, the channel impulse response was sufficiently sparse to support coherent mini-packet signaling with gaps between significant multipath clusters on the order of 15ms.

Both coherent and non-coherent signals were successfully decoded at effective data rates between 100bps and 500bps using the maximum likelihood receiver described above.

4.8. Multi-User Communications

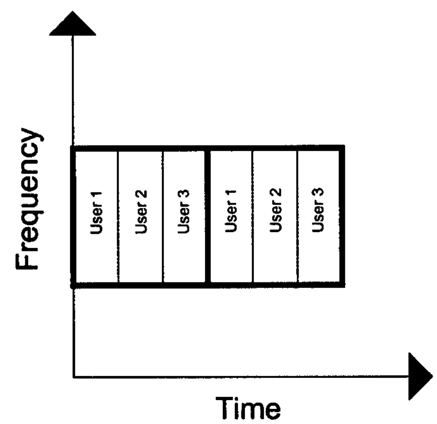
A multiuser communications system involves the transmission and reception of communications signals from several users. Ideally, a multiuser system will allow simultaneous signal transmissions, enabling efficient use of available bandwidth. In radio frequency (RF) communications, several different protocols have been developed to facilitate multiuser communications, including time-division multiple access (TDMA), frequency-division multiple access (FDMA), and code-division multiple access (CDMA). These RF protocols, however, are not easily applied to underwater acoustic communications. In some cases, they can be adapted to operate in the underwater acoustic environment. In this section, these protocols will be defined and relevant underwater adaptations will be described.

The applicability of existing network technology to the underwater acoustic environment is explored in [Sozer00]. Network topologies applicable to underwater communications include a centralized network, where communications is controlled by a hub, and multi-hop peer-to-peer networks, which are formed by establishing links only between neighboring nodes. Of the multiple access methods discussed, TDMA and FDMA are found to be less promising than CDMA for use in the underwater channel. Media access protocols suitable for an underwater network include Carrier Sense Media Access (CSMA) for low throughput scenarios and Multiple Access with Collision Avoidance (MACA) or its variant MACAW. Automatic Repeat Request (ARQ) methods with potential for use in underwater communications include GoBack N and Selective Repeat Protocol. Routing is feasible using ad hoc network routing protocols, since typical routing schemes are based on the shortest path algorithm with stationary nodes. Suitable ad hoc routing protocols include Destination Sequence Distance Vector (DSDV), Temporally Ordered Routing Algorithm (TORA), Dynamic Source Routing (DSR), and Ad hoc On-demand Distance Vector (AODV).

A design example is provided that incorporates a large number of nodes in a shallow water environment with depths between 50-100m. The nodes are mounted on the bottom at distances of up to 10km. The desired transmission rate is 100bps from each node, using a frequency band of 8-15kHz. Packet sizes are 256 bits, and each node is allowed to transmit up to five packets per hour at half-duplex. Relaying is utilized to minimize energy consumption, which is the dominant constraint. Both FDMA and CDMA are considered for multiple access. The media access protocol is based on MACA, and Stop & Wait ARQ is used. The overall network routing is ad hoc, so initialization is required to generate neighbor tables indicating each node's neighbors and the quality of the links between neighbors. The master node generates a routing tree based on the neighbor tables, and then assigns new routing as needed to the other nodes. The design was tested via simulation. Peak throughput is 10^{-6} packets per second. The design is expected to be tested as part of SEAWEB over the next few years.

4.8.1. Time-Division Multiple Access

The idea behind TDMA is to divide the time domain into frames and allocate a unique time slot within each frame to a different user. Each user can make full use of the frequency band of the channel within the assigned time slot. A master node synchronizes transmission frames. For N users, each frame is divided into N time slots that are appropriately assigned. The master node sends a frame synchronization signal to all users. Each user waits for the designated time slot within the frame and then transmits a message. When all users have transmitted, another frame sync is sent, and the process repeats. The following diagram illustrates a time-frequency viewpoint of two frames of a three-user channel allocation:



The key to the success of the TDMA protocol is the proper synchronization of each frame in time. If the transmissions from Users 1, 2, and 3 do not occur during the designated time periods of each frame, the receiver will not be able to distinguish the users. In the underwater acoustic environment, synchronization in time is difficult due to the occurrence of long propagation delays that can be many times the duration of a packet.

4.8.1.1. Time-Domain-Oriented Multiple Access

A variation on TDMA called Time-Domain-Oriented Multiple Access has been proposed [Hou99]. In this protocol, packets are interleaved through the channel so that no user receives a relevant packet while simultaneously transmitting. This is accomplished by fixing the length of time between successive packet transmissions to be an integer fraction of the round trip propagation times between users. For example, suppose User 1 initiates transmission by sending an initialization packet. User 2 receives the packet and immediately sends an acknowledgement packet. Upon receipt of this acknowledgment, User 1 now has knowledge of the propagation delay between Users 1 and 2. User 1 then designates the duration of subsequent packets for both users such that the propagation delay divided by the sum of the packet duration is an integer. This eliminates the need for a master node and the division of the time domain into frames, as times for transmission are dependent on actual propagation delays rather than arbitrarily fixed time slots. While this circumvents difficulties in time synchronization, it also leads to a substantial decrease in network throughput when there are several users.

4.8.1.2. TDMA with a Network Controller

In this concept, a modem with data to transmit must request a transmit time slice from the network controller. The controller responds with a permission to transmit, at which point the modem can begin transmission. Although this scheme requires use of at least three frequency bands (one for the request to transmit, one for permission to transmit, and at least one data channel), and does require that all nodes are able to communicate with the network controller, it can schedule transmissions so that the data channel is effectively used. This concept was pursued by WHOI after its initial success with collision detection multiple access (CoDMA) systems, and was called the Acoustic Local Area Network (ALAN). It was tested in the deep waters of the Monterey Canyon in 1994.

The deep water ALAN acoustic communication protocol was designed to allow efficient communication between many near-bottom sensors and a single surface or bottom network controller and designed to minimize idle time due to propagation delays and to provide high energy efficiency at the ocean-bottom sensors. This section explains the operation of the ALAN protocol through an illustration of packet flow through time and frequency, as shown in Figure 4.11. A more technical description is in [Catipovic93].

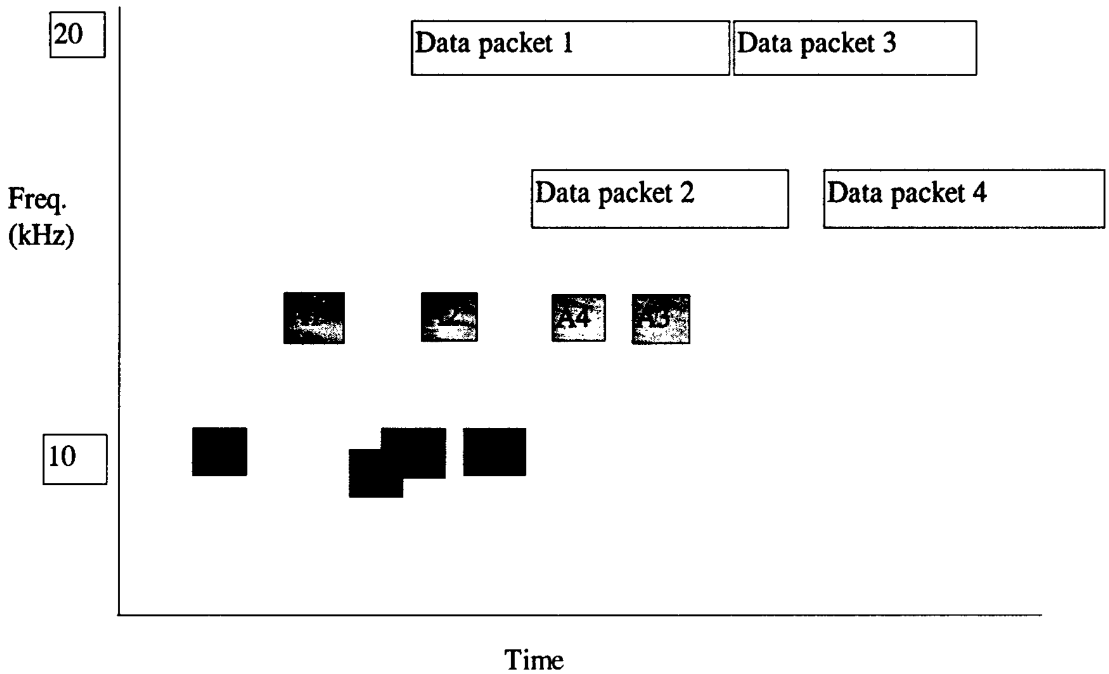


Figure 4.11 - Time-Frequency Representation of the ALAN Protocol

An ocean-bottom modem initiates the transmission process by asking for a reservation through a common request channel, and may remain in low power "sleep" mode until that time. Modems access the request channel asynchronously, simply asking for a reservation to transmit a certain number of data packets. Since each modem is unaware of a possible reservation request by another sensor, the possibility of packet "collisions" exists on this request channel. A "collision" means that two or more packets were received in the same frequency band at the same time. In the illustration above, requests have been made asynchronously by modems 1 through 4, with a collision of request packets from modems 2 and 3. ALAN used active interference rejection techniques to resolve the collision and thereby correctly receive the colliding requests [Brady94, Curtin93, Zvonar96].

The network controller decodes the reservations and replies to the requesting modems through an acknowledgment channel. Each ocean-bottom modem listens on this channel for an acknowledgment packet containing information on when, how, and how much data can be transmitted. The acknowledgment packets are preceded with unique modem password IDs so all the modems do not have to needlessly power up and decode all acknowledgment packets.

After receipt of the acknowledgment, the modem transmits immediately on the assigned data channel. The timing of the acknowledgment packets, in effect, synchronizes data flow on the data channels. In the example, acknowledgments A1 and A2 are issued immediately, since the controller has slots available on data channels 1 and 2. A3 is timed such that the round trip delay to modem 3 causes the data reception to start immediately following the end of transmission from modem 1. The controller is assumed to know the round trip travel time to the modems. Similarly, A4 is timed to allow data from modem 4 to follow that of modem 2. The small "quiet time" gap between data packet 2 and data packet 4 reflects the controller's uncertainty of the round trip delay to modem 4. The controller underestimates the delay to minimize collisions. However, if a collision is detected, the data is still recoverable using the collision resolution algorithm [Travis94]. Note that in the example, the order of the acknowledgment packets A3 and A4 is reversed from the order of the request packets or data sessions, since modem 4 is more distant than modem 3. Requests are scheduled sequentially, and a data channel is selected based on modem range and current channel quality estimates in order to minimize packet delay. If the network is congested and the controller has no channels available for a requesting modem, it does not reply, and the modem will re-request channel access after a time-out period.

Upon reception of the request packets, the controller resolves request packet collisions, schedules the requests on the available data channels, and notifies each packet of their reservation one round-trip propagation time before the expected data packet reception. Request packets are very short (<10 bytes) in order to reduce the probability of packet collision.

Upon reception of the data packets, the receiver updates the estimate of round-trip propagation time. The Forward Error Correction Automatic Repeat reQuest (FEC-ARQ) error correction protocol ensures error-free data by requesting retransmissions of uncorrectable corrupted data sequences. [Catipovic93, Talavage94] Each ocean-bottom modem retains the most recently transmitted packet in the event of a transmission error requiring partial retransmission. The holding period for each packet is designed to make the chance of lost data vanishingly small.

4.8.1.3. Shallow-water ALAN

The above deep water ALAN in Monterey canyon relied on the ability of each network node to reach the controller most of the time. Thus the protocol is a "star" protocol, where each node communicates only with the controller and can afford to neglect all its neighbors. This assumption is quite valid in the Reliable Acoustic Propagation (RAP) "cone" encountered in deep water. In Monterey Canyon, with the network controller at a depth of about 1000 meters, the RAP cone extended some 6-8km from the controller, encompassing most of the canyon area.

In shallow water, the RAP cone extends some 3-4 water depths from a surface node, and is not a useful feature. Then the network needs to rely on the time-variant, high multipath acoustic propagation encountered in shallow water. In these conditions, a reliable communication link between any two points in the network could not be guaranteed.

The key assumption in the design of a shallow water ALAN concept is the inability to guarantee a viable acoustic path directly from a network node to the network controller. A follow-on assumption is that the newly-deployed network node does not know its location relative to the network and does not know the nodes or controller(s) in the network with which it can directly communicate. An implicit assumption is that a newly-deployed node can communicate directly with at least one network member. Given the above constraints, an adaptive network initialization and path definition protocol is required [Talavage94].

An early demonstration of multipoint networking was the Arctic local area network, deployed in March and April 1994 approximately 370km north of Prudhoe Bay. The network consisted of 6 nodes deployed in a 10km x 10km area surrounding an Ice Camp.

The shallow-water ALAN protocol concept is based on the premise that node-to-node connectivity cannot be guaranteed over multi-day periods, but that node-to-node links are stable over a several minute duration. The ALAN nodes maintain a "neighbor" list, which includes all relevant network initialization and status information. Specifically, each node maintains:

1. A list of all "reachable" neighbors, defined as nodes that were communicated with at least once.
2. A list of node locations for all reachable nodes, indexed with respect to the "master node," i.e. the node with the surface-piercing RF connection.
3. The history (mean and variance) of the power settings used to reach each of the neighbors. When initiating a new transaction, the node initializes the power setting to one standard deviation above the mean power setting.
4. Channel multipath structure to and from each neighbor.
5. Neighbor node ID and related protocol and packet design information.
6. Status (busy or free) for each of the neighbor nodes and related channel utilization.

Each shallow water ALAN node would have three bi-directional data channels and a single command and control (C^2) channel. The data channels nominally occupy 5kHz of bandwidth centered at 15kHz, 25kHz, and 35kHz. The C^2 channel occupies the 8-10kHz band.

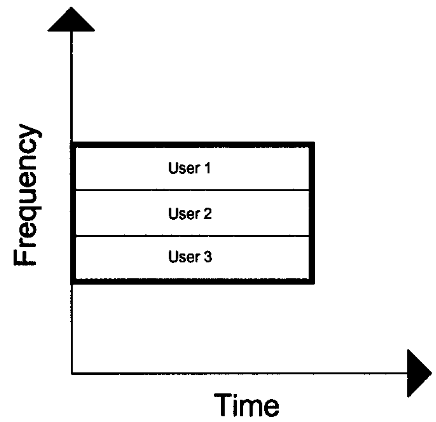
When a node desires to communicate, it (the "originator" node) broadcasts a request for a virtual link to the network, using the C^2 channel. The virtual link request contains the length of the message to be sent and the address of the terminal network node (in most cases the master node with the RF interface). If the terminal node is a neighbor and decodes the request, it allocates two available frequencies for the transmit and receive links and telemeters that information to the originator node. The other neighbor nodes mark the selected channels as busy.

The originator node transmits data on the assigned channel and expects an end of message acknowledgment from the terminator node. If it is received, the originator and all of the neighbor nodes mark the used channels as available, and become dormant. If the end of message is not received, the originator renegotiates the channel allocation after a timeout.

If the destination node is not a neighbor of the originator, a store-and-forward protocol can be used. This is described in [Talavage94], but it has not been implemented to date on an actual in-water network.

4.8.2. Frequency-Division Multiple Access

FDMA is the frequency domain analogue of TDMA. With FDMA, a unique portion of the available frequency spectrum is allocated to each user. There need not be any time domain restrictions on transmissions. The following diagram illustrates a time-frequency viewpoint of one packet of a three-user channel allocation:



The underwater acoustic environment is not amenable to the FDMA protocol. Acoustic channels can suffer from frequency-selective fading, meaning that one or more users may not be able to communicate at any given time.

4.8.2.1. An Experimental Result

A September 1998 study known as SeaWeb'98 tested an FDMA network in Buzzards Bay in Massachusetts [Green98b, Green98c, Rice99]. This network consisted of ten telesonar modems incorporating frequency hopped spread spectrum signaling. This type of signaling is discussed in section 4.9.2. The network was deployed in 10m deep water using a binary tree topology covering, with a master node at the base, oceanographic instruments at the outlying nodes, and branch nodes in between. It was designed to transmit oceanographic data from the outer nodes to the master node, which was equipped with an RF gateway to transmit the data to a support boat and ashore to command centers.

The SeaWeb network protocol consists of three layers. Layer 1 is for signaling between a pair of modems, and allows for rates from 100 to 2400bps using 4800Hz of bandwidth. Layer 2 is intended for communication of control information, to allow for the remote setting of parameters, to change a remote modem to a different operating state, and to transmit user information. Layer 3 is provides mechanisms for performing the network protocol, and aids in routing packets throughout the network. The protocol begins when an information sequence is obtained by Layer 3, either from instruments attached to a node, internally generated test data, or a transmission that has been received from another node. The node then determines if the information needs to travel up the tree towards the master node or down the tree towards the outlying nodes. Information is then passed down the layers, and the appropriate packet transmissions take place.

The modems operated according to a preprogrammed schedule (i.e. routing information was not provided during operation), but nodes could be operated by personnel on the support boat or at the control center when desired. Network performance was found to be very reliable. The following concepts were demonstrated using this network: store and forward of data packets, transmit retries and automatic repeat request, packet routing, cell-like node grouping to minimize interference between cells, wide-area coverage, robustness to shallow water multipath and shipping noise, low-power operation with sleep modes, affordability, and remote control. A follow-up study was scheduled for August 1999, and has not yet been reported.

4.8.3. Code-Division Multiple Access

CDMA does not involve division of the time domain or the frequency domain. Instead, each user is allocated a unique spreading code for use in modulating transmissions. Spreading codes are sequences resembling noise that have very low correlation with one another, and are discussed in more detail in the chapter on covert communications. Each user makes use of the same available channel bandwidth. Users are distinguished by correlating the received transmission with the set of spreading codes in use. For example, if User 1 modulates a transmission with the sequence {1 1 1} and User 2 modulates a transmission with the sequence {1 –1 1}, the correlation of {1 1 1} with itself is three times larger than the correlation of {1 1 1} and {1 –1 1}. Thus, if a transmission from User 1 is received, correlation by {1 –1 1} will more resemble noise, while correlation by {1 1 1} will yield the message from User 1.

4.8.3.1. Selected Experimental Results

There have been a few proposed implementations of CDMA modified for the underwater acoustic environment. Modification is necessary because of extensive multipath and the severely limited available bandwidth. These factors restrict the choice of spreading codes to short sequences with higher cross-correlations. One proposed method permits two-way data telemetry between ocean-bottom nodes and a central, surface-deployed receiver in the 10-40kHz vertical acoustical channel [Brady92, Brady94]. "Central" refers to the ability of the receiver to demodulate signals from more than one transmitter. The ocean bottom nodes initiate transmissions by requesting a data channel time slot through a common narrow-band channel. Request packets from different nodes may overlap, or collide, and the system is designed to resolve such collisions.

The system works as follows: each user modulates a pre-assigned length-3 spreading sequence by a sequence of symbols and transmits asynchronously through a unique channel. The received waveforms are processed with a bank of matched filters, one per user. The matched filter output for each user contains interference from other users (known as multiple access interference or MAI) due to the partially overlapping cross-correlations. The receiver estimates the MAI through soft decisions that depend on the received signal estimates. The estimated MAI is subtracted out and the resulting signal is used together with a channel estimate to form final decisions for the data. This strategy utilizes the attribute that packets that collide usually have portions that do not contain MAI.

Evaluation of this strategy was accomplished using recorded in-water data from 1992 [Brady92]. Two modems transmitted 700-bit BPSK packets in an asynchronous manner at a range of 2000m in deep water south of Martha's Vineyard with weather at sea state 5 and 6'-8' seas. Both the transmitters and the shipborne receiver were omni-directional. Each packet was initiated by a transmitter-specific 13-bit address. This was followed by a 4-bit silent period, used to initialize decoding and channel estimation, and provide data buffering. The method exhibited success even under conditions where one user was 10dB in signal energy below the other user.

Another type of implementation employs receivers that implicitly perform adaptive matched filtering, MAI suppression, linear equalization of ISI and timing error compensation. This method adaptively determines the component of the desired signal that is orthogonal to the interference. Implementations of this type include a decentralized receiver based on a decision feedback equalizer (DFE) [Zvonar93], a decentralized minimum mean squared error (MMSE) receiver with a fractionally spaced transversal filter [Zvonar94, Zvonar97], and a centralized receiver based on a multidimensional DFE [Zvonar96].

These methods were tested in a 1993 shallow-water experiment conducted by WHOI in the Woods Hole Harbor. The range between transmitters and receiver was 750m in 18-m deep water. The geometry of the setup precluded the use of beamforming techniques for interference cancellation. The carrier frequency was 15kHz and the data rate was 2000bps using BPSK modulation over length-3 spreading sequences. The bandwidth utilized by the transmitted signals was 9-21kHz. One transmitter generated a signal at a power of 185dB re 1μPa while the other transmitted a signal that was 10dB weaker. The methods generally worked, although conditions causing severe fading led to a loss of communications, even for the stronger user.

4.8.3.1.1. CDMA with Spatial Diversity

A further variation on these adaptive methods introduces spatial signal processing techniques to improve performance. As discussed earlier with regards to coherent acoustic communications, the spatial structure of the multipath can be exploited to reduce receiver complexity. Included in this category are a decentralized receiver based on multidimensional DFE with spatial precombining to reduce complexity [Stojanovic93c, Stojanovic94b] and a centralized receiver based on multidimensional DFE with spatial precombining [Stojanovic94b, Stojanovic96b].

In the aforementioned methods, the adaptive filters are updated at the symbol rate. Another implementation incorporating spatial signal processing takes a different approach. This method consists of a receiver with a multi-input, single output array processing filter followed by a single-channel adaptive equalizer [Gray94, Preisig95, Gray97]. The array processing filter is designed to exploit the deterministic multipath component and is updated infrequently at a rate commensurate with the rate of change of the macro-multipath structure of the environment, which is far slower than the symbol rate. The adaptive equalizer is designed to track the stochastic multipath component and is updated at the symbol rate. The array processing filter allows for a reduction in the complexity of the adaptive equalizer, just like the spatial precombiner.

A recent simulation analyzed a modified CDMA approach that uses spatially and temporally adaptive receivers with spreading codes of up to length-31 [Tsimenidis99], which are longer than previously studied. Longer code lengths improve the viability of a CDMA-based underwater acoustic communications network. This method has yet to be tested in the field.

4.8.4. Other Networking Protocols

Channels for terrestrial networks are relatively stationary and have few multipath arrivals. Undersea channels have multipath arrival structures that can span several tens to hundreds of symbols, whereas terrestrial multipath typically spans only adjacent symbols. Terrestrial channel time-variability is slow enough that almost all work in the research literature assumes that the channel can be estimated once with a probe, and that it then remains constant for the duration of the transmission. Also, terrestrial channels have "rapid" feedback, which enables cooperative communication techniques such as efficient energy distribution across frequency and requests for retransmission.

The nominal speed of sound is an undersea channel 1500m/s, making TDMA scheduling and network synchronization difficult. Worse than this, it makes any feedback difficult to use, since the channel may change significantly over the propagation time delay to-and-from the receiver. Without feedback, adapting techniques from terrestrial networks (i.e. Code Division Multiple Access (CDMA), Auto Repeat reQuest (ARQ), routing optimization, etc.) is difficult. CDMA techniques, which use orthogonal spreading sequences, may suffer because orthogonality would be impossible to maintain and timing would never be precise. In addition, the severe bandwidth restrictions in the undersea environment would cause CDMA to suffer from very low throughput. Because of these challenges, alternative networking protocols have been investigated for use in the underwater acoustic channel, such as Collision Detection Multiple Access, an Acoustic Local Area Network (ALAN) for deep water, an ALAN for shallow water, and the Autonomous Oceanographic Sampling Network (AOSN).

4.8.4.1. Collision Detection Multiple Access

Collision Detection Multiple Access (CoDMA) is appropriate for low-duty cycle networks. In this approach, any modem that has data to transmit simply transmits it. The recipient is expected to acknowledge the message. If no acknowledgement is received, or if the recipient indicates decoding errors, then the modem re-transmits.

This concept was pursued and tested with ONR University Research Initiative (URI) funding at Woods Hole Oceanographic Institution (WHOI) in 1987 and 1988. It was intended to replace electrical conductors for communicating along deep-sea mooring cables from mid-water instrumentation to the surface buoy and then ashore via satellite. The project used WHOI-developed, non-coherent multitone acoustic modems (the forerunners of the Datasonics ATM acoustic modem series). They communicated at 1200bps using 8-FSK in two frequency diversity bands.

A mooring was deployed with three acoustic modems at 3000m, 1500m, and 150m below the surface. The surface modem was attached to the buoy, which also provided the digital satellite link. The modems were interfaced to acoustic current meters but also broadcasted test data once every six hours.

Since the modems only broadcasted a single packet every six hours, a simple CoDMA protocol was used in order to allow the bottomed modems to broadcast asynchronously. If a transmission was garbled, the surface modem could poll each of the three subsurface units for a retransmission. The experiment lasted for six months and the system is fully described in the Oceans 1989 proceedings [Catipovic89b].

4.8.4.2. Autonomous Oceanographic Sampling Network

The Autonomous Oceanographic Sampling Network (AOSN) is a highly integrated remote sensing and communication system concept described in [Curtin93, Schmidt96, Herold97]. It was intended for deployment in Haro Strait, British Columbia to locate and track a 1km tidal front. It consists of a network of four acoustic remote sensing moorings supplemented by a network of small AUVs. Each mooring consists acoustic modems, tomography sources, and 16-element vertical arrays arranged over a 100m aperture. In addition, each mooring has a 900MHz Ethernet radio to generate a link with a land-based repeater. The moorings were deployed at depths ranging from 110-230m. The AUVs are equipped with local sensors that transmit real-time data. The objective was to improve the mapping resolution by combining the capabilities of the distributed moorings and the AUVs.

The required functions of the network included tomography, acoustic communications between all components of the system, and real-time remote control and configuration of the system. The network is linked by the modems on the AUVs and the moorings. The moorings communicate among themselves and establish protocols for information routing in response to changes in the acoustic channel and ambient noise levels. Transmission loss among the many internodal paths is measured periodically using 28 symbol Barker codes to probe the channel. A centrally-located network controller, either one of the moorings and/or an onshore node, processes the incoming data in real-time. The moorings would emit tones at different frequencies to steer an AUV (i.e. f_1 for turn left, f_2 for turn right, f_3 for turn up, f_4 for turn down). The network sources operate with carrier frequencies between 9 and 15kHz. The practicality of the AOSN is based on the number and capabilities of the AUVs, as well as the performance of the acoustic communications in the deployed environment for both navigation and data transmission.

4.9. Clandestine Communications

Clandestine (or non-overt) communications is a particularly challenging task in the underwater acoustic environment [Park86]. Transmissions are subject to frequency-selective fading and distance-dependent attenuation. As a result, it is impossible to attain a consistent level of covertness at different ranges. Clandestine communications signals ideally have a low probability of detection (LPD) and a low probability of interception (LPI). LPD means that there is a low probability that an adversary will determine that observed signal energy is not attributable to the environment. LPI means that there is a low probability that an adversary will be able to extract useful information from detected energy. Some methods that show promise in the underwater environment include direct sequence-spread spectrum (DSSS), frequency-hopped spread spectrum (FHSS), mimicry, high frequency transmission, and frequency-hopped mini-packets (see Section 4.7.3).

A clandestine communications system has no utility unless it provides the required communications functionality. So, measuring the performance of such a system must presume that the communications system is providing the specified amount of link availability. A generic measure of clandestine (or LPI) performance compares the signal strength needed to support a certain level of communications performance to the signal strength needed to support a specified level of intercept success. These signal strengths must be determined for the same noise and interference backgrounds. The "LPI margin" is therefore defined as the ratio of the signal power required for intercept success to the signal power required for successful communications. A communications system becomes more and more covert as the LPI margin increases. An alternative definition of the LPI margin is the ratio of the input SNR needed for intercept success to the input SNR needed for communications success. This definition is more convenient to use because the two SNR values can be determined separately and then combined as an LPI margin.

For the sake of simplicity, it is assumed in this section that the intercept system employs a wide-band energy detector (or radiometer). Then, the intercept performance equations for reasonable time-bandwidth product DSSS or FHSS waveforms are identical in additive white Gaussian noise (AWGN) given the waveforms occupy the same bandwidth and are transmitted over the same duration of time. In this case, any difference in the LPI margin between two waveforms is measured in terms of the communications performance, i.e. the SNR at which the specified probability of a bit error (P_e) is achieved.

4.9.1. Direct Sequence-Spread Spectrum

Direct sequence-spread spectrum signals were introduced in the section describing CDMA. In fact, CDMA is sometimes referred to as Spread Spectrum Multiple Access or SSMA. The description here will focus on the application of DSSS to clandestine communications. One distinguishing characteristic of DSSS signals is that the bandwidth is increased beyond the minimum necessary to exchange information. Another is that the spreading codes used to generate DSSS signals resemble noise to receivers other than the intended one and are thus sometimes referred to as pseudonoise sequences. These factors help make DSSS signals less overt, as signals may be hidden in the background noise by transmitting at a low average power level. Furthermore, without knowledge of the specific spreading code used to modulate a transmission, an adversary will not be able to easily decode a transmission. Without access to the specific spreading code in use, transmissions will have resistance to jamming since any interference will not correlate well with the specified code.

Figures 4.12 and 4.13 show P_e for BPSK DSSS without coding as a function of signal-to-noise power ratio (SNPR) for multiple values of the processing gain (*PG*) in AWGN and Rayleigh-fading channels, respectively. A greater "spread" in the signal frequency is characterized by a greater *PG* value in that

$$PG = \frac{T_b}{T_c} = \frac{W}{R_b}$$

where T_b and T_c are the bit and chip durations of the DSSS waveform, respectively, W is the signal bandwidth, and R_b is the uncoded bit rate. Each family of curves is derived by substituting

$$\frac{E_b}{N_0} = \frac{S}{N} PG$$

into the appropriate formulas for bit error performance [Proakis95].

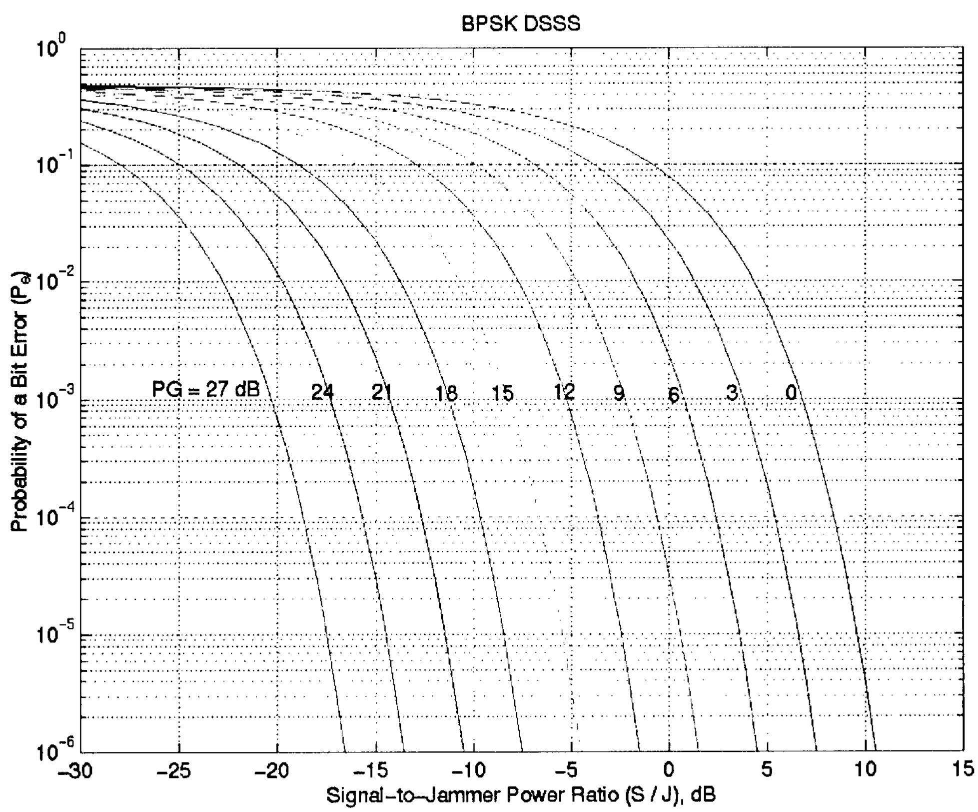


Figure 4.12 - BPSK DSSS Performance in an AWGN Channel

4.9.1.1. Selected Experimental Results

A 1995 French study [Hakizimana95] demonstrated the use of spread spectrum sequences called Yates-Holgate sequences in an underwater environment. A set of 2^k Yates-Holgate sequences is constructed from a parent sequence. The parent sequence is a maximal-length sequence (m-sequence), which is a sequence of length 2^k-1 that has an impulse-like autocorrelation function. When any Yates-Holgate sequence is correlated with the parent sequence, the cross-correlation function yields a uniquely identifiable shape as well as synchronization information. Each transmitted sequence provides 2^k bits of information. The advantage of Yates-Holgate sequences over other spread spectrum sequences is that only one correlator (namely the parent sequence) is required at the receiver. A set of 32 Yates-Holgate sequences (of length-31) were tested in a lake. The range was 13.5m, and the transmitter and receiver were located at a depth of 4m. BPSK modulation was used to implement the sequences using a 5kHz carrier frequency. Transmission rates were 188bps and 375bps, corresponding to 1250 and 2500 sequences per second, respectively. Two receive SNRs were analyzed: 8dB and –2dB. For the 8dB case, error rates were found to be worse than that computed for Gold sequences, a particular set of well-studied sequences that were used for purposes of comparison. However, at –2dB, the Yates-Holgate sequences exhibited slightly better error rates than that for Gold sequences.

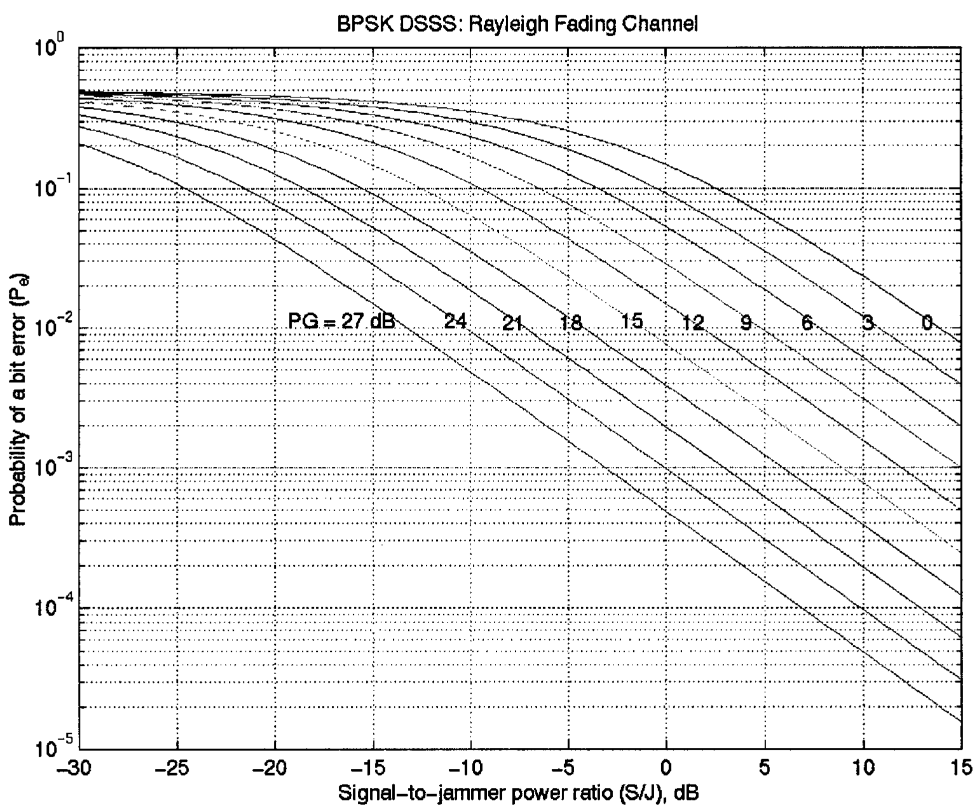


Figure 4.13 - BPSK DSSS Performance in a Rayleigh-Fading Channel

In 1997, research at France's CEPHAG-INP Grenoble Institute was reported [Loubet97], showing that spread spectrum techniques were successfully employed at receive SNRs as low as –5dB, maintaining a data rate of 35.7bps at 45 nautical miles, using a 375Hz bandwidth signal centered at 1.5kHz. These experiments were conducted at a military site, Cap Ferrat, in conjunction with the French Navy. The experiments involved transmitting length-63 Gold sequences. Gold sequences were selected for their outstanding autocorrelation properties, which allow them to be reliably detected at low SNR. Due to their length, the transmit time is 63 times longer than it would be without spreading the signal, while the power level is 18dB lower (which corresponds to 1/63).

A follow-up study looked at finding spreading codes that are more efficient than Gold sequences [Boulanger98]. The researchers developed spreading codes that minimized a criterion on both odd and even correlation functions. Previous spreading code development neglected the so-called odd correlation function. These new codes were found to have better performance than the Gold sequences. The sequences were tested at the same location, Cap-Ferrat, in the Mediterranean Sea at 20 and 50km ranges. The carrier frequency was 1.666kHz and the bandwidth was 520Hz. Receive SNRs were as low as –14dB with an error rate no greater than 0.2%.

Researchers at Northeastern University have developed a direct sequence-spread spectrum underwater acoustics communications system [Stojanovic98, Sozer99]. The transmitter uses two Gold sequences of length-2047 to spread differentially-coded data. The same data bit is spread both in the quadrature and the in-phase branches of the channel. The signaling rate is 100bps and the carrier frequency is 12kHz. The receiver makes use of a rake receiver to incorporate the energy present in multiple propagation paths. The self-noise generated within the taps of the rake receiver is eliminated via thresholding. The system was tested using data taken in 1999 from the Baltic Sea. The receiver was placed at a depth of 30m, the transmitter was placed at a depth of 6m, and the distance between the two was about 3km. Error-free transmissions were achieved at the SNR levels tested.

4.9.2. Frequency Hopped Spread Spectrum

An alternative to spreading the bandwidth of a signal in time is to spread the signal in frequency by employing FSK with frequency hopping. $M = 2^k$ tones are assigned with a spacing of $1/T$ between tones, where T is the tone duration. With this spacing, the tones are orthogonal to one another. T is chosen to be much greater than the multipath spread, similar to the FSK limitations discussed previously. Frequency diversity is obtained by means of frequency redundancy in the encoded signal to be transmitted. As the bandwidth of a tone is much smaller than the bandwidth of the channel, this serves as the method for combating fading within the channel.

A 1988 study [Solaiman88] discussed a slow frequency hopped binary FSK scheme. Theoretical bit error rates were derived for Rayleigh selective and Ricean selective fading channels. Improvements in the error rates were made possible by the use of convolutional coding. [Stojanovic98] described an 8-FSK frequency hopped spread spectrum modem design. By using a wide bandwidth, fading is expected to be uncorrelated. There is mention of the necessity of accurate Doppler tracking to reduce the effects of Doppler spread and Doppler shift.

[Green97] proposed a frequency hopped MFSK approach that uses a whitening filter and strong error correction coding to facilitate low SNR transmissions. Whitening is accomplished by varying the durations of the transmitted tones at different frequencies. This is done in such a manner as to enhance energy at known fade frequencies in the channel and reduce energy at good frequencies. This also makes cyclostationary counter detection techniques less effective. Transmissions are encoded using a constraint-length 10 nonbinary (the alphabet size is greater than 2) rate 1/2 convolutional code with interleaving. A sync signal is also transmitted with a message to indicate time of arrival and any needed frequency correction based on the range and data rate. At the receiver, sequential decoding is employed, which is termed a "classical solution" for processing signals in difficult channels.

The telesonar type-B communication relies on such frequency hopped MFSK signals with nonbinary error correction coding [Green98a, Green98b, Rice98a, Rice99, Green00]. The approach can involve either sequential decoding or maximum-likelihood decoding, depending on the power and bit error rate requirements of an application. It is intended to complement a coherent system to provide reliable, multiple access communications. One application of type-B waveforms is for handshaking to establish links between two network nodes. For the case of Node 1 seeking to communicate with Node 2, Node 1 transmits the hopping pattern for Node 2 (which is known by both nodes). Node 2 is continuously searching for said pattern, and receives the initial transmission. Further handshaking establishes the type of modulation and message format for communications. An additional feature is that the type-B waveform can serve as a channel probe for estimating characteristics of the channel.

A test of the type-B signal for channel characterization was conducted in San Diego Bay [Rice99, Green00]. The range was 400m in shallow water with a depth of 7m. Both the source and two receivers were located at 5m depths. The receivers were separated from the source by distances of either 1700m or 2600m. Successful transmission occurred only at the near receiver. The estimated channel spread for the type-B waveform was found to agree with a reference method incorporating a low-frequency modulated waveform.

The telesonar type-B signaling was also tested in March 1999 in the Baltic Sea [Green00]. This test involved a Datasonics modem operating in 80m shallow water at frequencies between 9-14kHz and transmitting at a source level of 180dB/μPa. For a source located 3km away from the receiver that was traveling away from the receiver at 8kts, a rate of 54bps was achieved.

4.9.2.1. Acoustic CHESS

Acoustic CHESS is the name given to the non-coherent mode of phase-encoded frequency-hopped signaling as outlined in Section 4.7.3.2. In acoustic CHESS data is encoded using a technique called differential frequency hopping (DFH). This can be defined in the following manner: Given a data symbol X_N and frequency of the previous hop F_{N-1}, the frequency of the next hop is defined as:

$$F_N = G(F_{N-1}, X_N)$$

where the function G can be viewed as a directed graph whose nodes are frequencies and whose vertices are labeled with data patterns. For a set of M frequencies (the nominal hop set), the graph will have M nodes, and each node will have some number of vertices $f = 2^k$, where k is the number of bits/hop being coded. The parameter f is called the fanout of the graph because it refers to the number of vertices emanating from each node. For example, for a CHESS system using a hopset size of 16 frequencies to encode 2 bits/hop, each of the 16 nodes in the trellis will have four vertices, one associated with each of the four possible inputs. A block of data is encoded by breaking it into words of k bits, and traversing the graph starting at some random node. This is done by executing a hop at each node to the next frequency specified by that node.

Acoustic CHESS holds great promise for clandestine communications systems. Figures 4.14 and 4.15 show an upper bound on P_e for soft-decision acoustic CHESS without coding as a function of SNPR for multiple values of PG in AWGN and Rayleigh-fading channels, respectively. For a fixed value of PG in an AWGN channel, it is easily observed by comparing Figures 4.12 and 4.14 that DSSS requires 2.5 dB greater SNR to achieve $P_e = 10^{-6}$ and 1.7 dB greater SNR to achieve $P_e = 10^{-3}$. DSSS performance is marginally better when $P_e = 10^{-1}$ but this error rate will not support most link availability requirements. This acoustic CHESS performance margin over BPSK DSSS in AWGN increases as the number of frequencies (M) increases.

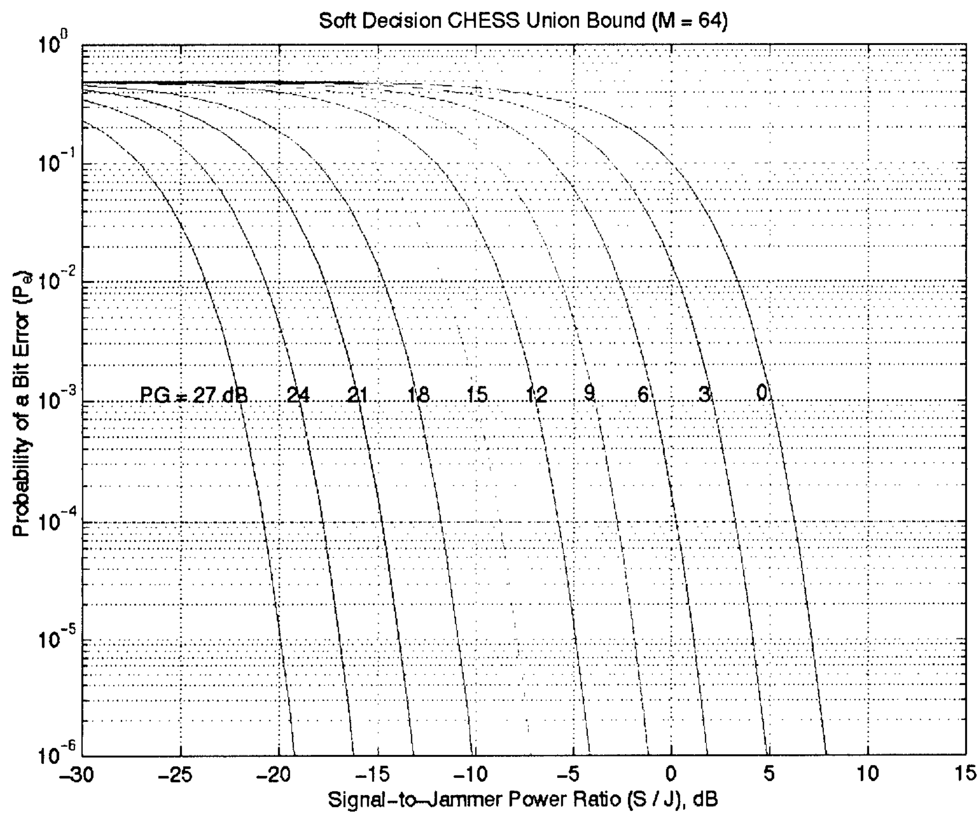


Figure 4.14 - Acoustic CHESS Performance in an AWGN Channel

The underwater acoustic channel is much more likely to be characterized by Rayleigh fading than just by additive noise. Therefore, it is the Rayleigh-fading channel performance comparison that is of most interest. In this case, the performance difference is more dramatic. For a fixed value of PG in a Rayleigh-fading channel, it is easily observed by comparing Figures 4.13 and 4.15 that DSSS requires 24 dB greater SNR to achieve $P_e = 10^{-4}$ and 17 dB greater SNR to achieve $P_e = 10^{-3}$. DSSS performance is again marginally better when $P_e = 10^{-1}$. As is the case with the AWGN channel, the acoustic CHESS performance margin over BPSK DSSS in a Rayleigh-fading channel increases as the number of frequencies (M) increases.

Similar performance gains by uncoded acoustic CHESS over uncoded M-ary FSK can also be shown.

4.9.3. Environmental/Biological Mimicry

Mimicry is a completely different method for providing covert communications. Instead of reducing the SNR of a transmission to a minimum, the idea here is to design a modulation waveform so that it appears to be naturally occurring in the underwater acoustic environment. Mimicry could include biologic mimicry, transmitting waveforms that resemble oceanic animals like dolphins, whales, or shrimp, or environmental mimicry, transmitting waveforms that resemble breaking waves or man-made sources like hull and machinery noises. A message is sent by modulating the phase and/or amplitude of an appropriately chosen mimicking waveform. To implement this method, an adaptive transmitter must be developed that can emulate sources in the present acoustic environment. If done properly, mimicry has the potential to provide truly covert underwater communications. This is because the waveforms being imitated are always present in the underwater environment, and as such, mimicked versions are unlikely to alert an adversary.

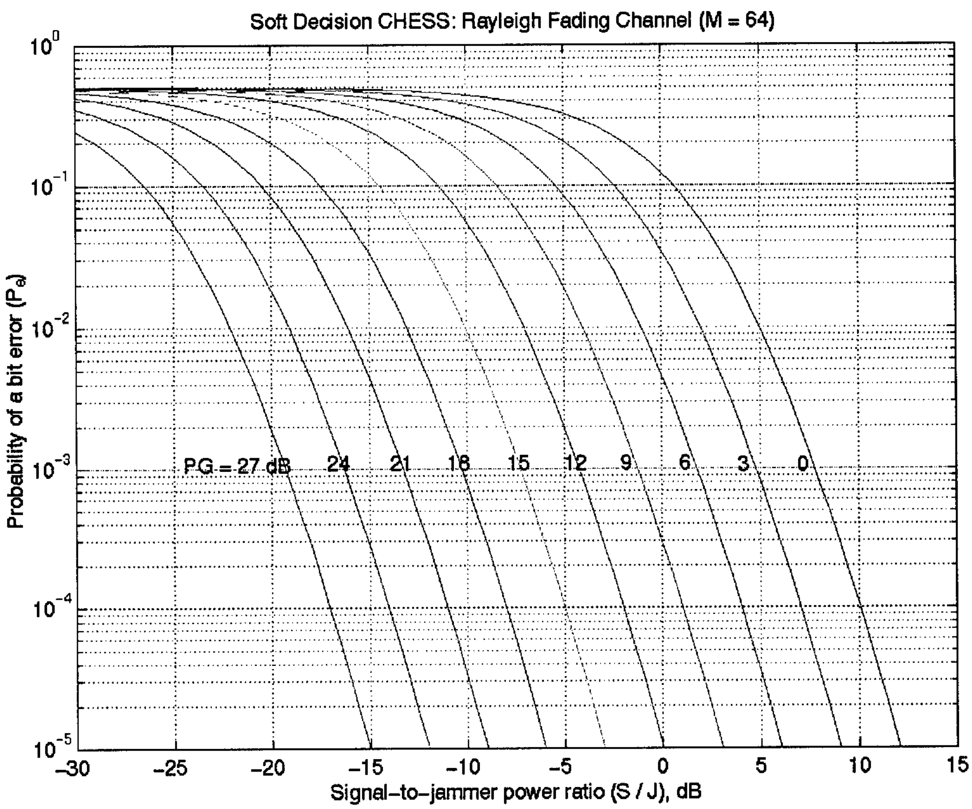


Figure 4.15 - Acoustic CHESS Performance in a Rayleigh-Fading Channel

4.9.4. High Frequency

A simpler method for achieving clandestine communications involves the use of high frequency acoustic transmissions. This method exploits the frequency characteristics of the underwater acoustic environment. Namely, high frequencies tend to attenuate more rapidly than lower frequencies. As such, high-frequency messages received from a source at close range are not likely to be detected by adversaries located further away from the source. This method is of limited utility, however, because such communications will be vulnerable to intercept if an adversary is located between the source and receiver or if an adversary is located close enough to the source that the transmission has not been sufficiently attenuated by the channel.

Intuitively, an interceptor is at an advantage over the intended receive node because the interceptor only needs to detect and not extract information from the signal. Furthermore, the intended receiver requires sufficient SNR to make multiple detections and/or estimates in contrast to the interceptor that only needs to make a single detection. Whether or not these claims are true requires a great deal of analysis and testing.

Consider two nodes spaced by a distance R_0. Assuming omni-directional transmit at a given frequency, omni-directional ambient noise, and equal spherical transmission loss at all horizontal angles, a circle of radius R_0 can be drawn through the receive node on which the SNR is constant. For simplicity, this circle is referred to as the 0dB circle. Two circles with radii $R_{-10} > R_0$ and $R_{+10} < R_0$ denote the corresponding -10dB and +10dB SNR ranges, respectively. This configuration is shown in Figure 4.16.

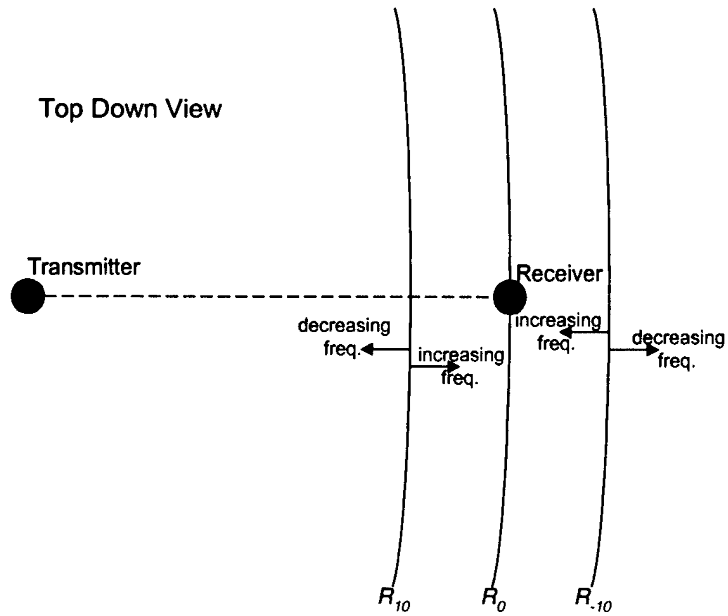


Figure 4.16 - High Frequency Acoustic Transmissions Configuration

If the transmit frequency is increased in this scenario, R_{-10} decreases and R_{+10} increases, thereby making a narrower annulus around the node spacing distance, R_0. However, if the transmit frequency is decreased from the original scenario, R_{-10} increases and R_{+10} decreases, thereby enlarging the annulus around the node spacing distance, R_0. This relationship to frequency implies that increasing the frequency "favors" the interceptor only if the interceptor's SNR requirements force him to be at a range less than or equal to R_0. However, if the interceptor's SNR requirements allow him to operate at ranges greater than R_0, increasing frequency "favors" the intended receive node. Obviously, the relationship is inverted when the frequency is decreased.

The basic challenge given this line of reasoning is to attempt to design a low-BER communications waveform/system that is undetectable at the receiver range.

4.9.5. Directional Transmit

In cases where a transmit array exists, some degree of LPI/LPD capability can be achieved through the use of directional transmission. If the transmitter aims his beam at the intended receiver, then eavesdroppers in other directions will see a reduced signal level, making detection and demodulation difficult.

Vertical directivity produces several advantages. Using vertical aperture enables the steering of a null toward the ocean surface. This will greatly reduce the sound pressure field reaching the ocean surface in a region directly over the transmit platform/node. Furthermore, if a thermocline is present in the transmission region, it is possible to design a transmit system with a half-beamwidth that is less than the critical angle of the thermocline. This design would force an interceptor to deploy a receiver below the knee in the sound speed profile, a sometimes difficult and potentially expensive procedure. It has the added benefit of greatly reducing the potential for inadvertent interception.

Horizontal directivity is advantageous in that the azimuthal area that is being ensonified is reduced basically to the azimuthal beamwidth of the array. This will reduce the probability of detection for omni-transmit by the probability that the potential interceptor is within the azimuthal area of ensonification.

5. Survey of At-Sea Results and Modem Implementations

5.1. Compiled At-Sea Results

We have compiled published results on a large selection of modems in terms of range-rate product, bandwidth efficiency, range, frequency, and power. The data points represent both deep and shallow channels. It is important to note that the operating points reported in the literature may not have been achieved reliably or more than once. However, these low-reliability operating points are presented along with those for which communications performance was satisfactory.

Figures 5.1-5.5 show this compilation of acoustic communications systems performance results. The points plotted in each figure are measured at-sea performance as described in the papers reviewed. Multiple data points from the same research team were included if reported in separate papers. On these plots, "x" indicates coherent systems (e.g. QPSK) and "o" indicates non-coherent systems such as FSK. Every figure is a log-log plot so that the number along an axis is really an exponent. For example, the range axis in Figure 5.1 is labeled "Log Range (km)" and has values ranging from –2 to 4. In this case, 0 corresponds to 10^0 km = 1 km and 2 corresponds to 10^2 km = 100 km.

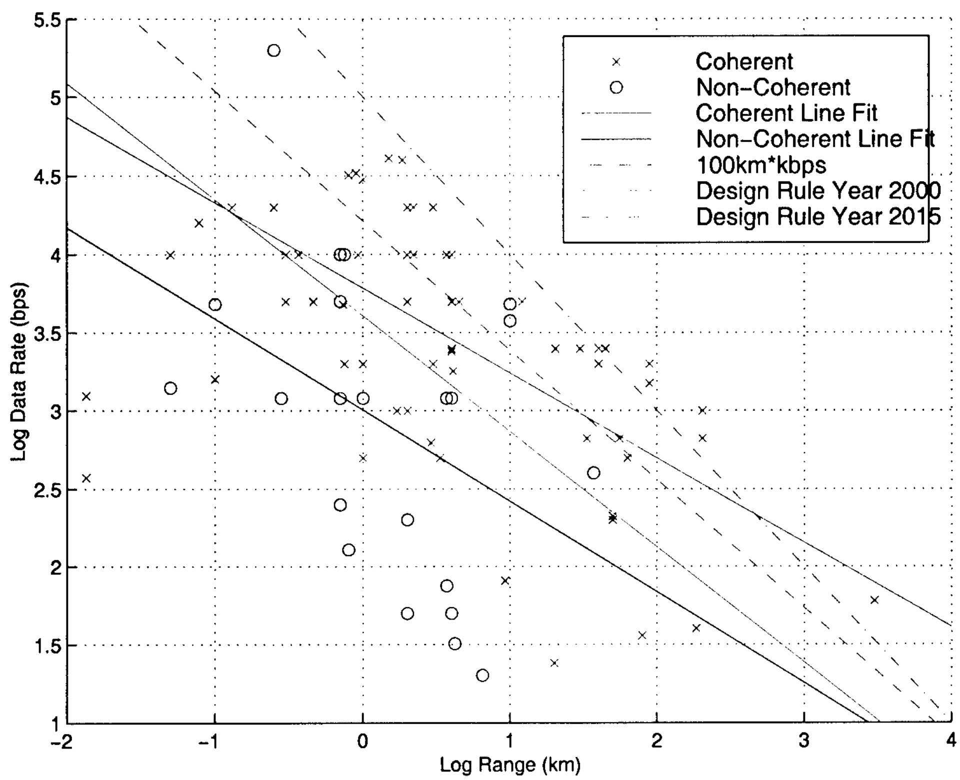


Figure 5.1 - Compilation of Published Range-Rate Results

A summary of range versus data rate experimental performance is presented in Figure 5.1. Sea tests have been conducted at ranges between 13m [Hakizimana87] and 3000km [Birdsall84]. Data rates vary from a small number of bits per second to 41kbps [Pointer99], with one test at

200kbps [Coates93]. Data rate drops with range, as expected due to propagation effects. For ranges greater than 10km, most work to date has been done with coherent systems, while at ranges less than 10km, both coherent and non-coherent systems have been studied. In general, coherent systems are seen to have a higher range-rate product than their non-coherent counterparts. This observation is borne out when comparing the solid red and blue lines in this figure. These lines represent the best line fits to the coherent (red) and non-coherent (blue) results, respectively. Note that since the scales of this plot are not linear, even small separations between these lines and individual data points can mean large differences in performance. The dashed green line in Figure 5.1 denotes the 100 km*kbps curve. Even though this range-rate product has sometimes been "advertised" as an underwater acoustic communications design rule for coherent systems, it is seen here to be more of a limit. The remaining two design rule lines will be discussed more in Section 7.2.

Figure 5.2 shows frequency versus range for at-sea testing to date. Since propagation range falls off with frequency, longer range systems must use lower frequencies. This issue of attenuation versus frequency is quite well understood in that it enforces a limit on the maximum achievable range, which can be approximated by [Dyer]

$$\alpha(f_0)R_{\max} = 10\,\mathrm{dB}$$

where $\alpha(f_0)$ is the attenuation at the center frequency and R_{max} is the maximum range. Of note in this figure is the green curve denoting $\alpha(f_0)R$ = 10dB in which $\alpha(f_0)$ is the attenuation at the center frequency calculated using Thorpe's equation and R is the range. This range indicates a theoretical limit in range that can be exceeded with substantially more complexity or cost in the sonar [2]. The green curve in Figure 5.2 shows this sonar design guide equation using Thorpe's equation to calculate the attenuation coefficient. The solid black line shows a combined line fit to all the coherent and non-coherent operating points shown in the figure.

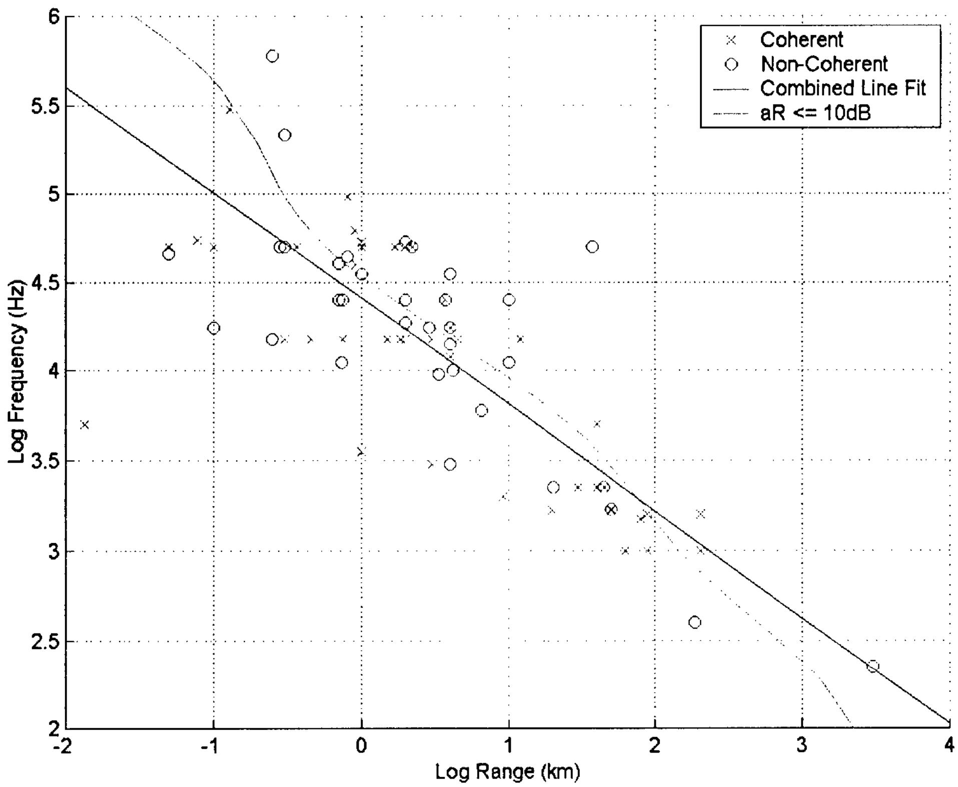


Figure 5.2 - Compilation of Published Range vs. Frequency Results

Figure 5.3 contains the subset of points plotted in Figure 5.2 for which source level information was reported. Shown in this figure is the same combined line fit used in the previous figure in combination with operating points represented by the reported source level. The three operating points shown in red, and all located well above the combined line fit, denote those experiments fortunate enough to have source levels greater than 200dB. The two operating points shown in blue, and located well below the combined line fit, denote those experiments with source levels less than 170dB. The remaining experiments (shown in green) utilized sonars with source levels between 170 and 200dB. As expected, greater ranges can be expected with greater source level for a given frequency.

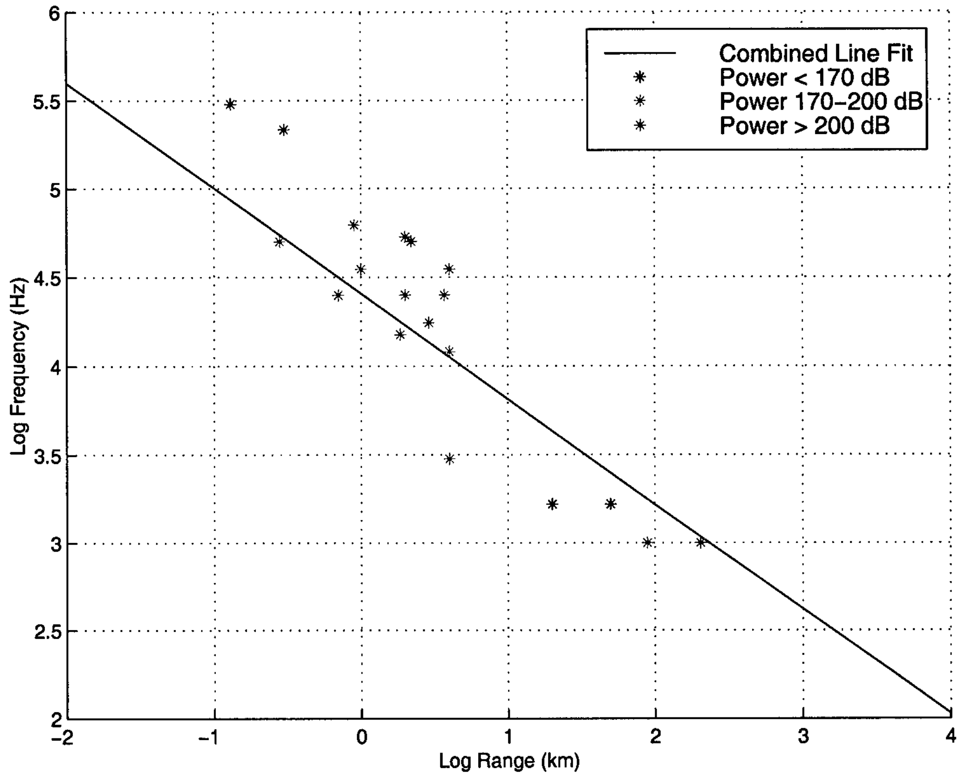


Figure 5.3 - Range vs. Frequency Results as a Function of Source Level

Figure 5.4 provides information on communications bandwidth efficiency, i.e. data rate as a function of bandwidth. An efficiency of 1.0 implies that the data rate is exactly equal to the bandwidth so that data rate should increase with bandwidth if the goal is bandwidth efficiency or data rates as high as possible. The dashed green line in Figure 5.4 shows efficiency equals 1.0 line. The operating points presented in this figure show that increasing bandwidth does generally increase the data rate. The combined line fit curve to the experimental operating points is approximately equal to an efficiency of 1/3, while the remaining two design rule lines in this figure will be discussed more in Section 7.2. However, it is interesting to note that the mean efficiency line and the design rule for the year 2015 are practically identical. The coincidence of this result emphasizes that the design rule was developed for robust, autonomous systems while the experimental results are mostly neither robust nor autonomous.

The trend toward increased data rate with increased center frequency (and the potential for wider bandwidths) is also apparent in Figure 5.5, which plots data rate as a function of frequency. In general, tested coherent systems are seen to have a higher data rate than the tested non-coherent systems. This observation is borne out when comparing the solid red and blue lines in this figure. These lines represent the best line fits to the coherent (red) and non-coherent (blue) results, respectively. Once again, note that since the scales of this plot are not linear, even small separations between these lines and individual data points can mean large differences in performance.

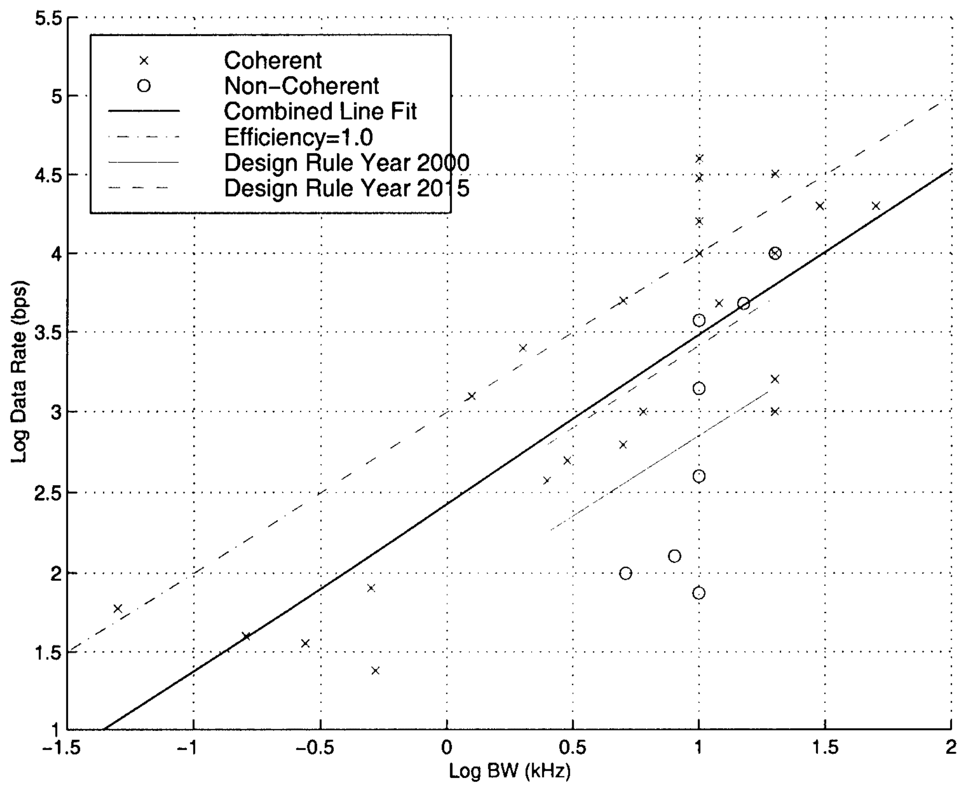


Figure 5.4 - Compilation of Published Efficiency Results

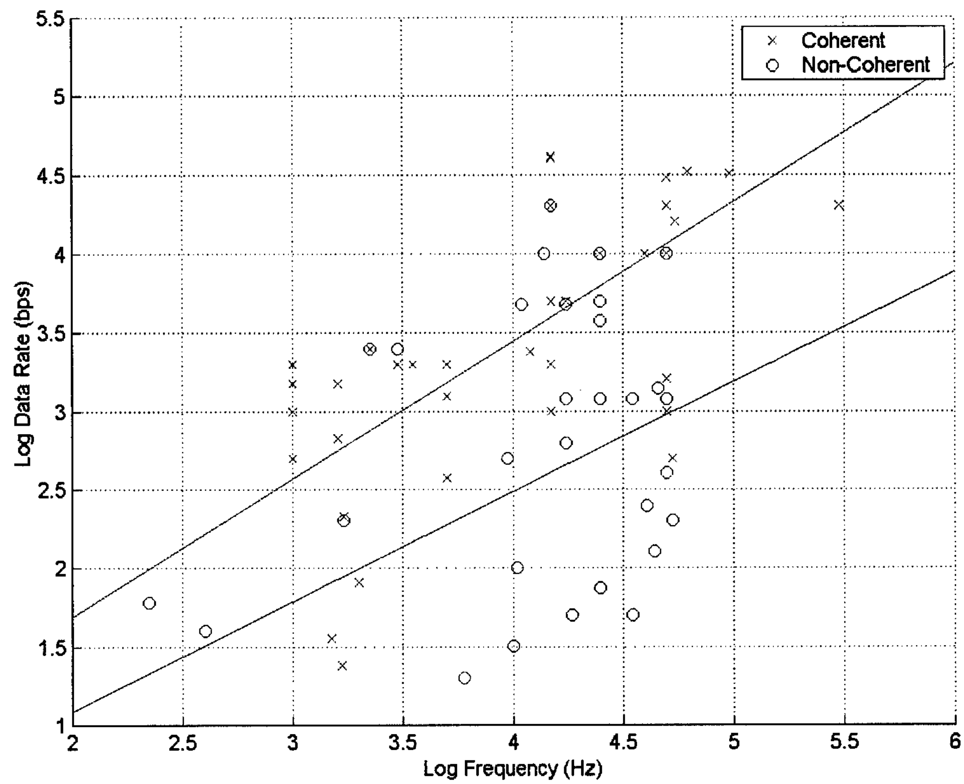


Figure 5.5 - Compilation of Published Data Rate vs. Frequency Results

5.2. Selected Modem Implementations

This section provides the available details on a selection of modems.

[Catipovic84] describes the design of the Digital Acoustic Telemetry System (DATS), a non-coherent MFSK system. This modem system was used to transmit data over short range underwater paths and measure the fading characteristics of CW tones. The system processed a variety of transmissions over ranges from approximately 20m to 1km. At the receiver, the data was quadrature demodulated and Fourier transformed. The modem used an 8085 central processing unit (CPU).

In 1988, Datasonics and WHOI developed the ATM-850 modem, which is a digital signal processor (DSP) -based acoustic modem capable of digital data transmission up to 1200bps [Freitag91, Scussel97]. The modem uses MFSK modulation with eight tones transmitted simultaneously on a double-sideband carrier. The modem originally used an AT&T DSP32C-R35-080D DSP. The operating frequency band is 15-20kHz [Rice98b]. The modem was installed in the Canadian AUV, Theseus, to lay fiber optical cables in ice-cover waters [Thorleifson97]. In October 1993, two AUVs, both with ATM-850 modems, were demonstrated to communicate with one another [Chappell94].

System designers at Applied Remote Technology in San Diego developed an acoustic telemetry system, ADATS (adjustable diversity acoustic telemetry system) [Mackelburg91, Smith91]. The system was developed for a variety of undersea applications including UUV command and control, and manned underwater vehicle data communications. A joint DARPA/Navy UUV program used the ADATs to demonstrate UUVs to meet specific Navy mission requirements [Pappas91]. The modem system supported data rates from 19.5 to 1200bps using 8-tone MFSK modulation. The modem system was tested at ranges of 300m at 1200bps and 2000m at 50bps. The newer generation ADATS II is able to support eight transmission rate options from 31 to 2500bps. ADATS II has been tested at ranges of 1000m at 1200bps and 4000m at 50bps.

ORCA instrumentation developed, in cooperation with IFREMER (Institut Français de Recherche pour l'Exploitation de la Mer) and ENSTB (Ecole Nationale Supérieure des Télécommunications de Bretagne), an acoustic communication system to transmit images and data on a vertical link channel in deep water. The system transmitted at 19.2kbps on a carrier frequency of 53kHz over a 2000m range using 2-DPSK modulation [Ayela94].

WHOI developed a high-performance acoustic modem for use in the Acoustic Local Area Network (ALAN), where higher data rates are needed in an acoustically-difficult shallow water environment [Herold94]. The DSP board is the TMS320C40 processor, chosen for its relatively low power consumption. The system contains eight-channel, 16-bit analog-to-digital converters (ADCs) and ten-channel, 16-bit digital-to-analog converters (DACs). The data rate of the system is 10kbps and QPSK modulation is used with a carrier frequency of 15kHz. A personal-computer (PC) based system controller manages the various subsections of the modem. There is a capability to attach radio Ethernet and global positioning system (GPS) to the system. This modem was installed on the MIT Odyssey IIB class AUVs for communicating between the AUV and a docking station [Singh96].

The multimodulation acoustic transmission system is an acoustic modem based on two low-power consumption microcontrollers and one 56002 DSP [Ayela94]. The modulation schemes and baud rates of the modem were remotely controlled by the operator. Three types of modulation were implemented on the system: PSK for vertical transmission, chirp and frequency hopping for horizontal transmission. Two frequency bands existed within the system, 10-14kHz (20bps) and 50-58kHz (100-200bps), which have both been tested in shallow water.

The hardware structure of the receiving unit for an acoustic system designed for use between an AUV and its surface base station is presented in [Barroso94]. The input has a pre-amplifier stage, followed by an automatic gain control unit (AGC) and a bandpass filter. The AGC ensures the signal level delivered to the remaining stages stays constant. The bandpass filter is centered around a carrier frequency of 53kHz. The non-coherent demodulation of the signal is broken out into the in-phase and quadrature components around 53kHz. The two signals are then sampled and delivered to a digital processing card, the Motorola DSP56002 at 40MHz with 8 K program RAM. The decoded bits processed by the DSP are sent to a 68HC11 microcontroller, which supervises the whole system. The emitter is based on the same DSP card, which outputs the PSK modulated signal after scrambling and differential coding. The modulated signal is then amplified before being applied to the acoustic transducer.

An underwater acoustic modem has been developed by Edge Tech for AUVs that is advertised as robust and low cost [LeBlanc96, LeBlanc99, LeBlanc00b]. The modem uses a single TI320C31 DSP to generate multi-frequency chirp pulses for transmission and uses continuous matched filter processing in the receiver mode. The modem can operate in either a Chirp MFSK or Chirp DPSK mode. The MFSK mode is more robust but lower in baud rate (2000 baud) versus the DPSK (4000 baud). The modem is presented as being reliable at ranges up to 10km range in shallow water. The Advanced Marine Systems (AMS) group at Florida Atlantic University developed a peer-to-peer communication protocol, which was implemented on the Edge Tech modem with the Ocean Explorer (OEX) AUV [Smith97].

The computer system on-board the AQUA EXPLORER 1000 AUV consists of a MC68040 CPU board, a hard drive, two DSP boards, a frame grabber board for video image processing, and an I/O board [Asakawa96, Kojima96]. The software is developed on a multi-task real-time operating system (VX Works). The modulation is performed with the software on one DSP (AT&T DSP32C), while the demodulation is performed with the software on two DSPs. The data transmission rate is 16kbps, the modulation used is QPSK, the acoustic frequency band is 50-60kHz, and the modem operates to ranges greater than 1200m. Almost all the signal processing, filtering, modulation, demodulation, and synchronization are performed with the DSPs.

The acoustic modem developed for the Autonomous Minehunting and Mapping Technology (AMMT) program is located in the forward and aft free-flood sections of the UUV, with a computer in the center section of the vehicle for control [Freitag96, Paglia96, Regan96]. A ruggedized PC consists of a commercially available DSP motherboard that holds up to four TI C40 processor modules and a custom multichannel analog front-end and A/D converter that interfaces directly to a C40 processor. Multi-pole fixed filters set for 10 to 20kHz perform the signal conditioning internal to the modem to reduce out-of-band interference. In the forward section of the AUV four directional receivers are located on each side. The aft freeflood area contains two directional projectors.

The second-generation AUV called the AQUA EXPLORER 2 (AE-2) has a low bit rate acoustic link to communicate non-video data and a high bit rate for video signal transmission [Kojima97]. The low bit rate is 125bps using an FSK modulation. The carrier frequency is apparently 48kHz. Almost all the signal processing, filtering, modulation, demodulation, and synchronization are done with one DSP. The video high transmission rate is 32kbps, using a QPSK modulation. The QPSK frequency band is 86 to 106kHz. Both the high and low bit signals use Reed-Solomon error correction coding.

The next-generation Datasonics modem, a telesonar type-A MFSK (ATM-875) acoustic modem is advertised to communicate at rates between 100 and 2400bps [Scussel97]. The modem preserves the 1-of-4 MFSK modulation mode used in the Datasonics ATM-850 for higher data rates, but incorporates Hadamard modulation code when detection resistance is preferred and/or when only low data rates are needed. The bandwidth of the ATM-875 is 5120Hz, operating in the frequency band between 8 and 13kHz. The modem uses a TI TMS320C50 processor [Rice98b]. The use of the ATM-875 modem in the Telesonar development effort is discussion in [McDonald98a, McDonald98b, McDonald99].

[Chang-Hong98] discusses a prototype MPSK underwater acoustic communication modem for an AUV application. The transmitter unit consists of a transducer, a transmitter, a TMS320C30 DSP board with a DAC, and a PC. The receiver consists of four hydrophones, a 4-channel receiver, a 16-channel ADC board, and a PC. The fast self-optimized LMS (FOLMS) algorithm is implemented as the DFE update algorithm.

The constraints of power and space on a UUV drive towards an on-board acoustic modem that is compact and low power. The Utility Acoustic Modem (UAM) designed by WHOI is a compact, low-power device with high-performance DSP, multichannel receiver, and efficient power amplifier [Freitag98b]. It is capable of transmitting and receiving PSK and FSK communication signals, offers both 10 to 1000bps and 2.5 to 10kbps transmission and reception for MFSK and PSK, respectively. It can be installed as a board set within an existing pressure housing or used stand-alone in a free-flood area with an internal lithium battery pack. The UAM can operate at frequencies up to 30kHz with the appropriate transducers. The unit draws four watts in receive mode and 30 watts when transmitting at 180dB. The modem achieved a raw burst data rate of 10kbps during a test in 1997 that was reduced to 6.7kbps with ECC. The UAM has achieved a maximum range from an AUV to a support ship of approximately 2km. The demonstrated range increases to greater than 3km for the down-link from the surface ship to the AUV.

The Underwater Digital Acoustic Telemetry (UDAT) system consists of a set of underwater acoustic telemetry modems used for test-range applications [Blackmon98, Blackmon99]. Each modem has a VME chassis and a notebook computer, which acts as the user interface. The VME chassis is controlled by a Force 5CE CPU running SUN OS 4.1.3-u1. The chassis contains a hard drive, a VME clock and timing board, and two octal TMS320C40 DSP boards. The transmit processing resides on two of the TMS320C40 processors. The receiver, which uses a FTF-update DFE, is implemented using eight TMS320C40 processors.

Three different users have successfully tested acoustic modems by LinkQuest, Inc. [Yu00]. LinkQuest, Inc. has four models to choose from, see Table 5-A, of which each test had a modem implemented for a specific use. All the modems maintained a consistent bit error rate of less than 10^{-7}. Global Marine Systems Ltd. of UK used a UWM2000 acoustic modem transmitted data from 500 meters below ocean surface at a data rate of 6,600 bps. Shell Oil Company also used a UWM2000 acoustic modem to upload current profiles from sub-sea ADCP in real-time instead of cable. Lastly, a commercial survey company, C&C Technologies customized a UWM4000 acoustic modem for AUV operations. Image files from the side-scan sonar sub-bottom profiler were transmitted to the surface to gain knowledge of the survey site.

Finally, a VME modem system has been developed at BAE SYSTEMS as part of the ACOMMS ATD. The system was used to develop the ACOMMS ATD software for various platforms and for all at-sea demonstrations on-board research vessels, SSNs, and surface ships. The entire system was designed to fit through an SSN hatch.

The software is a combination of commercial-off-the-shelf (COTS) products and code written in C. The Matlab version of the algorithms was ported to run on the Mercury PowerPC (PPC) environment implementing Mercury vector library code. Several of the Mercury PPC processors in the modem are dedicated to tuning/decimating and packet detection. The remaining DSP processors are arranged in a processor farm, each assigned to perform the signal preprocessing and data equalization for one received packet at a time. Additional processors may be easily added to support real-time operation at higher data rates. The software was written in a modular fashion to support the development effort and to support easy transition to fleet platforms.

The hardware is all COTS products. The VME modem system contains two digital signal processing (DSP) boards (Mercury PowerPC/RACEway). The total processing power is one GFLOPS (gigafloating point operating per second) to modulate and demodulate packets for transmit and receive. These processors were selected to maximize compatibility with near term upgrades to the BSY-1 and SQS-53C sonars and with the NSSN design.

There are several platform interface boards that provide access to sonar systems. The 16-channel, 100kHz analog-to-digital converter board provides input analog data from hydrophones and sonobuoys. The 100kHz digital-to-analog converter board transmits the data. The fiber optic interface board provides input digital data from the sonars aboard the SSN and DDG.

The VME modem system has three storing devices to collect data and process the results for later laboratory results. The internal hard drive can save up to 2GB of packets, while the JAZ disk drive can save up to 1GB of packets. The internal digital linear tape (DLT) drive is capable of archiving 10 hours of input data.

The graphical user interface (GUI) is located on a Unix-based notebook PC that provides an X11R6 display terminal for modem control. An Ethernet interface between the notebook PC and VME modem provides real-time access to results and allows on-site performance tuning.

Developer	Application	Channel	Modulation	ISI Compensation	Equalizer	Band	Data Rate	Processor
Oki Elec. Ind. Comp.	Robot comm/control	Very short (60m) shallow	16-QAM	LE (LMS)	--	1 MHz	500 kbps	--
JAMSTEC	Image tx.	Vertical (6,500m)	4-DPSK	LE (LMS)	--	20 kHz	16 kbps	--
IFREMER/ORCA {Ayela94]	Image and data	Vertical (2km)	2-DPSK	None	--	53 kHz	19.2 kbps	--
ENST-Br. /IFREMER	Digital speech	Test pool	4-DPSK	DFE (LMS)	--	-	6 kbps	--
Micrilor	Telemetry	Medium (1km) Shallow	2-DPSK	DSSS	--	30 kHz/ 100 kHz	600 bps	--
WHOI/Datasonics [Catipovic89b, Freitag91, Pietryka95, Scussel97]	Telemetry	Vertical and Horizontal	16 x 4-FSK	None	None	15 kHz	1,200 bps	DSP32C
DERA	Telemetry	Shallow and deep				4 kHz		DSP96002
Edge Tech [LeBlanc96, Smith97, LeBlanc99]	Telemetry/ AUV Comms.	Long shallow (10km)	MFSK; DPSK			8 – 16, 4 kHz	300 – 2,400 bps	TI320C31
WHOI – A size [Herold94 Johnson94]	Telemetry	Shallow	QPSK	DFE (RLS)	DFE (RLS)	15 kHz	Up to 10kbps	TMS320C40
U Birmingham, UK [Galvin94, Jones97]	Telemetry	Medium (0.9-2.8km) shallow; vertical	MDPSK,		None	50 kHz	M=2, 10 kbps M=4, 20 kbps M=8, 30 kbps	--
U. Newcastle, UK	Telemetry	Medium (0.9km) shallow	4-DPSK	DFE (LMS)	DFE (LMS)	50 kHz	10 kbps	--
Institute of Acoustics [Chang-Hong98]	Telemetry	Medium shallow (4km)	MPSK		DFE (FOLMS)	17.5 kHz	Up to 10 kbps	TMS320C30

Table 5.A - Summary of Acoustic Modems

Developer	Application	Channel	Modulation	ISI Compensation	Equalizer	Band	Data Rate	Processor
WHOI/Northeastern U.	Telemetry	Long deep (~185km) long shallow (~93km) med. Shallow (1.9km)	M-PSK 8-QAM M=4,8,16	DFE (RLS)	Multichannel DFE (RLS)	1 kHz, 1 kHz, 15 kHz	1 kbps, 1 kbps, 40 kbps	--
NSF ALAN (surface modem)	Telemetry	Vertical (~1.8km)	FSK, 2-PSK, 4-PSK	None, DFE (RLS)	DFE (RLS)	8-21 kHz	40 bps 10 kbps	DSP32C
WHOI VME Modem	Telemetry	Vertical deep (100m), shallow	MPSK	DFE (RLS)	DFE (RLS)	8-30 kHz	-	TMS320C40
Delphi VME Modem	Telemetry	Shallow	MPSK/QAM	DFE (RLS)	DFE (RLS)	-	-	1860
WHOI PC Modem (AMMT, Haro Strait) [Freitag96, Paglia96]	Telemetry/ UUV Control	Shallow	MPSK	DFE (RLS)	DFE (RLS)	1.5-30kHz	Up to 10 kbps	TMS320C40
LDUUV (VME Modem) [Cancilliere94]	Telemetry/UUV Control	Shallow	MPSK	DFE (RLS)	DFE (RLS)	<50kHz	Up to 30 kbps	TMS320C40
WHOI/BAE SYSTEMS – A Size modem [Fiore95 Herold95]	Telemetry	Shallow	MPSK	DFE (RLS)	DFE (RLS)			ADSP21062 (SHARC)
BAE SYSTEMS Lunchbox	Telemetry	Shallow	MPSK	DFE (RLS)	DFE (RLS)			ADSP21062 (SHARC)
Marine Acoustics Limited	Telemetry	Medium (6,000m)	FSK			8.5 kHz – 16 kHz		
WHOI UAM [Freitag98b]	Telemetry		MFSK and PSK	DFE (RLS)	DFE (RLS)	15kHz	10 – 1,000 bps 2,500 – 10,000 bps	TMS320C44

Table 5.A - Summary of Acoustic Modems, cont.

Developer	Application	Channel	Modulation	ISI Compensation	Equalizer	Band	Data Rate	Processor
Datasonics type-A [Scussel97, Rice98b]	Telemetry	Shallow, long, horizontal; Deep, vertical (6,000m)	MFSK and Hadamard MFSK			1 - 40 kHz	Up to 2,400 bps; Up to 1,200 bps	TMS320C50
LinkQuest, Inc. [Yu00]	Telemetry	200 m depth; 300 m range	Broadband Spread Spectrum			26.775 kHz to 44.625 kHz	9,600 bps standard; 19,200 bps optional	
LinkQuest, Inc. [Yu00]	Telemetry	1,000 m depth; 1,500 m range	Broadband Spread Spectrum			26.775 kHz to 44.625 kHz	9,600 bps standard; 19,200 bps optional	
LinkQuest, Inc. [Yu00]	AUV comms	6,000 m depth; 3,000 m range	Broadband Spread Spectrum			7.5 kHz to 12.5 kHz	2,500 bps standard; 5,000 bps optional	
LinkQuest, Inc. [Yu00]	Image and data	6,000 m depth; 4,000 m range	Broadband Spread Spectrum			12.75 kHz to 21.25 kHz	4.800 bps standard; 9,600 bps optional	
BAE SYSTEMS – ACOMMS ATD	Telemetry	Horizontal; Shallow and deep (>65km – MF, >4.5km – HF)	BPSK, QPSK; 8PSK	DFE (RLS&LMS)	DFE (RLS&LMS)	MF HF		P2A32B-D (PowerPC603)

Table 5.A - Summary of Acoustic Modems, cont.

6. Measures-of-Effectiveness / Requirements

Specific performance metrics need to be calculated for any acoustic communications system in order to measure the overall effectiveness of the system. Measures of effectiveness (MOEs) provide a detailed structure for the evaluation of the communications performance envelope as a function of environmental and platform conditions.

Underwater acoustic data telemetry can be viewed as a union of sonar and communications. As such, it suffers from the same environmental degradation as sonar and communications systems. Efforts undertaken to date have shown that system performance is dictated by received signal-to-noise ratio and multipath complexity, spread, and rate-of-change.

Comparisons between different signaling technologies to gauge their ability to satisfy a particular communications need are made using a variety of MOEs. These include:

- Data rate
  - Burst rate
  - Throughput realized
- Expected bit error rate
- Range
- Required bandwidth
- Required signal-to-noise ratio at the receiver
- Channel robustness
- Doppler tolerance
- Communications latency
- Networkability
  - Number of users that can be accommodated
  - Loss of bit rate per user
- Diversity requirements
- Jam resistance
- Probability of detection (if clandestine operation is desired)
- Computational complexity
- Processor architectural requirements

For any system design, the designer is presented with a set of requirements and the environment in which the system is expected to operate and to meet the requirements. Each requirement can either be viewed as a MOE that must be attained (i.e. a true requirement) or they can be system goals (i.e. performance measures or comparison facilitators). The system design will likely be evaluated for its performance relative to a well-defined set of MOEs. To address the design problem, a set of technologies is available as a toolbox to the designer. An example of this concept is presented in Table 6.A.

The individual entries in the first, third, and fourth columns of Table 6.A can be interchanged based on the problem at hand. For example, the sonars on an AUV may be defined based on its mission profile. In this case, the center frequency, bandwidth, and sonar configuration of the sonar available for communications will be defined and presented to the communications system designer as requirements. The use of MOEs in this fashion serves not only to compare system designs but also to guide the design of system concepts as described in Section 8.

Requirements	Environment	Technology	Performance Measures
Message size	Noise level	Coherent demodulation – PSK, etc.	Joules/bit – Transmit
Number of simultaneous users	Multipath complexity	Non-coherent demodulation – FSK, etc.	Joules/bit – Receive
Throughput	Temporal variability	Center frequency	Required SNR
Range	Spatial variability	Bandwidth	Required SL
Size	Range/depth ratio	Signaling rate	Probability of detection
Power	Propagation loss	Receive configuration	FLOPS/bit
BER	Bottom type	Transmit configuration	
Availability	Sea state		
Detection resistance	Sound speed profile		
Message latency			
Operating depths			

Table 6.A - Acoustic Communications System Design Table Example

7. Outlook

7.1. Acoustic/RF Communications Gateway

The requirement to provide connectivity to submerged Naval assets over very long ranges is unlikely to be met directly with an acoustic link due to the complex effects of acoustic propagation and limited propagation ranges. A solution to this requirement is to use a buoy to convert acoustic signals to and from RF signals creating an acoustic/RF communications gateway. This solution provides real-time two-way communications to distant surface ships, aircraft and/or satellites that can act as relays to integrate undersea communications into the RF-based communications network.

The conceptual system includes a free floating surface buoy or a bottom deployed node with a pop-up RF unit. Acoustic linkage would be via HF to the buoy unit to both support high data rates and to limit vulnerability counter-detection by supporting only short-range acoustic propagation. The bottomed unit would deploy a network of acoustic notes with remote commandable pop-up RF units able to cover a large undersea area. This system could operate either at MF or HF and further limit vulnerability by allowing an AUV or submarine to transmit in a downward direction, and by extending the range between the RF unit and the submarine.

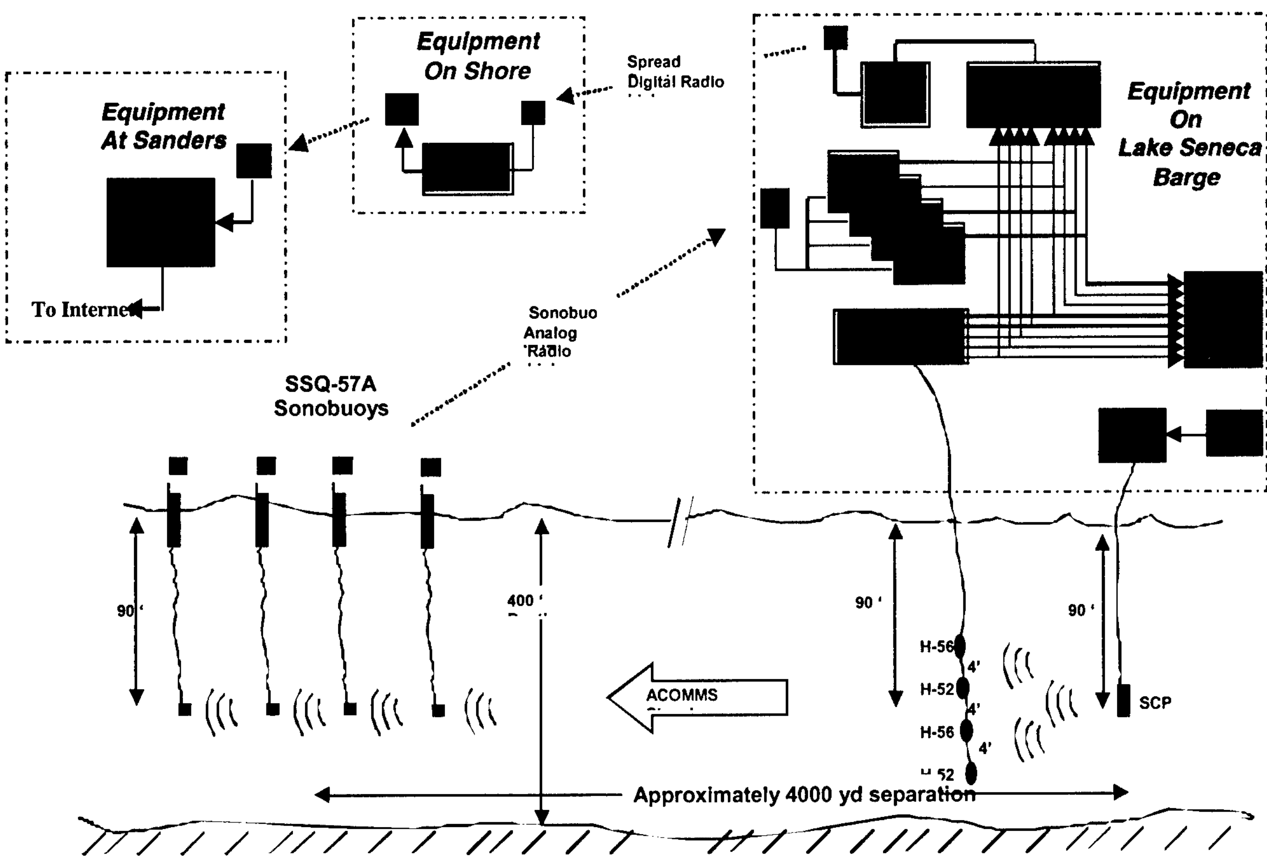


Figure 7.1 - Test Configuration

Coherent acoustic communications technology was employed to demonstrate this concept as a piggy-back to a source evaluation test at the Seneca Lake Sonar Test Facility in September 1999. The experimental setup is shown in Figure 7.1. A low frequency (LF) acoustic signal was transmitted using an LF Slotted Cylinder Projector manufactured by BAE SYSTEMS for SPAWAR. Sonobuoys were deployed at a range of 4km to act as acoustic receivers and to convert the acoustic signal into an RF signal. The analog RF signals were received on the barge via RF receiver and processed by the ACOMMS ATD acoustic modem. Processed results were transmitted ashore via spread spectrum digital RF link, where they were forwarded via Internet phone line to a web site for real-time presentation. Data received, decoded, and downloaded to the web site included assorted JPEG images and text files as shown in Figure 7.2.

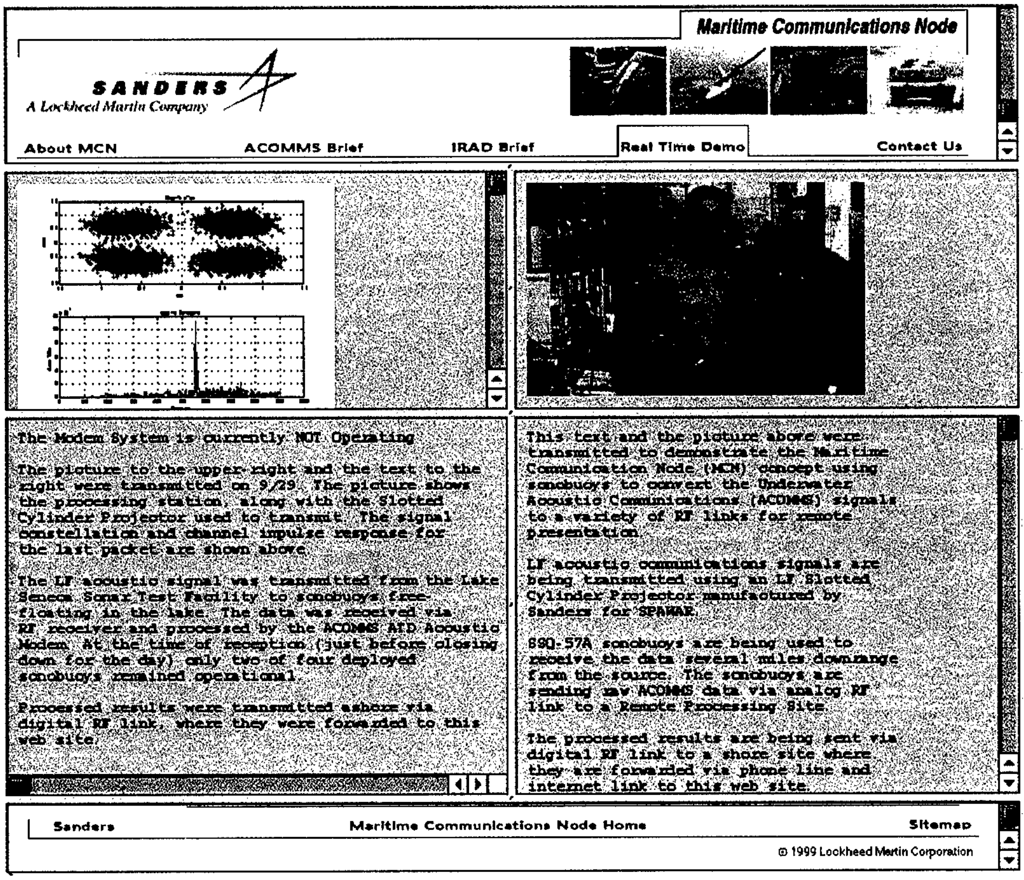


Figure 7.2 - Real-Time Web Site with Data from the Demo

This demonstration validated the use of sonobuoys as receivers for acoustic communications that would place the acoustic communications processing either in the receiving platform or in the expendable buoy. For example, this equipment configuration could be used today to establish an SSN-to-P-3C data link using SSQ-57A sonobuoys. Both analog and digital RF data links were demonstrated, as was remote processing and remote presentation of link data.

Data was transmitted at a rate of 500bps to limit the communications waveform to a center frequency and bandwidth compatible with the LF source. The source was driven at about an 85% duty cycle at low power for a period of approximately seven hours. Packets containing approximately 1-Kbyte of data and lasting just over 30 seconds were transmitted with an inter-packet gap of five seconds, resulting in about 90 packets transmitted per hour. Transmitted data sets rotated through a library of two text files and five images. The text files were each one packet in duration. Four images were each two packets in duration, and one was four packets in duration.

During this demonstration, ACOMMS signal libraries were recorded to a digital audio tape (DAT) from an ACOMMS VME modem. The DAT was then transmitted acoustically using the LF source and amplifier equipment located on the Seneca Lake barge.

Four SSQ-57A sonobuoys were moored approximately 3.2km away at the south mooring site. Four corresponding RF receivers were located on the test barge to provide acoustic data inputs to the VME modem. There were also four receive hydrophones (vertically spaced about 1.2m apart) deployed off of the barge to a depth of 30m. These signals were used for equipment setup verification as well as comparison and backup to the RF signals.

The acoustic data was processed by the modem system to recover the message data. The processed results from the VME modem, along with status information, were then transmitted in real-time via digital RF data link to the Seneca Lake shore facility where they were forwarded by telephone link to a computer at BAE SYSTEMS. This computer acted as a secure (password protected) web site host, allowing real-time viewing via Internet browser from anywhere in the world.

In addition to the normal passband recordings made by the modem system to a digital linear tape (DLT), an 8-channel DAT recorder was used to record all raw acoustic data received on the SSQ-57A sonobuoys and the local reference hydrophones. As normal, processed data from the VME modem was logged with time stamps. This data included all detected packets along with equalization parameters and data recovery results.

The digital RF link consisted of an 803.11 direct sequence-spread spectrum (DSSS) wireless local area network (LAN) using commercial equipment from Aironet. A single +12dBI yagi antenna was used on each end of the link, manually bore-sighted by eye. The horizontal distance from the barge antenna to the shore antenna was scaled from the map to be 2560m. By using the latitude and longitude of the barge (42.6975N, 76.92694W), as indicated on the map and the location of the building (42.68528N, 76.95308W) (as determined by using http://imap.chesapeake.net/) the distance is 2530m. A mean value of 2545m was used for link performance predictions.

In general, link operation was maintained even in the presence of concurrent source testing in a variety of frequency ranges. Some of the high-powered testing, however, negatively impacted the acoustic data link by saturating the sonobuoy preamplifiers. At one point, a nearly in-band, high-power test was interrupted for a short time to verify link operation without interference. Shipping traffic in the vicinity of the sonobuoy receivers also caused a breakdown of the link for the same reason.

7.2. Systems Design Rules-of-Thumb

Figures 7.3 through 7.6 show some design rules-of-thumb that can be used to ensure that proposed system designs have communication requirements that are reasonable. These rules of thumb are generated from simple physical models of the communications problem. For most rules, some minimum threshold required for operation had to be set, these were chosen as conservative estimates based on performance observed during the ACOMMS ATD.

It is interesting to compare these rules of thumb to the data points reported in the literature, as plotted in Figures 5.1 through 5.4.

For throughput versus range, shown in Figures 7.3 and 5.1, we see that the design rule from Figure 7.3 for year 2000 is in the center of the pack of sample points in Figure 5.1, and the design rule for year 2015 is at the upper edge of the sample points. While these rules may seem a bit conservative, it is important to remember that the sample points reported in the literature were undoubtedly achieved, but may not have been achieved reliably or more than once. Thus, we expect that what has sporadically been achieved today will be reliably achieved by year 2015. Throughput diminishes with increasing range because lower frequencies, which propagate further, support smaller bandwidths. Also, the available receive SNR decreases, causing more noise in the data estimates at the receiver, and thus, either more errors are made, or the data must be made more noise tolerant by consuming more time-bandwidth product.

For throughput versus bandwidth (i.e. efficiency), shown in Figures 7.4 and 5.4, we have selected a design rule for year 2000 that is toward the bottom of the pack of sample points, and a rule for year 2015 that is in the middle. These rules were selected because in the published literature, bit errors rates as high as 10^{-2} are reported, and still considered successful. Unfortunately, it is unlikely that a tactical system would operate reliably in an environment where 1% of all bits sent are received correctly (1% of all bits in error means that 8% of all bytes contain errors - or about every other word in written text contains a mistake). Error control coding must be applied, reducing the available throughput by about 50%. Throughput increases with bandwidth because the bandwidth can be used to transmit data symbols faster, or to add constellation points while maintaining a fixed symbol rate.

For range versus center frequency, refer to Figures 7.5 and 5.2. The design rule approximately matches the sample data in this case. This rule is a simple reflection of the fact that high frequencies don't propagate as far as low frequencies, and some minimum SNR is required to decode the received data. Although we expect that the minimum SNR required will decrease by the year 2015, these decreases do not add significantly to the communications range, since we have assumed "typical" operating conditions, for which the receiver may be in a shadow zone at the ranges of interest.

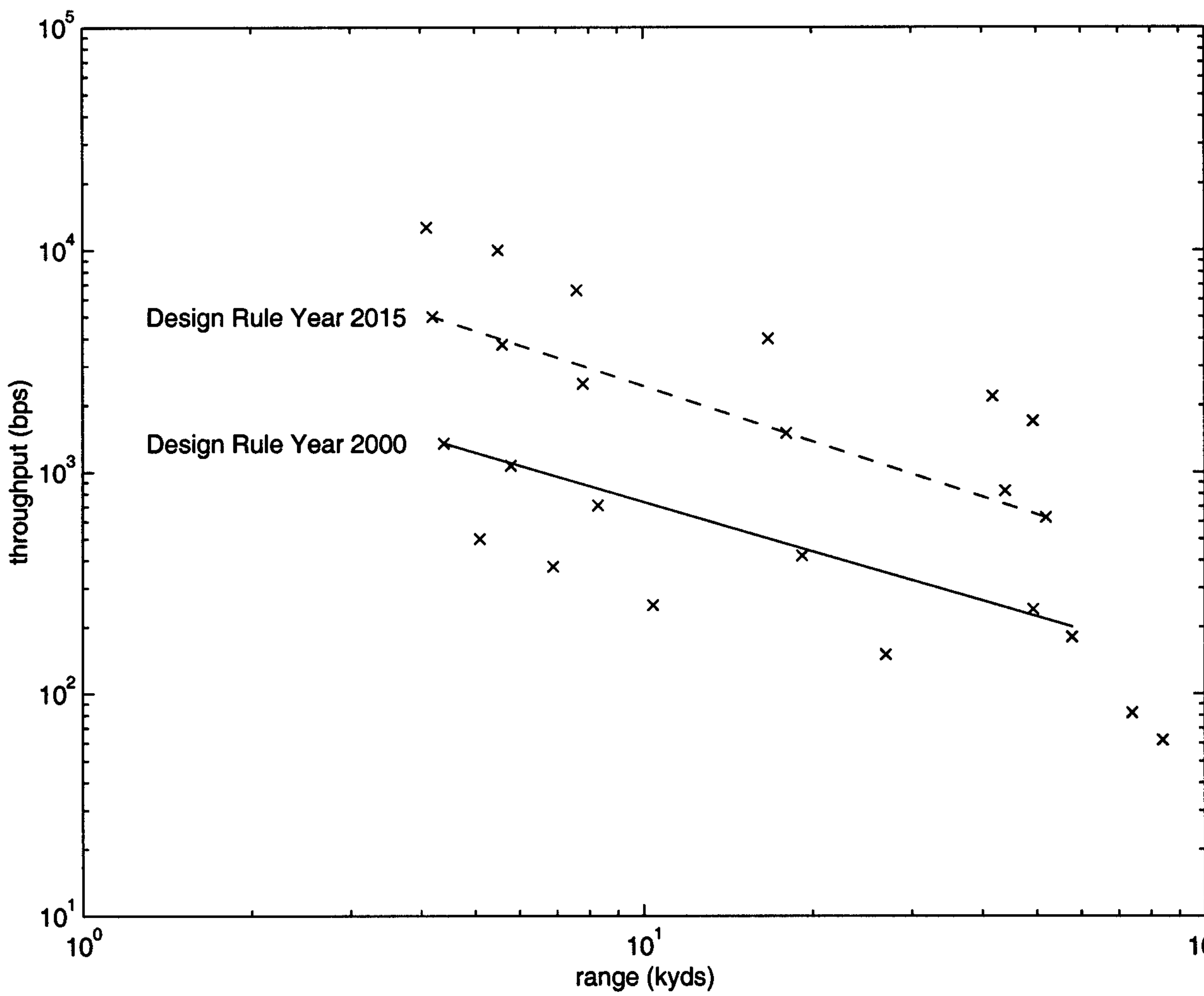


Figure 7.3 - Rules-of-Thumb, Throughput vs. Range

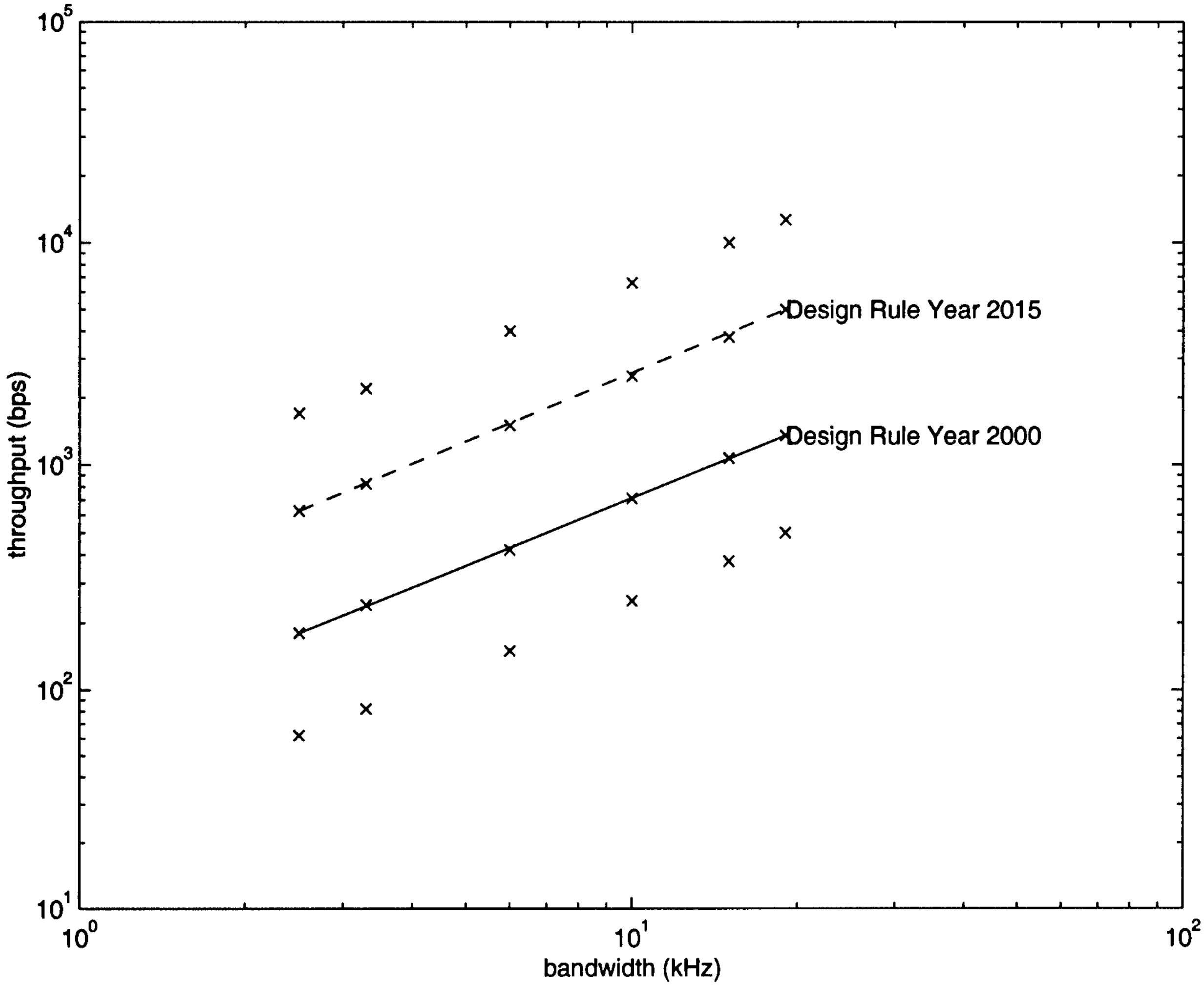


Figure 7.4 - Rules-of-Thumb, Throughput vs. Bandwidth

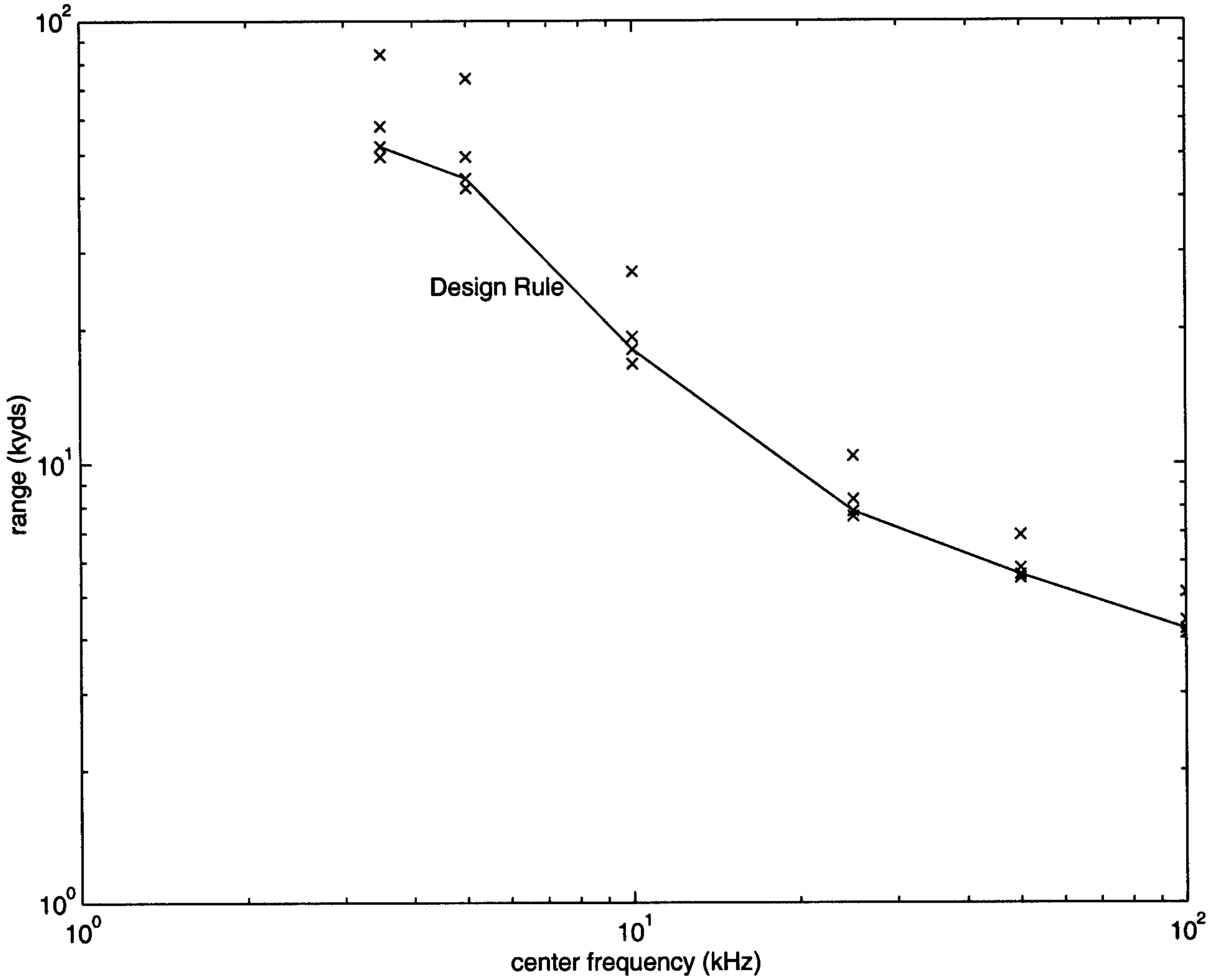


Figure 7.5 - Rules-of-Thumb, Range vs. Center frequency

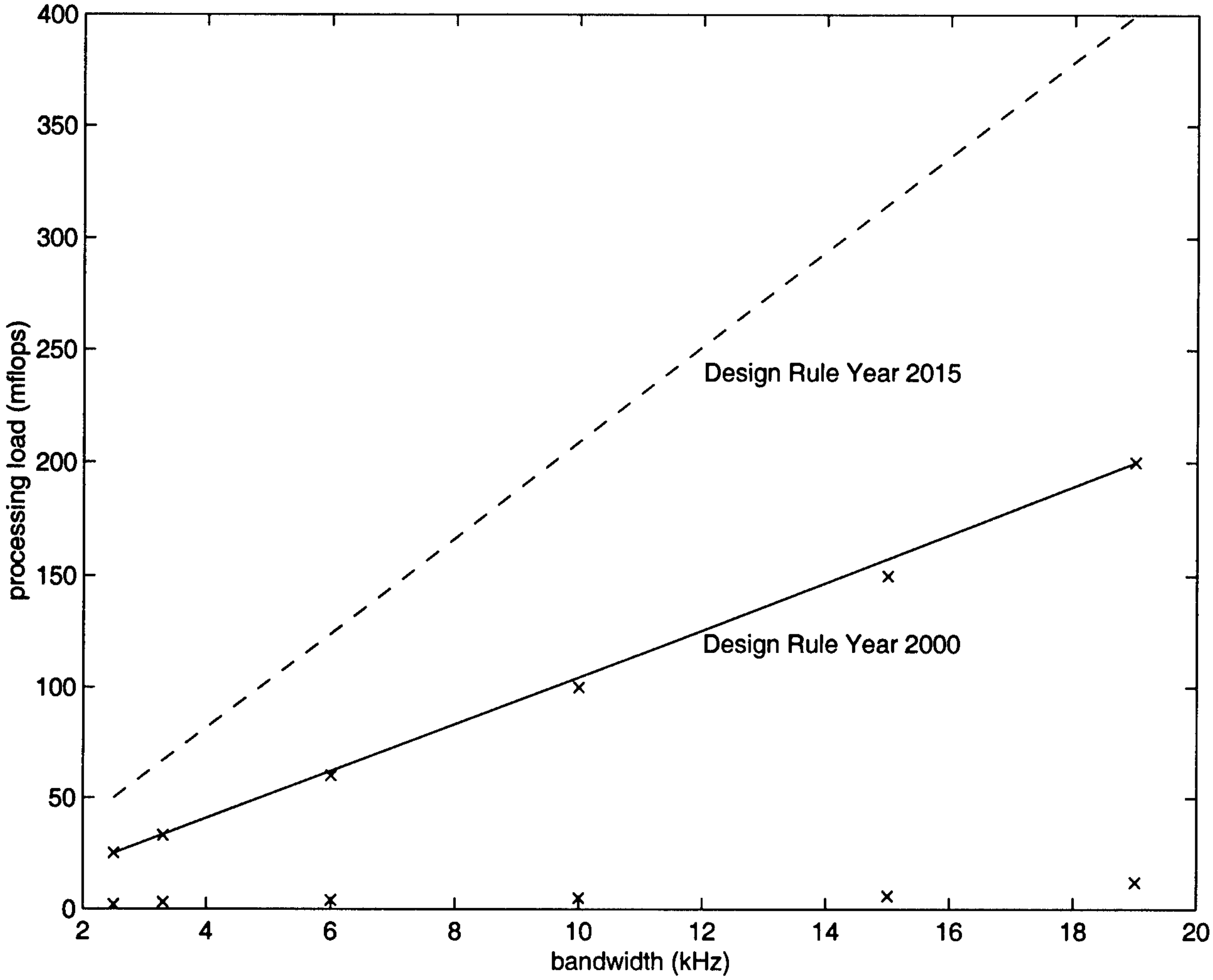


Figure 7.6 - Rules-of-Thumb, Processing Load vs. Bandwidth

7.3. *Candidate Technologies*

Here is a summary of some of the advances that are likely to take place in underwater acoustic communications over the next decade or so.

Improved channel modeling techniques. Increasingly accurate models of the ocean acoustic propagation environment are being developed. Current communications systems all but ignore these models, and instead rely on simple but highly parameterized (i.e. long) FIR (finite impulse response) channel models. Explicitly incorporating more elaborate channel models into modem design may result in significant parameter reduction for a given model accuracy. This point is strongly emphasized in [Kilfoyle99].

Multiuser protocols. Most acoustic communications today is focused on individual links -- a single transmit/receive pair. In the future, the emphasis will shift towards large numbers of transmitters and receivers operating simultaneously. To manage interference among the users, multiuser detection and other interference suppression techniques will be needed at the receivers of such systems, and power-control algorithms will be needed in transmitters to further optimize performance. In this context, various multiple-access protocols such as CDMA and/or TDMA will likely come into use. More generally, network issues will become more important than simply individual link issues.

Content-specific techniques. This is more speculative. The sense is that there may be a move towards modems tuned to specific types of applications and data. By designing transmitters and receivers that are adapted to and optimized for specific kinds of data (say corresponding to the MCM scenario), in principle, better performance can be obtained. Joint source and channel coding techniques are examples of techniques that can be viewed in this class. The basic idea is one of "application-specific" networks.

Iterative algorithms. There has been a lot of activity in the past decade on iterative decoding algorithms for error-correction coding. This is changing the ECC landscape significantly and new codes (such as Turbo codes) are emerging with attractive characteristics. More generally, a variety of iterative algorithms with attractive performance characteristics are finding their way into communication systems.

Universal (blind) algorithms. Most existing communication systems rely on some kind of explicit (and periodic) training phase to bootstrap the modem and keep it tracking. However, training is inherently inefficient. Progress is being made on universal modems that do not require training data, but learn the channel in the course of transmitting data and adapt accordingly, ultimately achieving the performance as if the channel were known a priori. Current blind algorithms (such as blind equalizers) are generally too slow in learning the channel to be practical in underwater applications. However, as these algorithms improve, they will naturally find their way into acoustic modems.

A blind receiver architecture was recently proposed and tested by [Waldhorst00] using data from the ROBLINKS project. The receiver is based on a self-trained adaptive equalizer designed to jointly process signals from a vertical array and perform the tasks of timing and carrier recovery. Blind system startup is permitted by use of a toggling decision rule that exploits the Offset Quadriphase-Shift Keying (OQPSK) modulation. With this rule, decision regions toggle between two perpendicular BPSK constellations depending on the parity of the symbol index. The equalizer coefficients are updated using a simple gradient-type complex LMS algorithm. For the 2km range test, a bit error rate of approximately 10^{-4} was observed with an estimated output SNR of 9.3dB. For the 5km range test, the bit error rate was $4\text{x}10^{-4}$ at an output SNR of 12.5dB. For the moving source test, a bit error rate of $2\text{x}10^{-3}$ with an output SNR of 8.5dB was observed.

Alternative coding and modulation techniques. This is also on the speculative side. The current canonical transmission model has the data being error-control coded, then modulated (possibly separately, possibly jointly). In current practice, we have been constrained to think of coding and modulation techniques quite narrowly. Many alternative strategies await discovery, possibly with very different performance-complexity tradeoffs.

Spatial modulation. A technique termed spatial modulation that seeks to control the spatial distribution of signal energy such that multiple parallel communication channels are supported by the single, physical ocean channel is introduced in [Kilfoyle00]. The basic idea consists of transmitting independent data streams over different, separable propagation paths. At a fundamental level, the availability of multiple, resolvable propagation paths can be interpreted as increased spatial bandwidth with signal design strategies and subsequent benefits similar to those associated with increased frequency bandwidth. The implementation of spatial modulation involves the transformation of a multiple transmitter, multiple receiver framework into a set of virtual parallel channels. The superiority of this approach is demonstrated both theoretically and experimentally. In two different experiments, a bandwidth efficiency of 4 was possible using two parallel channels with an observed relative improvement characterized by a 2.9dB diversity gain as compared to the use of a single channel. In a third experiment, the construction of three parallel channels in a shallow water environment achieved a bandwidth efficiency of 6 with an associated 4dB diversity gain. A data rate of 24kbps was attained for spatial modulation, whereas conventional signaling could not achieve such a rate.

8. Design Examples for the Application of Acoustic Communications to MCM Operations

8.1. Anticipated Uses of Acoustic Communications in MCM Operation

In this section, two possible scenarios are described. They are:

1. Mine search from a surface ship or submarine using multiple AUVs simultaneously to find mines at significant ranges from the host platform. The host platform is coordinating the operation in real-time.
2. Use of moored nodes in conjunction with AUVs to monitor activity in a potentially hostile area. The operation is expected to be semi-automatic, since real-time control will be limited to that available through an LPI/LPD long-range communications link.

8.2. AUV Operations with a Single Gateway Node

In order to analyze the acoustic communications requirements in MCM applications, some assumptions must be made about the scenarios and ancillary equipment that will be available. A single gateway node is used to convert acoustic signals to and from RF signals to provide real-time two-way communications over long distances. See Section 7.1 for further explanation.

We will make these assumptions about AUV operations:

- It is desirable to operate more than one AUV at a time, thus presenting a problem with contention for use of the communications channel.
- The sensor data is processed in the AUV so that images are transmitted back to the gateway buoy and not raw sonar data.

8.2.1. Communications Requirements: Gateway Node with Multiple AUVs

In this section, the likely communication requirements between the AUV and the gateway node are considered. Many possible message contents are considered and the requirements for communication will exceed the bandwidth available if all message types are selected. As the MCM system is designed, it will be necessary to pick only the messages that are needed and send them only as frequently as needed.

The requirements can be refined as system concepts firm up. In many cases, it will be possible to omit some message types and shorten others. For example, contact lists will likely not require target speed and course since most mines are stationary.

8.2.1.1. AUV Location and Health Feedback

The vehicle will need to provide the gateway node with location feedback periodically. Somewhere between once per minute and once per 10 minutes is probably a good frequency. More than once per minute will burden the communications channel unnecessarily (unless these AUVs move very quickly) and less than every 10 minutes presents the possibility of loosing the AUV without realizing it.

A probe pulse could be used to indicate health (still working) and angle to the host, which would be significantly more efficient than actually transmitting the location of the AUV as a digital message. However, it would be difficult to indicate health of each subsystem or the range to the AUV with this technique.

8.2.1.1.1. Throughput

The location can be expressed in any number of coordinate systems. If latitude and longitude coordinates are assumed, then 32 bits for each component is enough to specify location on the face of the earth to within 1 cm (circumference of the earth is about 40,000,000m), which should be sufficient. Depth can be specified to within 6cm in 16 bits (assuming a 4000m maximum depth). It is also desirable to get course and speed information (3 components at 16 bits each).

In addition to all this, the vehicle health must be monitored. We will assume 8 bits for the battery or fuel level and another 16 bits for other data (i.e. OK/failed status for 16 subsystems). All this location and health information totals approximately 19 bytes, or 152 bits, sent from the AUV to the host every minute or so.

Summary: Each AUV must be able to send the gateway about 19 bytes of location and health data every minute. A simplified version could indicate operational status (working/ not working) and direction using simple analog probes.

8.2.1.1.2. Bit Error Rate

Errors in the location and health messages are not likely to be critical. It is unlikely that the operator will need to take action as a result of a single incorrect message. If 9 out of 10 messages arrive intact, an intelligent operator can easily ignore an error in the tenth. The bit error rate is (1 error)/(10 messages*19 bytes each*8bits/byte)

Summary: For the vehicle health messages, a bit error rate of $7x10^{-4}$ or less is sufficient.

8.2.1.2. Update AUV Mission Profile On-the-fly

The AUV's mission profile is likely to consist of a sequence of waypoints and a short set of instructions to follow at each point (i.e. collect image data, collect sonar data, etc.).

8.2.1.2.1. Throughput

Each waypoint will consist of a latitude, longitude, and depth (or equivalent). As described previously, these require at most 32 bits for latitude and longitude, and 16 bits for depth (10 bytes total). Ten bytes of instruction data are added to each so that each waypoint and associated instruction set will require 20 bytes of data. If the mission profile consists of 10 hours of waypoints, with an update every minute, then the mission profile will contain a total of 600 waypoints, requiring 12 Kbytes or 96,000 bits of information.

Mission profiles would not be sent routinely but, when they need to be sent, it is likely to be a result of a high-priority change in the mission or a result of unexpected returns from one of the AUVs. In either case, the mission profile update occurs rarely, but must be serviced with high priority (e.g. all AUVs must be reprogrammed within 10 minutes of some alarm).

Summary: 12 Kbytes of mission profile data must be sent to each AUV within 10 minutes of an alarm.

8.2.1.2.2. Bit Error Rate

Errors in the mission profile are likely to cause serious problems. The AUV may not be able to determine that a message has an error by virtue of its content. Mission profile messages should have checksums or similar error detection features to allow the AUV to determine that a message is faulty. When a faulty message is detected, several options are available:

- If the error can be localized to a single waypoint, omit the waypoint and proceed with the mission.
- If the error can be localized to a single waypoint, request retransmission of that waypoint.
- Request retransmission of the entire mission profile.

All of these are serious errors, which may significantly affect mission performance, and should be avoided. An error rate of one error per 20 mission profiles is probably about the maximum tolerable.

Summary: For mission profile download, an error rate of about $5x10^{-7}$ is sufficient.

8.2.1.3. Vehicle Provides a List of Contacts

In this scenario, each AUV is capable of analyzing returns from is sensors (sonar, camera, magnetic, etc), and maintains a list of contacts that it is tracking. Furthermore, the AUV is somehow capable of initially classifying these contacts to provide the remote operator with a more complete picture of the environment near the AUV.

8.2.1.3.1. Throughput

If the AUV is capable of simultaneously managing 20 contacts, then 10 bytes are needed to specify the location and perhaps two bytes to specify the target type. Total: 12 bytes x 20 contacts = 240 bytes.

The AUV would like to return this data as frequently as possible, probably at least once a minute.

Summary: The AUV must be able to send 240 bytes of contact information every minute.

8.2.1.3.2. Bit Error Rate

As in the case of vehicle health messages, an error in a contact list message can be detected and ignored by an intelligent operator. Since a new contact list will come in another minute, the impact is typically minimal. An error rate of one bit per 5 to 10 messages should be acceptable, particularly since these errors should generally only affect one contact out of the entire message. At a rate of one bit per 10 messages, this will amount to an error rate of $5x10^{-5}$.

Summary: For contact list messages, a bit error rate of $5x10^{-5}$ or less is sufficient.

8.2.1.4. Direct Vehicle to Return Snapshot of Sensor Data

In this scenario, a remote operator is studying contact information returned from one of the AUVs and finds a contact of interest. The operator then requests sensor data regarding the contact.

8.2.1.4.1. Throughput

The contact number (8 bits) and possibly some information about which sensor to use or how much data to return (another 4 bytes) needs to be specified.

Summary: Each AUV must be able to receive 5 bytes of data occasionally, about every 30 minutes.

8.2.1.4.2. Bit Error Rate

These commands to investigate a target of interest are among the most important in the system. An error in one of these messages directly affects the performance of the mission. As a result, a low error rate is appropriate, probably 1 bit in 100 messages.

Summary: For messages directing the AUV to investigate a target, a bit error rate of $3x10^{-4}$ is sufficient.

8.2.1.5. Vehicle Images or Sensor Data for Selected Contacts

In this case, we assume that the AUV has identified a target of interest or the remote operator, reviewing contact information from the AUVs, has requested sensor data regarding some contact that is considered to be of interest.

8.2.1.5.1. Throughput

Assume that a 128x128 image is required with 8 bits per sample. This yields an image containing 128 Kbytes of data.

Some kind of data compression may be possible on this data, lets assume an optimistic 4:1, comparable to lossless text compression, but still far less than photo-quality lossy compression of images (10:1 or 20:1).

Summary: Each AUV must be able to send about 32 Kbytes of sensor data to the gateway on command, about once every 30 minutes.

8.2.1.5.2. Bit Error Rate

It is difficult to draw any reasonable conclusions about acceptable bit error rates for sensor data without any information about how the data is processed.

Assume that the entire snapshot of data is ruined if it contains a single bit error. Since the data constitutes critical information, it is unacceptable to loose a significant number of these messages. An error rate of 1 error in 100 messages will yield a bit error rate of $4x10^{-8}$.

If one error per snapshot can be accommodated, then the required error rate drops considerably to $4x10^{-6}$.

Summary: For sensor data snapshots, a bit error rate of between $4x10^{-6}$ and $4x10^{-8}$ is acceptable.

8.2.1.6. Propagation Time Measurements

In some cases, it may be desirable to use the AUVs to collect sound speed information. This can be accomplished with minimal communication overhead if (1) the AUVs have accurate clocks, which are synchronized to the gateway prior to departure, and (2) the AUV knows its location relative to the gateway or some other reference with high accuracy. Given these assumptions, the AUV only needs to add a time field to the location and health reports that it sends back to the host regularly.

8.2.1.6.1. Throughput

Specifying the transmit time with an accuracy of more than 1msec will require some attention in the design of the time-stamping logic. At a 1.5km range, this 1msec timing error would result in a speed of sound estimate error of about 1.5 m/s. (This assumes that the AUV's location is known with great accuracy, much better than 1.5 meters. Given that the AUV is likely to be several meters long, and thus have a location measurement error of several meters, we must conclude that realistic speed of sound errors will be more like 5 m/s).

To specify the time to within 1 millisecond on a 10-hour mission will require 25 bits. We will round up to 32 bits for this study.

Summary: Each AUV must time-stamp its outgoing messages to within 1 msec accuracy (requires 4 bytes) to provide useful speed of sound estimates.

8.2.1.6.2. Bit Error Rate

An error in one of these messages will typically yield unreasonable results, which are easily detected. Although these messages are likely to be non-critical, an error rate of 1 bit out of 5 or 10 messages is the maximum that is likely to be acceptable.

Summary: Propagation time measurements require a bit error rate of $3x10^{-3}$ *or less.*

8.2.1.7. Range Checking

If the AUV does not have a good self-location mechanism, then its range from the gateway can be determined by the travel time of a message from the AUV to the gateway. Range can be determined by measuring the round-trip time for a message to go from the gateway to the AUV and back. This provides no functionality beyond the scope of active sonar and requires the AUV to provide a return message to the gateway at a fairly exact time after the gateway's message is received.

8.2.1.7.1. Throughput

If the AUV has a synchronized 1-msec accuracy clock as noted previously, then the range can be determined to an accuracy of 1.5 meters, providing that the speed of sound is known perfectly. If there is a 10 m/s error in the speed of sound estimate, then the range estimate error is dominated by the speed of sound error and is 10 meters when the AUV is approximately 1.5km from the gateway.

Summary: Each AUV must time-stamp its outgoing messages to within 1 msec accuracy (requires 4 bytes) to provide ranging estimates.

8.2.1.7.2. Bit Error Rate

An error in one of these messages will typically yield unreasonable results, which are easily detected. Since these messages would be used to track the AUV, repeated errors may cause a loss of track on the vehicle. An error rate of 1 bit out of 10 messages should be acceptable.

Summary: Propagation time measurements require a bit error rate of $3x10^{-3}$ *or less.*

8.2.1.8. AUV Communications Requirements Summary

Table 8.A summarizes the message types complete with their communications burdens and provides an unsupported opinion, based on review of sea-test results, about whether or not each type of message transfer is feasible using acoustic communications.

It seems likely that the communications range requirement for these operations will be the full range of the AUV operations. There are alternatives, in which the AUV goes on a mission, returns near the gateway platform to dump its contact reports, downloads a new mission profile, and leaves. While this scenario is potentially useful, it is much more limiting than a scenario in which the AUV is in constant communication with the gateway.

Table 8.A lists a typical communications demand (item numbers 1, 3, and 6) of 263 bytes/minute/AUV (35 bits per second/node), with occasional burst traffic for mission profile changes (item number 2). With the exception of a mission profile change, most traffic is from the AUV to the gateway. Item number 5 will require approximately 1kbps of realized throughput, which should be straightforward at high frequencies.

Current communications technology should support about 10 AUV nodes sharing the same communications channel (35 bits/second per AUV * 10 AUVs = 350 bits/second. Fifteen percent of the total bandwidth has been allocated to network access overhead). The emphasis needs to be in coordinating multi-node access to the shared communications channel and in dealing with the disruptions caused by uploading sensor data snapshots.

Item Number	Message Purpose	Message Size (bytes)	Maximum BER	Message Frequency	Direction	Acoustic Comms Feasibility Today/10 years
1	Health and location feedback	19	$7x10^{-4}$	1/minute	AUV->Gateway	Today
2	Mission Profile change	12k	$5x10^{-7}$	Within 10 minutes of some alarm	Gateway->AUV	Today probably, 10 years definite
3	Contact Info	240	$5x10^{-5}$	1/minute	AUV->Gateway	Today
4	Request Sensor Info	5	$3x10^{-4}$	1/30 minutes average	Gateway->AUV	Today
5	Provide Sensor Info Snapshot	32k	Between $4x10^{-6}$ and $4x10^{-8}$	1/30 minutes average	AUV->Gateway	Within 10 years
6	Propagation Time/Range Check	4 + 1ms accuracy time-stamp	$3x10^{-3}$	1/minute	AUV->Gateway	Today

Table 8.A - Summary of Message Types

8.3. AUV Operations with Moored Nodes

In this section, we consider the application of acoustic communications to the organization and maintenance of a semi-autonomous network of sensor nodes. One or more of these nodes maintains contact with some host (perhaps a boat or submarine standing off at some distance, or via an RF or satellite link to an on-shore facility).

We assume that the sensor nodes are fixed in place, but the network may have one or more AUVs at its disposal to send out to investigate suspicious contacts. This approach is derived from the AOSN concept [Curtin93]. In addition, we assume that the link to the distant host may provide feedback in semi-real-time, enough to warrant having an AUV investigate a contact that interests the operator on the remote host.

8.3.1. Communication Requirements: Moored Nodes with AUVs

In this section we will consider the likely communication requirements between the sensor nodes, the remote host platform, and the AUV.

8.3.1.1. Network Organization

If the sensors in a network are distributed in semi-random locations (perhaps dropped from an airplane), then the network must organize itself. This organization process consists of allowing the nodes to calculate the geometry of the array based on acoustic measurements. When this task is completed the network must define a communications routing scheme that is optimized for one (or more) of a number of characteristics (e.g. maximized communications reliability, minimized probability of detection, minimized power use, maximized battery life).

Network organization is likely to depend on determining the physical arrangement of the nodes in the network. We can assume that the communications requirements for this process are modest, but the technique used to determine the locations is not obvious.

If the nodes are equipped with some self-localization equipment similar to GPS, then determining the geometry of the array is trivial, and the problem is reduced to optimizing the network of some characteristic such as maximized communications reliability.

In addition, the network must either have a master node that coordinates the communications traffic on the network, or it must utilize a self-regulating protocol. In either case, it is likely that a single node will need to be designated to communicate with the remote host, in order to reduce the need for long range communications equipment. For the remainder of this discussion, the node that contacts the remote host is referred to as the "network master".

Summary: The network must have a mechanism for the nodes to determine the initial geometry of the array. Given the geometry of the network, paths are determined to maximize communications reliability.

8.3.1.2. Nodes Provide List of Contacts to Master

In this scenario, each sensor node is capable of analyzing the returns from its neighboring AUV(s) and maintains a list of contacts that it is tracking. Furthermore, the nodes are assumed to be capable of classifying these contacts to provide the host platform with a complete picture of the environment near the node.

8.3.1.2.1. Throughput

If the node is capable of tracking 20 targets, then 10 bytes are needed to specify the location (see longitude/latitude discussion above) and two bytes are needed to specify the target type for each contact. Total: 12 bytes x 20 contacts = 240 bytes.

The nodes would like to return this data to the master node frequently, on the order of once every 10 minutes.

Summary: Each node must relay 240 bytes of information to the network master every 10 minutes.

8.3.1.2.2. Bit Error Rate

Since a new contact list will come in another 10 minutes, the impact of an error is minimized. An error rate of one per 10 messages should be acceptable, particularly since these errors should generally only affect one contact out of the entire message. At a rate of one bit per 10 messages, this will amount to an error rate of 5×10^{-5}.

Summary: For contact list messages, a bit error rate of 5×10^{-5} *or less is sufficient.*

8.3.1.3. Each Node Provides Status to Master

Each node can relay its health and battery status to the master along with its contact list.

8.3.1.3.1. Throughput

This function requires approximately 5 bytes every 10 minutes.

Summary: Each node must be able to relay 5 bytes of data to the network master every 10 minutes.

8.3.1.3.2. Bit Error Rate

Errors in the health messages are not likely to be critical. It is unlikely that the operator will need to take action as a result of a single incorrect message. If 9 out of 10 messages arrive intact, an intelligent operator can easily ignore an error in the 10th.

Summary: For health messages, a bit error rate of $3x10^{-3}$ is acceptable.

8.3.1.4. Master Relays Contacts List to Remote Host

The network master must periodically reconcile the contact reports from the entire network and relay it to the remote host.

8.3.1.4.1. Throughput

If there are 20 nodes maintaining 20 contacts each and each contact requires 12 bytes, then the network master will need to send about 4800 bytes of contact information to the remote host approximately every 10 minutes or so. Addition of node health information will increase this to approximately 4900 bytes every 10 minutes.

Summary: The network master must be able to relay 4900 bytes of contact and health data to the remote master every 10 minutes.

8.3.1.4.2. Bit Error Rate

Since a new contact list will come in another ten minutes, the impact of an error is minimized. An error rate of one bit per 10 messages should be more than acceptable, particularly since these errors should generally only affect one contact out of the entire message. At a rate of one bit per 10 messages, this will amount to an error rate of $3x10^{-6}$.

Summary: For contact list messages, a bit error rate of $3x10^{-6}$ or less is sufficient.

8.3.1.5. Master Node Assigns AUV to Investigate Contact

In this scenario, the network master, after reconciling contacts from individual nodes, finds a contact of interest. It assigns a local AUV to investigate.

8.3.1.5.1. Throughput

The network master needs to specify the location of the contact (10 bytes), the amount of data to return (4 bytes), and possibly some information about which sensor to use (another 4 bytes).

Summary: The network master must be able to send the AUV 18 bytes of data every 30 minutes.

8.3.1.5.2. Bit Error Rate

These commands to investigate a target of interest are among the most important in the system. An error in one of these messages directly affects the performance of the mission. As a result, only a low error rate is appropriate, say 1 in 100 messages.

Summary: For messages directing the AUV to investigate a target, a bit error rate of $7x10^{-5}$ is sufficient.

8.3.1.6. AUV Returns Sensor Data to Network Master

See Section 8.2.1.5.

8.3.1.6.1. Throughput

Summary: Each AUV must send 32Kbytes of sensor to the network master on command, approximately once every 30 minutes.

8.3.1.6.2. Bit Error Rate

Summary: For sensor data snapshots, a bit error rate of between $4x10^{-6}$ *and* $4x10^{-8}$ *is acceptable.*

8.3.1.7. Autonomous vehicle returns contact identification

In this scenario, the AUV is capable of analyzing returns from its sensors (sonar, camera, magnetic, etc). When sent to investigate a contact of interest, the AUV will form a contact list and possibly classify any contacts that it has detected.

8.3.1.7.1. Throughput

If the AUV maintains a contact list of 20 targets, then 10 bytes are needed to specify the location (see lat/long discussion in a previous section) and another two bytes to specify the target type. Total: 12 bytes x 20 contacts = 240 bytes.

The AUV would like to return this data as frequently as possible, probably at least once a minute.

Summary: The AUV must be able to send 240 bytes of contact information every minute.

8.3.1.7.2. Bit Error Rate

Since a new contact list will come in another minute, the impact of an error is minimized. An error rate of one bit per 10 messages should be acceptable, particularly since these errors should only affect one contact out of the entire message. At a rate of one bit per 10 messages, this will amount to an error rate of $5x10^{-5}$.

Summary: For contact list messages, a bit error rate of $5x10^{-5}$ *or less is sufficient.*

8.3.1.8. Remote Host Requests Sensor Data for Contact

In this scenario, an operator on the host platform is studying contact information returned from the nodes, and finds a contact of interest. A request for sensor data regarding the contact is then made.

8.3.1.8.1. Throughput

The host platform need only specify the contact number (8 bits), node number (8 bits) and possibly some information about which sensor to use or how much data to return (another 4 bytes).

Summary: The host platform must be able to send each network master 6 bytes of data approximately every 30 minutes

8.3.1.8.2. Bit Error Rate

These commands to investigate a target of interest are among the most important in the system. An error in one bit of these messages directly affects the performance of the mission. As a result, a low error rate of 1 bit in 100 messages is appropriate.

Summary: For messages directing the AUV to investigate a target, a bit error rate of $2x10^{-4}$ *is sufficient.*

8.3.1.9. Node-to-Node Propagation Time Measurements

In some cases, it may be desirable to use the sensor nodes to collect sound propagation speed information. This can be accomplished at minimal communication overhead if the nodes have accurate clocks, which are synchronized to the host prior to departure, and if the nodes know their locations with high accuracy. Given these assumptions, the nodes only need to add a time field to the location and health reports that they send back to the network master regularly.

8.3.1.9.1. Throughput

Specifying the transmit time with an accuracy of better than 1ms will require some attention in the design of the time-stamping logic. At a range of 1.5km, this 1ms timing error would result in a speed of sound estimate error of about 1.5 m/s. (This assumes that the node's location is known to an accuracy of better than 1.5 meters.)

To specify the time to within 1ms in a day will require 27 bits, which is rounded up to 32 bits for this study.

Summary: Each node must time-stamp its outgoing messages to within 1ms accuracy (requires 4 bytes) to provide useful speed of sound estimates.

8.3.1.9.2. Bit Error Rate

An error in one of these messages will typically yield unreasonable results, which are easily detected. An error rate of one bit out of 10 messages should be acceptable.

Summary: Propagation time measurements require a bit error rate of $3x10^{-3}$ *or less.*

8.3.1.10. Network Communications Requirements Summary

Table 8.B describes a typical communications demand (item numbers 1, 2, and 5) of 25 bytes/minute/node (4 bits/second/node) between any one node and the local network master. In addition,490 bytes/minute (66 bytes/second) will be exchanged between the network master and the remote host (item number 6). As in the case of the AUV system, most traffic is one-directional, from the sensor nodes to the network master, and then from the network master to the remote host. Item number 4 is straightforward at high frequencies.

Current communications technology should support about 100 semi-autonomous moored nodes sharing the same communications channel ((400 bits/second throughput)/(4 bits/second for each node)) = 100 nodes. Twenty percent of the channel bandwidth has been allocated to network management. The emphasis needs to be in coordinating multi-node access to the shared communications channel and in dealing with the disruptions caused by uploading sensor data snapshots.

Item Number	Message Purpose	Message Size (bytes)	Message Frequency	Maximum BER	Direction	Acoustic Comms Feasibility Today/10 years
1	Contact information	240	10 mins	$5x10^{-5}$	Node -> network master AUV -> network master	Today
2	Health feedback	5	10 mins	$3x10^{-3}$	Node -> network master AUV-> network master	Today
3	Contact list relayed to remote host	4900	10 mins	$3x10^{-6}$	Network master -> remote host	Today
4	Investigate contact	18	30 mins	$7x10^{-5}$	Network master -> AUV	Today
5	Provide sensor data snapshot	32k	30 mins average	Between $4x10^{-6}$ and $4x10^{-8}$	Node -> network master AUV -> network master Network master -> remote host	Today probably, 10 years definite
6	Request sensor info	6	~30 mins	$2x10^{-4}$	Network master -> node Remote host -> network master	Today
7	Propagation time/range check	4 + 1ms accuracy time-stamp	10 mins	$3x10^{-3}$	Node -> network master	Today

Table 8.B - Typical Communications Demand

9. Trade-Offs and System Design for the AUV MCM Mission

In this section, data are presented to generically address underwater acoustic communications for application to the AUV MCM mission. Rather than address all of the data types identified in [Kujawa00] in terms of each of the individual six final system concepts outlined by the editors of [Kujawa00], the data are shown in such a manner as to support the analysis of the utility and use of acoustic communications in these and other system concept designs.

Under the assumption that no new sensors will be added to a vehicle to facilitate the use of acoustic communications, the sonars under investigation for the different sized AUVs must be analyzed for use as communications sensors. In the section entitled UUV ALS Configurations of the *MCM Future Systems Study Workbook*, the ahead looking sonar (ALS) parameters for the six pre-defined vehicle diameter values are presented in two tables, one each for the narrow and wide vertical beamwidth options [Kujawa00]. The chosen vehicle diameters range from the smallest vehicle (4.875 in diameter) capable of performing an MCM mission up to the largest vehicle (54 in. diameter) that could reasonably be expected to be launched from a typical platform.

The two ALS tables from [Kujawa00] have been mimicked here in Tables 9.A and 9.B for application to communications and are meant to provide nominal maximum signaling rate and maximum design ranges for each center frequency / bandwidth combination. The signaling rate (as opposed to information rate) is one-half the sonar bandwidth, which was chosen based on both the coherent and non-coherent signaling results achieved to-date as reported in the literature and summarized in Figure 5.4. The information rate never exceeds the signaling rate and takes into account any synchronization and training data in addition to the redundancy introduced through coding.

UUV Diameter (ins)	54	36	21	12.75	7.5	4.875
Freq (kHz)	50	60	80	105	110	160
BW (kHz)	20.00	24.00	32.00	36.00	44.00	64.00
Max Source Level	220	220	220	220	220	220
Comms Pulse Width at Max SL (ms)	25.00	20.84	15.62	13.90	11.36	7.82
Max Comms Range (m)	2250	1660	1020	650	600	320
Comms Duty Cycle at Max SL	0.8%	0.9%	1.1%	1.6%	1.4%	1.8%
Max Single Transmission Message Length at Max SL (bits)	250	250	250	250	250	250

Table 9.A - Maximum Range Communications for Narrow Vertical Beamwidth AUV Sonars

UUV Diameter (ins)	54	36	21	12.75	7.5	4.875
Freq (KHz)	50	60	100	150	170	245
BW (KHz)	20.00	24.00	40.00	50.00	68.00	98.00
Max Source Level	220	220	220	220	220	220
Comms Pulse Width at Max SL (ms)	25.00	20.84	12.50	10.00	7.36	5.10
Max Comms Range (m)	2250	1660	700	355	290	160
Comms Duty Cycle at Max SL	0.83%	0.94%	1.34%	2.11%	1.90%	2.39%
Max Single Transmission Message Length at Max SL (bits)	250	250	250	250	250	250

Table 9.B - Maximum Range Communications for Wide Vertical Beamwidth AUV Sonars

The first four rows of Tables 9.A and 9.B are taken directly from [Kujawa00]. In each table, the center frequency and the bandwidth increase as the vehicle diameter decreases. This is mostly a practical issue in that the physical arrays are larger and heavier at the lower frequencies. The array design is based on current state-of-the-art array and electronics technology [Kujawa00] so the maximum achievable power is 220dB for all the sonars. To calculate the maximum range, a line fit through the 200dB operating points in Figure 5.3 is used and the vertical beamwidth of the sonars is not taken into account. For these reasons, some of the maximum ranges for communications may tend to be conservative. And, as expected, the maximum range for communications gets smaller as the center frequency of the sonar gets higher.

For each ALS sonar configuration, the longest possible communications pulse width at the maximum source level (SL) as shown in Tables 9.A and 9.B is twice that of the maximum pulse width when the sonar is configured for detection processing. This rule was arrived at based on discussions with the contributors to the ALS section of [Kujawa00]. The maximum message length for a single transmission at the maximum SL is then calculated as

$$\#\text{of bits} = R_s t_p = \frac{BW}{2} t_p$$

in which R_s is the signaling rate in bits per second (bps) and t_p is the pulse width in seconds. The maximum message length for a single transmission is exactly 250 bits for all the pre-defined ALS sonar configurations. In this case the 250-bit message must include all synchronization, training, information, and redundancy bits. The duty cycle values in Tables 9.A and 9.B become important when a total message transmit time needs to be determined for some arbitrary number of bits.

Tables 9.C through 9.L present a trade-off between the maximum message length for a single transmission and the maximum range that should successfully support acoustic communications as a function of the sonar center frequency. The signaling rate is fixed for each sonar to half the bandwidth as presented in Tables 9.A and 9.B. The single transmission message length is effectively increased by reducing the SL of the sonar. For every 3dB reduction in SL, the pulse length can be doubled. The resulting range is calculated using a simple version of the sonar equation. As expected, the appropriate communications range decreases as the SL decreases.

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	25.0	1%	250	2250
217	50.0	2%	500	2100
214	100.0	3%	1000	1950
211	200.0	7%	2000	1810
208	400.0	13%	4000	1660
205	800.0	27%	8000	1520
202	1600.0	53%	16000	1390
199	3000.0	100%	30000	1250

Table 9.C - Communications Ranges and Message Lengths for 50 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	20.8	1%	250	1660
217	41.7	2%	500	1540
214	83.4	4%	1000	1420
211	166.7	8%	2000	1310
208	333.4	15%	4000	1190
205	666.9	30%	8000	1080
202	1333.8	60%	16000	980
199	2213.3	100%	26560	880

Table 9.D - Communications Ranges and Message Lengths for 60 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	15.6	1%	250	1020
217	31.2	2%	500	940
214	62.5	5%	1000	850
211	125.0	9%	2000	770
208	249.9	18%	4000	690
205	499.8	37%	8000	620
202	999.7	74%	16000	540
199	1360.0	100%	21760	470

Table 9.E - Communications Ranges and Message Lengths for 80 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	12.5	1%	250	700
217	25.0	3%	500	630
214	50.0	5%	1000	570
211	100.0	11%	2000	510
208	200.0	21%	4000	440
205	400.0	43%	8000	390
202	800.0	86%	16000	330
199	933.3	100%	18667	280

Table 9.F - Communications Ranges and Message Lengths for 100 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	13.9	2%	250	650
217	27.8	3%	500	580
214	55.6	6%	1000	520
211	111.2	13%	2000	460
208	222.4	26%	4000	400
205	444.8	51%	8000	350
202	866.7	100%	15600	300

Table 9.G - Communications Ranges and Message Lengths for 105 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	11.4	1%	250	600
217	22.7	3%	500	540
214	45.4	6%	1000	480
211	90.9	11%	2000	420
208	181.8	23%	4000	370
205	363.5	45%	8000	320
202	727.0	91%	16000	270
199	800.0	100%	17600	220

Table 9.H - Communications Ranges and Message Lengths for 110 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	10.0	2%	250	360
217	20.0	4%	500	310
214	40.0	8%	1000	270
211	80.0	17%	2000	230
208	160.0	34%	4000	190
205	320.0	68%	8000	160
202	473.3	100%	11833	130

Table 9.I - Communications Ranges and Message Lengths for 150 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	7.8	2%	250	320
217	15.6	4%	500	280
214	31.3	7%	1000	240
211	62.6	15%	2000	200
208	125.1	29%	4000	170
205	250.2	59%	8000	140
202	426.7	100%	13653	110

Table 9.J - Communications Ranges and Message Lengths for 160 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	7.4	2%	250	290
217	14.7	4%	500	250
214	29.4	8%	1000	210
211	58.9	15%	2000	180
208	117.8	30%	4000	150
205	235.5	61%	8000	120
202	386.7	100%	13147	90

Table 9.K - Communications Ranges and Message Lengths for 170 kHz AUV Sonar

Signal Level (dB re 1uPa @ 1m)	Comms Pulse Length (msec)	Duty Cycle	Message Length (Bits)	Comms Range (m)
220	5.1	2%	250	160
217	10.2	5%	500	130
214	20.4	10%	1000	110
211	40.8	19%	2000	90
208	81.6	38%	4000	70
205	163.2	77%	8000	50
202	213.3	100%	10453	40

Table 9.L - Communications Ranges and Message Lengths for 245 kHz AUV Sonar

Figures 9.1 and 9.2 summarize the information given in the preceding tables. Figure 9.1 shows range as a function of the pre-defined ALS sonar configurations, which are denoted by their respective center frequencies, for single transmission message lengths ranging from below 256 bits to above 32 kbits. Figure 9.2 shows range as a function of single transmission message length for each of the pre-defined ALS sonar configurations. Once again, the single transmission message bits not only include the information bits but also all the overhead bits mentioned previously. The driving notion behind these two figures is that the AUV must transmit all the data as fast as possible. So, the basic assumptions behind these figures are that the AUV transmits

- Continuously
- Transmits at the highest possible signaling rate (one-half the bandwidth)
- Transmits a single message, i.e. the data is not divided into multiple transmissions.

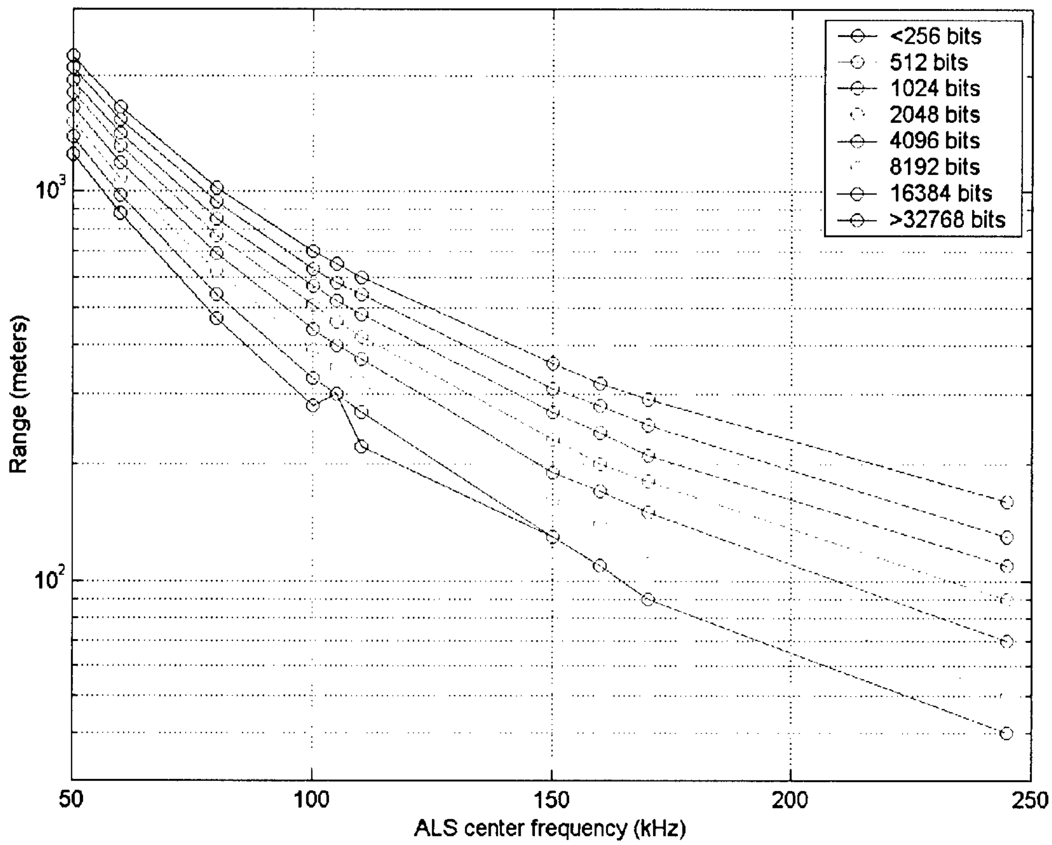


Figure 9.1 - ALS Center Frequency vs. Range as a Function of Message Length

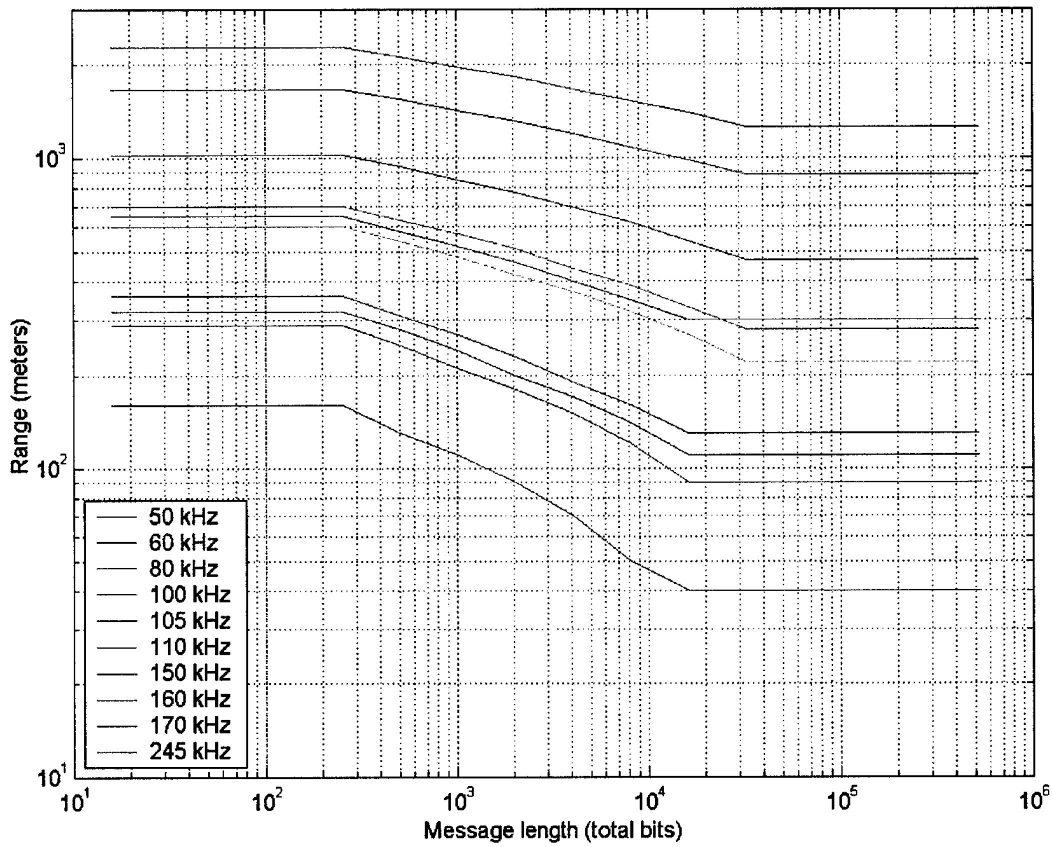


Figure 9.2 - Message Length vs. Range as a Function of ALS Center Frequency

An assortment of data types have been identified within the mission profiles of the final six system concepts identified by the editors of [Kujawa00]. These data types include

- Marker or vehicle identification and location
- Guidance, navigation, or acknowledgement
- Target type or parameters
- Target signature
- Target images
- Maps or area mosaic.

The number of information bits required for these data types ranges from 16 bits to 32 kbytes. This latter value assumes some significant compression of image data. Without this compression, the number of information bits for an image could grow as high as 512 kbytes. Based on the amount of overhead introduced by synchronization, training, and error control coding, the number of bits per message can grow significantly. For example, even without any synchronization or training bits, rate 2/3, 1/2, 1/3, and 1/12 codes will enlarge a 16-bit message to 24, 32, 48, and 192 bits, respectively.

The amount of time it takes to transmit an entire message is directly affected by the overall message size, the range between the transmitter and the receiver, and the sonar design parameters for communications, including the duty cycle. This message delivery time is shown for the ten different AUV ALS sonars (defined by center frequency) in Figures 9.3 through 9.12. These figures present the minimum message transmission (delivery) time in seconds along the vertical axis for multiple transmissions as a function of the overall message length (including overhead) in bits. Both of the axes are logarithmically scaled. Each curve corresponds to a given communications range. The continuous transmission (straight dashed green) line in each figure obscures the blue line depicting the transmission times at the minimum noted range.

The staircase effect in each family of curves is derived from the sonar's transmit restrictions, which is in turn caused by the duty cycle. A reduction in the sonar's peak power is not only accompanied by a reduction in achievable range but also by a corresponding increase in the duty cycle which then allows for longer transmission times. The delivery time is the summation of the time it takes to transmit all the bits and the waiting time between these transmissions. Each additional waiting period adds another "step" to the curve.

The effect of the 50 kHz sonar's duty cycle is clearly seen in Figure 9.3. At the maximum range of 2250 m, a step of 2.475 seconds is added to the delivery time every 250 bits. Whereas, at a range of 1390 m, a step of only 0.9 seconds is added to the delivery time every 16000 bits. If the technology allows, continuous data transmission is possible at ranges less than 1250 m. These values are either taken or derived from Tables 9.A and 9.C.

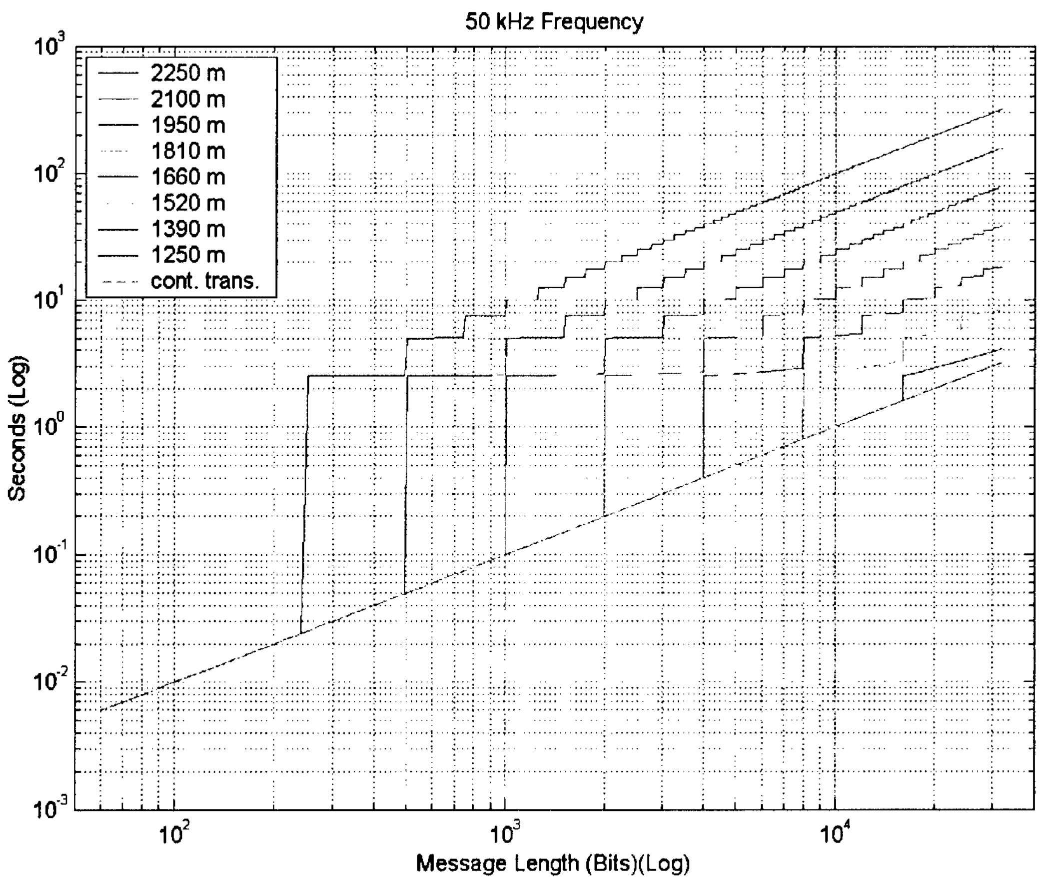


Figure 9.3 - Message Transmission Time vs. Message Length for 50 kHz AUV Sonar

The families of curves presented in Figures 9.3 through 9.12 present the trade space necessary for the AUV MCM system designer. They provide the data to test the utility of acoustic communications for an existing overall system concept. They can also be used as the case may be (1) to modify an existing concept to benefit from the added capabilities offered by acoustic communications or (2) to better accommodate the physical limitations presented by acoustic communications.

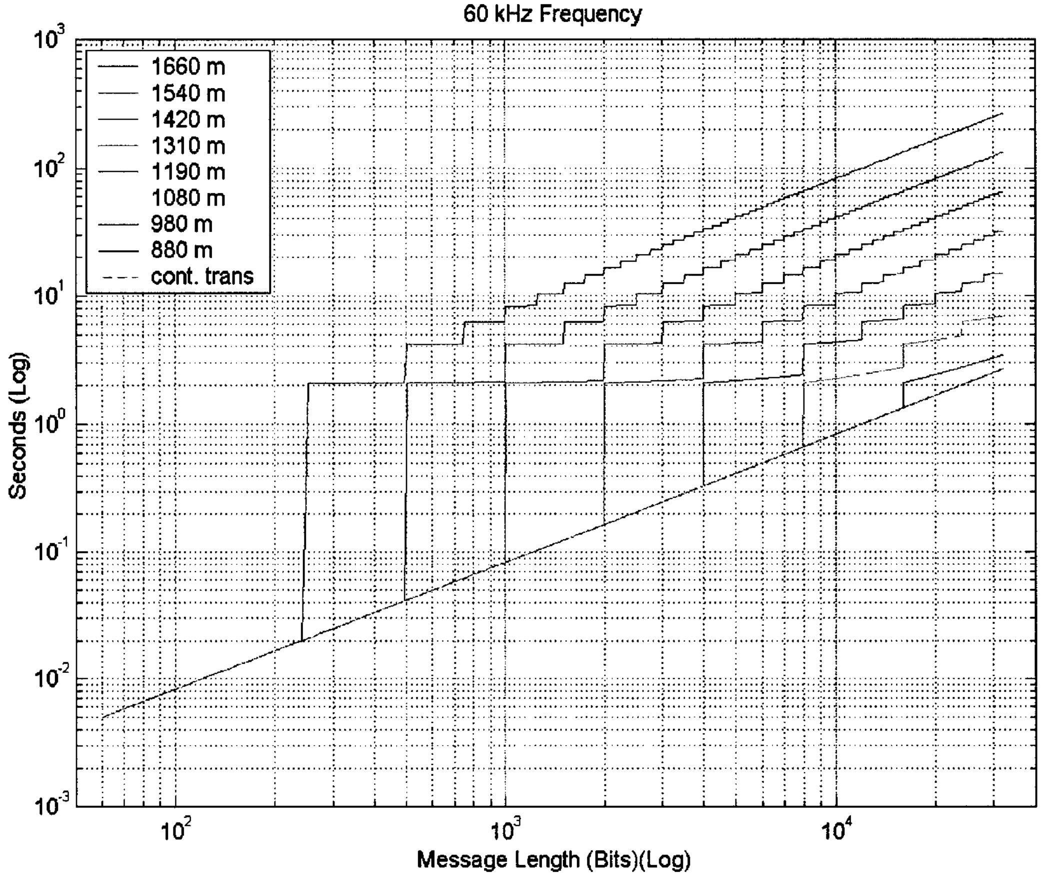


Figure 9.4 - Message Transmission Time vs. Message Length for 60 kHz AUV Sonar

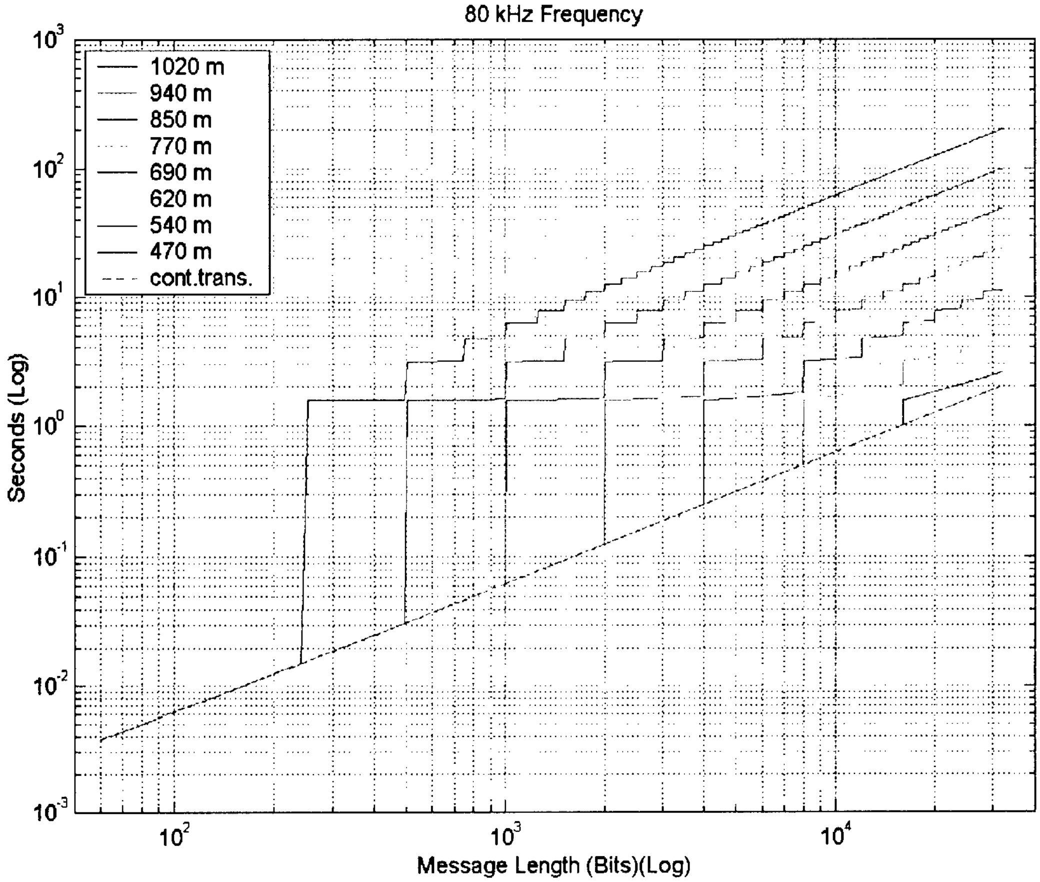


Figure 9.5 - Message Transmission Time vs. Message Length for 80 kHz AUV Sonar

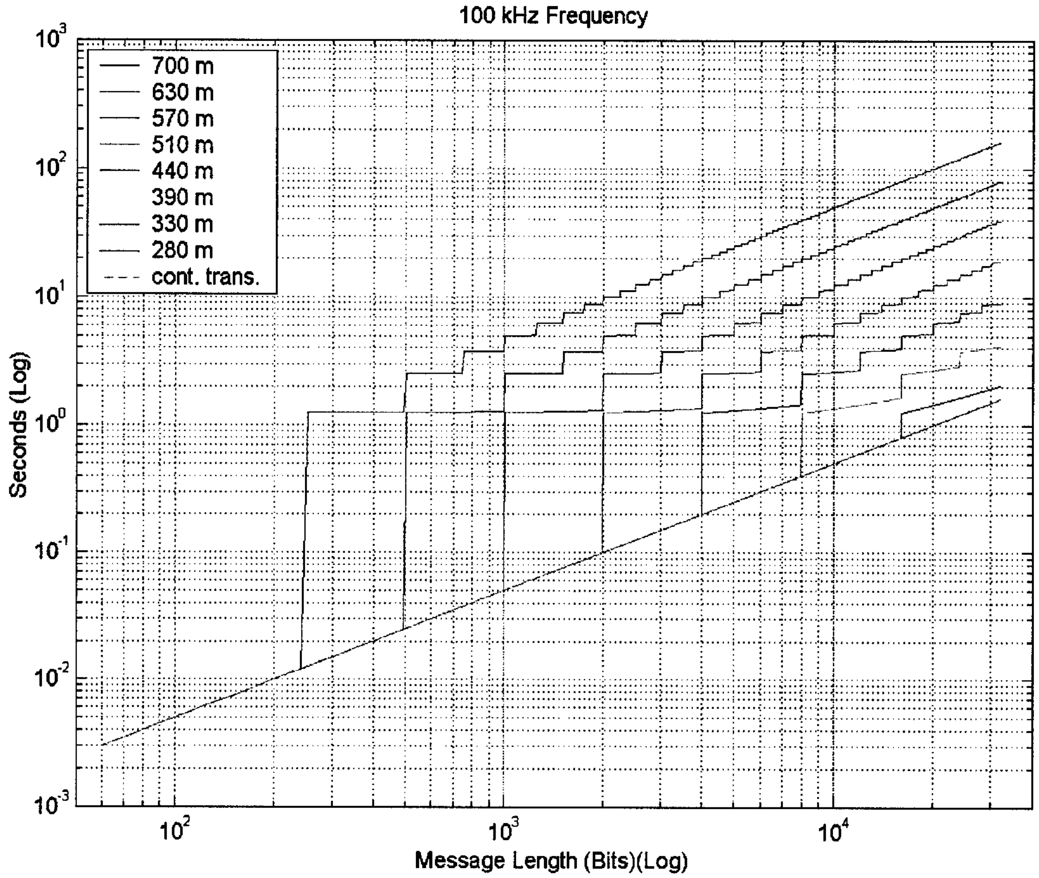


Figure 9.6 - Message Transmission Time vs. Message Length for 100 kHz AUV Sonar

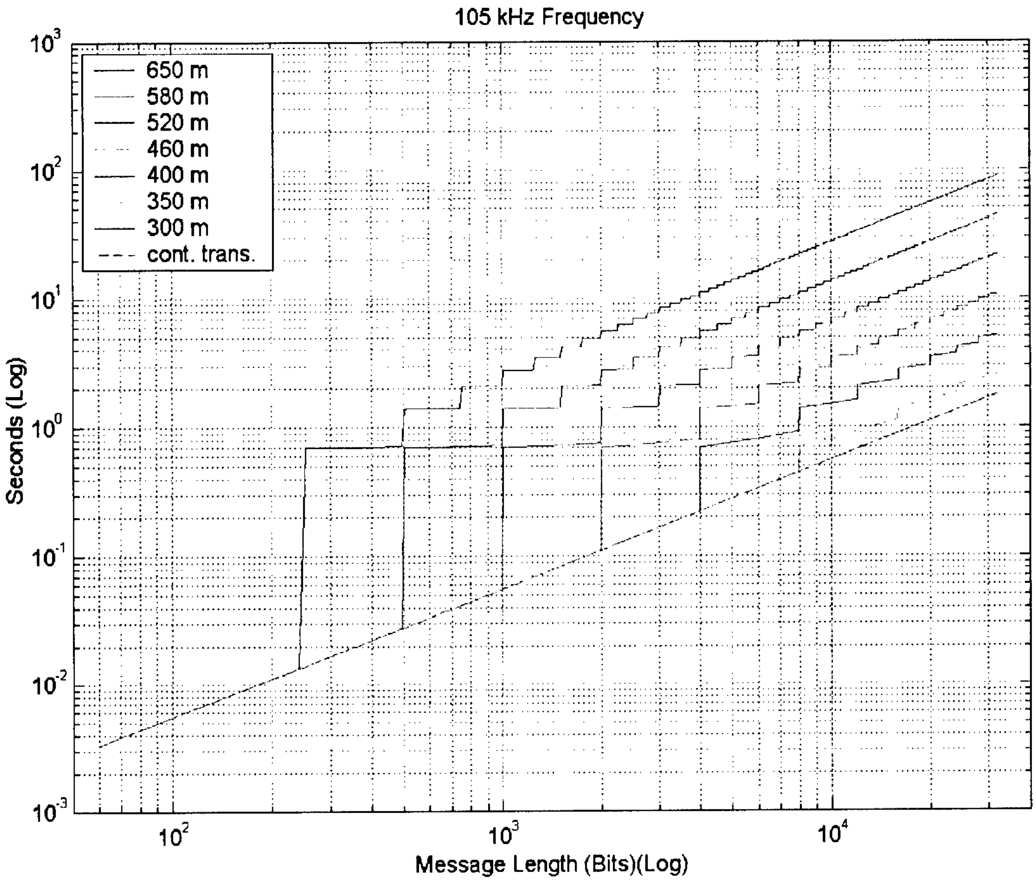


Figure 9.7 - Message Transmission Time vs. Message Length for 105 kHz AUV Sonar

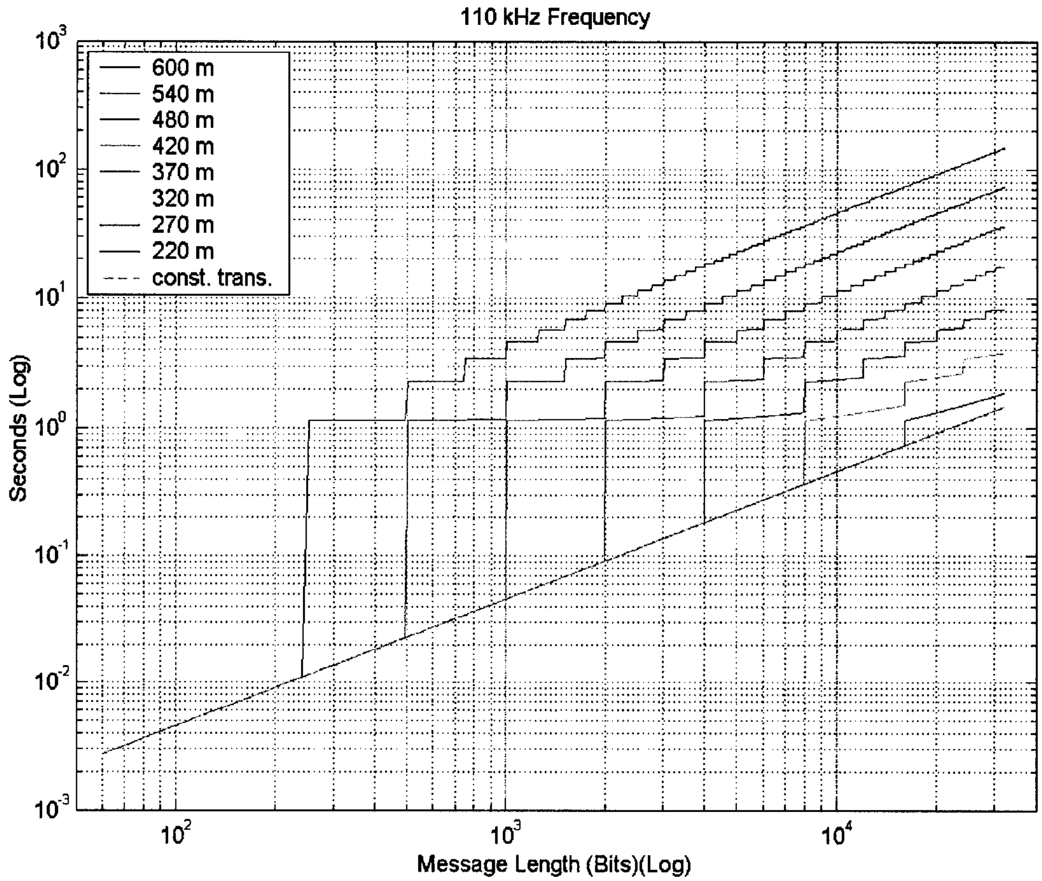


Figure 9.8 - Message Transmission Time vs. Message Length for 110 kHz AUV Sonar

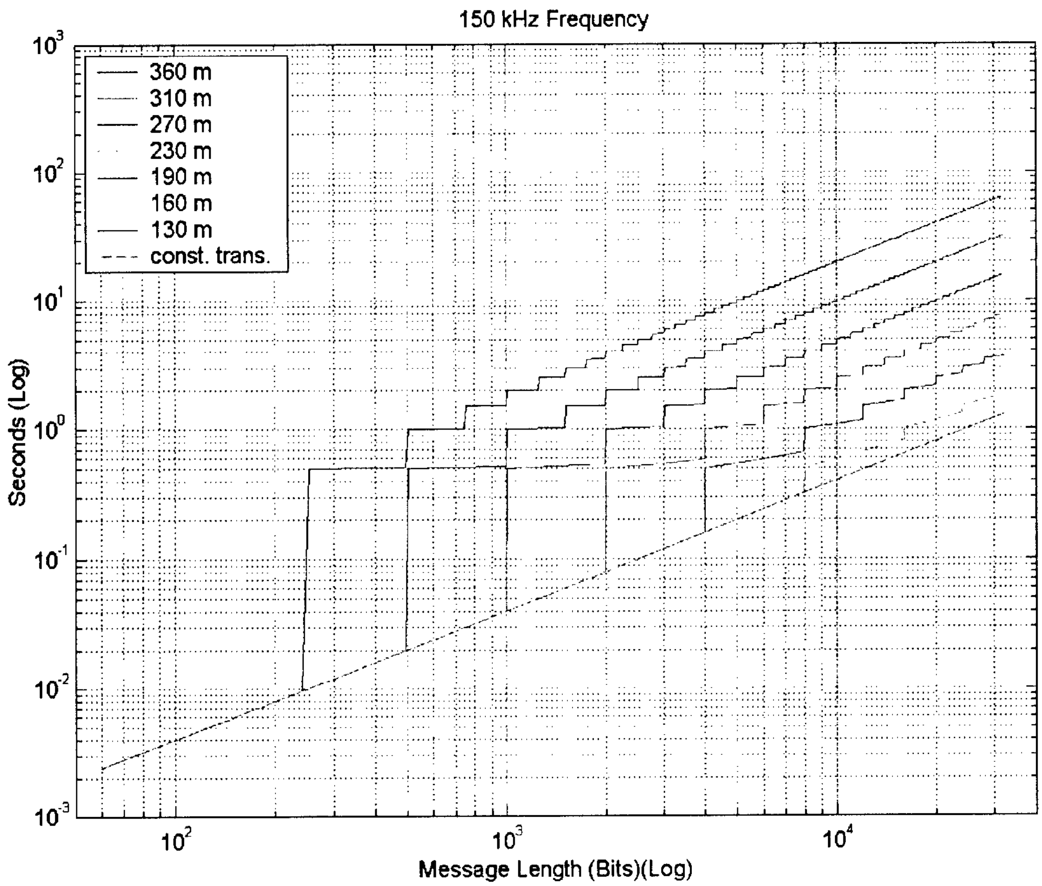


Figure 9.9 - Message Transmission Time vs. Message Length for 150 kHz AUV Sonar

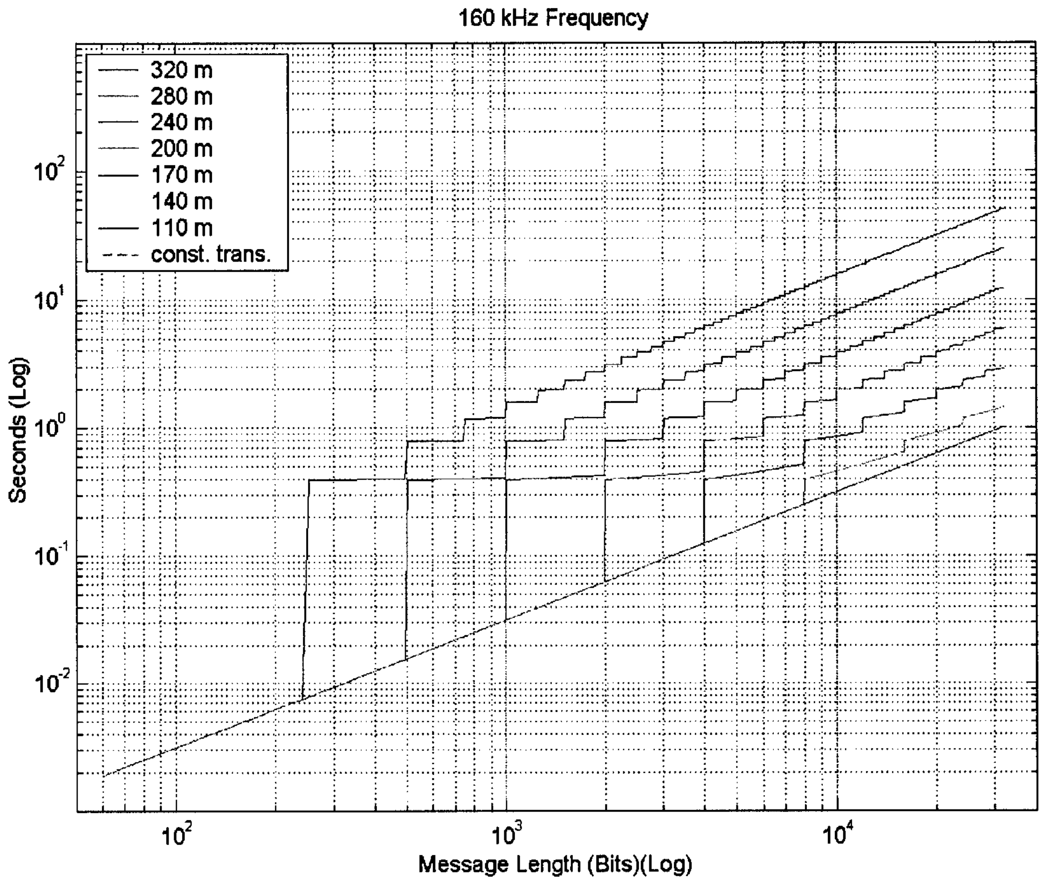


Figure 9.10 - Message Transmission Time vs. Message Length for 160 kHz AUV Sonar

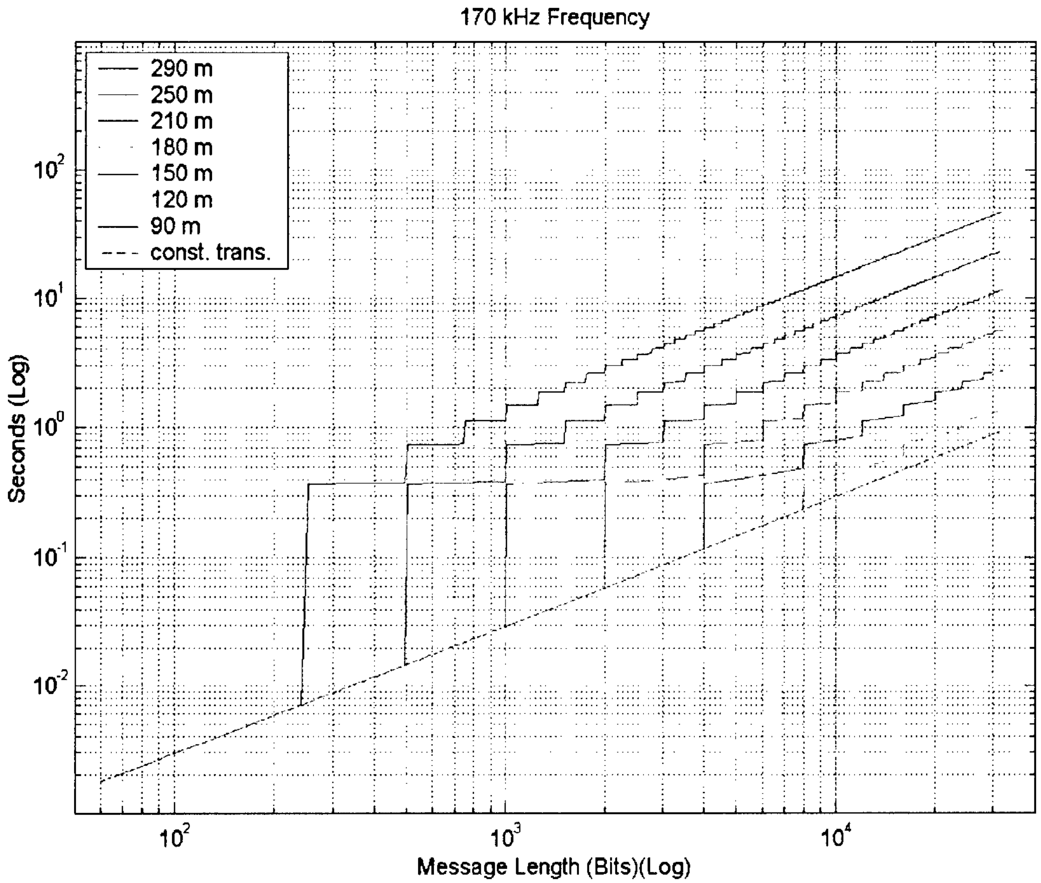


Figure 9.11 - Message Transmission Time vs. Message Length for 170 kHz AUV Sonar

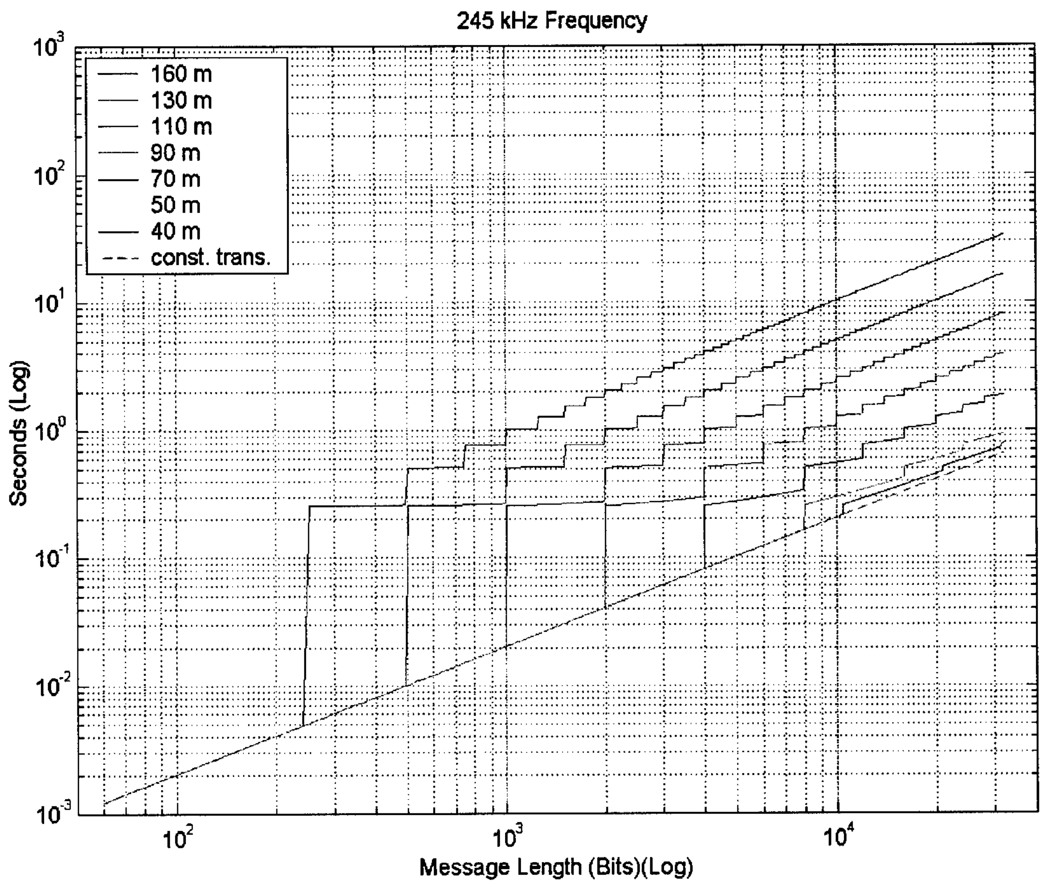


Figure 9.12 - Message Transmission Time vs. Message Length for 245 kHz AUV Sonar

10. Acronyms

A	
ACOMMS	Acoustic Communications
ADCs	Analog-to-Digital Converters
ADPCM	Adaptive Differential Pulse Code Modulation
AGC	Automatic Gain Control
ALAN	Acoustic Local Area Network
AM	Amplitude Modulation
AMMT	Autonomous Minehunting and Mapping Technology
AMS	Advanced Marine Systems
AODV	Ad-hoc On-demand Distance Vector
AOSN	Autonomous Oceanographic Sampling Network
ARQ	Automatic Repeat reQuest
ASK	Amplitude Shift Keying
ATD	Advanced Technology Demonstration
AUVs	Autonomous Undersea Vehicles
AWGN	Additive White Gaussian Noise
B	
BASS	Birmingham Acoustic Signaling
BCH	Bose-Chaudhuri-Hochquenghem
bps	Bits Per Second
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BSY-1	Submarine Integrated Sonar System (688I Class)
BW	Bandwidth
C	
C^2	Command and Control
CDMA	Code-Division Multiple Access
CoDMA	Collision Detection Multiple Access
COFDM	Coded Orthogonal Frequency Division Multiplexing
COTS	Commercial-Off-The-Shelf
CPB	Coherent Path Beamformer
CPU	Central Processing Unit
CSMA	Carrier Sense Media Access
CW	Continuous Wave
D	
DACs	Digital-to-Analog Converters
DARPA	Defense Advanced Research Projects Agency
DATS	Digital Acoustic telemetry System
dB	Decibel
DDG	Guided Missile Destroyer
DFE	Decision Feedback Equalizer
DFH	Differential Frequency Hopping
DLT	Digital Linear Tape

DPCM	Differential Pulse Code Modulation
DPLL	Digital Phase-Locked Loop
DPPM	Digital Pulse Position Modulation
DPSK	Differential Phase Shift Keying
DSDV	Destination Sequence Distance Vector
DSR	Dynamic Source Routing
DSP	Digital Signal Processor
DSSS	Direct Sequence-Spread Spectrum
E	
ECC	Error Control Coding
ENSTB	Ecole Nationale Supérieure des Télécommunications de Bretagne
EQSID	Equalization via System Identification
F	
FB	Feedback
FBE	Fleet Battle Experiment
FDMA	Frequency-Division Multiple Access
FEC-ARQ	Forward Error Correction Automatic Repeat reQuest
FF	Feedforward
FFF	Feedforward Filter
FHSS	Frequency Hopped Spread-Sprectrum
FM	Frequency Modulation
FOLMS	Fast self-Optimized Least Mean Squares
FOLMSPE	Fast self-Optimized Least Mean Squares Phase Estimator
FTF	Fast Transversal Filter
G	
GB	Gigabytes
GFLOP	Gigafloating Point Operating per second
GPS	Global Positioning System
GUI	Graphical User Interface
H	
HF	High Frequency
Hz	Hertz
I	
IC	Inner Code
ID	Identification
IFREMER	Institut Français de Recherche pour l'Exploitation de la Mer
I/O	Input/Output
ISI	Intersymbol Interference
J	
JMCIS	Joint Maritime Command Information System
K	
kbps	Kilo Bits Per Second
kHz	KiloHertz

km	Kilometer
L	
LE	Linear Equalizer
LF	Low Frequency
LMS	Least Mean Squares
LPD	Low Probability of Detection
LPI	Low Probability of Interception
M	
m	Meter
MA	Moving Average
MACA	Multiple Access with Collision Avoidance
MAI	Multiple Access Interference
MARIUS	Marine Utility System
MCM	Mine Countermeasures
MDPSK	M-ary Differential Phase-Shift Keying
MF	Mid-Frequency
MFSK	Multiple Frequency Shift Keying
MHz	MegaHertz
MIT	Massachusetts Institute of Technology
MLSE	Maximum-Likelihood Sequence Estimator
MMSE	Minimum Mean Squared Error
MOEs	Measures of Effectiveness
ms	Millisecond
MSE	Mean Squared Error
N	
NBOA	Narragansett Bay Operating Areas
NEU	Northeastern University
NSSN	New Attack Submarine, Nuclear
NUWCDIVNPT	Naval Undersea Warfare Center Division in Newport
O	
OEX	Ocean Explorer
OFDM	Orthogonal Frequency Division Multiplexing
ONR	Office of Naval Research
OOK	On-Off Keying
OQPSK	Offset Quadriphase-Shift Keying
P	
PAM	Pulse Amplitude Modulation
PC	Personal Computer
PCM	Pulse Code Modulation
PG	Processing Gain
PLL	Phase-Locked Loop
PPC	PowerPC
PPM	Pulse Position Modulation
PSK	Phase Shift Keying

Q

QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying

R

RAM	Random Access Memory
RAP	Reliable Acoustic Propagation
RF	Radio Frequency
RLS	Recursive Least Squares
RV	Research Vessel
RWLS	Recursive Weighted Least Squares

S

s	seconds
SFTF	Stabilized Fast Transverse Filter
SL	Source Level
SNR	Signal-to-Noise Ratio
SPM	Sequence Position Modulation
SQS-53C	Surface Ship Bow Sonar System
SSB	Single Sideband
SSMA	Spread Spectrum Multiple Access
SSN	Attack Submarine, Nuclear
STV	Short-Term Variability

T

TCM	Trellis Coded Modulation
TDMA	Time-Division Multiple Access
TORA	Temporally Ordered Routing Algorithm

U

UAM	Utility Acoustic Modem
UDAT	Underwater Digital Acoustic Telemetry
URI	University Research Initiative
UUV	Unmanned Underwater Vehicle
UWA	Underwater Acoustic

V

VME	Versa Module Europa
VSTV	Very Short-Term Variability

W

WHOI	Woods Hole Oceanographic Institution

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13. ABSTRACT *(Maximum 200 words)*
The purpose of this study is to provide an analysis and assessment of the state-of-the-art of underwater acoustic (UWA) data communications technology with application to multiple vehicle operation in support of shallow-water mine countermeasures (MCM) operations. This study is intended to provide system architects with a reference to what the current state of acoustic communications technology can provide and to support the execution of system architecture trades. Included are discussions and analyses that may be used to predict the performance, strengths, and limitations of different approaches to acoustic communications in facilitating the overall MCM system approach.

Not only does the underwater environment constrain the data rates achievable but it also limits the maximum ranges achievable. Acoustic attenuation in the ocean is frequency-dependent with higher frequencies propagating shorter distances. The requirement to provide connectivity to submerged Naval assets over very long ranges is unlikely to be met directly with a single acoustic link due to the complex effects of acoustic propagation and limited propagation ranges. One solution is to create an UWA network of communications nodes. Another solution to this requirement is to use a buoy to convert acoustic signals to and from radio frequency (RF) signals. This solution provides real-time two-way communications to distant surface ships, aircraft and/or satellites that can act as relays to integrate undersea communications into the RF-based communications network.

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