HIGH-FREQUENCY PROPAGATION FOR ACOUSTIC COMMUNICATIONS

MICHAEL B. PORTER, PAUL HURSKY, AND MARTIN SIDERIUS
Science Applications International Corporation
1299 Prospect St, Suite 303, La Jolla, CA 92037, USA

VINCENT K. MCDONALD, PAUL BAXLEY
SPAWARSSC
San Diego, CA 92152-6145, USA

In recent years there has been great progress in developing undersea wireless networks. The physical layer of these networks is generally an acoustic link operating in the 10-50 kHz band. Interestingly, our understanding of the acoustic propagation has not kept up with the elegant signaling schemes used to transmit the information. For instance, one common system uses adaptive equalizers to not only recombine the ocean multipath but to track the ocean dynamics. The tracking occurs on a millisecond time scale. Predicting system performance then requires an understanding of the transmission loss, the noise background, and the dynamics of the multipath. Here we use the term 'transmission loss' loosely, glossing over subtleties about the coherent and the incoherent field, which also play an important role in the system performance. This talk will summarize the issues and our knowledge about them, drawing upon results from an extensive set of sea tests conducted under the SignalEx program.

1 Introduction

It is not hard for today's cell phone user to appreciate the benefits of wireless connections. On land, 802.11a,b, and g are rapidly becoming established for wireless local area networks and Bluetooth™ for more specialized connections. In the ocean, this sort of technology is just emerging; however, the pay-off is arguably greater, given the costs of laying cable at sea.

One proposal under development would deploy a wide-scale network to monitor sewage outfall, plankton blooms, and nutrients, to observe the ocean ecosystem. The flexibility in terms of adapting the network configuration (including autonomous vehicles) to evolving features is a major benefit of the wireless structure. In the ocean, of course, wireless normally implies an acoustic rather than electromagnetic link.

On the human level this application presents another communication problem, namely that between the communications engineer and the underwater acoustician. The first group is generally only casually interested in the channel characteristics; the second group is only casually interested in the modulation schemes. Actually, the techniques from communication theory are very clever in finessing the effects of a varying multipath environment. However, we believe there is much to be gained for underwater acoustic communications in bridging these fields. That is the goal of this short paper.

2 Communications Schemes

Before we can discuss the role of the acoustic channel, we should understand something about the existing communication schemes [1,2]. There are many variations on these themes but a high-level taxonomy breaks these into *non-coherent* and *coherent* schemes. We will discuss some of the more widely used approaches. In general, increased receiver complexity provides increased bandwidth efficiency, i.e. we can get more bits through per second if we work harder.

It is useful to remember that with sufficient SNR, there is no limit on the achievable bit rate (we assume a discrete-time, memory-less, additive Gaussian noise channel). To understand this, consider a pulse-amplitude transmission scheme where the amplitude of a pulse is given by a 32-bit integer. (We assume the SNR is such that the 32-bits are resolvable.) If we have a 1-kHz bandwidth, we can send a new pulse every millisecond, providing 32 kbits/sec or 32-bits/Hz. Since the data rate is proportional to the number of bits we use to encode the pulse amplitude we can transmit as many bits as we want in the 1-kHz band.

Noise of course makes our ability to distinguish levels less reliable and as the SNR drops the bit-rate drops. Thus, the channel capacity, C, in bits/sec is a function of bandwidth, W, and SNR:

$$C = W \log_2 (1 + SNR):$$

Higher data rates can therefore be achieved by increasing one or the other. Practically speaking, bandwidth efficiencies in a wide variety of systems tend to be O(1) bits/Hz.

2.1 MFSK (Multiple Frequency-Shift Keying)

Imagine playing a piano underwater. Obviously, the pattern of notes can be used to encode an arbitrary data stream. Of course, the multitude of echoes could make it difficult to hear clearly. Simply playing one chord and waiting for all the reverberation to die down before playing the next can solve this multipath problem. This is the basis for *Multiple Frequency-Shift Keying* and is the technique of the widely used Benthos modems. It is also the basis of the so-called Interoperability Standard proposed as a sort of Esperanto for acoustic modems (which in turn may speak their own language for more floral expression).

The details of one particular scheme may be of interest to the reader. We synthesize a piano with 128 keys spaced 40 Hz apart from 8 kHz up to 13.2 kHz. The upper and lower 4 tones are reserved for pilot tones that can be used to compensate for Doppler. Since 40-Hz tones are readily resolved in an FFT over a 25-msec time frame, we strike the notes for 25 msec. We then allow 0-200 msec of clearing time before striking the next chord. In MATLAB the demodulator is little more than a one-line call to generate the spectrogram of the received waveform as seen in Fig. 1. (These data are taken from a SignalEx [3,4,5] test conducted near Ship Island, Mississippi.) One small trick here is that ‘1 of 4’ coding is used. This means that every 2 bits of the data stream are used to select within blocks of 4 tones. The decoder then decides which of the 4 tones is loudest rather than trying to address the SNR-dependent question of whether one specific tone has been struck.

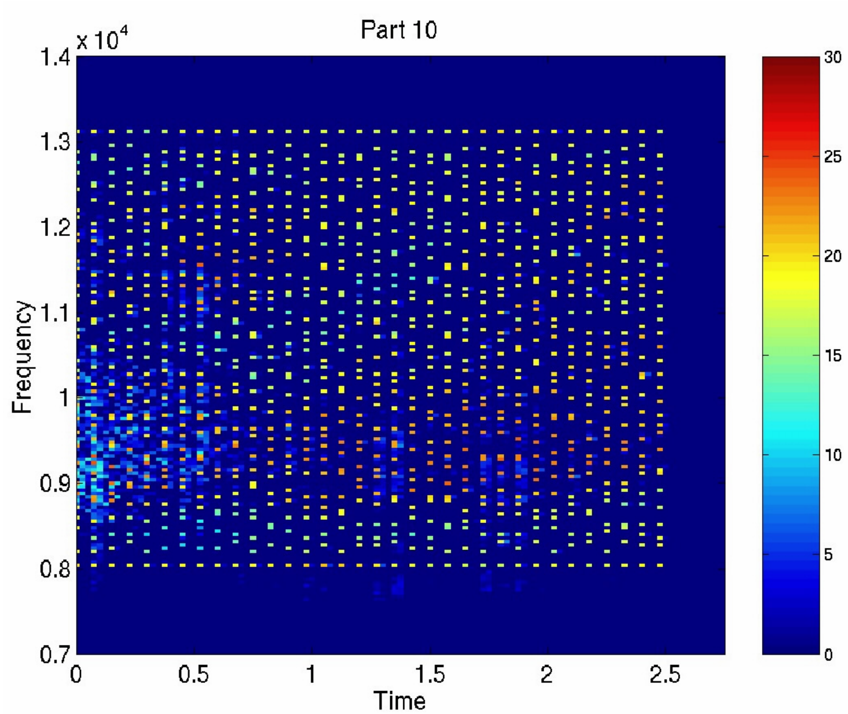

Figure 1. Spectrogram of an MFSK reception at a range of 5 km (SignalEx/Ship Island).

To get a feel for the data rates, consider that we have 120 tones available every 200 msec. Since we are using 1-of-4 coding, 30 tones are played in every chord and each tone encodes 2-bits. That implies 60 bits every 200 msec or 300 bps. It turns out that that amount of channel clearing time is seldom needed in shallow water and in Ship Island no clearing time at all was needed so a data rate of 2400 bps was reliably achieved. (You can see in Fig. 1 that there is little evidence of echoes.)

2.2 Direct-Sequence Spread Spectrum (CDMA)

The technology of Direct-Sequence Spread Spectrum (DSSS) or Code Division Multiple Access (CDMA) is embedded in the Telecommunications Industry Associate standard IS-95 and is widely used in cell phones. The technique begins with a repeating stream (a so-called *spreading code*) that looks like a random pattern of +/-1's. In the standard terminology these are called *chips*, a term that distinguishes them from the bits of the data stream. The chips flip (or do not flip) the phase of a sine wave or carrier and this is referred to as *Binary Phase-Shift Keying*.

For our testing we typically use a 12-kHz carrier with 4000 chips/sec. This implies we flip (or more precisely, control) the sign 4000 times per second, and thereby generate a transmitted wave with energy predominantly in the 8-16 kHz band. (The bandwidth spreading is proportional to the chip rate.)

The process described so far is not allowing us to transmit data. To get our data stream in we simply multiply the bit stream of our data by the chip stream. (We are considering our bit stream as +/-1 so that multiplication makes sense.) Typically our bit stream will run at 200 bps, which of course is much slower than the 4000 chips/sec. To summarize, the bits flip the sign of the chip stream 200 times/sec; the chip stream flips the sign of the carrier 4000 times/sec.

To recover the data on the receiver end, we correlate, i.e. matched-filter, the received time series with a BPSK signal encoded using the spreading code directly. The result is the carrier wave with phase determined by the bit stream. Note that where MFSK beats the multipath by waiting for the channel to clear, CDMA does so by the intrinsic orthogonality of the spreading code. There is an initial acquisition process to synch the receiver matched-filter to the frames of the data. However, once that is done a later (or earlier) arriving multipath has a piece of the spreading code that is de-correlated from the dominant path used in the acquisition. With that understanding we can see that as the multipath spread increases we will generally need to reduce the bit rate so that the time-window for each bit never sees an echo of a previous bit.

In a short space, it is difficult to do justice to the CDMA approach. It should be noted that the standard procedure also includes a RAKE receiver that decodes the received signal on a tapped-delay line and recombines the taps. This process provides a further benefit in a fading multipath environment. Multiple-access is handled by simply assigning a different spreading sequence for each user. For moving sources, a Doppler tracker is also typically required to account for drifts in the frame positions. Finally, the typical implementation is DPSK (Differential Phase Shift Keying) rather than BPSK. Benthos Corp. and Nautronix Corp. are currently testing DSSS/CDMA in their commercial modems.

2.3 Coherent Schemes with Adaptive Equalizers

Getting the highest data rates through a channel typically requires careful attention to channel equalization. On a superficial level one might imagine that for a channel that propagates lower frequencies better than high frequencies that you simply boost the high end at the receiver to recover the natural sound. The problem is actually far more difficult. The channel transfer function in the ocean is just the Fourier transform of the impulse response. Consequently it has a variation in the frequency domain that occurs on a scale proportional to the reciprocal of the multipath spread.

To take some round numbers, in the 8-16 kHz band we may see up to 200-msecs spread in a shallow water channel. That implies a 5-Hz scale to the variation of the transfer function. Over an 8-16 kHz band one may then have a very large number of deep fades. The equalizer must compensate for these variations across frequency. It must also track the time variations of the transfer function. This is done through an adaptive equalizer, which is actually implemented in the time-domain as a tapped-delay line. The output of the adaptive equalizer is roughly speaking a version of the received waveform with the multipath effects eliminated.

3 Implications for High-Frequency Modeling

Numerous sea tests in the lower frequency band have established that acoustic propagation can usually be understood in terms of a sequence of distinct echoes. These echoes are typically fairly undistorted copies of the transmitted waveform. There are far fewer published results in the 8-16 kHz band and one may well-wonder if an echo off a rough bottom or clouds of bubbles near the surface leaves much of a signal. A representative sample shown in Fig. 2 illustrates that even out to ranges of 6 km, one may see a clear multipath structure. (The colorbar is intensity in dB with an arbitrary reference.) These data from the SignalEx/New England Shelf test were taken during a fairly rough sea state with wave heights of approximately 3 m.

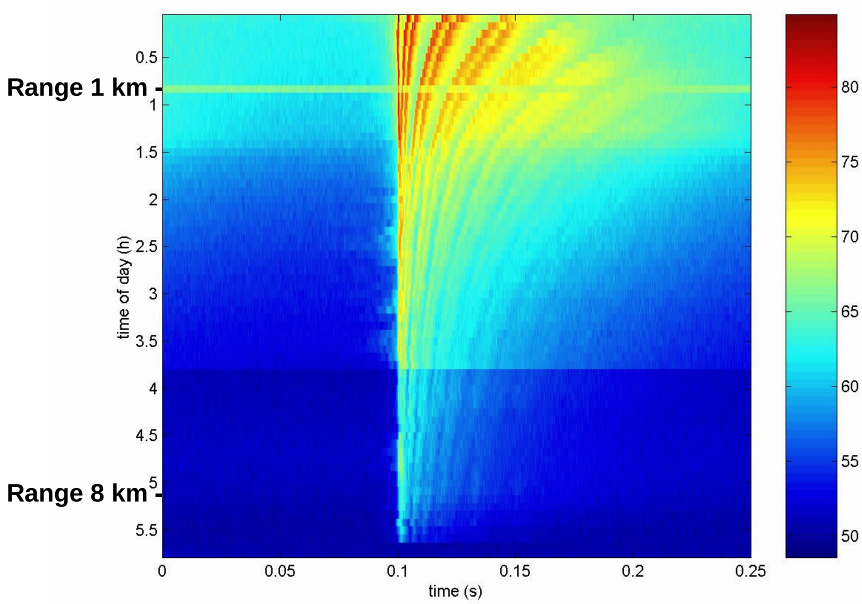

Figure 2:Variation of impulse response in a shallow-water channel (depth=45m) from SignalEx/New England Shelf. The response was measured as the source drifted out in range.

The first-order predictor of communications performance is simply SNR. In upward-refracting conditions the sole arrival is sometimes the surface reflection. A rough sea forms a murky acoustic mirror and simultaneously generates a lot of ambient noise. Network outages have been clearly correlated with wind speed.

The second key consideration is multipath spread. All of the above described modulation schemes are sensitive to this in some way. The MFSK approaches requires channel clearing time based on the spread; the DPSK scheme suffers from intersymbol interference; and the PSK approach with adaptive equalizers requires a tap-delay line accomodating the spread (and even so, will degrade in trying to equalize a more complicated channel). Of course, multipath spread can vary significantly from one environment to another. For instance, Fig. 3 shows a result taken during the SignalEx/Ship Island test in very shallow water (< 5m) and shows a multipath spread of about 2 msec. This type of channel is an easy one for most modulation schemes.

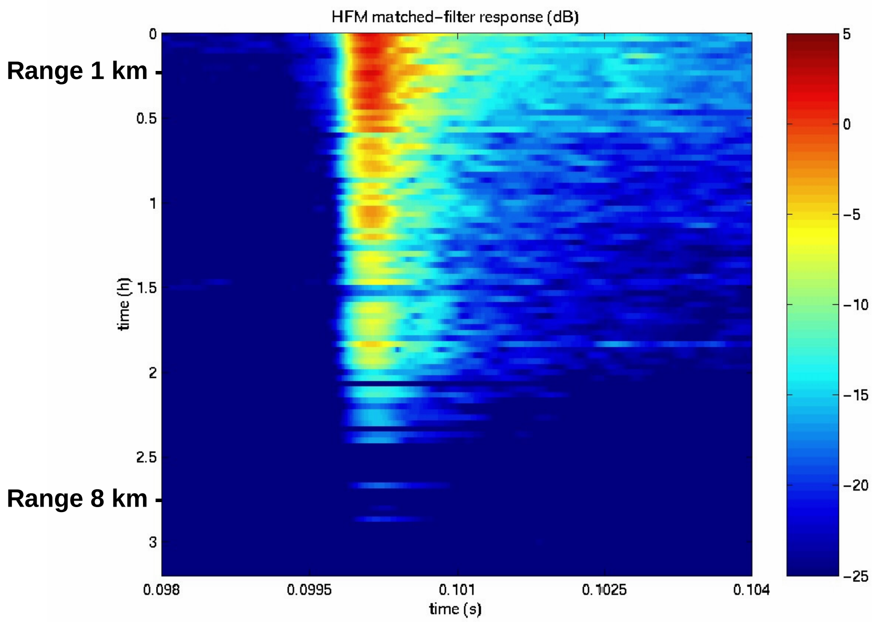

Figure 3: Variation of channel impulse response in a very shallow water (<5 m) site (from SignalEx/Ship Island, Mississippi).

The final consideration is the variability in the channel impulse response. This is sometimes characterized in terms of Doppler spread and can be measured by transmitting a tone and then forming its spectrogram. An example is shown in Fig. 4 from ModemEx conducted off the coast of San Diego. Note that the 10-kHz tone has been both shifted and spread. The former is principally due to the fact that the source is moving; the latter probably represents fluctuations due primarily to swell. It is typically easy to compensate for Doppler shift.

Variability is obviously a factor for MFSK schemes in that it limits the narrowness of the individual tones. The effects on other schemes are much more subtle. For instance, the adaptive equalizers are designed to track the dynamics of the multipath structure. To understand how they lose track, we must understand how the equalizer reacts to very short scale variations in the multipath structure. With a moving source, those variations can be significant on a 100 msec time scale.

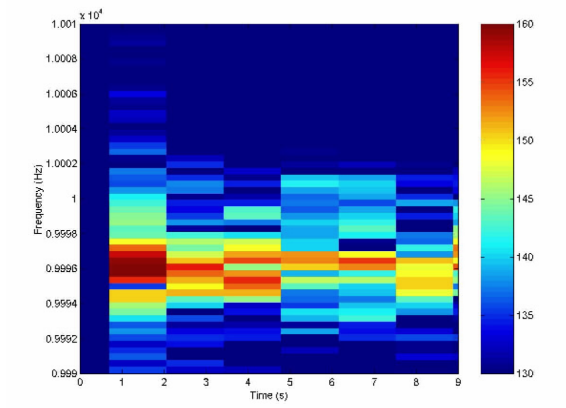

Figure 4: Doppler shift and spread of a 10-kHz tone (from SignalEx/Pilot Test San Diego).

Direct measurements of the time-dependent impulse response are also useful in characterizing the variability. The left panel in Figure 5 shows such a result from the SignalEx/Point Loma test with chirps transmitted every 250 msecs. We see that even at the 12-kHz carrier frequency the impulse response is very stable from ping to ping. Repeating the process 2 minutes later we again get a stable impulse response. However, comparing left and right panels we can clearly see the evolution in time. A fixed source/receiver geometry like this is clearly a much easier situation than one where there is relative source/receiver motion, e.g. in communicating to an AUV. Typically we find that with drifts of just 0.5 m/s the impulse response varies significantly between pings separated by 250 msec.

4 Summary

Communications engineers have developed clever schemes to mitigate the effects of a time-varying channel. The schemes have been optimized to yield the best performance subject to constraints on complexity. However, it is important to realize that understanding the channel characteristics is still a very important problem if we wish to predict the actual achievable performance. For instance, in deploying a network we have to select a spacing between nodes. That in turn will depend on our best guess of achievable data rates relative to bit-error rates.

The state of the art on the modeling end is fairly limited. Surface and bottom loss models in the 10-50 kHz band are available in HFEVA[8]. However, the dynamics of the ocean impulse response (usually characterized by the so-called scattering function) is also important. There are very few measurements of the scattering function in the ocean.

Acknowledgements

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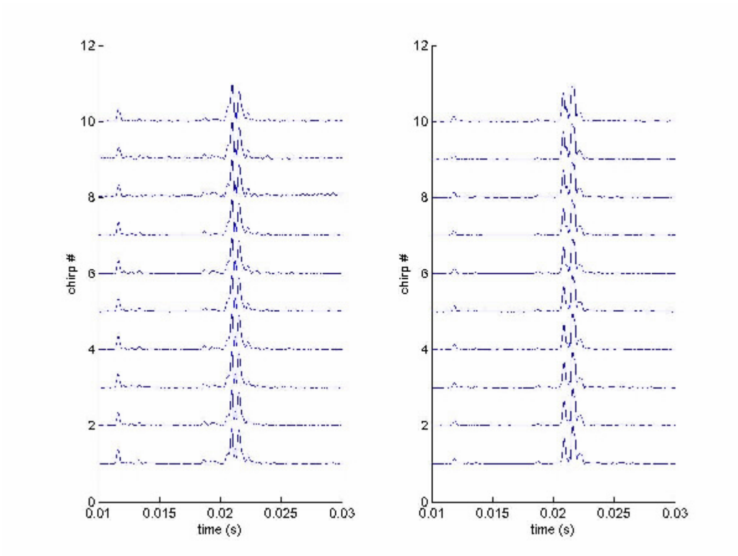

Figure 5: Successive snapshots of the impulse response taken every 250 msec. Right-panel recordings begin 2 minutes after left (from SignalEx/Point Loma, San Diego, California).

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