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DEMONSTRATION OF A LOW-FREQUENCY, LONG-RANGE ACOUSTIC COMMUNICATIONS SYSTEM

An acoustic communications system employing low frequencies to attain greater communications ranges is described. The waveforms are incoherent so that greater reliability can be obtained at the expense of potentially higher bit rates. The waveform generation process is described as are the results of three at-sea expeditions in which the system was tested in both bottom-limited and deep water conditions as well as in surface duct and convergence zone environments. The primary objective of the at-sea tests was to measure the attainable data rate from any fixed or moving transmitter to any fixed or moving receiver at a range of 100 nmi. The attainable data rate was measured as a function of several parameters, including the relative Doppler shift, error correction overhead, the acoustic environment, and trade-offs of time for bandwidth in the data modulation process.

INTRODUCTION

This article describes a low-frequency, long-range underwater acoustic communications link developed by APL. Development began in October 1991, and since then the system has been demonstrated during three different at-sea expeditions encompassing a variety of acoustic environments. The system provided highly reliable communications in both deep and shallow water as well as convergence zone and surface duct environments.

The motivation for this research is the need for a highly reliable communications system that operates over long ranges and in a wide variety of operational environments. High data rates have not been the driving requirement, although higher data rates without significant loss of reliability are of definite interest. Operational areas are expected to include the deep, open ocean (both summer and winter environments) in addition to shallower bodies of water.

This article addresses acoustic communications waveform design criteria, waveform generation equations, system design aspects, and the results of three at-sea expeditions conducted in the last two years.

WAVEFORM DESIGN

Overview

Figure 1 shows a generic acoustic communication waveform. For simplicity, the waveform shown contains only 16 bits, the values of which are 1010011111001000, as explained below.

The waveform consists of N_B blocks ($N_B = 2$ in this example), each of which is T_B seconds in duration. Successive blocks are separated by a T_G-second gap. Thus, the total waveform duration is $T_W = (N_B \times T_B) + [(N_B - 1) \times T_G]$ seconds. Each block spans the frequency band from F_{LO} to F_{HI} Hz. In each block, gated continuous-wave (CW; i.e., single-frequency) tones exist at F_{LO} and F_{HI}. These tones are used for automatic Doppler determination and correction, as discussed below. Each block contains N_L data bits ($N_L = 8$ in this example), each assigned to a unique, nonoverlapping frequency slice of B_L Hz width. Adjacent tones (bits) are separated by guard bands of B_G Hz bandwidth. Each bit is represented by the presence (bit is "on") or absence (bit is "off") of an upswept linear frequency modulated (LFM) tone that spans its B_L Hz frequency slice. The lower and uppermost B_L Hz bands in each block are separated from the CW tones at F_{LO} and F_{HI} by B_G Hz. Thus, the total bandwidth is $B_W = (F_{HI} - F_{LO}) = (N_L \times B_L) + [(N_L + 1) \times B_G]$ Hz.

Figure 2 shows a similar time/frequency display of an actual 32-bit acoustic communication received at sea after passing through 100 nmi of a surface duct acoustic environment. Such an environment constrains most of the acoustic energy to the top layer of the ocean relatively close to the surface. The display shows several channel effects on the signal that affect the way the signal needs to be processed to achieve satisfactory results. Three major effects are as follows:

1. Figure 2 shows a message having only 16 bits per block. More typical higher-data-rate messages have many more bits squeezed into the same bandwidth, resulting in a spectrum slice of only a few Hz per bit. The motion-induced Doppler shifts that are commonly encountered may change the frequencies of the higher-data-rate acoustic communications by a significant fraction of the slice bandwidth. Thus, an automatic Doppler compensation algorithm is incorporated into the system design. The two CW tones at the lower and upper frequency limits of the acoustic communication are used to measure the Doppler shift so that automatic compensation can be made.

2. Two of the higher-frequency tones in the first (lower) block are markedly faded because of the channel-fade

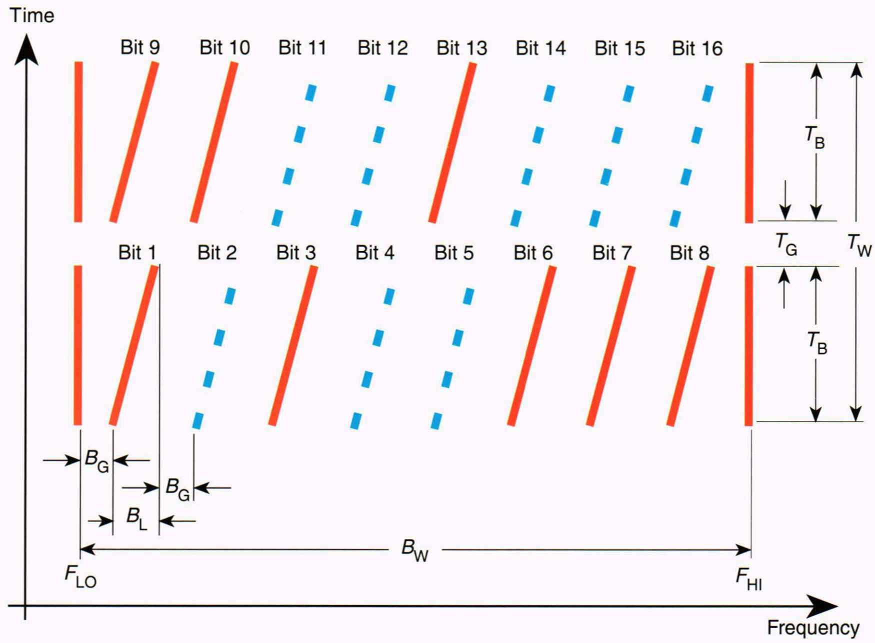

Figure 1. Generic acoustic communications waveform structure showing a 16-bit waveform. The waveform occupies a bandwidth of B_W Hz from F_{LO} through F_{HI} and a total of T_W seconds. The bandwidth of each tone corresponding to a bit is B_L Hz, and each tone is separated from adjacent tones by a gap of B_G Hz. Yellow dashed diagonal lines indicate the absence of a tone. The acoustic communication is divided into two 8-bit blocks separated by T_G seconds. Each block is T_B seconds in duration.

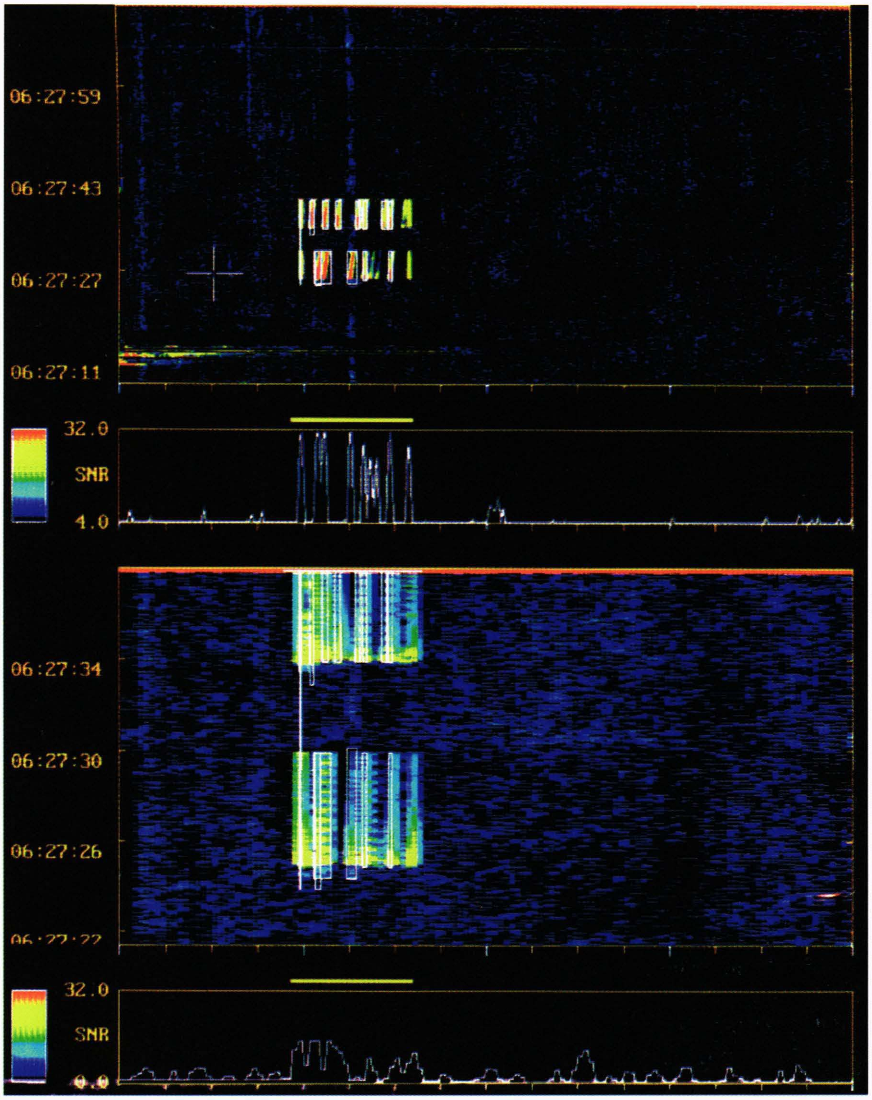

Figure 2. Time/frequency display of a 32-bit acoustic communication received at a range of 100 nmi from the acoustic communications source. The lower and upper displays show different time resolutions of the same waveform.

phenomenon in narrow frequency bands. Such fades can result in bits that are "on" being reported as "off." Similarly, narrowband noise can cause bits that are "off" to be reported as "on." To provide reliable communications under such conditions, an automatic error detection and correction algorithm is incorporated into the system. During the first at-sea test, 28 bits were added to each message for the purpose of error correction using a Bose–Chaudhuri–Hocquenghem (BCH) code.[1] These bits contain information that enables the receiver to correct up to 4-bit errors in the message. Later tests incorporated a more robust convolutional code described later.

3. A significant time spread of the waveform is seen in the lower display of Figure 2. Deep submergence paths result in a considerable amount of energy being received before the arrival of the peak energy traveling along the duct. The rectangular detection boxes show that the detector often triggers on the early, deep submergence arrivals. For waveforms having short-duration blocks, this detection time offset caused by multipath time spreading can result in the processed acoustic communications block being somewhat offset in time from the peak-energy block, thereby degrading performance. To compensate for this effect, the waveform is automatically searched for the exact time frame at which peak energy integrated over the duration of the block occurred.

Waveform Advantages

The method of encoding binary information onto an analog waveform, as shown in Figures 1 and 2, has the following advantages:

1. The receiver can be incoherent. In other words, the receiver merely detects energy in the frequency slices corresponding to each bit, and the presence or absence of energy indicates whether the bit is "on" or "off." Coherent signaling requires a stable phase reference that is phase locked to the carrier frequency. A stable phase reference is extremely difficult to maintain in a severe multipath environment. Incoherent signaling schemes do not require this stable reference and therefore are more robust in such environments. The disadvantage of using incoherent signaling is the greater signal-to-noise ratio required for a given probability of bit error. Thus, incoherent communications can be said to provide greater reliability at the expense of a lower potential data rate.

2. The spectrum of the acoustic communications waveform can be precisely constrained so as not to interfere with adjacent frequency bands of interest. Most coherent modulation schemes do not offer such precise spectral control.

3. Harmonics generated by clipping in the receiver are out of the fundamental acoustic communications band if 2 times the lower acoustic communications frequency is greater than the upper frequency. Thus, the detrimental effects of receiver clipping on data transfer reliability are minimized.

The examples shown in Figures 1 and 2 are acoustic communications messages of only 16 and 32 bits. The number of raw data bits in an actual message may be about 200. Twelve cyclic redundancy check (CRC) bits that guarantee message correctness are appended plus 6 flush bits required by the decoder. The resulting 218-bit stream is then encoded for error correction purposes by a rate 3/4 convolutional encoder,[2] which generates 4 encoded bits for every 3 bits of the message. Thus, the number of bits is increased to 291. A message having $N_B = 4$ may be used, which will send 291/4 = 73 bits per block.

Waveform Specification Parameters

The fundamental acoustic communications waveform specification parameters are shown in Figure 1. An additional parameter required to describe the waveform completely is $A_L(c)$, which is an amplitude scaling factor for each CW and LFM tone. The value of $A_L(c)$ is proportional to the inverse of the source output level. Its purpose is to flatten the spectrum of the acoustic communications waveform. A flat-spectrum constant envelope is required by the soft decision-decoding mechanism described later. The thresholds will vary from bit to bit if the signal spectrum is not flat. Also required is $\theta_L(c)$, which is a random phase variable uniformly distributed between 0 and 2π. This variable is constant for each CW and LFM tone. Its purpose is to minimize the effects of constructive and destructive interference between tones that otherwise causes large amplitude jumps in the time series output. If this parameter is not set appropriately, most of the time series waveform will be significantly below full scale, and the full power capacity of the sources will consequently not be used. An overall amplitude limit A_W is used to normalize the magnitude of the time series waveform to 1.0. Finally, the value of the bits being encoded is represented by $\chi(b)$, which equals 1 if the bit is "on" or 0 if it is "off."

Low-frequency modulated tones rather than CW tones are used to represent bit values because they provide an increase in the bandwidth occupied by a tone. The increase in bandwidth mitigates the effects of narrowband fades as long as the bandwidth of an LFM tone is greater than the width of the narrowband fade. By providing frequencies outside the bandwidth of the fade, the probability of a detection is increased. An LFM tone can therefore be thought of as providing frequency diversity.

In summary, the following parameters are required to define the acoustic communications waveform completely:

N_B = number of blocks (integer)
T_B = block duration (s)
T_G = time gap between blocks (s)
N_L = number of bits per block (integer)
B_L = bandwidth per LFM (Hz)
B_G = frequency gap between LFM's (Hz)
F_{LO} = lower frequency limit (Hz)
F_{HI} = higher frequency limit (Hz)
A_W = amplitude (waveform magnitude) limit
$A_L(c)$ = amplitude scale factor for each LFM and CW, where c = tone index ($c = 0$ for the CW at F_{LO}; $c = 1, 2, \ldots, N_L$ for the LFM's in the block; $c = N_L + 1$ for the CW at F_{HI})
$\theta_L(c)$ = phase offset for each LFM and CW (radians)
$\chi(b)$ = 1 if bit is "on" or 0 if it is "off," where b = the message bit index; [$b = 1, 2, 3, \ldots, (N_L \times N_B)$].

Waveform Generation Equations

The acoustic communication waveforms consist of the coherent summation of LFM and CW components organized into blocks. The time series data points $V_L(t)$ for one LFM and the time series data points $V_C(t)$ for the two CW tones can be obtained from

$$V_L(t) = \chi(b)A_L(c)\sin[2\pi F_L(c)t + \pi r t^2 + \theta_L(c)], \qquad (1)$$

and

$$V_C(t) = A_L(0)\sin[2\pi F_{LO}t + \theta_L(0)] + A_L(N_L + 1)\sin[2\pi F_{HI}t + \theta_L(N_L + 1)], \qquad (2)$$

where $F_L(c)$ is the start frequency for LFM tone c (Hz):

$$F_L(c) = F_{LO} + cB_G + (c - 1)B_L, \quad c = 1, 2, \ldots, N_L;$$

r is the frequency modulation rate for the LFM tones (Hz/s) = B_L/T_B; and $t = 0, (1/f_s), (2/f_s), \ldots, T_B$, which are the discrete time points (s), where f_s is the sampling frequency (Hz). All values of $V_L(t)$ and $V_C(t)$ for t not equal to $0, 1/f_s, 2/f_s, \ldots, T_B$ are zero.

Each block of the waveform is the point-by-point (coherent) summation of the LFM and CW tones that constitute the block. The time series data points $V_B(t)$ for the block for all LFM's that compose it can be derived from the relation

$$V_B(t) = R(t)[V_C(t) + \Sigma V_L(t)], \qquad (3)$$

where

$$R(t) = \begin{cases} 0.01 f_s t & t = 0, (1/f_s), (2/f_s), \ldots, (99/f_s) \\ 0.01 f_s (T_B - t) & t = [T_B - (99/f_s)], [T_B - (98/f_s)], \ldots, T_B \\ 1 & \text{all values } (99/f_s) < t < [T_B - (99/f_s)] \\ 0 & \text{all other values of } t. \end{cases}$$

The function $R(t)$ is a linear ramp that is applied to the first 100 data points (ramp up) and the last 100 data points (ramp down) in the block so that phase discontinuities at the block boundaries are avoided. The application of $R(t)$ also prevents abrupt source activation.

The time series data points $V_B(t)$ for each block are normalized to have a magnitude no greater than A_W. The normalized time series data points $Q_B(t)$ for the block are derived from

$$Q_B(t) = A_W V_B(t)/(\max|V_B|) \quad \text{for all } t, \qquad (4)$$

where max $|V_B|$ = the maximum absolute value of all data points in $V_B(t)$, for all t.

The total waveform is the point-by-point (coherent) summation of all the normalized blocks, each time-shifted by S_B seconds. Since the time-shifted blocks do not overlap in time, the contribution to the complete waveform at any given time will be due to, at most, one time-shifted block $Q_B(t-S_B)$. The time series waveform $W(t)$, which is the result of the waveform generation process, is derived from the following relation:

$$W(t) = \Sigma\, Q_B(t - S_B) \quad \text{for all blocks}, \qquad (5)$$

where S_B is the time shift for block B in seconds and equals $(B - 1) \times (T_B + T_G)$, where $B = 1, 2, \ldots, N_B$.

SYSTEM DESIGN

Figure 3 shows the principal modules in both the transmit and receive subsystems of the acoustic communications system. The waveform generation process is as follows:

1. Data items to be transmitted are determined by the system (message entry).
2. The corresponding bit stream is generated. A message of a nominal 200 bits has now been generated.
3. The 12 CRC bits are generated so that the receiver can automatically determine, with 99.998% reliability, whether the received message is exactly the same as that sent. An additional 6 flush bits are added for convolutional decoder purposes.
4. The 218-bit message is encoded to allow the receiver to correct errors introduced by the ocean medium. A convolutional encoding scheme of rate 3/4 is used (as noted earlier, this generates 4 bits for every 3 bits of the data message).
5. The resulting 291-bit sequence is then interleaved to reduce the adverse effect of burst errors on adjacent bits, such as may be caused by channel fades or narrowband noise spanning several bits in frequency. Adjacent bits are no longer adjacent after the interleaving process, thereby randomizing the effect of burst errors.
6. The time series waveform for the resulting 291-bit sequence is generated as described by the equations presented earlier.

Figure 3 also shows the functions performed by the receiver, which are as follows:

1. The receiver automatically detects and captures the acoustic communication. If the transmission meets certain fundamental criteria specific to acoustic communications (frequency coverage and duration), it is passed to the acoustic communications subsystem.
2. The acoustic communications subsystem determines a more exact start time for the transmission by searching in time for the maximum energy over the acoustic communications duration. Thus, the effects of multipath time spreading, which can cause the detector to trigger early, are minimized.
3. The CW tones embedded in the acoustic communication are used to determine precisely the relative Doppler shift for which the processor must account because of the tight packing density of the bits in the acoustic communication.
4. A high-resolution fast Fourier transform (FFT) process is used to sum the energy in each frequency slice of B_L Hz width. The energy values for the bits in each block are normalized to values between 0.0 and 1.0 so that bits that are "on" have energy values approaching 1.0, whereas those that are "off" have values close to 0.0.
5. The normalized energy values for the entire message are deinterleaved (i.e., unshuffled), thereby undoing the interleaving performed by the transmitter.
6. A Viterbi decoder[2] is used to determine the most likely message bit sequence. By operating on the floating-point normalized energy values, the decoder incorporates

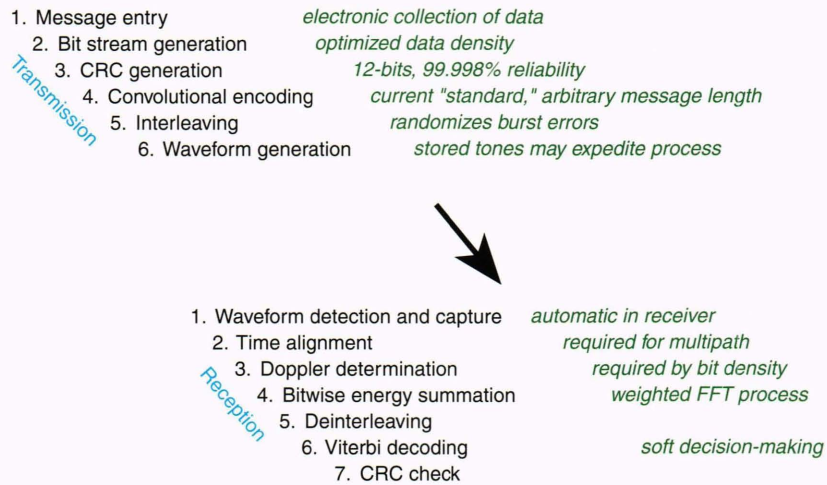

Figure 3. The principal modules in the acoustic communications transmit and receive processors. (CRC = cyclic redundancy check; FFT = fast Fourier transform.)

soft decision-making, which is a far more robust approach than if a prior decision had been made as to whether each bit was "on" or "off."

7. The CRC bits are used to determine if the bit sequence is correct. Acoustic communication "false alarms" are dismissed by this operation. Legitimate acoustic communications that are too garbled by the transmission path to be recognized are also dismissed under the premise that no information is better than false information, especially since the information transmitted via acoustic communications may be repeated periodically.

AT-SEA TEST RESULTS

The principal objective of testing acoustic communications at sea is to take in situ measurements to determine the reliable bit rate that can be attained in various acoustic environments of interest.

Reliable bit rates must be measured under a variety of conditions and functional parameters, including the acoustic environment, bandwidth/time trade-offs, and other variables in the acoustic communications encoding process. The goal is to achieve reliable communications under a wide variety of environmental conditions for communications waveforms generated in one specific manner yet to be determined.

To date, several trials have been conducted, and the outcome of each is described in the following sections.

Deep Ocean with Surface Duct

Messages sent via acoustic communications in the trial in the deep ocean with a surface duct were either 32, 64, or 128 bits in length. More than 90% of the 168 messages transmitted were successfully received. Success depended primarily on the signal-to-noise ratio at the receiver. The successful reception of all the 128-bit messages indicated that the channel bit-rate capacity was never reached.

During this first trial, a simple BCH code[1] was used to correct a limited number of errors in each message. Individual bit failure statistics (98.7% of the 10,272 bits transmitted were correctly received) were collected to guide the continued development of the system's error correction capability. The decision resulting from this trial, based in part on the complexities of normalizing the data, was to incorporate the more robust—although more complex—convolutional encoder described earlier for subsequent trials. The Viterbi decoder associated with the convolutional encoder allowed soft decision-making in which the signal processing does not have to decide before error correction which bits are "on" and which "off." Rather, soft values of "on" and "off" (i.e., 0.9 implies almost certainly "on," 0.3 probably "off," and 0.5 unknown) are fed directly to the decoder for a "best fit" code to the soft data.

Thus, this trial demonstrated that our acoustic communications concept is viable and provided direction for follow-on development.

Deep Ocean with Convergence Zone

A convergence zone environment provides a larger dynamic range in received signal level but a smaller time spread than that encountered in the ducting environment of the first trial because most of the acoustic energy tends to converge at certain ranges from the source. If the receiver is within such a convergence zone, the received signal level will be significantly greater than if the receiver were between convergence zones. Thus, the robustness challenges imposed on the acoustic communications system are different. A total of 171 pregenerated messages of varying bit rates and modulation schemes were transmitted during seven 30-min periods. A summary of the test results is presented in Figure 4.

The results were remarkably consistent. All acoustic communications having a bandwidth–time product of 3.75 per data bit or greater (150 messages out of 171) succeeded, without exception. The 21 remaining messages, which all had a bandwidth–time product of 1.99 per data bit or less, failed. The measure of bandwidth–time product per data bit is simply the bandwidth–time product for the entire acoustic communication (e.g., 12 s times 160-Hz bandwidth, which equals a total bandwidth–time product of 1920) divided by the number of raw data bits being transmitted, not including overhead bits for error correction or CRC.

The results just discussed were independent of the relative Doppler shift between the transmitter and receiver,

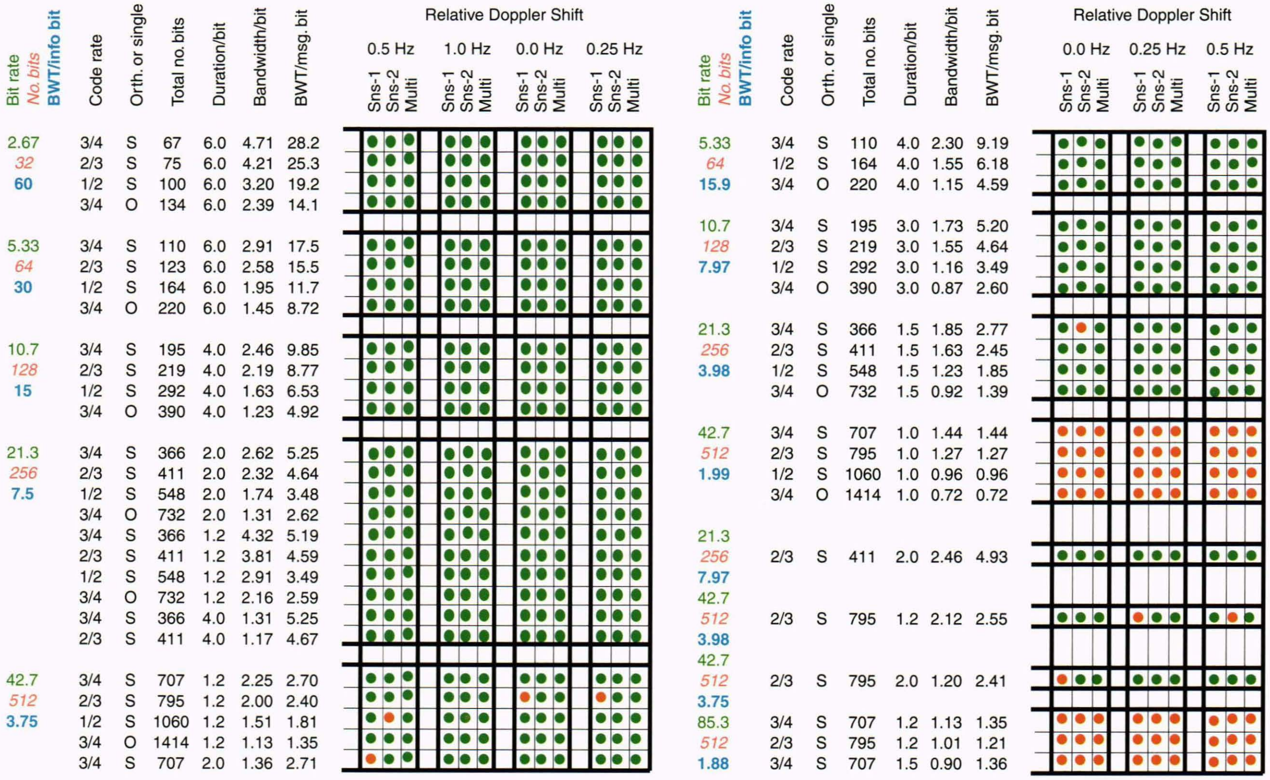

Figure 4. Acoustic communications results for the convergence zone environment. Seven 30-min sets of acoustic communications were received at the relative Doppler shifts indicated. Four sets contained 27 messages (left side of figure), and the remaining three sets contained 21 messages. Each message was decoded on three sensor sets: (1) a low-sensitivity sensor (Sns-1), (2) a high-sensitivity sensor (Sns-2), and (3) the sum of the two (Multi). Successful reception is indicated by a green dot, and a failure is indicated in red. Messages ranged in bit rate from 2.67 through 85.3 (indicated in green type), and corresponding bandwidth–time products per information bit (BWT/Info Bit) are shown in blue type. All messages having a BWT/Info Bit of 3.75 or greater succeeded, whereas those of 1.99 or less all failed. These results apply throughout the range of parameters indicated. The parameters that were varied included relative Doppler shift, the convolutional error correction code rate, data modulation scheme (orthogonal or single), and trade-offs of bandwidth for time for each data bit.

which ranged from 0.0 to 1.0 Hz. The amount of error correction overhead in the message (ranging from 35 to 110%) and any trade-offs of bandwidth per bit for time also did not affect the results. Furthermore, the results did not depend on whether the waveform used conventional bit encoding or an alternative "orthogonal" encoding scheme. In conventional encoding, the presence or absence of an LFM tone indicates whether the corresponding bit is "on" or "off," as described previously. In orthogonal encoding, two frequency slices are allocated for each bit, and an LFM tone resides in either the lower or upper frequency slice. The corresponding bit is declared "on" if the tone is in the upper frequency slice or "off" if it is in the lower.

These results demonstrated reliable acoustic communications under a wide range of conditions for data rates up to 40 bits/s.

Shallow, Highly Reverberant Conditions

The approach of transmitting a sequence of pregenerated messages spanning bit rates of interest was again repeated in shallow water. The waveforms in the sequence were used to evaluate bit-rate limitations in a shallow water environment, which was significantly different from the deep water acoustic environments encountered during the previously described duct and convergence zone trials. The breakdown of reliable communications in going from a bandwidth–time product of 3.75 per bit to 1.99 per bit was explored in more detail by using several messages having bandwidth–time products per bit between these two values.

This trial has just been concluded, and initial results indicate an acoustic communications reliability in highly reverberant shallow water that is comparable to that in deep water. An additional component of the trial was the transmission of about 700 individual acoustic communications during a 9-day period in both deep and bottom-limited, highly reverberant shallow water. Each message contained 180 raw data bits, resulting in a bandwidth–time product of 8.4 per data bit or about double the minimum requirement demonstrated during the scientific tests. More than 95% of the messages were successfully received.

SUMMARY

A low-frequency acoustic communications system has been developed and demonstrated at sea that achieves reliable communications over a range of 100 nmi at bit rates approaching those theoretically feasible for incoherent communications. The system sends binary data in the form of simultaneous LFM tones, each assigned to a unique slice of the acoustic communications spectrum. The received energy in each spectral slice is measured to determine if the corresponding bit is "on" or "off." The system incorporates automatic Doppler measurement and compensation as well as convolutional encoding with Viterbi decoding and soft decision-making for automatic error correction to overcome the effects of narrowband noise and fading. It also interleaves the data to minimize the adverse effect of burst errors on the Viterbi decoder, uses a 12-bit CRC code to assure data integrity, and incorporates automatic time alignment at the receiver to minimize the adverse effect of the significant time spread that can be encountered in long-range communications—especially through strongly ducted acoustic environments. An incoherent communications scheme was selected to achieve greater reliability at the expense of potentially higher data rates.

Test results show that waveforms of 160-Hz bandwidth and 12-s duration (i.e., a bandwidth–time product equal to 1920) can support the robust and reliable communication of messages having 512 raw, unencoded data bits at a range of 100 nmi. In general, communications having a bandwidth–time product of 4 per raw message bit (not including error correction and CRC overhead bits) were reliably received in a variety of acoustic environments, in both deep and shallow water, and at substantial ranges.

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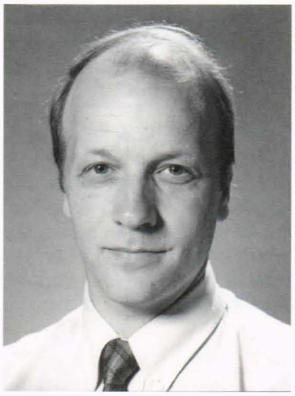
THE AUTHORS



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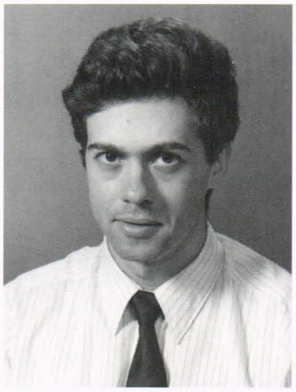

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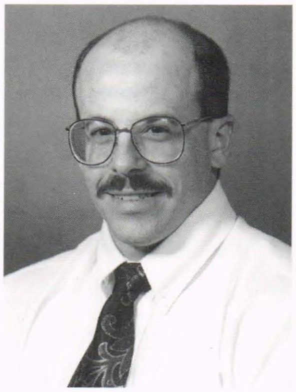

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