MFSK--THE BASIS FOR ROBUST ACOUSTICAL COMMUNICATIONS

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ABSTRACT

Subsea acoustic control and communications systems based on FSK modulation have been used in the offshore industry for the past few years. Prerequisites for reliable link operation included frequency and spatial diversity, signal coding, and channel equalization. These prerequisites limited previous systems to very low data rates.

In this paper, an acoustic modem capable of an order of magnitude increase in data rates is discussed.

This improved acoustic modem achieves higher data rates by simultaneous transmission of multiple tones using FSK modulation. While not new, until now this technique has been impractical because of the hardware constraints imposed by multitone signal processing. Rapid advancements in the areas of charge transfer devices and Fast Fourier Transform processors removed these constraints.

Tradeoffs discussed in this paper include the various signal-processing alternatives available and the use of error correcting or detecting codes to reduce the system undetected-error rate to acceptable values.

1. INTRODUCTION

Acoustic systems have proven their worth in offshore operations for vessel positioning, communications, and, particularly, BOP control.

Regulatory agencies have made emergency backup BOP control systems mandatory in some areas. Secure, reliable acoustic transmision of control commands and pilot-valve status is needed to control and monitor the BOP from a drilling vessel or support ship.

As the number of subsea completions increases, a wireless method for controlling and monitoring subsea wellheads and pipeline valves becomes more and more attractive.

These requirements show that using an acoustic modem as the core of a system that could be economically tailored to meet these applications would be useful to the offshore industry.

2. THE SHALLOW-WATER COMMUNICATION PROBLEM

One common but complex problem is operation in shallow water at long ranges at useful data rates, due to multipath phenomenon. The multipath phenomenon affects acoustic communication in several ways. First, it limits the rate at which signals can be sent sequentially on the same frequency. If they are sent too closely together, the second signal arrives at the receiver before the last of the preceding signal's arrivals have decayed to a low level. In communications, this "time-smearing" effect prevents the receiver from properly decoding the symbol.

Second, fluctuations in the lengths of the various paths result in frequency smearing of the signal. Because of this frequency-smearing effect, signals whose frequencies are too close together can overlap. This overlap, like time smearing, can result in improper symbol decoding.

A third effect of multipath is signal fading. Partial or total destructive cancellation of the signal caused by multiple paths varies with the previously described motion, causing fluctuations across the frequency spectrum of the received signal level.

To counter this problem the system design shall

- Provide sufficient time intervals between signals at any one frequency, to prevent time smearing
- Provide frequencies different enough to prevent frequency smearing
- Provide signal repetition with sufficient frequency diversity to overcome the effects of frequency-selective fading

The following sections of this paper address the design of signal waveforms and error control (coding) techniques needed to meet these requirements and to provide a reliable communications channel in the subsea acoustic environment.

3. SIGNAL WAVEFORM CHOICES

The signal waveform design begins with selection of a modulation technique for data transmission.

Possible schemes include Pulse-Position Modulation (PPM) Frequency Modulation (FM), Phase-Shift Keying (PSK), On-Off Keying (OOK), and Frequency-Shift Keying (FSK).

FSK can be treated as two interleaved OOK signals of different carrier frequencies. Simply, a tone of T time units is transmitted at one of two frequencies, F1 or F2, depending on its binary state, 1 or 0. FSK does not require a coherent channel for detection, and the bit-decision threshold level is independent of the carrier amplitude. It is precisely for these two reasons that FSK is preferred over the other modulation techniques in applications where fading is expected and coherent detection is not feasible. Thus, FSK was chosen as the modulation method for the acoustic modem.

4. SIGNAL WAVEFORM DESIGN

The signal waveform design used for the acoustic modem resulted from the pursuit of the following objectives:

- Minimum data rate of 40 bits per second
- Maximum undetected bit-error probability of 1×10^{-10}
- Reliable communication range of 3 nautical miles via a shallow-water path

Meeting the first objective would provide an order-of-magnitude improvement over the existing state of the art in reliable acoustic communications. The second objective would provide the same order of reliability as existing high-reliability communications systems. The third objective was predicated on meeting the control/telemetry range requirements of most subsea completions.

The basic channel bit-error probability of 1×10^{-5} can be attained through the use of frequency diversity, which simply means transmitting the same information at several different frequencies. Rayleigh fading is assumed for the channel, and each frequency channel is assumed to fade independently. Theoretical studies by Pierce (Reference 1) indicate that a bit-error probability of 1×10^{-4} can be attained on a channel with Rayleigh fading. It would use a frequency diversity factor of 4 at a signal-to-noise ratio of about 16 dB. This S/N ratio was considered practical, given assumptions of path loss, noise level, attainable source level, and data rate.

The maximum attainable signaling rate is controlled by the time-smearing effect of multipath and attainable S/N ratio at the desired operating range. Transmission tests through the shallow-water path indicated that channel-clearing times on the order of several milliseconds would be required before any particular frequency channel could be reused. Literature searches indicated that these times could extend to tens of milliseconds. The S/N ratio constant indicated that a signal duration of about 50 milliseconds would be required for reliable multitone communications. To maintain a constant power envelope during data transmission, "time-hopping" was introduced, which entailed transmitting each frequency-diversity channel in a different time slot. Since the baud interval used is 50 milliseconds and there are four diversity channels, a conservative clearing time of 150 milliseconds is available before a signaling frequency must be reused.

The scheme just described limits the signaling rate to 5 bits per second (i.e., a new data bit is started every 200 milliseconds). To increase the data rate, several bits are transmitted simultaneously by using additional frequency channels. The S/N ratio constraint and implementation requirements limited the number of bits sent in parallel to eight, or to one "byte." This number of parallel bits provides a data rate of 40 bits per second, which is the original objective.

The signaling concept is depicted in Figure 1. Each data bit in the eight-bit byte to be transmitted is mapped to one of two frequencies (channels) as a function of its logic state (Figure 2). Thus 16 frequency channels are provided per diversity band, and the four-way diversity technique increases the total number of frequencies required to 64. This use of multiple frequency channels to encode M bits per signaling interval is termed "M-ary" FSK or MFSK.

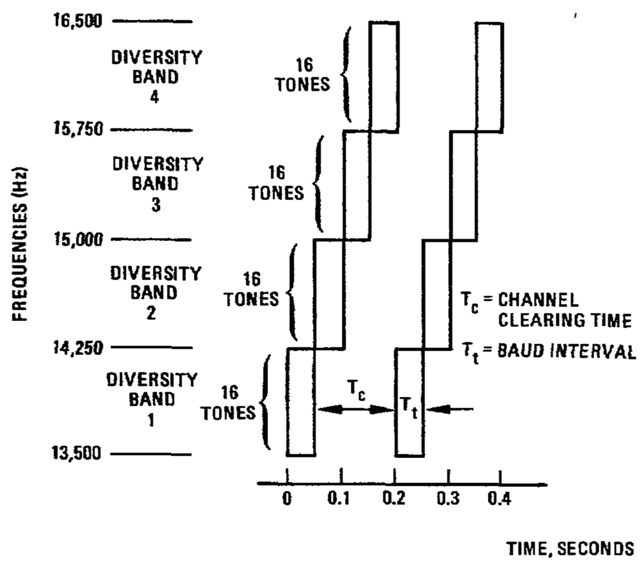

Figure 1. Signaling Approach

5. CODING

Two basic techniques are used to improve the reliability of data transmission over a communication channel. These are automatic-repeat-request (ARQ) and forward-error-control (FEC). ARQ tests a received data block for errors; if any errors are detected, then retransmission is requested through a feedback channel. FEC attempts to detect and correct errors in a received block of data without retransmission.

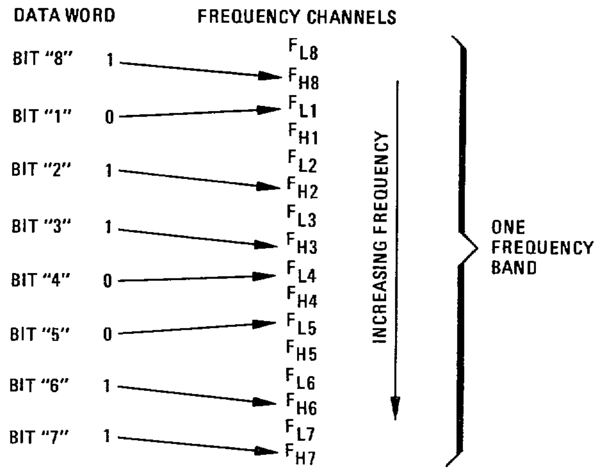

Figure 2. Byte Serial FSK

In both techniques, redundant bits are added to information bits to allow detection and correction. To attain a given undetected error rate, error-correction codes use more bits than error-detection codes. This is because, in addition to requiring bits for the error-detection process, it is necessary to identify the incorrect bit(s).

In a system designed to satisfy both reasonable data rates and low error rates, a tradeoff between the methods of error control (coding) is necessary. In order to compare the techniques, system throughput must be examined, not bit rate. If, for example, a 16-bit error-detecting code is added to an 80-bit message, the total length of the message will be 96 bits. At a bit-error rate of 10^{-4}, the probability of receiving an erroneous message is about 10^{-2}. Thus, one out of 100 messages will be received in error, requiring retransmission. Assuming stop-and-wait ARQ is used, the throughput may be computed from

$$T = \frac{n(1 - P(n))}{n + cxv} \times 100\% \qquad (1)$$

with T the throughput, n the block length, P(n) the block error probability, c the time delay from the end of transmission of one block to beginning of transmission of the next, and v the signaling rate. For our assumed block length of 96 bits, and further assuming a signaling rate of 20 bits per second and a round-trip delay of 4 seconds (range of 10,000 feet), the throughput is 54 percent. The effective bit rate is then 10.8 bits per second, on the order of the error-correcting code performance. If the message length is increased to 1,000 bits, then the throughput is 83 percent and the effective bit rate is 16.67 bits per second, substantially better than the error-correcting code. In addition, the undetected error rate has been reduced by a factor of 2^{-16}.

6. SYSTEM-DESIGN TRADEOFFS

The preceding sections have introduced the signaling and coding design concepts for the acoustic modem. This section will complete the design by discussing the optimum operating frequency, practical bandwidth values, and expected values of frequency and time dispersion in the multipath channel. The tradeoff between diversity and coding redundancy will be considered. These parameters will be used to compute the multipath-limited data rate to determine their acceptability in meeting the data-rate objective.

The acoustic carrier frequencies used for data transmission are centered about a frequency of 15 kHz. This frequency closely approximates the 20-kHz frequency shown to be optimum for transmission of maximum information at a range of 10 kiloyards (Reference 2). Operating at 15 kHz precludes interference with other acoustic systems operating in the 20 to 30 kHz band.

System bandwidth can be computed from the quality factor (Q) of the transducer at operating frequency (F), resulting in an available bandwidth (B) of F/Q kHz.

With a diversity factor D, it is possible to provide N channels of bandwidth W_a in the bandwidth B, according to the following relationship:

$$N = \left[\frac{B}{W_a}\right] \times \left[\frac{1}{D}\right] \qquad (2)$$

If these channels can be used every T_a seconds, then N/T_a signals per second can be sent. In general each signal can have 2^A states; therefore, the bit rate is AN/T_a bits per second. The rate R is

$$R = \frac{AB}{W_a T_a D} \text{ bits per second} \qquad (3)$$

Considering frequency and time dispersion (Figure 3),

$$W_a = WF + W_t \qquad (4)$$

$$T_a = T_t + T_c \qquad (5)$$

where W = frequency spreading factor
T_t = signal pulsewidth
T_c = channel clearing time
W_t = signal bandwidth

then

$$R = \left[\frac{AF}{DQ}\right] \times \left[\frac{1}{WF + W_t}\right] \times \left[\frac{1}{T_t + T_c}\right] \frac{\text{bits}}{\text{second}} \qquad (6)$$

It was previously mentioned that coding will be used to reduce the channel bit-error rate to the required value. There is, however, a tradeoff between the diversity factor D and the number of code bits required to reduce the channel-bit error

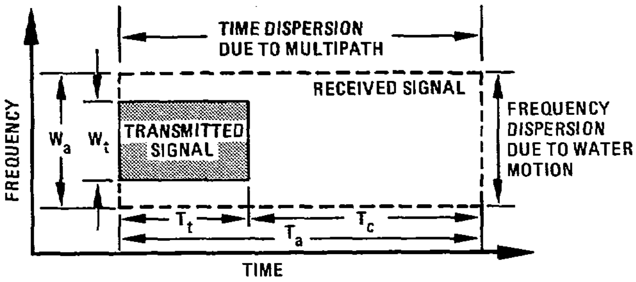

Figure 3. Frequency and Time-Dispersion Effects

rate. Reference 1 suggests that at a 16-dB signal-to-noise ratio, the use of threefold diversity instead of twofold decreases the probability of error by approximately an order of magnitude, and similar results hold for threefold to fourfold diversity. If, at a S/N ratio of 16 dB and threefold diversity, we can expect a bit-error rate of 10^{-4} (Reference 1), and an undetected bit-error rate of 10^{-10} is required, then at least 20 parity bits are needed. If, however, at the same 16-dB S/N, fourfold diversity is used, the bit-error rate is 10^{-5} and only 16 parity bits are needed. Cyclic-redundancy-check (CRC) codes using 16 bits of parity are used extensively in data communication systems and have well known error-detection capabilities. Therefore, the use of 16 parity bits appears to be a desirable choice.

A practical lower limit on the Q of a transducer is approximately 4. Assuming that a Q of 5 is readily obtained, operating at a center frequency of 15 kHz provides a system bandwidth of 3 kHz. For a quadruple-diversity system, this implies 750 Hz per band. A number of practical considerations affect the determination of the number of tone channels per band, including

- Number of bits encoded per diversity band
- Implementation of the comb filter
- Shape factor and bandwidth of the individual filters which make up the comb filter
- Anticipated Doppler shift
- Required noise bandwidth of each filter

If 16 tone channels are used to encode 8 data bits at a time, the required bandwidth per channel becomes about 47 Hz.

Since filters are not ideal, a wiser choice would be to provide a guard channel between adjacent operating channels. In addition, the noise bandwidth would be reduced, providing greater signal-to-noise ratio for a given in-band signal level. (Signal bandwidth matches noise bandwidth.) Therefore, a tone-channel bandwidth of 23.5 Hz will result from dividing the diversity band into 32 separate channels and using every other channel for signaling. Since the signal bandwidth, W_t, is optimally 23.5 Hz and $W_tT_t = 1$, then T_t = 42.6 milliseconds. A reasonable value of W is 4×10^{-4} (Reference 4). Using the signaling period of 42.6 milliseconds and diversity (D) of 4 provides an operating T_c of 0.13 second. Evaluating Equation 6 results in an R of about 150 bits per second. However, this value must be divided by four because alternate channels are not used, and two channels are used for each bit (FSK). Thus, the predicted bit rate is about 37 bits per second. A bit rate of about 47 bits per second is actually realizable, using the signal duration of 42.6 milliseconds, the clearing time of about 0.13 second, and the transmission of 8 bits per signal interval. The difference in predicted-versus-actual rate is due to an actual division factor of less than 4, because a portion of the guard band is used to accommodate the frequency spreading factor (W) by using slightly wider filter bandwidths while maintaining channel spacing.

7. SYNCHRONIZATION

The data transmission concept of the acoustic modem is to send "packets" of data. That is, a group of data bytes, with an appropriate parity-check character, is transmitted as an entity. If the packets are kept relatively short, say 32 bytes maximum in length, and if frequency offsets between the transmitter and receiver are small, baud synchronization can be done once, at the start of the data packet. By keeping the synchronization timing error to less than 10 percent of the baud period, no more than 1 dB of processing gain should be lost. For the signal period of 42.6 milliseconds, mentioned in a previous section of this paper, the initial synchronization error allowed is ± 4 milliseconds. Implementation of the synchronization technique is discussed in the following section.

8. SIGNAL PROCESSING DESCRIPTION

Figure 4 is a receiver block diagram. Note in the figure that the signal from the transducer is first linearly amplified and band-limited by the bandpass filter (BPF) to remove undesired out-of-band signal components. The AGC circuit maintains received signal levels within the dynamic range of the sampler and analog-to-digital converter (A/D). The

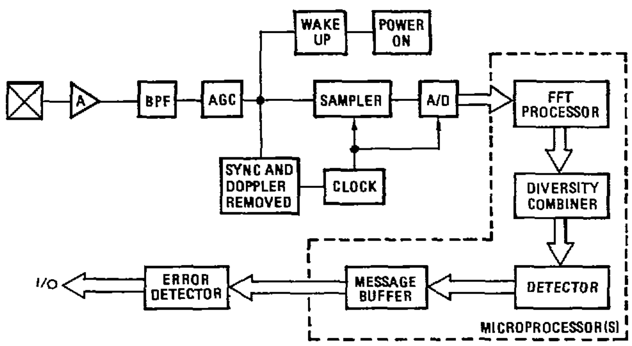

Figure 4. Receive Processing Block Diagram

sync circuit uses a frequency discriminator and threshold detector to determine when an initial carrier frequency shift occurs (sync signal). Bandwidth of the sync detector is set sufficiently wide to meet a sync uncertainty requirement of ± 4 milliseconds at a minimum specified S/N ratio.

A wake-up detector is provided to alert the receiver to the arrival of a data message. This detector has a narrow bandwidth (about 10 Hz) to minimize false-alarm rate.

The sampler demodulates the received signal and holds samples long enough for analog-to-digital conversion by the A/D. The bandwidth and selectivity of the input bandpass filter are set to keep aliasing errors below acceptable limits.

The FFT processor is an individual hardware unit, designed specifically to perform 512-point discrete Fourier transforms. (The same device also performs the inverse Fourier transform, IFFT, during data transmission.) The FFT processor functions as a comb filter, and can be implemented using either digital components or CCD Chirp-Z transform devices. The acoustic modem uses a fully digital design.

The diversity combiner operates on signals from associated redundant channels to maximize signal-to-noise ratio. The detector ratios the energy in the pairs of channels corresponding to the mark and space frequencies for each bit and decides whether a one or a zero was present. The word is temporarily stored in a buffer. Processing continues in a similar fashion until the whole message is complete, then it is checked for errors and placed in the memory for further use if no errors are detected.

Diversity combining, detection, buffering, and error detection are all performed by a microprocessor. Analog signal processing ends at the input to the sampler; all further processing is entirely digital. The sync and wake-up functions are analog.

Figure 5 depicts the transmit signal processing elements. First the message (assumed to be in a buffer) is encoded with error-detecting code bits. Diversity coding functions by assigning redundant channels to each data bit transmitted. The binary word from these operations is scanned by the FFT processor, which is performing the inverse algorithm to the receive processing. The imaginary component is output in a time sequence by the IFFT.

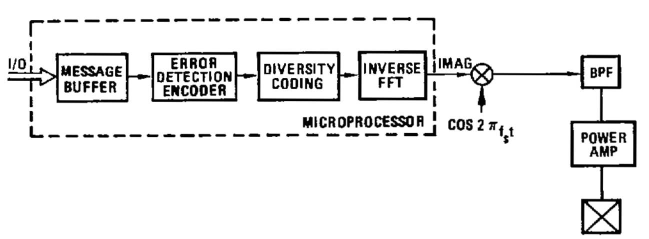

Figure 5. Transmit Processing Block Diagram

This component is double-sideband modulated by an oscillator operating at the sampling frequency. The bandpass filter selects the upper sideband and a linear power amplifier drives the transducer with the filtered and amplified signal.

9. PERFORMANCE PREDICTION

Based on feasibility studies and implementation of a breadboard system, the following performance is within reach for a shallow-water telemetry system based on the techniques presented in this paper:

Data rate ---------------	40 bits/second
Bit error rate ----------	10^{-5} @ 16-dB signal-to-noise ratio
Undetected error rate ---	10^{-10} (16-bit error detecting code)
Throughput --------------	75 percent at range of 10,000 yards
Range -------------------	3 nm

CONCLUSIONS

- The problems of time smearing and frequency smearing must be solved for a viable acoustical communications system.
- FSK is the optimal modulation technique in a long-range acoustic channel.
- Parallel FSK can increase the data rate of an acoustical communications system, provided the bandwidth is available and the signal processing can be implemented within economic constraints.
- Error-detection coding can reduce the undetected rate to extremely low values while maintaining a robust throughput.

NOMENCLATURE

FSK	Frequency-shift keying
BOP	Blowout preventer
FFT	Fast Fourier transform
PPM	Pulse-position modulation
FM	Frequency modulation
OOK	On-off keying
PSK	Pulse-shift keying
ARQ	Automatic repeat request
FEC	Forward error control
H.F.	High frequency
Q	Quality factor
AGC	Automatic gain control
A/D	Analog-to-digital converter
IFFT	Inverse FFT
I.C.	Integrated circuit

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