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FAST UNDERWATER ACOUSTIC DATA LINK DESIGN VIA MULTICARRIER MODULATION AND HIGHER-ORDER STATISTICS EQUALIZATION

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Abstract - There is a global current need to develop reliable wireless digital communications for the underwater environment, with sufficient performance and efficiency to substitute for costly wired systems. Scientific research conducted in the oceans and coastal areas could benefit through the reduction of cost for acquisition and downloading of data from remote monitoring sites. Undersea robotics could benefit through improvement of the communications features of manned and unmanned underwater vehicles for control and for near real-time conveyance of data and images.

The theoretical formulation of the underwater acoustic data communications problem includes modeling of the stochastic channel to incorporate a variety of impairments and environmental uncertainties, and proposal of compound strategies for parallel data transmission systems using multicarrier modulation (MCM) in a higher-order statistics (HOS) signal processing domain. An MCM system obtains enhanced usage of available bandwidth, as well as higher transmission rate and error threshold in comparison to the single carrier modulation (SCM) type, and thus establishes a faster communication link which is inherently more immune to impulsive type impairments.

INTRODUCTION

Objectives for any underwater acoustic communications system approach are to develop a low error rate digital modem for communication in both environments, shallow, and deep water, as well as to provide robust performance in presence of multipath distortion and reverberation. The system communications performance should adjust to the dynamically changing environment, remain robust, and potentially reach the reliability of wired systems. A hydroacoustic link, highly reliable at high speeds for digital acoustic telemetry (acoustic modem), and real-time wireless transmission of time-critical sea data from instruments located in remote sections in the sea, is of great significance for undersea automated surveillance, transmission and reception of data between submersibles, deep moored sensors or modems, and surface ships or land-based stations. A further advantage of an all-acoustic data communications system is the potential of deploying multiply moored oceanographic sensors that would exchange and potentially pass data to a master station, forming a centralized-type communication network. Such systems which are characterized by geographical sensor deployment freedom, would have a positive impact on environmental monitoring efforts and might be crucial to certain scientific programs.

The multipath propagation problem and its induced degradation effects such as ISI, have been encountered in many application fields: indoor, outdoor, and rural wireless terrestrial digital communications, tropospheric microwave digital radio links (troposcatter channel), etc.. The channel coherence bandwidth $(\Delta f)_c$ in the case of indoor communications (large offices and rooms) is in the range of 2-5 MHz; for smaller rooms $(\Delta f)_c > 10$ MHz; and for rural communications the multipath spread (T_M) is of the order $\sim 1\,\mu$ sec with a corresponding coherence bandwidth $(\Delta f)_c \sim$1MHz.

Similar effects appear in the underwater acoustic environment where the multipath spreads are orders of magnitude higher implying a reduced channel coherence bandwidth capable of accommodating only lower bit rates. Hence, in underwater applications the transmission parameters, rates, and environmental conditions differ significantly. The T_M in the underwater channel can vary from 50 ms up to the order of several seconds depending upon the application and associated range. For shallow-water, long-range communications (up to several km) $T_M \sim$ 1-3 sec, and the Doppler spread (B_D) may be of the order of several Hz. The available bandwidth W is of the order of several $100s\ KHz$ for practical short, and/or long range applications. Thus, the underwater channel consists a difficult medium to achieve high bit rates, leading to a definition of "very high speed transmission" that is at substantially lower rates as compared to radio-based systems. In this environment, a failure to meet temporal/frequential channel coherence constraints is common. Also, frequency selectivity is an unavoidable phenomenon for high speed underwater transmission due to time-varying multipath within the operating channel. This degradation, combined with additional high-frequency energy absorption, causes severe distortion reception with significant bandwidth limitations. Because of the signal-dependent type of distortion encountered, properly designed transmitted waveform pulses, and a complex receiver design are expected to be required. We deal with the effect of *reverberation* due to multipath, which creates a colored non-flat spectrum. To equalize a system with this kind of non-white input it is necessary to rely upon *a-priori information regarding the correlation properties of the colored received input* (2ndorder statistics model $\in \{S^{(2)}\}$). Other effects, affecting shallow water communications are path losses due to geometrical spreading and absorption, i.e., energy lost to heat in the water, and Gaussian ambient noise which decreases with frequency over the operative range. Usually it is encountered at low frequencies and localized. In addition, there is the flow noise turbulent boundary which is a Gaussian random process having a power spectrum which is a function of speed. Finally, there is Doppler spreading due to relative motion.

If the ocean acoustic impulse response (OAIR) can be estimated, then, in principle, the inverse of the estimated response can be used to remove the signal distortions. The propagation induced distortion can be removed by deconvolving the received data envelope with the model envelope and, consequently, source signal characteristics can be extracted. In the case of temporally and spatially invariant, range independent bottom-limited ocean, the OAIR will contain information about environmental loss parameters, i.e. rate decay of reverberant tail, etc. The transfer functions of acoustic systems often exhibit wide dynamic range and very long impulse response. Measurements of the OAIR with long reverberation is performed by using rectangular pulse as the source signal, whereby pulse width and amplitude are significant design parameters to be determined. Equalization has been applied on wireline channels, fiber optic, and wireless terrestrial communication links, where frequently the delay spread is less than 20% of the symbol duration $\left(T_M < 20\% \, T_S\right)$ so that need for equalization is not strictly necessary, but can provide improvements in reliability and performance. However, this rule of thumb is always violated in underwater communications system design, rendering equalization a necessity.

Applicability of blind equalizers has been investigated for frequency selective fading digital radio communication links. However, there exist underwater communications scenarios where the blind equalizer may be an integral part of the receiver; for instance, P-MP networks where the architecture is comprised of a master and multiple slave nodes. The master unit might potentially be located at a great distance from the furthest slave unit in the monitoring chain if the acoustic link were reliable enough. Each unit would be required to ***self-adapt*** its communication parameters to the particular environmental conditions typical of its location--a task that is not readily achieved with present field deployable system. Hence, the master station would not need to interrupt the global transmission towards each receiver in case of an individual unit link failure. Data could be recovered without disturbing the sensor mooring and without any wired connection. Environmental monitoring (satellite based service), AUV navigation, and search and recovery activities could potentially utilize the proposed communication scheme whereby a number of video channels at various locations relay information to a central point. Unlike satellite video distribution systems, the proposed scheme provides a final link to an underwater network of receivers without a direct satellite link.

PROBLEM DEFINITION

A statistical decision theory framework for general digital communications applications could be briefly described as follows (see Figure 1):

- The signal parameter space $\mathcal{Q} \subset \mathbb{C}, or \ \mathbb{R}$ (set of complex or real numbers respectively) of operating points q, or under a statistical framework, the state of nature q (statistical models of signal and noise spectra, with prior probability density functions pdf's $\pi(q)$),
- The space A of strategies, or decision rules, or actions α (filters, detectors, estimators) forming randomized or nonrandomized decision rules,
- The space $\mathcal{G}$ with observations $y^{(s_i)}$, and
- The space **M** of performance measures (Mean Squared Error MSE, SNR, Probability of Error P_e).

The communication channel, independent of the medium type, can be seen as the conditional pdf, that statistically characterizes the channel, and is mapping from signal space $\mathcal{Q} \rightarrow \mathcal{G}$ observation space. Receiver can be seen as a mapping from the observation space $\mathcal{G} \rightarrow \mathcal{A}$ decision space. Optimal receiver design is accomplished by determining a deterministic or probabilistic map based on some designated criterion.

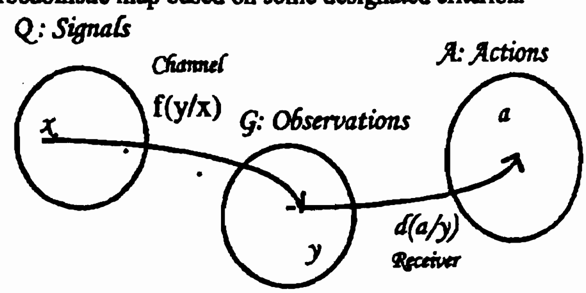

Figure 1 : The statistical communication model

The effects of the sound speed structure have a significant impact on the shape of the OAIR within the ducts and/or the low-intensity shadow zones where keen features can be predicted. Propagational distortions can be removed by deconvolving the data envelope with the model envelope, while knowledge of the sound speed dependent features enables deconvolutional schemes to restore useful signal parameters. Hence, a-priori knowledge of the transmitted signature sequences, prior data pdf's $\pi(q)$, or channel state information, i.e., some a-priori description of the communication channel OAIR is necessary. Therefore, conventional adaptive equalizers, e.g., based on Least Mean Square (LMS), and Recursive Least Squares (RLS) adaptive algorithms, would require transmissions previously known to the receiver training sequences. This waste of time and energy can be avoided by utilizing a *blind-type receiver* which only requires some statistical information about the transmitted data and can obtain equalization/identification directly from the received data. All adaptive equalizers follow the logic of the statistical sequential analysis [8].

OAIR Model

Impulse responses with long multipath spreads result in severe temporal signal dispersions. In spatial multichannel digital signaling, the frequency selective, slowly fading channels will be modeled by tapped delay lines with statistically independent, time-variant tap weights. A spatial diversity structure is implied for the vertical receive array consisting of two or more well separated sensors. The most accurate system model for OAIR is a nonlinear time-varying filter impulse response. However first-order approximation of OAIR would rely upon the linearity property and superposition of the surface and bottom reflections, so that time-varying models would be sufficient $h(\tau,t) \xrightarrow{\mathcal{F}(^*)} H(\omega,t)$. Frequency spreading channel characteristics can be measured via bifrequency functions that measure the amount of shifted energy from one frequency to another : $h(\tau,t) \xrightarrow{2-D\ \mathcal{F}(^*)} H_B(\omega_1,\omega_2)$. Model simplification is obtained by assuming a temporally (the impact of the temporal changes to the impulse response is neglected), and spatially invariant or range independent ocean, eventhough the OAIR depends on the source-receiver configuration geometry. We assume FIR transfer functions (non-minimum-phase systems) of the multiple independent paths. Provided the spacing of the sensors is adequate to control their spatial coherency (~ several wavelengths), the desired independence of the multiple measurements is achieved.

Ray acoustics with lossy specular reflection from boundaries may be used for the channel model. It is assumed that the propagation takes place along several paths with significant signal strength, or alternatively several normal modes are considerable :

$$h_n = \sum_{k=1}^{N} G_k \delta(n-k) \qquad (1)$$

where G_k, k stand for transmission losses, and propagation delays respectively, and they are independent of frequency, and independent of each other. The received signal is modeled as the sum of many scattered replicas of the transmitted sequence x_n, with spectra $X_n(\omega_0)$:

$$y_n = \sum_{k=1}^{N} G_k x_{n-k} \rightarrow Y_n = \sum_{k=1}^{N} G_k X_n e^{j\omega_0 n + \phi_k} \qquad (2)$$

By setting N=3 in eqn. (1), an averaged impulse response for the shallow water channel using a simplified 3-ray model, is formed by the contribution of a direct ray LOS, a reflected ray from the sea surface, and a bottom reflected ray:

$$\underset{\in \mathcal{G}}{y_n^{(s_i)}} = \underset{\in Q}{x_n^c} + \underbrace{\overbrace{a_S^{(s_i)}}^{G_k} x_{n-k}^c}_{Surface\ Reflection} + \underbrace{\overbrace{a_B^{(s_i)}}^{G_l} x_{n-l}^c}_{Bottom\ Reflection} + \underbrace{n^{(s_i)}}_{AGN},$$
$$i = 1, \dots, L$$

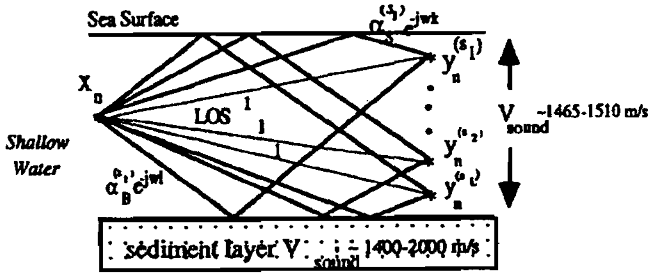

Figure 2 : The multisenor-based receiver for a 3-ray shallow water model

where $n_n^{(s_i)}$ stands for the total Additive Gaussian Noise (AGN), at each individual sensor element, and includes ambient and flow noise, assuming to follow a Gaussian distribution (i.i.d.), wide-sense stationary and statistically independent from the transmitted data. The ambient noise has a non-zero space-time cross-correlation, with a low frequency concentration, while the flow noise is spatially uncorrelated from element to element. The OAIR contains information about environmental loss parameters, i.e. rate decay of reverberant tail, or the overall attenuation parameters $a_s^{(s_i)}$, $a_B^{(s_i)}$ (surface and bottom attenuation losses respectively), and k, l are the relative time delays between the direct and multipath components. The assumption of a time invariant channel OAIR implies a stationary random variable for the received sequence y_n. However, the ergodicity property is not valid directly for the unconditioned received data process $E\{y_n^2\}$, but only for the conditioned random variable [9], i.e. the ensemble average equals the temporal one $E\{y_n^2|g_k\} = \langle y_n^2 \rangle$, where E{.}, <> stand for ensemble and time average, respectively. A non-Gaussian signal processing framework is assumed for the transmitted data distribution model. The white i.i.d. data x_n^w initially transmitted by the active sonar, acquire the colored properties due to reverberation effects. The sequence x_n^c is assumed to be a non-Gaussian if reverberation is present. Finally, $h_n^{(s_i)}$ are the unknown channel impulse response coefficients presented to the equalizer.

SCM is sufficient when we are dealing with frequency-selective, fading channels, and the resulting system is underspread, i.e., $B_D T_M < 1$, where T_M is limited to a couple of symbols (the length of the equalizer N can be relatively short) and B_D is also sufficiently small to guarantee coherent processing of all transmitted symbols. In general the ocean acoustic channel is highly dispersive in time, but not in frequency. However when a channel is characterized overspread, i.e., $B_D T_M > 1$ any waveform may become severely distorted by induced ISI, implying that a serial transmission mode is inadequate. In this case parallel schemes can prove adequate for achieving a desired degree of efficacy as discussed below.

Wideband transmissions can be designed to exploit the diversity in the channel response spread, and to operate in the presence of multipaths rather than be constrained by their presence. Spatial diversity, which is optimum in terms of energy and temporal eficiency but not in the number of the deployed receivers, could exploit the multidirectional scattering effects of the surface reflected signal. Other types of diversity can be obtained by transmitting over different time/frequency cells with time separation greater than the temporal channel coherence $(\Delta t_C) \sim 1/B_D$, and frequency separation greater than the frequency coherence bandwidth $(\Delta f_C) \approx 1/T_M$.

HOS Inverse Filter Estimator

If the OAIR can be estimated then, in principle, the inverse of the estimated response can be used to remove the signal distortions. In order to measure OAIRs, cross-correlation of the received signal with a source replica at the lower frequency band and within a limited bandwidth, has been most often used, and is equivalent to the propagation of the transmitted signal autocorrelation function R_x. A second order statistics $\{S^{(2)}\}$ (autocorrelation-based) domain-based inverse filter estimate $C^{(s_i)}(\omega)$ (e.g., a Best Linear Mean Square Error (BLMSE) channel transfer function estimate satisfying the orthogonality principle *error* $e_n \perp$ *data* x_n), suppresses the phase information which is significant in the recovery of non-minimum phase degraded systems.

More robust systems can be achieved via HOS-based estimators $\{S^{(l>2)}\}$, due to Gaussian noise suppression and phase information preservation. An increase of the SNR level in the working environment is obtained, while the algorithm is capable of dealing with non-minimum phase channels. Higher order moments or cumulants or their Fourier Transforms, so called polyspectra, can be used to identify and equalize distorted communication systems. More specifically it is the 4th-order (or higher) statistics domain that is most often used for communications applications, since communication signals have symmetric pdf, so that their 3rd-order statistics are identical to zero :

$$\begin{aligned}
&\textit{Third - Order Statistics (TOS) - domain:}\\
&\textit{Skewness} \qquad \beta = E\left\{x_n^3\right\} = 0\\
&\textit{Fourth - Order Statistics (FOS) - domain:}\\
&\textit{Kurtosis} \qquad \gamma_X = E\left\{x_n^4\right\} - 3\left\{E\left\{x_n^2\right\}\right\}^2 \neq 0
\end{aligned} \tag{3}$$

The fourth order cumulant of the received data can be used to provide channel phase estimates $\hat{\phi}_{(s_i)}$:

$$L_{m,n,l} = E\left\{y_i y_{i+m}^* y_{i+n} y_{i+l}^*\right\} - E\left\{y_i y_{i+l}^*\right\} E\left\{y_{i+m}^* y_{i+n}\right\} - E\left\{y_i y_{i+m}^*\right\} E\left\{y_{i+n} y_{i+l}^*\right\}$$

where E{*} stands for the expected value of the stochastic process due to the inherent environmental uncertainties, and its Z-transform, (the trispectrum) is: $L_{m,n,l} \xrightarrow{Z} T^{y}_{z_1,z_2,z_3}$. Note that the 2nd and 4th order moment estimates are updated iteratively with each new received symbol.

MODULATION FOR A HIGH-SPEED UNDERWATER MODEM

Single-Carrier Modulation (SCM)

In the following we would like to classify the priorities of the most critical criteria in the 4-D design space of a modem for underwater operation, comprising :

1. Bandwidth efficiency,
2. Power efficiency,
3. Receiver complexity,
4. Immunity to nonlinearities.

The optimal signal sets $\{x_i \in Q, \quad i = 1,\ldots,M\}$ lie in the class of linearly dependent sets of vectors since they present a greater P_D, $\forall SNR$ than their corresponding independent ones $\{x_i' \in Q, \quad i = 1,\ldots,M\}$. Furthermore, in 2-dimensions the optimal signal set that obtains $\max P_D$, $\forall SNR$, consists of the M signal vectors equally spaced around the unit circle $\{x_i = e^{j2\pi i/M}, i = 1,\ldots,M\}$. The signal constellations depending on the modulation type can be a subset of the set of real $\mathbf{R}$, or complex $\mathbf{C}$ numbers. Incoherent (FSK-type) detectors are simple in implementational sense, but they obtain poorer performance and low bandwidth efficiency compared to the coherent (PSK, QAM-type $\subset \mathbf{C}$) ones. Performance of the constellation against noise, is determined by the minimum distance between the signal points, e.g., $d_{\min}^{(16-QAM)} < d_{\min}^{(16-PSK)}$ (the spectral efficiency ($n \sim \log_2 \underbrace{2^n}_{\#\ of\ signal\ points}$) of a 16-point constellation is ~ 4 bps/Hz, while of a 2-point is ~ 1 bps/Hz). However the coherent detector's performance is degraded in the presence of multipaths and doppler shifts that exist in the underwater nonstationary applications. Addition of differential detection can overcome the above difficulty with some loss in performance. Large QAM signal constellations (i.e., 16-, 64-QAM) obtain an extremely high bandwidth efficiency, but they suffer in terms of power efficiency. QPSK shows best bandwidth efficiency, moderate immunity to nonlinearities and complexity. MSK is not recommended due to its poor bandwidth efficiency, and high implementation complexity. However, in parallel transmission schemes via frequency division mutiplexed channels (see next section), MSK presents lower *crosstalk* than QPSK if the guardband is > 0.7 bit rate of the individual channel. Finally, MFSK is not recommended either due to the very poor bandwidth efficiency.

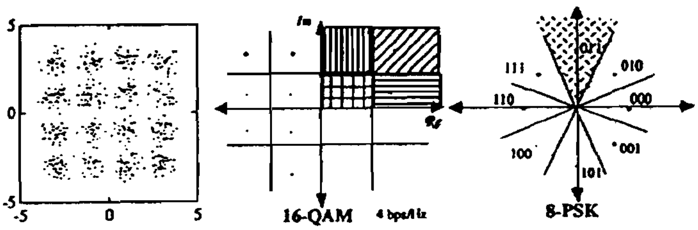

Figure 3 : Accumulation over time of noise and ISI-perturbed received 16-QAM constellations (left) - Decision regions for ML detector (right)

Channel coding has been successfully combined in satellite communications to compensate for power losses without sacrificing bandwidth efficiency. CPM-type modulations are coded modulations themselves. In general CPM shows an extraordinary joint power and bandwidth efficiency, at the expense of a higher hardware complexity. The reduced level of sidelobe energy in their spectra provides an additional advantage of system resistance to crosstalk effects. It is worthwhile to consider the same approach in underwater acoustic modem design. Thus, switching operation between uncoded mode for deep water and coded mode for shallow water with intense multipath effects, would be a natural consequence.

Multicarrier Modulation (MCM)

A novel approach to acoustic communication system design originates from the following significant observation about the sound propagation properties in the ocean. The surface duct and the deep sound channel are two disjoint propagation paths with different transfer functions $H_S(\omega_1), H_D(\omega_2)$ and optimum operational frequencies (see Figure 4).

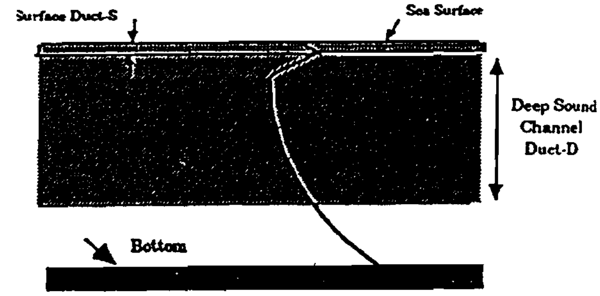

Figure 4 : Sound speed profile

The energy leakage among the two ducts may be considered to be negligible and can be exploited judiciously through a MCM transmission scheme that encodes data over two subcarriers. It allows operation at multiples of the single-carrier transmission rate, by performing a data volume split over the multiple carrier modes. The system's immunity to distortions is increased due to incorporation of symbol guard space, while the equalizer's complexity at the reception point is relaxed. This design philosophy is characterized as "equalization complexity split" between receiver and transmitter. More specifically, the longer the guard period the less sensitive to the multipath spread. However, there is a trade-off between the BER and the guard period, since the longer the guard period the larger the inefficiency in terms of power and bandwidth. Each carrier operates at low data rate to reduce the probability of frequency selective fades destroying the data integrity, while the overall system performance is the one of a high data rate system. MCM can deal with more severely overspread channels where serial-type transmission is inadequate, and alleviates the high complexity of the equalizers employed by the receiver. There exists a dual reasoning behind the obtained receiver's complexity reduction :

- A complexity split between the transmitter and the receiver is accomplished, by performing joint equalization at both ends.
- Equalization of a narrowband subchannel requires an equalizer of lower complexity, than that of the wideband (comparatively) channel, and reduces the adverse effects of linear equalization such as the additive noise enhancement.

$$x(t) = \sum_{i=-\infty}^{\infty} \sum_{k=0}^{1} \mathrm{Re}\left\{\alpha_{ki} e^{j\omega_k (t-iT_s)}\right\} p\left(t-iT_s\right)$$

where p(t) is the symbol pulse waveform $p(t) = \begin{cases} 1 & -\Delta \le t \le t_s \\ 0 & otherwise \end{cases}$, t_s is the observation period,

α_{ki} are mutually independent i.i.d. data sequences, Δ is the guard space length, $T_S = \Delta + t_S$ is the extended symbol period, and $\omega_k = \omega_0 + \frac{2\pi k}{t_s}$. The received waveform is given :

$$y(t) = \int x(t-\tau)h(\tau;t)d\tau + n(t) \tag{4}$$

$$y_{mi} = \frac{1}{t_s}\int_{iT_s}^{t_s+iT_s} y(t)e^{-j\omega_m(t-iT_s)}dt$$

For a time non-selective channel, it is assumed that the channel fluctuation can be neglected during a single transmission or even a few symbols, i.e., $T_S << \frac{1}{f_D}$.

MCM is inherently more robust to time-domain impulsive noise (e.g., biologically generated or other impulsive noises) due to the block processing nature in the frequency domain.

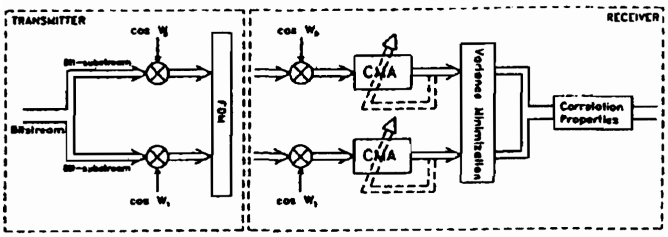

Figure 5 : Transmitter/Receiver Structures for MCM - Parallel Data Transmission

Performance analysis of the MCM system can be derived after determining the type of probability density function (pdf) $D_{mi}(\rho)$ that the random decision variable D_{mi} follows :

$$D_{mi} = R\left[y_{mi}\dot{y}_{m(i-1)}e^{j\left(\frac{\pi}{2}-\frac{\pi}{M}\right)}\right] \tag{5}$$

The bit-error-rate performance for MCM can be calculated based on the region that the decision variable D_{mi} takes value less than zero. Hence, given that the distribution function of the decision variable is as follows,

$$F_{D_{mi}}(x) = P\left(D_{mi} \le x\right) = \int_{-\infty}^{x} D_{mi}(\rho)d\rho \Rightarrow P_e = F_{D_{mi}}(0)$$

it can be easily shown that the bit-error rate (BER) performance for frequency-selective fading channel is a function of the correlation coefficient $R_y(i)$ between consecutive received symbols of the same carrier : $R_y(i) = E\{y_{mi}\dot{y}_{m(i-1)}\} / E\{y_{mi}\dot{y}_{mi}\}$. The higher the correlation degree the smaller the P_e:

$$P_e = P(D_{mi} < 0) = \int_{-\infty}^{0} D_{mi}(x)dx = f(R_y(i)) \tag{6}$$

CONCLUSIONS

In MCM, interleaved bit streams modulate more than one carrier. The rates on each carrier can vary providing a flexible transmission rate. MCM, can be considered as a predistortion/pre-equalization introduced at the transmitter, reducing some ISI, and allowing simplified equalization schemes with lower complexity and implementation cost. It provides additional efficiency for robust receiver design especially with overspread channels where serial transmission is inadequate. Frequency diversity may be implemented as an alternative for MCM methods to exploit the multiple modes of the ocean acoustic waveguide. Although improved performance can be obtained in terms of obtained SNR output levels, system degradation due to increased ISI effects is also expected.

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