Digital Signal Processing Research Program

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Introduction

The field of Digital Signal Processing grew out of the flexibility afforded by the use of digital computers in implementing signal processing algorithms and systems. It has since broadened into the use of a variety of both digital and analog technologies, spanning a broad range of applications, bandwidths, and realizations. The Digital Signal Processing Group carries out research on algorithms for signal processing and their applications. Current application areas of interest include signal enhancement and active noise cancellation; speech, audio and underwater acoustic signal processing; advanced beamforming for radar and sonar systems; and signal processing and coding for wireless and broadband multiuser/multimedia communication networks.

In some of our recent work, we have developed new methods for signal enhancement and noise cancellation with single or multisensor measurements. We have also been developing new methods for representing and analyzing fractal signals. This class of signals arises in a wide variety of physical environments and also has potential in problems involving signal design. We are also exploring potential uses of nonlinear dynamics and chaos theory of signal design and analysis. Another research emphasis is on structuring algorithms for approximate processing and successive refinement.

In other research, we are investigating applications of signal and array processing to ocean and structural acoustics and geophysics. These problems require the combination of digital signal processing tools with a knowledge of wave propagation to develop systems for short time spectral analysis, wavenumber spectrum estimation, source localization, and matched field processing. We emphasize the use of real-world data from laboratory and field experiments such as the Heard Island Experiment for Acoustic Monitoring of Global Warming and several Arctic acoustic experiments conducted on the polar ice cap.

A major focus of the group involves signal processing and coding for digital communications applications including wireless multiuser systems and broadband communication networks. Specific interests include commercial and military mobile radio networks, wireless local area networks and personal communication systems, digital audio and television broadcast systems,

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multimedia networks, and broadband access technologies. Along with a number of other directions, we are currently exploring new code-division multiple-access (CDMA) strategies, space-time techniques for exploiting antenna arrays in wireless systems, new multiscale methods for modeling and management of traffic in high-speed packet-switched networks, and new information embedding techniques for digital watermarking of media and related applications.

Much of our work involves close collaboration with the Woods Hole Oceanographic Institution, MIT Lincoln Laboratory, and a number of high technology companies.

Neural Signal Processing

Sponsors
National Science Foundation Graduate Fellowship

Project Staff
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In order to understand the detailed interworkings of many neurological processes, it is necessary to measure the firing patterns realized by individual neurons. Current measuring techniques involve inserting one or more electrodes into the region of interest, which make extracellular voltage recordings derived from the action potentials of nearby neurons. The difficulty is that firing patterns from many different neurons are superimposed at the electrodes, while we are interested in individual neuron behavior. Deriving this information from such measurements is referred to as separating multiple single-unit spike trains from a multi-unit recording.

The problem is therefore one of signal separation, and many approaches have been attempted based on pattern matching and feature clustering. In many of these approaches the inaccurate assumption that different neurons exhibit action potentials with unique waveforms is made. A new instrument called the tetrode was developed in 1994. A tetrode consists of four very closely spaced electrodes, which allows one to drop this assumption and therefore perform a more reliable separation. The best existing separation schemes for the tetrode are computer assisted they present waveform parameters in a graphical manner so that a well-trained user can visually cluster the features as arising from separate neurons. These techniques necessarily prevent a full exploitation of the information available in the tetrode measurements, since decisions must be made in a low enough dimension for human visualization.

We hope to improve upon these existing multi-unit separation schemes, and will then pursue related applications (e.g., action potential coding) of our new knowledge.

Dual Channel Signal Processing

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Many models for signal estimation systems assume only statistical information about the source signal to be recovered and about the channel through which the source is sent. In some scenarios, however, there also exists deterministic side information about the desired signal which can be used jointly with channel observations to assist recovery. For example an existing full-band, noisy analog communications infrastructure may be augmented by a low-bandwidth digital side channel. Our research is a study of a hybrid channel that is the composition of two channels: a noisy analog channel through which a signal source is sent unprocessed and a secondary rate-constrained digital channel. The source is processed prior to transmission through the digital channel. Using a signal processing framework for low latency and low complexity, we derive optimal encoder and receiver structures for hybrid channels.

Batch-Iterative Channel Equalization

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Project Staff

Albert Chan, Professor Gregory W. Wornell

The goal of channel equalization is to minimize the probability of error by compensating for channel distortion. We are currently working on a new and efficient class of nonlinear equalizers developed for intersymbol interference (ISI) channels. These ``iterated-decision equalizers'' use an optimized multipass algorithm to successively cancel ISI from a block of received data and generate symbol decisions whose reliability increases monotonically with each iteration. These equalizers have a complexity comparable to the decision-feedback equalizer (DFE), yet asymptotically achieve the performance of maximum-likelihood sequence detection (MLSD). Because their structure allows cancellation of both precursor and postcursor ISI, iterated-decision equalizers outperform the minimum mean-square error DFE by 2.507 dB on severe ISI channels even with uncoded systems. Moreover, unlike the DFE, iterated-decision equalizers can be readily used in conjunction with error-control coding, making them attractive for a wealth of applications.

Information Embedding and Digital Watermarking

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Digital watermarking and information embedding, which are also referred to as data hiding and steganography, refer to the process of embedding one signal, called the “embedded signal” or “digital watermark,” within another signal, called the “host signal.” The host signal is typically a speech, audio, image, or video signal, and the embedding must be done in such a way that the host signal is not degraded unacceptably. At the same time, the digital watermark must be difficult to remove without causing significant damage to the host signal and must reliably survive common signal processing manipulations such as lossy compression, additive noise, and resampling. Applications include copyright protection, authentication, transmission of auxiliary information, and covert communication.

In our work we are developing a general framework for designing digital watermarking systems, evaluating their performance, and understanding their fundamental performance limits. In the process we have developed a class of digital watermarking techniques called quantization index modulation, along with a convenient realization called dither modulation, that have considerable performance advantages over previously proposed methods. More information can be found at http://web.mit.edu/bchen/www/wmark-home.html and in the following publications.

[1] B. Chen and G. W. Wornell, “Preprocessed and postprocessed quantization index modulation methods for digital watermarking,” to appear in *Proc. of SPIE: Security and Watermarking of Multimedia Contents II* (part of Electronic Imaging 2000), vol. 3971, San Jose, CA, Jan. 2000.

[2] B. Chen and G. W. Wornell, “Provably robust digital watermarking,” *Proc. of SPIE: Multimedia Systems and Applications II* (part of Photonics East '99), vol. 3845, Boston, MA, pp. 43-54, Sept. 1999.

Three-Description Source Encoding

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Project Staff
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We are working in the area of source compression for correlated sources. Standard techniques lead to one of two options. One is to entropy encode the two sources into one codeword. This uses minimal memory resources, but leads to complicated decoding procedures. Alternately, the sources can be separately entropy encoded into two independent codewords. This uses greater memory resources, but leads to simple decoding procedures.

We are interested in a third set-up where the sources are encoded into three codewords. One codeword characterizes the “common information” between the two sources, and the other two codewords characterize the “marginal refinements” needed to reconstruct each source. By differentially weighting the cost of the common information rate versus the marginal rate, we can trace out a region bracketed by the two standard techniques described above.

Nonuniform Sampling and Filter Banks

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In our research the connection between nonuniform sampling and filter banks is being explored. It is well established that a bandlimited finite-energy signal can be uniquely determined from its samples at nonuniform time instances provided that the average sampling rate is at least the Nyquist rate. For some classes of sampling distributions explicit reconstruction formulas are known. However, the reconstruction typically involves time-varying systems. To efficiently implement the reconstruction, we suggest a filter bank configuration. Our research is directed at developing a general framework for reconstructing a signal from its nonuniform samples using filter banks. We are currently investigating sampling distributions that fit this framework.

Coding Approaches to Fault Tolerance in Dynamic Systems

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A fault-tolerant system tolerates internal failures while preserving desirable overall behavior. Fault tolerance is necessary in life-critical or inaccessible applications, and also enables the design of reliable systems out of unreliable, less expensive components. Our work discusses fault tolerance in dynamic systems, such as finite-state controllers or computer simulations, whose internal state influences their future behavior. Modular redundancy (system replication) and other traditional techniques for fault tolerance are expensive, and rely heavily -- particularly in the case of dynamic systems operating over extended time horizons – on the assumption that the error-correcting mechanism (e.g., voting) is fault-free.

The work develops a systematic methodology for adding structured redundancy to a dynamic system and introducing associated fault tolerance. Our approach exposes a wide range of possibilities between no redundancy and full replication. Assuming that the error-correcting mechanism is fault-free, we parameterize the different possibilities in various settings, including algebraic machines, linear dynamic systems and Petri nets. By adopting specific error models and, in some cases, by making explicit connections with hardware implementations, we demonstrate how the redundant systems scan be designed to allow detection/correction of a fixed number of failures. We do not explicitly address optimization criteria that could be used in choosing among different redundant implementations, but our examples illustrate how such criteria can be investigated in future work.

Our work relaxes the traditional assumption that error-correction be fault-free. We use unreliable system replicas and unreliable voters to construct redundant dynamic systems that evolve in time with low probability of failure. Our approach generalizes modular redundancy by using distributed voting schemes. Combining these techniques with low-complexity error-correcting coding, we are able to efficiently protect identical unreliable linear finite-state machines that operate in parallel on distinct input sequences. The approach requires only a constant amount of redundant hardware per machine to achieve a probability of failure that remains below any pre-specified bound over any finite time interval.

Antenna Arrays and Space-Time Coding

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Due to the rich scattering environment of an indoor setting, wireless communication is hindered by signal fading (loss of signal energy) due to multipath propagation of signals from transmitter to receiver. The frequent lack of line of sight signaling and time-varying nature of the communications channel (due to movement of people and objects for instance) limits the amount of data than can be reliably transmitted given power and bandwidth constraints. One way to mitigate these effects is with arrays of antennas at both the transmitter and receiver. Space-time codes are a class of codes which provide signal diversity in both space and time (as their name implies) by "spreading" information bits over many samples in time as well as over the spatially separated antennas to provide redundancy to aid in decoding.

In the context of a wireless indoor LAN, we are looking at developing space-time codes and designing algorithms for a system to achieve gigabit data rates over an indoor wireless channel. The available bandwidth will be divided into several subchannels, each of which will be adaptively coded given the varying signal quality in the frequency band. We are currently looking at Lucent's BLAST technology for coding, as well as the effects of the non-idealities that are inevitable in building an actual system, such as channel estimation errors and antenna isolation difficulties, and their impact on our ability to reach very high data rates.

Low-Complexity Diversity Transmission for Fading Channels

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In the mobile wireless communications setting, transmission quality is severely degraded by *fading* -- fluctuations in received signal energy induced by multipath propagation and relative motion of the transmitter and receiver. Diversity transmission via multiple transmit antennas, bandwidth expansion, or coding mitigates the effects of fading by essentially repeating information on independent realizations of the channel and averaging the received signal. This most basic form of diversity transmission in space, frequency, or time corresponds to a repetition code, and more powerful codes for fading channels have been developed over the years. Unfortunately, the complexity of maximum-likelihood decoding precludes the use of long codes and, hence, higher orders of diversity.

Motivated by two compelling examples, spread-response precoding [1] and signal space diversity [2] systems, our research explores low-complexity diversity transmission from primarily a signal processing viewpoint. Each of these examples creates temporal diversity by transmitting the output from a linear (unitary) transformation of an uncoded symbol sequence, and neither scheme requires additional power or bandwidth. In the case of spread-response precoding, a low-complexity receiver consisting of linear equalization followed by symbol-by-symbol detection can achieve very high orders of diversity with quite good performance. We are comparing and contrasting these examples, evaluating the performance and complexity tradeoffs associated with several receiver structures, and hoping to generalize these ideas to create more efficient schemes.

[1] Gregory W. Wornell. "Spread-Response Precoding for Communication over Fading Channels," *IEEE Transactions on Information Theory*, March 1996.

[2] Joseph Boutros and Emmanuele Viterbo, "Signal Space Diversity: A Power- and Bandwidth-Efficient Diversity Technique for the Rayleigh Fading Channel," *IEEE Transactions on Information Theory,* July 1998.

Distributed Signal Processing

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Networks of distributed processors and sensors can offer significant advantages over traditional signal processing platforms through improving resource sharing and fault tolerance. However, for networks in which resource availability changes dynamically, it is important to have a flexible execution strategy which allows algorithms to adapt to the current conditions of the network. Our research studies a number of questions arising from designing signal processing algorithms for dynamically changing networks of processors and sensors. We propose a framework in which the system is allowed to dynamically and optimally choose an execution strategy for each data block in adaptation to the computing environment. Our formulation of this strategy derives from an interesting interpretation of algorithms as similar to communications networks. A software simulation has been developed to demonstrate the ideas.

Transmit Antenna Arrays for Multiple-User Wireless Communication

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Antenna arrays have the potential to increase both the reliability and maximum performance of wireless systems. Tapping this potential using arrays at the transmit side, such as in a base station-to-mobile downlink, presents challenges not encountered with the better-understood receiver arrays. The problem becomes even more difficult when information (either common or user-specific) must be directed toward multiple receivers.

Single-user strategies have been proposed in the literature for scenarios where the transmitter has no knowledge of the channel parameters. We, on the other hand, are investigating the use of feedback to enhance performance when transmitting to a number of receivers. Especially useful are algorithms which control either the acquisition or the use of this feedback information in an efficient manner. It is also important to quantify the tradeoffs involved in transmitting to multiple users, such as how the value of having channel knowledge degrades as one attempts to send a common message to additional receivers.

Analog Authentication: Protecting Images, Video, And Other Signals from Forgery and Tampering

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Project Staff

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We have been exploring the problem of how to authenticate analog signals such as images, video, speech, etc. in order to protect them from tampering and forgery. One possible application is to protect ID photos such as the ones on passports or driver's licenses from forgery. For example, Massachusetts currently stamps a hologram on all driver's licenses to deter forgery. The idea is that criminals will not be able to produce a license with a hologram. The security is thus based on the assumption that producing a hologram is hard. We examine approaches where security could be based upon standard complexity theoretic assumptions such as the difficulty of factoring or information theoretic assumptions such as shared keys.

Analog authentication is a new idea and there is no general consensus on a rigorous problem formulation or solution. Most researchers work on balancing security, robustness, and processing distortion. Security relates to the likelihood that an unauthorized person can create a forgery or tamper with a valid signal. Robustness relates to the false positive rate. For example, somebody might accidentally smudge a license. Intuitively we consider a smudge to be noise and not a forgery. Most schemes are designed to allow small perturbations without declaring them to be forgeries. The final issue is distortion. Any kind of analog authentication scheme which modifies the original (such as stamping a hologram on a photo) distorts the original. One goal of a good analog authentication scheme is to keep the processing distortion small.

In our work we propose some rigorous definitions of the analog authentication problem. In our problem formulation we quantify the various issues and goals such as security, robustness, and distortion. We analyze the problem using tools from information theory and cryptography. We have found a coding theorem as well as a converse which yield some surprising results. For example, we have proved that there is no fundamental tradeoff between robustness and security; both can be achieved arbitrarily well as long as enough processing distortion is allowed. In fact we have derived a formula to calculate the fundamental distortion necessary to achieve the goals of analog authentication. Schemes exists which require processing distortions arbitrarily close to the fundamental distortion and achieve the goals of robustness and security. Furthermore any scheme which uses less processing distortion than the fundamental distortion can not achieve both robustness and security. These results are analogs of the classical capacity theorems where reliable communication at rates below capacity are possible and reliable communication at rates above capacity are impossible.

Our work grew out of Chen and Wornell's work on digital watermarking and has some interesting relations to watermarking. More about Chen and Wornell’s work can be found at http://web.mit.edu/bchen/www/wmark-home.html.

Applications of Extended Multi-rate Processing

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The standard multi-rate concepts of up-sampling and down-sampling can be extended to include continuous time (with possibly differing time scales). Systems with mixed discrete-time and continuous-time components are being analyzed. These theoretical constructs are very powerful, and are a natural extension to the well known discrete systems. These ideas are being applied to several problems, with significant advantages. For example, sample rate conversion done using these types of abstractions can be much more efficient in both computation and memory usage over conventional techniques. Other problems being addressed are those of non-uniform sampling, and the idea of finding better discrete representations of continuous signals.

Linear Models for Randomized Sampling of Discrete-Time Signals

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Efficient computation in digital signal processing is a practical problem that has been approached in various ways. Traditional solutions involve down-sampling the signal by a constant factor. Instead, in this project we explore the use of randomized sampling of the discrete-time signal as a method for reducing multiplications in the convolution sum. We have designed several algorithms based on randomized sampling of the input signal, the system function, and also iterative randomized sampling of the system function. In addition, we have formulated additive noise models for each algorithm which allowed linear filtering techniques to be applied.

Approximation and Autonomy in Large Scale Signal Processing Computations

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Matthew J. Secor, Professor George C. Verghese, Professor Alan V. Oppenheim

This research investigates issues related to the operation, dynamics and control of networks of processors performing large-scale DSP (digital signal processing) computations. Two key aspects of our framework are the introduction of approximate processing to provide flexibility in managing system resources effectively for enhanced performance and the use of market-based models for the computational environment on which to build the communication and control algorithms.

In a typical scenario, an individual user partitions a large problem into task packets that can be distributed on the network. Other processors attend to these task packets as best they can, and then approximate results are gathered back at the user's node for re-assembly into an approximate answer to the problem of interest. Associated with such a scenario are research challenges related to characterizing DSP algorithms appropriately for this form of approximate processing, developing partitioning and re-assembly methods that are robust to differential delays and varying levels of completion in the returned task packets, and managing the communication and computation on the network so that congestion and delay instabilities are avoided. The research focuses on the latter set of problems, dealing with the dynamics and control of the network.

Specifically, we are reformulating queuing network models and load balancing strategies for networks of processors, using richer models for jobs. Our models includes the ability to use approximate processing methods to trade off the resources required to process a job against the quality of the results. We have used such models to investigate the relationship between the stability of a queue and the distribution of the quality of the jobs after being processed by the queue.

Additionally, we have investigated the use of economic models and mechanisms for the efficient allocation of resources of computations. We have developed analytical models for the dynamic behavior of an existing implementation of an auction-based resource allocation system, in order to determine the fairness and stability properties of the scheme. Some numerical pitfalls in naive approaches to the simulation of such systems have also been exposed. We are also exploring the use of game-theoretic models in order to model the strategic aspects of the interaction between independent participants in the computational marketplace in their attempts to acquire computational resources.

Sinusoidal Analysis Synthesis

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Project Staff
Alan Seefeldt, Professor Alan V. Oppenheim

The problem of separating voice from background noise or sounds in a recording is being investigated. In particular, we are examining the use of an uncorrupted facsimile of the corrupted voice as an aid in the separation process. Possible directions include using the facsimile to generate a probabilistic model from which processing is derived and applying transformations to the facsimile to make it sound like the voice imbedded in the noise. These techniques will hopefully be applied to the separation of an operatic voice from the accompanying orchestra.

Speech Enhancement with Side Information

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Multi-sensor signal processing techniques been used successfully in a many applications. For example, arrays of sensors have been applied to geophysical, sonar, and radar signal processing problems. These traditional examples of sensor networks have tight restrictions on their structure. The arrays have a known a known, regular geometry. The measurements form each sensor are synchronized and communicated losslessly to a remote observer for processing. In some cases, the remote processor is able to process the data offline and has no complexity constraints on the processing algorithm.

In the most general case, however, the sensor network need not be constrained. Given the recent proliferation of data networks and wireless communications, it is possible to imagine implementing a more complex network in which each sensor processes local measurements as well as information from other sensors. The sensors can cooperatively perform signal processing tasks and eliminate the need to analyze the data remotely.

We will investigate signal processing algorithms in this distributed environment. We seek techniques that enable a network of flexible sensors to carry out high-performance statistical signal processing algorithms subject to the synchronization, communication, and processing complexity constraints of their environment.

Communications Using Chaotic Systems

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Chaotic signals have noise-like characteristics, are generally broadband, and are difficult to predict, therefore they are potentially useful for spread-spectrum communications. In particular, because chaotic signals are difficult to predict and noise-like, they are possibly well suited for situations in which it is desirable to have a low probability of intercept, or covert operation. Since it has been shown that chaotic systems can be synchronized through a weak, one-way coupling, many chaotic transmitter/receiver designs are possible, both in discrete-time and continuous-time. We are investigating an approach in which the information signal modulates the rate at which the chaotic transmitter system evolves. Conceptually, this can be viewed as a generalization of traditional phase- and frequency-modulation techniques. The major problems in designing this type of communication system are robust demodulation of the information signal and maintaining synchronization in the presence of noise.

Broadband Modal Coherence and Beamforming at Megameter Ranges

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Project Staff

Kathleen E. Wage, Professor Arthur B. Baggeroer

Our work develops a method for estimating the normal mode decomposition of broadband signals and uses it to analyze data from the Acoustic Thermometry of Ocean Climate (ATOC) experiment. Normal modes are the eigenfunctions of the ocean waveguide, derived from the frequency-domain wave equation. They are useful in underwater acoustics, particularly matched field processing and tomography, because the lowest modes provide an efficient description of the most energetic arrivals at long ranges. Extracting source or environmental information from the mode signals depends on understanding the effects of internal waves on coherence and the validity of adiabatic approximation. While much theoretical research has been done on long-range propagation of modes in deep water, there have been few opportunities to compare theoretical predictions with experimental measurements.

The first contribution of our work is a short-time Fourier framework for estimating broadband signals propagating in the lowest modes of the ocean waveguide. Since previous research has focused primarily on narrowband sources, this work concentrates on broadband processing issues. Specifically, it addresses the fundamental issue of frequency resolution required for mode estimation, analyzes the performance characteristics of two modal beamforming algorithms and explores the time/frequency tradeoffs inherent in STFT mode processing.

The second contribution of this research is a detailed analysis of the low-mode arrivals at megameter ranges using five months of data from the ATOC vertical line array at Hawaii (3515 km range). Short-time Fourier processing of these receptions revealed that each low mode contains a series of arrivals, rather than the single dispersive arrival that would characterize adiabatic propagation. Average coherence times of the mode signals are on the order of 6-8 minutes. The multipath structure changes significantly between receptions at 4-hour intervals, indicating that stochastic methods are required for mode tomography at megameter ranges. A statistical analysis found that modes do retain travel-time information at megameter ranges, *e.g.*, the centroids show the expected dispersion characteristics of a deep water channel. The centroids show statistically significant trends in mode arrival time over the period of the experiment.

Coding for Transmission over Unknown Channels

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In certain communication applications, such as wireless communications, the channel might not be known and might be time variant. In this scenario, one approach is to repeatedly estimate the channel using training sequences known to the receiver, taking up time and/or frequency allowance. This approach also become unsuitable when the channel is highly time variant. Our approach in the unknown channel scenario is not to use training sequences to help estimate the channel, but to build structures into the transmitted signal and use universal decoders to decode the messages directly without estimating the channel first. This class of approaches tend to have the shortcoming of low spectral efficiency. We are also considering using random codebooks to introduce diversity into the transmission to improve worst case performance.

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