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Flexible Signal Processing Algorithms for Wireless Communications

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Abstract: **Wireless communications systems of the future will experience more dynamic channel conditions and a wider range of application requirements than systems of today. The identification of specific modes of flexibility that enable efficient overall system operation. The approach then uses explicit knowledge of the relationships between the input and output samples to develop efficient algorithms that provide the desired flexible behavior. Using this approach, we have designed a suite of novel algorithms for essential physical layer functions. These algorithms provide both dynamic functionality and efficient computational performance. We present a new technique that directly synthesizes digital waveforms from precomputed samples, a matched filter detector that uses multiple threshold tests to provide efficient and controlled performance under variable noise conditions, and a novel approach to narrowband channel filtering. This is contrast to conventional approaches, where the complexity depends upon the input sample rate. Finally, we describe an implementation of these algorithms in a software radio system as part of the Speetrum Ware Project at MIT.**

Keywords: **MIT, Physical, Layer, Transmitter, Receiver**

1. INTRODUCTION

Communications- anytime, anywhere. This mantra is often repeated as designers and producers of communications networks advertise new capabilities and services available because of recent technological developments. An important part of this universal communications network will, of course, be the portions that provide mobility and access to the global infrastructure without wires. Today, it is this wireless portion of the system that seems to lag behind expectations.

To be sure, wireless communications systems have changed significantly in the past ten years. Most notable, of course, is the sheer *number* of people that use wireless communications services today. This trend of increased usage is expected to continue and has generated considerable activity in the area of communications system design. Another clear trend has been the type of information that these wireless systems convey: virtually all new wireless systems are *digital* communications systems; unlike earlier broadcast and cellular systems, newer systems communicate using digital data encoded into radio waves. Even analog source information such as voice and images are digitized and then transmitted using digital formats.

The reasons for this shift to digital wireless systems are several. Initially, the shift was made in cellular telephone systems to improve system capacity through more efficient usage of the limited radio frequency (EF) spectrum. This shift has also made available more advanced features, better

performance, and more security for users. Another reason for the shift in future systems, however, will be the fact that almost all information communicated through such systems will already be digital, both between individual users and between computers.

This shift toward digitizing all information for transmission does not mean that all data should be treated as equivalent. Another trend in future wireless communications should be differentiated support for heterogeneous traffic, such as voice, data and video. Different types of data traffic will have varying requirements for transmission through the communications system, including different requirements for latency, error performance and overall data rate.

At the same time that usage becomes more widespread, users will also desire better performance: not only higher data rates for future applications, but also better reliability and coverage. In fact, users will want these systems to work as well as conventional wire-line systems, just without the wires. The desire for wireless connectivity will include not only increased total area of coverage, but also the ability to transparently move within zones, maintaining reliable connectivity using lightweight, low-power communication devices. This will lead to a wider range of operating conditions within the wireless channel due to mobility and the requirement for communications services in diverse environments.

In order to meets these demands and fulfill the vision of universal connectivity, wireless systems of the future will have to provide a level of flexible and efficient service not seen in current systems. These systems will have to carefully manage limited resources such as spectrum and power as they provide services and performance far superior to any available today. Furthermore, this increased flexibility will have to extend to those layers of the system that interface with the wireless channel: the *physical layer.*

The algorithms that perform the processing in this layer are directly impacted by the dynamic conditions and increased demands for efficiency and performance implied by the trends above. These algorithms will have to provide flexibility in the way that they accomplish their work under changing conditions in the wireless channel, yet they will have to do this in a way that provides efficient use of power exceeding the best systems of today. In this work, we demonstrate that it is possible to design algorithms that provide both flexibility *and*

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efficiency to meet the challenges that lie ahead.

2. FLEXIBLE DESIGN OF WIRELESS SYSTEMS

Communication is the transfer of information from the source to the destination. In this process, resources are consumed: electrical power, EF spectrum, computational resources or elapsed time. In this chapter, we review how future trends will make it more challenging for mobile wireless systems to accomplish their communications objectives while efficiently using their limited resources.

We begin with a review of some of the functions to be performed in the physical layer processing for a wireless communications system and some of the relevant characteristics of the EF wireless channel. We then describe how decisions about the allocation of system resources impact the ability of the system to be flexible and efficient. In particular, we will show that a *static* allocation of system resources to individual users and a *static* design of the individual wireless links will result in unacceptable levels of inefficiency. It is this need for an adaptive, flexible physical layer implementation that motivates the work in this thesis.

2.1 Signal processing in the Physical Layer

The physical layer provides the interface between the higher layers of the system and the underlying physical communications medium, the analog wireless channel. Processing steps required in the physical layer are shown in Figure 2-1,

Processing in this layer has long been referred to as *signal processing* because it has involved continuous signals instead of discrete data. In the transmitter, the *modulation* process transforms digital information into continuous signals appropriate for the wireless channel. Because the wireless channel is a shared medium, the receiver performs functions that provide for *signal isolation* in addition to *demodulation*. After

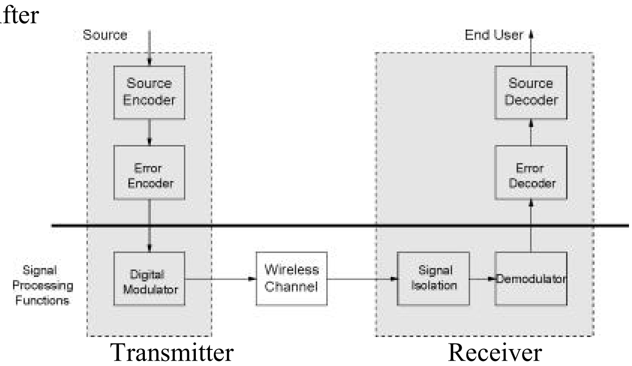

Figure 2-1: A wireless communications system.

Isolating the signal components corresponding to the desired transmitted signal, the demodulator recovers the digital data from these received waveforms. In the figure we see that the higher layer functions, such as source and error correction coding and decoding are separated from the signal processing functions.

In the earliest *analog* wireless communications systems, these signal processing functions encoded analog source signals directly into RF signals, so the source and error coding stages were not present. As systems were developed to communicate digital information, signal processing was still performed on continuous signals in the analog domain. In modern systems, however, significant portions of the physical layer processing are now performed using *digital signal processing* (DSP) techniques.

Digital signal processing still involves signals, but they are now sampled and quantized representations of continuous waveforms. The processing of these discretetime signals is performed as numerical operations on sequences of samples. Signal processing has now become a *computational* problem and systems are designed with ever greater portions of the "physical layer" implemented in the digital domain due to its relative advantages in cost, performance and flexibility [Frerking, 1994],

When we consider the different layers of the wireless communications system in Figure 2-1, we see that the physical layer is typically the largest consumer of resources in the system. For example, several studies have found that wireless network adapters for personal digital assistants often consume about as much power as the host device itself [Loreh and Smith, 1998, Stemm and Katz, 1997], Within the wireless network adapter, almost all of this power will be consumed in the physical layer implementation. This becomes clear when we consider that the upper layers of the protocol stack might require only a few operations *per bit* in an efficient implementation. In the physical layer, however, the signal processing might require tens of thousands of arithmetic operations to communicate a single bit from source to destination [Mitola, 1995],

Software Signal Processing

Moving the physical layer processing into software allows it to be treated as one part of the total processing required in the data communications process. The signal processing functions are computationally intensive and need to be well-matched to the properties of the wireless channel, but they are still only part of the processing chain whose overall goal is efficient and reliable communication. Although there has traditionally been a hard partition between the signal processing functionality and the higher layers of the system, that line in beginning to blur as more functionality is moved in to software. This allows signal processing functions to be more tightly integrated with the higher layers of the communication system, and also allows us to consider the design of the system as a whole, instead of separate parts.

In the chapters ahead, we demonstrate how to build the components of the physical layer in a way that provides not only the flexibility required for future systems, but also improved computational efficiency relative to existing techniques. As we have investigated this area, we have tried to incorporate and apply techniques and principles that have been used successfully to design efficient computer algorithms and software systems in the past. As a result, we have been able to identify new directions and new techniques that we believe will have significant impact on the design of future wireless

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communication systems.

2.2 Effects of Mobility on the Wireless Channel

In this section, we review some basic properties of the wireless KF channel and attempt to show how the desire for greater mobility and wider coverage leads to a more dynamic and challenging operating environments for wireless systems.

Effects of Diverse Communications Services

We have seen that the trends toward increasing mobility and more widespread coverage in future wireless systems can have a significant impact on the performance of a wireless systems. Another such trend is the **diversification** of available services, which results in increased heterogeneous traffic. Different types of traffic have significantly different demands on the communications system. In the ease of voice and data services, the different requirements for error rates, latency and jitter can lead to completely different system designs. These differences can be seen in many different parts of the system design, such as source coding schemes for compression, error coding techniques, medium access control (MAC) protocols, etc. In the ease of the MAC protocol, eonstant-bit-rate data, such as voice, is best handled using a reservation- based access scheme, whereas bursty data traffic leads to a contention-based access control scheme [Stallings, 1990], There are certainly hybrid approaches that try to provide better performance in the ease of mixed traffic, but any static solution will be a compromise that either inefficiently uses resources or limits service flexibility. The wider variety of channel conditions and desired services seen by the system will impact the design at the same time as capacity demands will require increased efficiency. The diversity of conditions and services may lead to different, even conflicting, demands on the system and will limit the ability of the system design to be optimized to improve efficiency and performance.

Design of Wireless Communications Systems

To identify aspects of the design process that might be profitably changed to meet the demands of designing the flexible and efficient systems of tomorrow, we begin with the end-to-end principle for system design. This principle provides guidance as to the best way to implement functionality in a communication system. In the classic paper on the subject by Saltzer, Reed and Clark [Saltzer et ah, 1981], the authors state:

3. BALANCING FLEXIBILITY AND PERFORMANCE

From the preceding chapter we see that there is a mismatch between the highly dynamic nature of the wireless channel and the use of static design techniques for the physical layer processing of the associated signals. While this mismatch has been tolerable in the past, growing demand for mobile wireless connectivity will make it necessary that the wireless systems of the future use their resources efficiently to support heterogeneous traffic over a wide range of channel conditions and user constraints.

In particular, wireless communications system will require the ability to:

- Adapt to significant changes in the operating environment or in the desired end-to-end functionality,
- Provide this dynamic functionality using, in some appropriate sense, a “minimal” amount of resources, thereby allowing the system to recover or conserve its limited resources, and
- Gracefully degrade performance in situations where desired performance is beyond the capabilities of the system for current conditions.

In this work, we make some initial steps in the direction of designing a more flexible communications system. We begin this process at a fundamental level: the level of the algorithms that perform the signal processing required for the wireless channel. These signal processing algorithms are the workhorses that couple the digital user to the physical EF channel. If low-level algorithms cannot operate effectively in changing conditions, then there is little hope of building from them a more complex system that can.

In this chapter, we present some of the major themes that have been significant in the algorithm development work that comprises this thesis. These themes can viewed as general approach to DSP algorithm design that has resulted in number of useful algorithms for flexible wireless systems. They also highlight some approaches to algorithm design that are relatively uncommon in the design of DSP algorithms. These kinds of approaches are more common in the field of computer science, particularly in the areas of computer system and computer algorithm design.

3.1 Key Elements for Flexible Algorithm Design

In this chapter, we present some of the common themes that have emerged in the development of the different algorithms we present in this thesis. These themes are:

- the identification of specific modes of flexibility that are useful in providing overall system efficiency under changing conditions and performance requirements,
- the identification of explicit relationships between input and output data samples for each processing function, and
- the use of the results of (1) and (2) to develop efficient algorithms through the removal of unnecessary intermediate processing steps and the use of techniques with statistical performance characteristics.

3.1.1 Flexibility to support function and performance

Up to this point, we have simply stated we want to design algorithms that are flexible. Here we attempt to provide a clear picture of what this flexibility looks like and how we decide which types of flexibility are appropriate.

What are the specific types of flexibility that are desirable and what kinds of flexibility are unnecessary? Before answering this question, we first note there are two general ways that flexibility can be manifested. In some cases, flexibility might provide a smooth change in an algorithm’s behavior as conditions change, for example, an algorithm needs to compensate as signal strength slowly fades as a mobile recedes from the wireless base station. In other cases, flexibility might provide a different choice in a set of discrete operating modes, A step change may be triggered by some gradual change that reaches a threshold value, or it may be due to a sudden change in conditions or desired functionality, for example, a mobile might hand-off to a different base station because of motion or to a different operating mode in response to a change in type of traffic (e.g,, voice to data)

It is also helpful to understand more precisely how providing flexibility within an algorithm can force us to give up **efficiency** in processing. One way to understand this is to realize that **efficiency** is often gained when multiple abstract processing steps can be combined for implementation. The principle is widely used in different areas of computer science, from computer algorithm design to compiler and computer language design.

Providing flexible operation affects performance because it introduces partitions that limit our ability to combine processing steps. To see this, consider a hypothetical processing system that consists of a series of abstract stages. Many factors influence where we place partitions in an actual implementation; we note a few that are relevant to our work. We might need to place partitions in the actual implementation at points where:

- subsequent processing depends on information not known at design time (late binding or re-binding), or
- there is a need to reduce excessive complexity that would occur with a larger composition of processing.

We will see examples of each of these partition decisions as we examine the different DSP functions in our wireless communications system.

This common step of determining specific desired modes of flexibility is an important first step of the algorithm design process. As we examine each function in our system, we try to compose functions where possible, but retain modularity to provide desired modes of flexibility. We do not have any general rules, other than to say that we determine the desired modes of flexibility in each stage by considering the basic functions of each stage in light of the flexibility desired of the overall system,

4. FLEXIBLE PROCESSING IN A DIGITAL TRANSMITTER

In this chapter, we begin our examination of specific signal processing functions in wireless communications systems. In particular, we start with the signal processing functions required in a wireless transmitter.

In chapter, we described a transmitter as being composed of two different functions: the source encoder and the channel encoder. We now further decompose the channel encoder into smaller components, as shown in Figure 4-1, This figure shows two distinct processing steps: error correction coding and digital modulation. The input to the channel coder is a bit-stream that has typically been processed to remove redundancy, thereby achieving an efficient representation. The error correction coding is a step that intentionally re-introduces limited redundancy in order to protect the data as it is transmitted over an unreliable channel. Sufficient re-dundancy is introduced to enable the receiver either to correct errors incurred in transmission, or to detect such errors and recover through other means such as retransmission, Error coders have been extensively studied elsewhere, see for exam-pie [Wicker, 1995, Berleeamp et al,, 1987], and will not be further discussed here.

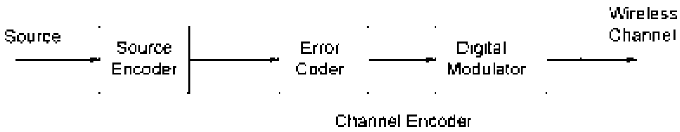

Figure 4-1: Stages of processing within the Channel Encoder

Instead, we focus on the second processing step of the channel encoder: the digital modulator. The modulator performs the processing required to transform the data into a form appropriate for transmission through a particular physical medium or channel. After examining the specific functions required in this modulation step, we review some of the conventional approaches used to perform the required processing in the transmitter.

The remainder of the chapter will then be devoted to a presentation of a novel technique for digital modulation that extends some of the ideas of DDS to more complex digital modulation waveforms. We describe how this technique, which we term *direct waveform synthesis* (DWS), enables the creation of an efficient digital modulator appropriate for many different types of wireless systems.

4.1 Overview: The Digital Modulator

In this work we concentrate on modulation techniques for encoding digital data into signals. Hence we are concerned here only with systems that transmit digital information, Many wireless communications systems are designed to communicate analog waveforms (often representing voice or images) directly using techniques such as amplitude modulation (AM) or frequency modulation (FM), but the current trend in wireless systems is to transform such waveforms into some digital representation for transmission.

Another important characteristic of the techniques we examine is that they are used to generate waveforms in the *digital domain.* We do not generate continuous waveforms (which can be used to represent digital data, e.g, a square-wave voltage signal) that can be coupled directly to an antenna for transmission, but rather sequences of discrete samples that *represent* continuous waveforms. This approach requires conversion, at some point, of this digital representation into an analog waveform for transmission, but there are many significant advantages to separating the functions, as we shall see.

In this Chapter we described some distinguishing characteristics of the wireless channel, Understanding these characteristics is crucial as we consider the choice of wave-

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forms to transmit digital information through these channels. One important characteristic is that a wireless EF channel is a *passband* channel, requiring the *translation* of generated signals to a higher frequency for transmission. This is precisely what happens in AM radio: the relatively **low-frequency** components of audible sound waveforms are translated to a much higher frequency band for transmission. The AM receiver then translates the signal back down to *baseband* to reproduce the original sounds.

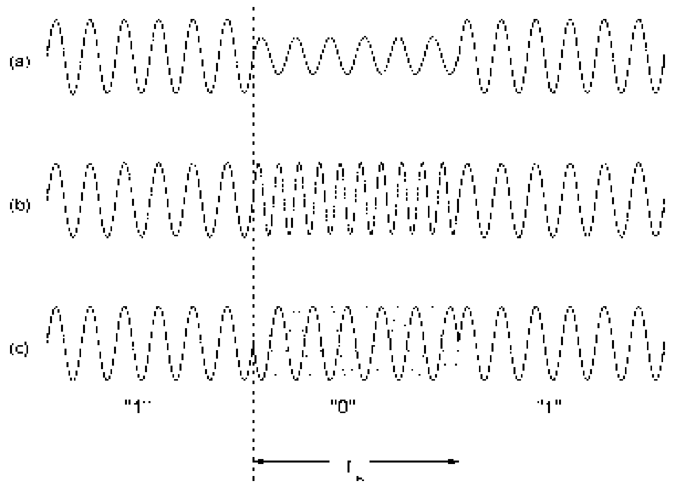

Figure 4-2: Typical modulation techniques for encoding bits into a continuous waveform: (a) amplitude modulation, (b) frequency modulation, and (e) phase modulation.

In a digital wireless system, it is useful to think of the modulation process as encoding the digital data into a high frequency signal. In Figure 4-2 we show several common techniques for encoding digital information into a continuous waveform. These various techniques modify, or modulate, a high frequency *carrier* signal according to the two possible values of the bits of the original data; in Figure 4-2, the modification is made to either the amplitude, the frequency, or the phase of the sinusoidal carrier signal, respectively. There are also modulation techniques that can modify the carrier waveform to produce more than two distinct patterns. These techniques might use multiple values for amplitude or combine modifications to both amplitude and phase to create a larger set of distinct patterns to encode data. These approaches can then encode larger alphabets than just zero or one. Instead, they can encode B bits as a single symbol into a section of the carrier waveform by using 2^B distinguishable modifications of the carrier sinusoid. In such a system, the task of the receiver is to distinguish among the 2^B different possible forms of the signal in order to determine which symbol was transmitted in each specific time interval.

Another point to note regarding Figure 4-2 is that, in each ease, the encoding of successive symbols is separated into disjoint time intervals. This is not true in general, and there are very good reasons why the sections of the encoded carrier waveform corresponding to continuous symbols often overlap significantly. We examine the implications of these overlapping segments more in the sections to come.

Many different modulation techniques have been devised for encoding a sequence of input data symbols into a waveform that can be transmitted through a wireless EF channel. Phase shift keying (PSK), frequency shift keying (FSK) and quadrature amplitude modulation (QAM) are just few of the more common techniques. The wireless environment can vary significantly for different systems, and modulation techniques are designed to provide different sets of trade-offs between system performance and resource consumption. Some modulation formats can be optimized for high data- rate, power-constrained applications, while others might enable relatively simple and inexpensive receivers [Lee and Messersehmitt, 1994, Proakis, 1995],

A flexible technique that can support many different modulation techniques is clearly useful. If we desire a system that can provide efficient operation in a wide range of environments while supporting different applications, it is evident that we may need to change between different modulation techniques as we encode data for wireless transmission.

4.2 Conventional Digital Waveform Generation

The first subsection presents a typical technique for encoding digital information into waveforms using pulse-amplitude modulation (PAM), which can be generalized to include a broad class of linear modulation techniques that are useful in different environments and applications. Following this, we review a technique known as direct digital synthesis (DDS) that has been used to efficiently generate simple discrete waveforms, such as sinusoidal carrier waveforms.

Following this review, we show how to extend some of the key ideas of DDS to the synthesis of more complicated waveforms, and we describe a new technique for digital waveform synthesis that provides many of the advantages of DDS in a more flexible and efficient digital modulator.

4.3 Direct Waveform Synthesis for Modulation

In the previous section, we discussed two conventional techniques for generating discrete waveforms for communications systems. We first discussed a typical imple-mentation of a digital modulator, which computes each output sample using the summation representation shown in (4,2), We also discussed DDS: a technique for synthesizing relatively simple waveforms, such as sinusoids, using a look-up table of pre-eomputed output samples. This technique required only a simple computation to find the correct index in a table for each output sample to be produced.

In this section we apply some of the ideas from DDS to produce a new simple and efficient digital modulation technique. Our goal here is to produce a digital modulator that can easily synthesize a discrete waveform using a table of pre-eomputed output samples, thereby greatly reducing the computation required in a digital transmitter. There are two challenges that must be overcome to produce a practical digital modulation technique that uses less computation. First, the technique requires a mapping from input bits to output samples that works in the general ease of overlapping transmit pulses. Equation (4,2) showed that each output sample might

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depend on a large number of input bits; we will show that this condition does not preclude look-up table technique similar to DDS, but that it can lead to prohibitively large look-up tables in some situations.

This is the second challenge: many DDS implementations use table with tens to hundreds of entries, and many tricks have been developed to reduce even these modest tables to smaller sizes [Frerking, 1994, Reed, 1998], The most straight forward lookup tables for a digital modulator can contain thousands or millions of entries. Such a modulator may not be feasible, even though 256 megabits of memory may soon be available on a single chip.

We overcome this second challenge by decomposing a large look-up table into smaller tables, whose total size is also much smaller than the single original. Different decompositions provide a wide range of operating points that enable a flexible tradeoff between computation and memory requirements.

In the next sections, we describe the *direct waveform synthesis* (DWS) technique for directly synthesizing a baseband digital waveform. We also extend the basic DWS technique to allow the synthesis of digital waveforms at passband, thereby removing the need to translate the baseband waveform to IF, and also to allow the mapping of multiple symbols at a time, thereby providing more flexibility in the computation- memory trade-off. Direct waveform synthesis provides a significant performance improvement over conventional approaches to digital modulation. This gain comes from its ability to directly synthesize waveforms using a simple table look-up based on the input bits to the modulator. We have also presented a technique that provides a flexible trade-off.

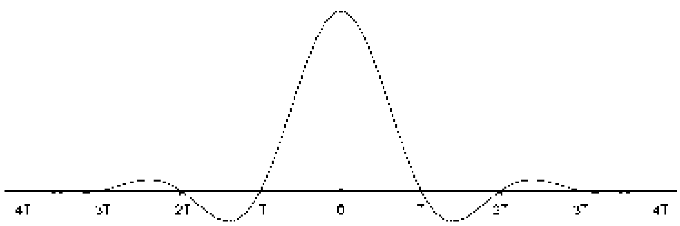


Figure 4-3: Typical sin(a:)/a: pulse shape for a bandwidth-efficient system, where $x = 7Tt/T$ and T is the symbol interval.

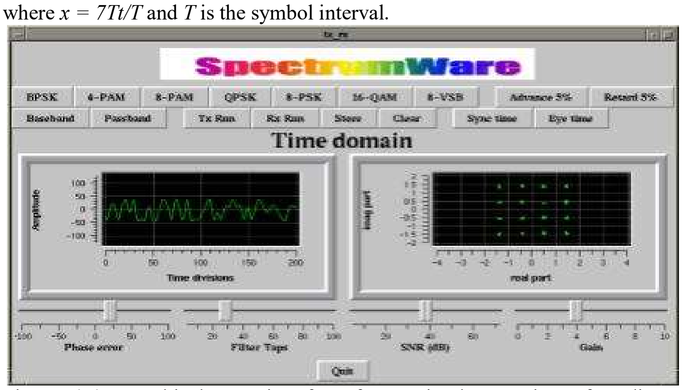
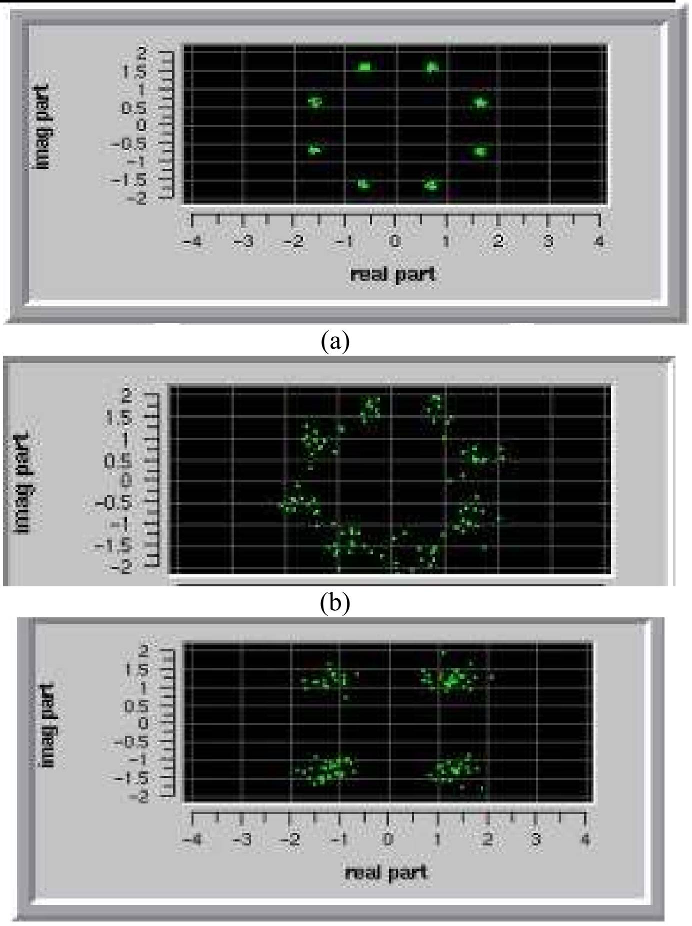


Figure 4-4: Graphical user interface for an implementation of a direct waveform synthesis digital modulator.



(a)

(b)

(c)

Figure 4-5: Received signal constellation diagrams for (a) 8-PSK under relatively high SXR (b) 8-PSK under low SXR and (e) 4-PSK under low SXR

In the context of a wireless communications system, the digital modulator does not need to provide much flexibility during operation. Any change to the modulation format will require coordination between receiver and transmitter, and such changes will therefore tend to be infrequent, DWS provides a good balance of efficient performance, flexible implementation properties and the ability to be reconfigured if channel conditions or system requirements so dictate.

5. FLEXIBLE PROCESSING IN A DIGITAL RECEIVER

We have characterized the channel coder function in a wireless transmitter as a one- to-one mapping from a sequence of data bits into a sequence of distinctive pulses for transmission. At the receiver, the primary task is to reverse this mapping, thereby recovering the original data bit-stream. After this channel decoding, source decoding reproduces the original source information.

In Figure we see a representation of the processing steps performed in the receiver. Also shown in this figure is a further decomposition of the channel decoder into several specific functions. The first two of these functions are channel separation and symbol detection.The sequence of bits produced at the output of the detector may contain errors.

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Because the original encoded sequence contained some controlled amount of redundancy, the error decoder can detect the presence of many errors and either correct them or take other appropriate actions to handle the corrupted data.

In this work, we examine the processing required for the functions of channel separation and symbol detection. Our first goal is to develop a more fundamental understanding of the functionality required by these two processing steps in the digital domain. We then use this understanding to develop algorithms that provide both flexibility and improved efficiency over conventional techniques in a wide range of situations.

The channel decoder is traditionally divided into the specific steps of channel separation and detection because the wireless channel is a shared medium that combines many signals. We therefore model the input to a digital receiver as a sum of an indexed sequence, s_n, representing the signal of interest, with I interfering signals from other nearby transmitters and a random noise component.

The overall process of channel decoding requires the recovery of the transmitted data from this received signal. Algorithms for this type of analysis are sensitive to interference in the input signal, but often perform well in the presence of only additive random noise.

In the next few sections we discuss the problems of channel separation in more detail and describe current techniques used to solve these problems. We then present several new techniques that enable us to perform the functions required in a channel separator in manner that provides both flexibility and greater efficiency than conventional approaches. We discuss the problem of detection, the second stage of processing, in the next chapter.

5.1 Overview: Channel Separation

We have seen that the discrete sequence of samples at the input to a wideband receiver is the sum of the desired signal and undesired interfering signals. We also assume that the signal contains additive random noise. This noise includes several effects, such as low level random signals picked up by the antenna and thermal noise generated inside the analog circuitry of the receiver.

The goal of channel separation is to prepare the received sequence, enabling the detector to recover the original data sequence from the received signal. Stated another way, the channel separator should produce a sequence containing components in the received signal corresponding to the signal of interest, while removing the interference. To facilitate separation, it is common to restrict, through regulation, the emissions of potential interfering transmitters, so that any interfering signals will occur in disjoint frequency bands in the RF spectrum. This technique for providing *'multiple access* to the physical medium is known as *frequency division multiple access* or FDMA, Other techniques such as time division multiple access (TDMA) and code division multiple access (CDMA) can also be used to allow multiple transmitters to share the same band of frequencies, but are not addressed here. In FDMA, separating the desired signal from potential *adjacent channel* interferers is equivalent to extracting from the received sequence only those components that occur within the band of frequencies corresponding to the signal of interest.

This separation is accomplished through the use of digital filters that pass signal components in the desired band of frequencies, the *passband*, and attenuate signal components outside of this band, in the region known as the *stopband.* In a wideband receiver the use of digital filters to perform channel separation can lead to an implementation with very high computational complexity. For a receiver in which the channel separation is performed in the digital domain, the work required to extract individual channels from the input of a wideband receiver provides a hrst-order estimate of the computational resources required for the *entire* receiver [Mitola, 1995, Wepman, 1995],

The reasons for this high computational burden are several. First, a wideband digital receiver will necessarily have a high input sample rate to adequately represent the wideband input signal. In addition, to cleanly extract a narrow band of frequencies, the receiver must accurately resolve those frequencies at the boundary of the band of interest. These two conditions combine to produce a situation that can require large amounts of processing to perform channel separation using conventional techniques. Before addressing this problem, however, we examine the basic steps that are widely used to perform the entire channel separation procedure using digital filters.

In Figure we see the conceptual steps that are often used to perform channel separation. The results of these same steps on a hypothetical wideband signal (represented in the frequency domain) are shown in Figure 5-3, First, there is a *frequency translation* step. Here the wideband signal is shifted in frequency to bring the band of interest into the passband of a digital filter and the interfering signals into the stopband, This translation is represented in Figure 5-2 by a multiplication of the received sequence and a complex sinusoidal sequence. The magnitude of the frequency shift is determined by the frequency of the complex sinusoidal sequence.

The second step in this process is *bandwidth reduction.* Here the filter produces Complex Sinusoid Frequency Bandwidth Sample Rate Translation Reduction Reduction

an output sequence in which the components at different frequencies are attenuated or amplified according to the frequency response of the **filter.** Because this output sequence does not contain any significant components outside the band of interest, it is now possible to represent the desired signal using a lower sample rate without significantly distorting those components within the band of interest. If there were significant signal components present outside the band of interest, reducing the sample rate would cause *aliasing* of those components into the band of interest [Oppenheim and Schafer, 1989], Hence a final step of *sample-rate reduction,* or *decimation,* reduces the rate of samples in the sequence by a constant factor *D,* the decimation rate. This rate reduction is also important because it reduces the processing load in subsequent stages.

5.4 Evaluation of Random Sub-sampling

Figure 5-11 shows the results of computer simulations that validate the performance of the sequence transformer analysis for the ease where the autocorrelation is not known (Case I), In these simulations, random WSS signals were generated to represent the received signal, A selection sequence z_n was produced to simulate independent random rounding decisions according to the transformation rule in (5,45), In the figure, the solid lines represent the result of (5,46) plotted for specific values of *Rr[0]* and Cft[0] over a range of values for *B.* in dB relative to the power of the received signal. The individual points on the plot are the measured results of the error variance produced when sub-sampling the random signal by zeroing specific proportions of the input samples. The plot shows results for five different values of qJ0], corresponding to different bandwidths for the desired signal relative to the wideband input. The results show that if we need to bound the distortion at some level, say —15*dB* relative to the received signal, the approach of Case I would provide some reasonable reduction in computation. The results shown are for the base ease where there are no interfering signals in adjacent channels and all of the signal energy is in the desired band. If any other signals are present in the wideband input signal, the respective curves would shift upward to reflect the increased value of R_r[0], the input signal variance,

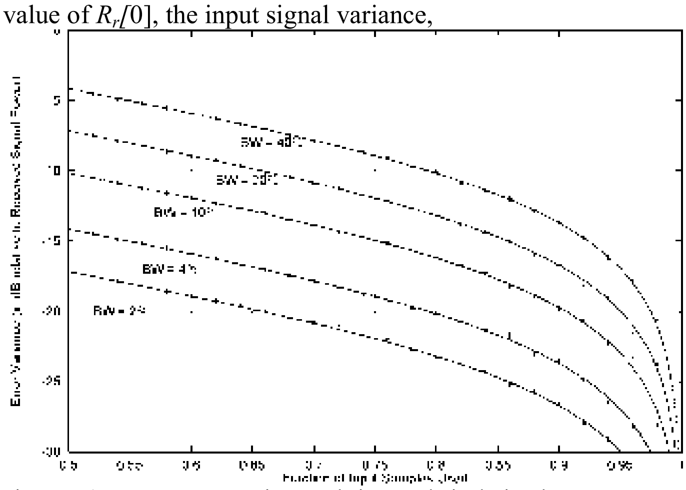

Figure 5-1: Output error variance relative to desired signal power versus proportion of input samples used for several eases of output signal relative bandwidth.

A contrast between the approach of Case I and an alternative, a uniform subsampling scheme, is their behavior in the presence of interference. The qualitative effect of the random sub-sampling approach is to provide more uniform behavior, regardless of the location of any interfering signals relative to the desired signal.

For example, consider the ease where a single interfering signal is present in the stopband of the channel filter. For the random sub-sampling approach, this would result in an upward shift of the curves in Figure 5-11 by 3 dB when the interferer is independent of the desired signal and has equal power (so [0] will be twice as much as with no interferer present). This result does not depend on the *location* of the interferer within the stopband: the interfering signal could have a center frequency anywhere in the stopband and the result would be additive uneorrelated distortion at about -15 dB relative to the desired signal (using half of the samples with a relative bandwidth of 2% in Figure 5-11).

If half of the samples are instead chosen *uniformly*, this is equivalent to decimation by a factor of two prior to filtering. The resulting effect would depend on the relative frequencies of the desired signal and the interferer. The interfering signal might be aliased into the stopband of the filter, resulting in negligible distortion. On the other hand, it is possible that the interferer would be aliased into the passband of the filter by the decimation, and this would result in significant distortion at the filter output (zero dB relative to the desired signal).

6. DATA SYMBOL DETECTION

The job of the channel decoder is to analyze the signal recovered from the channel in order to determine the original bits that were sent by the transmitter. In the previous chapter, we described this process as a manv-to-one mapping since there are many ways that the channel can corrupt the transmitted waveform sensed by the receiver, and the receiver must map each of these corrupted versions back to the original discrete data. We have already shown how the process of channel separation deals with the part of the channel distortion caused by interfering signals in other frequency bands. In this chapter, we will examine the second step of the process, detection.

We begin with an overview of the detection process, emphasizing the relationship between the input samples and the symbol to be estimated. We then review one widelv-used approach to detection that relies on a special property of certain pulse shapes enabling us to make optimal symbol-bv-symbol decisions. We then spend the remainder of the chapter describing a new approach to detection that enables the system to adapt the detector to efficiently provide a specified level of performance for existing channel conditions.

Overview: Symbol Detection

After channel separation has been completed, the receiver
Figure 6-2: Plot of bit-error rate versus SNE for a two level PAM system.
must make decisions about the individual symbols encoded in the received waveform. The sequence of processing steps in the receiver is shown in figure 6-1: in this figure, the symbol detector is divided into two different tasks, *synchronization* and *detection.* Synchronization is the process of determining the specific set of received samples that correspond to each particular symbol to be detected. The job of the detector is then to analyze the specific section of the received waveform that corresponds to a particular symbol and decide which is the most likely corresponding data symbol. In this chapter, we focus on the detection process under the assumption that the required synchronization has been accomplished, A more detailed discussion of techniques for synchronization in digital receivers is available in [Meyr et ah, 1997],

The output of the detection process is the sequence of symbols that the receiver believes was originally transmitted.

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This sequence is passed to the error decoder, which will either correct those errors it can, or simply detect errors and take other appropriate actions. It is in this context of protecting against errors in the receiver that the overall role of the detector in the system becomes clear. In a noisy environment, the detector will always produce some errors. One of the functions of the receiver as a whole is to control the level of errors through various mechanisms to ensure that *end user* sees only an acceptable level of errors. These mechanisms are not limited to the detector, or even the channel decoder, but include redundancy at higher layers in the communication system or other techniques such as protocols for retransmission. To design an efficient and flexible system, we need to provide the required error performance in a way that efficiently uses the available system resources.

When seen in this light, the detector does not necessarily need to produce an *optimal* estimate of the received symbols in every situation. Rather, it needs to recover symbols in such a way that the required error performance is efficiently achieved through the composite behavior of all the error control mechanisms. Figure 6-2 shows a typical performance curve for an optimal digital detector. In this plot, we see that the probability of bit errors depends on the SNR of the received signal. If a particular system is designed such that the detector needs to achieve a bit error rate (BER) of 10“^{4} at some worst-ease SNR, say 8,5 dB SNR, then it will provide much better BER SNR (dB) if the SNR is higher. For example, if the system is experiencing an SNR of 11 dB then the optimal detector will produce a BER of ICR^{6}, This extra performance will not hurt the overall system performance, but if it is achieved at the cost of unnecessary resource consumption, then it will not be the most efficient solution.

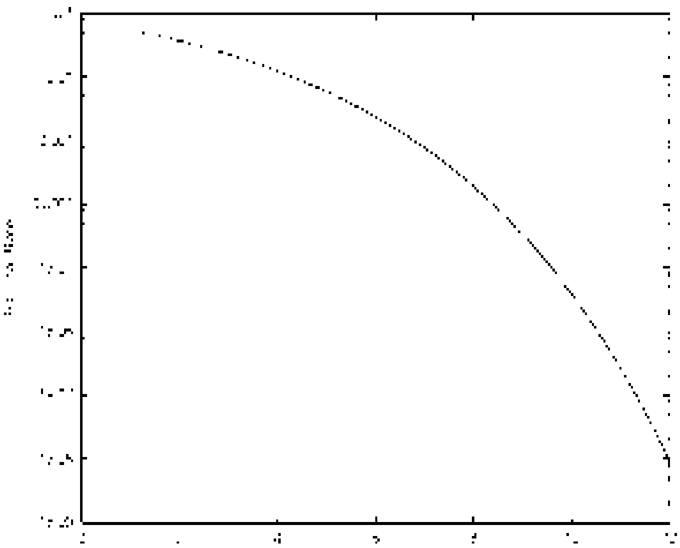

Figure 6-1: Plot of bit-error rate versus SNE for a two level PAM system.

In this chapter we will present an approach that enables the detector to efficiently provide a *desired* level of BER for a specific SNR from *current* channel conditions. Before presenting these results, we first review some conventional techniques to perform detection and try to develop an understanding of how this process can be modified to provide efficient and controllable performance trade-offs.

Conventional Approach to Detection

Conventional approaches to detection depend heavily on the type of distortion experienced in the wireless channel. In some eases multi-path reception causes a situation where a specific symbol in the encoded waveform is distorted by other preceding or following symbols. In such a ease, symbol-bv-symbol detection is not feasible, but the receiver must, in a sense, simultaneously estimate an entire sequence of symbols because of the *inter-symbol interference* (ISI) effects. We will not consider this ease in the present work, but will consider the ease of a system that experiences additive noise. We will consider this ease in the context of a system that has limited EF spectrum available and needs to efficiently use this limited bandwidth to achieve the best possible data rate.

As we consider some standard approaches to this problem, we will review how the system can use pulse shaping techniques to effectively use its spectrum as well as the concept of the matched filter detector,

Efficient Detection for Error Control

At this point, we have seen the conventional approach to a digital detector: the matched filter that provides an optimal decision under any SNR conditions and does so with a constant amount of computation for each decision. We now present a general framework for understanding the detection process that will not only allow us to represent conventional approaches to designing channel decoders, but will also enable us to develop a more general approach that allows more controlled symbol detection.

7. SUMMARY, CONTRIBUTIONS AND FUTURE WORK

In this thesis, we presented the results of an investigation of how physical layer processing algorithms can balance flexibility and efficiency in wireless communication systems. The work was motivated by a number of trends that are changing the type of service expected from wireless communications systems of the future. These trends include more dynamic environmental conditions, heterogeneous traffic, and increasing demands for efficiency and performance. All of these trends have implications on the physical layer implementation of the communication system.

Given these trends, we believe that it will be difficult, if not impossible, to design a static physical layer that will be able to provide efficient service in all of the different anticipated situations. This led us to investigate the design of DSP algorithms that provide processing that is both flexible and computationally efficient.

As we began to develop algorithms that could balance flexibility and efficiency, it became clear that we needed a precise understanding about the specific types of flexibility that would be useful in each situation. In other words, we needed an improved understanding of how a system could be designed to exploit better conditions: e.g,, how could it exploit better SNR conditions to lower processing requirements? In a number of eases addressed in this thesis, the answer to this question was that we could specify the requirements for processing in specific portions of the system so that in better conditions we could produce a sufficiently high quality result using less computation.

Furthermore, we began to understand that producing DSP algorithms to efficiently compute approximate results requires, in many eases, a more careful understanding of the specific relationship between input and output samples for the signal processing function in question.

In this thesis, this idea has been articulated through the concept of sub-sampling and the idea of the footprint of a desired result. The random sub-sampling techniques described in Chapter 5 recognize that an approximate result for a frequency selective filter needs to be computed from data that span the same interval of time, but need only use a subset of those samples in that interval. Similarly, the idea of a *symbol footprint* in Chapter 6 stems from the need to identify the precise set of samples that had information relevant to a specific symbol estimate in a detector. In this ease, an approximate result can be found using a subset of the samples in the footprint. This led to an investigation of which samples in the footprint were most useful and would therefore be used first.

In other situations, approximate results are not appropriate. In these eases, however, the same understanding of input-output data relationships led to algorithms that could produce the desired results using less computation by taking advantage of larger amounts of memory or removing intermediate processing steps.

In all of these eases, it was also important to understand the role of each processing stage within the large communications system. It is this understanding that led to the conclusion that approximate results would be sufficient in some eases, or that intermediate processing stages could be eliminated because they were unnecessary for overall system operation.

Conclusion

The dynamic nature of the wireless operating environment and the variable performance requirements of wireless applications is in conflict with the traditional static implementations of the physical layer. The dynamic channel and variable performance requirements preclude effective static optimization of the physical layer processing, and provide an opportunity for flexible implementations to dramatically improve overall system performance. This presents an opportunity for flexible algorithms that can enable the entire communication system to adapt to the changing conditions and demands, We believe that future wireless systems will have to incorporate this type of physical layer flexibility if they are to provide the types of efficient communications services expected by the users of tomorrow.

It is often assumed that flexibility in an algorithm comes at the cost of reduced computational efficiency. In this thesis, however, we have demonstrated a suite of flexible algorithms that take advantage of changing conditions and system demands to provide significantly improved performance relative to static designs.

Although the results of this work are very encouraging, we believe that the algorithms presented are just a start, and that there are still many significant and interesting problems to be solved. Complex system requirements and technological advances have led to a merging of the fields of digital signal processing, theoretical computer science and communications system design. We believe that effective designs for the systems of tomorrow will need to draw on results from all of these disciplines to meet the growing demand for communication services without wires.

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