CHAPTER I

INTRODUCTION

1.1 Literature Review

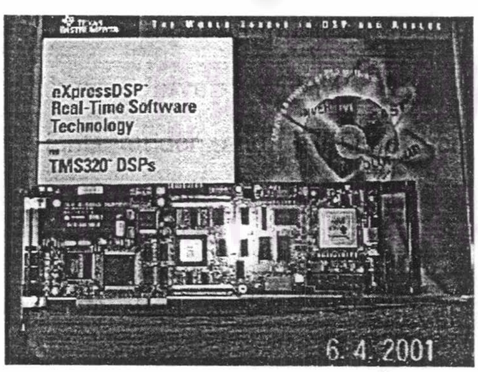
The concepts, theory, and basic algorithms of digital signal processing (DSP) have been established and applied for over thirty years. Recognizing the importance and intellectual challenges of this field, academic classes were initially taught at the graduate level and have more recently become core undergraduate classes for electrical and computer engineering programs [1]. Since the world has progressively decided to go digital using DSP, there has been a high demand for DSP trained professionals. Specific industries and applications that have experienced an increasing demand include: automobiles, digital and wireless communications, digital control systems, imaging and video, medical diagnostics, and data acquisition.

Since 1986, Roger Williams University [2] has devoted one section of the senior project course exclusively to DSP, initially based on the fixed point TMS32010/TMS320C25, and more recently based on the floating-point TMS320C30 digital signal processor. The course includes two parts; the first part is introducing the architecture and the instruction set of the digital processor by practical experiment, which helps students to be familiar with software and hardware tools. The second part is teaching about implementation of real-time final projects using C and/or TMS320C30 assembly code such as Ten-band multi-rate filter project.

At Bradley University, a number of design projects which involve the implementation of communications and control algorithms on digital signal processors have been carried out in recent years by undergraduate students in senior design project course and by graduate students in a graduate design project course. The projects include nonlinear Kalman-Bucy filtering on the TMS32010, FM demodulation on the TMS320C30, and the identification of plant transfer function on the DSP56001.

The California State University, Chico (CSUC), offers undergraduate students an electronics circuits for digital signal processing course, with two hours of lecture and three hours of laboratory per week. Two lab oriented texts [3,4] are used for the course which seeks to facilitate student understanding of DSP concepts by implementing real-time applications, and develop an appreciation for comparatives very long instruction word (VLIW) DSP architectures by working with a variety of DSP processors. Experiments similar to those performed at Roger Williams University.

An undergraduate digital signal processing (DSP) laboratory at Western Michigan University (WMU) [5] contains three main objectives, which to enhance the students' understanding of the theory taught in class, provide experience with DSP implementation issues, and to increase their interest and participation in this important field of electrical and computer engineering. The DSP laboratory have used the versatile, high-speed Texas Instruments' TMS320C6701 floating-point processor hosted on a TI Evaluation Module (EVM) as shown in figure 1(a) and 1(b). The TMS320C6701 EVM is the industry's first processor to cross the GFLOPS (giga floating operations per second) performance barrier. Applications for this processor are countless, including wireless communications, speech recognition, imaging systems, CAT scan, ultrasound, magnetic resonance imaging (MRI), and radar applications.



(a)

(b)

Figure 1 (a) TI TMS320C6701 EVM and (b) DSP laboratory workstation shows a typical PC with test equipment consisting of a signal generator, oscilloscope, and audio spectrum analyzer.

At the University of Southampton [6], students in electronic engineering can attend modules in communications, which are either taught or contain course work in terms of simulations that train advanced concepts but often omit basic but necessary communications problems such as synchronization and timing recovery. The fourth year module "real-time systems design" was therefore aimed at implementing a simple modem which combines the teaching of real-time system issues such as scheduling with essential knowledge of practical communications. The final project, which is the implementation of TMS320C6711 DSK, includes three main parts: transmitter, audio codec interface, and receiver (see in figure 2).

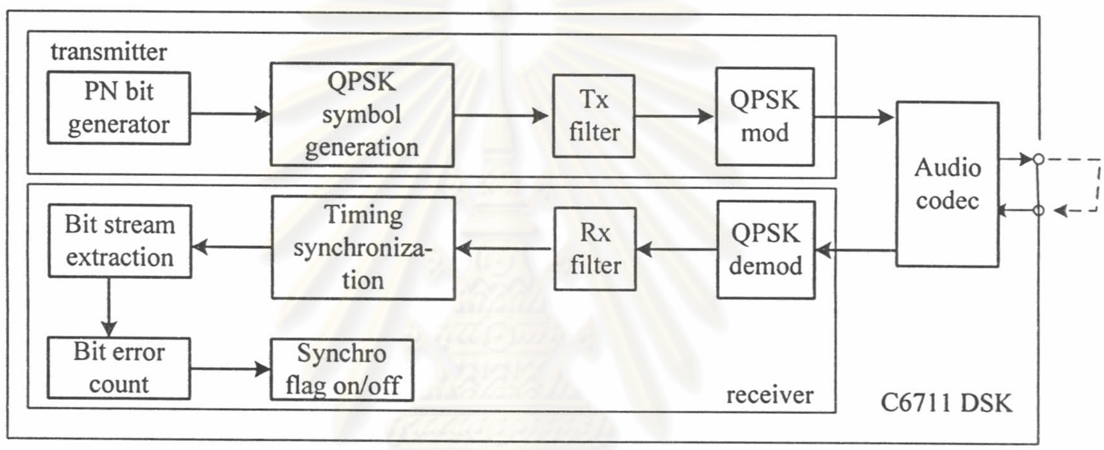


Figure 2 A differential QPSK modem using the TMS320C6711 DSK.

The Pseudo-Random Number (PN) Generator

Figure 3 is a 16 bit linear feedback shift register, which give maximum length sequence repeats every 2^{16} -1 iteration. The input to shift register is drawing by XORing the state values in registers 4, 13, 15, and 16.

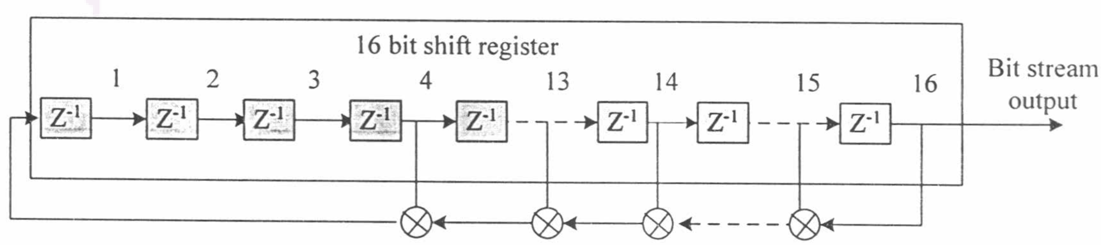


Figure 3 Maximum length sequence based pseudo random number generator.

The synchronization part is the correction of the bit stream in the receiver can be assessed by the system in figure 4, which fills the states of the same 16 bit shift register, comparing the feedback value with the most recent bit. Once in steady state, the output synchronization flag is permanently high in the absence of bit errors. The numbers of zeros occurs then represent the numbers of encountered bit error.

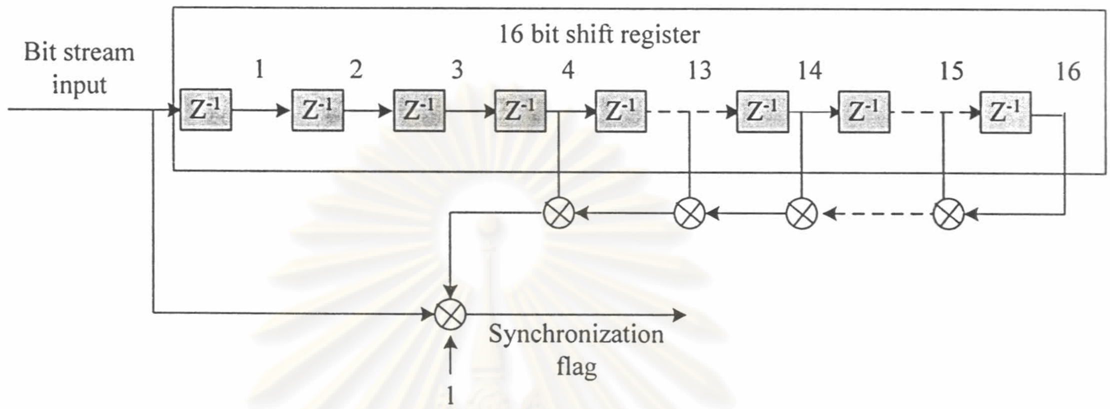


Figure 4 The linear feedback shift register of Fig 3 as a synchronization tool.

Quadrature Phase Shift Keying (QPSK)

QPSK maps two bit onto a complex symbol that can take on four possible values in the constellation plane. Multiplexing the bit stream into inphase and quadrature signals $I[n]$ and $Q[n]$ with $Q[n] \in \{0;1\}$,

$$x_0[n] = \sqrt{2}(\frac{1}{2} - I[n]) + j\sqrt{2}(\frac{1}{2} - Q[n]) = e^{j\psi[n]} \quad (1)$$

The original bit stream can be extracted from the reconstructed QPSK sequence in the receiver, $\hat{x}[n]$, by taking real and imaginary parts, and multiplexing the recovered inphase and quadrature components

$$\hat{I}[n] = \frac{1}{2} - \frac{1}{2} sign\{\Re\{\hat{x}_0[n]\}\} \quad (2)$$

$$\hat{Q}[n] = \frac{1}{2} - \frac{1}{2} sign\{\Im\{\hat{x}_0[n]\}\}, \quad (3)$$

Where $sign\{\cdot\}$ is the signum function.

Differential QPSK (DQPSK)

Since the QPSK constellation is invariant to shifts by $\pi/2$, π, and $3\pi/2$, the receiver suffers from an ambiguity. To mitigate this, differential encoding is applied to the phase of the transmitted QPSK symbols (see in figure 5).

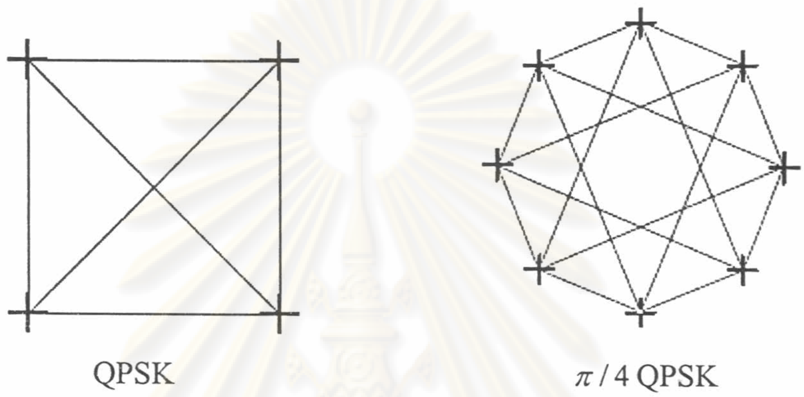


Figure 5 QPSK and $\pi/4$ QPSK constellation.

This can be based on the phase of the QPSK signal $x_0[n] = e^{j\psi[n]}$,

$$\psi_{diff}[n] = \psi_{diff}[n-1] + \psi[n] + \frac{\pi}{4} \tag{4}$$

using the circuit in figure 6. Alternatively, rather adding phase values, the QPSK symbols can be multiplied,

$$x[n] = x[n-1]\hat{x}_0[n]e^{j\pi/4} \tag{5}$$

whereby $x[n] = e^{j\psi_{diff}}$ is the differentially encoded QPSK (DQPSK) symbol stream.

Decoding can be based on the phase values of the received DQPSK signal $\hat{x}[n]$ as shown in figure 7 according to

$$\hat{\psi}[n] = \hat{\psi}_{diff}[n] - \hat{\psi}_{diff}[n-1] - \frac{\pi}{4} \tag{6}$$

with $x_0[n] = e^{j\hat{\psi}_{diff}[n]}$ or alternatively

$$\hat{y}_0[n] = \hat{y}[n]\hat{y}^*[n-1]e^{-j\pi/4} \tag{7}$$

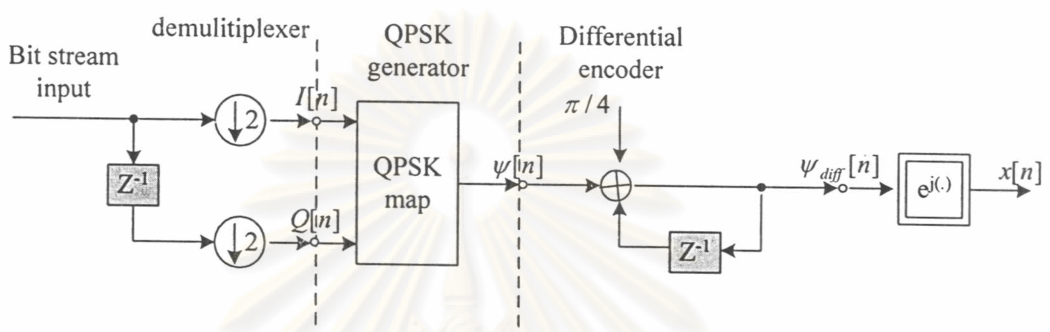


Figure 6 Differential QPSK symbol generations.

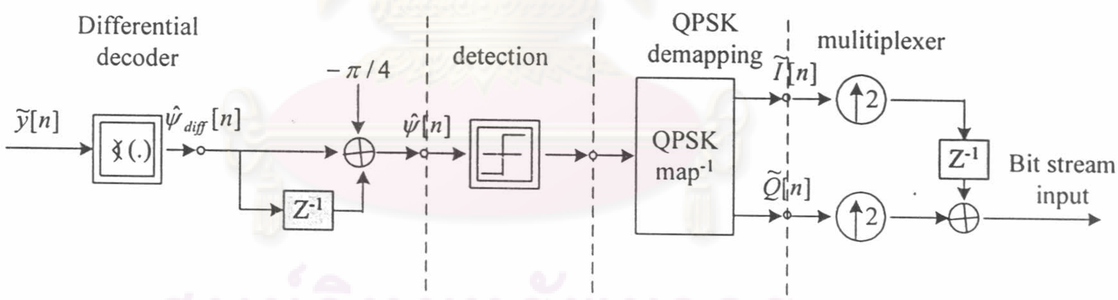


Figure 7 Differential QPSK decoding.

Transmit and Receive Filtering

In the transmitter they need to oversample the symbol stream by a factor of N = 8 in order to obtain a signal to be sent to DAC, by inserting N-1 zeros in between every original DQPSK sample. This is performed by the 8 fold expander shown in figure 1.8. The resulting signal is broadband, and needs to be band-limited through interpolation by a suitable transmit filter. In figure 9 shows a root-raised cosine filter with 81 coefficients and impulse response h[m] as characterized. Together with the receive filter in the receiver, the transmit filter forms a Nyquist system, i.e imposes no intersymbol interference (ISI) on the transmitted symbol stream.

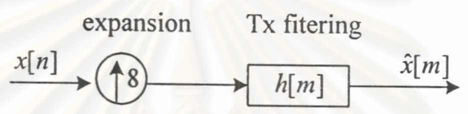


Figure 8 Transmit filtering with root raised cosine filter.

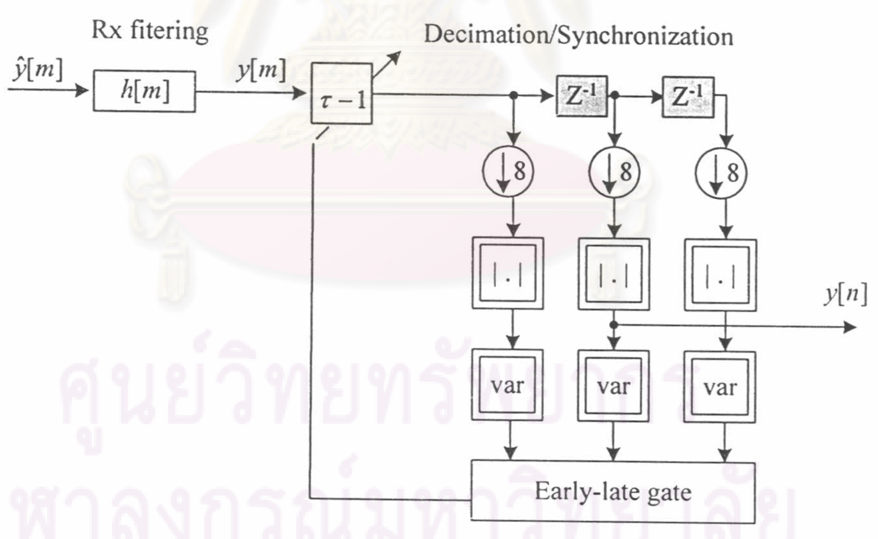


Figure 9 Receive filter with root raised cosine filter h[m] and decimation by N=8 after timing synchronization.

Modulation and Demodulation

To transmit the data, quadrature amplitude modulation of QPSK symbol stream is employed. This can be implemented as

$$s[m] = \Re\{\hat{x}[m]\} \cdot \cos(2\pi f_c / f_s m) + \Im\{\hat{x}[m]\} \cdot \sin(2\pi f_c / f_s m) \tag{8}$$

Where $f_c = 2000$ Hz, and $f_s = 8000$ Hz.

The signal r[m] received by the codec is demodulated as

$$\hat{y}[m] = r[m] \cdot e^{j2\pi f_c / f_s m} \tag{9}$$

Timing Synchronization

The most complex operation in the modem is timing synchronization in the receiver, at which stage a decision has to made on the exact sampling instance to retrieve the DQPSK symbol $\hat{y}[n]$ at 1kbaud (1000bits/second) from the 8 kHz receive filtered signal. The timing synchronization algorithm selected here is an early-late gate method [7].

TMS320C6711 DSK has been widely used as a teaching instrument in various universities. Nevertheless, at present a huge number of universities, such as the University of Wiscosin at Madison, the University of Wyoming, and Jonhs Hopkins University [8], have begun to incorporate MATLAB programming at the beginning of course before performing an implementation as show in figure 20. The main content of the laboratory is DSP theory, interactive demo, MATLAB program by student, and real-time DSK program by student (see in figure 10).

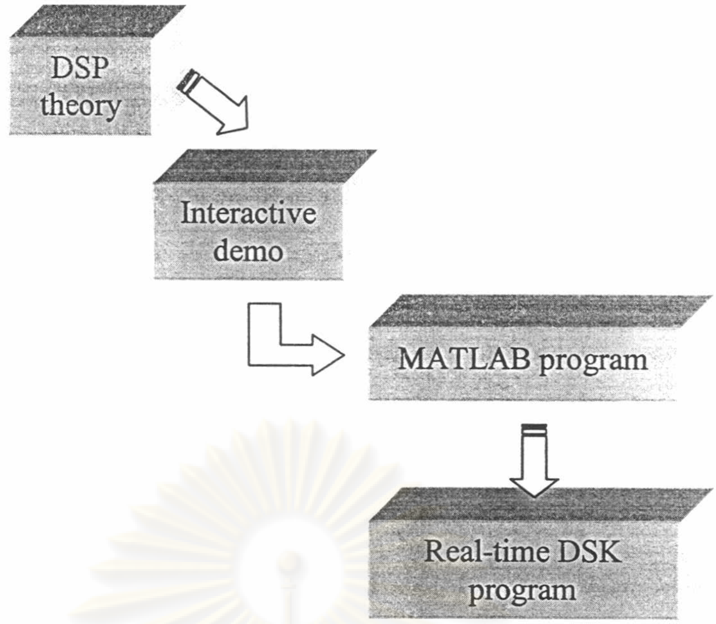


Figure 10 Teaching approach for DSP topics using the C6711 DSK.

Based on the above literature reviews, it is clear that there have been some efforts in development of real-time implementation used as teaching tools in many world-leading universities, inline with what we are aiming to do in this thesis. University of Southampton in particular adopted TMS320C6711 DSK in their experiment development, which is the same set of hardware we have at Chulalongkorn University. However, their implementations are intended for advanced experiments on digital communication. Therefore, students are expected to have sufficient knowledge prior to their experiments. On the contrary in our development the courseware is aimed at undergraduate students who have limited priori knowledge on both communications theory and hardware architecture. Therefore, our courseware design criteria will be fundamentally different form that of the university of Southampton. For our experiments, basic knowledge on communications theory will be provided in the first four experiments with comprehensive use of MATLAB. Once students acquire sufficient knowledge, they can continue to work effectively on hardware implementation. This proposed idea will result in a complete set of courseware on basic digital communications suitable for undergraduate students at Chulalongkorn University and National University of Laos.

1.2 Elements of an Electrical Communication System

Electrical communication systems are designed to send massages or information from a source that generates the massages to one or more destinations [8]. In general, a communication system can be represented by the functional block diagram show in figure 11. The information generated by the source may be of the form of voice (speech source), a picture (image source), or plain text in some particular language, such as English, Japanese, German, French, etc. An essential feature of any source that generates information is that its described in probabilistic terms. In other words, if the output of a source is not deterministic. There would be no need to transmit the message.

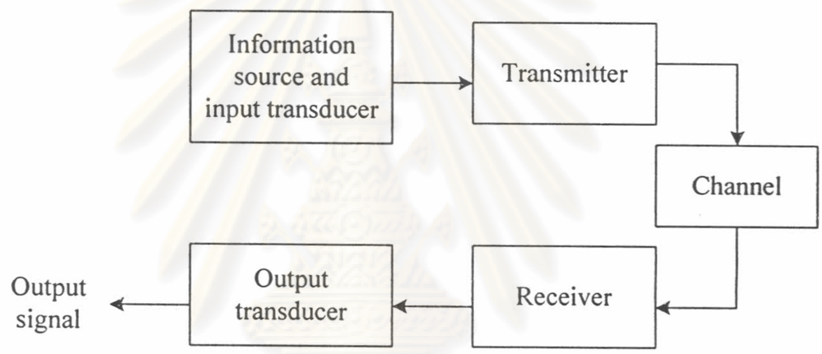


Figure 11 Functional black diagram of a communication system.

A transducer is usually required to convert the output of a source into an electrical signal that is suitable for transmission. For example, a microphone serves as the transducer that converts an acoustic speech signal into an electrical signal, and a video camera converts an image into an electrical signal. At the destination, a similar transducer is required to convert the electrical signals that are received into a form that is suitable for the user; e.g., acoustic signals, images, etc.

The heart of the communication system consists of three basic parts, namely, the transmitter, the channel, and the receiver. The functions performed by these three elements are going to be described.

1.2.1 The Transmitter

The transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. For example, in radio and TV broadcast, the Federal Communications Commission (FCC) specifies the frequency range for each transmitting station. Hence, the transmitter must translate the information signal to be transmitted into the appropriate frequency range that matches the frequency allocation assigned to the transmitter. Thus, signals transmitted by multiple radio stations do not interfere with one another. Similar functions are performed in telephone communication systems where the electrical speech signals from many users are transmitted over the same wire.

In general, the transmitter performs the matching of the message signal to the channel by a process called modulation. Usually, modulation involves the use of the information signal to systematically vary either the amplitude, frequency, or phase of a sinusoidal carrier. For example, in AM radio broadcast, the information signal that is transmitted is contained in the amplitude variations of the sinusoidal carrier, which is the center frequency in the frequency band allocated to the radio transmitting station. This is an example of amplitude modulation. In FM radio broadcast, the information signal that is transmitted is contained in the frequency variations of the sinusoidal carrier. This is an example of frequency modulation. Phase modulation (PM) is yet a third method for impressing the information signal on a sinusoidal carrier.

In general, carrier modulation such as AM, FM, and PM is performed at the transmitter, as indicated above, to convert the information signal to a form that matches the characteristics of the channel. Thus, through the process of modulation, the information signal is translated in frequency to match the allocation of the channel. The choice of the type of modulation is based on several factors, such as the amount of bandwidth allocated, the types of noise and interference that the signal encounters in transmission. In any case, the modulation process makes it possible to accommodate the transmission of multiple messages from many users over the same physical channel.

In addition to modulation, other functions that are usually performed at the transmitter are filtering of the information bearing signal, amplification of the modulated signal, and in the case of wireless transmission, radiation of the signal by means of a transmitting antenna.

1.2.2 The Channel

The communications channel is the physical medium that is used to send the signal from the transmitter to the receiver. In wireless transmission, the channel is usually the atmosphere (free space). On the other hand, telephone channels usually employ a variety of physical media, including wirelines, optical fiber cables, and wireless (microwave radio). Whatever the physical medium for signal transmission, the essential feature is that the transmitted signal is corrupted in a random manner by a variety of possible mechanism. The most common from of signal degradation comes in the form of additive noise, which is generated at the front end of the receiver, where signal amplification is performed. This noise is often called thermal noise. In wireless transmission, additional additive disturbances are man-made noise, and atmospheric noise picked up by a receiving antenna. Automobile ignition noise is an example of man-made noise, and electrical lightning discharges from thunderstorms is an example of atmospheric noise. Interference from other users of the channel is another form of additive noise that often arises in both wireless and wire line communication systems.

1.2.3 The Receiver

The function of the receiver is to recover the message signal contained in the received signal. If the message signal is transmitted by carrier modulation, the receiver performs carrier demodulation in order to extract the message from the sinusoidal carrier. Since the signal demodulation is performed in the presence of additive noise and possibly other signal distortion, the demodulated message signal is generally degraded to some extent by the presence of these distortions in the received signal. As we shall see, the fidelity of the received message signal is a function of the type of modulation, the strength of the additive noise, the type and strength of any other additive interference, and the type of any non-additive interference.

Besides performing the primary function of signal demodulation, the receiver also performs a number of peripheral functions, including signal filtering and noise suppression.

1.3 Overview of Hardware

This section will describe about TMS320C6x (C6x) family, some applications, DSP processor with real-time signal processing, DSP package, codec AD535, and feature of DSK board.

Digital signal processors such as the TMS320C6x family of processors are like fast special-purpose microprocessors with a specialized type of architecture and an instruction set appropriate for signal processing. The C6x digital signal processor is very well suited for numerically intensive calculations. Based on a VLIW architecture of the C6x is considered to be Texas Instrument (TI)'s most powerful processor.

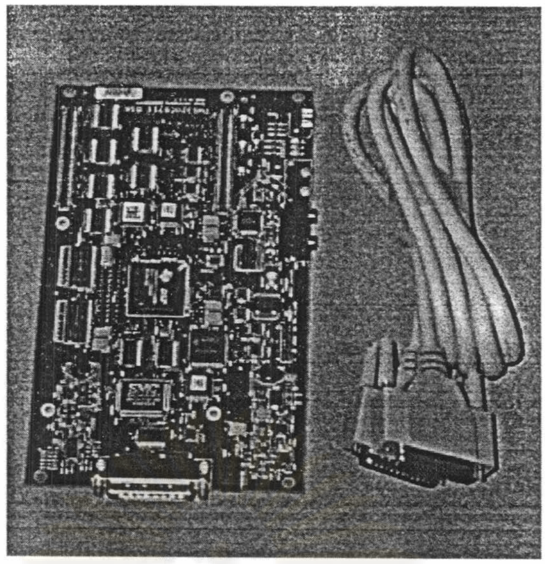
Digital signal processors are used for a wide range of applications, from communications and controls to speech and image processing. The general purpose digital signal processor is dominated by applications in communications (cellular). Applications embedded digital signal processors are dominated by consumer products. They are found in cellular phones, fax/modems, disk drives, radio, printers, hearing aids, MP3 players, high-definition television (HDTV), digital cameras, and so on. These processors have become the products of choice for a number of consumer applications, since they have become very cost-effective. They can handle different tasks, since they can be reprogrammed readily for a different application. DSP techniques have been very successful because of the development of low-cost software and hardware support. For example, modems and speech recognition can be less expensive using DSP techniques.

DSP processors are concerned primarily with real-time signal processing. Real time processing requires the processing to keep pace with some external event, whereas non-real time processing has no such timing constraint. The external event to keep pace with is usually the analog input. Whereas analog based systems with discrete electronic components such as resistors can be more sensitive to temperature changes, DSP based systems are less affected by environmental conditions. DSP processors enjoy the advantages of microprocessors. They are easy to use, flexible, and economical.

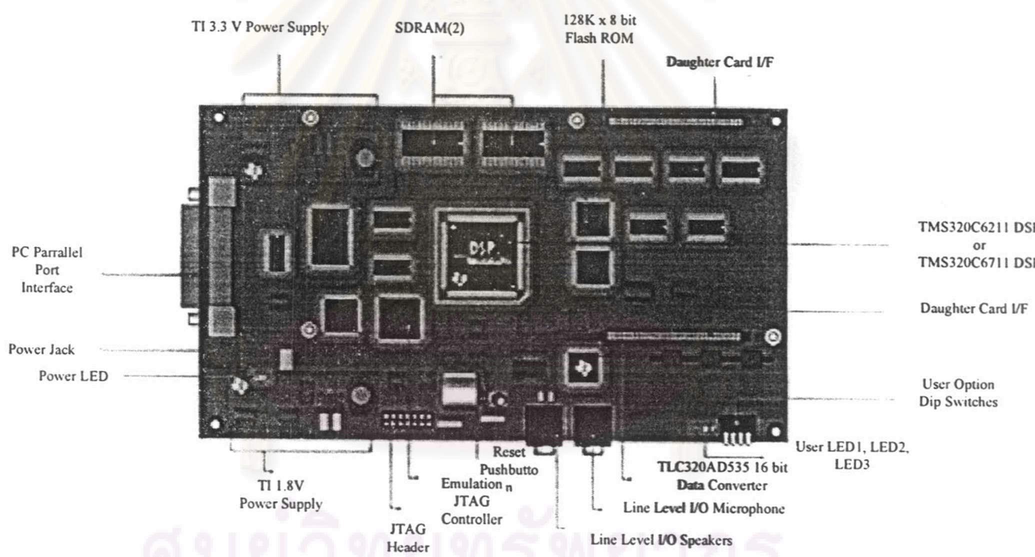
The DSK package, shown in figure 12(a), is powerful, yet relatively inexpensive ($295), with the necessary hardware and software support tools for real time signal processing [9-20]. It is a complete DSP system. The DSK board, with an approximate size of 5x8 in, includes the C6711 floating-point digital signal processor [10] and 16 bit stereo codec AD535 for input and output.

The onboard codec AD535 [21] uses a sigma-delta technology that provides analog to digital conversion (ADC) and digital to analog conversion (DAC). A 4-MHz clock onboard the DSK connects to this codec to provide a fixed sampling rate of 8 kHz.

The DSK board, shown in figure 12(b), includes 16MB (megabytes) of synchronous dynamic random access memory (SDRAM) and 128 kB (kilobytes) of flash ROM. Two connectors on the board provide input and output and are labeled IN (J7) and OUT (J6), respectively. Three of the four user dip switches on the DSK board can be read from a program. The DSK operates at 150 MHz. Also onboard the DSK are voltage regulators that provide 1.8V for the C6711 core and 3.3V for its memory and peripherals.



(a)



(b)

Figure 12(a) TMS320C6711 based DSK board, and **(b)** TMS320C6711 based DSK diagram.

1.4 Objective

The objective of this work is to develop an educational courseware on digital modem used as a teaching tool in communication laboratory primarily for undergraduate students in Chulalongkorn University and National University of Laos, covering basic topics on digital communications. The experiments are intended to provide students with not only essential theory background through MATLAB simulation but also realistic hands-on experiences on hardware development using C6711 DSK.

1.5 Scope

- The simulation and real-time implementation contains each part of digital communication system.
- The course will contain 2 parts to teaching.
    1. Simulation on MATLAB program part
        - Signals and linear system.
        - Baseband digital transmission.
        - Digital transmission through bandlimited channels.
        - Digital transmission via carrier modulation.
    2. Implementation on board DSK part
        - Experiments to Test the DSK Tools
        - BPSK Transmitter and Receiver with Phase Lock Loop (PLL) on Single Board
        - QPSK Modem on C6711 DSKs
- The laboratory package includes complete detail about experiments for helping students on real implementation.

1.6 Procedure

- Install Code composer studio (CCS)
- Study TMS320C6711 documents
- Study how to control components on TMS320C6711 DSK
- Study how to create project with Board Support library (BSL), Chip Support Library (CBL), peripheral devices, which related to my work.
- Using MATLAB for describing the step of implementation on hardware.
- Implement the experiments.

1.7 Expected Benefits

- A project of setting digital communication laboratory for undergraduate students in Chulalongkorn University and National University of Laos.
- Give an understanding on digital communication theory by simulation on MATLAB and using software CCS for implementing digital communication system on the TMS320C6711 DSK.
- Give interaction of study in digital communication theory.
- Give an ability to analyze the problems that occur in practical work and gain understanding about the complexity circuit design on board.
- Give an important skill on hardware controlling in laboratory.
- Especially, educate undergraduate students in Laos who are still lacking of lecturer can teach about real-time laboratory.