Digital modulation and demodulation

Chapter 1 Digital Communications

1. 0 Digital Communication

1. 1 Introduction

Communication Process:

When we think of communication, we usually think of people talking or listening to each other. This may happen face to face, or it may occur through the assistance of a telephone, radio, or television.

Basically, communication is the transfer of information. Life In our modern, complex world depends more and more on the transfer of information. The increasing dependency on the transfer of information has stimulated the growth of more and more communication systems. This surge in communication and communication systems has been referred to as a technological revolution.

This shows understand the transfer of information in a communication system

The communication system will consist of at least the three parts shown. The channel can be as simple as the air that carries the sound of your voice, or as complex as the satellite network required to carry a television program around the world.

The most common problem encountered by the communication process is interference. Interference is any force that disrupts or distorts the information or message while it is being " channeled." It could be noise, as in the case of normal conversation, or atmospheric weather changes, as In the case of radio or television

The biggest cause of interference, however, is a simple misinterpretation of the intended message. Cultural, economic, and political diversities allow people to receive the same message but interpret it differently.

Communication Systems:

Communication system is a combination of processes and hardware used to accomplish the transfer of Information (communication).

A system is a group of interrelated parts. We find that there are systems all around us. In nature, we can also find examples of systems that have been created by people. An automobile, a washing machine, and an electric drill are examples.

1. 2 TYPES OF COMMUNICATION:

Based on the requirements, the communications can be of different types:

Point-to-point communication: In this type, communication takes place between two end points. For instance, in the case of voice communication using telephones, there is one calling party and one called party. Hence the communication is point-to-point.

Point-to-multipoint communication: In this type of communication, there is one sender and multiple recipients. For example, in voice conferencing, one person will be talking but many others can listen. The message from the sender has to be multicast to many others.

Broadcasting: In a broadcasting system, there is a central location from which information is sent to many recipients, as in the case of audio or video broadcasting. In a broadcasting system, the listeners are passive, and there is no reverse communication path.

In simplex communication, the communication is one-way only.

In half-duplex communication, communication is both ways, but only in one direction at a time.

In full-duplex communication, communication is in both directions simultaneously.

Simplex communication: In simplex communication, communication is possible only in one direction. There is one sender and one receiver; the sender and receiver cannot change roles.

Half-duplex communication: Half-duplex communication is possible in both directions between two entities (computers or persons), but one at a time. A walkie-talkie uses this approach. The person who wants to talk presses a talk button on his handset to start talking, and the other person's handset will be in receiving mode.

When the sender finishes, he terminates it with an over message. The other person can press the talk button and start talking. These types of systems require limited channel bandwidth, so they are low cost systems.

Full-duplex communication: In a full-duplex communication system, the two parties-the caller and the called-can communicate simultaneously, as in a telephone system. However, note that the communication system allows simultaneous transmission of data, but when two persons talk simultaneously, there is no effective communication! The ability of the communication system to transport data in both directions defines the system as full-duplex.

1. 3 ANALOG VERSUS DIGITAL TRANSMISSION:

In analog communication, the signal, whose amplitude varies continuously, is transmitted over the medium. Reproducing the analog signal at the receiving end is very difficult due to transmission impairments. Hence, analog communication systems are badly affected by noise.

In a digital communication system, 1s and 0s are transmitted as voltage pulses. So, even if the pulse is distorted due to noise, it is not very difficult to detect the pulses at the receiving end. Hence, digital communication is much more immune to noise as compared to analog communication.

1. 4 Digital Modulation:

Firstly, what do we mean by digital modulation? Typically the objective of a digital communication system is to transport digital data between two or more nodes. In radio communications this is usually achieved by adjusting a physical characteristic of a sinusoidal carrier, the frequency, phase, amplitude or a combination thereof. This is performed in real systems with a modulator at the transmitting end to impose the physical change to the carrier and a demodulator at the receiving end to detect the resultant modulation on reception.

* Modulation is the process of varying some characteristic of a periodic wave with an external signal.

* Modulation is utilized to send an information bearing signal over long distances.

* Radio communication superimposes this information bearing signal onto a carrier signal.

* These high frequency carrier signals can be transmitted over the air easily and are capable of traveling long distances.

* The characteristics (amplitude, frequency, or phase) of the carrier signal are varied in accordance with the information bearing signal.

* In the field of communication engineering, the information bearing signal is also known as the modulating signal.

* The modulating signal is a slowly varying signal – as opposed to the rapidly varying carrier frequency.

The principal of a digital communication system is that during a finite interval of time, it sends a waveform from a finite set of possible waveforms, in contrast to an analog communication system, which sends a waveform from an infinite variety of waveform shapes, with theoretically infinite resolution. In a DCS (digital communication system), the objective of the receiver is not to reproduce a transmitted waveform with precision. The objective is to determine from a noise-perturbed signal which waveform from the finite set of waveforms was sent by the transmitter.

Why Digital?

· The primary advantage is the ease with which digital signals, compared with analog signals, is regenerated. The shape of the waveform is affected by two basic mechanisms.

1. As all transmission lines and circuits have some non-ideal frequency transfer function, there is a distorting effect on the ideal pulse.

2. Unwanted electrical noise or other interference further distorts the pulse waveform.

Both of these mechanisms cause the pulse shape to degrade.

* With digital techniques, extremely low error rates producing high signal fidelity are possible through error detection and correction but similar procedures are not available with analog.

* Digital circuits are more reliable and can be reproduced at a lower cost than analog circuits.

* Digital hardware lends itself to more flexible implementation than analog circuits.

* The combination of digital signals using Time Division Multiplexing (TDM) is simpler than combining analog signals using Frequency Division Multiplexing (FDM).

Metrics for Digital Modulation

- Power Efficiency

– Ability of a modulation technique to preserve the fidelity of the digital message at low power levels

– Designer can increase noise immunity by increasing signal power

– Power efficiency is a measure of how much signal power should be increased to achieve a particular BER for a given modulation scheme

– Signal energy per bit / noise power spectral density: Eb / N0

- Bandwidth Efficiency

– Ability to accommodate data within a limited bandwidth

– Tradeoff between data rate and pulse width

– Throughput data rate per hertz: R/B bps per Hz

- Shannon Limit: Channel capacity / bandwidth

– $C/B = \log_2(1 + S/N)$

Disadvantages of Digital Systems

* Digital systems tend to be very signal processing intensive compared with analog.

* Digital systems need to allocate a significant share of their resources to the task of synchronization at various levels. With analog signals synchronization is accomplished more easily.

* One disadvantage of digital communication system is non-graceful degradation. When the SNR drops below a certain threshold, the quality of service can change form very good to very poor. Most analog systems degrade more gracefully.

Formatting

The goal of the first essential processing step, formatting is to ensure that the source signal is compatible with digital processing. Transmit formatting is a transformation from source information to digital symbols. When data compression in addition to formatting is employed, the process is termed source coding.

The digital messages are considered to be in the logical format of binary 1's and 0's until they are transformed by pulse modulation into base band (pulse) waveforms. Such waveforms are then transmitted over a cable.

No channel can be used for the transmission of binary digits without first transforming the digits to waveforms that are compatible with the channel. For base band channels, compatible waveforms are pulses.

The conversion from a bit of streams to a sequence of pulse waveforms takes place in the block labeled, modulator. The output of a modulator is typically a sequence of pulses with characteristics that correspond to the digits being sent. After transmission through the channel the pulse waveforms are recovered (demodulated) and detected to produce an estimate of the transmitted digits.

Formatting in a digital Communication System

Symbols

When digitally transmitted, the characters are first encoded into a sequence of bits, called a bit stream or base band signal. Group of K bits can then be combined to form new digits, or symbols, from a finite or alphabet of M = 2^K such symbols. A system using a symbol set size of M is referred to as M-array system.

Waveform Representation of Binary Digits

Digits are just abstractions – way to describe the message information. Thus we need something physical that will represent or carry the digits.

Thus binary digits are represented with electrical pulses in order to transmit them through a base band channel. At the receiver, a determination must be made regarding the shape of pulse. The likelihood of correctly detecting the pulse is a function of the received signal energy (or area under the pulse).

PCM Waveform Types

When pulse modulation is applied to a binary symbol, the resulting binary waveform is called a PCM waveform. There are several types of PCM waveforms. These waveforms are often called line codes. When pulse modulation is applied to non-binary symbol, the resulting waveform is called an M-ary pulse modulation waveform.

The PCM waveforms fall into the following four groups.

1) Non return to zero (NRZ)

2) Return to zero (RZ)

3) Phase encoded

) Multilevel binary

The NRZ group is probably the most commonly used PCM waveform.

In choosing a waveform for a particular application, some of the parameters worth examining are

1) DC component

2) Self clocking

3) Error detection

) Bandwidth compression

5) Differential encoding

6) Noise immunity

The most common criteria used for comparing PCM waveforms and for selecting one waveform type from many available are

1) Spectral characteristics

2) Bit synchronization capabilities

3) Error detection capabilities

) Interference

5) Noise immunity

6) Cost and complexity of implementation

Bits per PCM Word and Bits per Symbol

Each analog sample is transformed into a PCM word up to group of bits. The number of quantization levels allowed for each sample can describe the PCM word size; this is identical to the number of values that the PCM word can assume. We use

L= 2^l

Where L is the number of quantization levels in PCM word, l is the number of bits needed to represent those levels.

M-ARY Pulse Modulation Waveforms

There are three basic ways to modulate information onto a sequence of pulses; we can vary the pulse's amplitude, position, or duration. This leads to the names

1) PAM (pulse amplitude modulation)

2) PPM (pulse position modulation)

3) PDM/PWM (pulse duration modulation/ pulse width modulation)

When information samples without any quantization are modulated on to the pulses, the resulting pulse modulation can be called analog pulse modulation. When the information samples are first quantized, yielding symbols from an M-ary alphabet set, and the modulation on to pulses, the resulting pulse modulation is digital and we refer to it as M-ary pulse modulation.

Base-band modulation with pulses has analogous counterparts in the area of band-pass modulation. PAM is similar to amplitude modulation, while PPM and PDM are similar to phase and frequency modulation respectively.

Spectral Density

The spectral density of a signal characterizes the distribution of the signals energy or power in the frequency domain.

Energy Spectral Density

We can relate the energy of a signal expressed in time domain to the energy expressed in frequency domain as:

âˆž

Ex = âˆ« x^2(t) dt

-âˆž

âˆž

= âˆ« | X (f) $|^2$ df

-âˆž

Where X (f) is the Fourier transform of the non periodic signal x (t).

Let Ïˆ (t) = | X (f) $|^2$

âˆž

Ex = 2 âˆ« Ïˆx (f) df

-âˆž

Power Spectral Density

The power spectral density function Gx (t) of the periodic signal x (t) is real, even and nonnegative function of frequency that gives the distribution of the power of x (t) in the frequency domain.

âˆž

Gx (t) = âˆ' | Cn$|^2$ âˆ«(f-nfo)

n =-âˆž

PSD of a periodic signal is a discrete function of frequency.

âˆž

Px = âˆ« Gx (t) df

-âˆž

âˆž

= 2 âˆ« Gx (F) df

0

If x (t) is a non-periodic signal it cannot be expressed by a Fourier series, and if it is a non-periodic power signal (having infinite energy) it may not have a Fourier transform. However we still express the PSD of such signals in a limiting sense.

Chapter 2 Modulation and Demodulation

2. 0 Modulation and Demodulation

Since the early days of electronics, as advances in technology were taking place, the boundaries of both local and global communication began eroding, resulting in a world that is smaller and hence more easily accessible for the sharing of knowledge and information. The pioneering work by Bell and Marconi formed the cornerstone of the information age that exists today and paved the way for the future of telecommunications.

Traditionally, local communication was done over wires, as this presented a cost-effective way of ensuring a reliable transfer of information. For long-distance communications, transmission of information over radio waves was needed. Although this was convenient from a hardware standpoint, radio-waves transmission raised doubts over the corruption of the information and was often dependent on high-power transmitters to overcome weather conditions, large buildings, and interference from other sources of electromagnetic.

The various modulation techniques offered different solutions in terms of cost-effectiveness and quality of received signals but until recently were still largely analog. Frequency modulation and phase modulation presented certain immunity to noise, whereas amplitude modulation was simpler to demodulate. However, more recently with the advent of low-cost microcontrollers and the introduction of domestic mobile telephones and satellite communications, digital modulation has gained in popularity. With digital modulation techniques come all the advantages that traditional microprocessor circuits have over their analog counterparts. Any shortfalls in the communications link can be eradicated using software. Information can now be encrypted, error correction can ensure more confidence in received data, and the use of DSP can reduce the limited bandwidth allocated to each service.

As with traditional analog systems, digital modulation can use amplitude, frequency, or phase modulation with different advantages. As frequency and phase modulation techniques offer more immunity to noise, they are the preferred scheme for the majority of services in use today and will be discussed in detail below

2. 1 Digital Frequency Modulation:

A simple variation from traditional analog frequency modulation can be implemented by applying a digital signal to the modulation input. Thus, the output takes the form of a sine wave at two distinct frequencies. To demodulate this waveform, it is a simple matter of passing the signal through two filters and translating the resultant back into logic levels. Traditionally, this form of modulation has been called frequency-shift keying (FSK).

2. 2 Digital Phase Modulation:

Spectrally, digital phase modulation, or phase-shift keying, is very similar to frequency modulation. It involves changing the phase of the transmitted waveform instead of the frequency, these finite phase changes representing digital data. In its simplest form, a phase-modulated waveform can be generated by using the digital data to switch between two signals of equal frequency but opposing phase. If the resultant waveform is multiplied by a sine wave of equal frequency, two components are generated: one cosine waveform of double the received frequency and one frequency-independent term whose amplitude is proportional to the cosine of the phase shift. Thus, filtering out the higher-frequency term yields the original modulating data prior to transmission.

* Modulate and demodulate/detect blocks together are called a modem.

* The frequency down conversion is performed in the front end of the demodulator.

* Only formatting, modulation, demodulation/detection and synchronization are essential for a digital communication system.

* FORMATTING transforms the source information into bits.

* From this point up to pulse modulation block, the information remains in the form of a bit stream.

* Modulation is the process by which message symbols or channel symbols are converted to waveforms that are compatible with the requirements imposed by transmission channel. Pulse modulation is an essential step because each symbol to be transmitted must first be transformed from a binary representation to a base band waveform.

* When pulse modulation is applied to binary symbols, the resulting binary waveform is called a PCM waveform. When pulse modulation is applied to non-binary symbols, the resulting waveform is called an M-ary pulse modulation waveform.

* Band pass modulation is required whenever the transmission medium will not support the propagation of pulse like waveforms.

* The term band pass is used to indicate that the base band waveform gi (t) is frequency translated by a carrier wave to a frequency that is much larger than the spectral content of gi (t).

* Equalization can be described as a filtering option that is used in or after the demodulator to reserve any degrading effects on the signal that were

caused by the channel. An equalizer is implemented to compensate for any signal distortion caused by a no ideal hi(t)

* Demodulation is defined as a recovery of a waveform (band pass pulse) and detection is defined as decision-making regarding the digital meaning of that waveform.

2. 3 Linear Modulation Techniques

* Digital modulation techniques may be broadly classified as linear and non-linear. In linear modulation techniques, the amplitude to the modulation signal S (t) varies linearly with the modulating digital signal m (t).

* Linear modulation techniques are bandwidth efficient.

* In a linear modulation technique, the transmitted signal S (t) can be expressed as:

S (t) = Re [Am (t) exp (j2pfct)]

= A [mr(t)cos(2pfct) – mI(t)sin(2pfct)]

Where

A is the amplitude

fc is the carrier frequency

m (t) = mr(t) + mI(t) is a complex envelope representation of the modulated signal which is in general complex form.

* From the equations above, it is clear that the amplitude of the carrier varies linearly with the modulating signal.

* Linear modulation schemes, in general do not have a constant envelope.

Linear modulation schemes have very good spectral efficiency.

Normalized Radian Frequency

Sinusoidal waveforms are of the form:

X (t) = Acos (wt+f) ——————————- (1)

If we sample this waveform, we obtain

X[n] = x (nTs)

= Acos (wnTs+f)

= Acos (wn+f) ——————————— (2)

Where we have defined w to be Normalized Radian Frequency:

w= wTs

The Signal in (2) is a discrete time cosine signal, and w is the discrete time radian frequency. w has been normalized by the sampling period. w has the units of radians/second, w= wTs has the units of radians; i. e. wis a dimensionless quantity. This is entirely consistent with the fact that the index n in x[n] is a dimensionless. Once the samples are taken from x (t), the time scale information is lost. The discrete time signal is just a sequence of numbers, and these numbers carry no information about the sampling period, which is the information required to reconstruct the time scale. Thus an infinite number of continuous time sinusoidal signals can be transformed into the same discrete time sinusoid by sampling. All we need to is to change the sampling period with changes in frequency of the continuous time sinusoid.

2. 4 Baseband Transmission

Baseband Demodulation/Detection

· The filtering at the transmitter and the channel typically cause the received pulse sequence to suffer from ISI (Inter Symbol Interference), thus the signal is not quiet ready for sampling and detection.

· The goal of the demodulator is to recover the pulse with best possible signal to noise ratio (SNR), free of any ISI.

· Equalization is a technique used to help accomplish this goal. Every type of communication channel does not require the equalization process. However equalization process embodies a sophisticated set of signal processing techniques, making it possible to compensate for channel induced interference.

· A received band pass waveform is first transformed to a base band waveform before the final detection step takes place.

· For liner systems, the mathematics of detection is unaffected by a shift in frequency.

* According to the equivalence theorem, all linear signal-processing simulations can take place at base band (which is preferred for simplicity)

with the same result as at band pass. Thus the performance of most digital communication systems will often be described and analyzed as if the transmission channel is a base band channel.

Chapter 3 p/4 Quadrature

3. 0 p/4 Quadrature Phase Shift Keying (p/4 QPSK)

3. 1 Linear Modulation Techniques

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Linear modulation schemes have very good spectral efficiency.

There are three major classes of digital modulation techniques used for transmission of digitally represented data:

* Amplitude-shift keying (ASK)

* Frequency-shift keying (FSK)

* Phase-shift keying (PSK)

All convey data by changing some aspect of a base signal, the carrier wave, (usually a sinusoid) in response to a data signal. In the case of PSK, the phase is changed to represent the data signal. There are two fundamental ways of utilizing the phase of a signal in this way:

* By viewing the phase itself as conveying the information, in which case the demodulator must have a reference signal to compare the received signal's phase against; or

* By viewing the change in the phase as conveying information – differential schemes, some of which do not need a reference carrier (to a certain extent).

A convenient way to represent PSK schemes is on a constellation diagram. This shows the points in the Argand plane where, in this context, the real and imaginary axes are termed the in-phase and quadrature axes respectively due to their 90° separation. Such a representation on perpendicular axes lends itself to straightforward implementation. The amplitude of each point along the in-phase axis is used to modulate a cosine (or sine) wave and the amplitude along the quadrature axis to modulate a sine (or cosine) wave.

In PSK, the constellation points chosen are usually positioned with uniform angular spacing around a circle. This gives maximum phase-separation between adjacent points and thus the best immunity to corruption. They are positioned on a circle so that they can all be transmitted with the same energy.

In this way, the moduli of the complex numbers they represent will be the same and thus so will the amplitudes needed for the cosine and sine waves. Two common examples are binary phase-shift keying (BPSK) which uses two phases, and quadrature phase-shift keying (QPSK) which uses four phases, although any number of phases may be used. Since the data to be conveyed are usually binary, the PSK scheme is usually designed with the number of constellation points being a power of 2.

3. 2 Amplitude Shift Keying (ASK)

Amplitude shift keying – ASK – in the context of digital communications is a modulation process, which imparts to a sinusoid two or more discrete amplitude levels. These are related to the number of levels adopted by the digital message.

For a binary message sequence there are two levels, one of which is typically zero.

Thus the modulated waveform consists of bursts of a sinusoid. In Amplitude Shift Keying the Amplitude varies whereas the phase and frequency remains the same as shown in following .

One of the disadvantages of ASK, compared with FSK and PSK, for example, is that it has not got a constant envelope. This makes its processing (eg, power amplification) more difficult, since linearity becomes an important factor. However, it does make for ease of demodulation with an envelope detector.

Thus demodulation is a two-stage process:

å Recovery of the band limited bit stream

å Regeneration of the binary bit stream

3. 3 Frequency-shift keying (FSK)

Frequency-shift keying (FSK) is a method of transmitting digital signals. The two binary states, logic 0 (low) and 1 (high), are each represented by an analog waveform. Logic 0 is represented by a wave at a specific frequency, and logic 1 is represented by a wave at a different frequency. In frequency Shift Keying the frequency varies whereas the phase and amplitude remains the same.

Phase shift keying (PSK)

Phase Shift Keying (PSK) was developed during the early days of the deep-space program. PSK is now widely used in both military and commercial communication systems.

In phase shift Keying the phase of the transmitted signal varies whereas the amplitude and frequency remains the same.

The general expression for the PSK is as

Where,

ji(t) = the phase term will have M discrete values, given by,

ji(t) = 2pi /M

3. 4 Binary PSK

In binary phase shift keying we have two bits represented by the following waveforms;

S0(t) = A cos (wt) represents binary " 0"

S1(t) = A cos (wt + p) represents binary " 1"

For M-array PSK, M different phases are required, and every n (where M= 2n) bits of the binary bit stream are coded as one signal that is transmitted as

A sin (wt + qj)

where j= 1,….., M

3. 5 Quadra phase-Shift Modulation

Taking the above concept of PSK a stage further, it can be assumed that the number of phase shifts is not limited to only two states. The transmitted " carrier" can undergo any number of phase changes and, by multiplying the received signal by a sine wave of equal frequency, will demodulate the phase shifts into frequency-independent voltage levels.

This is indeed the case in quadraphase-shift keying (QPSK). With QPSK, the carrier undergoes four changes in phase (four symbols) and can thus represent 2 binary bits of data per symbol. Although this may seem insignificant initially, a modulation scheme has now been supposed that enables a carrier to transmit 2 bits of information instead of 1, thus effectively doubling the bandwidth of the carrier

Euler's relations state the following:

Now consider multiplying two sine waves together, thus

From Equation 1, it can be seen that multiplying two sine waves together (one sine being the incoming signal, the other being the local oscillator at the receiver mixer) results in an output frequency double that of the input (at half the amplitude) superimposed on a dc offset of half the input amplitude.

Similarly, multiplying by gives which gives an output frequency double that of the input, with no dc offset.

It is now fair to make the assumption that multiplying by any phase-shifted sine wave yields a " demodulated" waveform with an output frequency double that of the input frequency, wh