Module 3

INTRODUCTION

The purpose of a Communication System is to transport an information-bearing signal from a source to a user destination via a communication channel.

Basically, there are two types of communication systems:

1. Analog Communication System
2. Digital Communication System

In an analog communication system, message (information bearing) signal is continuously varying in both amplitude and time. The message signal modulates the amplitude, frequency or phase of a high frequency sinusoidal carrier.

In digital communication system, the informal bearing signal is processed so that it can be represented by a sequence of discrete messages.

1.1 Factors contributed for the growth of digital communication:

1 Impact of ‘computer’ as sources of data and as a tool for communication.
2 Demands for digital services such as telex, facsimile, etc…
3 The use of digital communication offers ‘flexibility’ and ‘compatibility’ in adoption of common digital format for information from many different sources.
4 The improved reliability.
5 The availability of wide-band channels divided by geostationary satellites, optical fibers and co-axial cables.
6 The ever increasing availability of integrated solid state electronic technology which has made it possible to increase system complexity by orders of magnitude in a cost effective manner.

1.2 Basic Signal Processing operations in digital communications

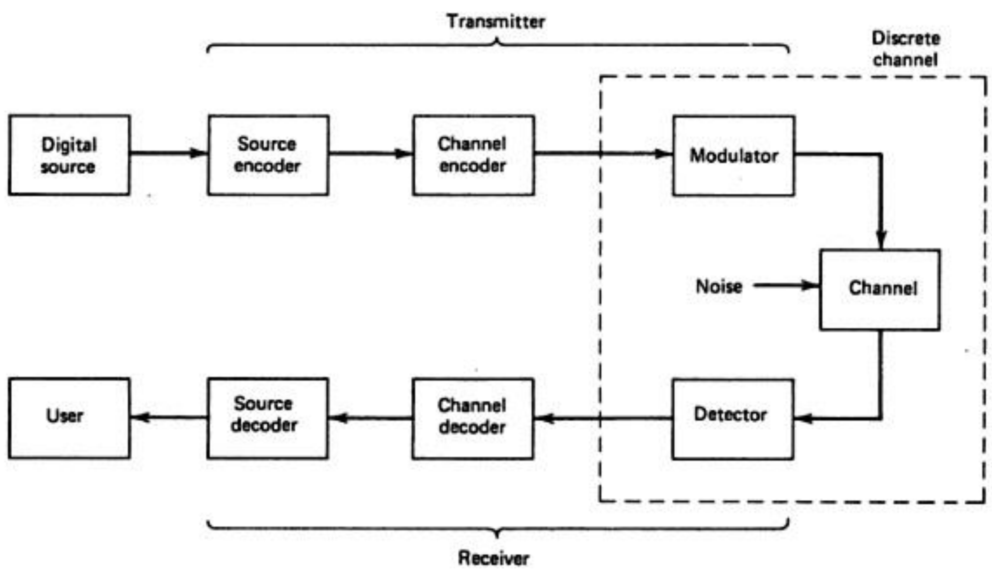


Figure 1.1

Figure 1.1 shows basic elements of a digital communication system. In this diagram three basic signal processing operations are identified:

- Source coding/decoding
- Channel coding/decoding.
- Modulation/demodulation.

The source of information is either analog or digital. If it is analog, it must be converted to digital by a process called 'formatting'.

Formatting:

Formatting is the first essential signal processing step, which makes the source signal compatible with digital processing. Formatting includes sampling, quantization and PCM (Pulse Code Modulation). Format converts analog information like voice, videos, etc… into digital signal.

Source coding:

In source coding the encoder maps the digital signal generated at the source output into another signal in digital form. The mapping is one-to-one. The objective of source coding is to *reduce redundancy in source output* so as to provide an efficient representation of the source output. The primary benefit thus gained from source coding is a reduced bandwidth requirement. The decoder performs the inverse mapping.

Channel coding:

The objective of channel encoder is to map the incoming digital signal into a channel input and for the decoder to map channel output into an output digital signal in such a way that the effect of channel noise is minimized. Channel coding provides reliable communication over a noisy channel. This is done by introducing 'controlled redundancy' in a prescribed fashion in channel encoder and exploiting it in decoder to decode the channel encoded signal.

Modulation:

The purpose of modulation is to provide for the 'efficient transmission' of the signal over the channel.

The modulator operates by keying shifts in the amplitude, frequency or phase of a sinusoidal carrier wave to the channel encoder output. The detector performs demodulation digitally modulated signals.

The combination of modulator, channel & detector is called a 'discrete channel'.

There are many other signal processing operations in digital communication like

- Encryption/decryption.
- Spreading/de-spreading.
- Multiplexing/de-multiplexing.
- Multiple access.

But, these are not mandatory operations.

SAMPLING PROCESS

The sampling process is the first process that is performed in analog to digital conversion.

In the sampling process, a continuous-time signal is converted into a discrete-time signal by measuring the signal at periodic instants of time. For the sampling process to be of practical utility, it is necessary that we choose the sampling properly, so that the discrete time signal resulting from process uniquely defines the original continuous time signal.

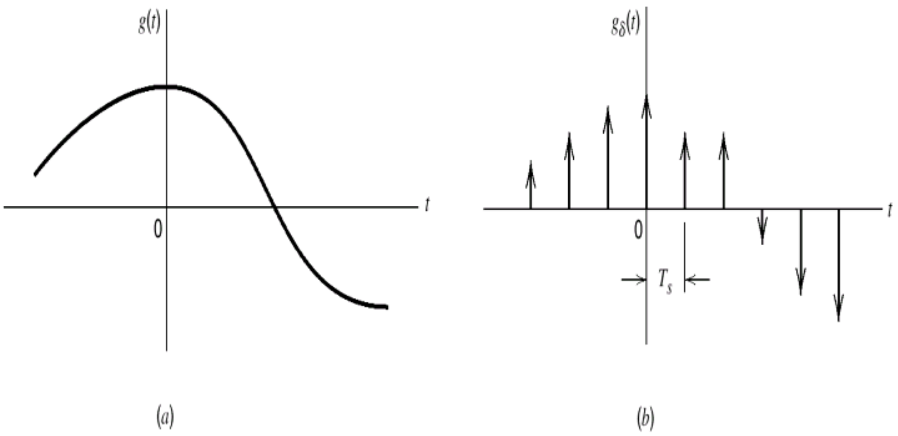
Sampling Theoram:

Statement:

1. *If a finite energy signal, g(t) contains no frequencies higher than 'W' Hz, it is completely determined by specifying its ordinates at a sequence of points spaced 1/2W seconds apart.*
2. *If a finite energy signal g(t) contains no frequencies higher than 'W' Hz, it may be completely recovered from its ordinates at a sequence of points spaced 1/2W seconds apart.*

Ideal Sampling (Instantaneous sampling or Impulse sampling)

Consider an analog signal g(t) that is continuous in both time and amplitude. Let g(t) be an infinite duration, finite energy signal, band-limited to 'W' Hz.



Ideal sampling is produced by multiplying g(t) by an ideal sampling function, $\partial_{TS}(t)$ which is also called 'Dirac Comb'. It is a Dirac delta function ie...

$$\delta_{TS}(t)= \sum_{n=-\infty}^{\infty} \delta(t-nT_S)$$

Taking FT of above function,

$$\delta_{FS}(f)= f_S \sum_{m=-\infty}^{\infty} \delta(f-mf_S) \quad \text{where } f_S=1/T_S$$

f_S = Sampling Rate (samples/sec) & T_S = Sampling Period(sec)

The circuit theoretic interpretation of the ideal sampling process is as follows:

Time domain interpretation:

$$g_\delta (t) = g(t)\, \delta_{TS}(t)$$

$$= g(t) \sum_{n=-\infty}^{\infty} \delta(t-nT_S)$$

$$g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_S)\ \delta(t-nT_S)$$

In $g_\delta(t)$, each delta function is 'weighted' by the corresponding sample value of the input signal g(t).

From the definition of delta function, we have,

$$g(nT_S)\ \delta(t-nT_S) = g(t)\ \delta(t-nT_S)$$

$$\therefore\ g_\delta(t) = \sum_{n=-\infty}^{\infty} g(t)\ \delta(t-nT_S)$$

Frequency domain interpretation:

Let $g(t) \Leftrightarrow G(f)$ & $g_\delta(t) \Leftrightarrow G_\delta(f)$

Multiplication in time domain is equivalent to convolution in frequency domain.

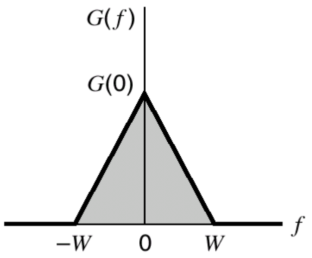
$\therefore\ G_\delta(f) = G(f) * \delta_{fs}(f)$ from equation 3

$$G_\delta(f) = G(f) * f_s \sum_{m=-\infty}^{\infty} \delta(f-mf_s)$$

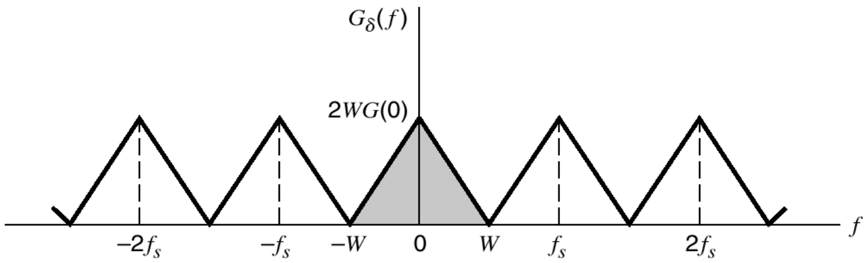
$$= f_s \sum_{m=-\infty}^{\infty} G(f) * \delta(f-mf_s)$$

$$G_\delta(f) = f_s \sum_{m=-\infty}^{\infty} G(f-mf_s)$$

$G_\delta(f)$ represents spectrum that is periodic in frequency with period f_s, but not necessarily continuous. In other words, "the process of uniformly sampling a signal in the time domain results in a periodic spectrum in the frequency domain with a period equal to the sampling rate" hence, $G_\delta(f)$ represents a periodic extension of the original spectrum, G(f).



(a)



(b)

Another useful expression for $G_\delta(f)$ is obtained as follows:
From 4

$$g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(t-nt_s)$$

By taking FT of above expression,

$$g_\delta(f) = \sum_{n=-\infty}^{\infty} g(nT_s)\, e^{-j2\Pi nfTs} \qquad [\delta(t-t_0) \xleftrightarrow{FT} e^{-j2\Pi fT0}]$$

Suppose that we choose sampling period, $T_S = \frac{1}{fs} = \frac{1}{2W}$

Then,

$$g_\delta(f) = \sum_{n=-\infty}^{\infty} g(\frac{n}{2W})\, e^{-j(\frac{2\pi nf}{2W})}$$

The above relation may also be viewed as a complex Fourier series representation of the periodic spectrum $g_\delta(f)$.

Therefore, if the sample values $g(\frac{n}{2W})$ of the signal g(t) are specified for all time, then Fourier transform, g(f) is uniquely determined by using the Fourier series of equation 9. Because g(t) is related to g(f) by IFT, it follows that the signal g(t) is itself uniquely determined by sample values $g(\frac{n}{2W})$ for $-\infty \le n \le \infty$. In other words, the sequence $\{g(\frac{n}{2W})\}$ contains all of the information of g(t).