(EC-603) Digital Communication System

Concept and Elements of Digital Commⁿ :

In Analog Commⁿ system, the meaningful or information bearing ~~system~~ signal is continuously varying with respect to time, either ~~both~~ in amplitude or phase and it is used directly to modify some characteristics of a sinusoidal carrier wave, such as amplitude, phase or frequency.

But, in Digital commⁿ system, the information bearing signal is processed so that it can be represented by a sequnce of discrete messages.

Basic Block Diagram :

Analog Signal → Sampling → Quantizing → Encoding → Digital Signal

ADC

Source of signal (message) → Analog
→ Digital

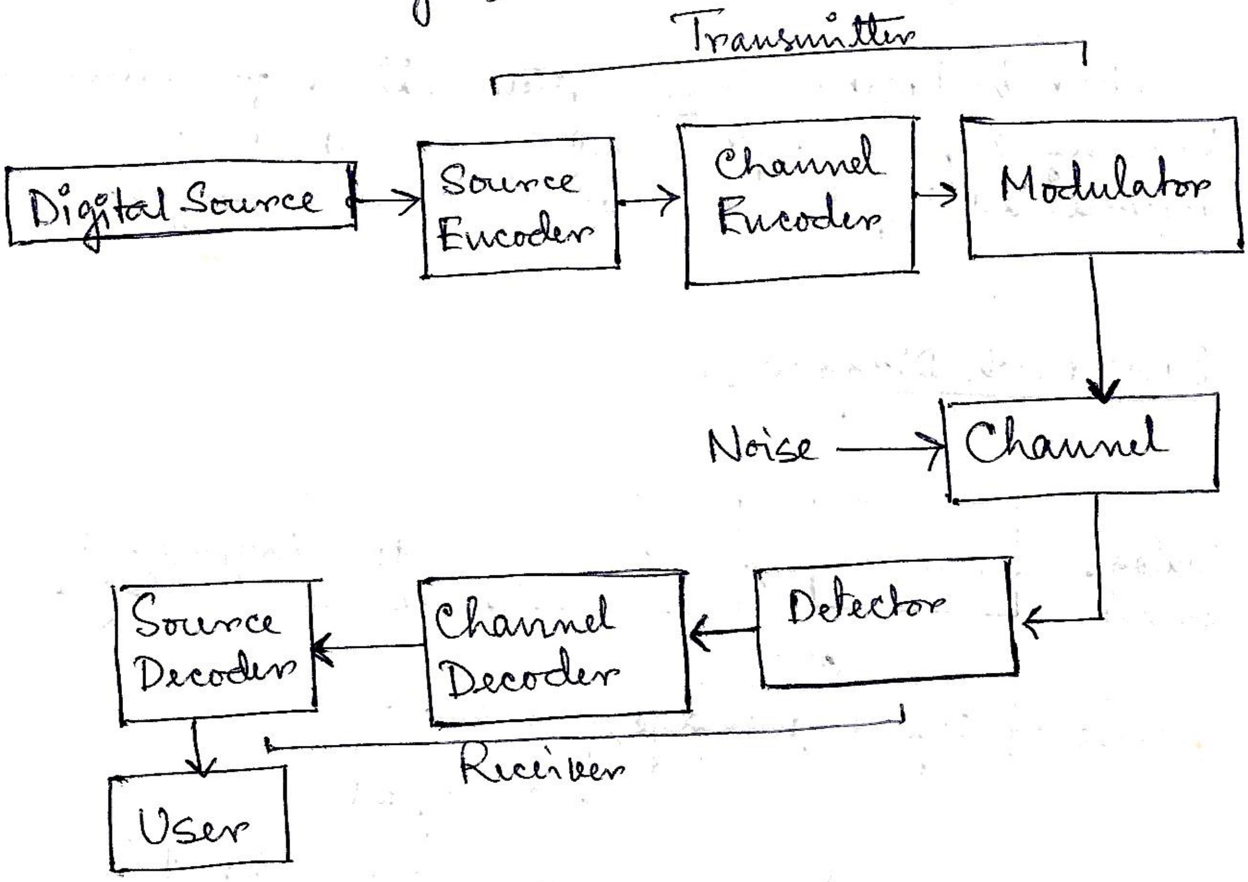
Analog → voice signal, o/p of analog ckts, television signals etc.

Digital → Data from Computers, o/p of ADC ckts, telegraphic signals etc.

Sampling : The process of retaining Sample values of the analog signal at uniformly spaced discrete instants of time.

Quantization : Approximation of sampled values to nearest level in a finite set discrete levels.

Encoding : In this operation the selected level is represented by a code word which consists of prescribed no. of code elements. (combination of binary digits)



Block Diagram of Digital Comm.

Source encoding / decoding :

It maps the digital signal generated at the source output into another signal one-to-one, which eliminate or reduce redundancy, as well as bandwidth requirement will also be less.

The decoder performs the reverse job.

<u>Channel Coding</u> : Here the purpose is to map the incoming digital signal in such a way that the effect of channel noise is minimized.

Hence, combination of source & channel encoder ensures a reliable communication over a noisy channel.

<u>Modulator</u> : It is performed for the purpose of providing efficient transmission of signal over the channel. For digital modulation the techniques are generally ASK, FSK, or PSK.

<u>Channels for Digital Comm</u> :

Selection of channel basically depends on two factors : (1) Bandwidth, & (2) Power.

Other is <u>linear</u> or <u>non-linear</u>.

<u>Example</u> : 1. Telephone Channel
2. Co-axial Cable.
3. Optical Fibre
4. Microwave Radio
5. Satellite channel.

(4)

Advantages of Digital Communication :

1. Digital comm. can withstand more channel noise and channel distortion is comparatively less compared to analog comm.

2. The greatest advantage of digital comm. is the viability of regenerative repeaters. For analog comm. as signal travels along a channel it becomes more and more weaker and noise goes on increasing, which is a cumulative process. Ultimately, after a very long distance the signal is multilated, as amplification increases both signal and noise power.
   But, for digital comm. repeater station is set up at such a short distance, hence signal is detected and new clean pulses are sent.

3. Digital hardware implementation is flexible and permits use of μP, micro-controller, digital switching etc.

4. Digital signals are coded to give extremely low error rate.

5. It is easier and more efficient to multiplex several digital signals.

6. Improved SNR and better bandwidth.

7. Digital signal storage is relatively easy and easy to access data from remote places also.

8. Reproduction of digital data is easy to ~~and reproduce~~ and also does not fade with time.

Sampling Theorem :

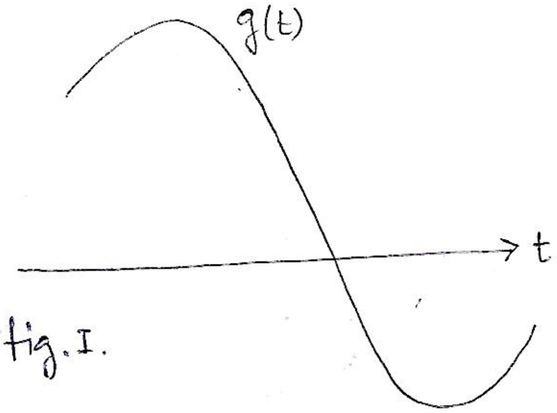


fig. I.

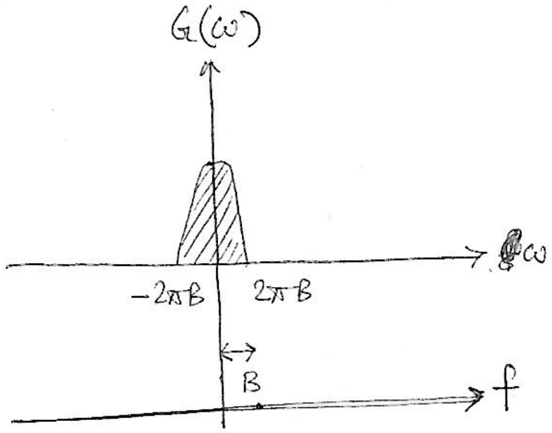


fig. II

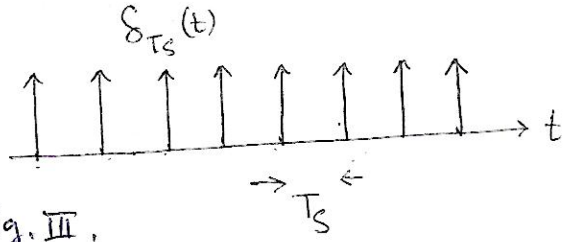


fig. III.

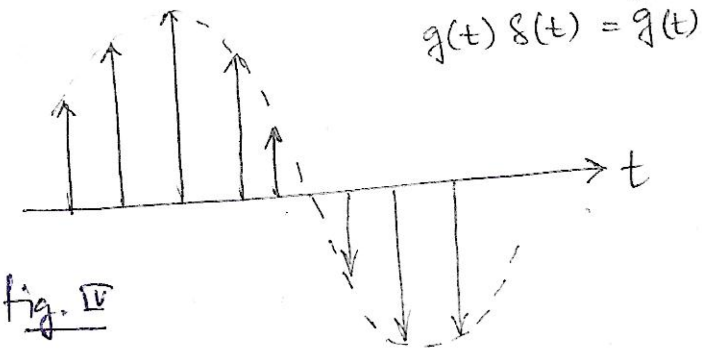


fig. IV

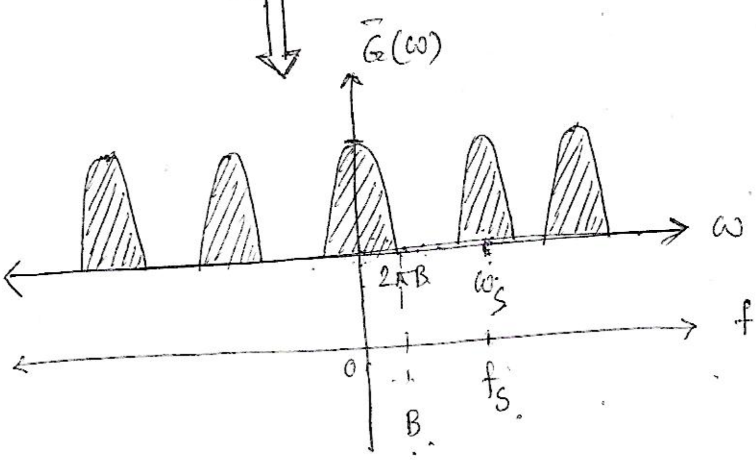


fig. V

Let us now consider a signal whose spectrum is band-limited to B Hz can be reconstructed exactly without any error from its samples taken uniformly at a rate $R \geq 2B$ Hz samples per second. That means, sampling freq. $f_s = 2B$ Hz.

Fig. I shows a time-varying signal and fig. II represents its fourier transform, which is band-limited to B Hz. Sampling of $g(t)$ at a rate of f_s Hz can be accomplished by multiplying $g(t)$ by an impulse train $\delta_{T_s}(t)$ as shown in fig. This results in sampled signal $\bar{g}(t)$ as shown in fig IV.

The sampled signal consists of impulses spaced every T_s. Therefore, in general, the nth impulse located at $t = nT_s$, has a value equal to $g(nT_s)$.

Hence, $\bar{g}(t) = g(t)\,\delta_{T_s}(t)$

$$= \sum_n g(nT_s)\,\delta(t - nT_s) \quad \text{-----} (1)$$

As the impulse train $\delta_{T_s}(t)$ is a periodic signal of period T_s, it can be expressed as Fourier Series.

The trigonometric fourier series, is.

$$\delta_{T_s}(t) = \frac{1}{T_s}\left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t + \cdots\right], \quad \omega_s = \frac{2\pi}{T_s} = 2\pi f_s$$

$$\text{----} (2)$$

So, $\bar{g}(t) = g(t)\,\delta_{T_s}(t)$

$$= \frac{1}{T_s}\left[g(t) + 2g(t)\cos\omega_s t + 2g(t)\cos 2\omega_s t + \cdots\right]$$

$$\text{-----} (3)$$

Now, fourier transform of $\bar{g}(t)$ is $\bar{G}(\omega)$. For convenience, we will find it term by term;

$$g(t) \longleftrightarrow G(\omega)$$

$$\therefore \quad 2g(t)\cos\omega_s t \longleftrightarrow G(\omega-\omega_s)+G(\omega+\omega_s)$$

$$\therefore \quad 2g(t)\cos 2\omega_s t \longleftrightarrow G(\omega-2\omega_s)+G(\omega+2\omega_s)$$

$$\vdots$$

So, we can see ~~that~~ the sepictrum $G(\omega)$ is shifted by ω_s, $2\omega_s$, $3\omega_s$, ... ~~to~~ ~~ma~~ etc.

That means the spectrum $\bar{G}(\omega)$ consists of $G(\omega)$ repeating periodically with period, $\omega_s = \frac{2\pi}{T_s}$ or $f_s = \frac{1}{T_s}$ Hz.

Hence, we can write,

$$\bar{G}(\omega) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} G(\omega - n\omega_s)$$

If we are to reconstruct $g(t)$ from $\bar{g}(t)$, we should be able to recover $G(\omega)$ from $\bar{G}(\omega)$.

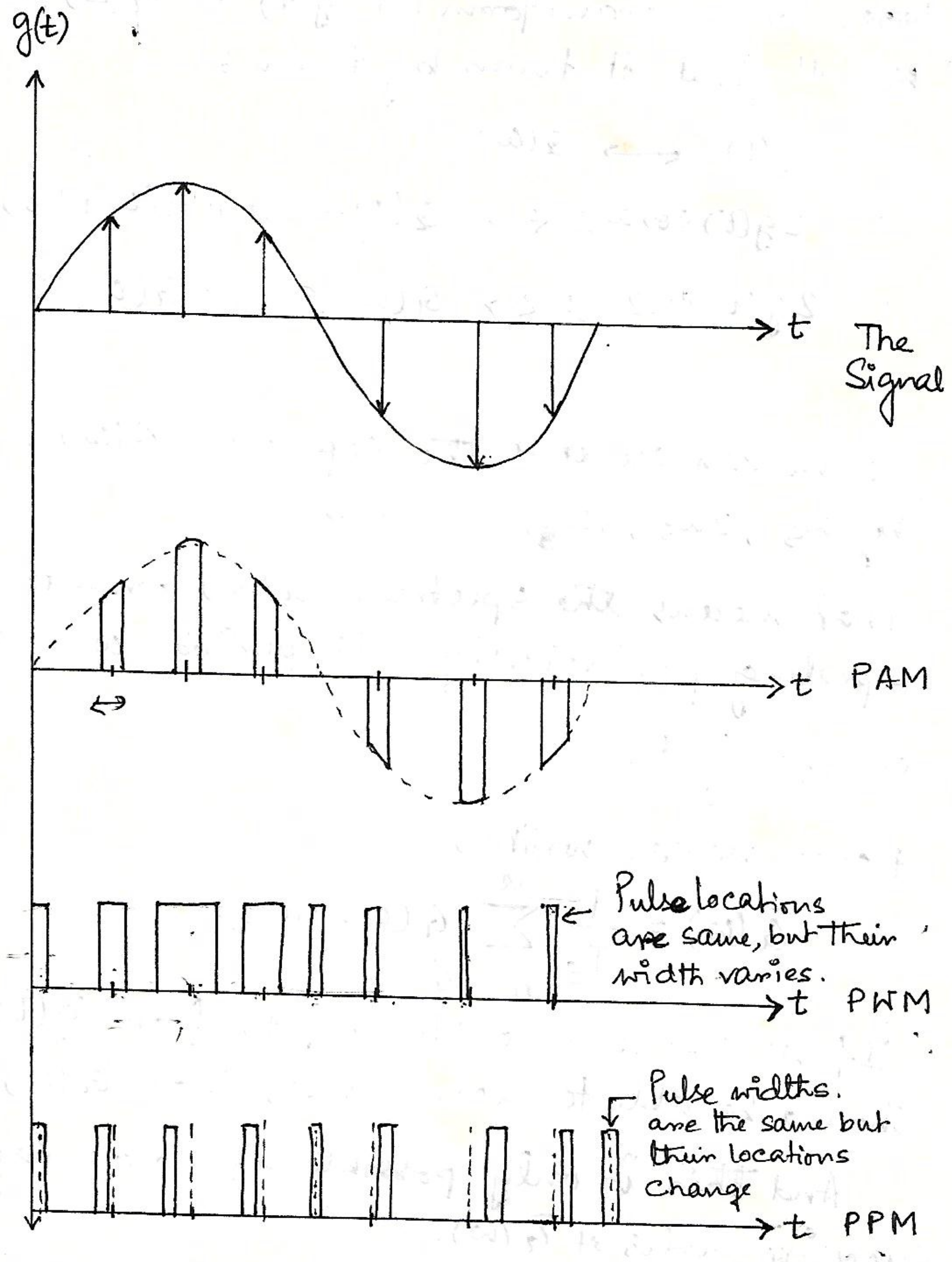
And this is only possible if no overlap between successive cycles of $\bar{G}(\omega)$.

The fig. V. shows that this requires,

$$f_s > 2B \quad \text{or} \quad T_s < \frac{1}{2B}$$

So, as long as the sampling freq. f_s is greater than twice the signal bandwidth B, $\bar{G}(\omega)$ will consists of non-overlapping repeatations of $G(\omega)$. When this is true $g(t)$ can be recovered from samples $\bar{g}(t)$ efficiently and without any error.

The min. sampling rate is $f_s = 2B$ Hz, which is known as <u>Nyquist Rate</u>, and corresponding [illegible] interval $T_s = \frac{1}{[illegible]}$ is called as Nyquist Interval



Some Applications of Sampling Theorem.