... be derived in an identical way. The phasor diagrams are also identical.

3. If you compare this circuit with the Foster Seeley discriminator discussed you will notice that these two circuits are identical except for the following changes :
   1. The direction of diode D_2 is reversed.
   2. A large value capacitor C_5 has been included in the circuit.
   3. The output is taken somewhere else.

Operation and phasor diagrams :

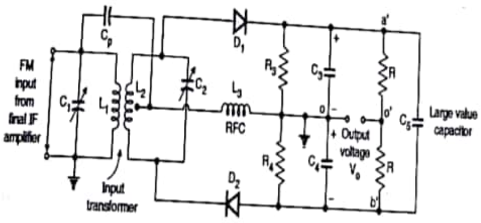


Fig. 5.23 : Ratio detector circuit

It can be shown that the ratio detector output voltage is equal to half of the difference between the output voltages from the individual diodes.

$$\therefore \quad V_o = \frac{V'_{ao} - V'_{bo}}{2}$$

The additional feature of the ratio detector is the amplitude limiting action which is incorporated due to the large capacitor C_5. Due to this the amplitude limiter is not required prior to the ratio detector.

Q. 13 Explain Foster Seeley discriminator with neat block diagram and compare the performance with ratio detector. Dec. 15, Dec. 16

Ans. :

Sr. No.	Parameter	Phase discriminator	Ratio detector
1.	Alignment/tuning	Not critical	Not critical
2.	Output characteristics depends on	Primary and secondary phase relation	Primary and secondary phase relation.
3.	Linearity of output characteristics	Very good	Good
4.	Amplitude limiting	Not provided inherently	Provided by the ratio detector
5.	Applications	FM radio, satellite station receiver etc.	TV receiver sound section, narrow band FM receivers.

Chapter 6 : Pulse Analog Modulatio

Q. 1 Write a short note on : Sampling Theorem.

Dec. 03, May 04, May 06, Dec. 07, May 09, May 10, Dec. 10, May 11, Dec. 11, May 12, Dec. 12, May 14, Dec. 15, May 16, Dec. 16

Ans. : In order to represent the original message signal "faithfully" (without loss of information), it is necessary to take as many samples of the original signal as possible. Higher the number of samples, closer is the representation.

The number of samples depends on the "sampling rate" and the maximum frequency of the signal to be sampled. Sampling theorem was introduced to the communication theory in 1949 by Shannon. Therefore this theorem is also called as **"Shannon's sampling theorem".**

The statement of sampling theorem in time domain, for the bandlimited signals of finite energy is as follows :

Statement :

1. If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2 W) seconds apart.
2. If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (1/2 W) seconds apart.

Combined statement of sampling theorem : A continuous time signal x (t) can be completely represented in its sampled form and recovered back from the sampled form if the sampling frequency $f_s \geq 2$ W where "W" is the maximum frequency of the continuous time signal x (t).

Q. 2 State and prove sampling theorem.

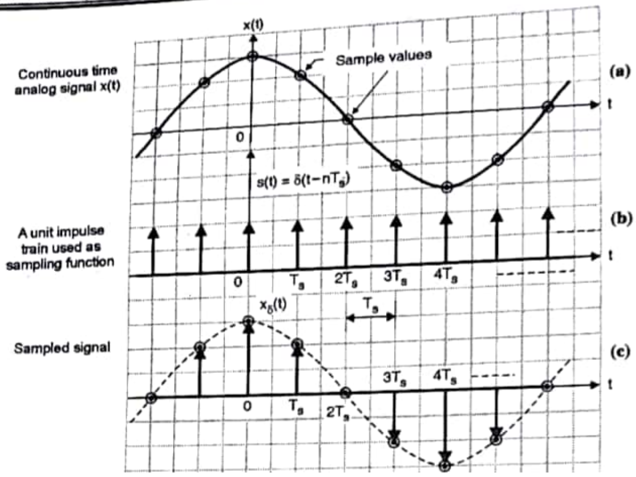
May 10, Dec. 10, Dec. 13, Dec. 15, May 16, Dec. 16

Ans. :

The assumptions made for this proof are as follows :

Assumptions : Let x (t) be a continuous time analog signal as shown in Fig. 6.1.

Let x (t) be a signal with finite energy and infinite duration. Let x (t) be a strictly bandlimited signal. That means it does not contain any frequency components above "W" Hz.



than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2 W) seconds apart.

Let x (t) be a strictly bandlimited signal. That means it does not contain any frequency components above "W" Hz.

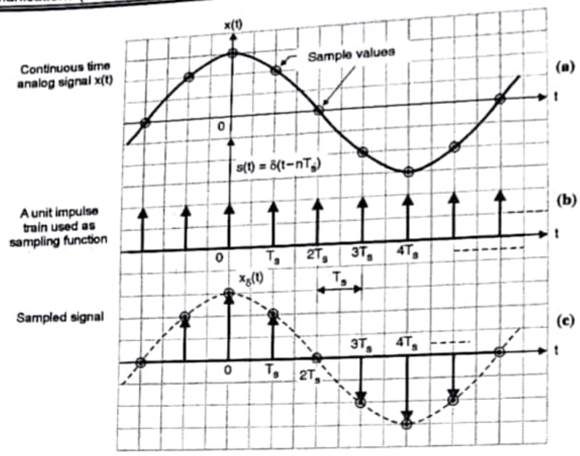


Fig. 6.1 : Sampling of a continuous time signal x (t)

Let s (t) be the sampling function as shown in Fig. 6.1. It is a train of unit impulses, spaced by a period of T_s seconds. This sampling function samples the original signal at a rate of "f_s" samples per second. Therefore "T_s" represents the sampling period such that,

$$T_s = \frac{1}{f_s} = \text{Sampling period} \quad ...(1)$$

$$\text{and} \quad f_s = \frac{1}{T_s} = \text{Sampling rate.}$$

Procedure to be followed :

We are going to follow the steps given below to prove the sampling theorem:

Step 1	:	Represent the sampling function s (t) mathematically.
Step 2	:	Represent the sampled signal x_δ (t) mathematically.
Step 3	:	Obtain the Fourier transform of the sampled signal.
Step 4	:	Prove that the sampled signal x_δ (t) completely represents x (t).
Step 5	:	Represent x (t) as summation of sinc functions (interpolation).
Step 6	:	Graphical representation of the interpolation process.
Step 7	:	Actual recovery of x (t) using an ideal low pass filter.

Part 1 : Sampling theorem :

Spectrum of the sampled signal

Step 1 : Represent the sampling function s (t) mathematically

Fig. 6.1 shows the sampling function s (t) which is a train of unit impulses. The spacing between the adjacent unit impulses is T_s seconds, therefore the frequency of the sampling function is equal to the sampling frequency f_s.

The sampled signal is denoted by x_δ (t) and it is as shown in Fig. 6.1. The sample function s (t) can be represented mathematically as follows :

$$s(t) = \text{.........}\delta(t+2T_s) + \delta(t+T_s) + \delta(t) + \delta(t-T_s) + \delta(t-2T_s) + \text{.....}$$

$$\therefore s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad ...(2)$$

Step 2 : Represent the sampled signal x_δ (t) mathematically :

Fig. 6.1 shows the sampled signal x_δ (t) graphically. It is present only at the sampling instants i.e. T_s, 2 T_s etc. and its instantaneous amplitude is equal to the amplitude of original signal x (t) at the sampling instants.

This is shown by the encircled points in Fig. 6.1. Let us represent the instantaneous amplitude of x (t) at the various sampling points t = n T_s as x (n T_s). This is the amplitude of the encircled points of Fig. 6.1.

Looking at the sampled signal x_δ (t) we can say that the sampled signal is obtained by multiplying x (t) and s (t).

$$\therefore x_\delta(t) = x(t) \times s(t) = x(nT_s) \times s(t) \quad ...(3)$$

Substituting the expression for s (t) from Equation (2) we get the mathematical expression for the sampled signal x_δ (t) as,

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s) \quad ...(4)$$

Step 3 : Obtain the Fourier transform of the sampled signal :

The fourier transform of a train of impulses (dirac delta function) is given by,

$$X(f) = f_0 \sum_{n=-\infty}^{\infty} \delta(f - nf_0)$$

Spectrum of the sampled signal

Step 1 : Represent the sampling function s (t) mathematically

Fig. 6.1 shows the sampling function s (t) which is a train of unit impulses. The spacing between the adjacent unit impulses is T_s seconds, therefore the frequency of the sampling function is equal to the sampling frequency f_s.

Step 3 : Obtain the Fourier transform of the sampled signal :

The fourier transform of a train of impulses (dirac delta function) is given by,

$$X(f) = f_0 \sum_{n=-\infty}^{\infty} \delta(f - nf_0)$$

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Here we have the similar pulse train as sampling function s (t). Therefore the Fourier transform of the sampling function is given by,

$$S(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \quad ...(5)$$

Note that f_0 has been replaced by f_s in the above equation.

The sampled signal in the time domain is represented as product of x (t) and s (t).

i.e. $x_\delta(t) = x(t) \times s(t) \quad ...(6)$

Taking the Fourier transform of both the sides we get,

i.e. $X_\delta(f) = X(f) * S(f) \quad ...(7)$

This is because the Fourier transform of the product of two signals in the time domain is the convolution of their Fourier transforms. Substituting the value of S(f) from Equation (5) we get,

$$X_\delta(f) = X(f) * \left[f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \right] \quad ...(8)$$

where * denotes convolution. Interchanging the orders of convolution and summation results in :

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f) * \delta(f - nf_s) \quad ...(9)$$

From the properties of delta function, we find that the convolution of X (f) and $\delta(f - nf_s)$ is equal to $X(f - nf_s)$. Hence the above equation can be simplified as follows :

F.T. of the sampled signal,

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \quad ...(10)$$

where X (f) = Fourier transform of the original signal x (t).

4. **Prove that sampled signal $x_\delta(t)$ completely represents x (t)**

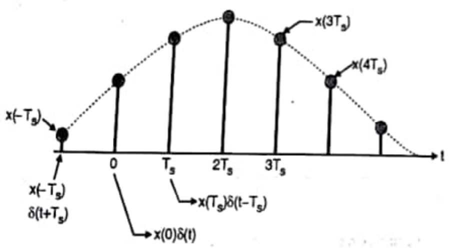

Fig. 6.2 : Sampled signal $x_\delta(t)$

$x_\delta(t)$ can be represented in the summation form as follows (Refer Fig. 6.2).

$$x_\delta(t) = ... x(-T_s)\,\delta(t + T_s) + x(0)\,\delta(t) + x(T_s)\,\delta(t - T_s) + ... \quad ...(11)$$

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s)$$

We can obtain another useful expression for the fourier transform $X_\delta(f)$ by taking the fourier transform of both the sides of the equation stated above as,

$$X_\delta(f) = \sum_{n=-\infty}^{\infty} x(nT_s)\, e^{-j2\pi n f T_s} \quad ...(12)$$

This equation is the fourier transform of a discrete time signal $x_\delta(t)$. Therefore it is called as the Discrete Fourier Transform (DFT). Compare it with the definition of fourier transform of a continuous time signal. i.e.

$$X(f) = \int_{-\infty}^{\infty} x(t)\, e^{-j2\pi f t}\, dt$$

As the signal is discrete, the integration sign has been replaced by the summation sign and "t" has been replaced by "nT_s".

Now consider

$$X_\delta(f) = f_s X(f) + \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} f_s X(f - nf_s)$$

$$\therefore \quad X(f) = \frac{1}{f_s} X_\delta(f) - \sum X(f - nf_s)$$

But in the range $-W \le f \le W$ the second term of the above expression will not be present

$$\therefore \quad X(f) = \frac{1}{f_s} X_\delta(f) \quad ...(13)$$

Substitute $f_s = 2W$ and $X_\delta(f)$ from Equation (12) to get,

$$X(f) = \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(nT_s) \cdot e^{-j2\pi n f T_s} \quad ...(14)$$

This is the frequency spectrum of x (t) in terms of $x(nT_s)$ i.e. the sampled signal.

Substitute $T_s = 1/2W$ to get,

$$X(f) = \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(n/2W) \cdot e^{-j2\pi n f/2W} \; .... -W \le f \le W \quad ...(15)$$

Part 2 of the sampling theorem :

5. **Reconstruction of signal from samples**

This is the second part of the sampling theorem. From Equation (15) we can obtain x (t) by taking the Inverse Fourier Transform (IFT).

$$x(t) = \text{IFT}\,\{X(f)\}$$

$$= \text{IFT}\left\{ \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(n/2W).\, e^{-j\pi f n/W} \right\}$$

Using the definition of inverse Fourier transform,

$$x(t) = \int_{-W}^{W} \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(n/2W) \cdot e^{-j\pi f n/W} e^{j2\pi f t}\, df$$

Interchanging the order of summation and integration we get,

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W) \frac{1}{2W} \int_{-W}^{W} e^{j2\pi f\left(t - \frac{n}{2W}\right)}\, df$$

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Fig. 6.2 : Sampled signal $x_\delta(t)$

$x_\delta(t)$ can be represented in the summation form as follows (Refer Fig. 6.2).

$$x_\delta(t) = \ldots x(-T_s)\,\delta(t+T_s) + x(0)\,\delta(t) + x(T_s)\,\delta(t-T_s) + \ldots \quad \ldots(11)$$

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\cdot\delta(t-nT_s)$$

$$= \mathrm{IFT}\left\{\frac{1}{2W}\sum_{n=-\infty}^{\infty} x(n/2W).\,e^{-j\pi n f/W}\right\}$$

Using the definition of inverse Fourier transform,

$$x(t) = \int_{-W}^{W}\frac{1}{2W}\sum_{n=-\infty}^{\infty} x(n/2W)\cdot e^{-j\pi f n/W}e^{j2\pi ft}\,df$$

Interchanging the order of summation and integration we get,

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W)\frac{1}{2W}\int_{-W}^{W} e^{j2\pi f\left(t-\frac{n}{2W}\right)}\,df$$

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W)\cdot\frac{1}{2W}\times\frac{1}{j2\pi\left[t-\frac{n}{2W}\right]}\cdot\left[e^{j2\pi f(t-n/2W)}\right]_{-W}^{W}$$

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W)\frac{1}{j4\pi W\left[t-\frac{n}{2W}\right]}\cdot\left[e^{j2\pi W(t-n/2W)}-e^{-j2\pi W(t-n/2W)}\right]$$

$$= \sum_{n=-\infty}^{\infty} x(n/2W)\cdot\left[\frac{e^{j2\pi W(t-n/2W)}-e^{-j2\pi W(t-n/2W)}}{j4\pi W\left[t-\frac{n}{2W}\right]}\right]$$

The term inside the square bracket is a "sine" function.

$$\therefore\; x(t) = \sum_{n=-\infty}^{\infty} x(n/2W)\frac{\sin(2\pi Wt-n\pi)}{(2\pi Wt-n\pi)} \quad \ldots(16)$$

We can simplify the equation above by using the definition of the "sinc function". The sinc function is defined as :

$$\mathrm{sinc}\,x = \frac{\sin(\pi x)}{\pi x} \quad \ldots(17)$$

Therefore Equation (16) can be written as :

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W)\,\mathrm{sinc}(2Wt-n) \quad \ldots(18)$$

Equation (18) provides an interpolation formula for reconstructing the original signal x (t) from the sequence of sample values {x (n /2 W)}. The "sinc" function plays the role of an interpolation function. Each sample x (n / 2 W) is multiplied by a delayed version of the interpolation function i.e. sinc function. Then all these resulting waveforms are added to obtain x (t).

6. Graphical representation of the interpolation process

Let us re-arrange Equation (18) as follows :

$$x(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\mathrm{sinc}\,2W\left(t-\frac{n}{2W}\right)$$

This is because $\frac{1}{2W} = T_s$

$$\therefore\; x(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\mathrm{sinc}\,2W(t-nT_s) \quad \ldots(19)$$

Let us expand this equation to write,

$$x(t) = x(0)\,\mathrm{sinc}\,2Wt + x(\pm T_s)\,\mathrm{sinc}\,2W(t\pm T_s) + x(\pm 2T_s)\,\mathrm{sinc}\,2W(t\pm 2T_s) + \ldots \quad \ldots(20)$$

(a) First term : $x(0)$ sinc $2Wt$

This will have a maximum amplitude at t = 0. The maximum amplitude is equal to the sample value x (0) at t = 0. This sinc function will pass through zeros at t = ± 1/2 W, ± 1/4 W ...etc. This is as shown in Fig. 6.3.

(b) Second term : $x(\pm T_s)$ sinc $2W(t \pm T_s)$:

This sinc function will have maximum amplitude at $t = \pm T_s$. The maximum amplitude is equal to the sample value $x(\pm T_s)$ at $t = \pm T_s$ respectively. Thus sinc 2 W $(t \pm T_s)$ represents shifted sinc function i.e. "sinc 2 Wt" by a period $\pm T_s$. This is as shown in Fig. 6.3.

Similarly the third term, $x(\pm 2T_s)$ sinc $2W(t \pm 2T_s)$ represents shifted sinc function "sinc 2 Wt" by a period of $\pm 2T_s$ and so on. We can plot all these sinc functions along with the sampled signal $x_\delta(t)$ as shown in Fig. 6.3. Note that the peak amplitude of any sinc function is equal to the corresponding sample value $x(nT_s)$.

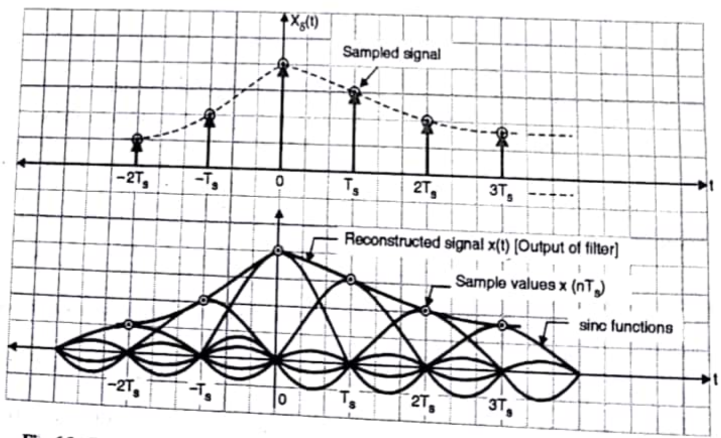


Fig. 6.3 : Reconstruction of the original signal x (t) from its samples using the interpolation

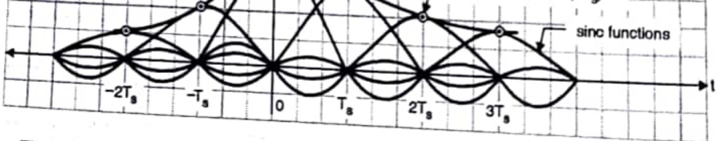


Fig. 6.3 : Reconstruction of the original signal x (t) from its samples using the interpolation

[redacted]

7. Actual reconstruction of the original signal by using a low pass filter :

Thus the peaks of the sinc pulses represent the amplitudes of the samples.

Fig. 6.4(a) : Reconstruction filter

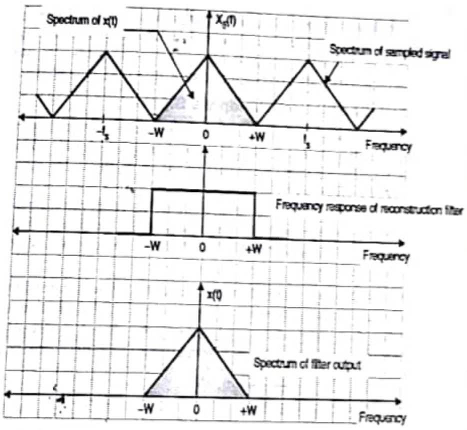


Fig. 6.4(b) : Operation of reconstruction

The signal x (t) expressed in Equation (18) is then passed through an ideal low pass filter to recover the original signal x (t). This low pass filter is therefore called as the reconstruction filter. This is shown in the graphical representation of Fig. 6.4(a).

Assume that the cut-off frequency of the ideal low pass filter is adjusted precisely to W Hz. The frequency response of the reconstruction filter is shown in Fig. 6.4(b).

When the sampled signal $x_\delta(t)$ is applied at the input, this filter will allow only the shaded portion in the spectrum of $x_\delta(t)$ to pass through to the output and will block all other frequency components. Thus the frequency components only corresponding to x (t) will be passed through to the output and the original signal x (t) is recovered.

Q. 3 State and prove sampling theorem and explain the aliasing error. Dec. 04. Dec. 07. May 11. Dec. 11. May 14. Dec. 16

Ans. :

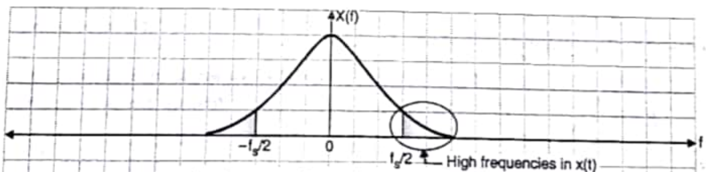
If the signal x (t) is not strictly bandlimited and / or if the sampling frequency f_s is less than 2 W, then an error called **aliasing** or **fold over error** is observed. The adjacent spectrums overlap if $f_s < 2$ W. This is shown in Fig. 6.5(b).

The signal x (t) is not strictly bandlimited. The spectrum of signal x (t) is shown in Fig. 6.5(b).

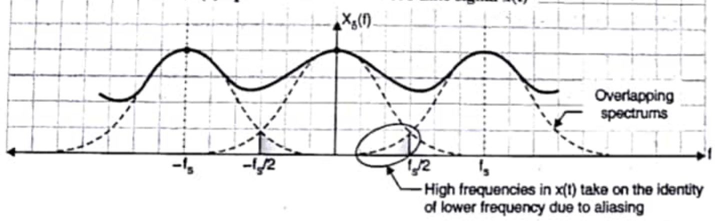
The spectrum $X_\delta(f)$ of the discrete time signal $x_\delta(t)$ is shown in Fig. 6.5(b) which is nothing but the sum of X (f) and infinite number of frequency shifted replicas of it as explained.

Consider the two replicas of X (f) which are centered about the frequencies f_s and $-f_s$. If we use a reconstruction filter with its pass-band extending from $-f_s/2$ to $+f_s/2$ then its output will not be an undistorted version of the original signal x (t). Some distortion will be present in the filter output.

The distortion occurs due to the overlapping of the adjacent spectrums as shown in Fig. 6.5(b). Due to this overlapping, it is seen that the portions of the frequency shifted replicas are "folded over" inside the desired spectrum.



(a) Spectrum of a continuous time signal x(t)



(b) Spectrum of the sampled version of x(t) with $f_s < 2W$

Fig. 6.5

[redacted]

Fig. 6.5

Due to this "fold over", high frequencies in X (f) are reflected into low frequencies in X_δ (f). This can be understood by comparing the shaded portions of the spectra shown in Fig. 6.5(a) and (b).

Aliasing : This phenomenon of a high frequency in the spectrum of the original signal x(t), taking on the identity of lower frequency in the spectrum of the sampled signal x_δ (t) is called as aliasing or fold over error.

Effect of aliasing

Due to aliasing some of the information contained in the original signal x (t) is lost in the process of sampling.

Q. 4 Prove sampling theorem for low pass signals. What is the use of antialiasing filter ? May 12, Dec. 14

Ans. : Aliasing can be completely eliminated if we take the following action : Use a bandlimiting low pass filter and pass the signal x (t) through it before sampling as shown in Fig. 6.6(a).

This filter has a cutoff frequency at f_c = W, therefore it will strictly bandlimit the signal x (t) before sampling takes place. This filter is also called as antialiasing filter or prealias filter.

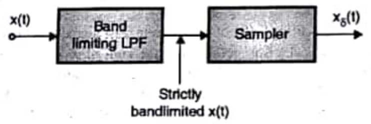


Fig. 6.6(a) : Use of a bandlimiting filter to eliminate aliasing

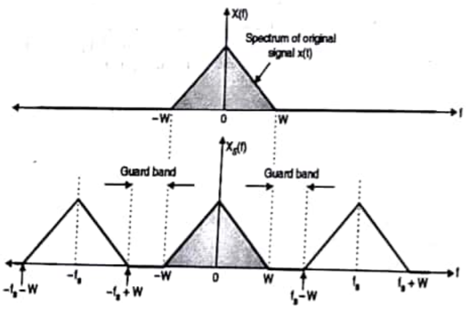


Fig 6.6(b) : Spectrum of a sampled signal for $f_s > 2$ W

Increase the sampling frequency f_s to a great extent i.e. $f_s \gg 2$ W. Due to this, even though x (t) is not strictly bandlimited, the spectrums will not overlap. A guard band is created between the adjacent spectrums as shown in Fig. 6.6(b).

Thus aliasing can be prevented by :

1. Using an antialiasing or prealiasing filter and
2. Using the sampling frequency $f_s > 2$ W.

Q. 5 State sampling theorem for low pass signals. What is Nyquist rate ? Dec. 12

Ans. :

The minimum sampling rate of "2 W" samples per second for a signal x (t) having maximum frequency of "W" Hz is called as "Nyquist rate". The reciprocal of Nyquist rate i.e. 1/2 W is called as the Nyquist interval.

Nyquist rate = 2 W Hz

Nyquist interval = 1/2 W seconds

Q. 6 Explain sampling theorem for bandpass signals with proof and also explain anti-aliasing filter. Dec. 14

Ans. : The sampling theorem for the bandpass signals can be stated as follows :

A bandpass signal x (t), having a maximum bandwidth of 2 W Hz can be completely represented in its sampled form and recovered back from the sampled form if it is sampled at a rate which is at least twice the maximum bandwidth. (i.e. $f_s \geq 4$ W.)

Quadrature Sampling of Bandpass Signals

In this section, we consider a scheme called "quadrature sampling" for the uniform sampling of bandpass signals. This scheme is actually a natural extension of the sampling of low pass signals. The scheme is as follows :

In this scheme, we do not sample the bandpass signal directly. Instead, before sampling we represent the bandpass signal x (t) in terms of its "in-phase" and "quadrature" components, x_I (t) and x_Q (t) respectively.

The in-phase and quadrature components can be obtained by multiplying the bandpass signal x (t) by cos $(2\pi f_c t)$ and sin $(2\pi f_c t)$ respectively and then by suppressing the sum frequency components by means of low pass filters as shown in Fig. 6.7(a).

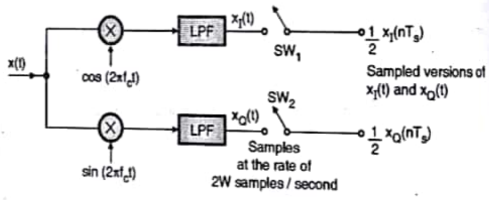


Fig. 6.7(a) : Generation of in-phase and quadrature samples from the bandpass signal x (t)

If x_I (t) = In-phase component and

x_Q (t) = Quadrature component.

Then we can express the bandpass signal x (t) in terms of x_I (t) and x_Q (t) as follows :

$$x(t) = x_I(t)\cos(2\pi f_c t) - x_Q(t)\sin(2\pi f_c t) \quad ...(21)$$

Under the assumption of $f_c > W$, it is found that x_I (t) and x_Q (t) both are "low pass signals" extending from – W to + W as shown in Fig. 6.7(b).

Then both the in-phase and quadrature components are separately sampled at a rate of 2 W samples per second by the switches SW_1 and SW_2 as shown in Fig. 6.7(a) to obtain the sampled versions of x_I (t) and x_Q (t).

$X_I(f), X_Q(f)$

Q. 8 With proper waveforms explain principles of qPWM, PAM, PPM systems of modulat

... aliasing can be prevented by :

1. Using an antialiasing or prealiasing filter and
2. Using the sampling frequency $f_s > 2$ W.

Q. 5 State sampling theorem for low pass signals. What is Nyquist rate ? **Dec. 12**

... in Fig. 6.7(b).

Then both the in-phase and quadrature components are separately sampled at a rate of 2 W samples per second by the switches SW_1 and SW_2 as shown in Fig. 6.7(a) to obtain the sampled versions of x_I (t) and x_Q (t).

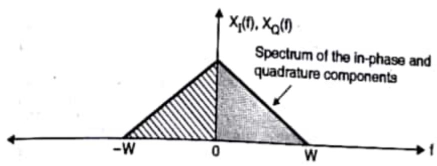


Fig. 6.7(b) : Spectrum of the in-phase and quadrature components of x (t)

In order to reconstruct the original bandpass signal from its quadrature sampled version, we first reconstruct the in-phase component x_I(t) and quadrature component x_Q (t) from their respective sampled versions x_I (nT_s) and x_Q (nT_s) by means of reconstruction filters. Then multiply x_I (t) and x_Q (t) by cos (2 π f_c t) and sin (2 π f_c t) respectively and add the result. The reconstruction process of x (t) is shown in Fig. 6.7(c).

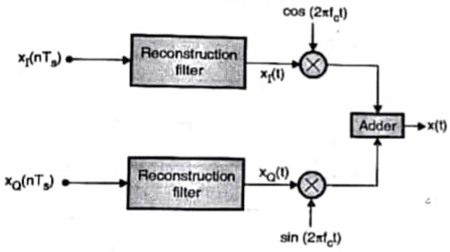


Fig. 6.7(c) : Reconstruction of the bandpass signal x(t)

Q. 7 A bandpass signal has a spectral range that extends from 30 kHz to 75 kHz. Find the sampling frequency.

Ans. :

Given : f_1 = 30 kHz and f_2 = 75 kHz.

Since f_1 = 30 kHz and f_2 = 75 kHz.

∴ Bandwidth $B = f_2 - f_1 = 75 - 30 = 45$ kHz ...(1)

Let us assume that $f_s = 2\,B = 2 \times 45 = 90$ kHz ...(2)

Note that here $f_s = 3\,f_1$ i.e. f_s is the third harmonic of f_1 or f_1 and f_s are harmonically related with each other. Therefore we have to use the sampling theorem stated below to find out the value of f_s.

Sampling theorem for bandpass signals

The sampling theorem for the bandpass signals can be stated as follows :

A bandpass signal x (t), having a maximum bandwidth of 2 W Hz can be completely represented in its sampled form and recovered back from the sampled form if it is sampled at a rate which is at least twice the maximum bandwidth. (i.e. $f_s \geq 4$ W.)

Hence the sampling frequency for the given bandpass signal is given by,

$$f_s = 2\,B = 90 \text{ kHz} \quad \text{...Ans.}$$

Q. 8 With proper waveforms explain principles of qPWM, PAM, PPM systems of modulat **Dec. 03, May 16**

Ans. : The other type of a pulse analog modulation is the Pulse Width Modulation (PWM). In PWM, the width of the carrier pulses varies in proportion with the amplitude of modulating signal. The waveforms of PWM are as shown in Fig. 6.8

As seen from the waveforms, the amplitude and the frequency of the PWM wave remains constant. Only the width changes. That is why the "information" is contained in the width variation. This is similar to FM. As the noise is normally "additive" noise, it changes the amplitude of the PWM signal.

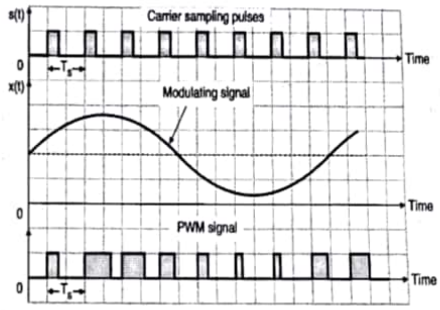


Fig. 6.8 : PWM signal [Trail edge modulated signal]

At the receiver, it is possible to remove these unwanted amplitude variations very easily by means of a limiter circuit.

As the information is contained in the width variation, it is unaffected by the amplitude variations introduced by the noise. Thus the PWM system is more immune to noise than the PAM signal.

Q. 9 Draw the block diagram of PWM generator. Explain the working giving waveforms at the output of each block. **Dec. 15, May 16**

Ans. : The block diagram of Fig. 6.9(a) can be used for the generation of PWM as well as PPM.

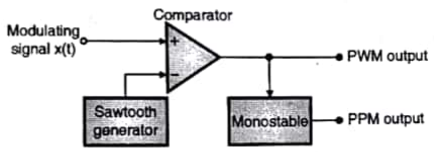


Fig. 6.9(a) : PWM and PPM generator

A sawtooth generates a sawtooth signal of frequency f_s, therefore the sawtooth signal in this case is a sampling signal. It is applied to the inverting terminal of a comparator. The modulating signal x (t) is applied to the non-inverting terminal of the same comparator. The comparator output will remain high as long as the instantaneous amplitude of x (t) is higher than that of the ramp signal.

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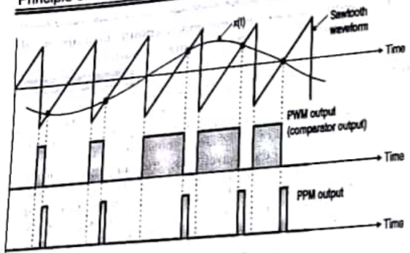

Fig. 6.9(b) : Waveforms

This gives rise to a PWM signal at the comparator output as shown in Fig. 6.9(b). Note that the leading edges of the PWM waveform coincide with the falling edges of the ramp signal. Thus the leading edges of PWM signal are always generated at fixed time instants. However the occurrence of its trailing edges will be dependent on the instantaneous amplitude of x (t). Therefore this PWM signal is said to be trail edge modulated PWM.

Q. 10 Explain PWM generation and degeneration method. May 03, Dec. 15

Ans.: The block diagram of PWM detector is as shown in Fig. 6.10.

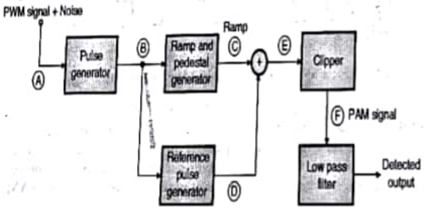

Fig. 6.10 : PWM detection circuit

Operation: The PWM signal received at the input of the detection circuit is contaminated with noise. This signal is applied to pulse generator circuit which regenerates the PWM signal. Thus some of the noise is removed and the pulses are squared up. The regenerated pulses are applied to a reference pulse generator. It produces a train of constant amplitude, constant width pulses. These pulses are synchronized to the leading edges of the regenerated PWM pulses but delayed by a fixed interval.

The regenerated PWM pulses are also applied to a ramp generator. At the output of it we get a constant slope ramp for the duration of the pulse. The height of the ramp is thus proportional to the widths of the PWM pulses. At the end of the pulse a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse.

The constant amplitude pulses at the output of reference pulse generator are then added to the ramp signal. The output of the adder is then clipped off at a threshold level to generate a PAM signal at the output of the clipper. A low pass filter is used to recover the original modulating signal back from the PAM signal. The waveforms for this circuit are as shown in Fig. 6.11.

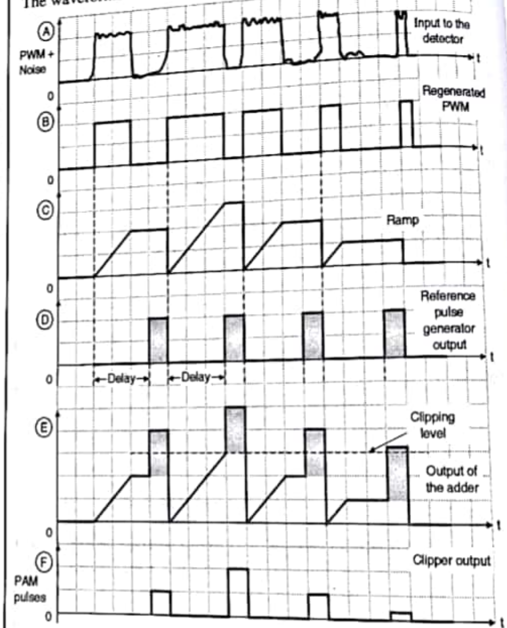

Fig. 6.11 : Waveforms for PWM detection circuit

Frequency Spectrum of PWM Wave : Fig. 6.12 shows the spectrum of a PWM signal for a sinusoidal modulating signal with a frequency f_m. The spectrum shows that the modulating frequency f_m and many of its sidebands are present.

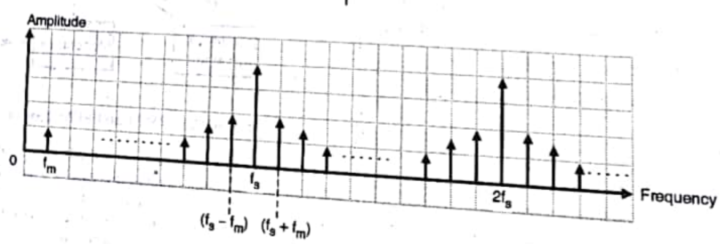

Fig. 6.12 : Spectrum of PWM signal

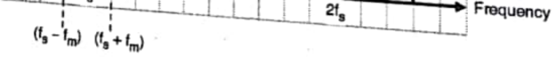

Fig. 6.12 : Spectrum of PWM signal

Q. 11 Explain generation and demodulation of PPM.

Dec. 03, May 15, Dec. 16

Ans. : In PPM the amplitude and width of the pulsed carrier remains constant but the position of each pulse is varied in proportion with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.

The PPM pulses can be derived from the PWM pulses as shown in Fig. 6.13. Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.

The vertical dotted lines drawn in Fig. 6.13 are treated as reference lines to measure the shift in position of PPM pulses. The leading edge of each PPM pulse coincides with the trailing pulse of a PWM pulse.

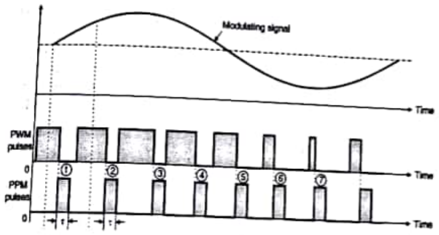

Fig. 6.13 : PPM pulses generated from PWM signal

The PPM pulses marked 1, 2 and 3 etc. in Fig. 6.13 go away from their respective reference lines. This is corresponding to increase in the modulating signal amplitude. Then as the modulating voltage decreases the PPM pulses 4, 5, 6, 7 come progressively closer to their respective reference lines.

Q. 12 Explain generation and demodulation of PPM.

Dec. 04, Dec. 14, May 15, Dec. 16

Ans. : The PPM signal can be generated from PWM signal as shown in Fig. 6.9(a). The same block diagram has been repeated in Fig. 6.14 as shown.

The PWM pulses obtained at the comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered. Hence corresponding to each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its own RC components. Thus as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting as shown in Fig. 6.9(b).

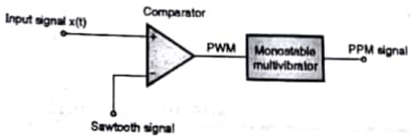

Fig. 6.14 : Generation of PPM signal

Demodulation of PPM : The PPM demodulator block diagram is as shown in Fig. 6.15.

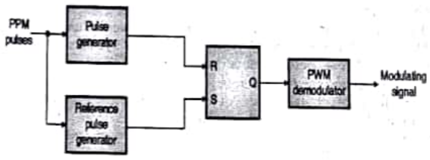

Fig. 6.15 : PPM demodulator circuit

The operation of the demodulator circuit is explained as follows :

1. The noise corrupted PPM waveform is received by the PPM demodulator circuit.
2. The pulse generator develops a pulsed waveform at its output of fixed duration and apply these pulses to the reset pin (R) of a SR flip-flop.
3. A fixed period reference pulse is generated from the incoming PPM waveform and the SR flip-flop is set by the reference pulses.
4. Due to the set and reset signals applied to the flip-flop, we get a PWM signal at its output. The PWM signal can be demodulated using the PWM demodulator.

Q. 13 Write a short note : Comparison of PWM, PAM and PPM.

May 06, Dec. 07

Ans. :

Sr. No.	Parameter	PAM	PWM	PPM
1.	Type of carrier	Train of pulses	Train of pulses	Train of pulses
2.	Variable characteristic of the pulsed carrier	Amplitude	Width	Position
3.	Bandwidth requirement	Low	High	High
4.	Noise immunity	Low	High	High
5.	Information is contained in	Amplitude variations	Width variation	Position variation
6.	Transmitted power	Varies with amplitude of pulses	Varies with variation in width	Remains constant
7.	Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
8.	Complexity of generation and detection	Complex	Easy	Complex

Sr. No.	Parameter	PAM	PWM	PPM
	Similarity with other	Similar to AM	Similar to FM	Similar to PM

a PWM pulse. ... with the trailing pulse of

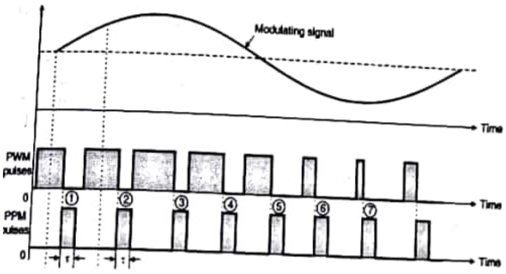

Fig. 6.13 : PPM pulses generated from PWM signal

The PPM pulses marked 1, 2 and 3 etc. in Fig. 6.13 go away om their respective reference lines. This is corresponding to crease in the modulating signal amplitude. Then as the odulating voltage decreases the PPM pulses 4, 5, 6, 7 come ogressively closer to their respective reference lines.

12 Explain generation and demodulation of PPM. Dec. 04, Dec. 14, May 15, Dec. 16

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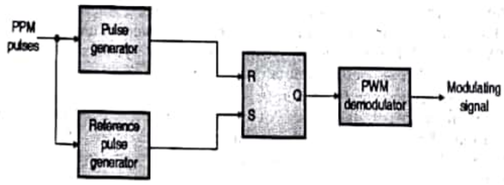

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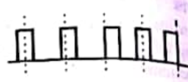
r. No.	Parameter	PAM	PWM	PPM
1.	Type of carrier	Train of pulses	Train of pulses	Train of pulses
2.	Variable characteristic of the pulsed carrier	Amplitude	Width	Position
3.	Bandwidth requirement	Low	High	High
4.	Noise immunity	Low	High	High
5.	Information is contained in	Amplitude variations	Width variation	Position variation
6.	Transmitted power	Varies with amplitude of pulses	Varies with variation in width	Remains constant
7.	Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
3.	Complexity of generation and detection	Complex	Easy	Complex

Sr. No.	Parameter	PAM	PWM	PPM
9.	Similarity with other modulation systems	Similar to AM	Similar to FM	Similar to PM
10.	Output waveforms	 (D-463)		

Chapter 7 : Digital Modulation Techniques

Q. 1 Describe PCM and also explain the PCM encoder and decoder with block diagram. Dec. 07, Dec. 15

Ans. : PCM is a type of pulse modulation like PAM, PWM or PPM but there is an important difference between them. PAM, PWM or PPM are "analog" pulse modulation systems whereas PCM is a "digital" pulse modulation system.

That means the PCM output is in the coded digital form. It is in the form of digital pulses of constant amplitude, width and position. The information is transmitted in the form of "code words". A PCM system consists of a PCM encoder (transmitter)

higher than "W". This will eliminate the possibility of aliasing.

2. The band limited analog signal is then applied to a sample and hold circuit where it is sampled at adequately high sampling rate. Output of sample and hold block is a flat topped PAM signal.
3. These samples are then subjected to the operation called "Quantization" in the "Quantizer". Quantization process ... the process of approximation as will be explained later o... The quantization is used to reduce the effect of noise. Th... combined effect of sampling and quantization produces t... quantized PAM at the quantizer output.
4. The quantized PAM pulses are applied to an encoder which