Comparison between PAM, PWM, and PPM

The comparison between the above modulation processes is presented in the table.

PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is high	System complexity is low	System complexity is low
Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulation

Pulse Amplitude Modulation

Pulse Amplitude Modulation (PAM) is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal. The pulse amplitude modulated signal, will follow the amplitude of the original signal, as the signal traces out the path of the whole wave. In natural PAM, a signal sampled at the Nyquist rate is reconstructed, by passing it through an efficient Low Pass Frequency (LPF) with exact cutoff frequency.

Pulse amplitude modulation is a type of modulation in which the amplitudes of regularly spaced rectangular pulses vary according to instantaneous value of the modulating or message signal. In fact, the pulses in a PAM signal may be of flat top type or natural type or ideal type. Out of all the three pulse amplitude modulation methods, the flat top PAM is most popular and is widely used. The reason for using flat top PAM is that during the transmission, the noise interferes with the top of the transmitted pulses and this noise can be easily removed if the PAM pulse has flat top. However, in case of natural samples PAM signal, the pulse has varying top in accordance with the signal variation. Now, when such type of pulse is received at the receiver, it is always contaminated by noise. Then it becomes quite difficult to determine the shape of the top of the pulse and thus amplitude detection of the pulse is not exact. Due to this, errors are introduced in the received signal.

The following figures explain the Pulse Amplitude Modulation.

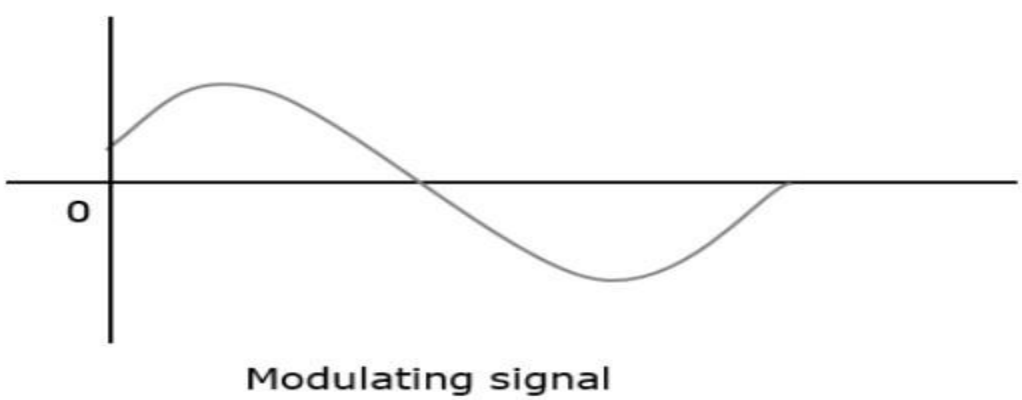

Figure 5.3.1(a) Modulating signal

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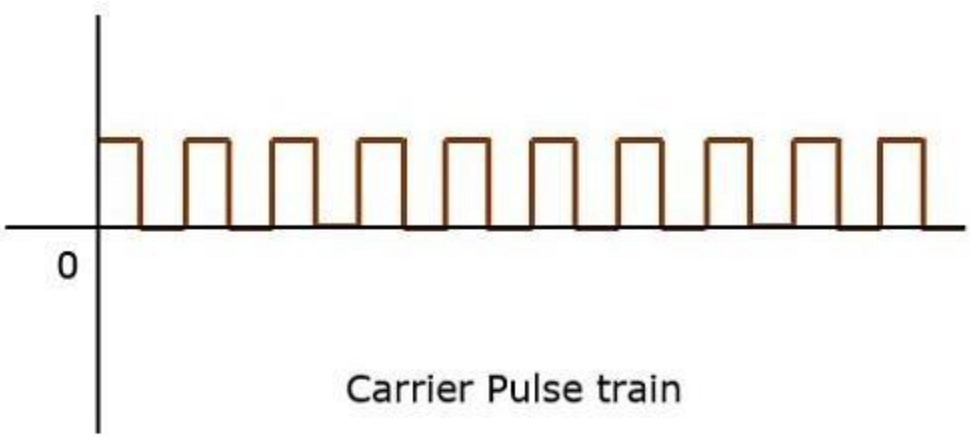

Figure 5.3.2(b) Carrier Pulse Train

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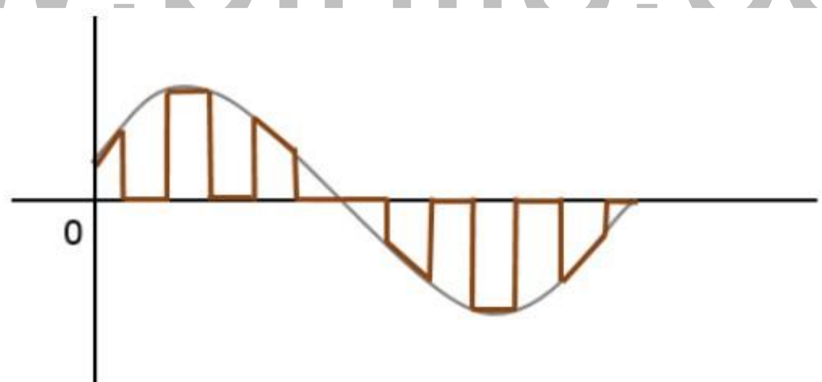

Figure 5.3.3(c) Natural PAM

Diagram Source Brain kart

Though the PAM signal is passed through an LPF, it cannot recover the signal without distortion. Hence to avoid this noise, flat-top sampling is done as shown in the following figure. Therefore, flat top sampled PAM is widely used.

Fig. Fig 5.3.1(c) shows the sample and hold circuit to produce flat top sampled PAM and the waveform for flat top sampled PAM.

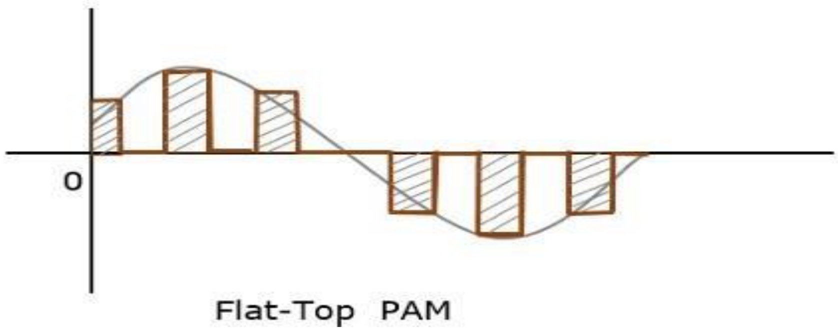

Figure 5.3.4(c) Flat top sampled PAM

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Flat-top sampling is the process in which sampled signal can be represented in pulses for which the amplitude of the signal cannot be changed with respect to the analog signal, to be sampled. The tops of amplitude remain flat. This process simplifies the circuit design.

Pulse amplitude modulation is a technique in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal. It is a modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling. This technique transmits the data by encoding in the amplitude of a series of signal pulses.

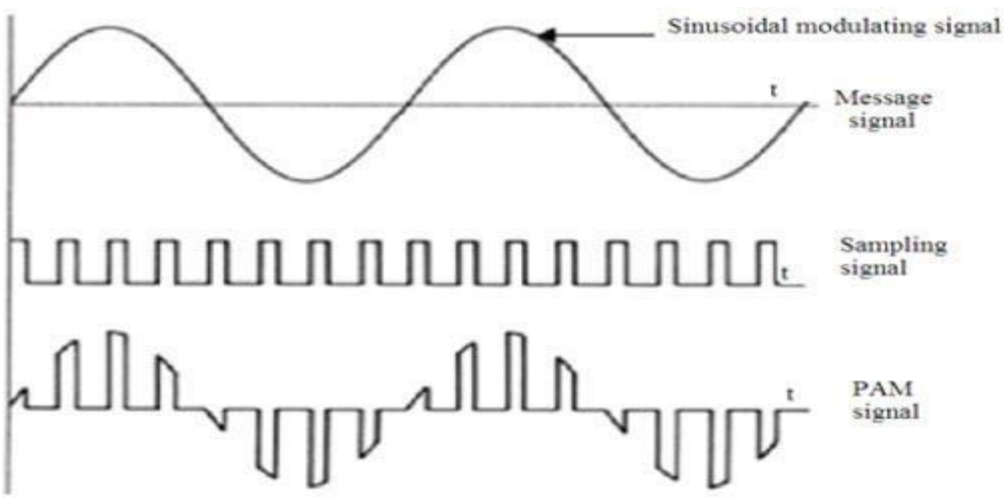


Figure5.3.5 Pulse Amplitude Modulation Signal

Diagram Source Brain kart

There are two types of sampling techniques for transmitting a signal using PAM. They are:

(i) Flat Top PAM

(ii) Natural PAM

Flat Top PAM

The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. The amplitude of the signal cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.

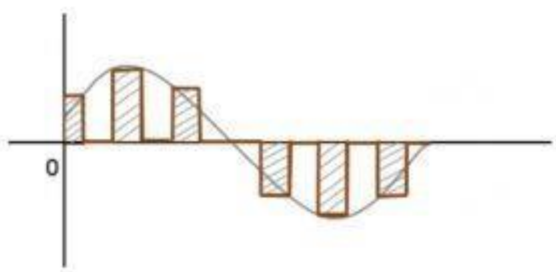


Figure 5.3.6 Flat Top Pulse Amplitude Modulation

Diagram Source Brain kart

Natural PAM

The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. Then follows the amplitude of the pulse for the rest of the half-cycle.

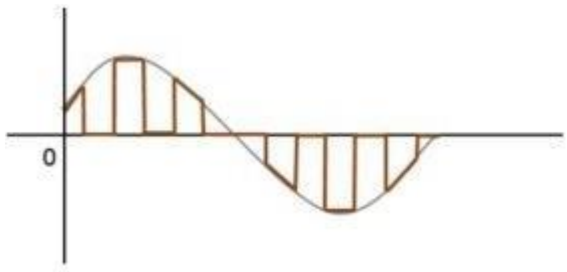


Figure 5.3.7 Natural Pulse Amplitude Modulation

Diagram Source Brain kart

In Pulse modulation, the unmodulated carrier signal is a periodic train of signals. So the pulse train can be described like the following.

$$up(t) = \sum_{k=-\infty}^{\infty} A \operatorname{rect}\left(\frac{t - kTs}{\tau}\right)$$

Where 'A' is the unmodulated pulse amplitude 'τ' is pulse width . The pulse trains periodic amplitudes can be changed based on the modulating signal. Here, the modulating signal like m(t), PAM time can be denoted as 'Ts' In PAM, the signal can be achieved through multiplying the carrier signal with the modulating signal. The o/p is a set of pulses, where the amplitudes of signals can be changed on the modulating signal.

The specific type of PAM can be referred to as normal PAM, as the pulses follow the outline of the modulating signal. The pulse train works like a periodic switching signal toward the modulator. Once it is switched ON, and then allows the samples of modulating signals to supply toward the output. The pulse train's periodic time is called the sampling period.

Fs = 1/Ts

PAM Generation and Principle of Working

A sample and hold circuit shown in fig.5.3.5(a) is used to produce Flat top sampled PAM. The working principle of this circuit is quite easy. The sample and Hold (S/H) circuit consists of two field effect transistors (FET) switches and a capacitor. The sampling switch is closed for a short duration by a short pulse applied to the gate G1 of the transistor. During this period, the capacitor 'C' is quickly charged upto a voltage equal to the instantaneous sample value of the incoming signal x(t).Now, the sampling switch is opened and the capacitor 'C' holds the charge**.**

The discharge switch is then closed by a pulse applied to gate G2 of the other transistor. Due to this, the capacitor 'C' is discharged to zero volts. The discharges switch is then opened and thus capacitor has no voltage. Hence, the output of the sample and hold circuit consists of a sequence of flat top samples as shown in fig.5.3.5(b).

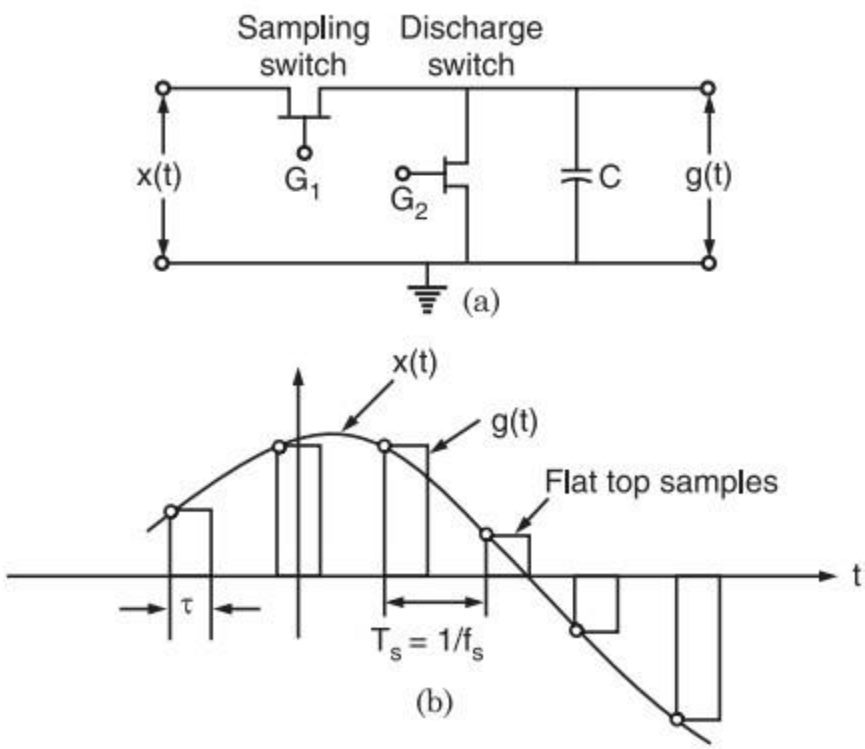


Figure.5.3.8 : (a) Sample and hold circuit generating flat top sampled PAM, (b) Waveforms of flat top sampled PAM

Diagram Source Brain kart

Mathematical Analysis

In a flat top PAM, the top of the samples remains constant and is equal to the instantaneous value of the baseband signal x(t) at the start of sampling. The duration or width of each sample is τ and sampling rate is equal to,

$$f_s = \frac{1}{T_s}$$

From fig.5.3.5 (b), it may be noted that only starting edge of the pulse represents instantaneous value of the baseband signal x(t).Also, the flat top pulse of g(t) is mathematically equivalent to the convolution of instantaneous sample and a pulse h(t) as depicted in fig.5.3.6.

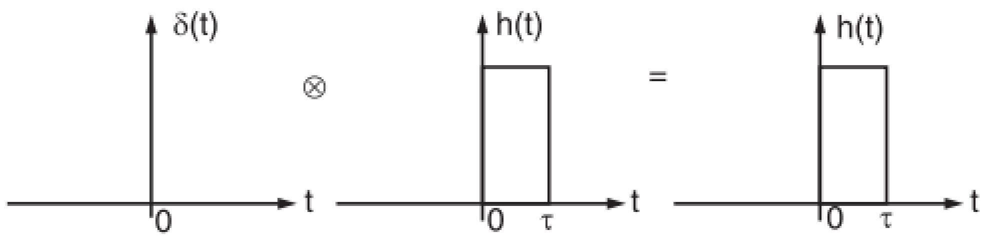


Figure.5.3.9: Convolution of any function with delta function is equal to that function

Diagram Source Brain kart

This means that the width of the pulse in g(t) is determined by the width of h(t) and the sampling instant is determined by the delta function. In fig.1 (b), the starting edge of the pulse represents the point where baseband signal is sampled and width is determined by function h(t). Therefore, g(t) will be expressed as,

$$g(t) = s(t) \otimes h(t)$$

This equation has been explained in fig.5.3.7 (a), (b),(c),(d)below.

Now, from the property of delta function, we know that for any function f(t),

$$f(t) \otimes s(t) = f(t)$$

This property is used to obtain flat top samples. It may be noted that to obtain flat top sampling, we are not applying the equation directly here i.e., we are applying a modified form of equation Thus, in this modified equation, we are taking s(t) in place of delta functions δ(t).

Observe that δ(t) is a constant amplitude delta function whereas s(t) is a varying amplitude train of impulses. This means that we are taking s(t) which is an instantaneously sampled signal and this is convolved with function h(t) as in equation

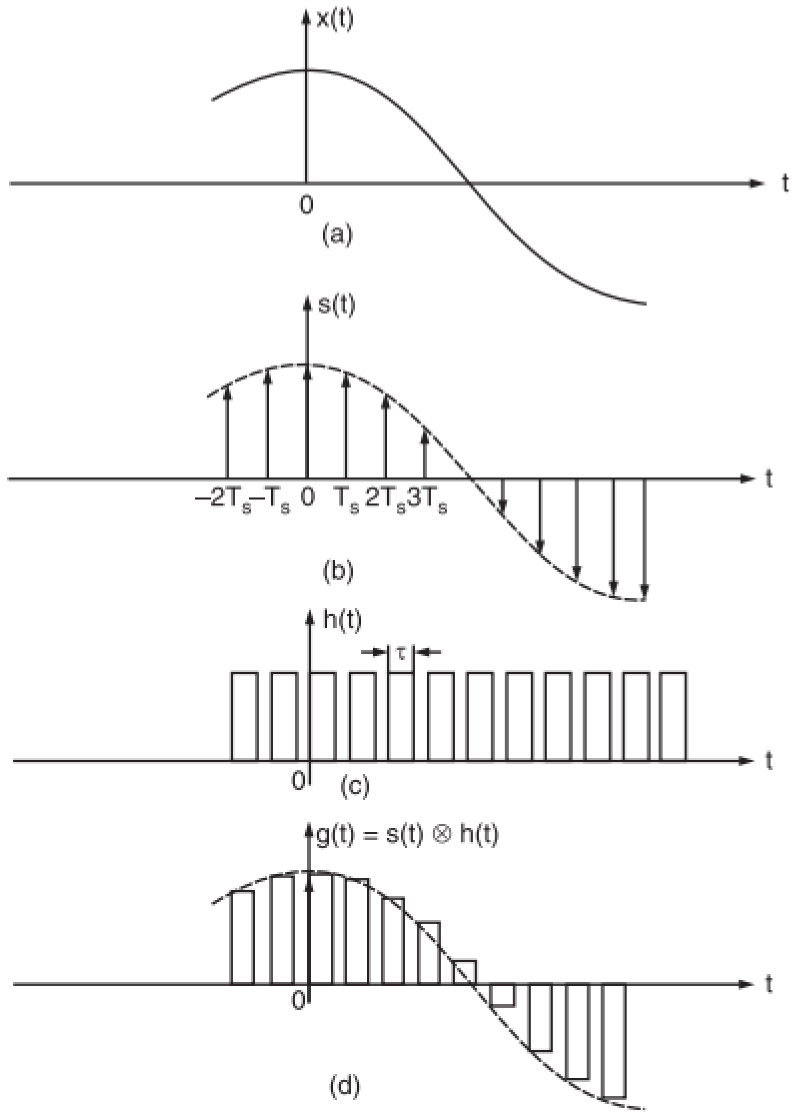


Figure.5.3.10 (a) Baseband signal x(t), (b) Instantaneously sample signal s(t), (c) Constant pulse width function h(t), (d) Flat top sampled PAM signal g(t) obtained through convolution of h(t) and s(t).

Diagram Source Brain kart

Therefore, on convolution of s(t) and h(t), we get a pulse whose duration is equal to h(t) only but amplitude is defined by s(t). Now, we know that the train of impulses may be represented mathematically as,

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

The signal s(t) is obtained by multiplication of baseband signal x(t) and δTs (t). Thus,

$$s(t) = x(t) \cdot \delta_{T_s}(t)$$

Or,

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

Now, sampled signal g(t) is given as equation (1)

$$g(t) = s(t) \otimes h(t)$$

or $$g(t) = \int_{-\infty}^{\infty} s(\tau)\, h(t-\tau)\, d\tau$$

or $$g(t) = \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(\tau - nT_s)\, h(t-\tau)\, d\tau$$

or $$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \int_{-\infty}^{\infty} \delta(\tau - nT_s)\, h(t-\tau)\, d\tau$$

According to shifting property of delta function, we know that,

$$\int_{-\infty}^{\infty} f(t)\, \delta(t - t_0) = f(t_0)$$

Hence,

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t - nT_s)$$

This equation represents value of g(t) in terms of sampled value x(nTs) and function h(t – nTs) for flat top sampled signal. Now, again from equation (1), we have

$$g(t) = s(t) \otimes h(t)$$

Taking Fourier transform of both sides of above equation, we get

$$G(f) = S(f)\, H(f)$$

We know that S(f) is given as

$$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

Therefore,

$$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)\, H(f)$$

Thus, spectrum of flat top PAM signal:

$$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)\, H(f)$$

Here, H(f) is the Fourier transform of the rectangular pulse. The spectrum of this rectangular pulse is shown in fig.4(b).

Let the spectrum of s(t) be the rectangular pulse train as shown in fig.4(a) and the spectrum of h(t) i.e., H(f) is shown in fig.5.3.8(b).

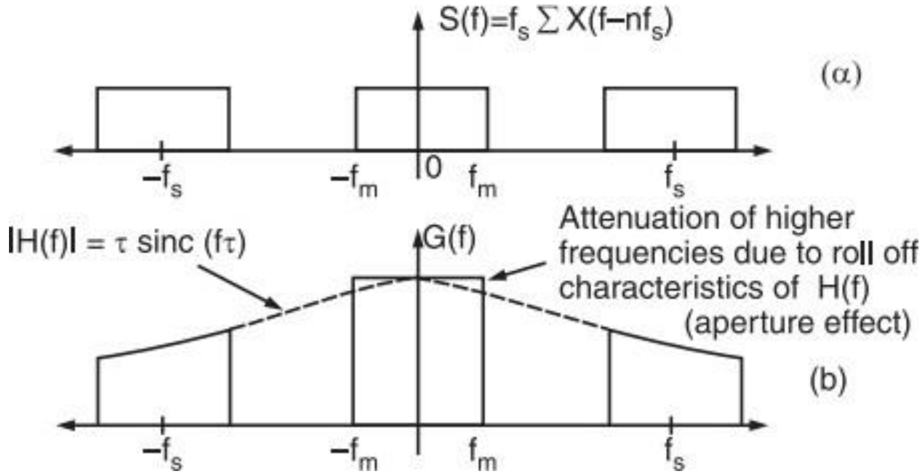


Figure.5.3.11 : (a) Spectrum of some arbitrary signal. The signal is sampled at fs and maximum frequency in the signal is fm, (b) Spectrum of flat top signal. The dotted curve is H(f) = τ sin c(fτ)

Diagram Source Brain kart

we know that,

$$G(f) = S(f) \cdot H(f)$$

Thus, according to above equation, we can plot the spectrum G(f) as shown in fig.4(b).

It may be observed in fig.5.3.8 (b) that higher frequencies in S(f) are attenuated due to roll-off characteristics of the 'sinc' pulse. This effect is popularly known as aperture effect. An equalizer is needed to overcome this effect.

Pulse Code Modulation

Pulse-code modulation or PCM is known as a digital pulse modulation technique . In fact, the pulse-code modulation is quite complex as compared to the analog pulse modulation techniques i.e. PAM, PWM and PPM, in the sense that the message signal is subjected to a great number of operations . In PCM an anlog signal or information is converted into a binary sequence, i.e.,' **1's** and '**0's**. The output of a PCM resembles a binary sequence. The following figure 5.7.1 shows an example of PCM output with respect to instantaneous values of a given sine wave.

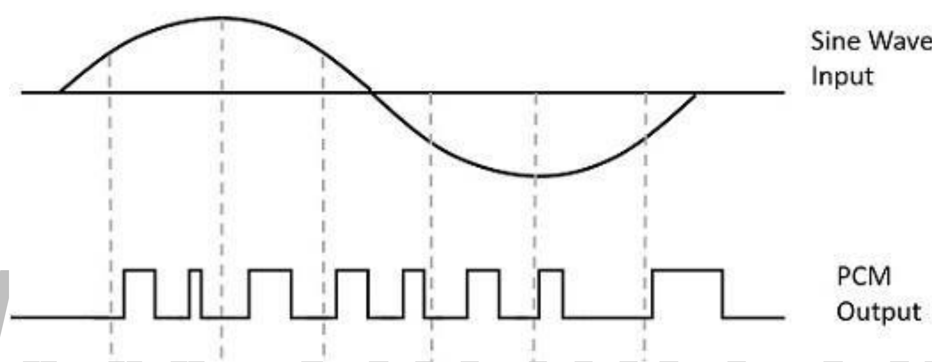


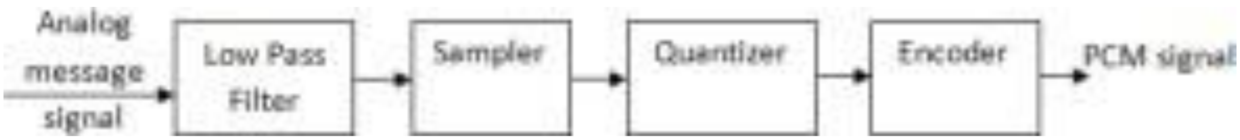
Figure 5.7.1 PCM Output

Diagram Source Brain Kart

PCM produces a series of numbers or digits instead of a pulse train,. Each one of these digits, in binary code, represent the approximate amplitude of the signal sample at that instant. In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

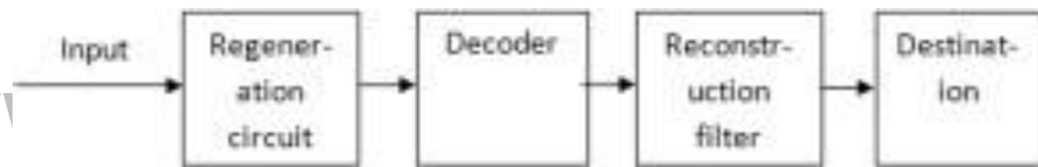
Elements of a PCM System

Fig.5.7.2 shows the basic elements of a PCM system .



(a) : PCM Transmitter

(b) : Transmitter Path



(c) : Receiver

Figure.5.7.2 : The basic elements of a PCM System

Diagram Source Brain Kart

It consists of three main parts i.e. ,

1. Transmitter
2. Transmission path
3. Receiver

The essential operation in the transmitter of a PCM system are :

1. Sampling
2. Quantizing
3. Encoding

As discussed earlier, sampling is the operation in which an analog (continuous-time) signal is sampled according to the sampling theorem resulting in a discrete-time signal .

The quantizing and encoding operations are usually performed in the same circuit which is known as an analog-to-digital converter (ADC) .

The essential operations in the receiver of a PCM system are :

1. Regeneration of impaired signals
2. Decoding and demodulation of the train of quantized samples

These operations are usually performed in the same circuit which is known as a digital-to-analog converter (DAC) .

Further, at intermediate points, along the transmission route from the transmitter to the receiver, regenerative repeaters are used to reconstruct (i.e. regenerate) the transmitted sequence of coded pulses in order to combat the accumulated effects of signal distortion and noise .

As discussed before, the quantization refers to the use of a finite set of amplitude levels and the selection of a level nearest to a particular sample value of the message signal as the representation for it . In fact, this operation combined with sampling, permits the use of coded pulses for representing the message signal . Thus, it is the combined use of quantizing and coding that distinguishes pulse code modulation from analog modulation techniques .

Few Important Points :

Now, let us summarize PCM in the form of few points as under :

1. PCM is a type of pulse modulation like PAM, PWM or PPM but there is an important difference between them i.e. PAM, PWM or PPM are analog pulse modulation systems whereas PCM is a digital pulse modulation system .
2. This means that the PCM output is in the coded digital form . It is in the form of digital pulses of constant amplitude, width and position .

3. The information is transmitted in the form of code words . A PCM system consists of a PCM encoder (transmitter) and a PCM decoder (receiver) .
4. The essential operations in the PCM transmitter are sampling, quantizing and encoding .
5. All the operations are usually performed in the same circuit called as analog-to digital converter .
6. It should be understood that the PCM is not modulation in the conventional sense . Because in modulation, one of the characteristics of the carrier is varied in proportion with the amplitude of the modulating signal . Nothing of that sort happen in PCM .

1. PCM Transmitter

Fig.5.7.3 shows a practical block diagram of a PCM generator or transmitter.

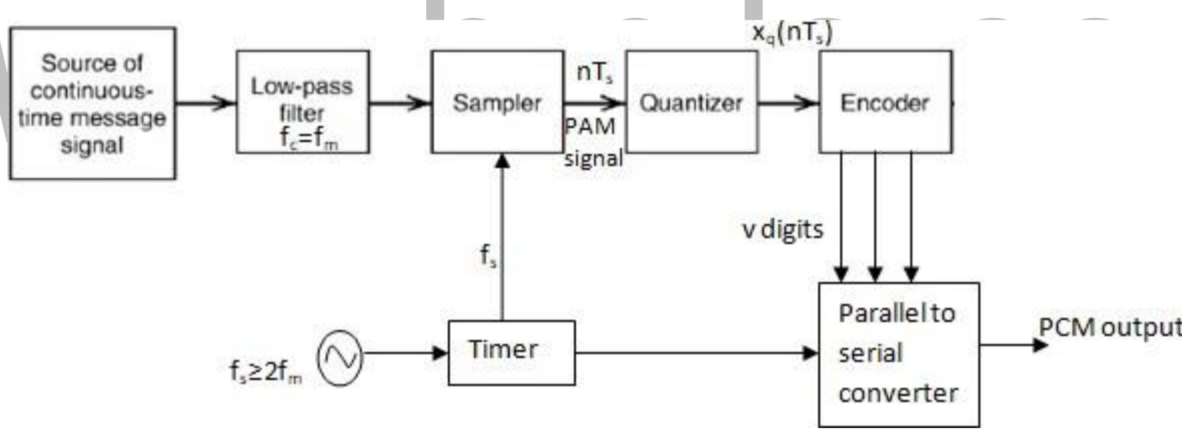


Figure 5.7.3 : PCM Transmitter

Diagram Source Brain Kart

In PCM transmitter , the signal x(t) is first passed through the low-pass filter of cut-off frequency f_m Hz . This low-pass filter blocks all the frequency components above f_m Hz.This means that now the signal x(t) is bandlimited to f_m Hz .

The sample and hold circuit then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficiently above nyquist rate to avoid aliasing i.e.,

$$f_s \geq 2f_m$$

The output sample and hold circutit is denoted by $x(nT_s)$. This signal $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compares input $x(nT_s)$ with its fixed digital levels. It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. This error is called quantization error. Thus output of quantizer is a digital level called $x_q(nT_s)$.

Now the quantized signal level $x_q(nT_s)$ is given to binary encoder.This encoder converts input signal to 'v' digits binary word.This encoder is also known as digitizer. In addition to these, there is an oscillator which generates the clocks for sample and hold circuit and parallel to serial converter.

In PCM, sample and hold, quantizer and encoder combinely form an analog to digital converter (ADC).

2. PCM Transmission Path

The path between the PCM transmitter and receiver over which the PCM signal travel, is known as PCM transmission path and it is shown in Fig.5.7.4

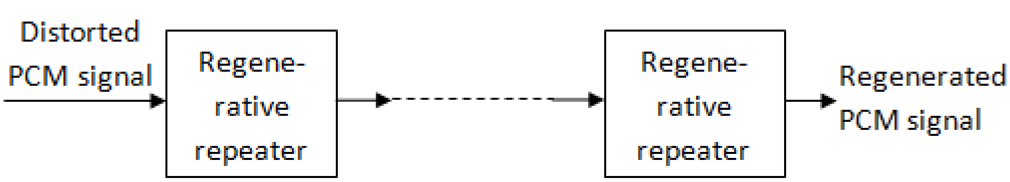


Figure 5.7.4: PCM Transmission path

Diagram Source Brain Kart

The most important feature of PCM system lies in its ability to control the effects of distortion and noise when the PCM wave travels on the channel. This is accomplished by means of using a chain of regenerative repeaters as shown in Such repeaters are spaced close enough to each other on the transmission path.

The repeaters perform three basic operations such as : quantization, timing and decision making. Hence, each repeaters actually reproduces the clean and noise free PCM signal. This improves the performance of PCM in presence of noise.

Repeater

Fig.5.7.5 shows the block diagram of a repeater.

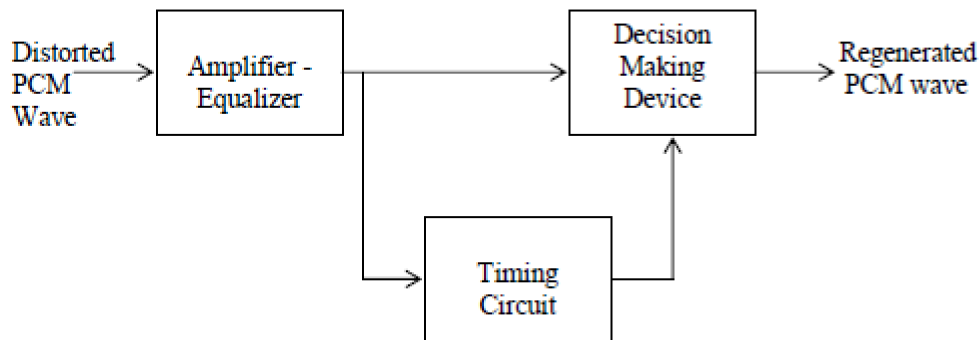


Figure 5.7.5: Block Diagram Of Regenerative Repeater

Diagram Source Brain Kart

The amplitude quantizer shapes the distorted PCM wave so as to compensate for the effects of amplitude and phase distortions. The timing circuit produces a periodic pulse train which is derived from the input PCM pulses. This pulse train is then applied to the decision making device. The decision making device uses this pulse train for sampling the equalized PCM pulses. The sampling is carried out at the instants where the signal to noise ratio is maximum. The decision device makes a decision about whether the equalized PCM wave at its input has 0 value or 1 value at the instant of sampling. Such a decision is made by comparing equalized PCM with a reference level called decision thershold . At the output of the decision device, we get a clean PCM signal without any noise.

PCM Receiver

Fig. 5.7.6 shows the block diagram of a PCM receiver .

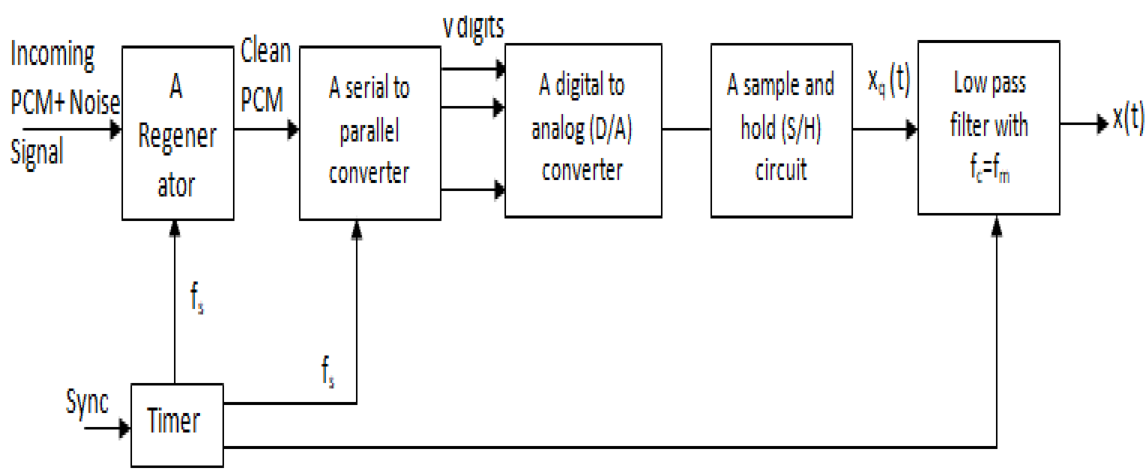


Figure 5.7.6 PCM Receiver

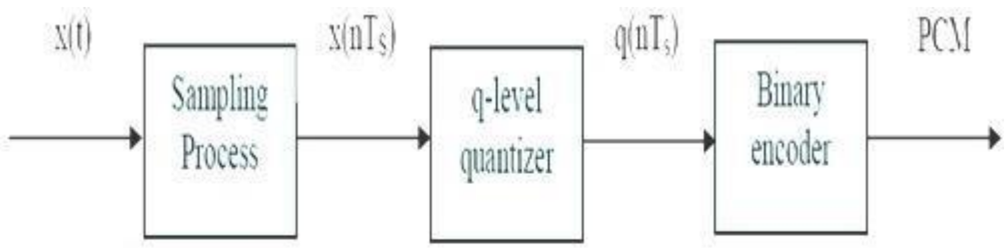
Diagram Source Brain Kart

The regenerator at the start of PCM receiver reshapes the pulse and removes the noise. This signal is then converted to parallel digital words for each sample. Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit is allowed to pass through a low-pass reconstruction filter to get the original message signal x(t) .

Pulse code Modulation:

The pulse code modulator technique samples the input signalx(t) at a sampling frequency. This sampled variable amplitude pulse is then digitalized by the analog to digital converter. Figure 5.7.7 shows the PCM generator.

The signal is first passed through sampler which is sampled at a rate of (fs) where:The output of the sampler x(nTs) which is discrete in time is fed to a „q" level quantizer. The quantizer compares the input x(nTs) with it's fixed levels. It assigns any one of the digital level to x(nTs) that results in minimum distortion or error. The error is called quantization error, thus the output of the quantizer is a digital level called q(nTs). The quantized



$f_s \geq 2f_m$

Figure .5.7.7 Analog To Digital Conversion

Diagram Source Brain Kart

signal level q(nTs) is binary encode. The encoder converts the input signal to ***v*** digits binary word.The receiver starts by reshaping the received pulses, removes the noise and then converts the binary bits to analog shown in the figure 5.7.8. The received samples are then filtered by a low pass filter; the cut off frequency is at fc.

fc= fm.

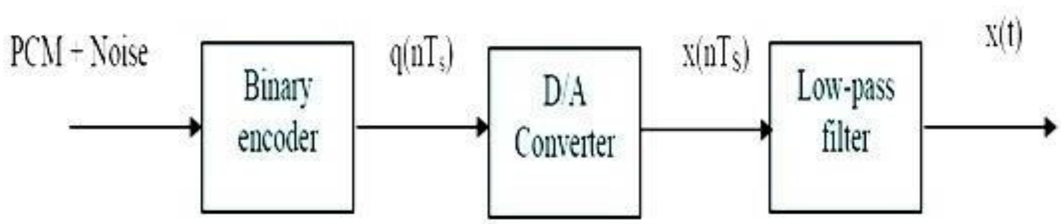


Figure.5.7.8 Digital To Analog Converter

Diagram Source Brain Kart

It is impossible to reconstruct the original signal x(t) because of the permanent quantization error introduced during quantization at the transmitter. The quantization error can be reduced by the increasing quantization levels. This corresponds to the increase of bits persample(more information). But increasing bits ***(v***) increases the signaling rate and requires a large transmission bandwidth. The choice of the parameter for the number of quantization levels must be acceptable with the quantization noise (quantization error).

Signaling Rate in PCM

Let the quantizer use 'v' number of binary digits to represent each level. Then the number of levels that can be represented by ***v*** digits will be : q=2*v*

The number of bits per second is given by : (Number of bits per second)=(Number of bits per samples)x(number of samples per second) = ***v*** (bits per sample) x fs (samples per second). The number of bits per second is also called signaling rate of PCM and is Signaling rate= ***v***fs

Quantization Noise in PCM System

Errors are introduced in the signal because of the quantization process. This error is called "quantization error".

$$\varepsilon = x_q(nT_s) - x(nT_s)$$

Let an input signal x(nTs) have an amplitude in the range of *x*max to - *x*max The total amplitude range is : Total amplitude

$$x_{max} - (-x_{max}) = 2x_{max}$$

If the amplitude range is divided into 'q' levels of quantizer, then the step size 'Δ'. Δ= q/2 X max If the minimum and maximum values are equal to 1,

*x*max,=1, - *x*max=-1, Δ= q/2.

Signal to Quantization Noise ratio in PCM

The signal to quantization noise ratio is given as

$$\boxed{\frac{S}{N_q} = \frac{\text{Normalized signal power}}{\text{Normalized noise power}}}$$

$$= \frac{\text{Normalized signal power}}{\frac{\Delta^2}{12}}$$

The number of quantization value is equal to: q=2***v***

$$\Delta = \frac{2X_{max}}{2^{\nu}}$$

$$\frac{S}{N_q} = \frac{\text{Normalized signal power}}{\left[\frac{2X_{max}}{2^{\nu}}\right]^2 * \frac{1}{12}}$$

Let the normalized signal power is equal to P then the signal to quantization noise will be given by:

$$\boxed{\frac{S}{N_q} = \frac{3P * 2^{2\nu}}{X_{max}^2}}$$

Advantages of PCM

1. Effect of noise is reduced.
2. PCM permits the use of pulse regeneration.
3. Multiplexing of various PCM signals is possible.

Pulse Position Modulation

Pulse Position Modulation (PPM) is an analog modulating scheme in which the amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal. The transmitter has to send synchronizing pulses (or simply sync pulses) to keep the transmitter and receiver in synchronism. These sync pulses help maintain the position of the pulses. The following figures explain the Pulse Position Modulation.

Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing of the pulse width modulated signal becomes the starting point for pulses in PPM signal. Hence, the position of these pulses is proportional to the width of the PWM pulses. In PPM, the amplitude and width of the pulses is kept constant but the position of each pulse is varied in accordance 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.5.4.1. Here, it may be noted that with increase in the modulating voltage the PPM pulses shift further with respect to reference. Figure 5.4.2 (a) and . Figure 5.4.2 (b) shows Constant Amplitude Periodic Pulse Train and Pulse Position Modulated Signal.

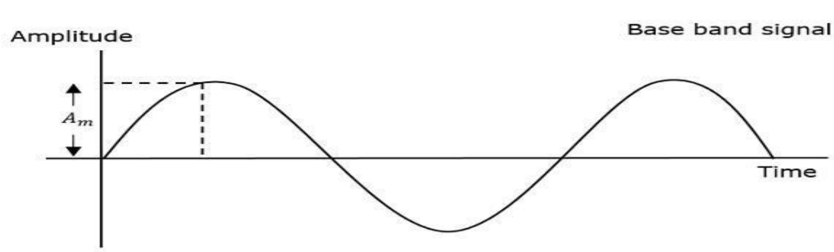


Figure 5.4.1 Base band Signal or Modulating Signal

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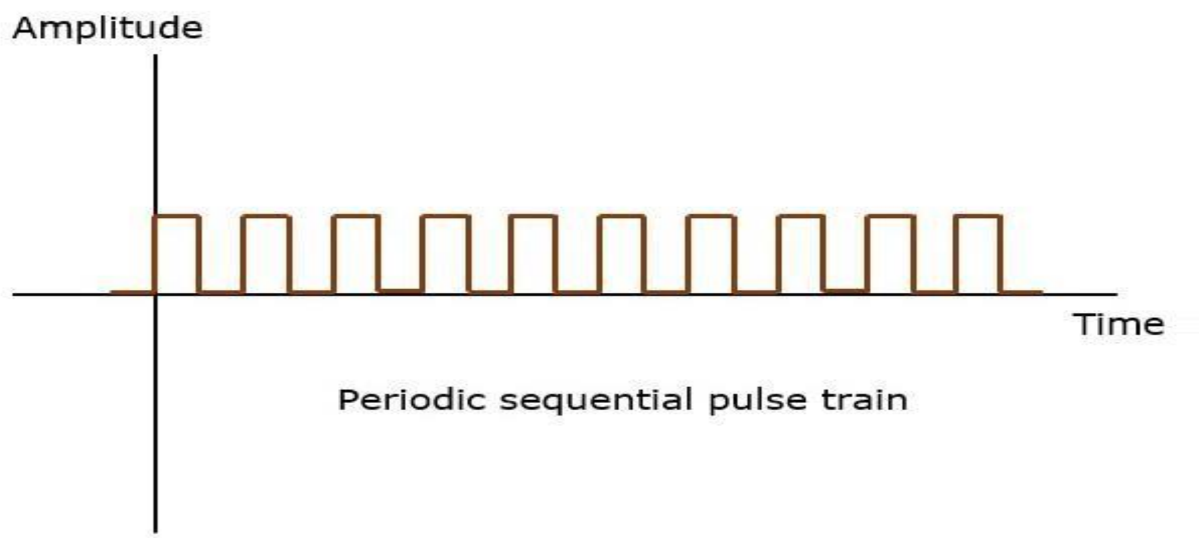

Figure 5.4.2 (a) Constant Amplitude Periodic Pulse Train

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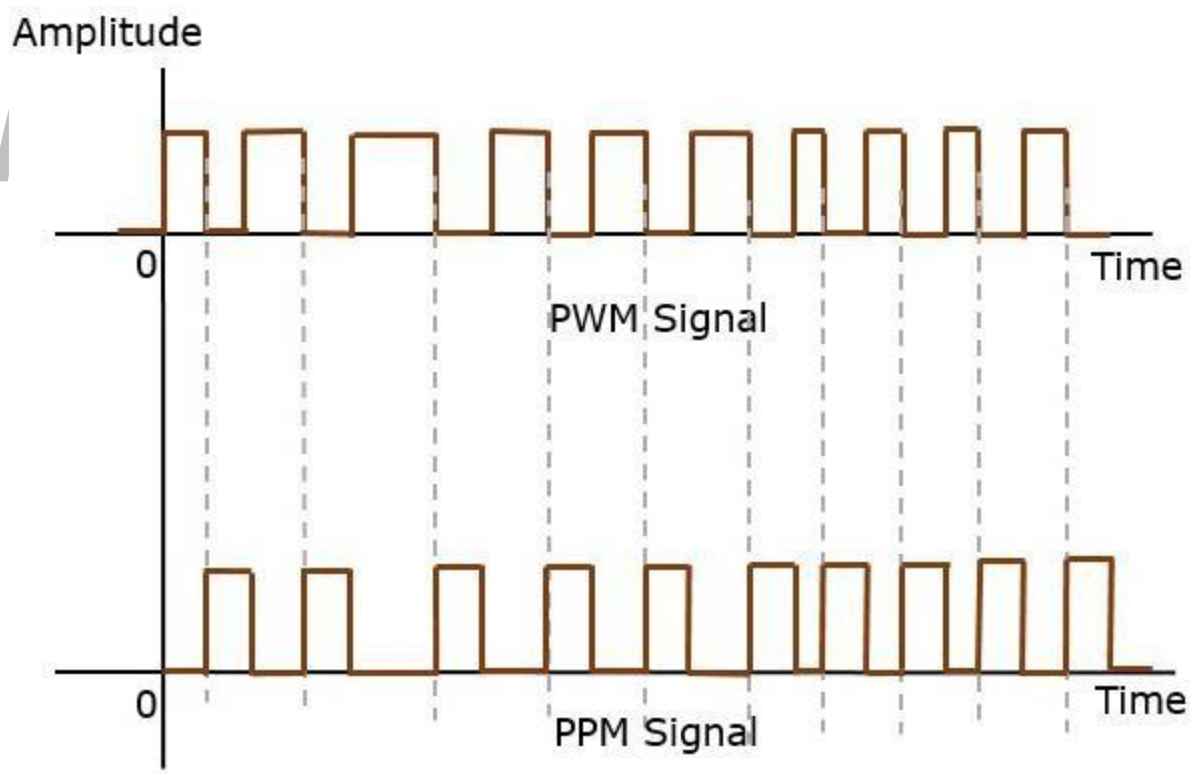

Figure 5.4.3 (b) Pulse Position Modulated Signal

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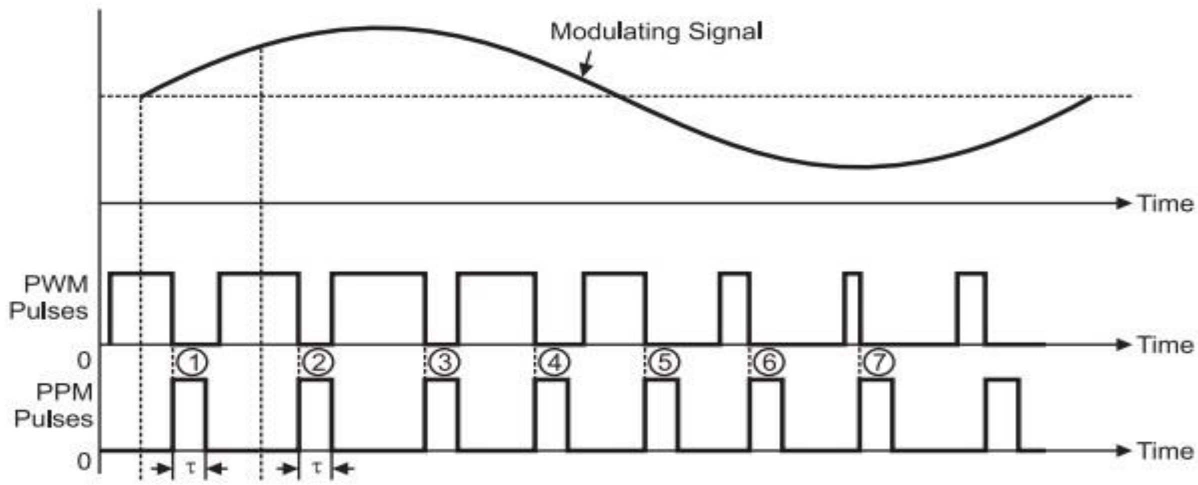

Figure 5.4.4 PPM pulses generated from PWM signal

Diagram Source Brain Kart

The vertical dotted lines drawn in fig.1 are treated as reference lines to measure the shift in position of PPM pulses. The PPM pulses marked 1, 2 and 3 in fig.5.4.3 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.

Generation of PPM Signal

The PPM signal can be generated from PWM signal as shown in fig.5.4.4 (a).

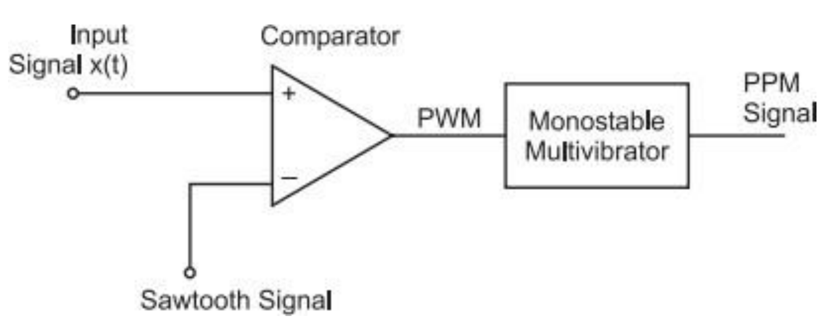

Figure 5.4.5 Generation of PPM signal

Diagram Source Brain Kart

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.5.4.5.

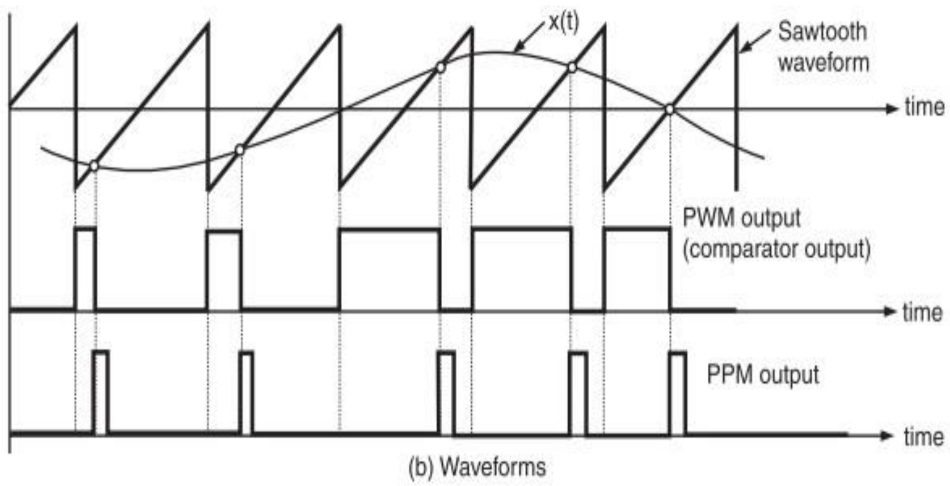

Figure 5.4.6 PPM from PWM Wave Forms

Diagram Source Brain Kart

Demodulation of PPM Signal

The PPM demodulator block diagram has been shown in fig.5.4.6

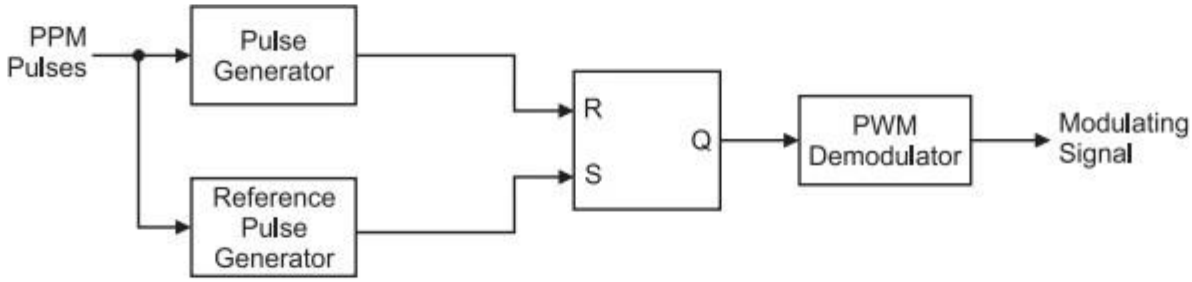

Figure 5.4.7 PPM Demodulator

Diagram Source Brain Kart

The operation of the demodulator circuit may be explained as under: The noise corrupted PPM waveform is received by the PPM demodulator circuit. The pulse generator develops a pulsed waveform at its output of fixed duration and applies these pulses to the reset pin (R) of a SR flip-flop. A fixed period reference pulse is generated from the incoming PPM waveform and the SR flip-flop is set by the reference pulses. 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.

Advantage

As the amplitude and width are constant, the power handled is also constant.

Disadvantage

The synchronization between transmitter and receiver is a must.

Pulse Width Modulation

Pulse Width Modulation (PWM) or Pulse Duration Modulation (PDM) or Pulse Time Modulation (PTM) is an analog modulating scheme in which the duration or width or time of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

The width of the pulse varies in this method, but the amplitude of the signal remains constant. Amplitude limiters are used to make the amplitude of the signal constant. These circuits clip off the amplitude, to a desired level and hence the noise is limited. The following figures explain the types of Pulse Width Modulations. Fig 5.4.1 (a), (b),(c) Pulse Width Modulated Waves with different time slots.

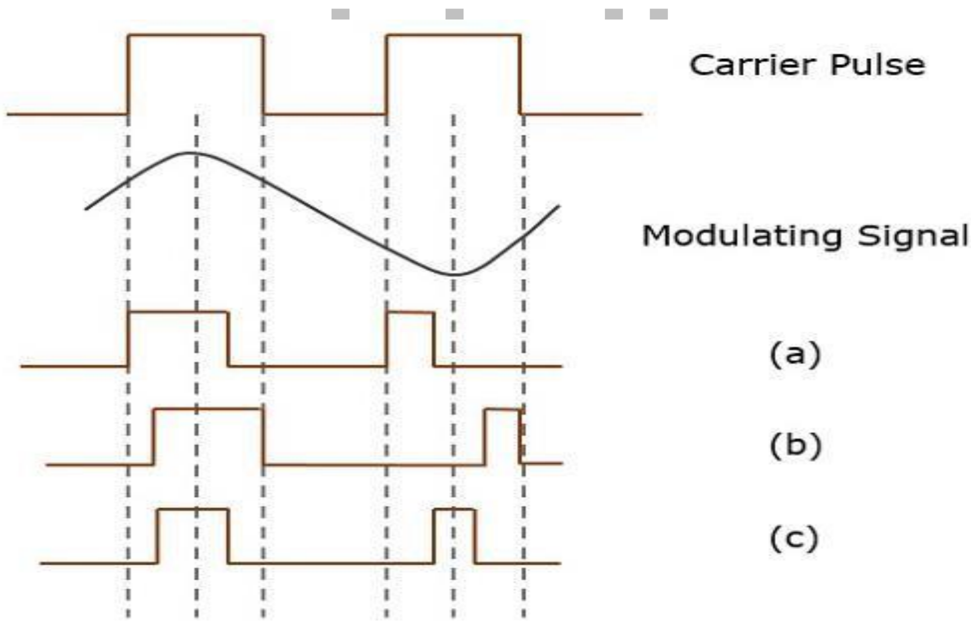

Figure 5.4.1 (a), (b),(c) Pulse Width Modulated Waves with different time slots

Diagram Source Brain Kart

There are three variations of PWM. They are −

The leading edge of the pulse being constant, the trailing edge varies according to the message signal. The trailing edge of the pulse being constant, the leading edge varies according to the message signal. The center of the pulse being constant, the leading edge and the trailing edge varies according to the message signal. These three types are shown in the above given figure, with timing slots.

Pulse Width Modulation (PWM) Waveform Representation

In PWM, the width of the modulated pulses varies in proportion with the amplitude of modulating signal. The waveforms of PWM is shown in fig.1 below.

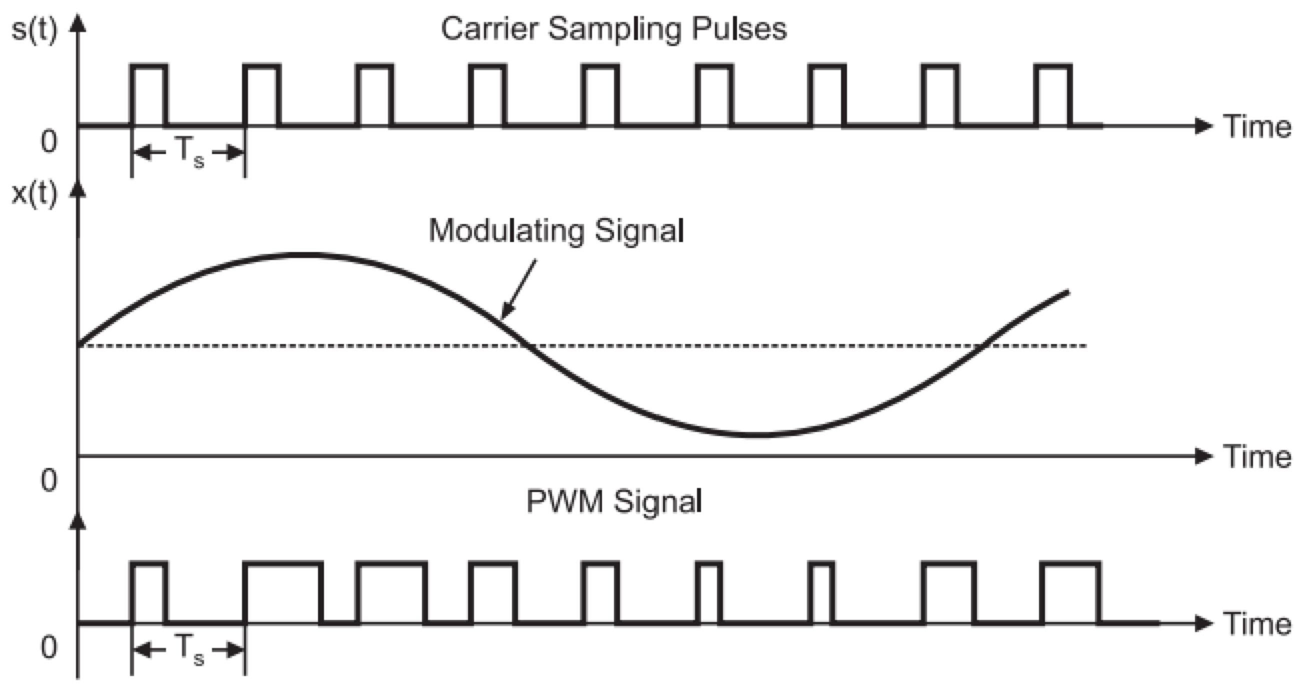

Figure.5.4.2 :Wave form Representations of PWM

Diagram Source Brain Kart

As we can observe, the amplitude and the frequency of the PWM wave remain 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.5.4.2 Wave form Representations of PWM.

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

Generation of PWM Signal

The block diagram of a PWM signal generator is shown in fig.5.4.3 below. This circuit can also be used for the generation of PPM signal.

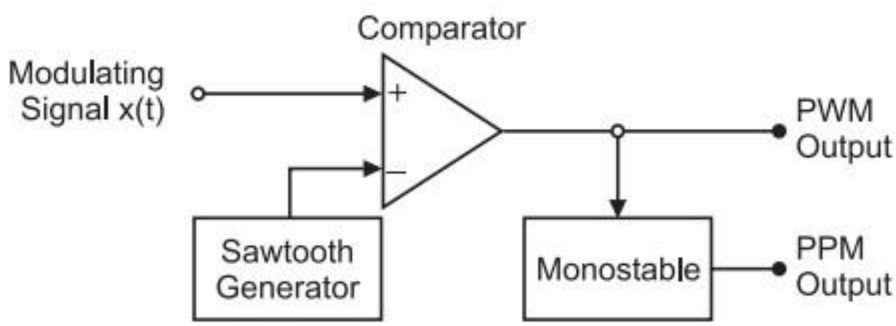

Figure.5.4.3 : PWM Generator

Diagram Source Brain Kart

A sawtooth generator generates a sawtooth signal of frequency fs, and this sawtooth signal in this case is used as 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. This gives rise to a PWM signal at the comparator output as shown in fig.5.4.4.

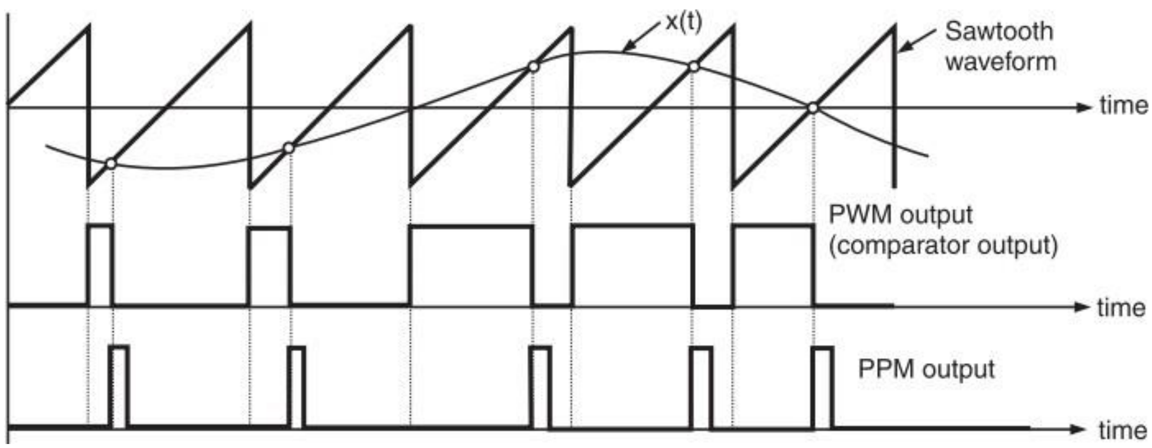

Figure.5.4.4: Waveforms of PWM and PPM

Diagram Source Brain Kart

Here, it may be noted 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 occurance 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.

Detection of PWM Signal

The circuit for the detection of PWM signal is shown in fig.5.4.5 below.

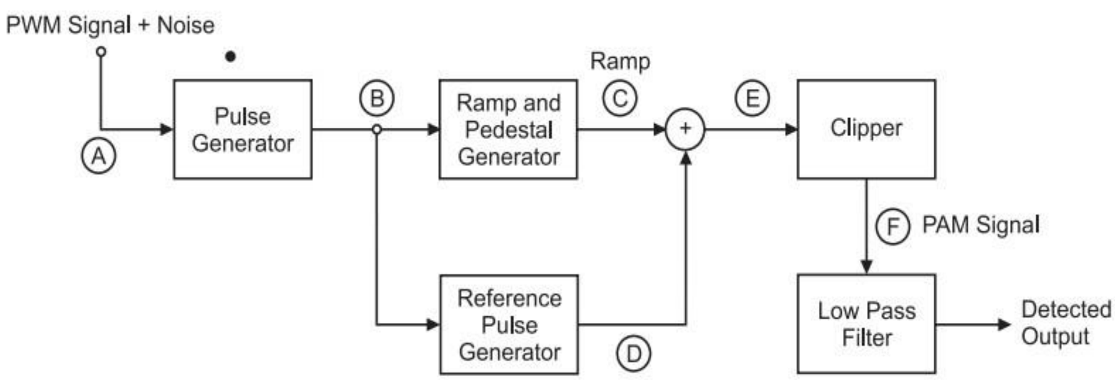

Figure.5.4.5 : PWM Detection Circuit

Diagram Source Brain Kart

The working operation of the circuit may be explained as under: 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 width 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 gtenerator are then added to the ramp signal. The output of the adder is then clipped off at a thereshold 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 have been shown in fig.5.4.6.

Advantages of PWM

Less effect of noise i.e., very good noise immunity. Synchronization between the transmitter and receiver is not essential (Which is essential in PPM).It is possible to reconstruct the PWM signal from a noise, contaminated PWM, as discussed in the detection circuit. Thus, it is possible to separate out signal from noise (which is not possible in PAM).

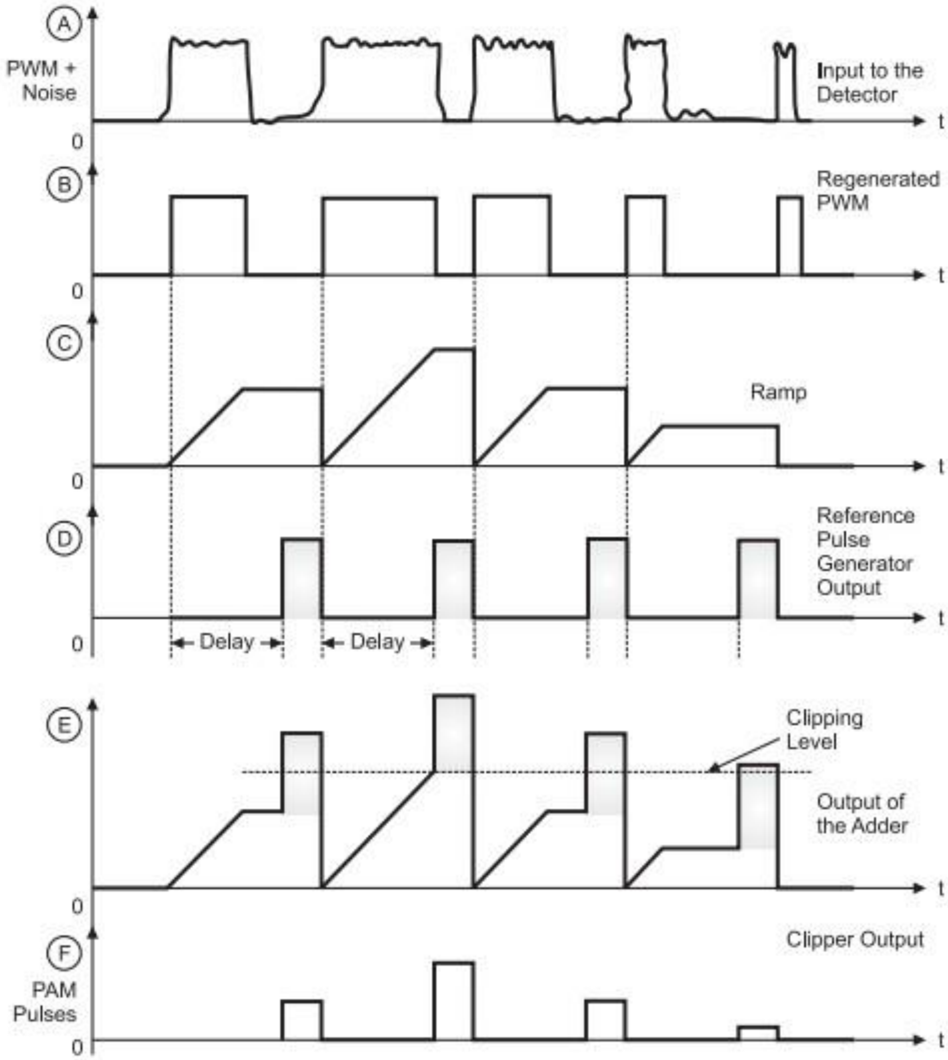

Figure.5.4.6 : Waveforms for PWM detection circuit

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Disadvantages of PWM

Due to the variable pulse width, the pulses have variable power contents. Hence, the transmission must be powerful enough to handle the maximum width, pulse, though the average power transmitted can be as low as 50% of this maximum power. In order to avoid any waveform distortion, the bandwidth required for the PWM communication is large as compared to bandwidth of PAM.

UNIT IV

QUANTIZATION

Quantization

The process of transforming Sampled amplitude values of a message signal into a discrete amplitude value is referred to as Quantization. The quantization Process has a two-fold effect:

1. The peak-to-peak range of the input sample values is subdivided into a finite set of decision levels or decision thresholds that are aligned with the risers of the staircase, and

2. The output is assigned a discrete value selected from a finite set of representation levels that are aligned with the treads of the staircase.

A Quantizer is memory less and the Quantizer output is determined only by the value of a corresponding input sample, independently of earlier analog samples applied to the input.

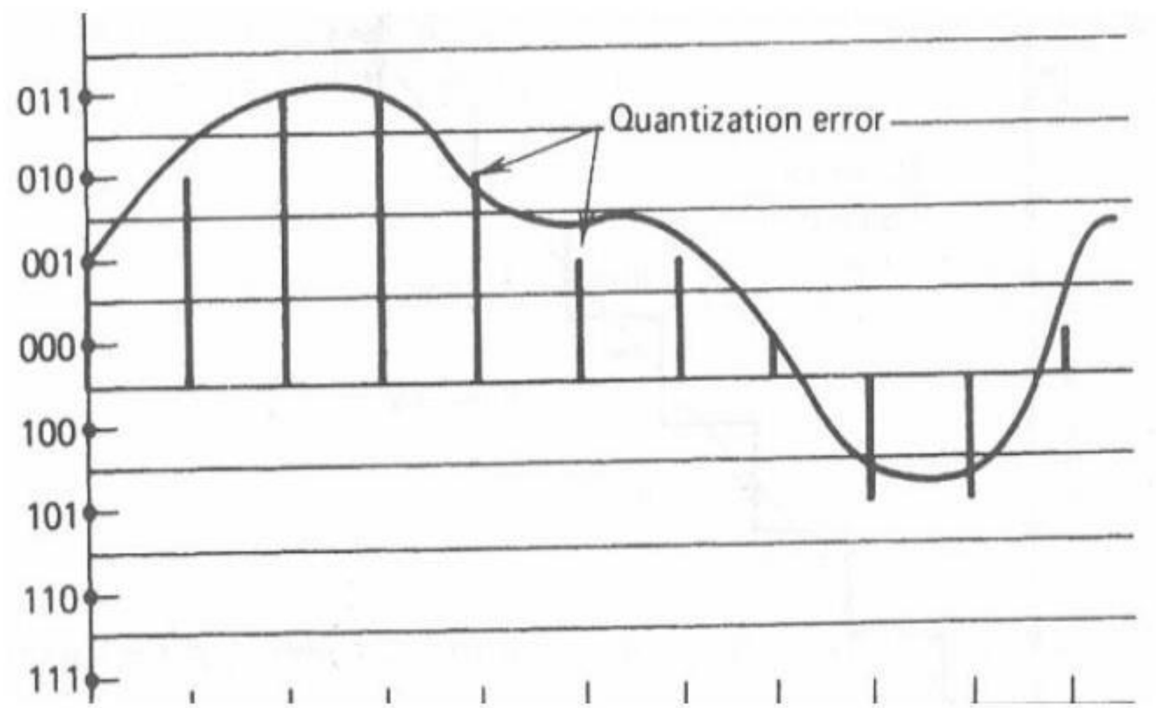

Fig 5.2.1 Quantization Process

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Types of Quantizers

1. Uniform Quantizer
2. Non- Uniform Quantizer

Uniform Quantizer: In Uniform type, the quantization levels are uniformly spaced, whereas in non-uniform type the spacing between the levels will be unequal and mostly the relation is logarithmic.

Types of Uniform Quantizers: (based on I/P - O/P Characteristics)

1. Mid-Rise type Quantizer
2. Mid-Tread type Quantizer

In the stair case like graph, the origin lies the middle of the tread portion in Mid –Tread type where as the origin lies in the middle of the rise portion in the Mid-Rise type.

Mid – tread type:Quantization levels – odd number.

Mid – Rise type: Quantization levels – even number.

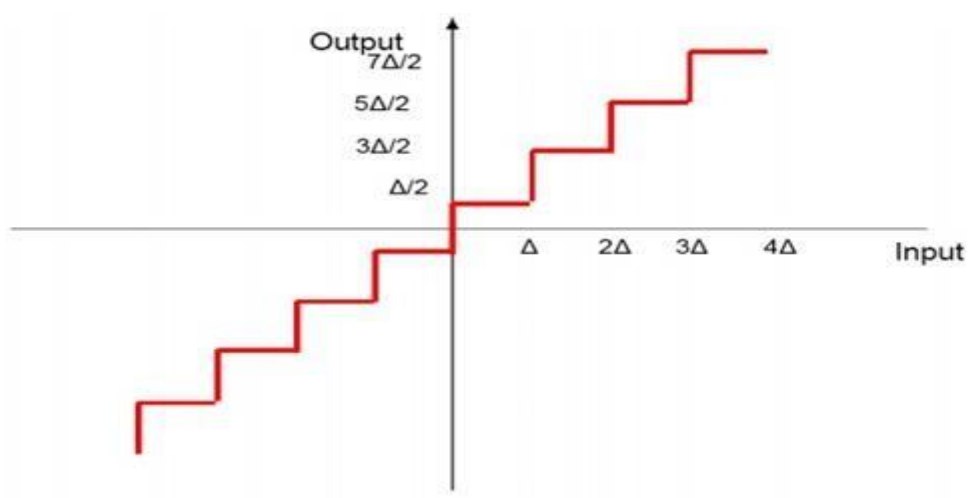

Figure 5.2.2 Input Output Characteristics of Mid Rise Quantizer

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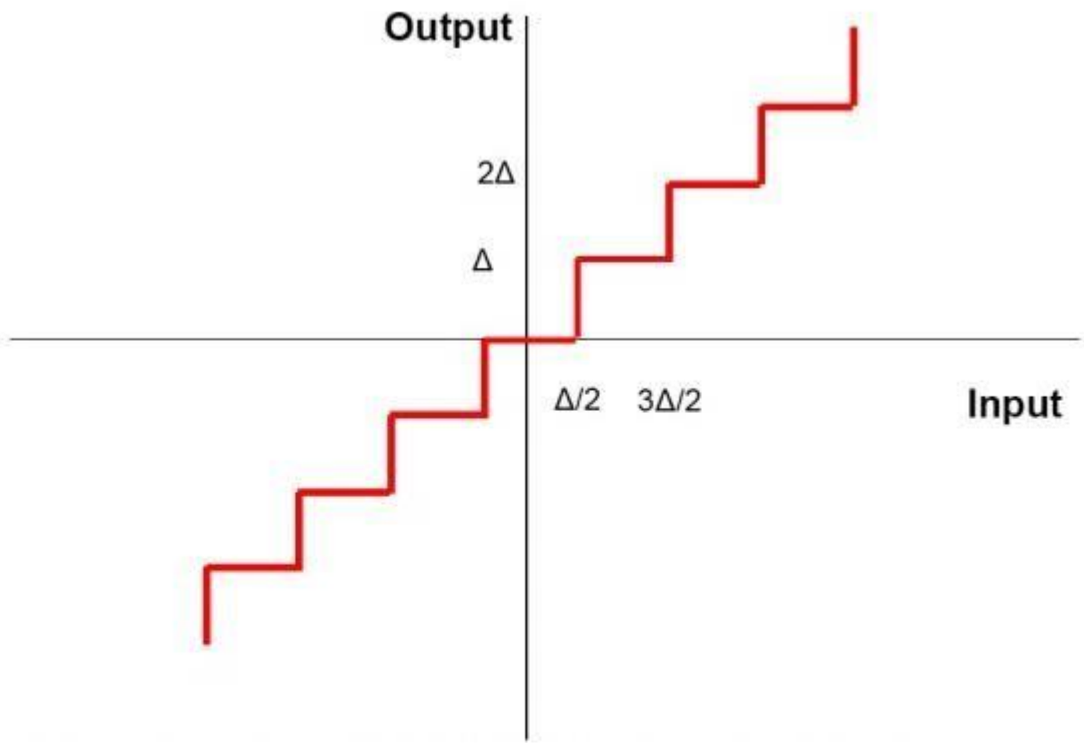

Figure 5.2.3 Input Output Characteristics of Mid Thread Quantizer

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Quantization Noise and Signal-to-Noise

"The Quantization process introduces an error defined as the difference between the input signal, x(t) and the output signal, yt). This error is called the Quantization Noise."

$$q(t) = x(t) – y(t)$$

Quantization noise is produced in the transmitter end of a PCM system by rounding off sample values of an analog base-band signal to the nearest permissible representation levels of the quantizer. As such quantization noise differs from channel noise in that it is signal dependent. Let "Δ" be the step size of a quantizer and L be the total number of quantization levels. Quantization levels are 0, ± ., ± 2 ., ±3 ....The Quantization error, Q is a random variable and will have its sample values bounded by [-(Δ/2) < q < (Δ/2)]. If is small, the quantization error can be assumed to a uniformly distributed random variable. Consider a memory less quantizer that is both uniform and symmetric.

L = Number of quantization levels

X = Quantizer input

Y = Quantizer output

The output y is given by

$$Y=Q(x)$$

which is a staircase function that befits the type of mid tread or mid riser quantizer of interest.

Companding of Speech signal

Compander = Compressor + Expander

In Non - Uniform Quantizer the step size varies. The use of a non – uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer. The resultant signal is then transmitted.

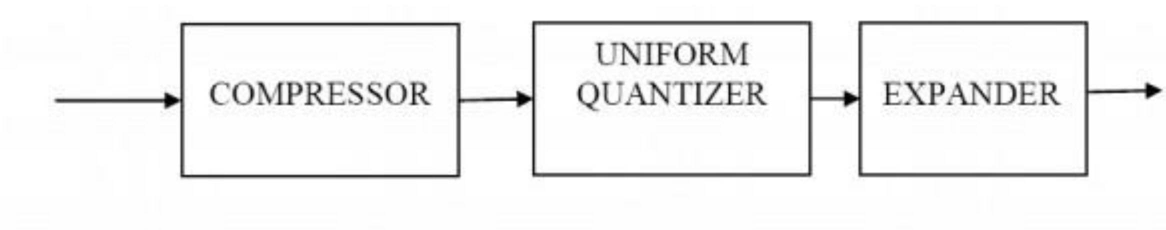

Figure 5.2.3 Model of Non Uniform Quantizer

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At the receiver, a device with a characteristic complementary to the compressor called Expander is used to restore the signal samples to their correct relative level. The Compressor and expander take together constitute a Compander.

- The quantization of the analog form of the signal to discrete form takes place in Quantizer. The sampled analog signal is still analog because though discrete in time the signal amplitude can take any value as it may wish

- The Quantizer forces the signal to takes same discrete values form the continuous amplitude values.
- From the sampled signal $m_s(t)$ new quantized signal $m_q(t)$ is created .
- Whereas it can take any value ,but $m_q(t)$ can take only L discrete values.

$$Vpp = L * q$$

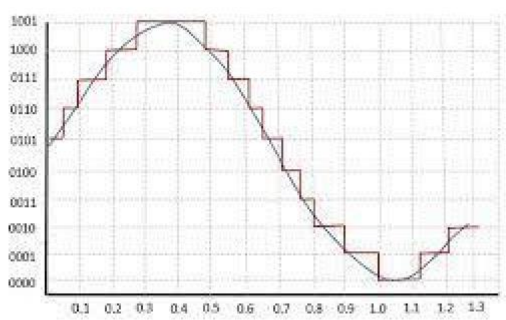

Figure 5.2.4 a Analog signal Quantization

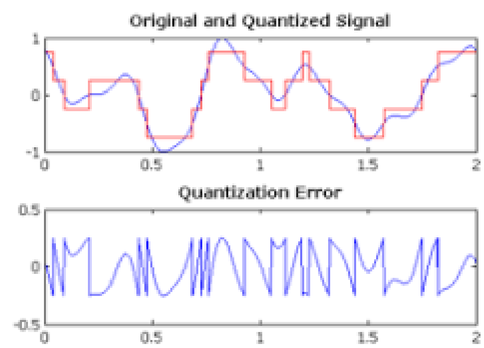

Figure 5.2.4 b Quantization Error

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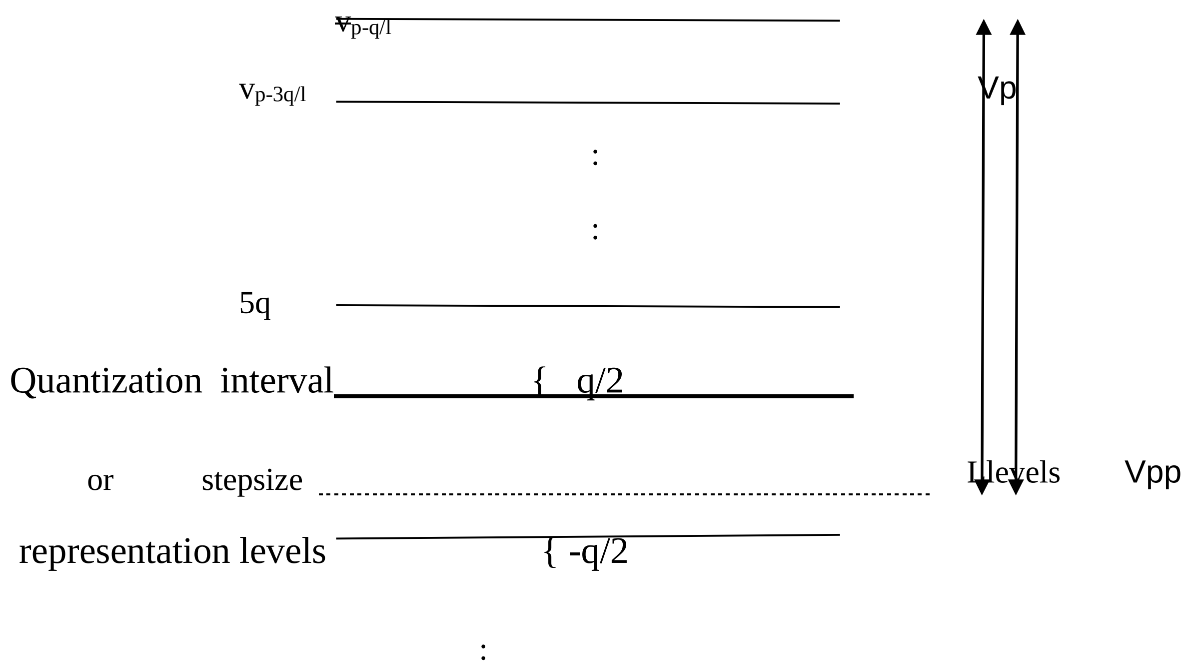

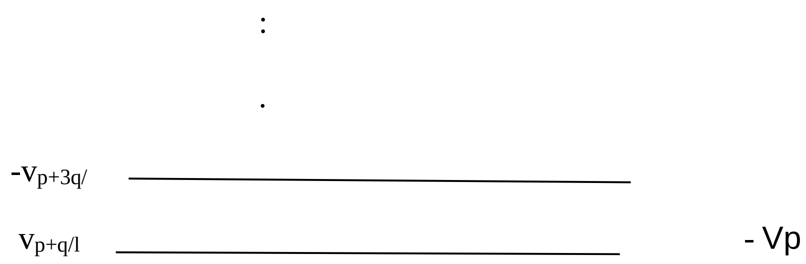

Figure 5.2.5 Signal Quantization with representation levels

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Figure shows the L level linear Quantizer for analog sampled signal $m_s(t)$ for an analog sampled signal $m_s(t)$, Vpp -→ peak to peak voltage range

$$Vpp=Vp-(-Vp)$$

$$Vpp=2Vp \text{ volts}$$

The quantized levels levels may assume positive and negative values. The step size between quantization levels are called quantized interval denoted q volts . If the quantization levels are uniformly distributed over the full range ten the Quantizer is called a uniform Quantizer each sample value of the analog waveform is approximated to a Quantizer level.

This approximation process gives rise to an error quantization noise, given by

$$e_q(t)=m(t)-m_q(t)$$

maximum error in quantization =1/2 q ie)half of the quantile interval ie) ±q/2 volts, let f(m) be the pdf of the Quantizer input signal m(t) in the range ±vp the mean1/square error

$$eq^2 = \int_{m1-q/2}^{m1+q2} f(m)(m-m1)^2 + \int_{m2-q/2}^{m2+q2} f(m)(m-m2)^2 + \ldots\ldots\ldots \int_{mL-q/2}^{mL+q2} f(m)(m_L)^2$$

For large L the quantile interval q becomes quite small .so that assume f(m) to be constant in an interval .without committing any significant error, this constant value may be taken to be the value of f(m) at the quantized values mi in the I the interval

Hence the noise expression becomes,

$$eq^2 = (f^I + f^{II} + f^L) \int_{-q/2}^{q/2} x^2 dx$$

$$\overline{eq^2} = (f^I + f^{II} + \ldots.. f^L) q^3 / 12$$

$$\overline{eq^2} = (f^I + f^{II} + \ldots.. f^L) q^2 / 12$$

Now are the respective probabilities that when m(t) at the I st , IInd ………… L th quantile interval 1/3 For large L the RMS inside the bracket approximately gives the area under the PDF, curve ,So

$$(f_q^I + f_q^{II} + \ldots.. f_q^L) = 1$$

So the value the quantization noise power becomes

$$\overline{eq^2} = q^2 / 12$$

So it is clear that quantization noise increases with increase in the size of quantile interval.

The uniform quantization in quantile interval is given by

□ = □□□/□ − 1

In most practical Quantizer L is indeed large. so e increase no of levels, so the quantization noise decreases. The peak signal power of the unquantised signal is

□□^{2} = (□□□/2)^2)

Noise power=□^{2}/12

Noise power=signal power/noise power

□□□ = 3□^{2}

Non uniform quantization

The step size is not fixed .it values according to the input amplitude

Case 1

The step size is small at low input signal levels. Hence the quantization error is also small the quantization noise power ratio is also improved at low signal levels.

Case 2

The step size is higher at high input signal levels .here the signal to noise power ratio remains almost same throughout the range of quantizer.

Necessity of non-uniform quantization

In uniform quantization the step size remains same throughout the range of Quantizer

For low signal amplitude the maximum quantization error is quite high ,but for high signal amplitude the maximum quantization error is small. This problem arises because of uniform quantization .,

Speech and music signals are characterized by large crest factor .ie) for such signal the ratio of peak to RMS value is very high,

Crest factor =Peak value/RMS value

Quantization noise is directly related to step size

P-→normalised signal power

if we normalize the signal x(t) then xmax-=1

crest factor $=1/\sqrt{P}$

The round of process in the quantization introduces an error. The difference between input signal m(t) and output signal mq(t) is called as quantization error.

$$q(t) = m(t) - m_q(t)$$

Consider an input m of continuous amplitude in the range of $m_{max} - (-m_{max}) = 2m_{max}$ volts.

The uniform Quantizer of the mid riser type

The total amplitude range$= m_{max} - (-m_{max}) = 2m_{max}$

The step size $\Delta = 2m_{max}/L$

If Δ is small the number of representation levels 'L' is sufficiently large

The probability during function of the quantization error Q as

$$f_Q(q) = \begin{cases} 1/\Delta & -\frac{\Delta}{2} < q < \frac{\Delta}{2} \\ 0 & \text{otherwise} \end{cases}$$

The variance $\sigma_Q^2 = E[Q^2] = 1/\Delta \int_{-\Delta/2}^{\Delta/2} q^2 dq$

$$= 1/\Delta \left[q^3/3\right]_{-\Delta/2}^{\Delta/2}$$

$$= \Delta^2/12$$

Let R denote number of bits per sample usd as the construction of binary code.

$$L = 2^R$$

The peak signal power

$$m_{max}^2 = \Delta^2 L^2/4$$

Companding

The crest factor can be given by

□□□ = □□□□ □□□□□/□□□ □□□□□

Speech signal has high crest value. This implies that the speech signal are having amplitude values near zero. So if uniform Quantizer are used for speech most of the time quantizes will be more Than the signal slope This is known as granular noise One solution is tapering quantizing levels.ie) step size can be made smaller signals and larger for larger signals But this type of Quantizer with varying step is practically difficult to implement.

A most practical approach is to predictor the signal by a logarithmic compression and then put into an uniform Quantizer. This compressed and quantized signal is transmitted through the channel and then can be undistorted at the receiver by the logarithmic compression algorithm This process is known as companding A device which is used for companding is known as compander In a compander the true function has a larger slope for small signals and smaller slope for larger signals. Thus a given signal change at small magnitudes will carry uniform Quantizer through more steps. After compression the distorted signal is used as a input of a linear [uniform Quantizer] At the receiver an inverse of compression called expansion is applied so that over all transmission is not distorted This processing pair(compression and expansion)is called Companding. It is similar to pre emphasis and de emphasis in FM. This in amplitude domain and in frequency domain.

Advantages of Non- Uniform Quantization

1. Higher average signal to quantization noise power ratio than the uniform Quantizer when the signal pdf is non uniform which is the case in many practical situation.
2. RMS value of the Quantizer noise power of a non – uniform Quantizer is substantially proportional to the sampled value and hence the effect of the Quantizer noise is reduced.

UNIT IV
SAMPLING

Introduction

In digital communication system input signals should be in digital form so that DSP can be employed on the signals. Electrical signal at the output of the transducer needs to be converted into sequential of digital signals. The block performing this task is formatter. To represent digital signals into few digits as possible, a coding system can be employed. This minimizes the number of digits. This process is called source coding and the block is called source encoder. This block compresses or minimizes the number of the digits for signals transmission. To reduce the noise in the communication cannel some redundancy bits is added with the message. This is done by channel encoder.

Base band processor

The channel encoded signals are not modulated. This transmission takes place in base band. For proper detection at the receiver and to reduce noise some line coding is used. Some pulse shaping is also done. Some special filter are also used to combat noise. All these are collectively called as baseband processor. This is in case of low speed wired transmission fixed telephony. For high speed data the digital signals to be modulated.

Band pass processor

The primary purpose of this band pass modulator is to map the digital signals into high frequency analog signals. Efficiency (spectral efficiency) is calculated according to number of bits send per second. The two block are (BASEBAND/BANDPASS) are mutually exclusive blocks. In the communication channel the transmitted signals get corrupted by random noise .the noise is from various sources .Thermal noise, shot noise, electro magnetic interference .At the receiver band pass modulator block processes the transmitted (corrupted)waveform and maps them back to sequence of number. In case of baseband the task of converting back the line coded pulse waveform to transmitted data sequence is carried out by baseband decoder block.

Channel decoder

Its used to reconstruct to the original sequence from the channel encoded digital by removing the extra added redundancy bits.

Source decoder

This estimates the digital signal.. In the De formatter if the original information source was not in digital data form ,this block needed to convert back the digital data to discrete or analog form. The Output transducer Converts the estimate of digital signal to analog signal.

Sampling

A message signal may originate from a digital or analog source. If the message signal is analog in nature, then it has to be converted into digital form before it can transmit by digital means. The process by which the continuous-time signal is converted into a discrete–time signal is called Sampling. Sampling operation is performed in accordance with the sampling theorem.

Sampling Theorem For Low-Pass Signals

Statement: - "If a band –limited signal g(t) contains no frequency components for $|f| > W$, then it is completely described by instantaneous values g(kTs) uniformly spaced in time with period $T_s \leq 1/2W$. If the sampling rate, fs is equal to the Nyquist rate or greater ($f_s \geq 2W$), the signal g(t) can be exactly reconstructed.

Formatting can be done in 3 steps

Sampling-------- > discretization in time

Quantization--------- >discretization in amplitude

Encoding-------- >obtaining quantized value

Sampling discretizes an analog signal in time domain .hence instead of continuous time wave form we get continuous values of signal at discrete point of time,

Analog to Digital Conversion

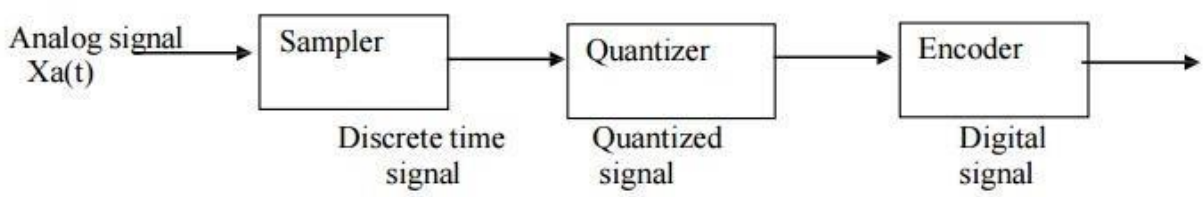

Figure 5.1.1 Analog to Digital Conversion

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Sampling theorem Definition

A bandlimited signal having no spectral components above fm Hz can be determined uniquely by values sampled at uniform intervals of $T_s \leq 1/2f_m$ sec

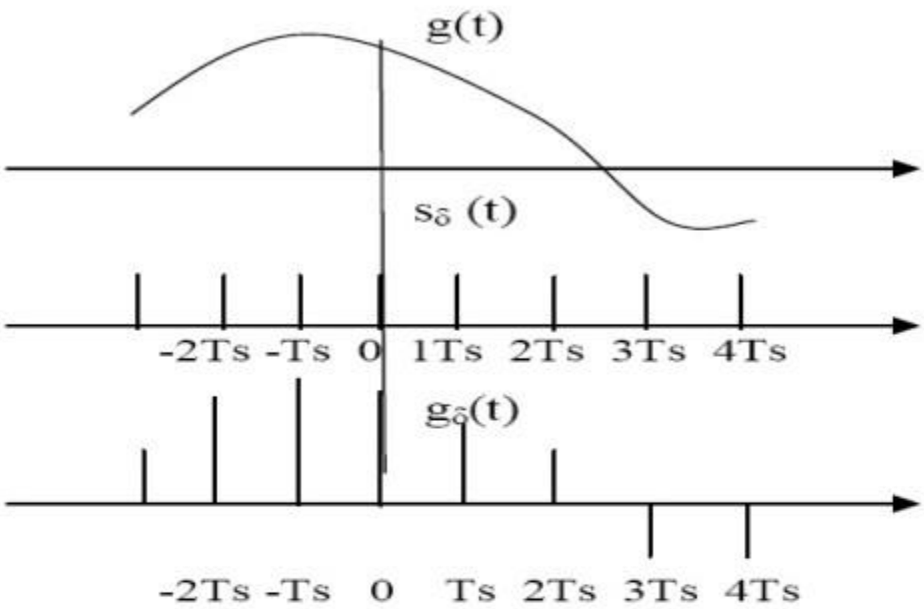

Figure 5.1.2 Sampling Process

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Nyquist rate

Minimum sampling frequency needed to reconstruct the analog signal from sampled waveform

$$fs \leq 2\, fm$$

Impulse sampling

Let as sample an analog waveform x(t) by a sequence of unit impulses (δfn) Assume that the spectrum of x(t) is band limited ,it is zero outside the interval -fm<f<fm

We sample x(t) at times t=nTs by aperiodic train of delta ($X\delta(t)\, fn$) can be represented as

$$X\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nTs)$$

Where Ts =1/2fm-----> to satisfy Nyquist criterion

The Fourier transform of the impulse train $X\delta(f) = 1/Ts \sum_{n=-\infty}^{\infty} \delta(t - nfs)$,Fs = 1/Ts

The sampling can be modeled mathematically as a product of $X(t)$ with $X\delta(t)$

$$Xs(t) = X(t)X\delta(t)$$

$$= \sum_{n=-\infty}^{\infty} X(t)\delta(t - nTs)$$

$$== \sum_{n=-\infty}^{\infty} X(nTs)\delta(t - nTs)$$

Now the spectrum of the sampled signal $X\delta(f)$ is

$$Xs(f) = X(f)X\delta(f)$$

$$= X\delta(f) * 1/Ts \sum_{n=-\infty}^{\infty} \delta(t - nfs)$$

$$= 1/Ts \sum_{n=-\infty}^{\infty} X(f) * \delta(t - nfs)$$

$$= 1/Ts \sum_{n=-\infty}^{\infty} X(t - nfs)$$

The sampled function has the same spectrum within a constant factor 1/Ts. This is within same as that of the spectrum of the input band -fm<f<fm. This spectrum repeats periodically in frequency every fs Hz. When the sampling rate just satisfies Nyquist rate just satisfies Nyquist rate fs=2fm,the upper end of each replicate touches the lower end of the higher band neighbor replicate .Here the extraction of the original waveform is possible without filtering. In practical digital system fs always>Nyquist rate for filtering rate

Natural sampling

Uses flat –top rectangular pulses of finite width to sample analog waveform. This is called natural sampling, because top of the each pulse in the sampled sequence remains the shape of the original signal during that pulse interval .

Note the flat –top pulse train of width T as $Xp(t)$

$$Xp(t) = \sum_{n=-\infty}^{\infty} 1/Ts \prod t - nTs/Ts$$,Ts =1/2fm

The flat top pulse train is a periodic waveform .In Fourier series it is represented by

$$Xp(t) = \sum_{n=-\infty}^{\infty} Cn e^{j2\pi fst}$$

The Fourier coefficient Cn of the pulse train is Cn =1/Ts sinc (nTs/Ts) . The magnitude of the pulse train has the train has the character of the sinc shape.

The sampled waveform can be written as

$$Xp(t) = X(t) \sum_{n=-\infty}^{\infty} Cn e^{j2\pi fst}$$

The transform $Xs(f)$ of the sampled waveform is given by

$$Xp(t) = X(t) \sum_{n=-\infty}^{\infty} F\{X(t) Cn e^{j2\pi fst}\}$$

Using the frequency translation property of F.T

$$Xs(f) = Cn \sum_{n=-\infty}^{\infty} X(t - nfs)$$

The spectrum of natural sampled wave form is a replication of $X(f)$ periodically repeated in every frequency fs H. This is similar to impulse sampling .One difference the spectrum is weighted by Fourier series coefficient of the pulse train is compared to constant value of the impulse samples for natural sampling the pulse width should be reduced. It can be shown calculating Cn value for flat –top pulses with pulse width Ts approaching zero.

$$Cn = 1/Ts \int_{-Ts/2}^{Ts/2} X_{\delta}(t)\, e^{j2\pi fst} dt$$

With ih the range of integration Ts/2 to Ts/2 the only conversion of $X\delta(f)$

$$Cn = 1/Ts \int_{-Ts/2}^{Ts/2} \delta(t)\, e^{j2\pi fst} dt = 1$$

The spectrum of flat pulse approaches the spectrum of impulse samples for Ts-→0

Sampler implementation

Sampler implementation is done by sample and hold circuit In this circuit a Switch and store mechanism is used to form a sequences of samples These samples are analog waveform. They look like PAM, the amplitude of the pulses vary continuously recovery. The original analog waveform can be recovered from these PAM samples by low pass filtering technique.

If fs<fnyquist =>under sample then aliasing occurs

Aliasing =>overlapping of adjacent spectrum replicates if fs<fnyquist

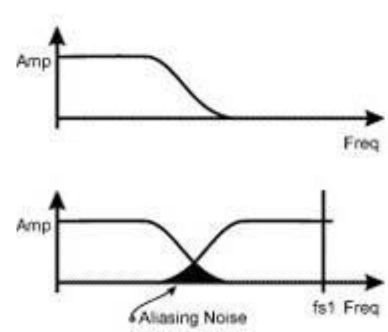

Figure 5.1.3 Aliased spectral components

Diagram Source Brain Kart

This aliased spectral components represent data appear in the frequency band fs-fm .Due to the practical difficulties in achieving Nyquist rate of sampling

,sometimes under sampling is done There are two ways to avoid aliasing both methods using antialiasing filter

The two methods are

i)pre filtering antialiasing

ii)post filtering

i)Pre filtering

Here the analog signal itself is pre filtered so that the new maximum frequency is fm

fm is reduced to fs/2 or less. Therefore sampled spectrum does not overlap

ii)Post filtering

In the method aliasing terms are eliminated after sampling wit help of low pass filter

the cut off frequency fm" needs to be less than fs-fm

Pre filtering in spectral domain

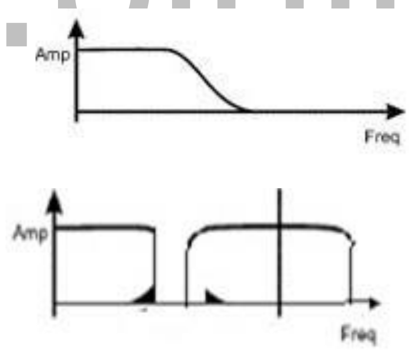

Figure 5.1.4 Pre filtering in spectral domain

Diagram Source Brain Kart

The antialiasing filter are commonly analog filters. There is alternate to them. We can oversample the signal there by removing antialiasing. The large number of samples can be filtered by digital filters instead of analog filter . This is the economic solution for sampling or the economy here A/D converter are required.

Without oversampling

In this method the signal passes through a high performance analog low pass antialiasing filter to limit its bandwidth. The antialiasing filter has a passband equal to the signal bandwidth plus the transition bandwidth. For transition bandwidth of f_T the Nyquist sampling rate $2f_m$ becomes $2f_m+f_T$. The additional spectral interval does not represent any useful signal content . It protect the signal by reserving free spectral interval between two spectral replicas. The filtered signal is sampled at Nyquist rate for the approximately Band limited signal. The samples are processed by D/A converter that's maps the continuous valued samples to a finite list of discrete output levels.

Post filtering

The disadvantages of both these antialiasing filter is same information always lost due to filtering. All realizable filters requires a nonzero bandwidth for transmission between the passband and stop band. This is known as transition bandwidth. Filter complexity and cost rise sharply with narrower transition band width.

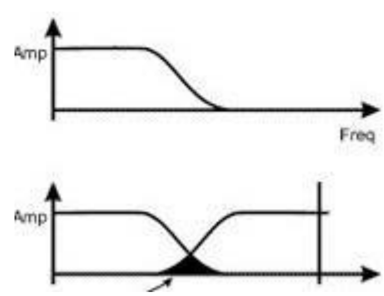

Figure 5.1.5 Post filtering for Sampling

Diagram Source Brain Kart

If we want to keep the sample rate down we should go for narrower transition bandwidth trade off is required between the cost of small transition bandwidth and cost of higher sampling rate .If we have 20%transitiuon B.W of the antialiasing filter, we have a practical Nyquist sampling rate.

$$fs \geq 2.2\ fm$$

With over sampling

Step:1

The signal is passed through a low performance analog LPF . Prefilter to limit its bandwidth. So bandwidth is reduced.

Steps:2

The prefilterd signal is over sampled at a much higher sampling rate higher than the Nyquist rate for the bandlimited signal.

Steps:3

The samples are processed by an A/D converter that maps continuous valued samples to a finite list of discrete output levels.

Steps:4

The digital samples bare then processed by a high performance a low cost digital filter to reduce the bandwidth of the digital samples

Steps:5

The sample rate at the output of the digital filter is reduced in proportion to the and width reduction obtained by the digital filter.

Cloud digital signal; processing techniques combine the filtering and resampling into a single structure .they also compensate the distortion in the first level and improve the signal quality.

Aliasing and signal reconstruction:

Nyquist's theorems as stated above and also helps to appreciate their practical implications. Let us note that while writing Eq.(1.4), we assumed that x(t) is an energy signal so that its Fourier transform exists. With this setting, if we assume that x(t) has no appreciable frequency component greater than W Hz and if fs> 2W, then

Eq.(1.4) implies that Xs(f), the Fourier Transform of the sampled signal Xs(t) consists of infinite number of replicas of X(f), centered at discrete frequencies n.fs, $-\infty < n < \infty$ and scaled by a constant fs= 1/Ts.

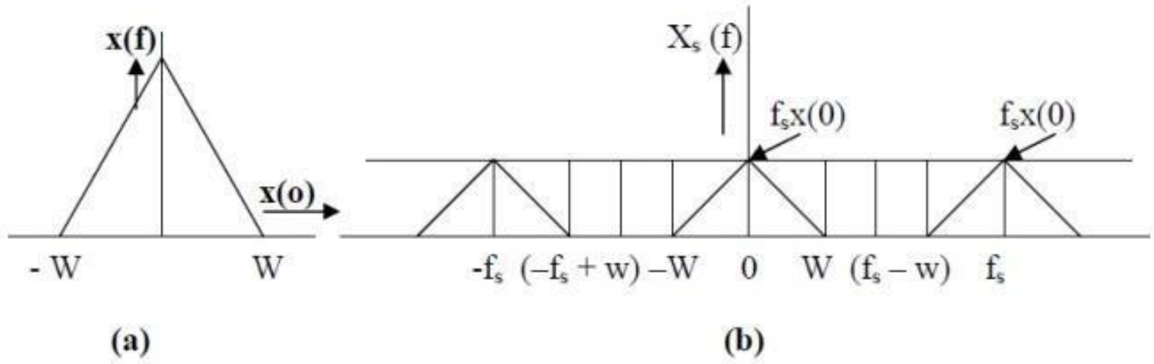

Figure 5.1.6 Spectra of Analog signal and its sampled Signal

Diagram Source Brain Kart

Fig. 5.5 indicates that the bandwidth of this instantaneously sampled wave xs(t) is infinite while the spectrum of x(t) appears in a periodic manner, centered at discrete frequency values n.fs. Part – I of the sampling theorem is about the condition fs> 2.W i.e. (fs – W) > W and (– fs + W) < – W. As seen from Fig. 1.2.1, when this condition is satisfied, the spectra of xs(t), centered at f = 0 and f = ± fs do not overlap and hence, the spectrum of x(t) is present in xs(t) without any distortion. This implies that xs(t), the appropriately sampled version of x(t), contains all information about x(t) and thus represents x(t).

The second part suggests a method of recovering x(t) from its sampled version xs(t) by using an ideal lowpass filter. As indicated by dotted lines in Fig. 1.2.1, an ideal lowpass filter (with brick-wall type response) with a bandwidth $W \leq B < (fs – W)$, when fed with xs(t), will allow the portion of Xs(f), centered at f = 0 and will reject all its replicas at f = n fs, for $n \neq 0$. This implies that the shape of the continuous time signal xs(t), will be retained at the output of the ideal filter.