ANALOG COMMUNICATION
(204189)

PULSE ANALOG MODULATION

Unit 6

Objectives

- Builds the concepts of Pulse Analog Modulation.
- Understand the difference between analog modulation and pulse analog modulation.
- Figures out building a time division multiplexing signal.
- Introduces Sampling and hence reconstruction.
- Familiarizes the words such as Nyquist rate, Nyquist interval, and states the Nyquist Theorem.
- Construction of different types of Pulse Modulated signals.
- Presents Pulse Code Modulation technology and
- Links pulse analog communication and digital communication.

Books

1. **Communication Systemes**

   Simon Haykin

   4th Edition

   Wiley Publications

2. **Principles of Communication Systems**

   Herbert Taub

   Donald Schilling

   Goutam Saha

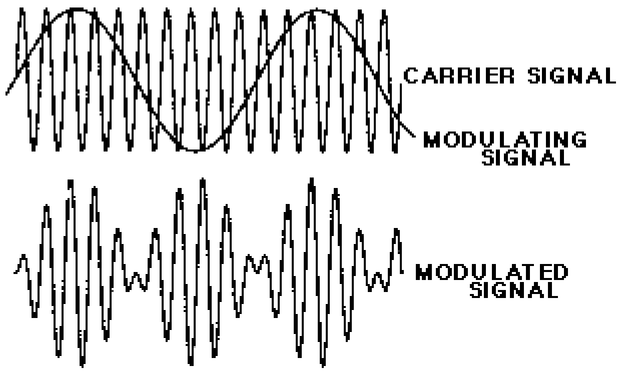
   Third Edition

   Tata-McGraw-Hill Publications

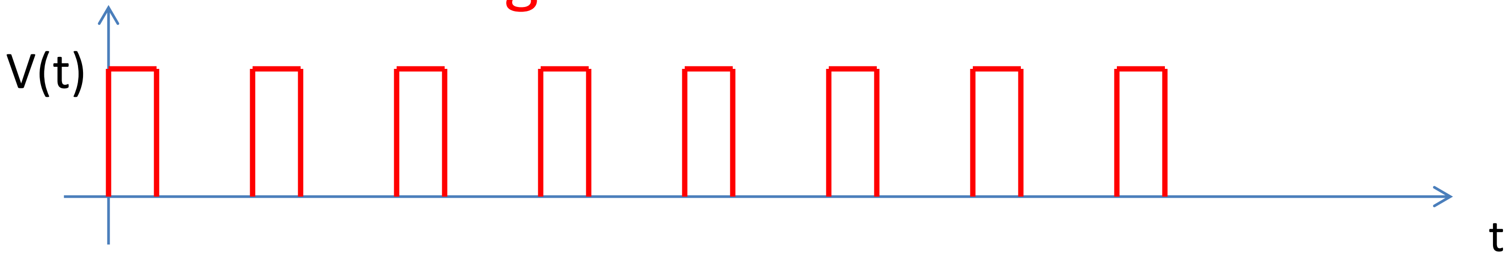
NPTEL Lecture Series

Introduction(1)

- All the Modulation schemes discussed so far – has carrier – a sinusoidal signal.



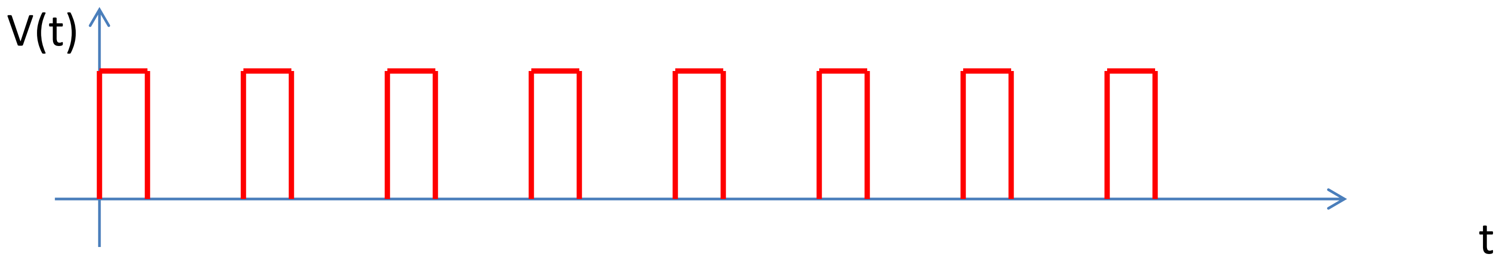
- A non-sinusoidal signal is used as the carrier. That is a Pulse Signal.



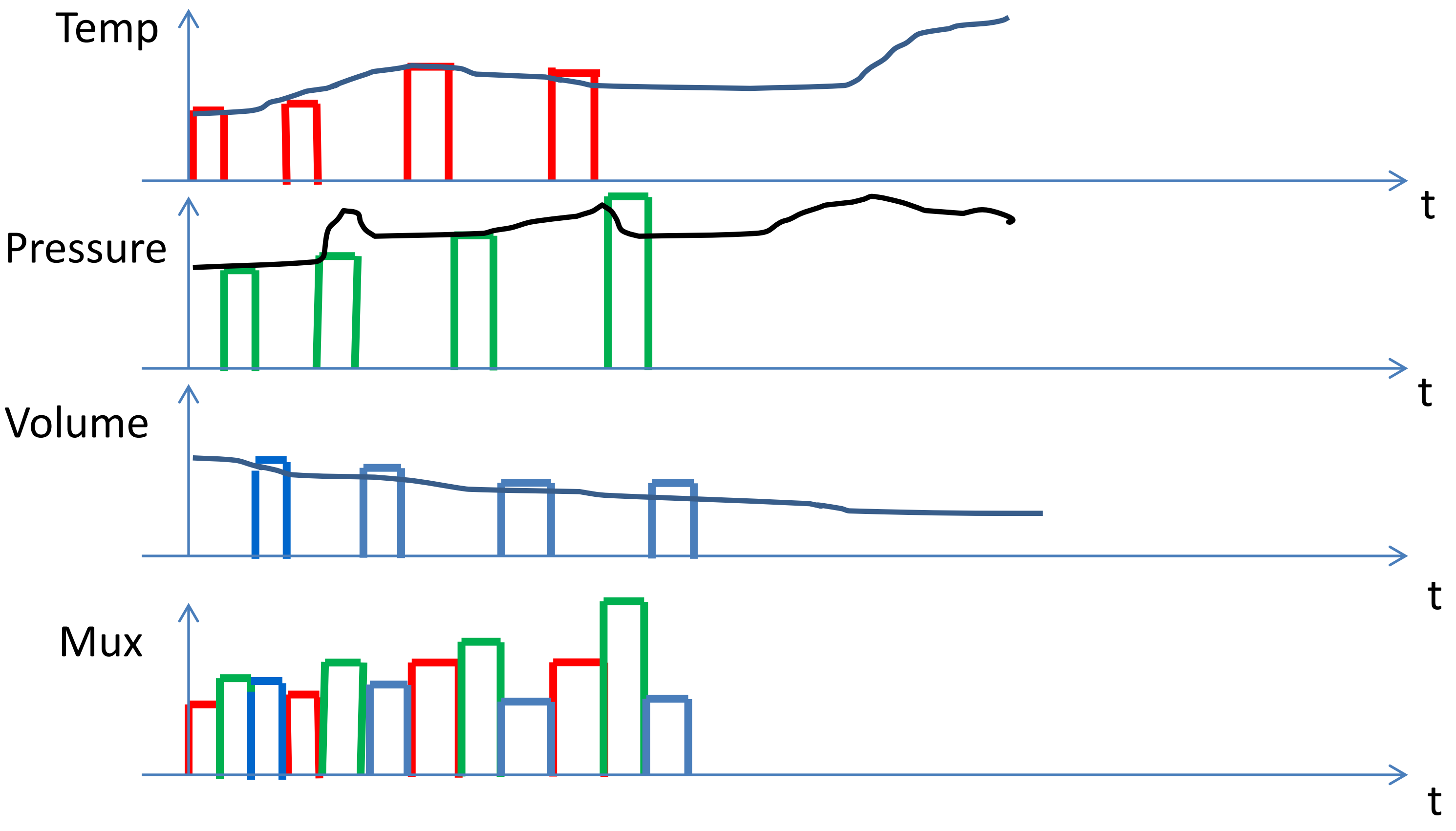
Introduction(2)

- The Pulses carry the information present in the message signal.

- Advantages



Introduction(3)



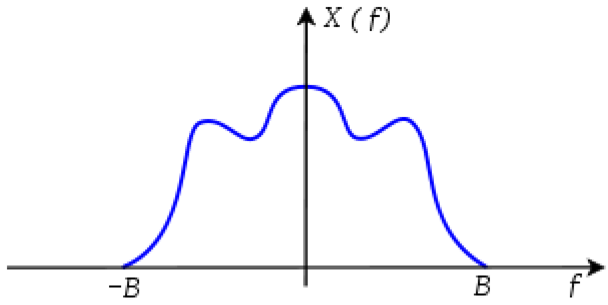
Band-Limited and Time-Limited Signals

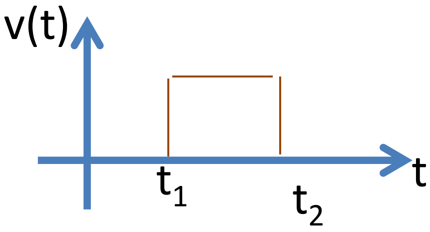
Band limited signals : A signal x(t) is said to be band limited if there exists a highest frequency "B" Hz, such that

$$X(f) = 0 \quad ; \quad |f| > B$$

Time limited signals : A signal x(t) is said to be time limited if it exists over certain finite duration of time,

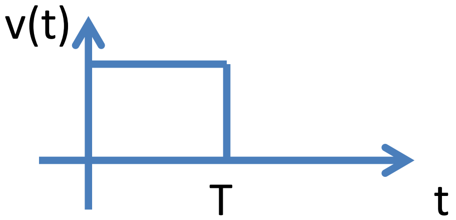
i.e. $$x(t) = 0 \quad ; \quad t \le t_1 \text{ and } t \ge t_2$$

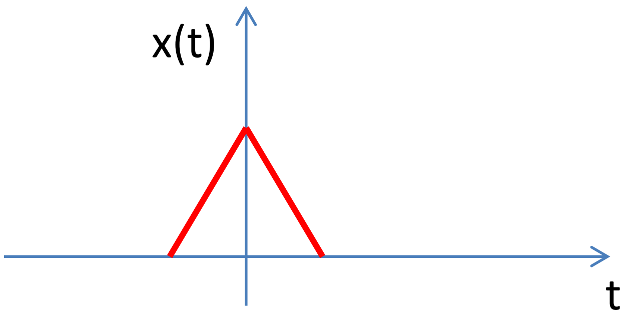


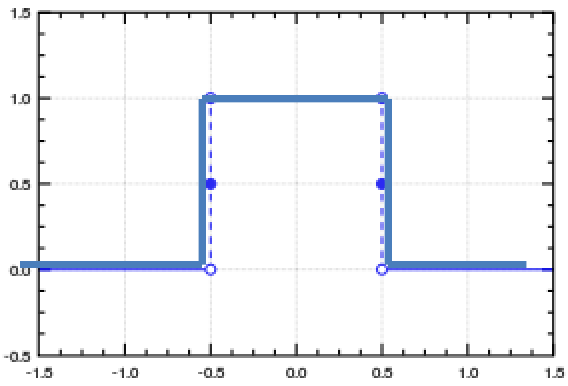


Band-Limited and Time-Limited Signals(4)

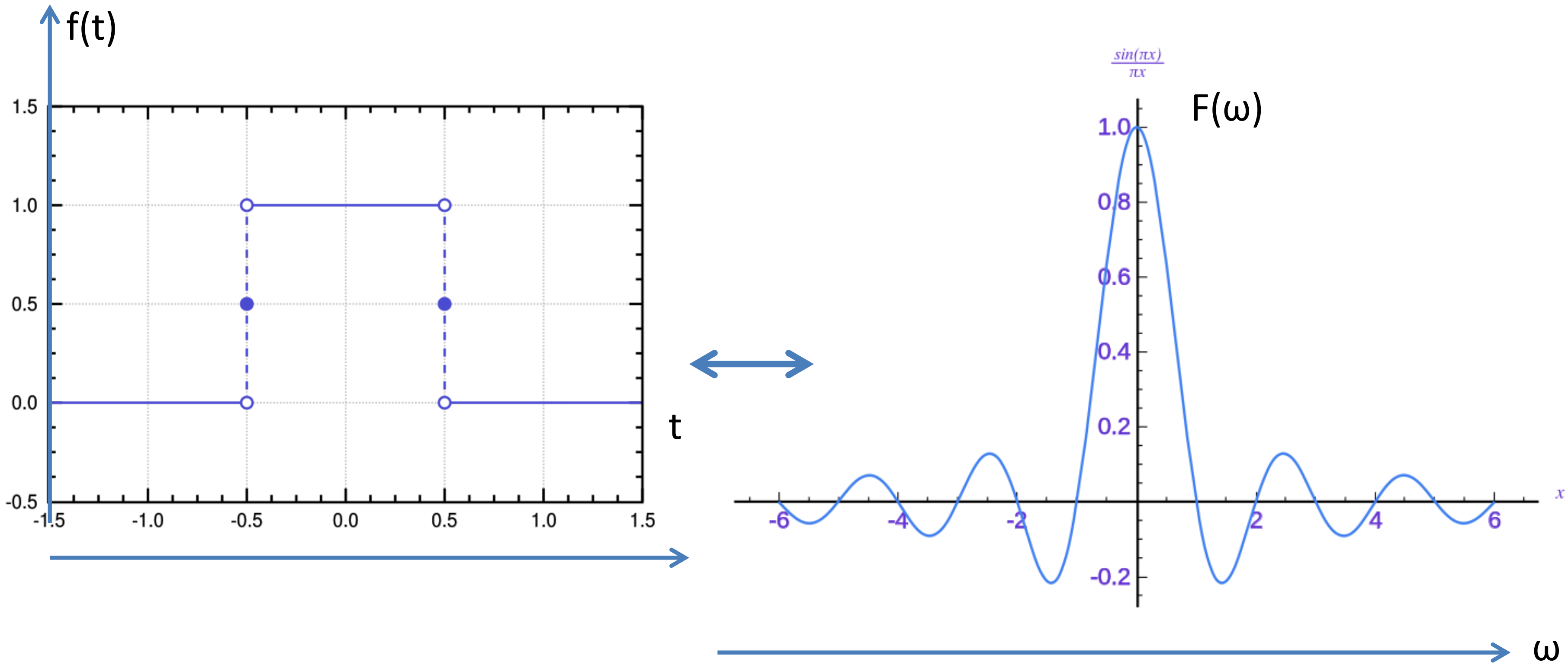
- **A time-limited signal is the one that is non-zero only for a finite length of time interval.**





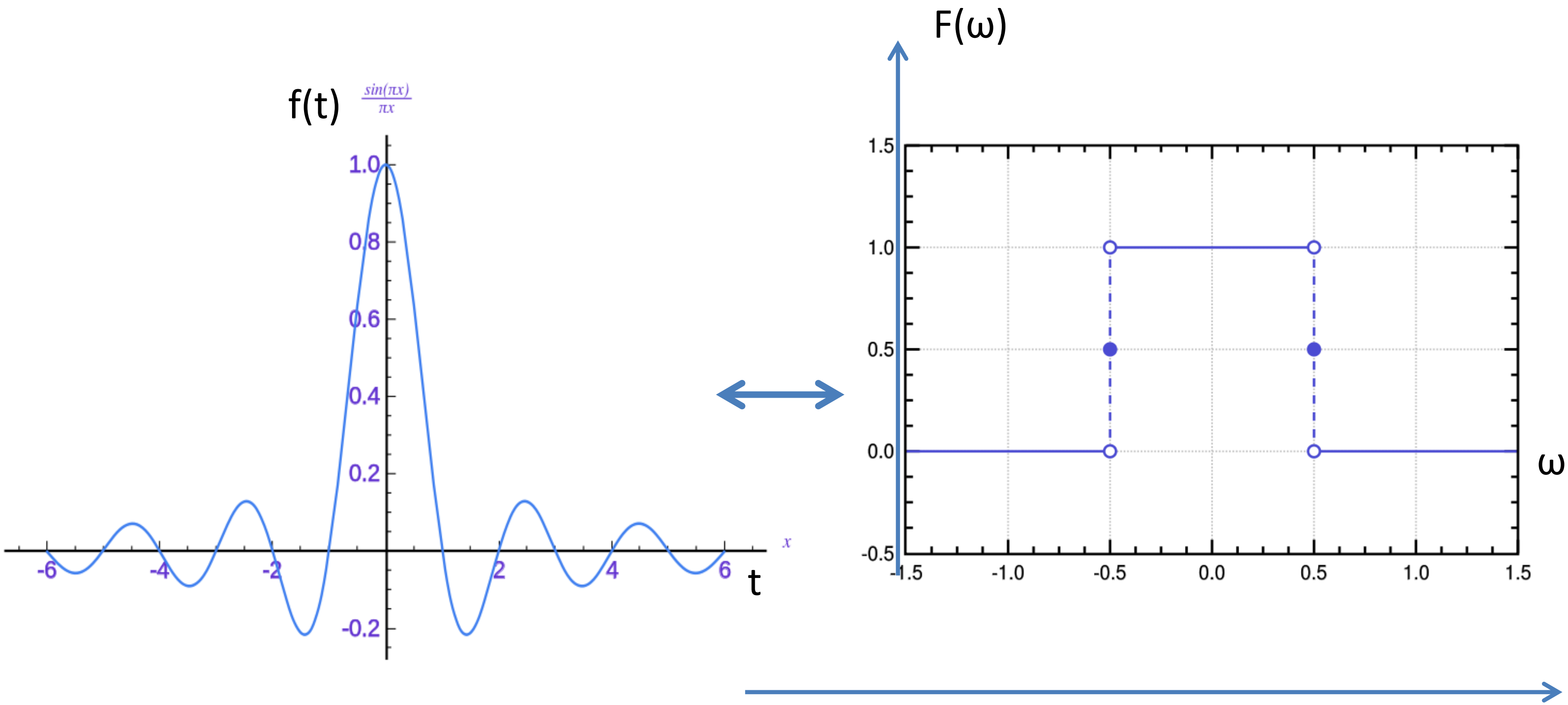


Band-Limited and Time-Limited Signals(5)

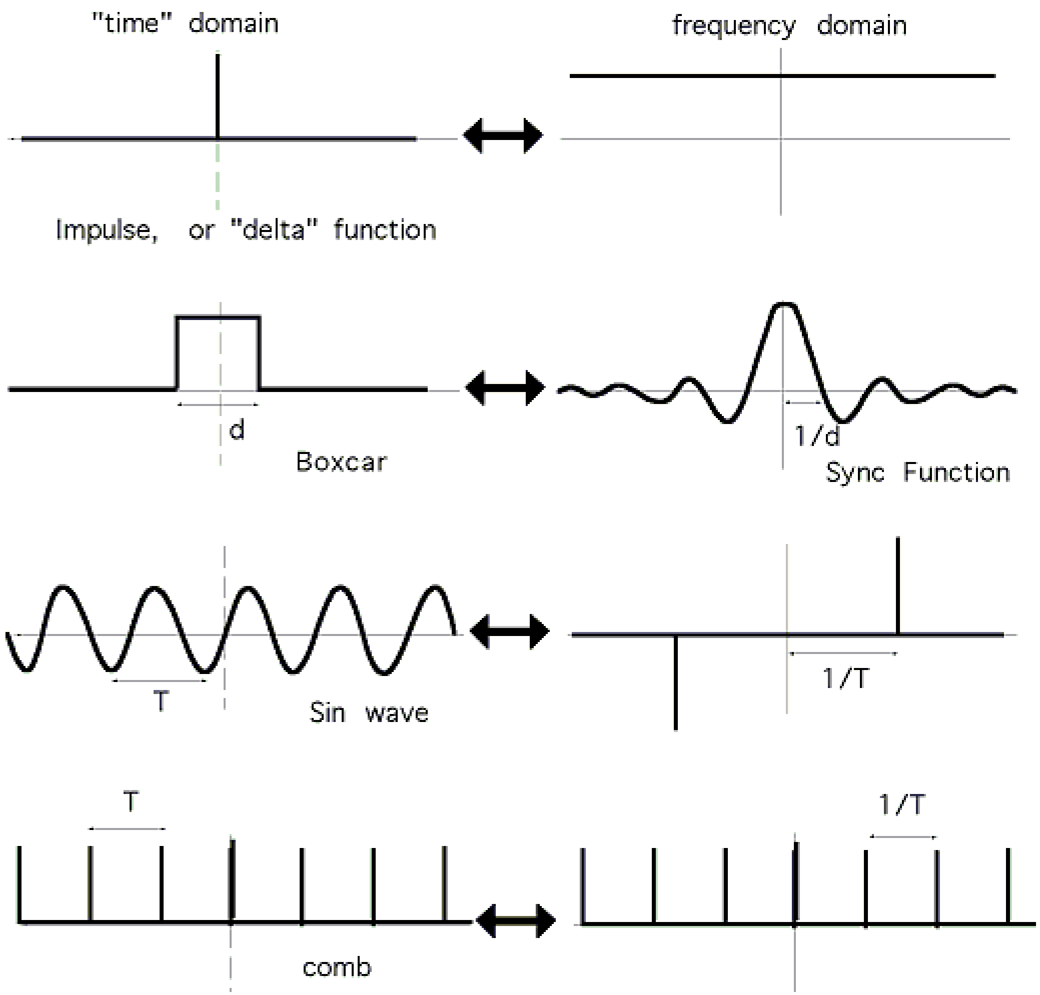


- **A time - limited signal cannot be also band limited.**

Band-Limited and Time-Limited Signals(6)



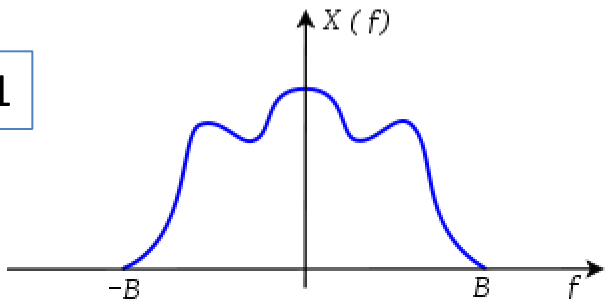
- **A band - limited signal cannot be also time-limited.**


"time" domain
frequency domain
Impulse, or "delta" function
d
Boxcar
1/d
Sync Function
T
Sin wave
1/T
T
1/T
comb

Band-Limited and Time-Limited Signals(3)

Mention which is a band limited signal?

1



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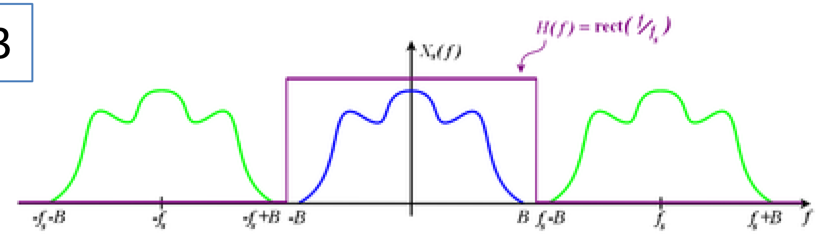

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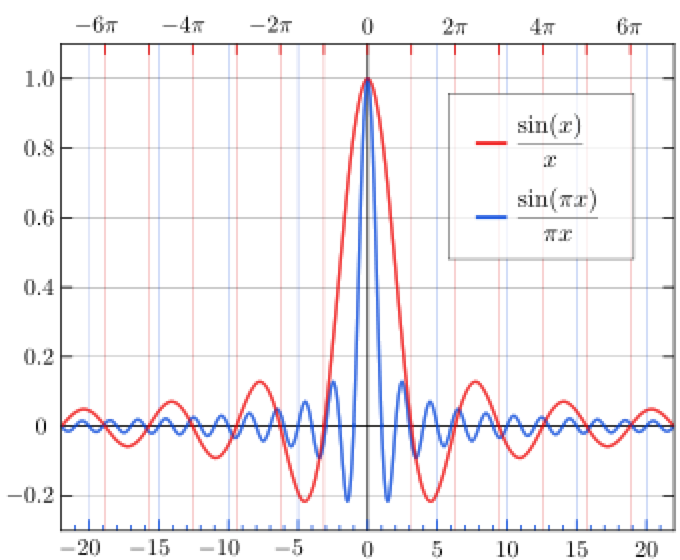

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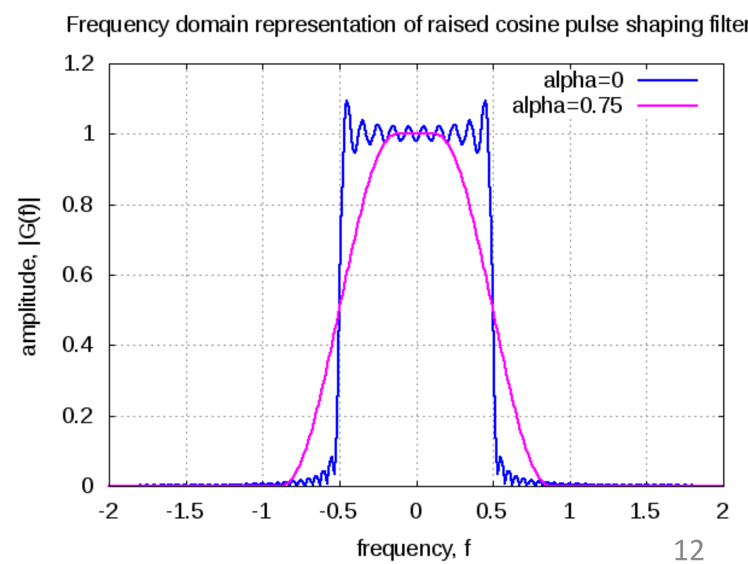


Narrowband signals and systems

Narrowband signals :

Narrowband signals has **frequency components within only a small band "B".**

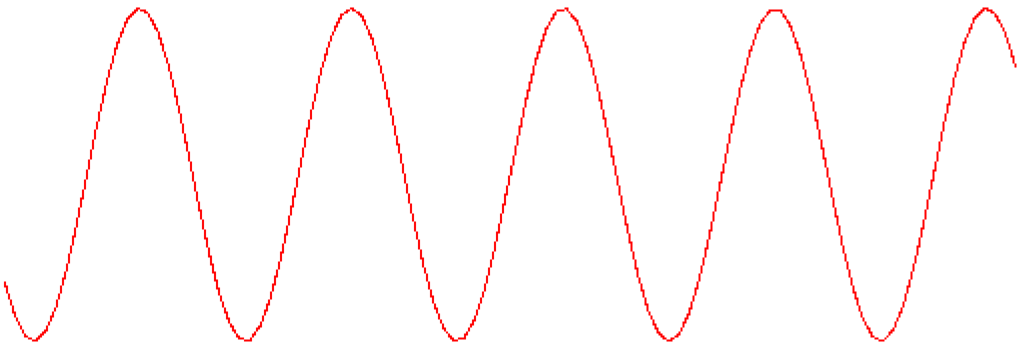
Modulation can produce narrowband signals.

Narrowband affects transmission bandwidth, design of receiver and transmitter.

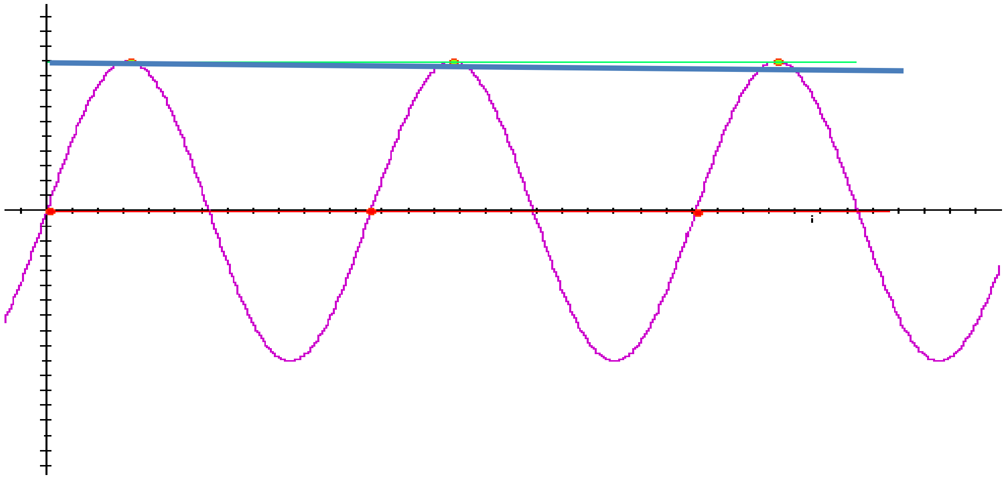
Narrowband systems :

Modulators, filters, transmitters, receivers that **process narrowband signals**, are called Narrowband systems.

Sampling Rate(1)



A Sine Wave

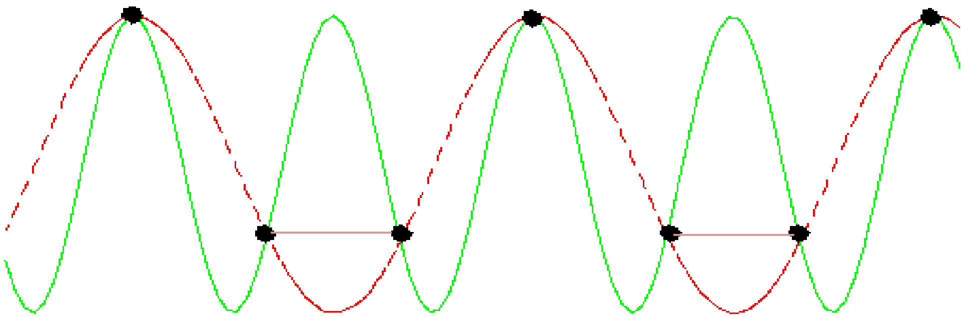


A Sine Wave Sampled at I time per cycle
And reconstructed

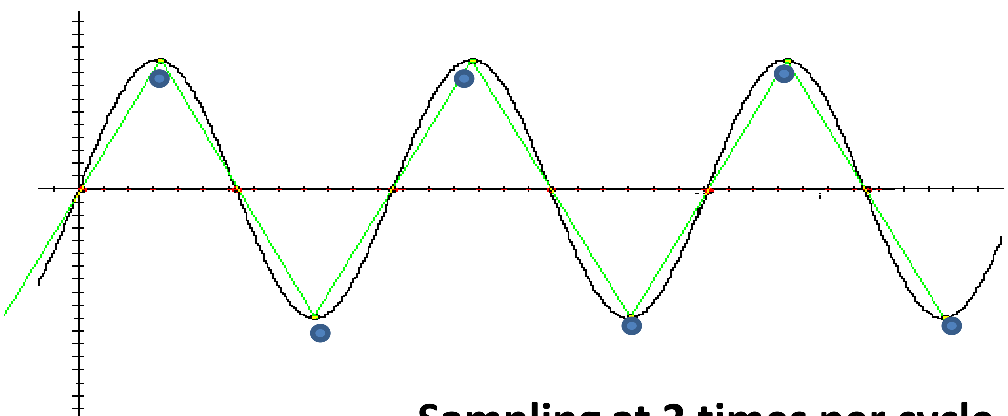
If a sine wave has frequency of 1Hz,
Then, Nyquist rate = 2Hz

But, if we will keep sampling rate = 1Hz,
Then, recovered signal might be as shown as blue horizontal line, from which it is not at all possible to recover or estimate the original signal.

Sampling Rate(2)



Sampling at 1.5 times per cycle

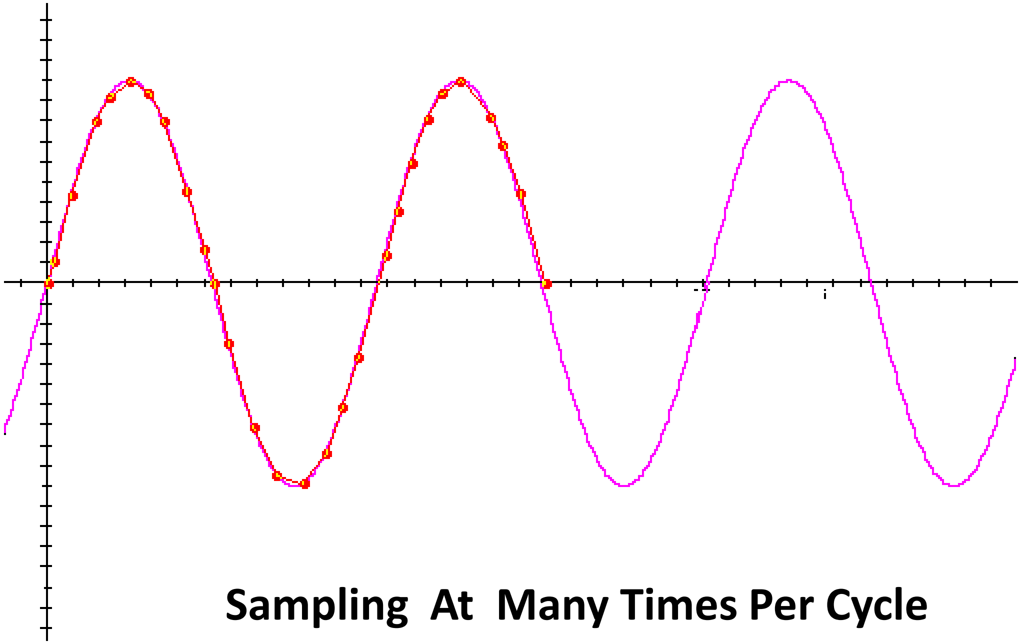


Sampling at 2 times per cycle

If a sine wave has frequency of 1Hz, Then, Nyquist rate = 2Hz

But, if we will keep sampling rate = 1.5 Hz or 2Hz, Then, recovered signal might be as shown as red (1st case) or green (2nd case) as shown, from which it is very hard to recover or estimate the original signal.

Sampling Rate(3)

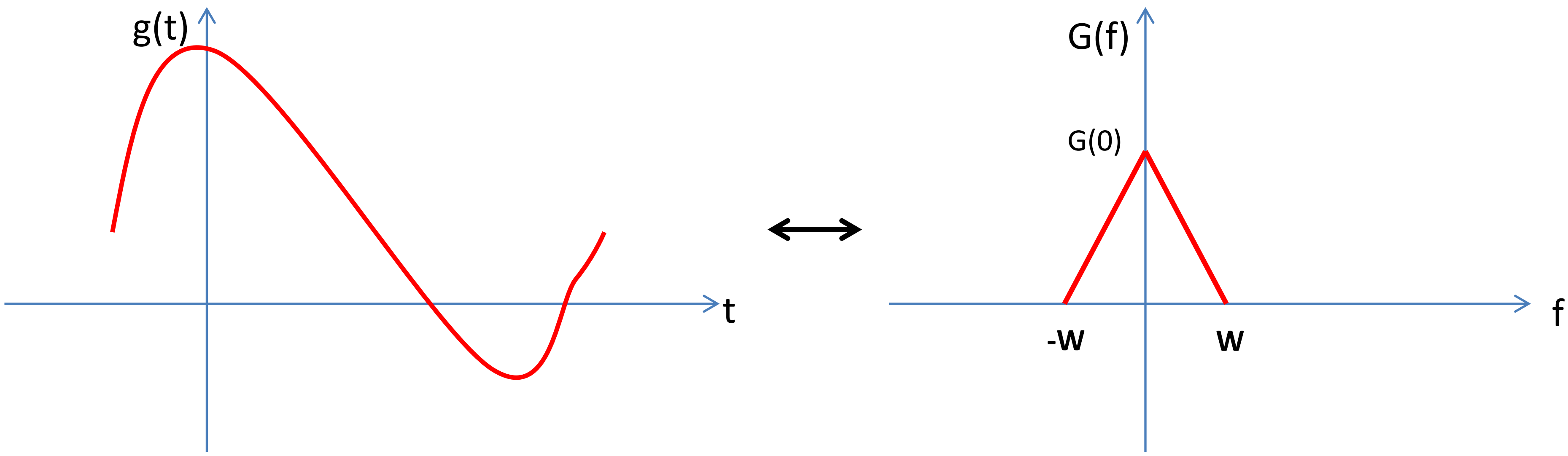


Sampling At Many Times Per Cycle

If a sine wave has frequency of 1Hz, Then, Nyquist rate = 2Hz

But, if we will keep sampling rate >> 2Hz i.e. sampling many times per cycle, Then, recovered signal will be as shown by red dotted outline, from which it is possible to recover or estimate the original signal.

Fourier Transform of a Strictly Band-limited Signal



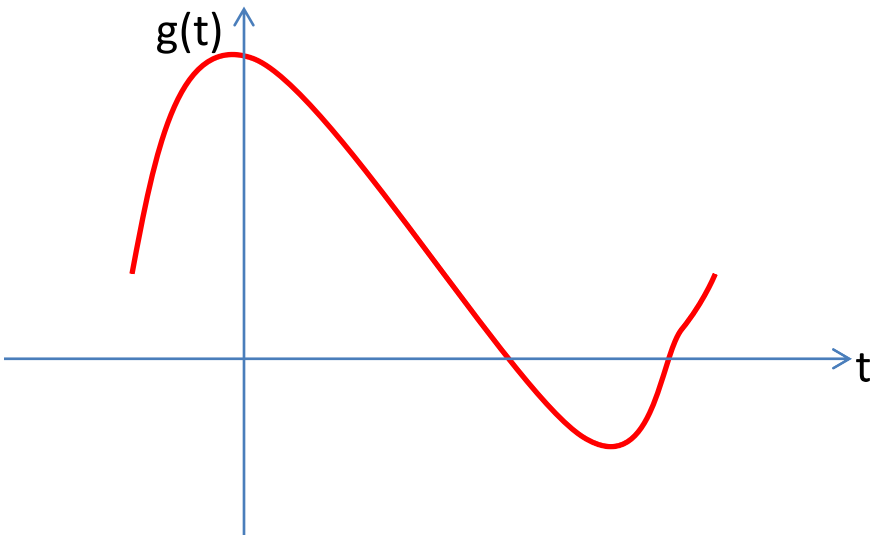
Analog signal

Spectrum of g(t)

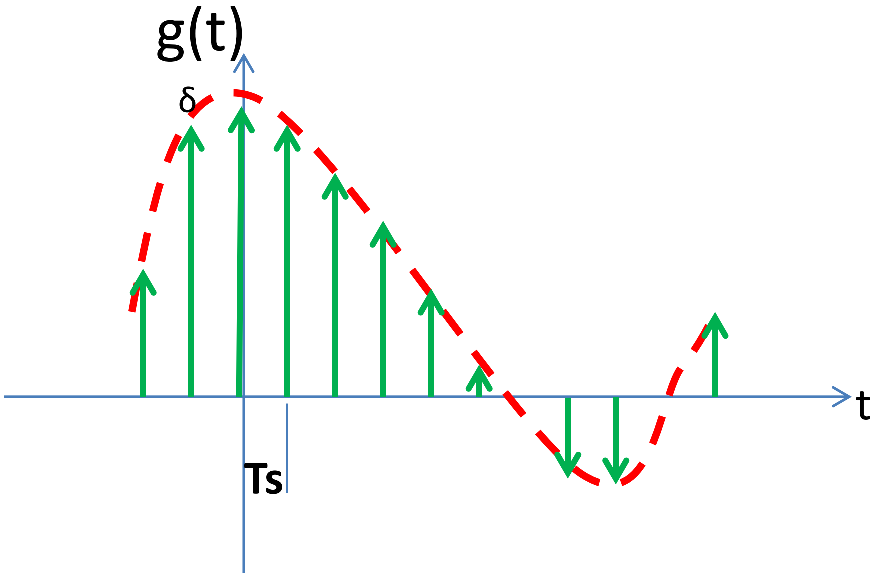
Sampling theorem :

Let, g(t) be a band limited signal whose highest frequency component is W.
Let, the signal is periodically sampled at every T_s seconds i.e. T_s>>(1/2W),
Then, these samples $g(nT_s)$ uniquely determines the signal and original signal may be recovered from these samples with no distortion.
The time T_s is the sampling time.

- Sampling process is described in time domain.



Analog signal

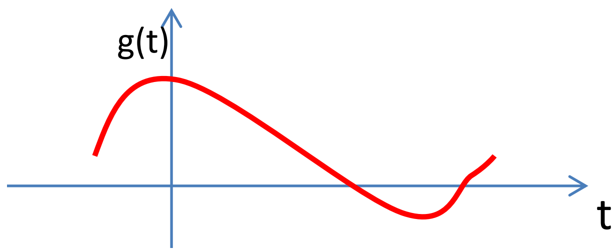


Instantaneously sampled version of analog signal

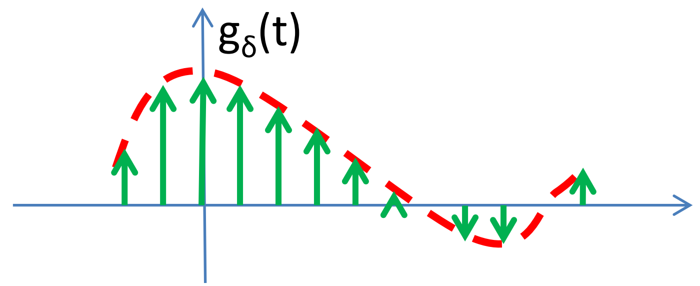
Ts: Sampling Period
1/Ts = fs   Sampling frequency

Sampling Process

Let, g(t) be a band limited signal whose highest frequency component is W.
$\delta_T(t)$ is a sampling signal with frequency f_S.{ where, $f_S >> 1/(2W)$}and
$g_\delta(t)$ is a sampled signal.
After multiplication of g(t) and $\delta_T(t)$ in time domain, we generate sampled signal $g_\delta(t)$.
Multiplication in time domain results into convolution in frequency domain.

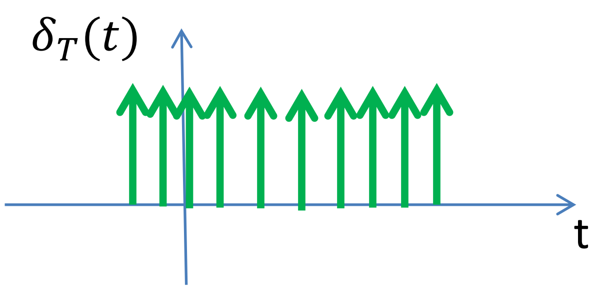


Analog signal: g(t)



Instantaneously sampled version of analog signal

$$g_\delta(t) = g(0)\,\delta(t) + g(Ts)\,\delta(t-Ts) + g(2Ts)\,\delta(t-2Ts) + \ldots$$

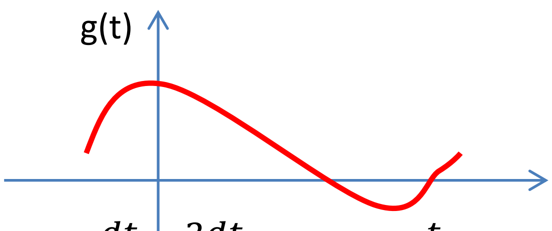


Sampling signal

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

Application of Fourier Transform Property (Sampling theorem and low pass signal)

- Frequency convolution property: Multiplication of two functions in time domain is equivalent to convolution in frequency domain.



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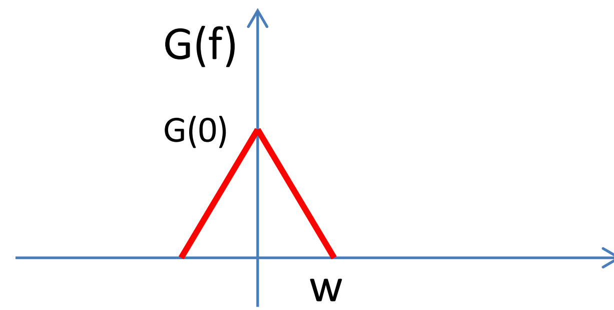
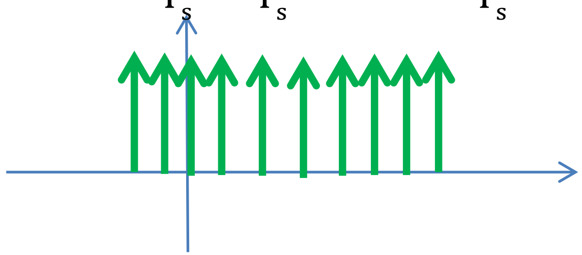
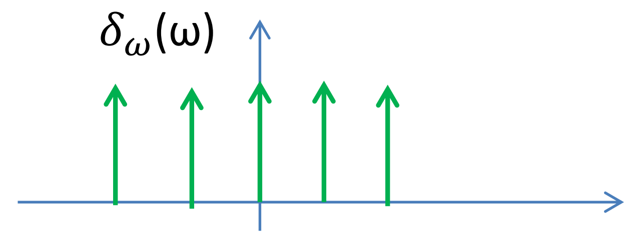


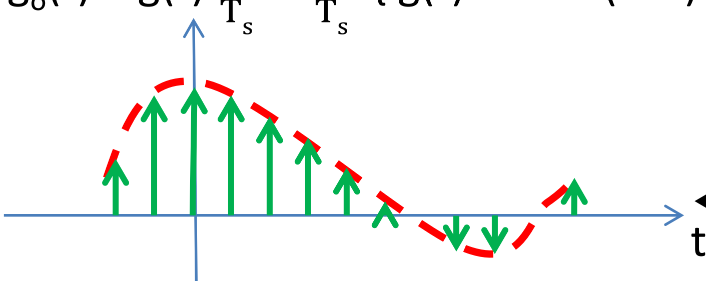
$$\delta_T(t) = \frac{dt}{T_s} + \frac{2dt}{T_s}\{\cos 2\pi \frac{t}{T_s} + \cos 2\times 2\pi \frac{t}{T_s} + \_\_\_\_\_\_\_\}$$





↔

$$g_\delta(t) = g(t)\frac{dt}{T_s} + \frac{2dt}{T_s}\{g(t)\cos 2\pi(2W)t + g(t)\cos 2\times 2\pi(2W)t + \_\_\_\_\_\}$$



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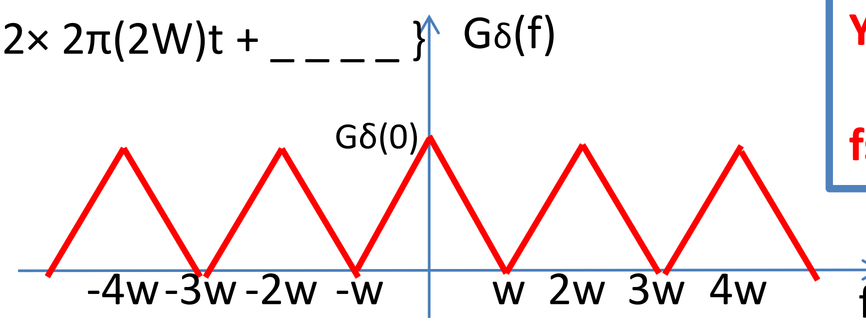


FREQUENCY fs

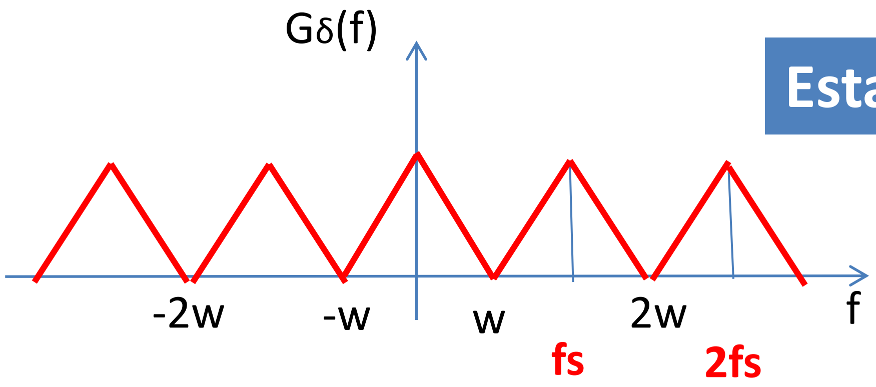
Sampled version of analog signal

Spectrum of sampled g(t)

Different Possible Versions of Periodic Spectrum

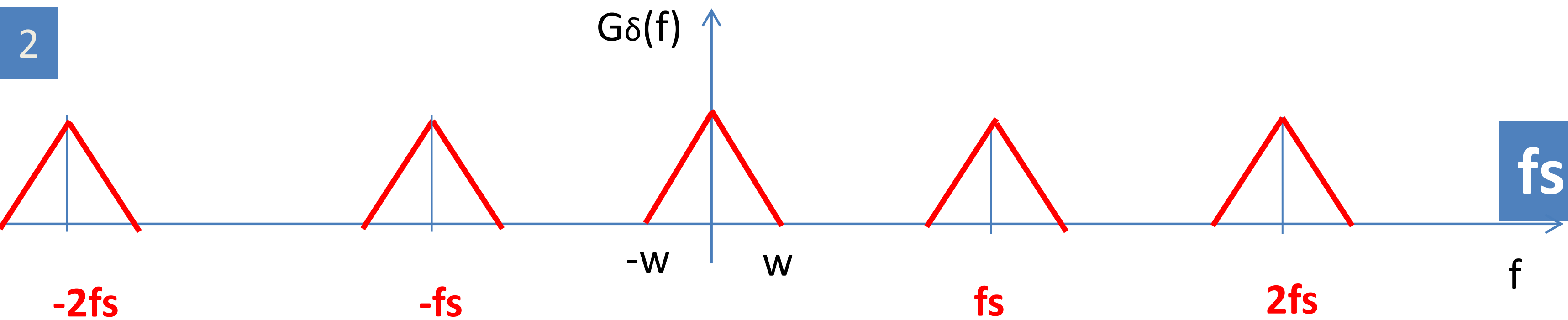
Establish the relation between fs and w.

1



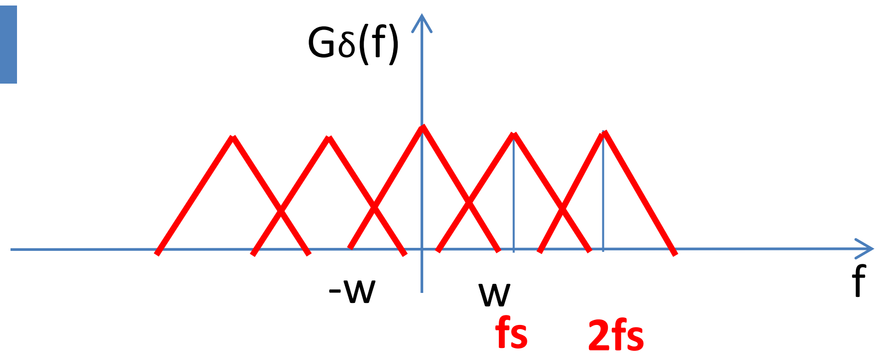
fs = 2w

2



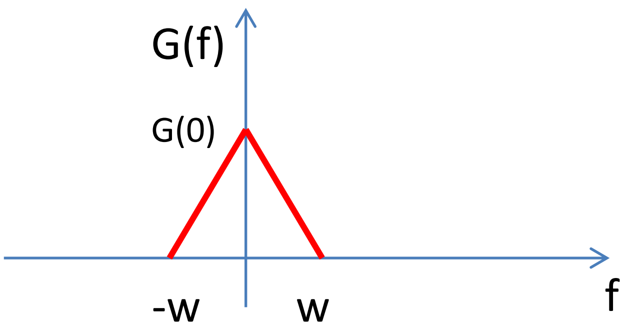
fs > 2w

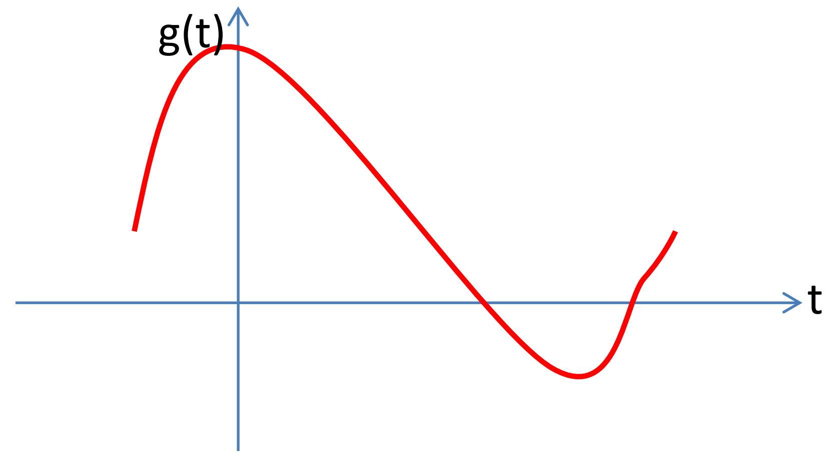
3



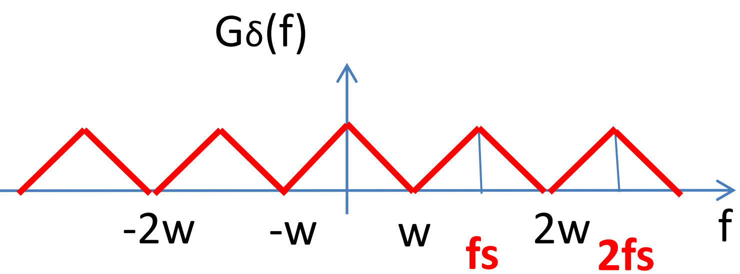
fs < 2w

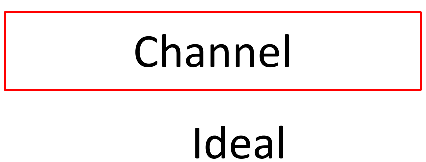
Reconstruction (1)

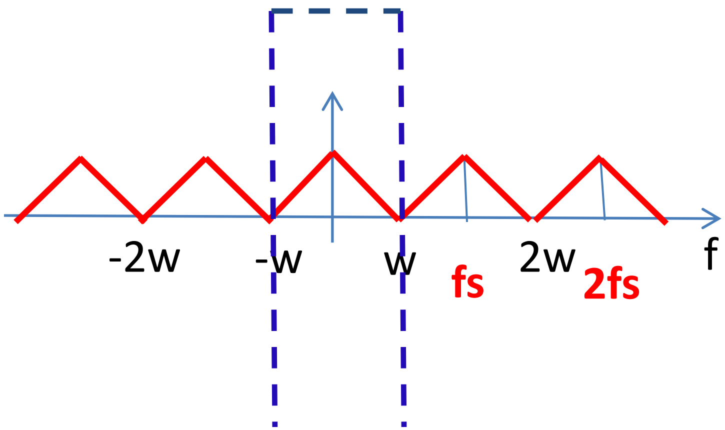


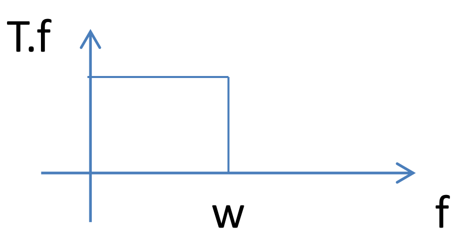


1. fs = 2w





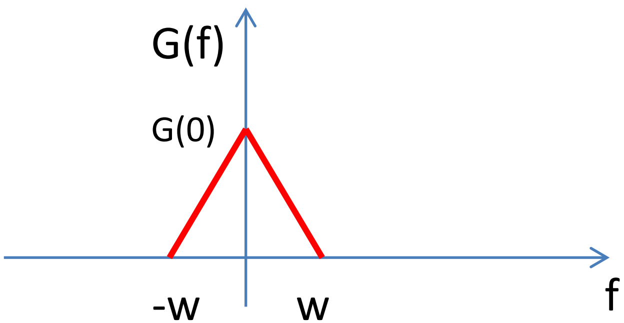


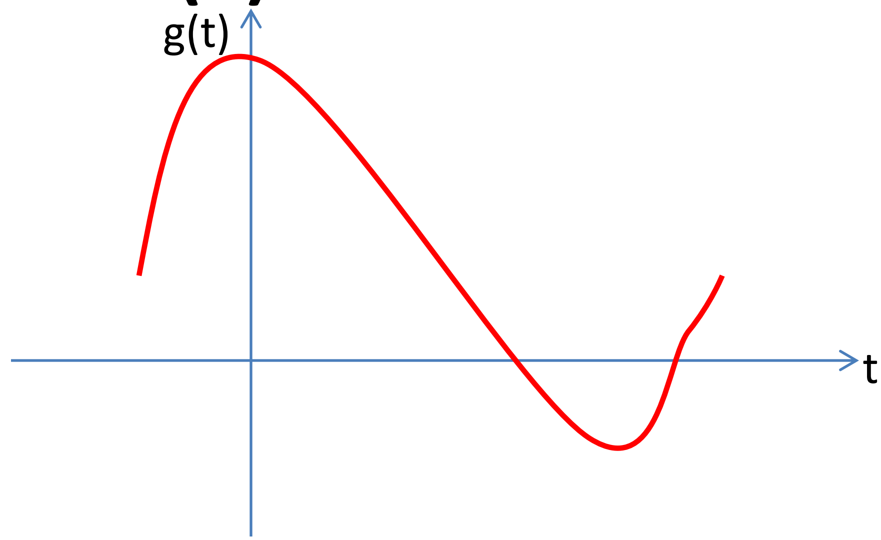


Pass it through a LPF

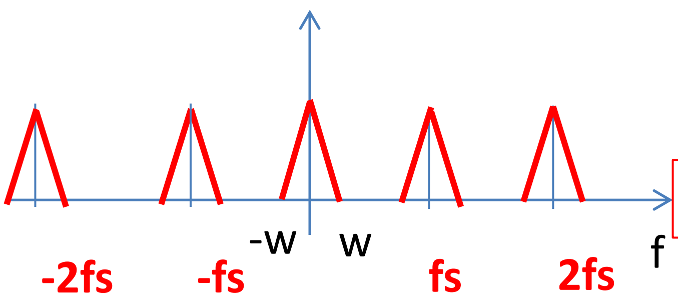
LPF: ideal frequency response

Reconstruction (2)



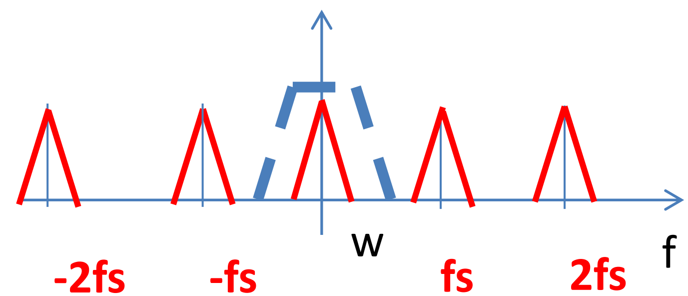


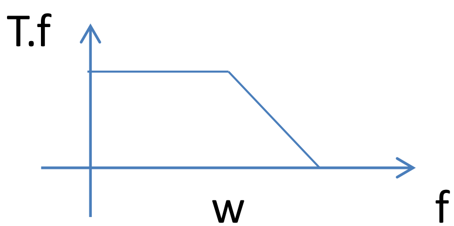
2. fs > 2w



Channel

Ideal





Pass it through a LPF

LPF: practical frequency response

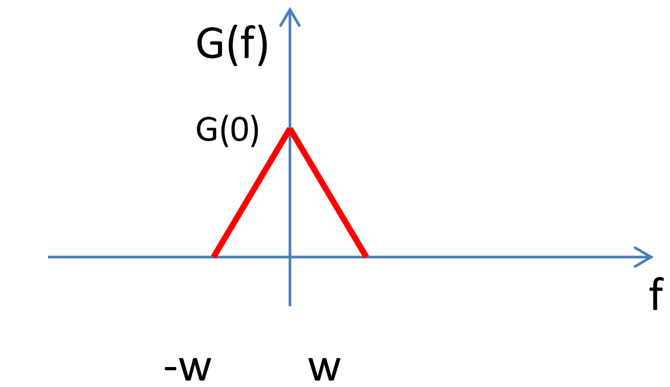
Guard band

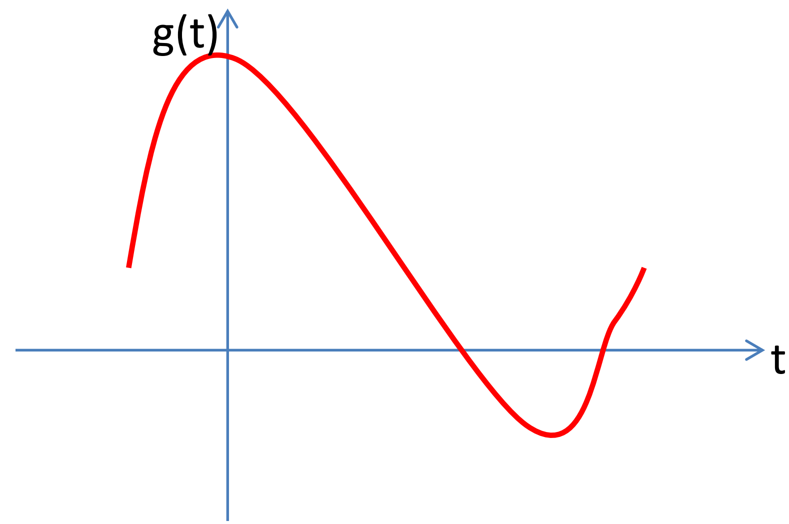
If f_s=8KHz,
W= 3.3KHz

Then,

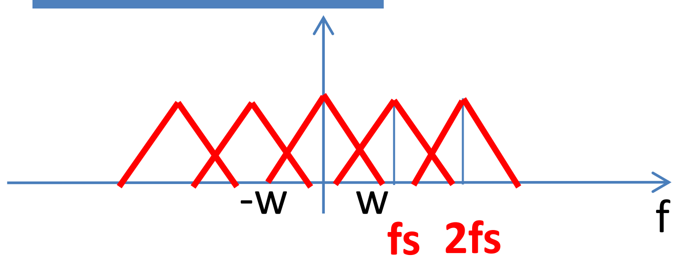
Guard band = $(f_s$-W) – W
= f_s- 2W
= (8KHz-2*3.3KHz)
= 1.4 KHz

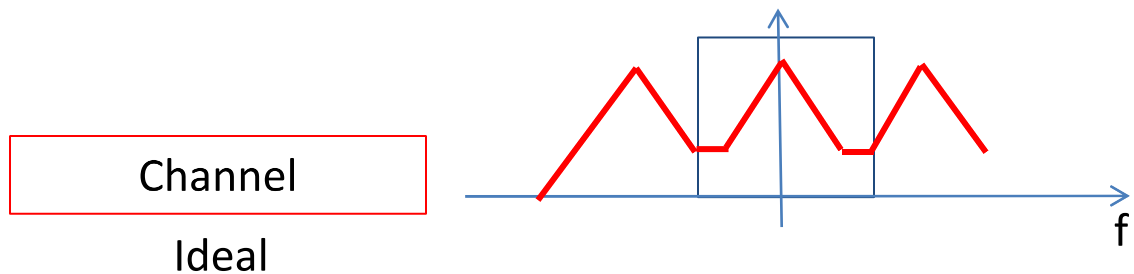
Reconstruction (3)





3. fs < 2w





Not possible to reconstruct the original signal

Here, we find the overlap between spectrum of g(t) and spectrum of DSBSC centered around f_S. Hence, no filtering action will allow recovery of g(t). This phenomenon is called as **Aliasing** in frequency domain.

ALIASING

Nyquist criteria

Nyquist rate : When the **sampling rate becomes exactly Equal to ‘2w’ samples per second**, for a given bandwidth Of ‘w’ Hz, then it is Nyquist rate.

Nyquist interval: **Time interval** between any two adjacent Samples **when sampling rate is Nyquist rate.**

Nyquist Theorem

- A band limited signal of finite energy, which has no frequency components higher than W or fm Hz, is completely described by specifying the values of the signal at instants of time separated by 1/2w or 1/2fm seconds.

- A band limited signal of finite energy, which has no frequency components higher than W or fm Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W or 2fm samples per second.

- 2W or 2fm sps: Nyquist Rate; 1/2w or 1/2fm : Nyquist interval

Sampling of Band pass signal

If m(t) has highest frequency $\mathbf{f_M}$ and lowest frequency **0 Hz**,
then $\mathbf{f_s \geq 2f_M}$.

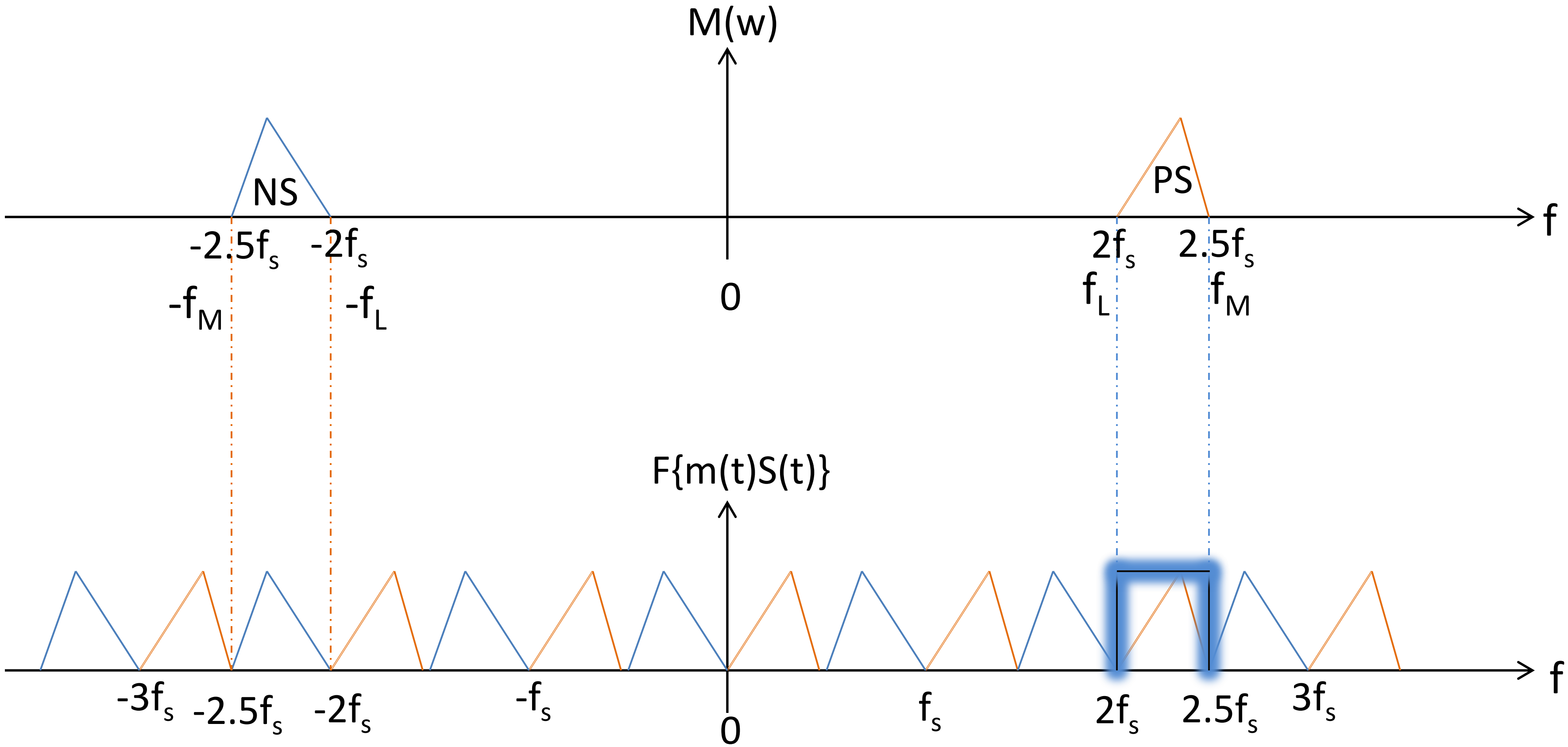
<u>case (i) $f_L = n.f_s$</u> ; where, n = integer n=2(here) ; $\mathbf{f_s=2(f_M-f_L)}$

If m(t) has highest frequency $\mathbf{f_M}$ and lowest frequency $\mathbf{f_L}$ Hz,
then $\mathbf{f_s}$ need not be greater than $\mathbf{2(f_M-f_L)}$.

e.g. If spectral range of signal extends from 10.0 MHz to 10.1MHz,
then $f_s= 2(f_M-f_L) = 2(10.1-10.0) = 0.2$MHz.

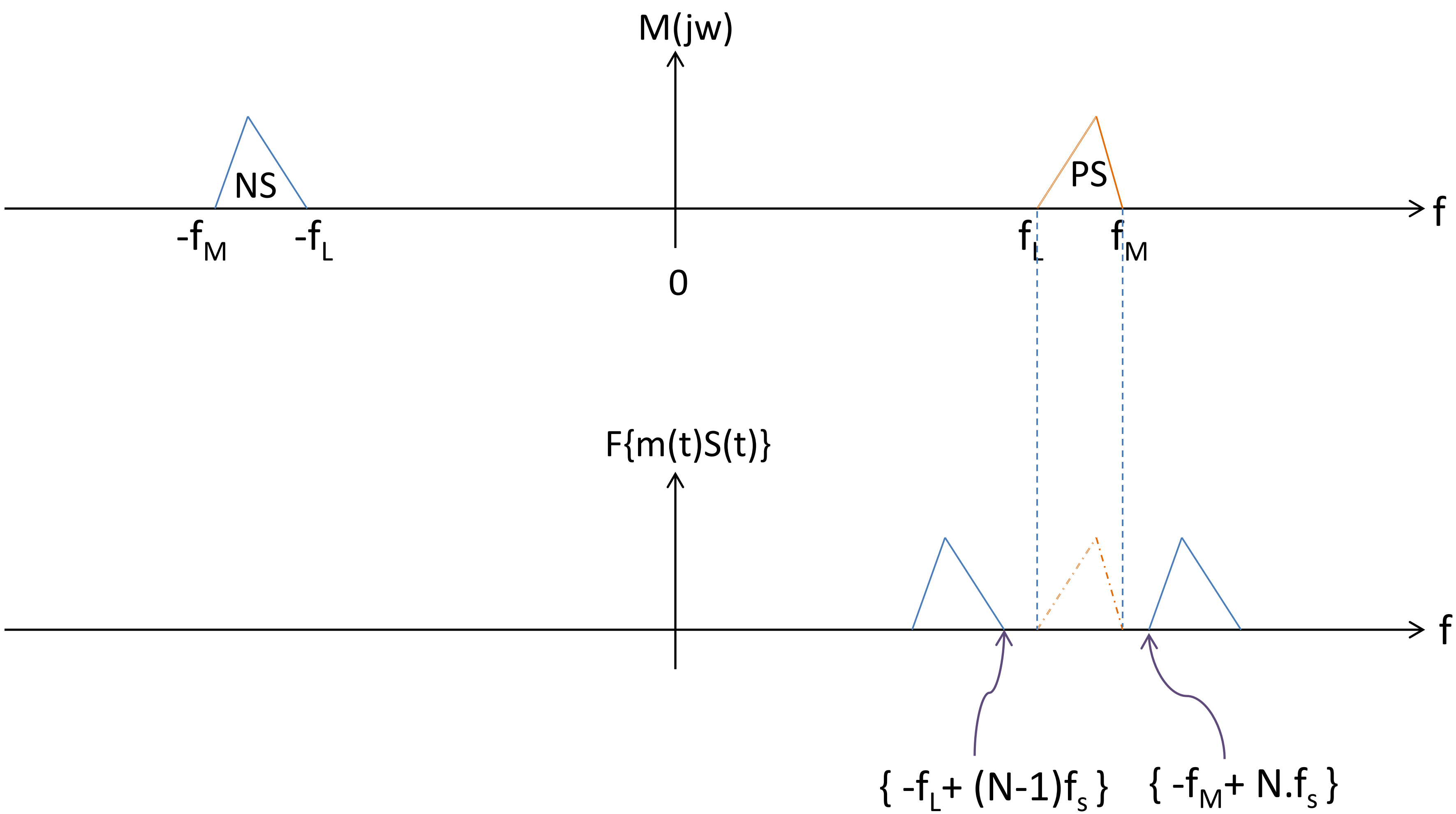
- To establish the sampling theorem for bandpass signals, select sampling frequency as $\mathbf{f_s=2(f_M-f_L)}$ *provided that f_M or f_L is a harmonic of f_s.*

case (i) $f_L = n.f_s$; where, n = integer n=2(here) ; $\mathbf{f_s=2(f_M-f_L)}$



If " F{ m(t)S(t) }" is passed thr' sharp cutoff BPF from f_L to f_M,
signal m(t) will be recovered exactly.

case (ii) $f_L \neq n.f_s$; *f_M and f_L are not harmonics of f_s* ; To find f_s which will give no overlaps.



$-f_L + (N-1)f_s \leq f_L$

$(N-1)f_s \leq 2f_L$

Let, $f_M - f_L \equiv B$, $k \equiv f_M/B$

$(N-1)f_s \leq 2(f_M - B)$

$f_s \leq 2B\left(\frac{k-1}{N-1}\right)$

$-f_M + N.f_s \geq f_M$

$N.f_s \geq 2f_M$

$N.f_s \geq 2f_M$

$f_s \geq 2B\left(\frac{k}{N}\right)$

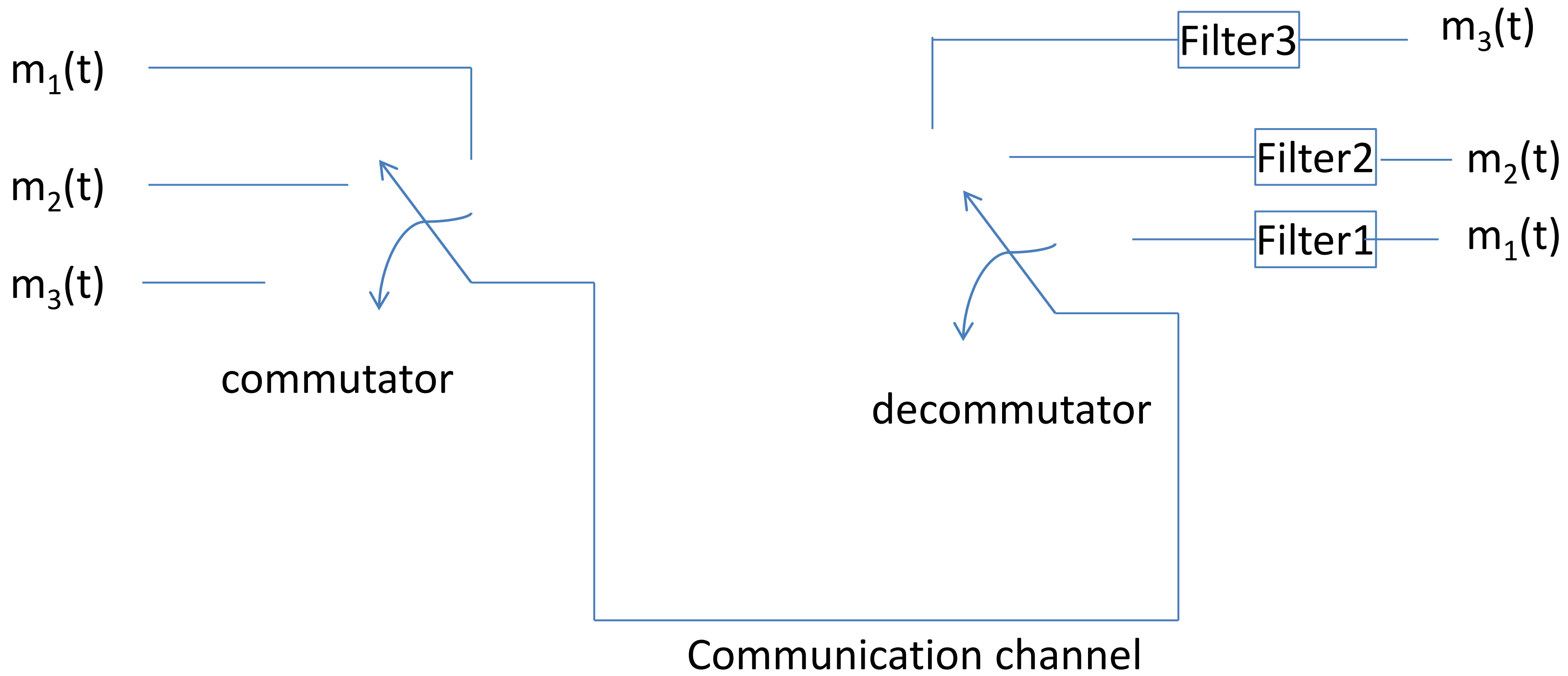
This means that sampling frequency should be in between above two f_s values to avoid the overlap.

As $f_s >> 2B$; $k>>N$

Band pass sampling theorem :- A band pass signal with highest frequency f_M and bandwidth B, can be recovered from it's samples through band pass filtering, by sampling it with frequency $f_s = 2 f_M / k$, where k is the largest integer not exceeding f_M/B.

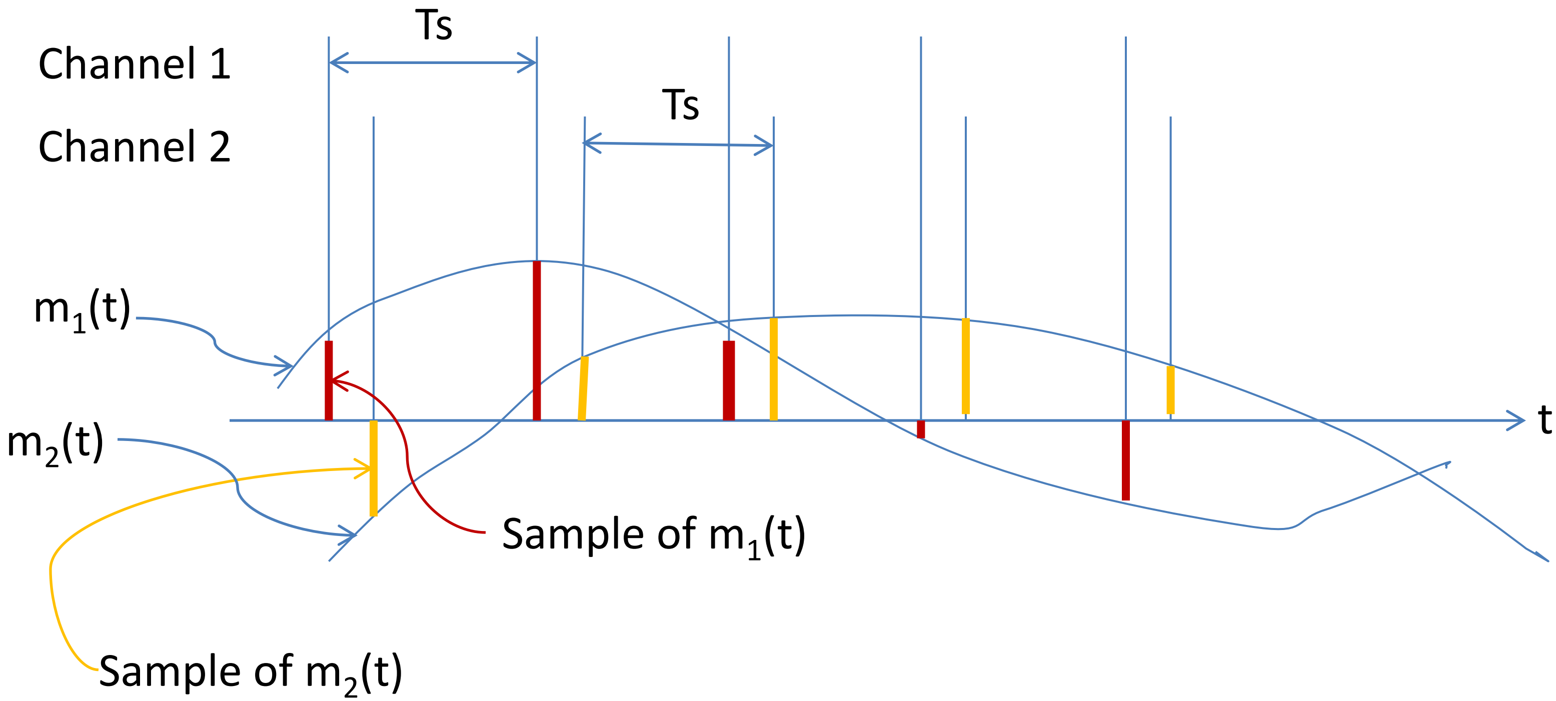
Numericals :

Pulse amplitude modulation and concept of time division multiplexing :



Transmission of no. of band limited signals over single Communication channel

Interlacing of two baseband signals :



These samples are inputs to the corresponding filters in decommutator.

Maximum no. of signals can be multiplexed = N = f_c / f_M

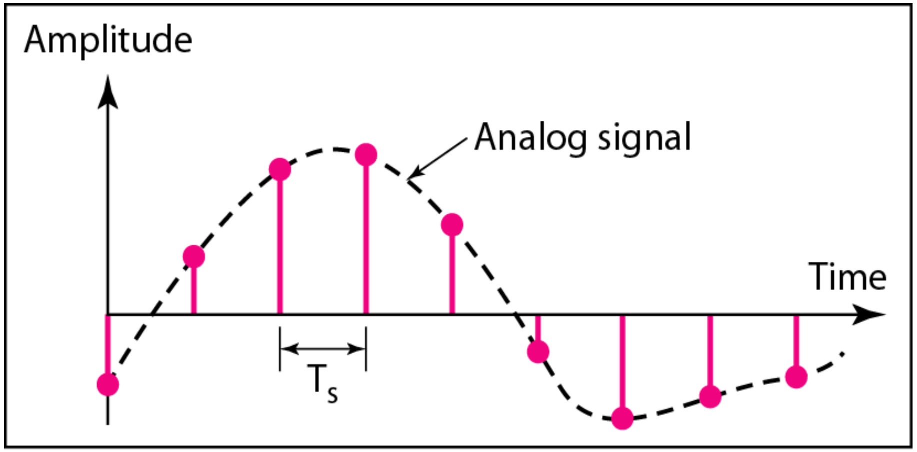
where, f_c => channel bandwidth

f_M => baseband signal bandwidth i.e. $m_1(t)$, $m_2(t)$, …… are bandlimited to f_M.

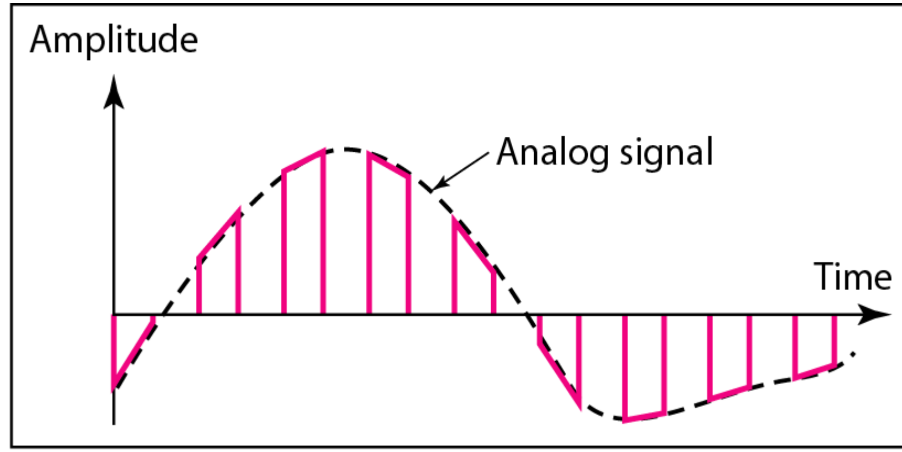
* The instantaneous samples at the transmitting end will have infinitesimal energy, thus have infinitesimal peak value after transmission, which can be lost in background noise.

* Thus more reasonable sampling will be **natural sampling**.

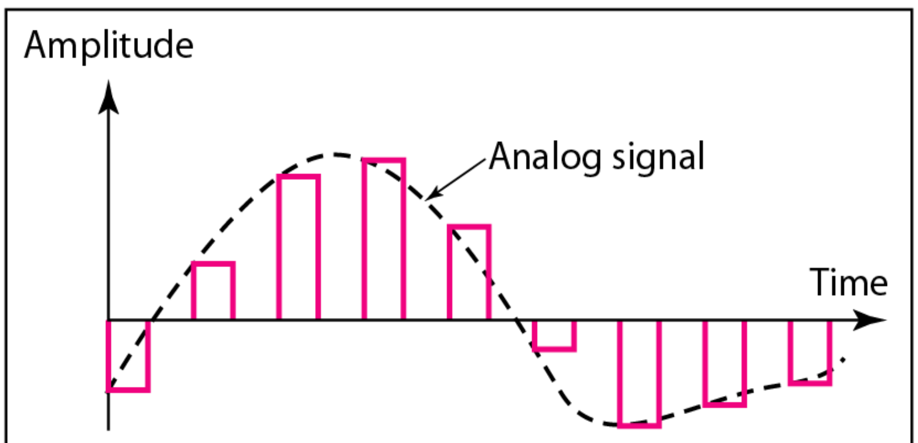
Types of sampling



a. Ideal sampling

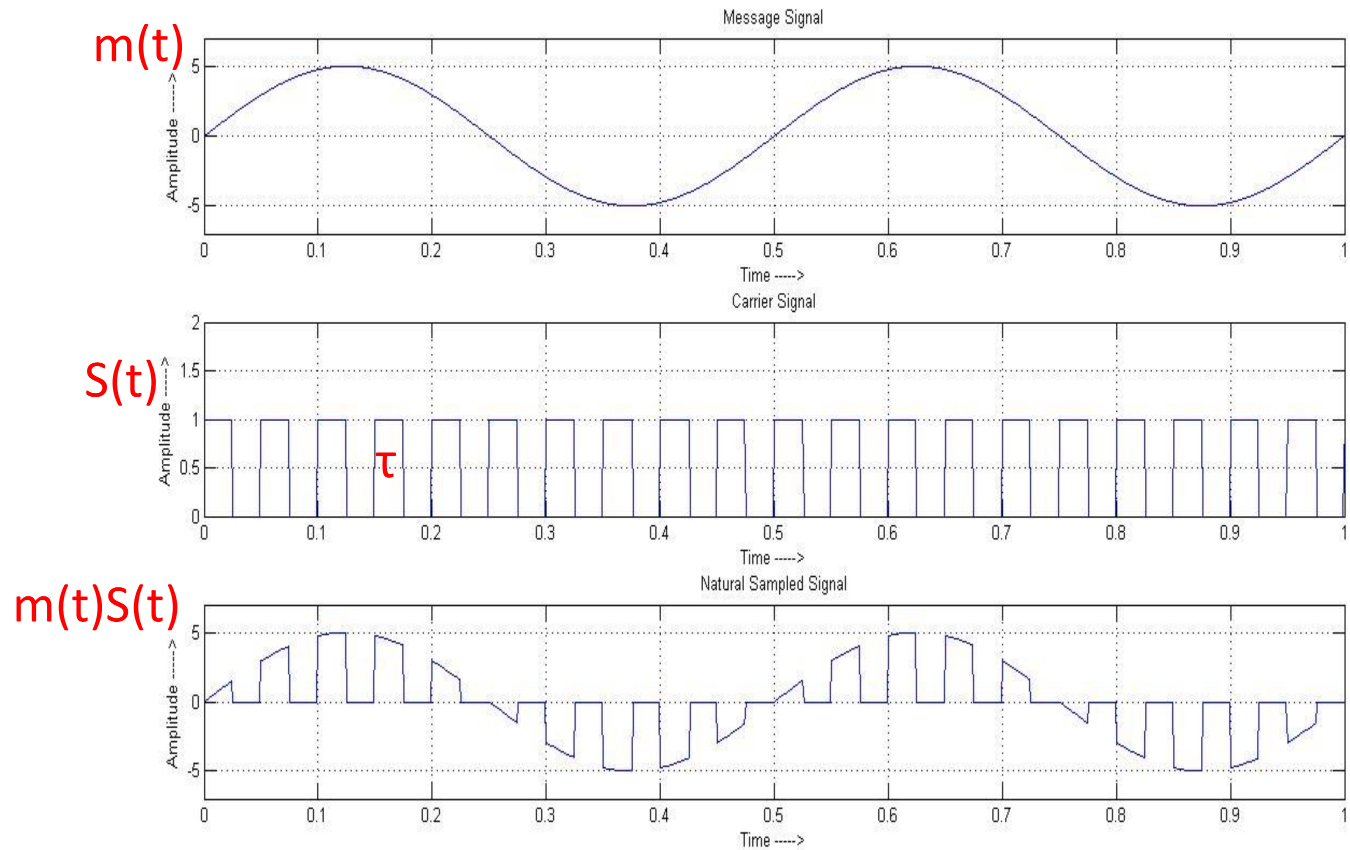


b. Natural sampling



c. Flat-top sampling

Natural sampling :



$$S(t) = \frac{\tau}{T_s} + \frac{2\tau}{T_s}\left\{ c_1 \cos 2\pi \frac{t}{T_s} + c_2 \cos 2\times 2\pi \frac{t}{T_s} + \_\_\_\_\_\_\_\_ \right\}, \quad c_n = \frac{\sin\left(\frac{n\pi\tau}{Ts}\right)}{\left(\frac{n\pi\tau}{Ts}\right)}$$

$$m(t)S(t) = \frac{\tau}{T_s}\mathbf{m(t)} + \frac{2\tau}{T_s}\left\{ m(t)\, c_1 \cos 2\pi \frac{t}{T_s} + m(t)\, c_2 \cos 2\times 2\pi \frac{t}{T_s} + \_\_\_\_\_\_\_ \right\}$$

Passed thr' LPF $S_0(t) = \frac{\tau}{T_s} m(t)$

If ‘N’ signals are to be multiplexed,

then the maximum pulse width $\tau = \frac{Ts}{N}$

If τ ↑, output $s_o(t)$ ↑ ; *where* $S_0(t) = \frac{\tau}{T_s} m(t)$

But with ↑ in τ,

crosstalk ↑

because guard band between adjacent pulse ↓.

So, $\boldsymbol{\tau \ll \frac{Ts}{N}}$.

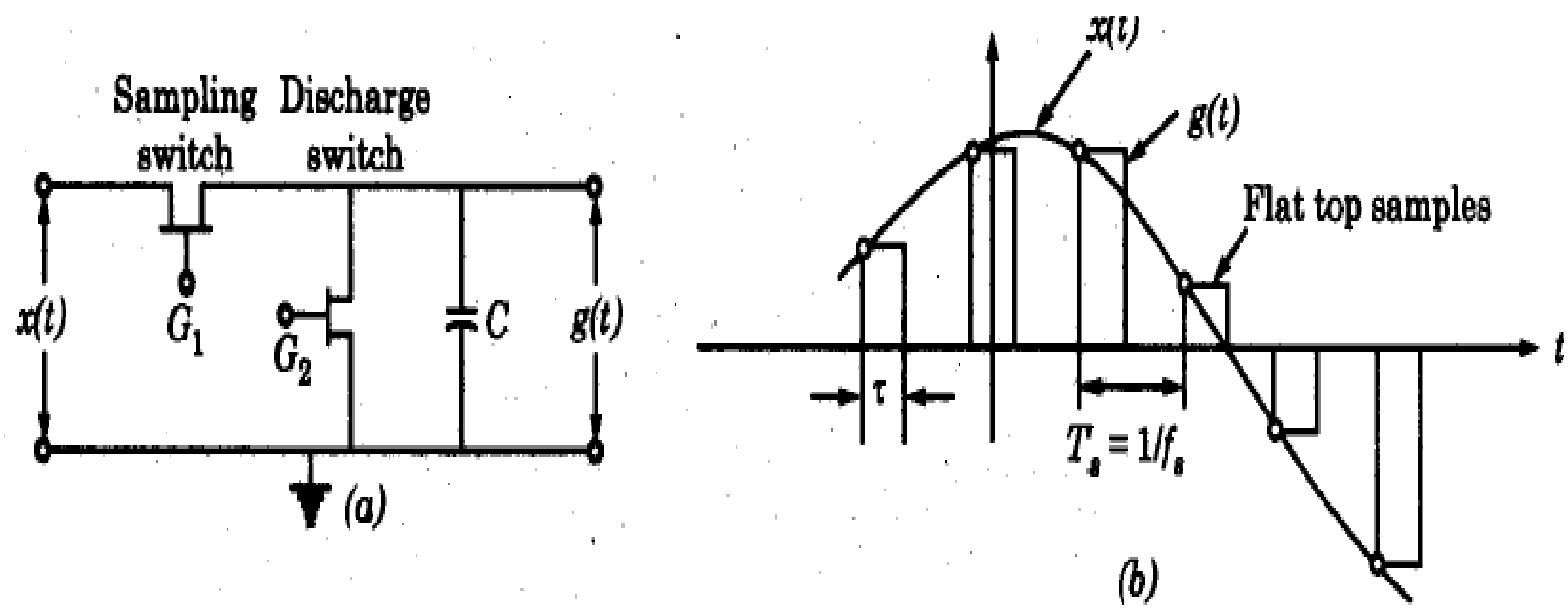
Natural sampling is not generally used,

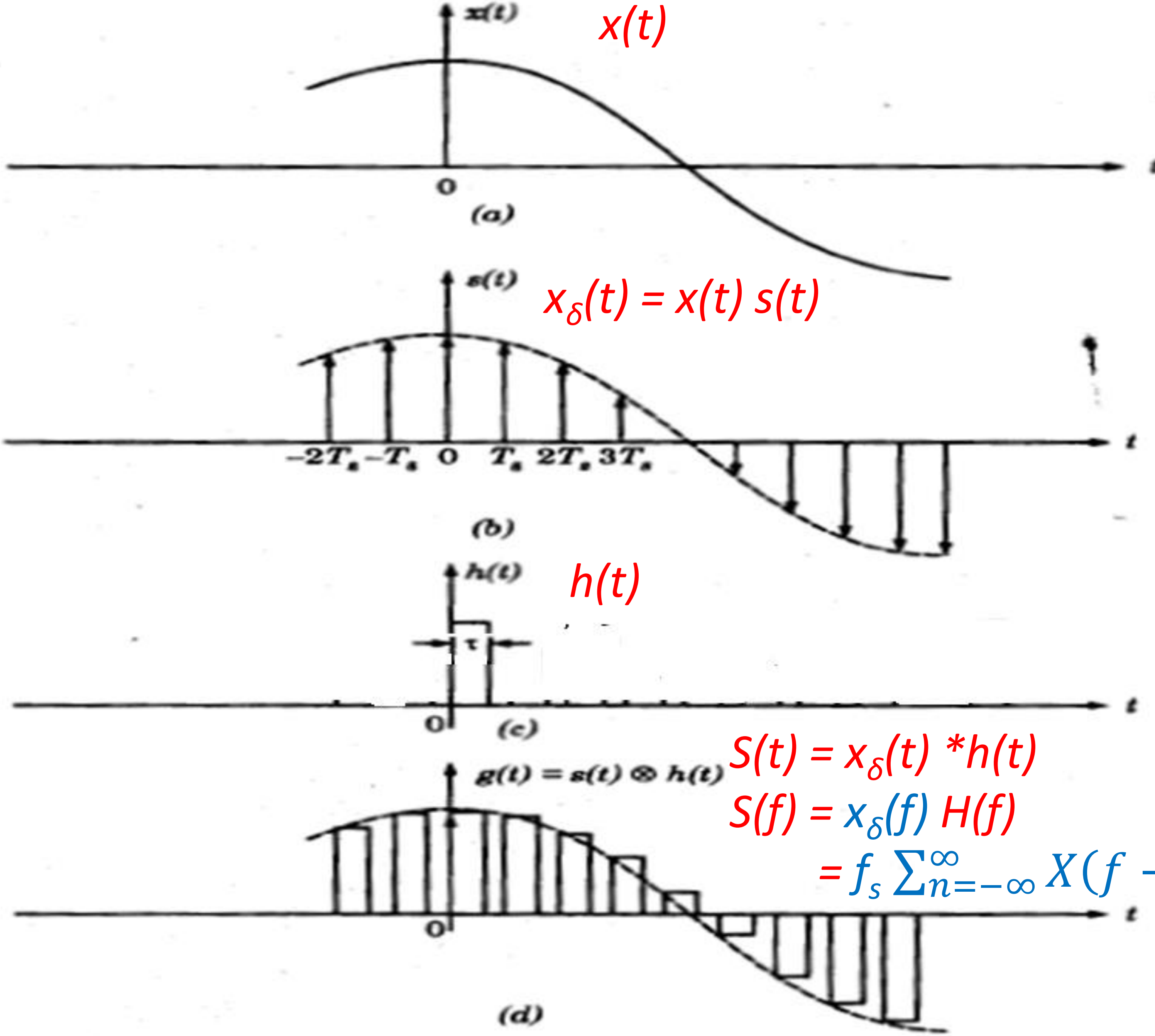
but instead **Flat top sampling** is used which **simplifies the circuitry for sampling operation**.

Flat top sampling :

- A Flat top pulse has a const. amplitude equal to sample value of signal at the beginning of the pulse.

- A gate pulse at G_1 briefly closes the sampling switch and the capacitor holds the sampling voltage until discharged by a pulse applied at G_2.





$x(t)$

$x_\delta(t) = x(t)\, s(t)$

$h(t)$

$S(t) = x_\delta(t) * h(t)$

$S(f) = x_\delta(f)\, H(f)$

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

Aperture effect :

The **high frequency roll off** characteristic of **H(f)** acts as low pass filter and **attenuates upper portion of message spectrum**. This loss of high frequency content is called as Aperture effect.

The aperture effect can be compensated by
(i) selecting pulse width τ very small i.e. " $\tau << T_s$ ".
(ii) using equalizer circuit

Recovering of x(t) :

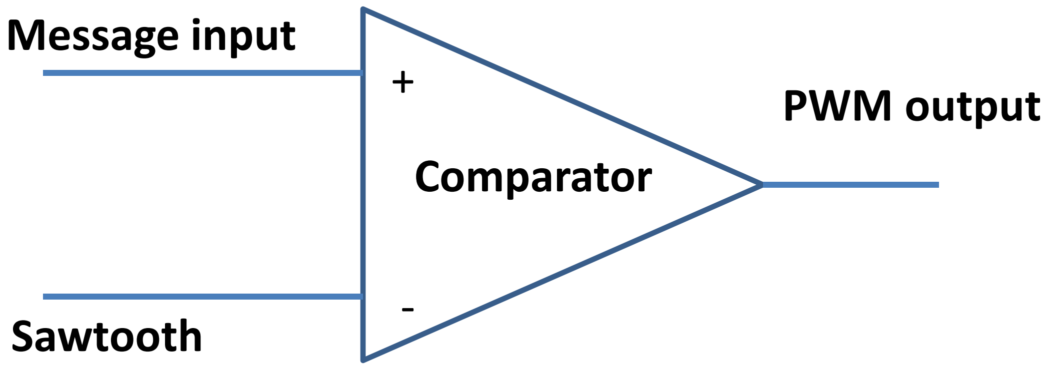
PAM signal → Reconstruction filter → Equalizer

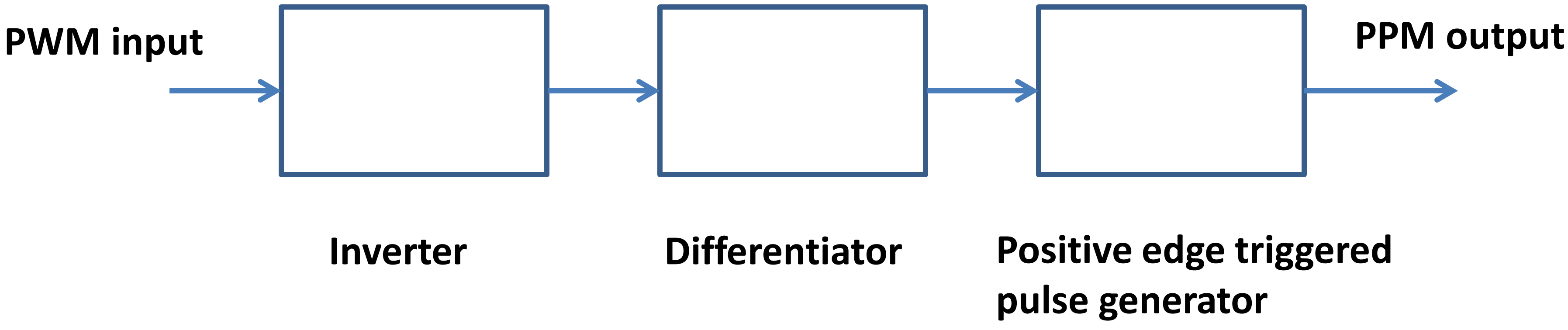
Equalizer compensates aperture effect and also compensates the attenuation caused by reconstruction filter.

Forms of Pulse Modulation

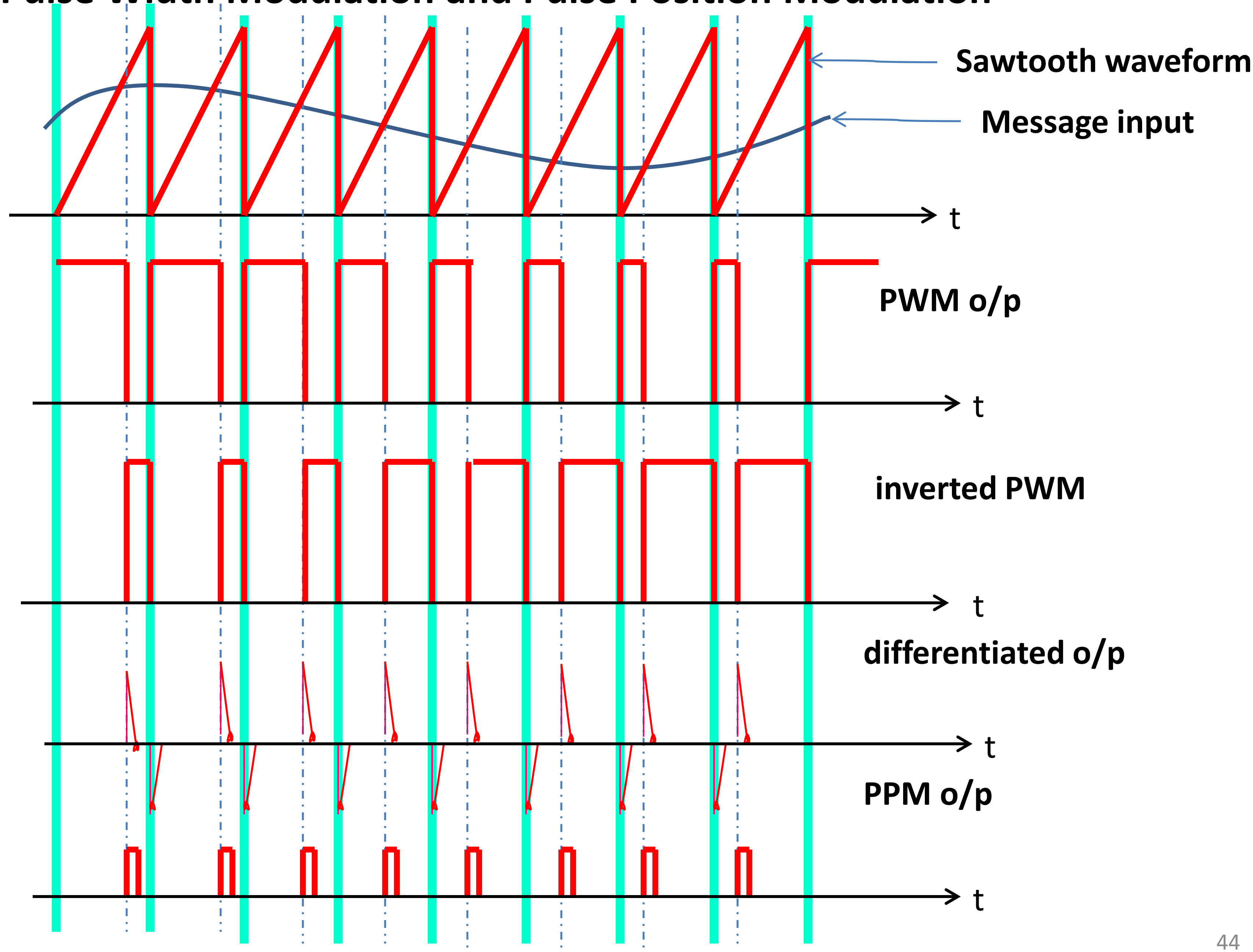
- PAM, PWM ad PPM Analog Modulation schemes.
- A parameter of the pulse is varied in accordance with message signal
- PAM- Amplitude – Analog
Width - Discrete
- PWM – Width- Analog
Amplitude – discrete
- PPM – Position – Analog
Amplitude - Discrete

PWM and PPM Generation





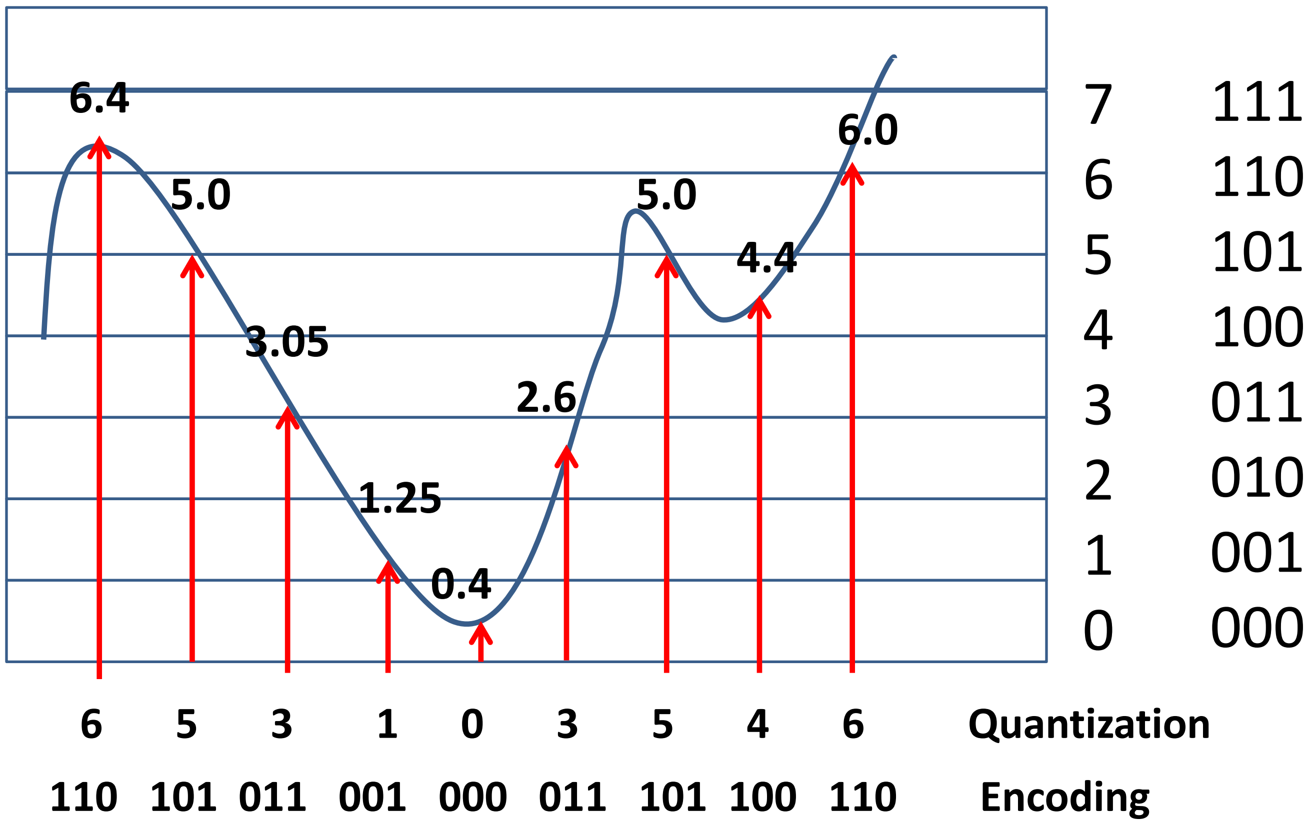
Pulse Width Modulation and Pulse Position Modulation



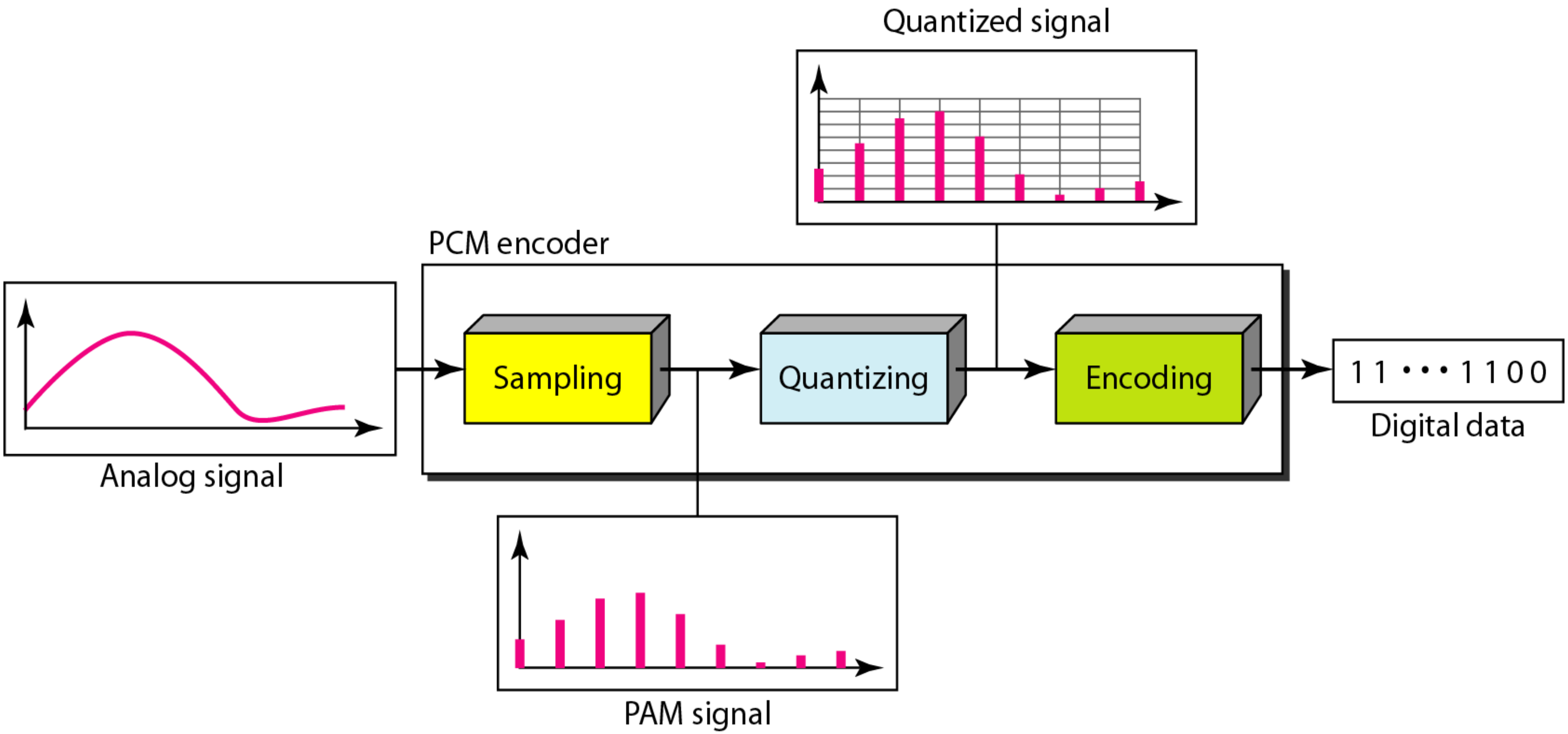
Pulse Code Modulation(1)

- Digital Scheme
- PCM is a method of converting an analog into digital signals.
- PAM, Quantization, Unique code word

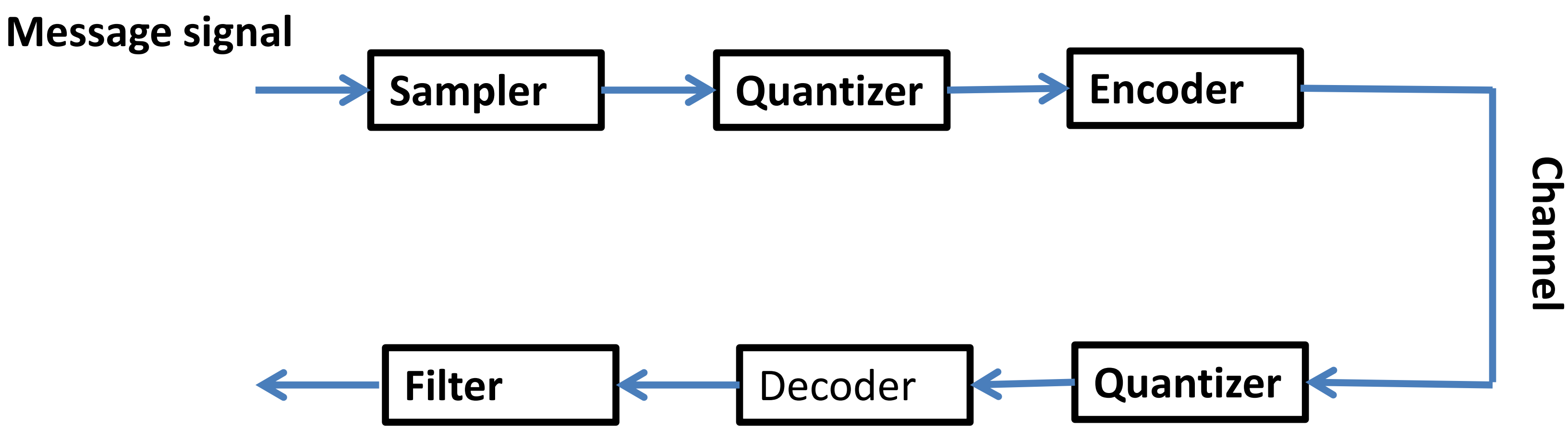
Pulse Code Modulation (2)

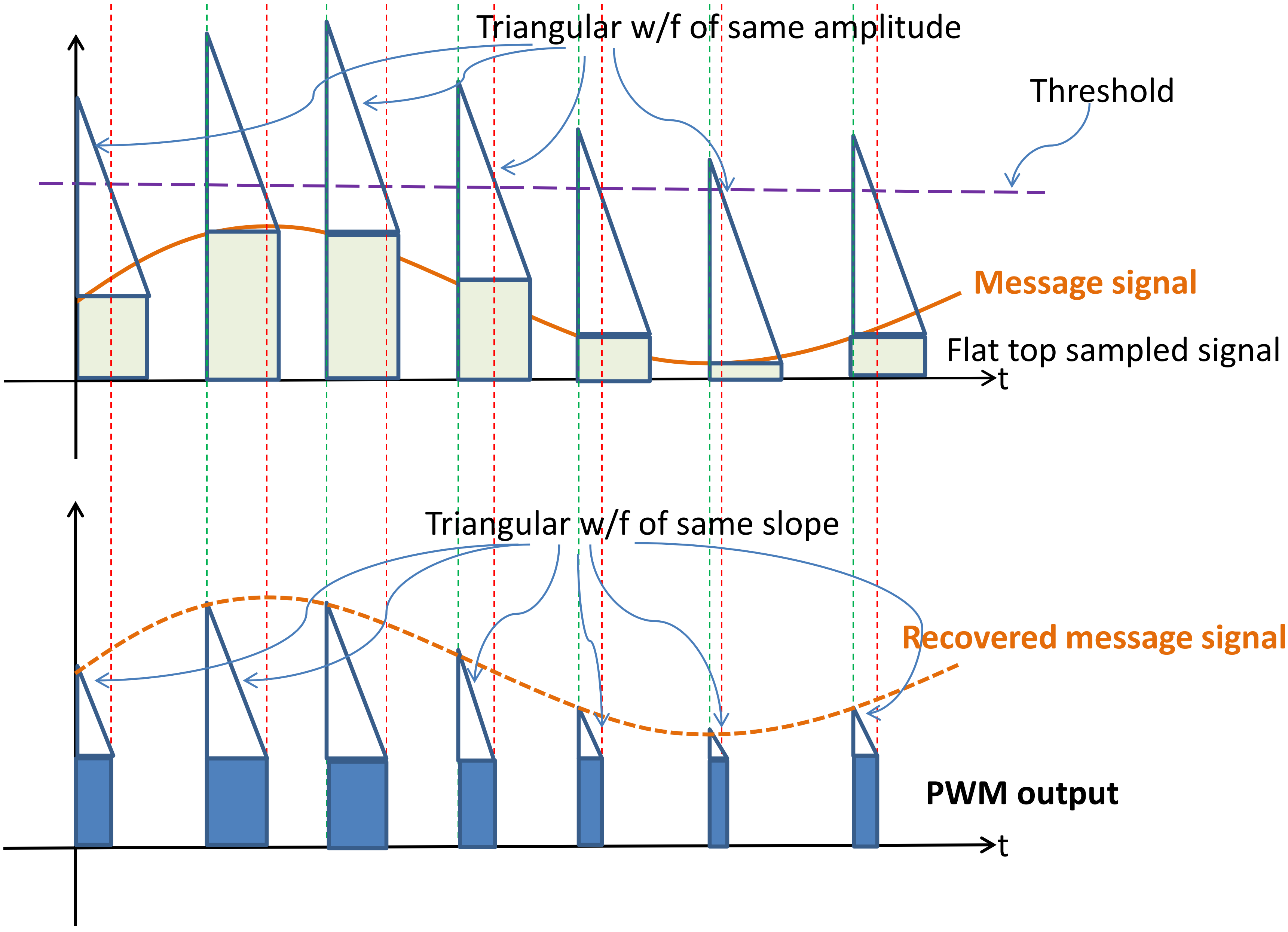


PCM generation



PCM Generation and Reception

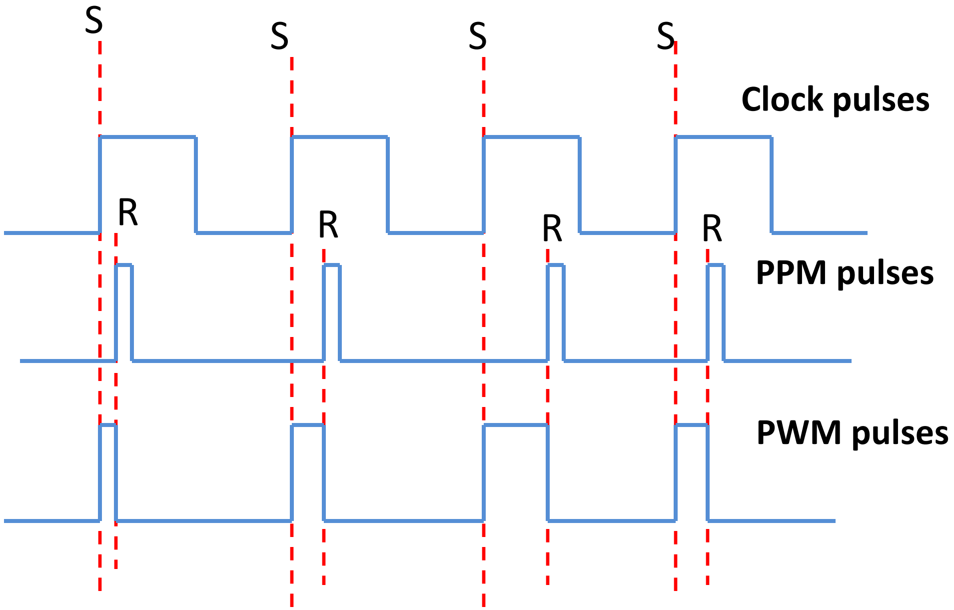




PWM generation (From PAM signal) and detection

PWM from PPM


PPM pulses
Clock pulses
R
S
Q
PWM pulses


S
S
S
S
Clock pulses
R
R
R
R
PPM pulses
PWM pulses

PPM Demodulation