Unit-6: Sampling

* Sampling definition: converting continuous time signal to discrete time signal.

$$x(t) = x(t) |_{t=0.75}$$

* Sampling interval: The time interval between 2.

samples is known as sampling interval (Ts)

Ts = 1/fs { fs = sampling frequency or rate }

* Time domain representation of sampling signal.

$$= \sum_{n=-\infty}^{\infty} x_s(t) = x_s(t) - \sum_{n=-\infty}^{\infty} s(t-n\tau s)$$

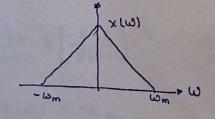
Sampling theorem of Band limited signals:

Statement: Any signal x(t) which is band limited

to wm Hz (ie x(w)=0: lwl-wm) can be completely

reconstructed back from its sample signal

taken at a rate ws > 2wm or fs > 2fm



Band width = 2 Wm

Here 2 wm is the nyquist rate.

Nyquist rate: It is the minimum sampling rate at which the signal can be sampled and can be completely reconstructed back from its samples without any distortions.

N.R = 2wm or 2fm { wm = max frequency of sign

Nyquist interval: (Time): The time interval blue 2 adjacent samples when the sampling rate is nyquist rate. N. I = $\frac{1}{N \cdot R} = \frac{1}{2 \cdot f_m} = \frac{1}{2 \cdot \omega_m}$

* Based on nyquist rate sampling is divided into

ws > 2 wm or fs > 2 fm

ws = 2wm or fs = 2fm

ws < 2 wm or fs < 2 fm

* frequency domain representation of sampling signal.

we know $x_s(t) = x(t) \cdot \sum_{n=-\infty}^{\infty} 8(t-n\tau s)$

Apply Forier Hansform on both sides.

* FT { xs(+)} = FT { x(+). } \$ (+-nTS)}

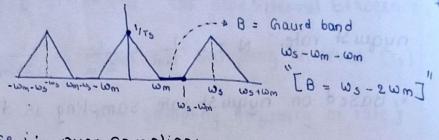
$$X_{s}(\omega) = \frac{1}{2\pi} \left[FT \left\{ x(H) \right\} \times FT \left\{ \sum_{n=\infty}^{\infty} \delta(H-nTS) \right\} \right]$$

$$= \frac{1}{2\pi} \left[x(\omega) \times \frac{2\pi}{T} \sum_{n=\infty}^{\infty} s(\omega-n\omega_{S}) \right]$$

$$= \frac{1}{TS} \left[x(\omega) \times \sum_{n=\infty}^{\infty} \delta(\omega-n\omega_{S}) \right]$$

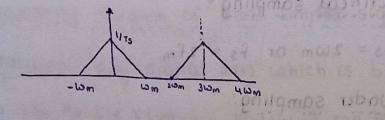
$$= \frac{1}{TS} \sum_{n=\infty}^{\infty} x(\omega-n\omega_{S})$$

 $X_{S}(\omega) = \frac{1}{T} \left[X (\omega + n\omega_{S}) + ... + X(\omega + \omega_{S}) + X(\omega) + X(\omega - \omega_{S}) + X(\omega - \omega_{S})$



case-i: Over sampling: Ws> 2 wm 3 200

$$X_{S}(\omega) = \frac{1}{T_{S}} \left[\dots + \chi(\omega) + \chi(\omega - 3\omega_{m}) + \chi(\omega - 6\omega_{m}) + \dots \right]$$



$$B = \omega_{m}$$

$$\text{for an order property monop property }$$

case-ii : critical sampling :

$$X_{S}(\omega) = \frac{1}{T_{S}} \left[\dots + \chi(\omega) + \chi(\omega - 2\omega_{n}) + \chi(\omega - 4\omega_{n}) + \dots \right]$$

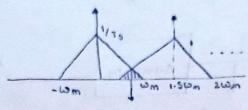
B - 2Wm - 2Wm

case in Under sampling

WS < LWm

Let Ws = 1.5 Wm

X5(W) = 1 [... + X(W) + X(W-1) + X(W-2) + ...]



Aliazing effect

Aliazing effect: Overlapping of replicas with original signal. In under sampling, lower frequencies of xs(w) overlap with higher frequencies of shifted x(w). This is/known as Aliazing effect. This overlapping leads to distortion & this is known as Aliazing effect.

* Aliazing occurs due to ...

when sampling rate is less than nyquist rate ws < N.A or 20m

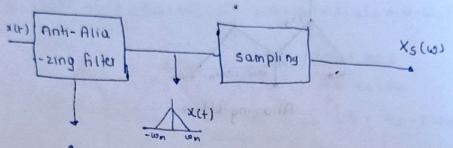
21 If the signal is not band limited to finite range

* for example, signal is like this...

1

original signal is not formed.

take a filter called low pass filter to cut the fluctual transport the filter considered should have frequencies from -wm to wm



by low pass filter.

as Allasino chicci.

Aliazing occurs due to

10 8 H 3 60

Find the nyquist rate of x(+) = sin 4000 TH

1000 = mw

Nyquist ws = 2wm

 $\omega_{s} = 2 \left(4000 \pi \right) \qquad \left\{ \omega_{s} = 2\pi f_{s} \right\}$ $\omega_{s} = 8000 \pi \qquad \left\{ \omega_{s} = 2\pi f_{m} \right\}$

Find the nyquist rate of x(t) = ,1+ cos 2000 nt + sin4000 nt.

10m1 = 2000 7 & 10m2 = 40007

0

Q

9

A

ws = 2wm = 8000T

Find the nyquist rate of Sinc 2000t.

$$Sinc 2000t = \frac{Sin 2000\pi t}{2000\pi t}$$

Wm = 20007

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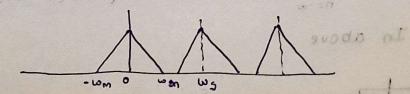
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Ws = 2 Wm = 4000 K (1) + (1) + (1) = (1) = (1)

Signal deconstruction or Recovery:

The process of getting back the original signal from the sample signal is called signal recon- Struction or signal recovery.

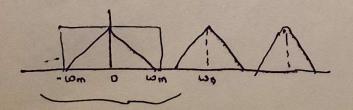
consider sample signal



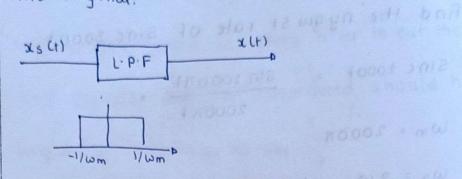
Filter of frequency - wm to wm of amplitude I

Now we apply it to sample signal

we get the original signal



Here the band width of low pass filter should be of the range won < B < ws - wm to retrieve the original.



$$x_{3}(t) = x_{5}(t) * h(t)$$

$$= (x(t) \cdot \sum_{n=-\infty}^{\infty} S(t-n\tau s)) * h(t)$$

$$= \left(\sum_{n=-\infty}^{\infty} x(t) \cdot S(t-n\tau s) * h(t)\right)$$

$$= \left[\sum_{n=-\infty}^{\infty} x(n\tau s) h(t-n\tau s)\right]$$

$$x_{T}(t) = \sum_{n=-\infty}^{\infty} x(nTS) 2fm \cdot Sin(2fm(t-nTS))$$

In above

$$h(t) = \frac{2fm}{2fm} \frac{\sin 2fmt}{2fmt}$$

we get the originalistiqual

* Band Pass signal:

A signal which has band of frequency from non-zero value to another non-zero value.

