

Q. Describe the need of Antialiasing filters.

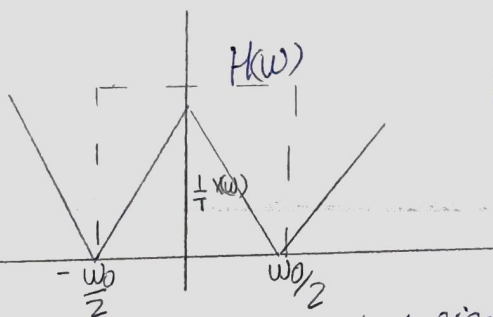
→ During recovery, while designing lowpass filter, it is essential to keep in mind of 2 points:

→ In practise real signal have infinite band width and it is necessary to make it into band limited signal. The real signal is filtered to get power rich signal.

→ Following Nyquist rate, the sampling frequency is high, which could keep higher frequency noise components from channel introducing aliasing, the lowpass filter removes such high frequency terms.

Q. For the example used in the experiment, explain how low pass filters recovers the message signal. Use spectrum diagram.

→ For the reconstruction of signal, we use a low pass filter to sampled signal.



The filter has a frequency response $H(\omega)$ and impulse response as

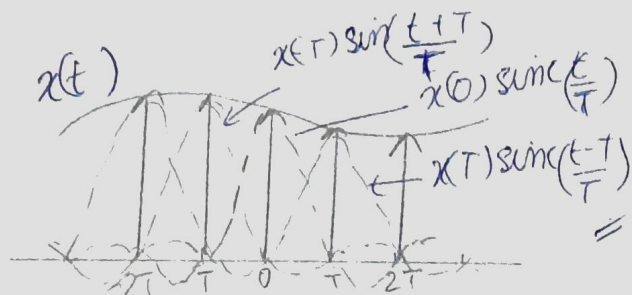
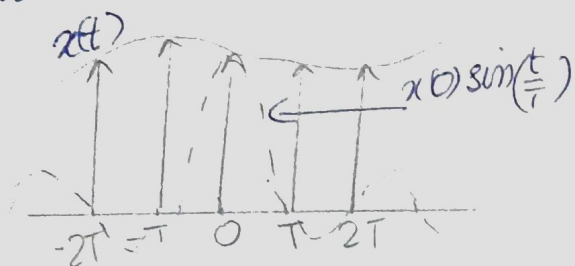
$$h(t) = \text{sinc}\left(\frac{t}{T}\right)$$

Thus the reconstructed signal can be given as-

$$\begin{aligned} x_r(t) &= \underbrace{x(t)}_{\text{sampled signal}} * h(t) = \sum_{n=-\infty}^{\infty} x(nT) \delta(t-nT) * h(t) \\ &= \sum_{n=-\infty}^{\infty} x(nT) \text{sinc}\left(\frac{t-nT}{T}\right) \end{aligned}$$

$$\begin{aligned} x[n] = x(t) \Big|_{t=nT} &\xrightarrow{\text{sinc Pulse generator}} x_r(t) = \sum_{n=-\infty}^{\infty} x[n] h(t-nT) \\ &= \sum_{n=-\infty}^{\infty} x[n] \text{sinc}\left(\frac{t-nT}{T}\right) \end{aligned}$$

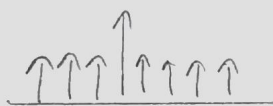
The reconstructed signal is a train of sinc pulses scaled by samples $x[n]$. The interpolated signal is a sum of shifted sincs weighted by the sample $x(nT)$. The sinc function shifted to nT is equal to one at nT & zero at all other samples.



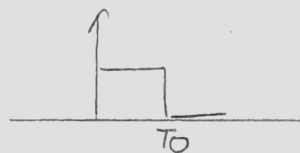
Q3. Explain how we recover larger amplitudes signals from a wave sampled with a higher duty cycle.

→ A wave sampled with a higher duty cycle implies a higher pulse width. Since this pulse is convoluted with the impulse train, it will be multiplied in the frequency domain i.e. sinc function.

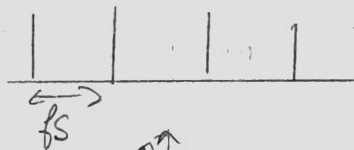
In time domain:



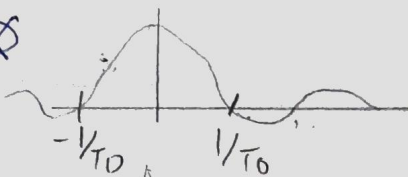
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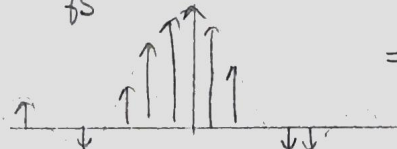
In frequency domain:



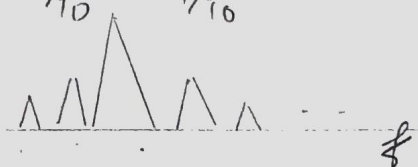
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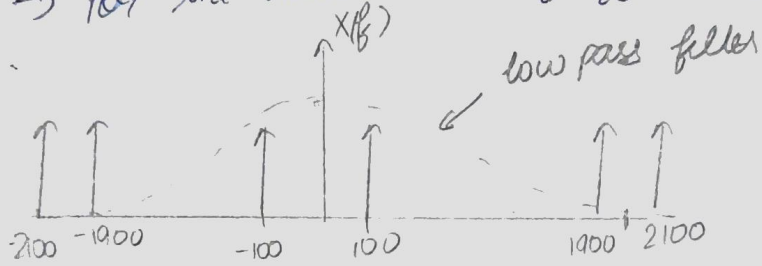
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Now as ' T_0 ' increases, Duty cycle \uparrow , the sinc waves which has a factor of T_0 , starts to achieve a higher central value & rate at which it dies out increases. Therefore the amp. of spectrum after sinc is convoluted with the signal \uparrow . Less recovery power of original signal is increased. This is how larger amplitudes signals are recovered.

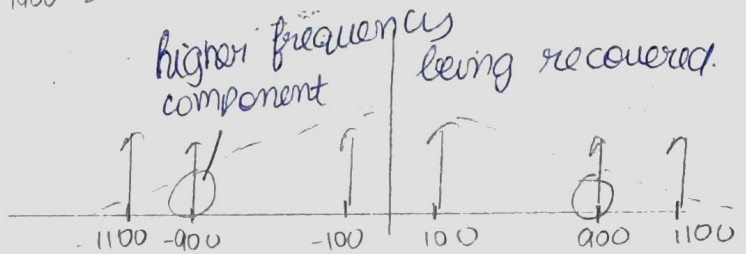
Q4. For the recovery system used in the experiment, changing the sampling frequency from $20f_m$ to $10f_m$ leads to the extra frequency signal in the recovered signal. Can this phenomenon be called aliasing? Explain.

→ For the initial case of $f_s = 20 \times f_m = 2000\text{Hz}$ ($f_m = 100\text{Hz}$)



• If $f_s = 10 \times f_m$.

Since the sampling frequency follows the Nyquist criterion i.e. $f_s \geq 2f_m$, this is not an case of aliasing. The problem in this case is of use of poor low pass filter. This is not an case of improper sampling hence it is not aliasing.



Q5. How can we fix the problem caused in the question above?

→ The poor choice of low pass filter in the above question cause the problem. Using a better performance low pass filter can be used to resolve the problem.

Q6. In practice, what is the sampling frequency used for audio & for speech transmission.

→ Audio transmission :- 44.1 KHz.

→ Speech transmission :- 8 KHz.

Q7. Briefly describe the concept of band pass sampling.
 → A band pass signal is a signal containing a band of frequencies that are not adjacent to (or not centered at) zero frequency i.e. lowest frequency in signal, $f_L > 0$ Hz



Here, the Bandwidth (BW), lowest frequency (f_L) & highest freq (f_H)

$$\Rightarrow BW = f_H - f_L$$

For sampling of such signal, the sampling rate (f_s):

$$\left[\frac{2f_H}{k} \leq f_s \leq \frac{2f_L}{k-1} \right]; \text{ where } k=1, 2, 3, \dots, N \text{ and}$$

$$N = \text{Integer of } \left[\frac{f_H}{BW} \right]$$

* This type of sampling allows the signal to be sampled at much lower rate than it is permitted if the Nyquist condition is used:

Eg: A band pass signal with f_L & f_H as 4 kHz & 6 kHz respectively. can be sampled at a rate of 4 kHz effectively in contrast to 12 kHz rate required as per Nyquist condition.

Q8. Briefly compare the ideal, flat top & natural sampling.

Ans. Ideal sampling: It is also known as instantaneous or impulse sampling. Train of impulse is used as a carrier. The sampling function is a train of impulses & principle used is multiplication principle.

Spectrum of ideally sampled signal is given as:

$$G(f) = f_s [\sum x(f - n f_s)]$$

Flat top sampling:- This sampling is practical in nature and is easily obtained. The top of the samples remain constant and is equal to the instantaneous value of the message signal $x(t)$ at the start of the sampling process. Sample & hold circuit are used. Spectrum is given as:

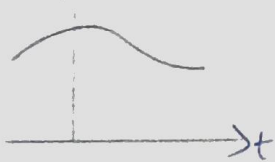
$$G(f) = f_s \cdot \left[\sum x(f - n f_s) \cdot \text{H}(f) \right]$$

Natural sampling: It is also a practical method with pulses having finite equal width T . Sampling is done in accordance with carrier signal (digital in nature). Spectrum is given as:

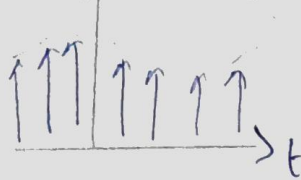
$$G(f) = \frac{A_T}{T_s} \sum \sin((n f_s T) \times (f - n f_s))$$

Sampled signal is multiplication of

message signal
($x(t)$)



Ideal sampling



Flat top sampling



Natural sampling

