

# Signal Processing - Assignment - 4

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## Q3. Digital Storage:

Sampling rate : 44.1 kHz, Bitrate : 32 bits

$$a) \text{ Sampling time period} = \frac{1}{44.1 \times 10^3} \text{ s} = 2.267 \times 10^{-5} \text{ s}$$

$$\text{Length of each song} = 5 \text{ min} = 300 \text{ s}$$

$$\Rightarrow \text{No. of samples} = \frac{300}{2.267 \times 10^{-5}} = 1.323 \times 10^7 \text{ Samples}$$

$$\Rightarrow \text{Size of each channel} = 32 \times 1.323 \times 10^7 = 4.2336 \times 10^8 \text{ bits}$$

$$\Rightarrow \text{Size of each song} = 2 \times 4.2336 \times 10^8 = 8.4672 \times 10^8 \text{ bits}$$

$$\Rightarrow \text{Total size occupied} = 7 \times 8.4672 \times 10^8 = 5.92704 \times 10^9 \text{ bits}$$

b) Since the quality of the audio must be the same, the no. of bits used for each sample must be the same (32 bits).

Since the audio signals have a maximum frequency, we can sample at a lower rate of 24 kHz (as per Nyquist theorem, sampling frequency must be atleast twice the max. frequency, i.e.,  $12 \times 2 = 24 \text{ kHz}$ .)

$$\text{Sampling time period} = \frac{1}{24 \times 10^3} = 4.167 \times 10^{-5} \text{ s}$$

$$\text{No. of samples} = \frac{300}{4.167 \times 10^{-5}} = 7.199 \times 10^6$$

$$\text{Size taken by each song} = 2 \times 32 \times 7.199 \times 10^6 = 4.6073 \times 10^8 \text{ bits}$$

$$\text{Total size Occupied} = 7 \times 4.6073 \times 10^8 = 3.22511 \times 10^9 \text{ bits}$$

$$\begin{aligned} \text{No. of bits saved} &= (5.92704 - 3.22511) \times 10^9 \text{ bits} \\ &= 2.70193 \times 10^9 \text{ bits} \end{aligned}$$

$$\text{Proportion of Space Saved} = \frac{2.70193 \times 10^9}{5.92704 \times 10^9} \times 100 \approx 45.586\%$$

$\therefore$  Using a reduced sampling frequency of 24 kHz reduces the space needed by 45.586 %.

Q4. Continuous-time to Discrete-time :-

a) To Prove: If the frequency  $f_s$  is above the Nyquist rate for  $x(t)$ , then it is still above the Nyquist frequency for  $y(t)$ , where  $y(t) = x(t) * h(t)$ ,  $\forall$  LTI system.

The o/p of any LTI system  $y(t)$  is denoted as  $y(t) = x(t) * h(t)$ .

In the Fourier Domain, the equation becomes,

$$Y(\omega) = X(\omega) \cdot H(\omega)$$

Since  $x(t)$  is band-limited,  $X(\omega) = 0 \forall |\omega| \geq B, B \in \mathbb{R}$ .

(Let the bandwidth of  $x(t)$  be  $B$ )

$$\Rightarrow Y(\omega) = 0 \forall |\omega| \geq B$$

$\Rightarrow y(t)$  is band-limited with maximum possible bandwidth  $B$ .

Since the Nyquist frequency of a band-limited signal depends on its bandwidth only and both  $x(t)$  and  $y(t)$  have a maximum possible bandwidth of  $B$ ,  $f_s$  will still be above the Nyquist rate for  $y(t)$ .

b) To find:  $h[n]$  such that  $y[n] = h[n] * x[n]$ , where  $y[n]$  and  $x[n]$  are the sampled  $y(t)$  and  $x(t)$ .

$$\begin{array}{ccc} \text{let } y[n] & & Y_D(\omega) \\ & \xleftrightarrow{\text{DTFT}} & \\ x[n] & & X_D(\omega) \\ h[n] & & H_D(\omega) \end{array}$$

$$\Rightarrow Y_D(\omega) = X_D(\omega) H_D(\omega)$$

Let the sampled version of  $x(t)$  be  $x_s(t)$ .

$$x_s(t) = \sum_{k \in \mathbb{Z}} x(t) \delta(t - n/f_s)$$

The o/p of  $x_s(t)$  is ,

$$y_s(t) = h(t) * x_s(t)$$

$$y_s(t) = \int_{-\infty}^{\infty} h(\tau) x_s(t - \tau) d\tau$$

$$= \int_{-\infty}^{\infty} h(\tau) \sum_{n \in \mathbb{Z}} x(t - \tau) \delta(t - n/f_s - \tau) d\tau$$

$$= \sum_{n \in \mathbb{Z}} \int_{-\infty}^{\infty} h(\tau) x(t - \tau) \delta(t - n/f_s - \tau) d\tau$$

$$= \sum_{n \in \mathbb{Z}} h(t - n/f_s) x(t - (t - n/f_s))$$

$$y_s(t) = \sum_{n \in \mathbb{Z}} x(n/f_s) h(t - n/f_s)$$

$$\Rightarrow y_s(t) = \sum_{n \in \mathbb{Z}} x[n] h(t - n/f_s) \quad \left[ x[n] = x(nT_s) = x\left(\frac{n}{f_s}\right) \right]$$

W.k.t ,  $y[m] = y_s(mT_s) = y_s(m/f_s)$

$$\Rightarrow y[m] = \sum_{n \in \mathbb{Z}} x[n] h(m/f_s - n/f_s)$$

$$\Rightarrow y[m] = \sum_{n \in \mathbb{Z}} x[n] h((m-n)/f_s)$$

$$\Rightarrow y[m] = x[m] * h(m/f_s)$$

Also, for discrete-time systems we know that,

$$y[m] = x[m] * h_d[m]$$

On comparing, we see  $h_d[m] = h(m/f_s)$

$\therefore h[n] = h(n/f_s)$ , i.e., the impulse response in discrete-time is the sampled signal of the continuous-time impulse response.