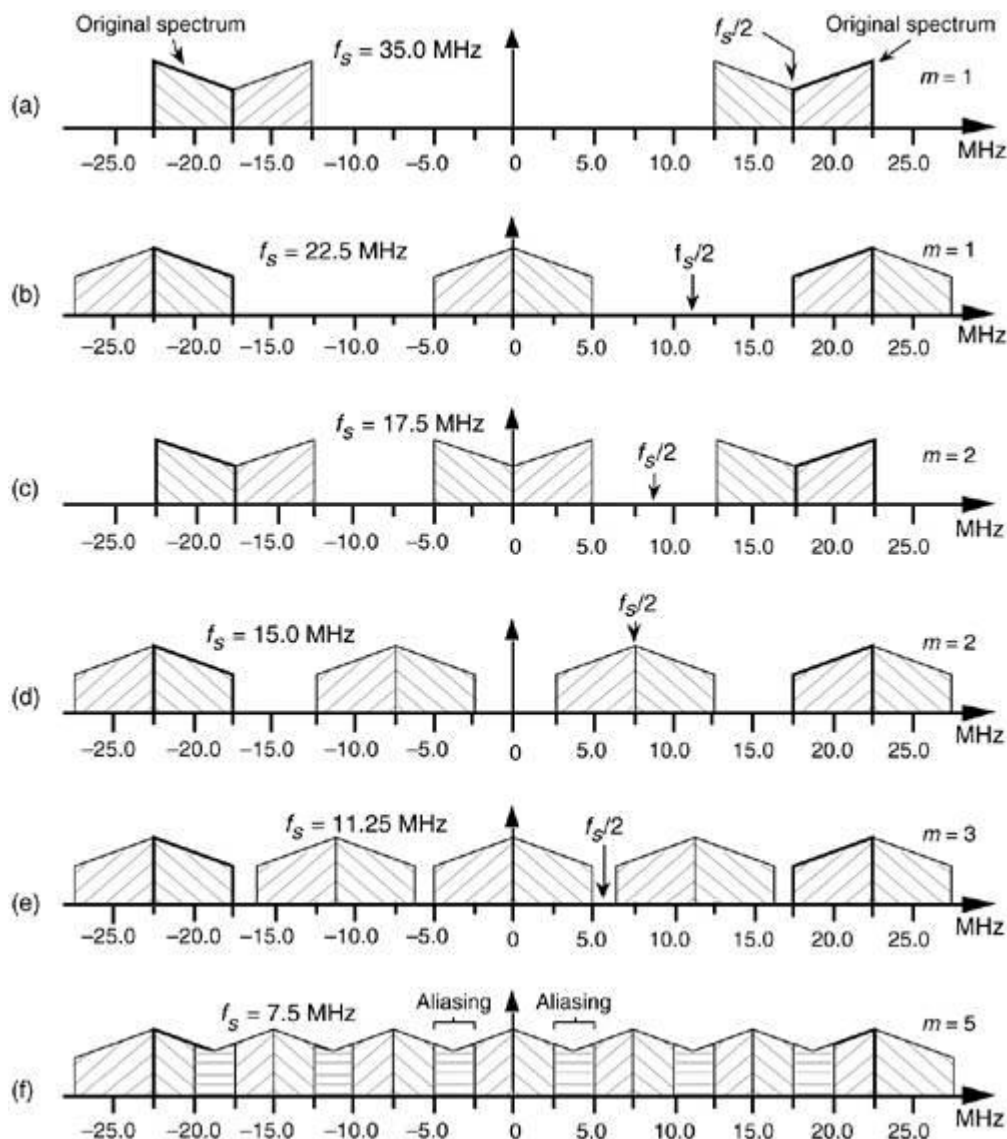


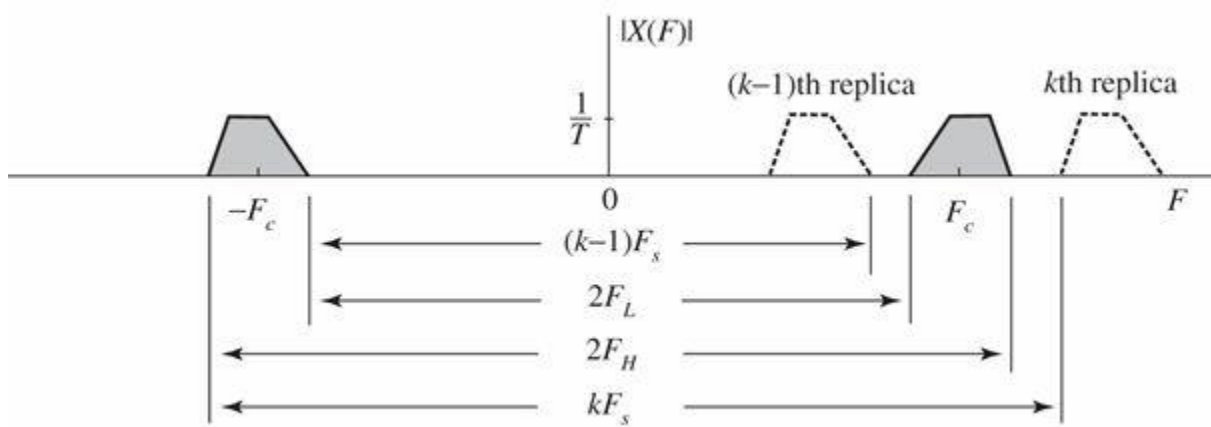
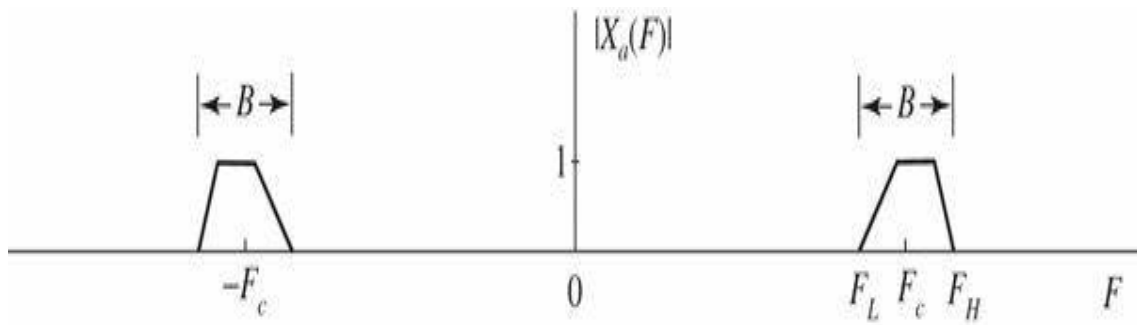
THEORY

SAMPLING THEORM

Statement: A continuous time signal can be represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal i.e. $f_s \geq 2f_m$.

Thus the minimum sampling frequency of a bandpass signal can be derived henceforth as:-

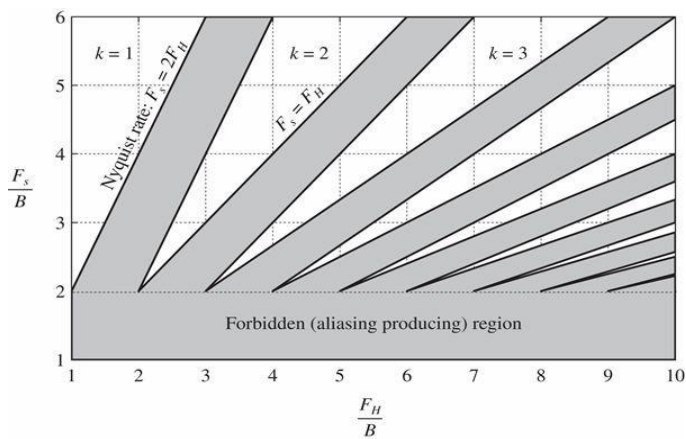




- $2 * f_h \leq k * F_s$

- $(k - 1) * F_s \leq 2 * f_l$

$\Rightarrow 2 * \frac{f_h}{k} \leq F_s \leq 2 * \frac{f_l}{k-1}$ where k is an integer given by $1 \leq k \leq \left\lfloor \frac{F_H}{B} \right\rfloor$



For $k=1$, we obtain,

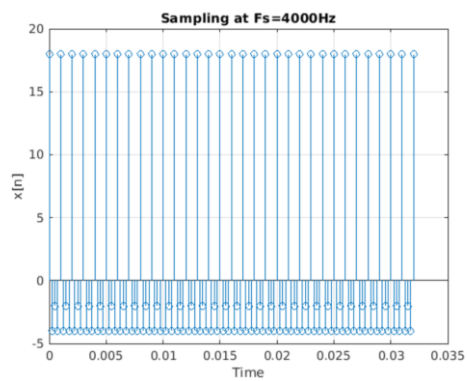
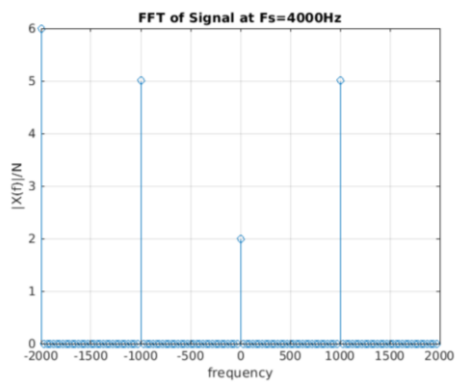
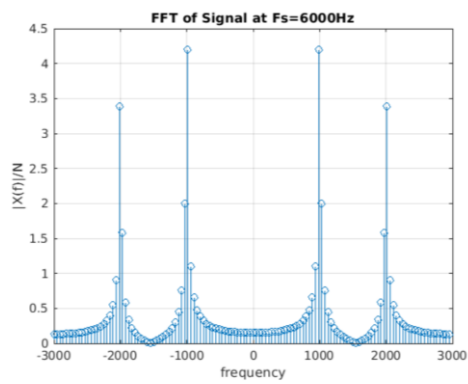
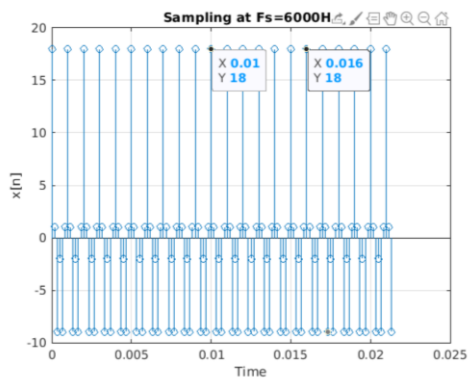
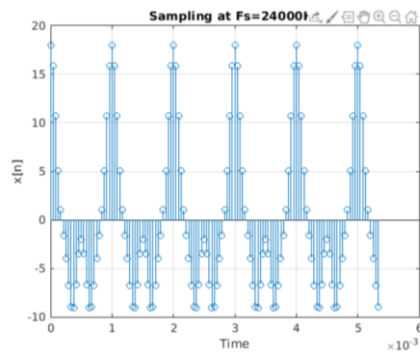
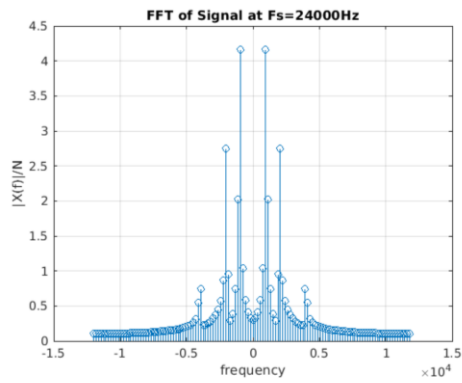
$$2f_H \leq F_s \leq \infty$$

Theoretically, the minimum sampling rate is $F_s = 2B$

MATLAB CODE

```
N = 128;
i=0;
for Fs=[24000,8000,6000,4000]
    % Generating the Function
    t=0:1/Fs:N/Fs;
    x=10*cos(2*pi*1000*t)+6*cos(2*pi*2000*t)+2*cos(2*pi*4000*t);
    figure(2*i+1);
    stem(t,x);
    xlabel('Time');
    ylabel('x[n]');
    title(['Sampling at Fs=',num2str(Fs),'Hz'])
    grid on;
    % Finding The N point FFT for the above Signal
    % where N =128
    X=fft(x,N);
    X=abs(X)/N;
    Y=fftshift(X);
    a=-Fs/2:Fs/N:Fs/2-Fs/N;

    %Plotting the FFT Plot
    figure(2*i+2);
    stem(a,Y);
    xlabel('frequency');
    ylabel('|X(f)|/N');
    title(['FFT of Signal at Fs=',num2str(Fs),'Hz'])
    grid on;
    i=i+1;
end;
```



RESULT

$F_{s\min}=6000\text{Hz}$

FROM THE TIME PLOTS :-

1) We can see from figure where $F_{s1}(24\text{kHz}) > F_{\text{min}}$, the sampling gives $x[n]$ almost resembles the actual signal $x(t)$.

2) While in figure where $F_{s2}(6\text{kHz}) = F_{\text{min}}$, the $x[n]$ is not as better as that as F_{s1} .

3) In figure, where the $F_{s3}(4\text{kHz}) < F_{\text{min}}$ we can see $x[n]$ doesn't resemble $x(t)$ at all.

4) We can conclude that as we increase the frequency (greater than F_{min}), $x[n]$ tends to $x(t)$.

FROM FFT results of the signal :-

5) We can see from figure where $F_{s1} > F_{\text{min}}$, the peaks are present at frequencies

$\pm 1000\text{Hz}, \pm 2000\text{Hz}, \pm 3000\text{Hz}, \pm 4000\text{Hz}$. This shows that there are no intervening of signals.

6) In figure where $F_{s2} = F_{\text{min}}$, we find peaks at $\pm 3000, \pm 2000, \pm 1000$. This also shows that there are no intervening of signals.

7) In figure where $F_{s3} < F_{\text{min}}$, we find a peak at 0Hz . This shows that there is intervening of signals which is why there a peak at 0Hz and hence aliasing has occurred.

8) We can conclude that the sampling frequency must be greater than Minimum frequency (i.e $F_{\text{min}} = 2B$) in order to avoid aliasing and to recover the original signal from the sampled signal.

PART -2

- **Sampling at a frequency higher than nyquist rate is called Over Sampling. Over sampling leads to an increase in computation, but it comes along with some advantages. The following are its practical applications:**
 1. Medical images are over sampled to avoid the loss of information. The reconstructed images shouldn't lose any high frequency signal components of the original signal as it may cost lives.
 2. To sample a signal at F_s we must pass the original signal through anti-aliasing filter which filters out frequency components greater than $F_s/2$. For example, let us consider $F_s = 24\text{KHz}$. Let us consider a filter cut-off frequency at 18KHz . That implies the filter attenuates the frequencies above 24KHz . If the signal is over sampled i.e at 192 KHz , the filter needs to attenuate the signal above 96KHz . Thus we can design a relatively simple circuit as a filter.
 3. Quantization noise power is spread from 0 to $F_s/2$. If we assume quantisation of the noise levels to be same with and without over-sampling, then the frequency spread of the same power is more in case of over-sampling. Thus, the input signal occupying a range much narrower than this band will experience less noise effect. Thus, over-sampling will reduce the effect of quantisation noise on input signal.

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