Tribhuvan University

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Telecommunication Lab 1

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Date: Feb 28, 2025

a) **Generate three 3.5 kHz sine signals (3 seconds duration), first signal at 12 kHz sample frequency, second signal at 8 kHz sample frequency and third signal at 6 kHz.**
b) **On the same graph, use the plot function to display the three signals versus t in the range $0 \le t \le 5$ msec.**
c) **Listen to the three signals one after another using the function soundsc(x, fs).**
d) **Give your interpretation of this listening.**

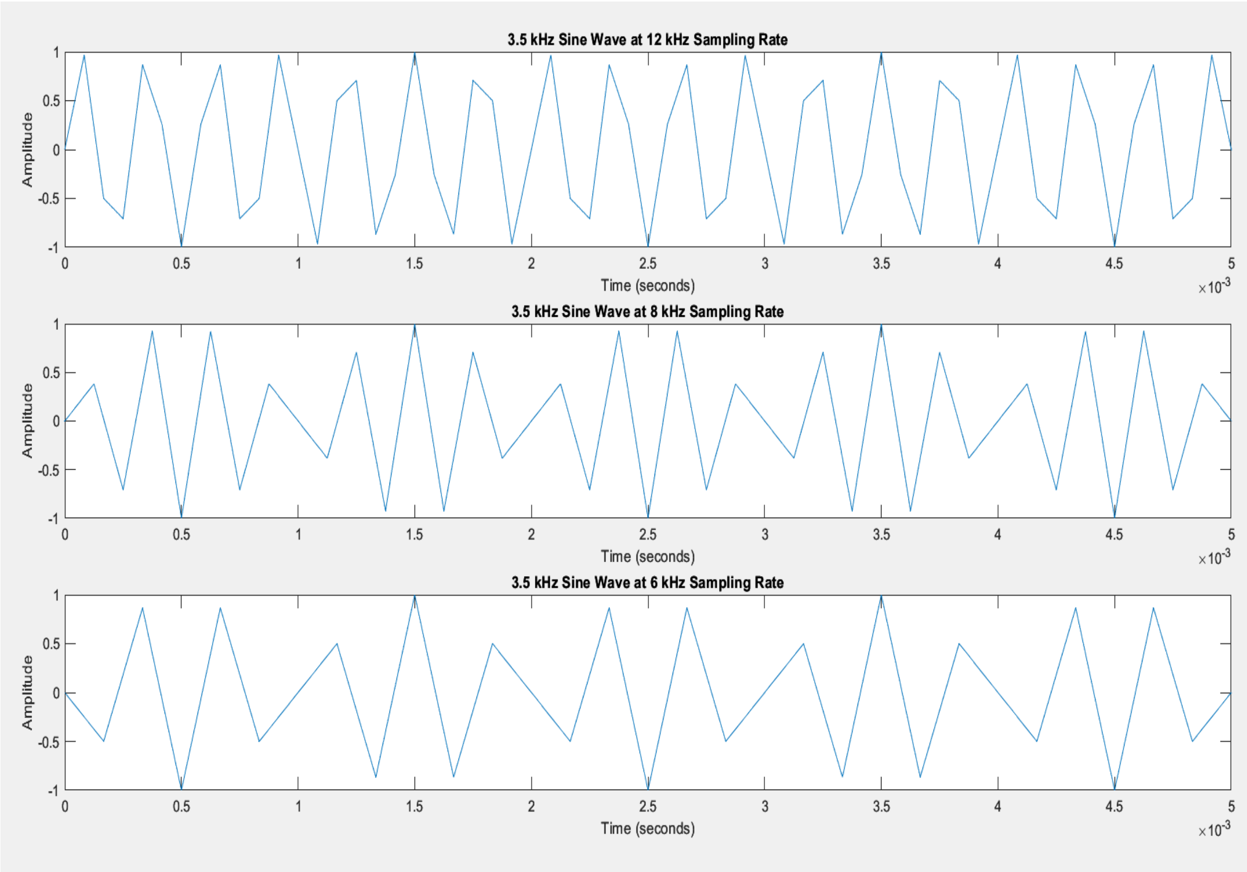
In digital signal processing, the Nyquist-Shannon Sampling Theorem plays a fundamental role in determining the appropriate sampling rate for accurate signal reconstruction. This theorem states that to avoid distortion and loss of information, the sampling rate must be at least twice the highest frequency present in the signal, known as the Nyquist rate. In this experiment, a 3.5 kHz sine wave is sampled at three different rates: 12 kHz, 8 kHz, and 6 kHz, each demonstrating the impact of varying sampling frequencies on the quality of the signal. When the sampling rate satisfies the Nyquist criterion, the signal is reconstructed accurately. However, when the sampling rate falls below the Nyquist rate, aliasing occurs, leading to distortion and a misrepresentation of the signal's frequency. This setup provides an opportunity to explore the relationship between sampling rates and signal fidelity, visually through waveforms and audibly by listening to the effects of each sampling rate on the reproduced sound.

Source Code:

```
duration = 3;
f_signal = 3500;
fs1 = 12000;
fs2 = 8000;
fs3 = 6000;
t1 = 0:1/fs1:duration-1/fs1;
t2 = 0:1/fs2:duration-1/fs2;
t3 = 0:1/fs3:duration-1/fs3;

signal1 = sin(2*pi*f_signal*t1);
signal2 = sin(2*pi*f_signal*t2);
signal3 = sin(2*pi*f_signal*t3);

time_limit = 0.005;

figure;

subplot(3,1,1);
plot(t1, signal1);
xlim([0 time_limit]);
title('3.5 kHz Sine Wave at 12 kHz Sampling Rate');
xlabel('Time (seconds)');
ylabel('Amplitude');

subplot(3,1,2);
plot(t2, signal2);
```

```
xlim([0 time_limit]);
title(‘3.5 kHz Sine Wave at 8 kHz Sampling Rate');
xlabel('Time (seconds)');
ylabel('Amplitude');

subplot(3,1,3);
plot(t3, signal3);
xlim([0 time_limit]);
title(‘3.5 kHz Sine Wave at 6 kHz Sampling Rate');
xlabel('Time (seconds)');
ylabel('Amplitude');

disp('Playing first signal (3.5 kHz at 12 kHz sample rate)...');
soundsc(signal1, fs1);
pause(duration + 1);
disp('Playing second signal (3.5 kHz at 8 kHz sample rate)...');
soundsc(signal2, fs2);
pause(duration + 1);
disp('Playing third signal (3.5 kHz at 6 kHz sample rate)...');
soundsc(signal3, fs3);
pause(duration + 1);
```

Output:



Interpretation of the Listening Experience:

i. At 12 kHz Sampling Rate:
   a. The sine wave sounds clear and well-represented, as the Nyquist criterion (sampling rate $\geq 2 \times$ signal frequency) is satisfied.
   b. Since 12 kHz > 2 × 3.5 kHz = 7 kHz, this means there is sufficient sampling to reconstruct the signal accurately without aliasing.
   c. The sound is smooth, without distortion.

ii. At 8 kHz Sampling Rate:
   a. The signal is still above the Nyquist rate (8 kHz > 7 kHz), so it is still be clear.
   b. However, it is closer to the Nyquist limit, meaning there may be a slight reduction in fidelity compared to the 12 kHz version.
   c. Some high-frequency details can be lost, but the overall shape of the waveform is still preserved.

iii. At 6 kHz Sampling Rate:
   a. This sampling rate is below the Nyquist limit (6 kHz < 7 kHz), leading to aliasing.
   b. Aliasing means that the signal will not be reconstructed correctly, and the perceived frequency may be different from the intended 3.5 kHz.
   c. Instead of hearing a 3.5 kHz tone, a lower-frequency artifact (folded frequency) is heard, making it sound different or distorted.

This experiment demonstrates the importance of choosing an appropriate sampling rate to avoid signal degradation due to undersampling.