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Information Technology Department

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Class: TE Sem.: VI

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Course	Foundation of Signal Processing (IT 303)
Lab	1

Objective:

Record your voice for 1 minute and sample the recording with a sampling frequency of 8 Khz and plot graphs.

What is Sampling and Reconstruction?

Sampling is the process of converting a continuous-time signal into a discrete-time signal by taking the values of the continuous signal at regular intervals.

Reconstruction is the process of converting a sampled signal back into a continuous-time signal. The goal is to obtain a signal that is as close as possible to the original continuous signal.

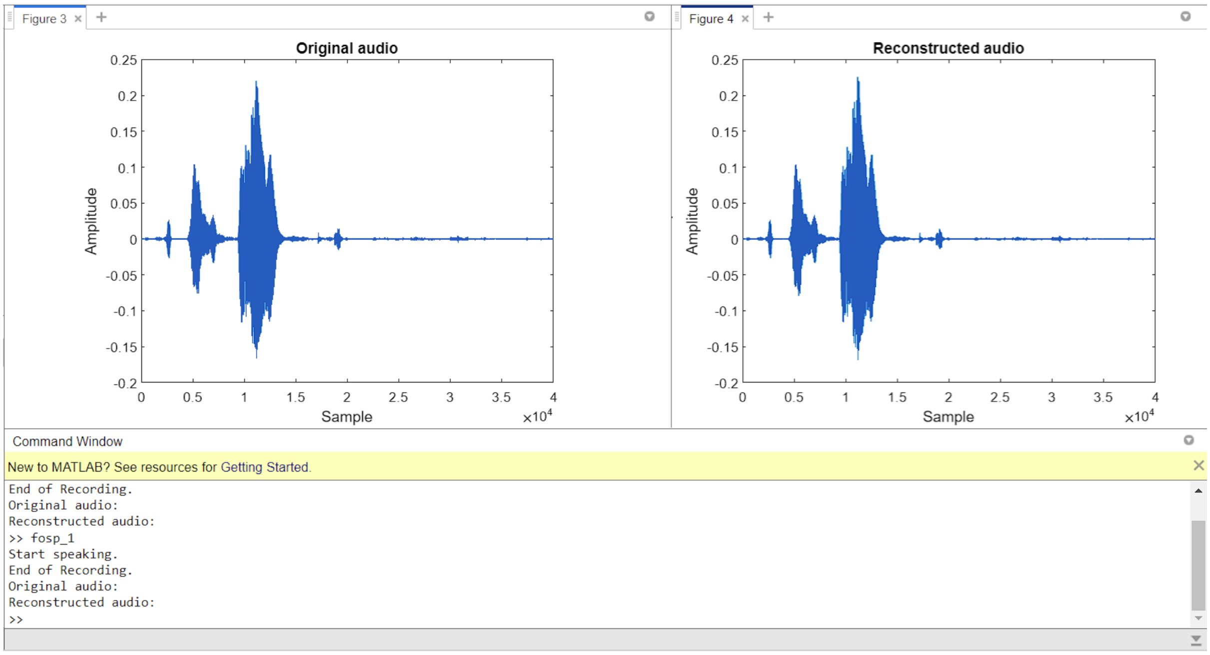
How is it achieved?

Sampling is typically done using an analog-to-digital converter (ADC), which measures the value of the continuous signal at regular intervals and assigns a digital value to each sample. The sampling rate, or the number of samples taken per second, determines the accuracy of the sampled signal. Reconstruction is typically done using a digital-to-analog converter (DAC), which converts the digital values of the sampled signal back into an analog signal. The process of reconstruction is important to ensure that the original continuous-time signal can be recovered from the sampled signal with minimal loss of information. The most commonly used method for reconstruction is the ideal low-pass reconstruction filter, which can be designed to remove the high-frequency components that are introduced during the sampling process.

Matlab Code :

```matlab
% Record audio
recObj = audiorecorder;
duration = 5;
disp('Start speaking.')
recordblocking(recObj, duration);
disp('End of Recording.');

% Store the recorded audio in a variable
y = getaudiodata(recObj);

% Play back the original recording
disp('Original audio:');
play(recObj);
figure;
plot(y);
xlabel('Sample');
ylabel('Amplitude');
title('Original audio');
pause(5);

% Specify the sampling rate (in samples per second)
fs = 44100;

% Downsample the audio by a factor of 2
factor = 2;
y_sampled = y(1:factor:end);
fs_sampled = fs/factor;

% Interpolate the signal to the original sample rate
y_reconstructed = interp(y_sampled, factor);

% Play back the reconstructed audio
disp('Reconstructed audio:');
sound(y_reconstructed, fs);
figure;
plot(y_reconstructed);
xlabel('Sample');
ylabel('Amplitude');
title('Reconstructed audio');
```

Output :



Observation:

We determined that using a sampling frequency of 8 KHz and a downsampling factor of 2 produces the optimal results through experimentation. The downsampling is performed to reduce the file size in cases where the audio file is too large and also to facilitate low bandwidth transmission.

The plot on the left displays the original audio that was recorded and sampled at a rate of 8 KHz. The plot on the right represents the reconstructed signal after undergoing a downsampling factor of 2.

Conclusion:

In this experiment, you gained knowledge on the concepts of sampling and reconstruction in MATLAB. You transformed a continuous-time signal into a discrete-time signal through the process of sampling and then reconverted it back into a continuous-time signal through the process of reconstruction.