Dynamic Measurement and Signal Processing
Lab 6: Order Tracking

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Abstract

Hello!

Some comment

name

name

name

1 Background and Objectives

The purpose of this lab is to understand and apply order tracking algorithms in real data. A Fast Fourier Transform (FFT) based and a Time Varying Discrete Fourier Transform (TVDFT) are used to visualize the order of a test bench motor. This section will overview important aspects of rotating machine testing and the two order tracking methods.

When working with rotating machines, .

rotating machine overview

The FFT order tracking .

background of FFT OT

$$R = \frac{V_{high} - V_{low}}{2^Q} \tag{1}$$

The TVDFT order tracking .

Show FFT equation

$$R = \frac{V_{high} - V_{low}}{2^Q} \tag{2}$$

background of TVDFT OT

2 Apparatus

Show TVDFT equation

The material used in the lab activity is listed bellow:

- Speed controlled DC motor with a unbalanced mass;
- Tachometer;
- Accelerometer;
- Data Acquisition NI cDAQ 9172;
- Data Acquisition NI 9234;
- Desktop computer with MATLAB, LabVIEW and device drivers installed ...

3 Experimental Procedures

The lab activity contains two components: an in-lab experimental procedure and a simulation performed with the MATLAB software. Both activities are described in detail in the lab manual, so this section describes the exceptions found in the activity.

It wasn't possible to decrease the signal generator output amplitude such that it would be mixed in the noise floor. Because the amplitude value is adjusted in discrete steps and the minimal value is well above the noise floor.

4 Data Summary

A sinusoidal signals with frequency of 356 Hz and amplitude of 1.24 V_{RMS} were acquired during the

laboratory section. The time history of the experimental and MATLAB generated data was added to the Appendix, Figure 2, 3 and 4. The experiment run using a range of $\pm 5\ V$ is presented in the frequency domain in the Figure 1.

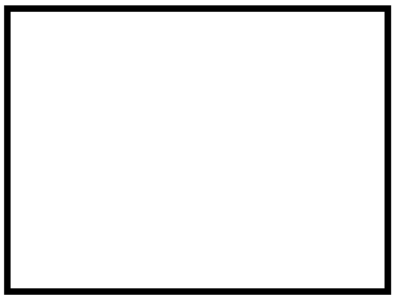

Figure 1: Overlayed plot of experimental and simulated data in the frequency domain for $\pm 5\ V$ of range and 8, 12 and 24 number of bits.

5 Interpretation and Analysis

Equation 1 demonstrates that a broad input range or low number of bits decrease the resolution. The time history of the simulation data confirms this statement: Figure 2 and 3 both show that the samples are closer to the nominal curve as the number of bits increase, and this distance is smaller in the experiment with the smaller range, $\pm 5\ V$. Also, voltage saturation was observed in the Figure 4, where the simulated signals saturates on $\pm 0.75\ V$.

Another point from Figure 1 is that the noise floor of the experimental data is higher than the simulated 24 bits counterpart, even thought the acquisition system also has a 24 bits precision. The experimental data also accounts with noises in the environment and in the DAQ electronics.

The oscilloscope's experimental SNR analysis was not possible because the signal generator couldn't reduce the sine amplitude low enough. But according to their respective data-sheet, the oscilloscope has a 8 bit resolution, while the acquisition board has 24 bits. And both systems have $\pm 5\ V$ as the narrower acquisition range. So according to Equation 1, the minimum signal amplitude detectable is 19 mV and 298 ηV, respectively. This follows that the NI system has higher SNR.

Finally, in respect to the critical thinking section, for a sine wave with period of $T = 56\mu s$ and RMS voltage of $A_{RMS} = 0.35\ V$, the desired sampling rate, F_s, voltage range, L, and number of bits, Q, for the requirements described in the manual are

$$F_s = 524288\ Hz \approx 2^{nextpow2(20/T)} \tag{3}$$

$$L = \pm 1\ V \approx \pm 2\ (A_{RMS}\ \sqrt{2}) \tag{4}$$

$$Q = 8 > log_2 20 \tag{5}$$

being $n = nextpow2(x)$ a function that returns the next power of 2 of x, or in other words, an integer n, $2^{n-1} < x \leq 2^n$.

6 Conclusions

On this assignment a physical signal was measured thought a data acquisition system in order to observe the effects of quantization. This data was also compared to simulated signals on MATLAB. Finally, the studies of acquisition parameters were explored with an example case.

From the frequency analysis of the discretized sine waves, higher SNR was observed in signals with higher number of bits and higher input voltage band usage. But experimental data has an inherent process noise floor that might be more significant than the quantization error, thus limiting the SNR to a maximum value. Check [2]. Testing saving capabilities.

References

[1] Steven W. Smith. *Digital Signal Processing. A Practical Guide for Engineers and Scientists.* Demystifying Technology Series. Newnes, 2003. ISBN: 9780750674447.

[2] Albert Einstein. "Zur Elektrodynamik bewegter Körper. (German) [On the electrodynamics of moving bodies]". In: *Annalen der Physik* 322.10 (1905), pp. 891–921. DOI: http://dx.doi.org/10.1002/andp.19053221004.

Appendices

A Effects of Number of Bits and Voltage Range

Figure 2: Time history of the experimental data and quantized simulated data for a ADC input range of $\pm 5\ V$

Figure 3: Time history of the experimental data and quantized simulated data for a ADC input range of $\pm 300\ V$

Figure 4: Time history of the experimental data and quantized simulated data for a ADC input range of $\pm 0.75\ V$

B Main Script

```
%% DSP - Laboratory #6
% Guilherme Aramizo Ribeiro
%% Definitions
rpm = 15000; % motor operating speed [rpm]

%% Getting calibration factor
% Calibrating a signal
%     Input frequency range: [Fmin Fmax] = [149.2 149.2] Hz
%     Input voltage range: [Vmin Vmax] = ?
%
%     Fs = 13.1072e6/(256*31) > 300 Hz = 2*Fmax
%     winsize = 476, minimize leakage
%     noavg = 32

% acquisition parameters
Fs = 13.1072e6/(256*31);
winsize = 476;
noavg = 32;
range = 5;
bits = 24;

% Simulate signal
time = linspace(0, (winsize*(noavg+2))/Fs, winsize*(noavg+2)).';
calib_factor_ = 9.81*3/5; % (m/s^2)/V
y = myDSP.discretize( ...
    (9.8*sqrt(2))*sin(2*pi*149.2*time) / calib_factor_, bits, range);

%{
plot(time*1e3, y, 'o-')
xlim([0 30])
xlabel('time [ms]'); ylabel('acc voltage [V]')
%}

% Visualize auto-power
yreshaped = myDSP.reshape(y, winsize, 0);

win = window(@flattopwin, winsize);
gain = [1; 2*ones(winsize-1, 1)] / (mean(win)*winsize);

GY = bsxfun(@times, fft(bsxfun(@times, yreshaped, win)), gain);
Gyy = squeeze(mean(conj(GY) .* GY, 2));

f = linspace(0, Fs, winsize);
```

```
figure
subplot(211); semilogy(f, sqrt(Gyy)); grid on; grid minor
xlim([0 Fs/2]); xlabel('frequency [Hz]')
subplot(212); plot(time*1e3, y, 'o-'); xlabel('time [s]'); xlim([0 30])

calib_factor = (9.81*sqrt(2))/2.355; % Calibration factor

%% Set #1

% Steady state 0.5*rpm speed
%     Input frequency range: [Fmin Fmax] = [149.2 149.2] Hz
%     Input voltage range: [Vmin Vmax] = ?
%
%     Fs = 13.1072e6/(256*31) > 300 Hz = 2*Fmax
%     winsize = 476, minimize leakage
%     noavg = 32

% acquisition parameters
Fs = 13.1072e6/(256*31);
winsize = 476;
noavg = 32;
range = 5;
bits = 24;
```

C Auxiliary DSP Functions

```
classdef myDSP
methods(Static)

function [SIGNAL, nowin] = reshape(signal, win_size, overlap_p)
%RESHAPE Reshape vector in window blocks, with overlaps
%     Example:
%     % Generate data
%     % Define dynamic system
%     sys = rss(3, 2, 1);
%     Fs = -100*min(real(eig(sys.A)));
%     N = 10000;
%
%     time = linspace(0, N/Fs, N).';
%     u = reshape(repmat(randn(round(N/20),1), [1 20]).', [N 1]);
%     y = lsim(sys, u, time);
%
%     winsize = 1000;
%     overlap = 0.3;
%     yreshaped = myDSP.reshape(y, winsize, overlap);
%     ureshaped = myDSP.reshape(u, winsize, overlap);
%
%     win = window(@hann, winsize);
%     gain = [1; 2*ones(winsize-1, 1)] / (mean(win)*winsize);
%
%     GY = bsxfun(@times, fft(bsxfun(@times, yreshaped, win)), gain);
%     Gyy = squeeze(mean(conj(GY) .* GY, 2));
%
%     GU = bsxfun(@times, fft(bsxfun(@times, ureshaped, win)), gain);
%     Guy = squeeze(mean(bsxfun(@times, conj(GU), GY), 2));
%
%     f = linspace(0, Fs, winsize);
%
%     figure
%     subplot(311); semilogy(f, sqrt(Gyy)); grid on; grid minor
%     xlim([0 Fs/2]); xlabel('frequency [Hz]')
%     subplot(312); semilogy(f, sqrt(abs(Guy))); grid on; grid minor
%     xlim([0 Fs/2]); xlabel('frequency [Hz]')
%     subplot(313); plot(time, y); xlabel('time [s]')

    len = size(signal, 1); % signal length
    nocha = size(signal, 2); % number of channels

    if win_size > len
```

```
        error('Window size is greater than signal size')
    end
    if overlap_p > 1 || overlap_p < 0
        error('Overlap should range between 0 and 1')
    end

    overlap = round(overlap_p*win_size);
    nowin = floor((len-win_size)/(win_size-overlap) + 1); % # of windows

    SIGNAL = zeros(win_size, nowin, nocha);
    for k = 1 : nocha
        % make overlap multiple of win_size, percentual to absolute
        idx = repmat((1:win_size).', [1 nowin]) + ...
            repmat((0:nowin-1)*(win_size-overlap), [win_size 1]);
        SIGNAL(:,:,k) = reshape(signal(idx,k), [win_size, nowin]);
    end
end


function Y = discretize( U, N, V )
    % Discretize vector U on the range +-V into N bits

    U = uencode( U, N, V, 'signed');

    U = cast( U, 'double' );
    N = cast( N, 'double' );
    V = cast( V, 'double' );

    Y = interp1( [ -2^(N-1) 2^(N-1)-1 ], [-V V], U );
end

function [blocksize, noc] = optimize_blocksize(F0, Fs, noc_range, Fmin)

    min_time = 2/Fmin;
    min_blocksize = min_time/Fs;

    nocs = noc_range(1) : noc_range(2);
    approx_blocksizes = nocs * Fs/F0;
    % disregard too small block sizes
    approx_blocksizes(approx_blocksizes < min_blocksize) = [];
    if isempty(approx_blocksizes)
        error('Slower component do not fit window choices')
    end
    blocksizes = round(approx_blocksizes);

    [err, idx] = min(abs(approx_blocksizes - blocksizes));
```

```
    blocksize = blocksizes(idx(1));
    noc = blocksize * F0/Fs;

    if err > 1e-3
        warning(['Bad block size. Round error: ' num2str(err)])
    end
end

end
end
```