Neuroprothetik – Exercise 1: Introduction

1 Generate a Signal

The following Matlab code generate a signal following eq. 1 and takes into account following input arguments: array of frequencies in Hz, array of amplitudes, signal duration in s, sampling rate in Hz.

$$f(t) = A_0 + \sum_{i=1}^{n} A_i \cdot sin(2\pi F_i \cdot t) \tag{1}$$

1.1 Plot the signal

a)

```
function [s,t] = generate_signal(f_v,A_v,duration,fs)

%% Signalgeneration
% f : Frequenz-Vektor in Hz
% A : Amplituden-Vektor
% duration : Dauer des Signals in Sekunden
% fs : Abtastfrequenz in Hz
%
sz_f = size(f_v); % Groessee des Frequenz-Vektors bestimmen
t = linspace(0,duration,duration*fs); % Zeitvektor erstellen
offset = A_v(1);

%% Signal berechnen
s = zeros(1,length(t));

%%
for i = 1:sz_f(2)
s = s + A_v(i+1)*sin(2*pi*f_v(i)*t);
end

s = s + offset;  % Gleichanteil drauf addieren

%% Signal plotten
figure;
plot(t*1e3,s, 'r');
title('Signal s(t)');
xlabel('Time t in ms'), ylabel('Amplitude A');
xlim([0 100]);

end
```

b)

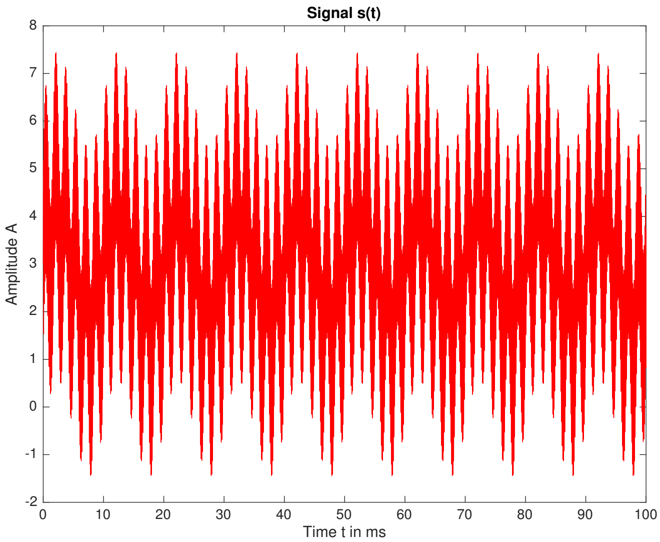

Figure 1: Sinusoidal signal s(t) consisting of superposed frequencies 100 Hz, 600 Hz and 9 kHz with amplitudes 1, 1.5 and 2 as well as an offset of 3 at a sampling rate of 100 kHz.

2 Calculate the Spectrum

The following code calculates the single-sided amplitude spectrum using the FFT.

a)

```
function [X f] = np_spectrum(s,fs)
% INPUT
% s : Signal im Zeitbereich
% fs : Sampling-Frequenz des Signals im Zeitbereich
%
% OUTPUT
% X : Spektrum von Zeitsignal
% f : FFT-Frequenzen

%% FFT berechnen
FFT = fft(s)/length(s);
n = length(s)/2; % Haelfte der Koeffizienten verwerfen
X = FFT(:, 1:floor(n)+1);

%% Amplitude der Koeffizienten anpassen (ausser Gleichanteil)
```

```
X(2:end) = X(2:end)*2;

%% Frequenz-Vektor f berechnen
f=(0:floor(n))/n * fs/2;

end
```

2.1 Plot the Spectrum

a)

The following Matlab-Code plots the amplitude spectrum for the different sampling rates (100 kHz, 20 kHz, 10 kHz):

```
%% Amplitudenspektrum plotten
figure;
plot(f,abs(X));
xlabel('Frequenz f'), ylabel('Amplitude X');
xlim([0 10e3]);
```

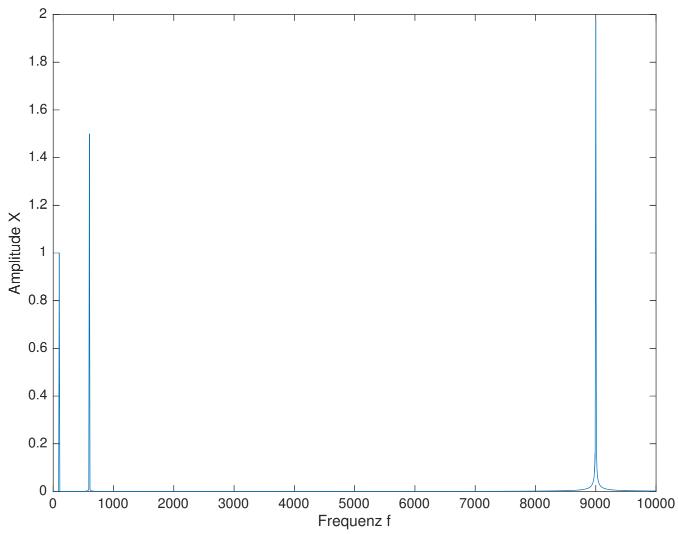

Figure 2: Amplitude spectrum for signal from section 1, with sampling rate of 100 kHz.

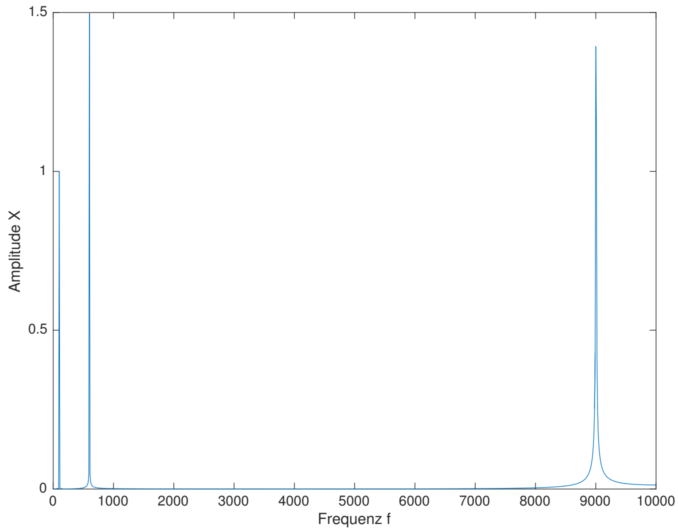

Figure 3: Amplitude spectrum for signal from section 1, with sampling rate of 20 kHz.

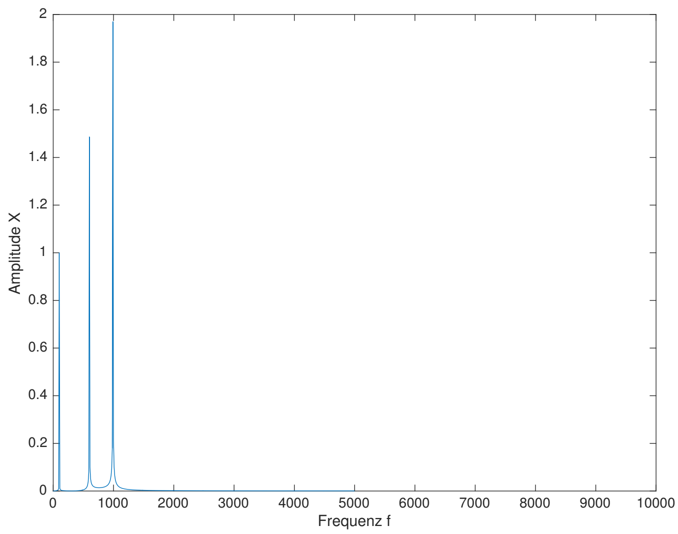

Figure 4: Amplitude spectrum for signal from section 1, with sampling rate of 10 kHz.

b)

The last spectrum (the one that was calculated based on the signal sampled at a rate of 10 kHz) violates the Nyquist-Shannon sampling theorem. The Nyquist-Shannon sampling theorem states that for a given bandlimit B of a signal, the sampling frequency f_s needs to be at least twice the bandlimit:

$$2 \cdot B < f_s \tag{2}$$

This condition is **not fulfilled**, as the maximum frequency in signal $s(t)$ is 9 kHz, i.e.:

$$2 \cdot 9 \text{ kHz} = 18 \text{ kHz} < 10 \text{ kHz} \tag{3}$$

Therefore, aliasing artifacts occur in the amplitude spectrum of signal $s_3(t)$ between the two main spikes at 100 Hz and 600 Hz. The maximum admissible frequency in any signal $s(t)$ must not exceed $f_s/2$ for proper sampling and perfect reconstruction of the signal.