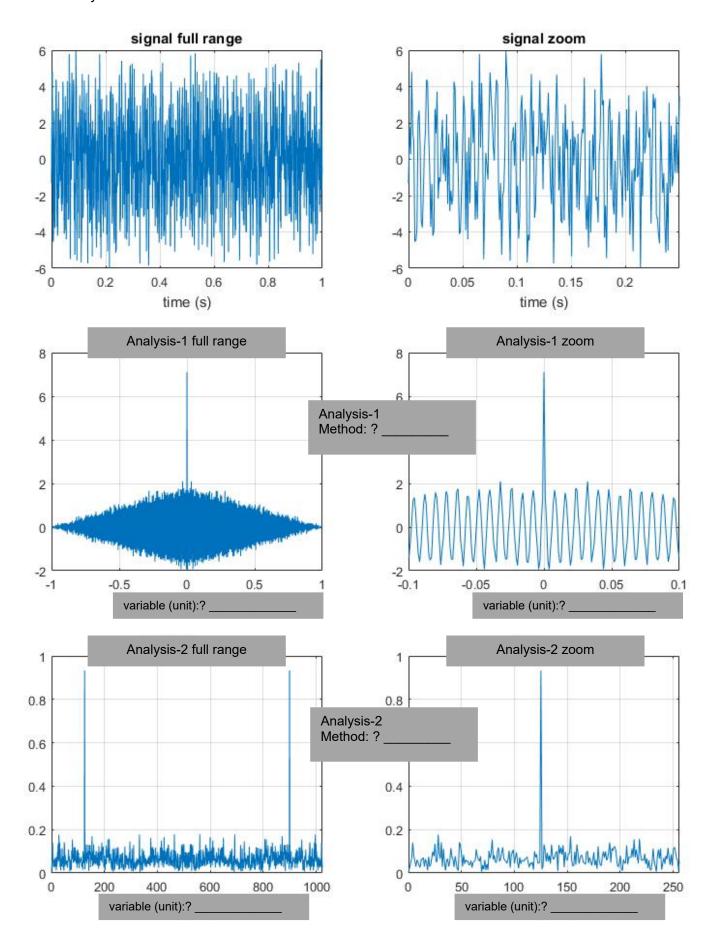


SiSy Semester Exam

Duration: 90 Minutes Open book exam, without calculator. Your calculations and solution approach need to be readable and comprehensible in order to get the full points. Please write your final results in the reserved gray fields and use the provided spaces for the sketches. Do not forget to label your axes.

Name:							Class:			
1:	2:	3:	4:	5:					Points:	Grade:

A noisy input signal s(t) is analysed using 2 methods: FFT and auto-correlation. The plots generated in these analyses are shown below.



- (A) Identify in the previous page, which plots belong to which analysis method. Then, make a supposition about the variables and units used for the horizontal axes in the plots, and write your assumptions in the reserved grey fields along the axes.
- (B) Which characteristics of the signal s(t) can you identify and confirm with the outputs of these two analyses?

(C) Complete the extract of code below, used to generate the plots on the previous page

Exercise 2 Fourier Series with Complex Coefficients

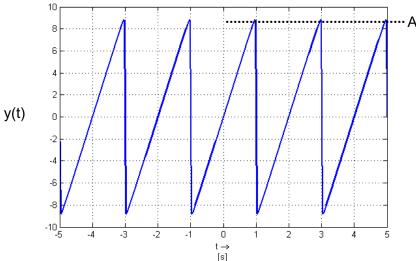
$$[2+4+4+4+5 = 19 Points]$$

The complex Fourier coefficients of y(t) and a plot of y(t) in the time domain are given below: Obs.: in order to identify these coefficients as related to y(t), we call them cky.

 $c_{ky}=j.rac{A}{k\pi}.cos(k\pi)$ für $k\in N^+$ with

$$c_{-ky} = \left(c_{ky}\right)^*$$

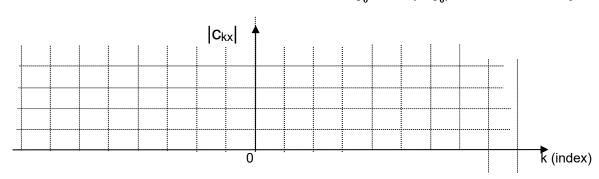
and $A = 3\pi$



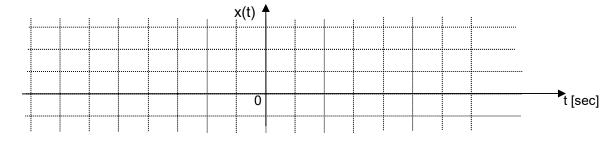
(A) Explain how the symmetry observed in y(t) can be confirmed by the expression of the corresponding c_{kv} coefficients.

(B) The complex Fourier coefficients of another time signal, c_{kx} for x(t), are given below. Draw a sketch of the amplitude spectrum of x(t) in the axis below.

$$c_{kx} = A \cdot \frac{\tau}{T_0} \cdot sinc\left(k \cdot \frac{\tau}{T_0}\right)$$
 with $\tau = \frac{T_0}{4}$



(C) Draw a sketch of x(t) in the time domain. Explain, how did you find out the characteristics of x(t). Hint: compare to a reference signal discussed in lectures and labs.



 $t\,[s]\to$

(D) The signals y(t) and x(t) are summed up to generate a new signal s(t). Determine the c_{ks} coefficients of s(t) and complete the Matlab code below, which implement the synthesis of s(t).

```
% Fourier Synthesis to check coeffs
% PARAMETERS & TIME-VECTOR
T0 = 2; w0 = 2*pi/T0;
t = -2*T0:T0/200:2*T0;
A= 3*pi;
tau = T0/4;
% Number of harmonics and DC-Offset
Kmax = 30;
c0x =
x t = c0x * ones(1, length(t));
                                     % initialise even part with DC-value
                                     % initialise odd part with zero
y t =
for k = -Kmax:1:Kmax
    if k \sim = 0
                                     % jump zero, cause cky expression not valid
      cky = j*(A/(k*pi))*cos(k*pi);
                                             % odd part
      y t =
      ckx = (A*tau/T0)*sinc(k*tau/T0);
                                                    % even part
                                                               Periodic Synthesised Time Function: odd part
      x t =
                                                        £
                                                          0
    end
                                                          -20
end
                                                                             t[s]
% combine even+odd parts for complete signal
                                                               Periodic Synthesised Time Function: even part
                                                          20
s_t = x_t + y_t;
                                                        £
                                                          0
figure() ...
                                                          -20
                                                                     -2
                                                                               0
subplot(311),plot(t,y t),...
                                                                             t[s] \rightarrow
subplot(312), plot(t, x t),...
                                                            Periodic Synthesised Time Function: complete signal
subplot(313), plot(t,s t),...
                                                          20
                                                          0
                                                          -20
                                                                 -3
                                                                     -2
                                                                               0
                                                                                             3
```

(E) Suppose the signal s(t) passes now through an ideal

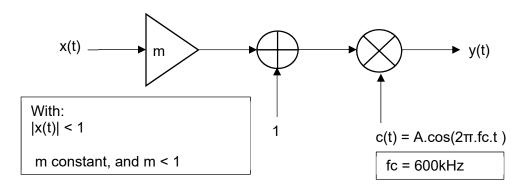
low pass filter, and gets band limited. The cut frequency

of the ideal LPF equals to the 1^{st} zero crossing of the spectrum of x(t) (first lobe of the sinc-shape).

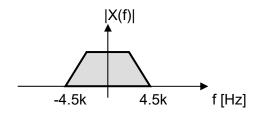
Explain how would you calculate the percentage of the power of the original s(t) signal, which gets through the LPF.

(A) The block diagram below shows a signal processing chain. Assume an input signal x(t), with the corresponding amplitude spectrum abs(X(f)) shown below.

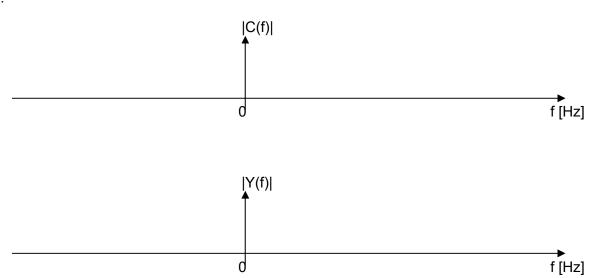
Then, complete the sketches and comments required (marked in grey).



Given:



Asked:



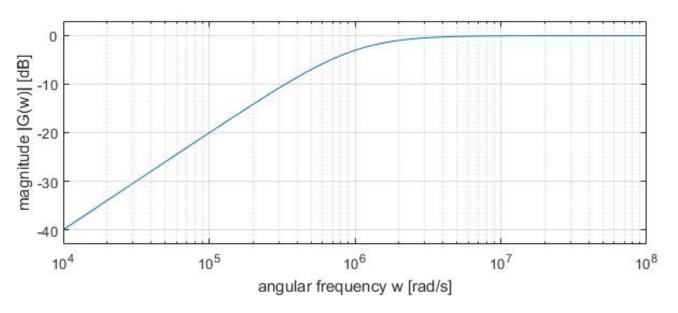
Related Fourier Transformation Property: (name + mathematical expression)

(B) What is the common denomination for the signal processing chain shown in item (a)? Which commercial application used it for a long time (from about 1920 until 2015 in many countries around the world)?

(C) In another signal processing application, you want to add a reverberation effect to an audio data, which reproduces the acustic characteristics of a cathedral. Explain how you can do that. Which measurements and which processing steps are required? Add equations and/or a block diagram to illustrate your answer.

Exercise 4 Frequency response of an electrical LTI System [4+3+6+6+3=22 points]

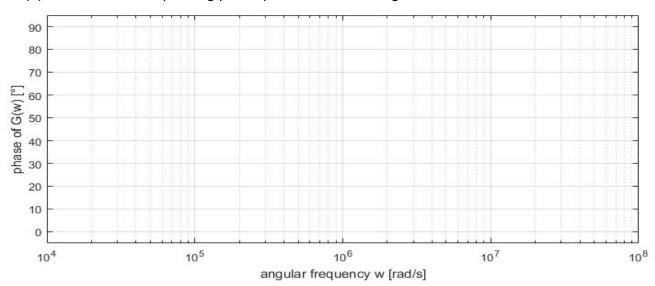
The amplitude part of the Bode Diagram of a frequency response $G(\omega)$ is given below:



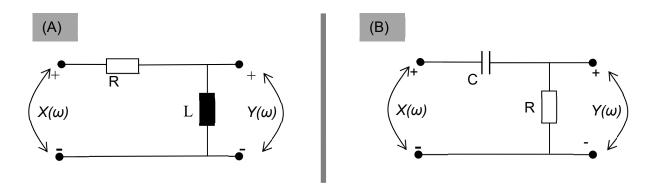
(a) Determine the equation describing $G(\boldsymbol{\omega})$.

Hint: The term in the numerator describes a line with constant slope of +20dB/decade, and the term in the denominator describes a curve with a corner and slopes of 0dB/decade before the corner, and -20dB/decade after the corner point.

(b) Sketch the corresponding phase part of the Bode diagram.



(c) Which of the following circuits can be used to implement the frequency response of item (a)? Justify your response by calculating the frequency response of both circuits using the method of the complex impedances.



(d) Using R=1k Ω , determine the corresponding value for L and/or C .

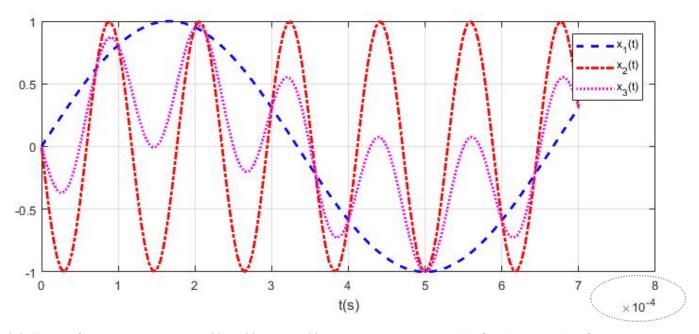
(e) Determine the response^(*) of a system with the frequency response $G(\omega)$ above, for an input signal $x(t) = \cos(10^6 \cdot t)$. Justify your answer with a short sentence.

^(*)Only the stationary part of the response is expected

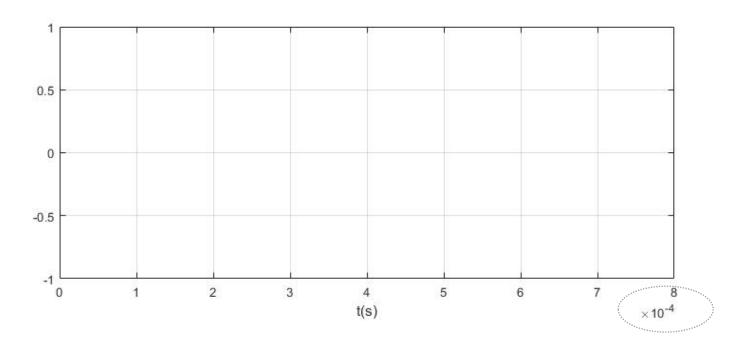
Exercise 5 ADC & Aliasing & FFT [3+4+8 = 15 Points]

The equation of three continuous time signals $x_1(t)$, $x_2(t)$, and $x_3(t)$ and the corresponding plots are given below:

$$x_1(t) = \sin(2\pi f_1.t)$$
 with $f_1 = 1.5 \text{ kHz}$
 $x_2(t) = -\sin(2\pi f_2.t)$ with $f_2 = 8.5 \text{ kHz}$
 $x_3(t) = (0.5).x_1(t) + (0.5).x_2(t)$



(a) Each of the three signals $x_1(t)$, $x_2(t)$, and $x_3(t)$ are converted by an ADC with sampling frequency Fs = 10kHz. Prepare a sketch of the corresponding discrete outputs $x_1[n]$, $x_2[n]$, and $x_3[n]$. Hint: Identify in the plot above the sampling points, starting at t=0s.



(b)	What would change in the discrete outputs $x_1[n]$, $x_2[n]$, and $x_3[n]$, if one added an ideal anti-
	aliasing filter before the ADC?
	Justify your answer, explaining the characteristics (filter type and cut-off frequency) of the anti-
	aliasing filter, and its effect on the input signals.

- (c) Suppose the three signals $x_1(t)$, $x_2(t)$, and $x_3(t)$ are now converted by an ADC with a higher sampling frequency of Fs= 32kHz. And the discrete outputs $x_1[n]$, $x_2[n]$, and $x_3[n]$ are recorded in a DSP, which calculates the corresponding spectra $X_1[k]$, $X_2[k]$, and $X_3[k]$ using the FFT algorithm. The DSP records the sampled signals in data blocks of 64 values. Answer the questions below, indicating the expected characteristics of the spectra $X_1[k]$, $X_2[k]$, and $X_3[k]$
 - \circ What is the frequency range represented in the spectra $X_1[k]$, $X_2[k]$, and $X_3[k]$?
 - \circ What is the frequency step (or frequency resolution) in the spectra $X_1[k]$, $X_2[k]$, and $X_3[k]$?
 - o For which frequencies (and corresponding k index) do you expect to find non-zero X[k] coefficients? Fill your answers in the table.

Spectrum	Non-Zero Coeffs for f = ?	Non-Zero Coeffs for k = ?
X ₁ [k]		
X ₂ [k]		
X ₃ [k]		
\\\ 3[K]		