Initial Remarks

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Course of the Exercise

- you'll get hand-outs with problems and we recommend you to solve them at home as preparation for the exam
- the solutions to the problems will be presented in the exercises
- at any time you can ask your questions on the material

Exams

- see information in the lecture

Literature

- see information in the lecture

For updates and downloads please visit the teaching webpage http://dss.kirat-online.de/index.php/teaching/lectures/ and click on the ADSP section.

Notation

Symbol	Meaning/Usage
t	Continuous time variable
n	Discrete time variable
$\delta_0(t)$	Continuous time impulse signal
$\gamma_0(n)$	Unit impulse signal (discrete)
$\gamma_{-1}(n)$	Unit step signal (discrete)
ω	Analog frequency in radians per second $\omega = 2\pi/T$
T	Sampling period in seconds
f	Analog frequency in Hz
f_s	Sampling frequency in Hz
Ω	Digital frequency in radians $\Omega = 2\pi f/f_\mathrm{s}$
$v(t) \circ\!\!-\!\!\bullet V(j\omega)$	Continuous Time Fourier transform
$v(n) \circ\!\!-\!\!\bullet V(e^{j\Omega})$	Discrete Time Fourier Transform
$v(n) \circ\!\!-\!\!\bullet V(\mu)$	Discrete Fourier Transform

<u>Problem 1</u> (relationship between continuous and discrete signals)

A complex-valued continuous-time signal $v_a(t)$ has the Fourier transform shown in figure 1. This signal is sampled to produce the sequence $v(n) = v_a(nT)$.

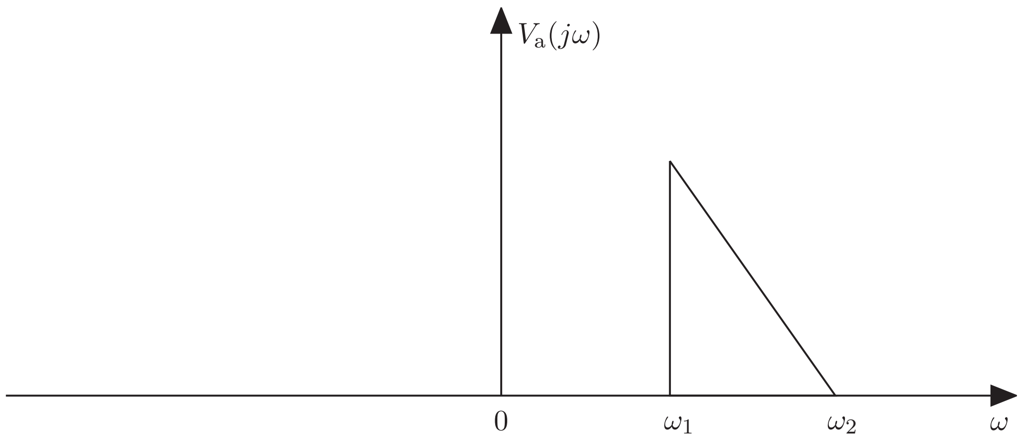

Figure 1: Fourier transform of $v_a(t)$

(a) Sketch the Fourier transform $V(e^{j\Omega})$ of the sequence $v(n)$ for $T = \frac{\pi}{\omega_2}$.

(b) What is the lowest sampling frequency that can be used without incurring any aliasing distortion, i.e. so that $v_a(t)$ can be recovered from $v(n)$?

Problem 2 **(overall system for filtering a continuous-time signal in digital domain)**

Figure 2 shows an overall system for filtering a continuous-time signal using a discrete-time filter. The frequency response of the ideal reconstruction filter $H_\mathrm{r}(j\omega)$ and the discrete-time filter are shown below.

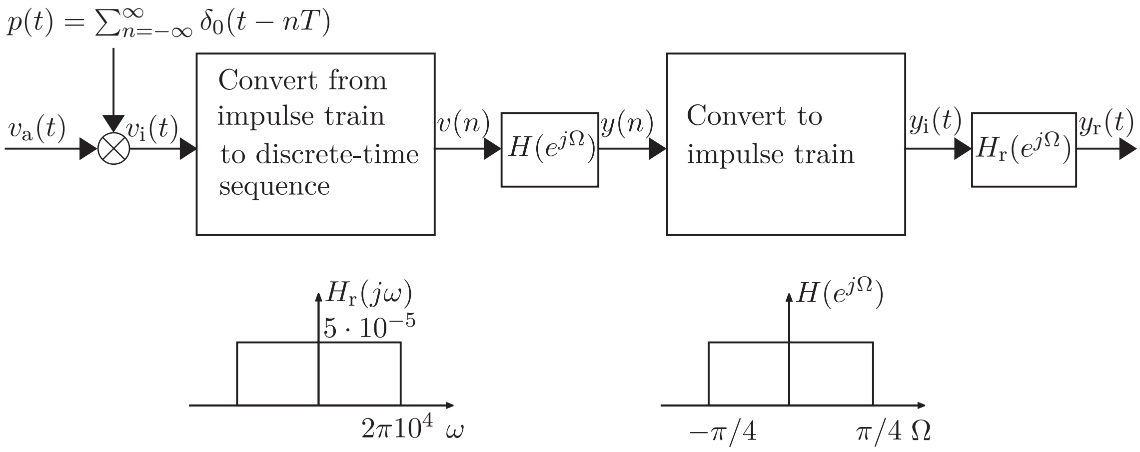

Figure 2: Overall system.

(a) For $V_\mathrm{a}(j\omega)$ as shown in figure 3 and $1/T = 20kHz$ sketch $V_\mathrm{i}(j\omega)$ and $V(e^{j\Omega})$.

Figure 3: Spectrum of $V_\mathrm{a}(j\omega)$ and $H_\mathrm{eff}(j\omega)$

For a certain range of values of T, the overall system, with input $v_\mathrm{a}(t)$ and output $y_\mathrm{r}(t)$, is equivalent to a continuous-time lowpass filter with frequency response $H_\mathrm{eff}(j\omega)$ sketched in figure 3.

(b) Determine the range of values of T for which the information presented above is true, when $V_\mathrm{a}(j\omega)$ is bandlimited to $|\omega| \leq 2\pi \cdot 10^4$ as shown in figure 3.

(c) For the range of values determined in (b), sketch ω_c as a function of $1/T$.

Note: This is one way of implementing a variable-cutoff continuous-time filter using fixed continuous-time and discrete-time filters and a variable sampling rate.