Exam

Digital Speech Transmission

Winter Term 2021/2022

Date: Thursday, March 17, 2022

Process Time: 90 minutes

Name: ______________________

Matriculation Number: ______________________

Signature: ______________________

Prob.	Points	Initials
1	/ 6	
2	/ 8	
3	/ 7	
4	/ 7	
Σ	/ 28	

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Advices for the Exam Digital Speech Transmission

- The camera and audio link are to be left on throughout the exam.
- In the beginnung, label each sheet used with your name and matriculation number.
- You are **not allowed** to use lecture notes, exercises notes, and any other book or written material.
- You are **not allowed** to use programmable electronic devices, such as laptops, notebooks, smart watches, tablets, etc. However, you are **allowed to use conventional pocket calculators**.
- Extent: 4 problems for 28 points on 5 pages excluding cover.
  Please check the exam for completeness!
- Duration: 90 minutes
- **Do not use red color, green color, or pencil**!
- Please consider that the distribution of points of the single problems can be different and that the subproblems might be solvable independently.
- The PDF, which can be downloaded in Dynexite, contains all problems. Below the download button, you will find the upload button where you can upload your solutions at the end of the exam.
- The time to scan and upload the exam sheets is 25 minutes. During this time, you remain in the Zoom meeting.
- You are allowed to use the same device as for video surveillance for scanning/photographing the exam sheets.
- If you have a problem, raise your virtual hand or wave at the camera, and one of the attendants will go into a separate breakout session with you. Do not leave your seat without being asked.
- If you need to cancel the exam for health reasons, immediately go to a doctor and get a medical certificate. The medical certificate needs to document date and time as well as a confirmation that the health reasons could not be detected before the exam.

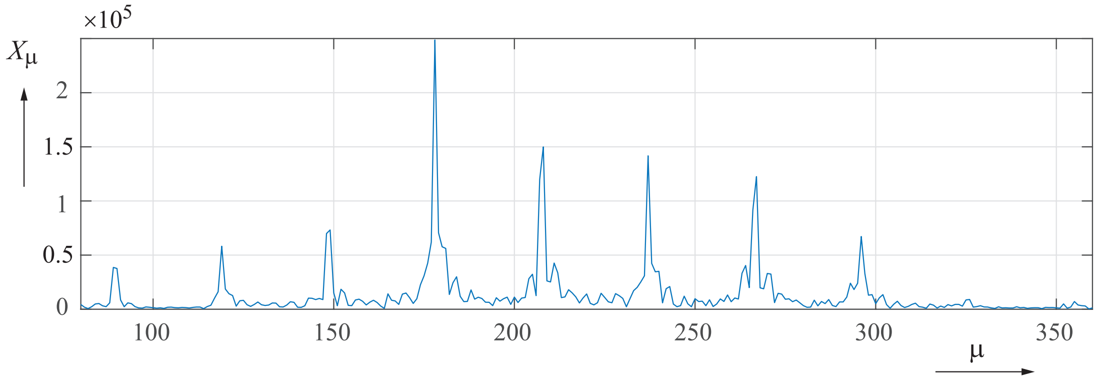
Other Dates

- Results will be available via RWTHonline on **Thursday, March 31, 2022**.
- Inspection of the exam will take place online on **Friday, April 8, 2022** from 11:00 to 12:00.
- You have to register for the online inspection. More details will be announced.

Problem 1 6 points

a) In a simplified model of speech production only white noise excitation shall be used. Draw qualitatively the short-time magnitude spectrum of a vowel generated with such a simplified model. (2)

b) The figure below shows an extract of the DFT X_μ of a speech signal $x(k)$. The speech signal is sampled at $f_s = 8\,\text{kHz}$ and the DFT length is $M = 2048$. Determine the fundamental frequency f_0. (2)



c) Explain the principle and the different steps of quantization with companding. Why is it useful for the quantization of speech signals? What are the most common companding standards in speech processing? (2)

Problem 2 **8 points**

Consider a digital ARMA filter with the transfer function

$$H(z) = \frac{1 - z^{-1} + z^{-2}}{1 - \frac{3}{2}z^{-1} + \frac{9}{4}z^{-2}}.$$

The filter $H(z)$ shall be implemented using a lattice structure. The block diagram of one lattice section is shown in Figure 2.1.

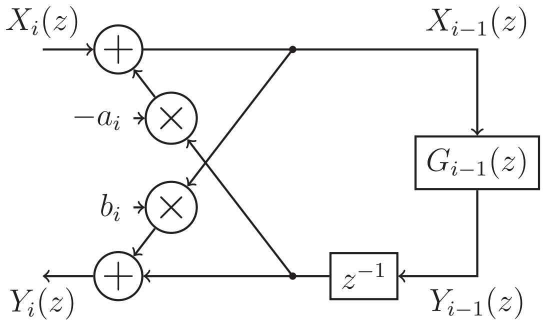


Figure 2.1: Block diagram of one section of the desired lattice structure.

a) Show that the transfer function of one lattice section is given by (2)

$$G_i(z) = \frac{Y_i(z)}{X_i(z)} = \frac{b_i + z^{-1}G_{i-1}(z)}{1 + a_i z^{-1}G_{i-1}(z)}.$$

b) Two lattice sections, as shown in Figure 2.2, shall now be used to implement the filter. Determine $G_2(z) = Y_2(z)/X_2(z)$ as a function of a_1, a_2, b_1, and b_2. Then, determine the values of a_1, a_2, b_1, and b_2 such that $G_2(z) = H(z)$. (4)

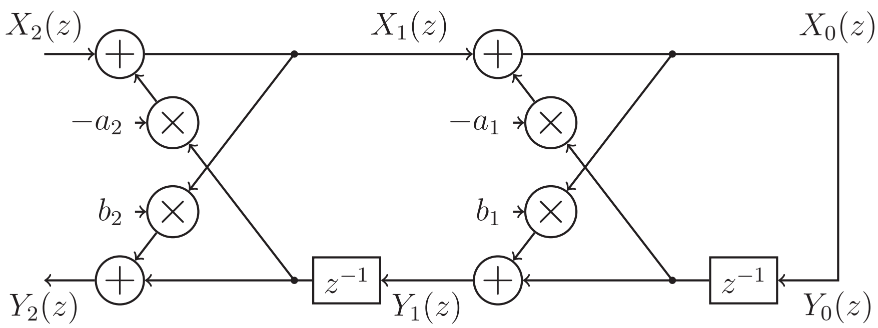
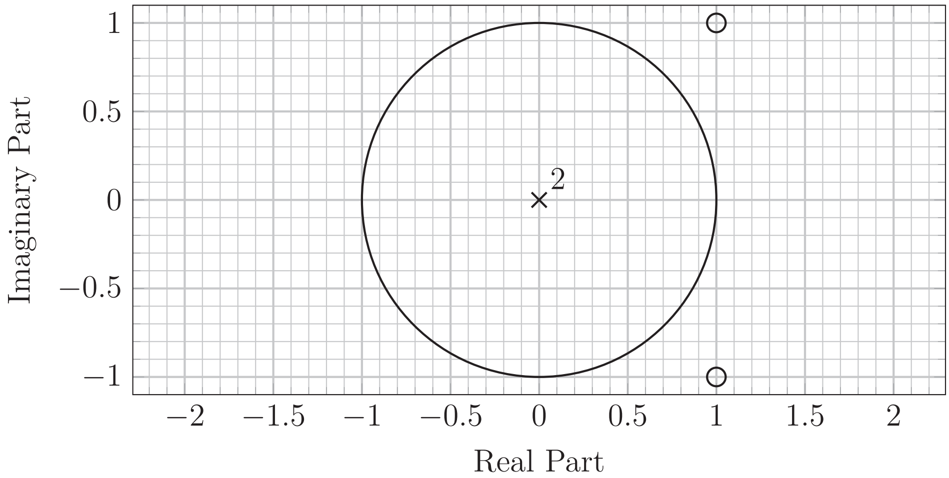


Figure 2.2: Block diagram of desired lattice structure.

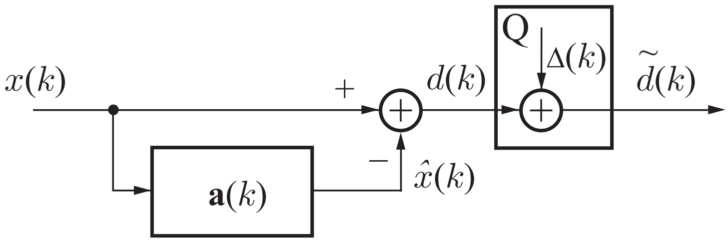
Now, a **different** filter is considered, of which the pole-zero plot in the z-domain is given below:



c) Give a reason why the filter is not minimum phase. Then, calculate the poles and zeros $z_{\mathrm{p},1}$, $z_{\mathrm{p},2}$, $z_{0,1}$, and $z_{0,2}$ of a minimum-phase filter with the same magnitude response. (2)

Problem 3 **7 points**

Consider a DPCM system with forward prediction and quantization. The predictor is optimally adjusted such that the difference signal $d(k)$ becomes a white noise signal. For simplicity, we assume in addition that difference signal $d(k)$ is uniformly distributed in the range $[-d_{\max} \;\; d_{\max}]$.



For the DPCM system the autocorrelation function of the input signal $\varphi_{xx}(0)$, the prediction gain G_p, and the word length w of the quantizer are given.

a) Give an expression for the calculation of $\varphi_{dd}(0)$ as a function of the given parameters $\varphi_{xx}(0)$ and G_p. (1)

b) Derive an expression for $d_{\max}$ as a function of $\varphi_{xx}(0)$ and G_p. (2)

In the next step, we want to prove the 6dB-per-bit-rule.

c) Provide an expression for the power of the quantization noise $\varphi_{\Delta\Delta}(0)$ as a function of $\varphi_{dd}(0)$ and word length w. (2)

d) Determine an expression for the signal to noise ratio at the quantizer $\mathrm{SNR_Q}$ and prove the 6-dB-per-bit-rule. (2)

Problem 4 7 points

In "Code Excited Linear Prediction" (CELP) speech codecs, the excitation of the LPC synthesis filter is realized using a **fixed** and an **adaptive** vector code book. The vectors of the fixed codebook are denoted as $\mathbf{c}_i$ $(i = 1, \ldots, N)$. The vectors of the adaptive codebook are denoted as $\mathbf{u}_j$ $(j = 1, \ldots, M)$. The corresponding gain factors are $g_{\mathrm{f},i}$ and $g_{\mathrm{a},j}$. In the sequel, we assume that in the search procedure the complexity of calculating the error for each codebook entry is identical in the fixed and adaptive codebook.

a) For reduced complexity, the codebook search is performed independently, i. e. **sequentially**, in the adaptive and the fixed codebook. By which factor does the sequential search reduce the complexity compared to the **combined** search over all combinations of $\mathbf{c}_i$ and $\mathbf{u}_j$? (2)

For the **combined** codebook search, the error vector is given by

$$\mathbf{e} = \mathbf{v} - g_{\mathrm{f},i} \cdot \mathbf{H}\mathbf{c}_i - g_{\mathrm{a},j} \cdot \mathbf{H}\mathbf{u}_j \, ,$$

where $\mathbf{v}$ is the target vector and $\mathbf{H}$ the synthesis filter matrix. The energy of this error, i. e. $E = \mathbf{e}^{\mathrm{T}}\mathbf{e}$, shall be minimized. In the following, assume that $\mathbf{H}$ is an identity matrix and use the simplified notation (with $\alpha_i = g_{f,i}$ and $\beta_j = g_{a,j}$)

$$\mathbf{e} = \mathbf{v} - \alpha_i \cdot \mathbf{c}_i - \beta_j \cdot \mathbf{u}_j .$$

b) Determine the linear equation system that yields the optimal gain factors α_{opt} and β_{opt}. Use a matrix/vector notation, i. e. the form (3)

$$\mathbf{A} \cdot \begin{pmatrix} \alpha_{\mathrm{opt}} \\ \beta_{\mathrm{opt}} \end{pmatrix} = \mathbf{b} \, .$$

Note: Please distinguish between scalars and vectors in your handwriting by underlining vectors.

c) Which conditions should the optimal code vectors $\mathbf{c}_{\mathrm{opt}}$ and $\mathbf{u}_{\mathrm{opt}}$ fulfill, such that the **sequential** code book search yields the same result as the **combined** search from b)? Justify your answer using the equation system determined in b)! (2)