WHAT TO READ FOR THE EXAM OF ADVANCED FILTER DESIGN?

This is more or less a combination of the descriptions of what to read for the exam included in the earlier courses entitled "Digital Linear Filtering I" and "Digital Linear Filtering II"–They are still visible in this messy homepage. The present description is divided into Part A: Digital Linear Filtering I, Part B: Digital Linear Filtering II, and Part C: Other Topics.

Part A: Digital Linear Filtering I

I. PART I

Learning Outcomes for Digital Linear Filtering I:

(a) What is a digital filter and how to analyze its performance using a difference equation, its transfer function and frequency response as well as its magnitude, phase, group delay, and phase delay responses?

(b) There are numerous structures implementing the very same transfer function–**Among these structures concentrate only on direct-form structures, transposed-form structures, and cascade-form structures consisting of second-order and first-order blocks.**

(c) Advantages and drawbacks when comparing infinite-impulse response (IIR) and finite-impulse response (FIR) filters with each other. Overall filter synthesis procedure in nutshell and various approximation criteria for the filter responses.

(d) **What to read in Part I? Pages definitely not needed: 1, 19-27, 36-37, 40-44, 46, 69-70, 80**

II. PART II:

(a) **No questions directly from this part.**

III. PART III:

Learning Outcomes for Digital Linear Filtering I–See also Part B: Digital Linear Filtering II for more reading

The characteristics of the four types of linear-phase FIR filters, their use in practical applications, and their synthesis using the windowing technique and the Remez algorithm

(a) **The characteristics of the four types of linear-phase FIR filters:** Definitions of these four filter types by means of the impulse response; Time- and frequency-domain characteristics as well as how to exploit the coefficient symmetry in the implementation. Furthermore, given a transfer function or an impulse response, there is need to determine the filter type as well as its frequency-domain characteristics by means of the zero-phase frequency response and the phase term, that is, there is a need to express the frequency response in the simplest possible form.

(b) **The use of the four types of linear-phase FIR filters in practical applications**: No additional comments needed; see the lecture notes

(c) **The windowing technique and the Remez algorithm**: Basic principles described clearly enough as well as the benefits and drawbacks in the design of linear-phase FIR filters. **Concerning the present course more topics will be given under Part B: Digital Linear Filtering II**

(d) **Pages to be skipped when reading for the exam: 21-22, 26-27, 31-32, 36-37, 45-47, 64-82, 96-99, 126-162, 179-191, 222-227, 236-239, 249-…These page numbers remain intact after going through Part B: Digital Linear Filtering II.**

IV. PART IV

Learning Outcomes for Digital Linear Filtering I

The characteristics of classical analog filters and their digital IIR equivalents, that is, classical recursive digital filters; The synthesis of low-pass IIR filters by transforming analog filters to their digital equivalents using the bilinear transformation, and the synthesis of high-pass, band-pass, and band-stop filters by applying appropriate z-plane transformations to low-pass IIR filters

(a) **The characteristics of classical analog filters and their digital IIR equivalents, that is, classical recursive filters**: **There is a need to remember only the simple formulas for synthesizing Butterworth filters!**

(b) **The bilinear transformation**: How to use the bilinear transformation for designing recursive digital filters with the aid of analog filters? Why is the bilinear transformation still a good technique for generating recursive digital filters? Design of a digital Butterworth filter for meeting the given criteria.

(c) **The synthesis of high-pass, band-pass, and band-stop IIR filters by applying appropriate z-plane transformations to low-pass IIR filters**: **NO QUESTIONS** because we just briefly went through this part of the lecture notes. However, please read also this part for the probable later use. It contains material that does not exist in any textbook

(d) **Pages to read: 0-55 with the emphasis on the above-mentioned goals.**

V. PART V

Learning Outcomes for Digital Linear Filtering I

(a) Attractive properties of the two's complement arithmetic; The commonly used model for estimating the output noise due to the multiplication round-off errors; Scaling of the cascaded-form IIR filters by using three commonly used scaling norms and the resulting trade-offs between the probabilities of overflows and the output noises; Finite word-length effects on the variations in the filter coefficients and their impact on various kinds of oscillations in IIR filters.

(b) **A Very Likely Question earlier but now because there is a competitor in Part B: Digital Linear Filtering II:** For two's complement arithmetic, scale a simple IIR filter according to one of the three commonly used scaling norms and estimate the variance of the output noise due to the multiplication round-off errors.

(c) **Comment: No question on** finite word-length effects on the variations in the filter coefficients and their impact on various kinds of oscillations in IIR filters. However, please read also this part for the probable later use.

(d) **Pages to read: 0-41, 45-68, 77-78, and Appendix B at the end of Part V with the emphasis on the above mentioned goals. These page numbers remain intact after going through Part B: Digital Linear Filtering II.**

Part B: Digital Linear Filtering II

VI. DIRECT-FORM FIR FILTERS

(a) In Part III under "Part A: Digital Linear Filtering I" and in "Design of FIR filters with constraints in the time and/or frequency domains", numerous techniques have been proposed for the design of direct-form linear-phase FIR filters. What are the key ideas behind these techniques? What are the benefits and drawbacks of these techniques? How to compare them with each other in terms of the simplicities of accomplishing these algorithms and in terms of the desired characteristics of the resulting linear-phase FIR filters.

(b) For linear-phase FIR filters designed in the minimax sense, the lecture notes contains altogether three simple ways of converting a narrow-band prototype filter transfer function into other ones. What are these simple ways? See pages 228–233 in Part III.

(c) How to design low-pass minimum-phase FIR filters in a simple manner?

(d) What are Lth-band linear-phase FIR filters? How to design a half-band linear-phase FIR filter using a "trick" proposed in the lecture notes?

VII. SYNTHESIS OF LINEAR-PHASE FIR FILTERS USING PERIODIC FILTERS AS BUILDING BLOCKS

(a) What are the main benefits (and minor drawbacks) of using a transfer function of the form $F(z^L)$ as a building block in the synthesis schemes proposed in the lecture notes?

(b) The key idea behind the frequency-response masking approach.

(c) Forget the overall Jing-Fam approach! What are the key ideas of using a periodic sub-filter, that is, the transfer function is of the form $F(z^L)$, when constructing a narrow-band or a wide-band linear-phase FIR filter?

VIII. IIR FILTERS

(a) Concentrate on the main lecture notes telling on how to construct a low-pass–high-pass or a band-pass–high-pass filter pair as a parallel connection of two all-pass filters.

(b) What are the block diagrams for these pairs?; What are the order differences in the above-mentioned two cases?; How the phase responses of the two all-pass filters should behave in order to arrive at the desired filter pairs?; What are the benefits of implementing the desired filter pair in the above-described manners compared to other implementation options?

IX. FINITE WORD-LENGTH EFFECTS

(a) What are the commonly used scaling norms for fixed-point arithmetic, especially for 2's complement arithmetic (basic definitions)?

(b) How do they differ from each other in terms of the probability of overflows and output noise variance due to the multiplication round-off errors.

(c) Very likely there is a question concerning the implementation of a sixth-order filter that is considered earlier in Part V in pages 45–68 as well as in pages 77–78. The lecturer feels that this study case is very illustrative.

Part C: Other Topics

X. FROM THE EALIER COURSE "SGN-2156 System Level DSP Algorithms": FINITE WORD-LENGTH EFFECTS IN PRACTICE

(a) **No questions from this part.**

XI. FROM THE EALIER COURSE "SGN-2106 Multirate Signal Processing": PART I: BASICS AND MOTIVATION

(a) **No questions from this part.**

In summary, the lecture feels that the course outcomes, according to the up-dated way of describing courses, are met provided that you are pretty aware of the above-mentioned requirements. O tempora o mores–The present course is a combination of four courses produced by the lecturer fifteen years ago. The learning and the needs of the students have been changed significantly during this time period. **Therefore, the lecture is eagerly waiting for feedback on how to update this course to be as satisfactory to you as the four courses, out of which the present course was constructed, were fifteen years ago. Please send an e-mail to ts@cs.tut.fi.** That's all folks! Good luck for the exam and please study intensively! Your lecturer following basically the name Tapio, but, thanks to my international friends, there are variations as Tap, Tap(10), TOP10. Nowadays, the best one seems to be Tap_Top10.