Scope of first partial exam: Problems 1-40
Scope of second partial exam: Problems 41-72

1. Sketch the general block diagram for a digital transmission system including the central functions of the transmitter and receiver and a simple channel model. Describe briefly the purpose of each block. 
   (Hint: Include coding, modulation, synch, etc.)

2. Explain the meaning of terms information and entropy.

3. Entropy of a binary random variable (example from lecture notes; entropy formula should be remembered).

4. What is meant by the prefix-property when talking about source coding? Which problem related to the decoding can be solved conveniently utilizing this property?

5. Describe the principle of Huffman-coding.

6. Develop a Huffman code starting from given symbol probabilities (like examples in the course material).

7. Describe the principle of run-length coding.

8. What information about the source characteristics is needed for a Huffman coder but not, e.g., for a Lempel-Ziv coder?

9. What are the typical compression ratio -values which can be achieved in lossless source coding (Huffman, Lempel-Ziv)?

10. Assuming that the source produces \( r \) symbols per second and the entropy of the symbol stream (or the stochastic process modeling it) is \( H(X) \), what is theoretically the lowest bit-rate that can be used in the transmission?

11. The capacity of a binary symmetric channel (example from lecture notes).

12. Considering discrete-time channels, sketch a chart showing the following plots:
   (a) Continuous-valued AWGN channel capacity (bits/symbol) as a function of SNR
   (b) Maximal conveyed information as a function of SNR for the following alphabets: 2-PSK, 4-PSK, 16-QAM, 64-QAM
   The characteristic shapes of the curves and asymptotic behaviors are essential.

13. The capacity per symbol for a continuous-valued discrete-time channel is \( C_s = \frac{1}{2} \log_2(1 + \frac{B}{\sigma^2}) \). Based on this, derive the capacity formula for a continuous-time AWGN-channel (Hartley-Shannon-law).

14. Compute the channel capacity for a telephone channel. Assume that the bandwidth is 3.3 kHz and S/N-ratio is 40 dB.
15. There are two different principles for shaping the spectrum of a PAM signal. Describe briefly these approaches. What are the central goals and properties (advantages and disadvantages) of these approaches? How does the transmitted signal spectrum depend on the used pulse shape?

16. What are the goals of line coding? What is the meaning of running digital sum (RDS) and baseline wander ISI? What requirements are typically set for the spectrum of the coded signal?

17. Show the bit sequence 0 1 0 0 0 1 1 1 0 0 in coded form using (a) (antipodal) NRZ-code (b) biphase- (Manchester) code (c) AMI-code. Sketch the signal power spectra using these codes. Compare the properties of these line codes.

18. Sketch the general block diagram for a baseband PAM transmission chain utilizing pulse-shaping (the central blocks of the transmitter and receiver and a simple channel model). Explain briefly the purpose of each block and the nature of the signals in the block diagram (discrete/continuos-time, discrete/continuos-valued, real/complex). What is meant by alphabet? How does the symbol rate depend on the bit-rate and the size of the alphabet?

19. (a) What is meant by intersymbol interference (ISI)? Why is it important to minimize it?
   (b) Describe the baseband pulse-shape, corresponding to a single transmitted symbol, which results in zero ISI.
   (c) Which frequency-domain criteria the pulse spectrum should satisfy to guarantee zero ISI?
   (d) Where in the transmission chain is it essential to satisfy this condition?
   (e) Which parts of the PAM/QAM/PSK transmission chain have an effect on the received pulse shape?
   (f) What are the essential aspects when designing the transmit and receive filters?
   (g) What is a raised-cosine pulse? (the formulas are not required)
   (h) Assume that raised-cosine pulse is used in the transmitter and the channel is ideal. Then what kind of receiver filtering is required to reach zero ISI?

20. In a PAM/QAM/PSK-system the symbol-rate is \( 1/T \).
   (a) What is the smallest bandwidth that can be used in baseband transmission and in passband transmission?
   (b) How does the size of the alphabet effect the signal spectrum and bandwidth when the symbol-rate is fixed?
   (c) Give a sensible range for typical bandwidths (use the term excess bandwidth). Sketch the baseband signal spectrum for 50 \% excess bandwidth.
   (d) In what sense can the waveform be improved by increasing the bandwidth (i.e., the excess bandwidth)?

21. In a PAM/QAM/PSK system, the transmit filter transfer function is \( G(f) \) and the channel transfer function is \( B(f) \). Which criterion should the receive filter transfer function satisfy to achieve zero ISI? Which problems this solution may have, i.e., why is this not necessarily optimal?

22. (a) What is an eye-diagram?
   (b) How does the eye-diagram show the sensitivity of the system to noise and timing errors?
   (c) Sketch a good eye-diagram in the case of 2-level PAM. Show the optimal sampling instant.

23. Describe the principle of quadrature amplitude modulation (I/Q-modulation) used in linear digital bandpass transmission systems. Show the transmitter and receiver block diagrams (in simple form) and
the signal spectra in different parts of the system. How do the parameters of a constellation point (angle and distance from the origin) show up in the waveform?

24. Show the equivalent baseband models for a linear digital passband transmission system:
   (a) Continuous-time model
   (b) Discrete-time model
   What is the output noise variance and noise power spectrum in each case? The channel noise is assumed to be white Gaussian noise.

25. (a) Show the constellations corresponding to the following alphabets: 2-PSK, 8-PSK, 16-QAM.
   (b) How does the effect of noise appear in the constellation plot (e.g. for 16-QAM)? What can be said about the noise distribution when the channel noise is white Gaussian?
   (c) Show the optimal decision regions for the constellations of part (a) (assume white channel noise and zero ISI).

26. Give a formula for symbol error rate for 2-PSK (symbol levels -a and a, noise variance \( \sigma^2 \)).

27. How does the symbol error probability of a QAM constellation depend on the minimum distance \( d \)?

28. Assuming that the alphabet and symbol error probability are fixed, how does the Gray-code help in minimizing the bit error probability?

29. Sketch a chart showing the symbol error rate as a function of S/N-ratio for 2-PSK and 16-QAM (typical shape, accuracy of a few dB is sufficient).

30(a) What is the maximum spectral efficiency for an alphabet of size \( M \) in baseband and passband PAM/QAM/PSK systems?
   (b) Give a typical value for a spectrally efficient system (e.g. voice-band modem).
   (c) What is meant by power efficiency? How can high power efficiency be achieved?

31. Starting from a given real baseband PAM system, the spectral efficiency should be made 1 bit/s/Hz higher than earlier. How much should the transmission power be increased to keep the error rate the same?

32. Explain the meaning of Shannon limit. (Please note that this is different from the Hartley-Shannon capacity theorem).

33. Describe the idea of differential modulation methods (such as DPSK)? What are the advantages and disadvantages of these methods.

34. (a) Describe the principle of FSK
   (b) What is meant by continuous phase? What are the advantages of continuous-phase systems?
   (c) Describe the principle of MSK.
   (d) Show the incoherent receiver structure for FSK. What advantages does it have?
   (e) Compare the required transmission power for binary FSK and binary PSK when the error probabilities are the same.
   (f) Describe the principle of GMSK. What advantages does GMSK have comparing with MSK?
35. Which advantages can be achieved using coherent modulation methods/receiver structures? Which are the drawbacks of coherent modulation methods? Which different possibilities there are for using noncoherent detection?

36. Explain the MAP and ML detection principles and illustrate the used probability distributions and the resulting threshold levels, using some basic noise distribution (e.g., Gaussian).

37. Describe the Maximum Likelihood detection principle in the case of symbol vector detection after a continuous-valued, discrete-time channel with additive white Gaussian noise.

38. Describe the Maximum Likelihood vector-detection principle in the case of binary symmetric channel.

39. Compute the Hamming distance of two given bit sequences.

40. Describe the principles for evaluating the error probability in sequence detection (e.g., through a simple example).

41. For which purposes the Viterbi algorithm can be used? Describe (briefly, e.g., through an example) the principle of Viterbi algorithm.

42. Considering Viterbi algorithm, what is meant by truncation depth?

43. Illustrate the ML sequence detection principle considering a simple example:
   * draw the trellis diagram and compute the branch metrics
   * formulate the principle of ML sequence detection with this diagram (i.e., for which question the detector gives an answer?) (Finding the shortest path, of course; tell also how the detected sequence can be determined from the path.)
   * show how the Viterbi algorithm works in this example case.

44. Explain what is meant by the term sufficient statistics when talking about detection in digital transmission systems.

45. In the case of QAM modulation, compare correlation receiver and the receiver structure based on matched filters. How do these two receiver architectures differ in performance?

46. The received baseband pulse shape is $h(t)$ (possibly complex).
   (a) What is then the impulse response of the matched filter?
   (b) How are the Fourier-transform of $h(t)$ and the frequency response of the matched filter related?
   (c) For a passband PAM/QAM/PSK system, show the receiver structure based on matched filtering which is optimal in the ML-sense for detecting a single symbol (channel noise is assumed to be white Gaussian).
   (d) What is the effect of the matched filter on frequencies in the signal band (on passband of the transmit filter) which are attenuated in the channel? (Does it compensate the attenuation or does it attenuate more? How much in relation to the channel attenuation?)

47. (a) A receiver for passband PAM/QAM/PSK-system includes demodulation (frequency shift), matched filtering, and sampling at symbol rate. Is this structure good in the ML-sense? Under which conditions no essential information is lost, for example, due to aliasing?
(b) The following figure shows a discrete time channel model including the sampled matched filter of part (a).

\[ A_k \quad S_h(e^{j\omega}) \quad P_k \]

\[ Z_k \]

How does the frequency response \( S_h(e^{j\omega}) \) depend on the received pulse shape? What is it in the case of zero ISI? What is the power spectral density of the equivalent noise source \( Z_k \) (assuming that the actual additive channel noise at the receiver input is white)?

48. Considering linear equalizers, describe the zero-forcing and MSE design principles. Assuming that the equivalent discrete-time channel transfer function and channel noise variance are known, what is the linear equalizer transfer function in each case? Which of these two principles results in lower mean-squared error at the equalizer output? Linear equalizers have a problem known as noise enhancement. What does it mean?

49. Show a receiver structure based on fractionally spaced equalizer (FSE). What advantages and disadvantages does it have comparing with structures where the sampling is done at symbol rate?

50. Describe the principle of decision-feedback equalizer (DFE). In which way do the different filter blocks help in reducing the intersymbol interference? Which advantages/disadvantages this technique has comparing with linear equalizers?

51. Show a receiver structure based on ML-sequence detection and describe briefly the principle. Which similarities and differences does this principle have comparing with decision-feedback equalizer, DFE?

52. Compare the MSE-performance of the following equalizer/receiver types: MF-MSE, DFE-MSE, LE-MSE, LE-ZF, DFE-ZF, ML-sequence detection. *(Notice that it is not always possible to say which is better from two structures.)*

53. Sketch the structure of an adaptive equalizer and describe briefly the principle. What kind of filters are used in this case? Describe also the operation modes: decision directed and training mode.

54. Describe the principles of MSE gradient algorithm and LMS algorithm for adaptive equalization. Sketch a block diagram implementing the LMS algorithm showing the calculations for each tap coefficient of the filter. Which factors have an effect on the choice of the step-size parameter \( \beta \)? How can an adaptive DFE be implemented using the LMS algorithm?

55. Show a shift-register structure which can be used for generating pseudorandom sequences. How long pseudorandom sequence can be generated using an \( n \)-stage shift-register? Describe the principle and goals of scrambling.

56. Show a structure which can be used for generating the pseudo random sequence with generator polynomial

\[ h(D) = 1 \oplus D^{14} \oplus D^{17}. \]
57. Describe the principles of hard- and soft-decision decoding for error-correcting codes. Compare them considering the performance and implementation aspects.

58. Let’s consider a digital transmission system for a given information transfer rate based on a given QAM- or PSK-type alphabet. Explain how the following factors depend on the code type (block code, convolution code, trellis-coded modulation) and how the code-rate (k/n) effects in each case:
   - the signal bandwidth in transmission
   - S/N-ratio at the receiver (assuming AWGN channel)

59. What is a systematic code? Give some simple example.

60. (a) Assume that all the code-words of a linear code are given in an array. How can you easily determine the minimum Hamming distance of the code?
   (b) How does the error correction capability, t, depend on the Hamming minimum distance in hard-decision decoding?

61. How large benefit (in dB’s) can typically be achieved by using simple block codes and convolution codes? How do soft-decision decoding and hard-decision decoding differ in this respect?

62. How many errors Hamming codes can correct? Why?

63. The simple block-codes from pages 342 and 343 of lecture notes: how to get the generator matrix and parity check matrix, how to find the minimum Hamming distance and minimum Euclidian distance, error-correction/detection capability in hard decoding, coding and (hard-decision) decoding of given bit-sequences.

64. Let’s consider (n, k) Reed-Solomon codes. Which values for n and k are possible? How many bits are included in each symbol? How many bit-errors/symbol-errors an (n, k) Reed-Solomon code can correct?

65. Let’s consider (255, 235) Reed-Solomon code. How many source bits/symbols are coded in one code word? What is the structure of the code word (length, bits/symbol). What is the error correction capability of this code? From this basic code, a shortened (220, 200) R-S code can be constructed. What is the error correction capability of this shortened code? How can this be realized if a coder and a decoder for the basic code are available?

66. Present a (non-trivial) example of convolution coder (a coder with one or two shift-register stages is recommended). What is the code rate? Show the generator matrix and parity check matrix for this coder. Show the trellis diagram. Code the bit-sequence 0 0 1 0 1 0 (assuming that the initial state is zero). Make a single bit-error to the coded sequence and do hard-decision ML-decoding for the erroneous bit sequence. Determine the Hamming minimum distance for the code. Determine also the Euclidean minimum distance assuming PSK modulation.

67. Problem 66 for the convolutional code of page 371 when the generator matrix or coder structure are given. In the case of generator matix, you may also be asked to draw the coder structure.

68. Consider convolution coding with code rate k/n when the information transfer rate is fixed. Assume that the pulse-shape and channel alphabet are fixed (e.g., PSK).
   - How does the bandwidth of the modulated signal depend on the code rate?
- How does the receiver S/N-ratio depend on the code rate (assuming white Gaussian channel noise).

69. Considering convolutional codes, what is meant by puncturing?

70. Describe the principle of trellis-coded modulation and compare it with convolutional coding. How large advantage can be typically achieved with trellis-coding? Illustrate a (simple) example of trellis coding.

71. For which different purposes Viterbi algorithm can be used in digital transmission systems? In which cases Euclidian distance and Hamming distance are used as path metrics?

72. Describe the effects of (a) carrier synchronization errors and (b) timing errors on the received signal/constellation.