1. What is data compression and what is it good for? Give advantages and drawbacks.
2. What is the definition of redundancy of a code for a given message X? In which units it is measured?
3. What is the basic principle (idea) of data compression? Which phenomena it usually employs?
4. What is the difference between lossy and lossless data compression? In which situations can we afford to use lossy compression and when not?
5. What is compression ratio? Is it constant for a given algorithm? If not, which factors may influence it?
6. Is Morse alphabet a compression method? Give reasoning. Is it unambiguously decodable? Is it necessary to use symbol delimiters? Give reasoning.
7. Which methods of data compression do you know? In which situations would you use each individual method and why?
8. What is the difference between a continuous and a discrete signal? Give examples, draw...
9. What is a periodic signal? Define mathematically.
10. Explain the following terms: period, frequency, amplitude, phase. Demonstrate them using an example, draw...
11. In a Fourier series, can a ratio of two frequencies be irational? Give reasoning.
12. Let's have a signal containing two spectral components with frequencies of 400 Hz a 500.sqrt(2) where sqrt() denotes square root function. Is this signal periodic?
13. Is spectrum of a periodic signal given unambiguously, uniquely? Which algorithms/mathematical tools for transformation of a signal (both periodic and nonperiodic) from temporal to spectral domain do you know?
14. Which frequencies in spectrum correspond to fast (abrupt) changes in temporal course of a signal? Low or high? Draw an example of such a signal.
15. Draw amplitude spectrum of an ideal Dirac pulse.
16. Draw amplitude spectrum of signal given as: x(t) = sin(200\*pi\*t+5) - 0.5\*sin(400\*pi\*t+5).
17. Is spectrum of an analogue nonperiodic signal continuous or discrete? Which mathematical tool can be used for computation of such spectrum?
18. Explain these terms: sampling, quantization. Demonstrate them on an example, draw...
19. Give and explain the sampling theorem.
20. Explain what aliasing in a sound signal means, explain the principles by which it emerges. How can we avoid it?
21. What is the manifestation of quantization in a 1D signal, what does it look like? What happens with its spectrum? Draw an example of the original and quantized signal and their approximate amplitude spectra.
22. What is an antialiasing filter, what is its purpose? What is its cutoff frequency?
23. What will happen to spectrum of a signal (e.g. a sound) when it is clipped in amplitude (for example, when the input range of A/D converter is exceeded). Draw a examples of the original and clipped signals along with their approximate spectra.
24. How many spectral components we get by application of a DFT or FFT on a signal of N samples? Will all the output samples be useful?
25. Let's have a coder using a DFT or FFT for signal processing, the block (window) is 512 samples long. For some reason we need to increase frequency resolution at its output. What should we do with the window length and why?
26. What is sound? What acoustic pressure (in Pa), approximately, has a very loud and a very quiet sound?
27. What is sound pressure level (SPL)? In which units is it measured? What are the reasons to use SPL instead of acoustic pressure?
28. What is spectrogram? Draw a simple example of a temporal wave and its spectrogram.
29. To which frequency band are people sensitive the most (approximately)? Which frequencies are perceived as less loud?
30. Let's have two harmonic sounds at frequencies of 50 Hz and 500 Hz at 60 dB SPL. Will both sound be perceived with the same loudness? If not, which one will be subjectively louder?
31. What is masking (in human's hearing)? Draw an approximate shape of masked threshold of a sinusoidal tone.
32. For which purposes and how is the auditory masking used in audio processing?
33. What is the device which transforms sound from analogue to digital form? Which basic operations does it perform?
34. What is the device which transforms sound from digital to analogue for? Which basic operations does it perform?
35. What is the device which transforms sound from mechanical vibrations to electric signal? What is the device which transforms sound in an opposite direction?
36. What is quantization noise? Draw a simple example.
37. Which negative phenomenon occurs when decreasing bit depth of a digital sound? Can it be compensated?
38. What is the theoretical dynamic range of an audio CD?
39. Which bit depths are usually used in digital audio today? In which situations are they used?
40. Which sampling frequencies are usually used with digital audio today? In which situations are they used?
41. Explain: fade-in, fade-out, cross-fade, audio normalization.
42. Explain what is dynamic compression. How does it affect temporal wave and spectrum of a signal? Draw an example...
43. DYnamic range of an orchestral composition is 120 dB. We want to record it on an audio CD. Which audio processing operation has to be performed so that we do not completely lose the loudest and/or quietest parts?
44. What is equalizer? What is its influence on temporal wave and signal spectrum?
45. What is resampling and requantization of an audio signal? In which situations is it used?
46. What are approximate compression ratios of MP3, OGG and FLAC coders (assume a CD quality signal)? Can the compression ratio be influenced by some setting?
47. Is it useful to use LZW compression for compressing digital audio? Explain.
48. Explain the basic priciple (idea) of perceptual lossy compression, such as MP3.
49. Which formats of digital audio files do you know? Describe their basic properties.
50. What is a psychoacoustic model? Where is it used and what is its purpose?
51. Describe what happens after a repeated compression of audio by a lossy coder such as MP3?
52. Do all the MP3 coders have the same sound quality at a given bitrate? If not, explain why.
53. Which types of signal are affected the most by the degradations given by lossy audio compression?
54. In data compression, we often use a mapping from a set of input symbols S to a set of codewords C. What property does this mapping have to have so that the compression be lossless?
55. In our data, symbol A occurs twice more often than symbol B. Does A have larger or smaller entropy than B? By how many bits?
56. Entropy corresponds to information content of a message, it is measured in bits. Let's have a message X with entropy H. We encode it using a code K and we get an output D bits long. Can D be larger or smaller or equal to H? What is the redundancy of the code in such cases?
57. Describe the principle of LZ77 algorithm.
58. Describe the principle of LZW algorithm.
59. Describe the principle of RLE compression method. Design an input message for which the compression ratio will be 10 %, and a message for which the compression ratio will be 150 %.
60. Let's have data containing only three different symbols, which have the same probability of occurrence. We construct a Huffman code to encode these data. Will this code have zero redundancy or not? Explain.
61. Explain these terms: Fourier series, discrete spectrum. Which type of signal can be transformed to Fourier series?
62. If we transform a signal from temporal to spectral domain, do we lose some information or do both representations contain the same information? Is it the same in theory and in practical situations?
63. Explain terms: Fourier transform, continuous spectrum. In which situations do we use Fourier transform and in which situations Fourier series? Is there any transform for discrete signals?
64. A signal containing frequencies 15000 Hz, 19000 Hz and 22000 Hz will be sampled at 40000 Hz sampling frequency. Which spectral components will the resulting digital signal contain?
65. What is (in general) the output of a discrete Fourier transform of a signal of N samples? What is the relationship of this output to the real spectrum of the input signal?
66. Let's have a digital signal sampled at 2000 Hz. We process this signal with DFT with window of 1000 samples. What will the resulting frequency resolution be?
67. Describe possible principles of reconstruction of a signal from discrete to analogue (continuous) form. Draw examples.
68. Let's increase te amplitude of some sound 10 times. What will be the corresponding change of sound pressure level (SPL) in dB? What will be the total change of SPL in dB if we amplify the sound once more 10 times?
69. Let's decrease te amplitude of some sound 100 times. What will be the corresponding change of sound pressure level (SPL) in dB? What will be the total change of SPL in dB if we attenuate the sound once more 10 times?
70. Spectrum of cembalo sound contains frequencies up to ca 80 kHz. Can a cembalo piece be recorded to an audio CD without loss on information? If not, what will be the highest recorded frequency? Explain.
71. Spectrum of cembalo sound contains frequencies up to ca 80 kHz. If we want to record a cembalo piece to an audio CD, what type of processing must be included between the microphone and the A/D converter? What would happen if we omitted this processing?
72. Let's have a sinusoidal signal recorded in a CD quality with amplitude of 10000 quantization steps. During processing, we at first amplify it (desctructively) by 20dB and then attenuate it again by 20dB. What will the signal look like (approximately) after such processing? Draw...
73. Let's have a sinusoidal signal recorded in a CD quality with amplitude of 50 quantization steps. During processing, we at first attenuate it (desctructively) by 20dB and then amplify it again by 20dB. What will the signal look like (approximately) after such processing? Draw...
74. Let's have a sinusoidal signal recorded in a CD quality with amplitude of 10000 quantization steps. During processing, we at first amplify it (desctructively) by 20dB and then attenuate it again by 20dB. What will the signal's spectrum look like (approximately) after such processing? Draw...
75. Describe the basic principle of resampling of audio signal. Describe fundamental properties - advantages, drawbacks, risks and how to avoid them.
76. Describe the basic principle of changing bit depth (quantization) of audio signal. Describe fundamental properties - advantages, drawbacks, risks and how to avoid them.
77. Describe and explain the principle of a lossy perceptual coder of digital audio, such as MP3.
78. Let's have a lossy perceptual coder of digital audio, such as MP3, which uses block processing. Describe problems related to block processing and possibilities of their reduction.
79. Describe typical distortions resulting from perceptual lossy compression of digital audio. What are the reasons of these distortions?
80. Compute compression ratio for an MP3 compression with bitrate of 256kbps relatively to an audio CD signal.
81. Compute how much space will one minute of a stereo digital audio in CD quality take, and how much will the same audio take after compression by an MP3 coder at 160 kbps.