- Define the parametric all-pass filter, calculate its transfer function, show that the magnitude response is 1, and calculate its parameter for a cut-off frequency f_c where the phase response is $-90°$.
- How do you get a parametric low- and high-pass filter from the parametric all-pass filter? Sketch the phase response of the all-pass filter and the magnitude response of the low- and high-pass filter. Explain the relationship.
- The second-order parametric all-pass filter is given by

  $$y[t] = (a_2 * x)[t] = -dx[t] + c(1-d)x[t-1] + x[t-2] - c(1-d)y[t-1] + dy[t-2]\,.$$

  Show that the magnitude response is 1. Sketch its phase response. How do you get a parametric band-pass filter from it?
- What is a shelving filter? What types are there? Why is its parameter calculated differently in the cut-case?
- What is a phaser? How is it constructed?
- How is a Wah-Wah effect constructed?
- What are fractional delays used for? Explain one method to implement them.
- How is a rotary speaker simulated?
- In which effects does a comb filter arise? Why and how is the gain of an IIR comb filter corrected?
- Show how upper and lower sidebands arise in a ring modulator.
- How do you get rid of the lower sideband in modulation?
- Describe the chain of functional units in dynamics processing.
- How does an averager with different attack- and release-times work?
- Explain noise gate, expander, compressor, limiter, clipper.
- Why and how does non-linear distortion introduce new frequencies? (Answer: Taylor $\to \cos^n(\omega t) \to \cos(k\omega t)$, and $(\cos + \cos)^n \to \cos \cdot \cos \to \cos((\omega_1 \pm \omega_2)t)$, exact formulas not necessary).)
- How can octavers shift the pitch up and down an octave?

- Why does aliasing occur in non-linear transformations of discrete (sampled) signals? How can it be avoided?
- Define and explain the overlap-add method for signal re-synthesis based on STFT. What is the summing condition for the used windows?
- Explain why phase unwrapping is needed in STFT-based time stretching, and how it works.
- How is sound mutation (morphing, vocoder effect) implemented with STFT?
- How is denoising implemented with STFT?
- Derive the peak detection by fitting a parabola to the logarithmic magnitudes of three neighboring frequency bins.
- In the digital resonator $x[t+1] = bx[t] - x[t-1]$, derive the value of b that produces a sinusoid of frequency ω.
- Explain the inverse Fourier transform method for sinusoid synthesis. Show what values have to be stored in a table to avoid evaluating cos- and sin-functions.
- What is the residual signal in the sinusoidal+residual signal model? How is it represented, analyzed and re-synthesized?
- Define the method of linear predictive coding. Derive the equation system to find the optimal filter coefficients.
- What is the Levinson-Durbin recursion algorithm used for? What is the basic idea? What is its complexity compared to conventional equation solvers.
- How is sound mutation (morphing, vocoder effect) implemented with LPC?
- What is the cepstrum? How can it be used for source-filter separation?
- How is sound mutation (morphing, vocoder effect) implemented by using the cepstrum?
- Explain SOLA and PSOLA.
- Which methods can be used to modify the apparent source direction with stereo loudspeakers?

- Explain “inter-aural intensity/time difference” and “head related transfer function”. Are they used with loudspeakers or headphones?
- What is the effect of decorrelation of stereo signals? How can it be achieved?
- Explain Ambisonics.
- Explain the room-within-a-room model for reverberation.
- What are normal modes? How are they used in artificial reverberation?
- Explain the components of Moorer’s reverberator.
- What is a feedback delay network? Define it.
- What are maximum-length-sequences? How and why are they used for recording room impulse responses?
- Explain the Fourier methods to implement convolution with long impulse responses. How can the latency by reduced?
- How does lossless coding with LPC work? What are Rice codes and why are they used here?
- What is the difference between forward and backward adaptive prediction?
- What is the MDCT? (Formula not required.) Why and how is it used in lossy audio coding?
- What are the two main effects in psychoacoustics? How are they used in lossy audio coding?