DSP topics

1. Basics of Digital Signal Processing
   1.1. Introduction to DSP
   1.2. Discrete-time signals, basic examples
   1.3. Operations with discrete-time signals, energy and power
   1.4. Classes of discrete-time signals, finite length and infinite length signals.
   1.5. Discrete-amplitude signals, noise robust transmission
   1.6. Discrete-time sounds. Sampling rate, sampling time. Digital and real frequencies.
   1.7. Karlup-Strong algorithm.
   1.8. Complex exponentials

2. Vector spaces
   2.1. Signal processing and vector spaces
   2.2. Vector spaces (linear spaces)
   2.3. Inner product, Hilbert spaces.
   2.4. Signal spaces
   2.5. Orthogonal systems, Bases
   2.6. Gram-Schmidt orthogonalization, Legendre polynomials
   2.7. Subspaces and approximations, orthogonal projection

3. Basics of Fourier Analysis
   3.1. Discrete Fourier Transform (DFT)
   3.2. Examples of DFT calculus
   3.3. Physical interpretation of DFT
   3.4. DFT analysis and synthesis
   3.5. Short-Time Fourier Transform, Spectrogram
   3.6. Discrete Fourier Series (DFS)
   3.7. Discrete-Time Fourier Transform (DTFT)
   3.8. Existence and properties of DTFT
   3.9. DTFT of basic discrete time signals.
   3.10. Sinusoidal Modulation
   3.11. Adding two sinusoids with near frequencies, tuning a guitar.

4. Introduction to Filtering
   4.1. Linear time-invariant systems (filters)
   4.2. Convolution, impulse response of system
   4.3. Acoustics of room, reverberation
   4.4. Moving average filter
   4.5. Leaky integrator
   4.6. Filter stability

4.7. Filter classification in the time domain, FIR, IIR
4.8. Frequency response. Convolution Theorem
4.9. Examples of frequency responses
4.10. Filter classification in the frequency domain
4.11. Amplitude response, phase response, examples
4.12. Ideal lowpass filter
4.13. Ideal filters derived from the ideal lowpass filter
4.14. Demodulation with lowpass filtering
4.15. Impulse truncation (window method) as an approximation of ideal (low pass) filter, Gibbs phenomenon
4.16. Frequency sampling as an approximation of ideal (low pass) filter
4.17. Realizable Filters. Constant Coefficients Differentiable Equations.
4.18. Z-transform
4.19. Z-transform of Realizable Filters, Filter coefficients.
4.20. Pole-zero interpretation of Linear Filters.
4.21. Region of convergence (ROC), system stability.
4.22. Design of filters. Leaky integrator, DC removal, resonator, hum removal.
4.23. Filter specifications.

5. Sampling and Quantization
5.1. Continuous-time signals, Fourier Transform (FT).
5.2. Polynomial interpolation, local interpolation, sinc interpolation.
5.3. Sampling of bandlimited functions, Sampling theorem (Nyquist theorem) formulation and application, consequencies.

6. Audio processing
6.1. Mel Frequency Cepstral Coefficients (MFCC)
6.2. Overlap Add (OLA) method.
6.3. Real time processing, e.g. reverberation computation.
6.4. Popular audio processing problems.