Speech signal processing using MATLAB

Basics and applications

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Slides and MATLAB scripts and data at https://github.com/murtex/spl

Outline

Digital signals
- Sampling
- Time domain
- Frequency domain
- Filters

Acoustic signals
- Short-time analysis
- Spectrograms
- Activity detection
- Landmarks detection
- Formants detection

Digital signals/Sampling

Sampling

- **continuous signal** (normalized magnitude, length L in seconds)

$$x(t) \in [-1, 1] \quad \text{with} \quad t \in [0, L]$$

```
>> x = @( t ) sin( 2*pi*f * t ); % continuous sine with frequency f
```

- **sampling rate** f_S, quantization of time

$$t \to t_i = \frac{i-1}{f_S} \quad \text{with} \quad i \in \{1, \ldots, N\} \quad \text{and} \quad N = \lfloor L f_S \rfloor$$

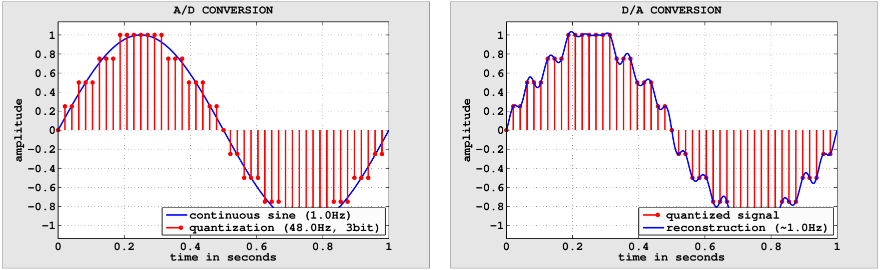
```
>> N = floor( L * fS ); % number of samples
>> ti = (0:N-1) / fS; % quantized time values
```

- **bits per sample** n_S, quantization of amplitude

$$x(t) \to x_i = \frac{\lfloor 2^{n_S - 1} x(t_i) \rceil}{2^{n_S - 1}}$$

```
>> xi = round( 2^(nS-1) * x( ti ) ) / 2^(nS-1); % quantized amplitudes
```

- **example:** `matlab/sampling.m`



- **exercise:**
  - verify from reconstruction that **Nyquist frequency** holds

$$f_{\mathrm{Ny}} = \frac{f_{\mathrm{S}}}{2}$$

  - compare commonly used **sampling standards** (telephony, Audio-CD, professional audio equipment, ...)

Digital signals/Time domain

- **total energy**, **average power** and **root mean square**

$$E = \sum_{i=1}^{N} x_i^2, \quad P = \frac{1}{N}\sum_{i=1}^{N} x_i^2 \quad \text{and} \quad RMS = \sqrt{\frac{1}{N}\sum_{i=1}^{N} x_i^2}$$

```
>> E = sum( xi .* xi ); % total energy
>> P = mean( xi .* xi ); % average power
>> RMS = sqrt( mean( xi .* xi ) ); % root mean square
```

- **decibel full scale**, different for **power-** and **magnitude-like** quantities, e. g.

$$P_{\mathrm{dB}} = 10\log_{10}(P) \quad \text{and} \quad RMS_{\mathrm{dB}} = 20\log_{10}(RMS)$$

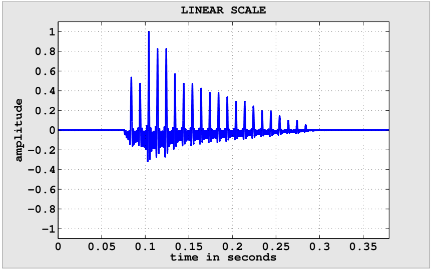
```
>> PdB = 10 * log10( P ); % power-like
>> RMSdB = 20 * log10( RMS ); % magnitude-like
```

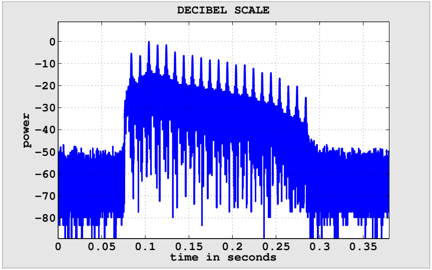
- **zero-crossings rate**

```
>> fZ = sum( abs( diff( xi >= 0 ) ) ) / N * fS;
```

Time domain

- **example:** `matlab/decibel.m` (`matlab/sound.wav`)





- **exercise:**
  - compare **linear** and **logarithmic scales**
  - explain **negative decibel values** (e. g. −3 dB power, −6 dB magnitude)
  - specify the power of silence in decibels

Digital signals/Frequency domain

Frequency domain

- **discrete Fourier transform**, time domain → frequency domain

$$X_k = \sum_{i=1}^{N} x_i \, \mathrm{e}^{-2\pi \mathrm{i} \frac{(i-1)(k-1)}{N}} \in \mathbb{C} \quad \text{with} \quad k \in \{1, \ldots, N\}$$

```
>> Xk = fft( xi ) / N; % complex Fourier coefficients
```

- k is a **frequency index** (as i was a time index)

$$k \to f_k = \frac{k-1}{N} f_\mathrm{S}$$

```
>> fk = (0:N-1) / N * fS; % frequency values
```

- frequencies beyond Nyquist frequency are **negative frequencies**

$$f_k \to \begin{cases} f_k - f_\mathrm{S} & \text{if } f_k > f_\mathrm{Ny} \\ f_k & \text{otherwise} \end{cases}$$

```
>> fk(fk > fNy) = fk(fk > fNy) - fS; % imply negative frequencies
```

Frequency domain

- **power spectral density** (also known as **power spectrum**)

$$P_k = |X_k|^2 \in \mathbb{R} \quad \Leftarrow \quad \sum_{k=1}^{N} P_k = P$$

```
>> Pk = abs( Xk ) .^ 2; % power spectral density
```

- **real valued signals** ($x_i \in \mathbb{R}$) imply a special symmetry

$$X_{f_k} = X^*_{-f_k} \quad \Rightarrow \quad P_{f_k} = P_{-f_k}$$

- restrict to **one-sided spectrum**

```
>> Pk(fk < 0) = []; % remove negative frequency components
>> Xk(fk < 0) = [];
>> fk(fk < 0) = [];
>> Pk(2:end) = 2 * Pk(2:end); % rescale to match total power
>> Xk(2:end) = sqrt( 2 ) * Xk(2:end);
```

- P_1 is **DC offset**, $P_{k>1}$ are **contributions of sines** with frequencies $f_{k>1}$

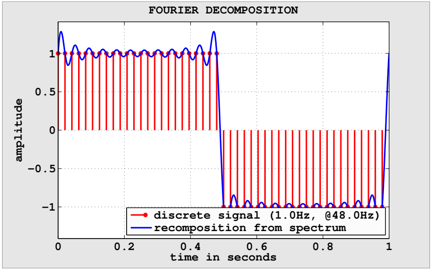
$$x(t) = \sqrt{P_1} + \sqrt{2} \sum_{k>1} \sqrt{P_k} \sin(2\pi f_k t)$$

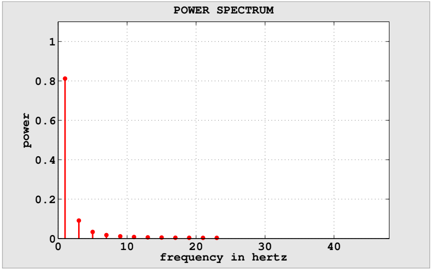
Frequency domain

- complex valued but without loss of phase information

$$x(t) = X_1 + \sqrt{2} \sum_{k>1} X_k e^{2\pi i f_k t}$$

- **example:** `matlab/fdomain.m`



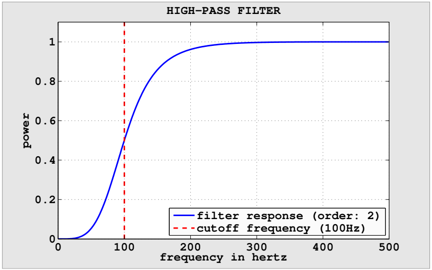


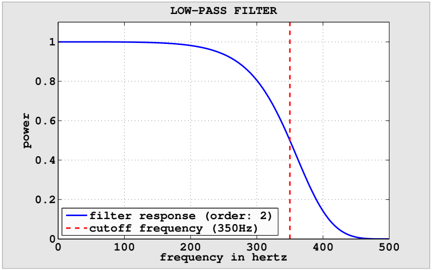
- **exercise:**
  - examine spectra of different wave forms (sines, square, sawtooth, ...)
  - examine spectral **frequency range**
  - verify loss of **phase information** in (real valued) power spectra

Digital signals/Filters

Filters

- general **filter types:**
  - **low-pass:** passes low frequencies (cuts high ones)
  - **high-pass:** passes high frequencies (cuts low ones)
  - **band-pass:** passes a range of frequencies (combination of low- and high-pass)
  - **band-stop** (notch): cuts a range of frequencies (opposite of band-pass)
- **cutoff frequency** at which output power is (generally) reduced by −3 dB
- **example:** `matlab/filters.m`

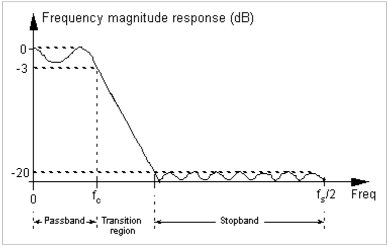


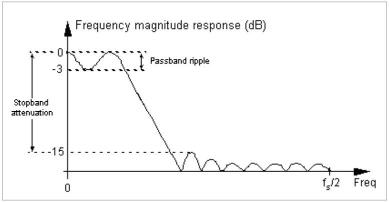


- filters are represented by **filter coefficients** b_j (feedforward) and a_j (feedback)
- high **filter order** m increases computational complexity but thereby quality

$$y_i = \overbrace{\frac{1}{a_1}\left(\underbrace{\sum_{j=0}^{m} b_{j+1} x_{i-j}}_{\text{FIR}} - \sum_{j=1}^{m} a_{j+1} y_{i-j}\right)}^{\text{IIR}} \quad \text{with} \quad i \in \{1, \ldots, N\}$$

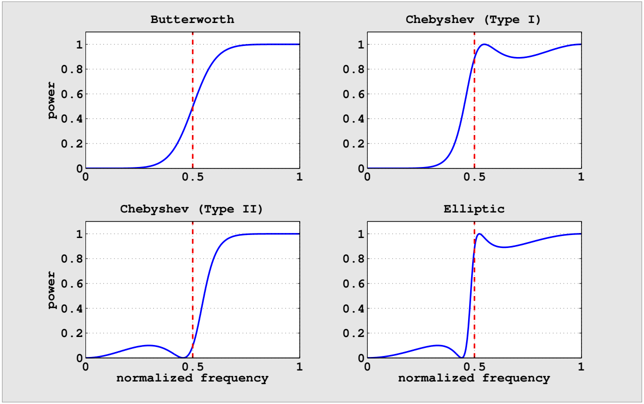
- **FIR filters** (finite impulse response) are slow to compute but stable
- **IIR filters** (infinite impulse response) are fast to compute but might be unstable
- some often used **additional terms** (images from `http://dspguru.com`)





Filters

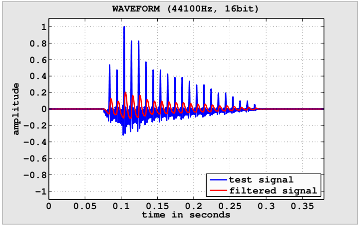
- **example:** `matlab/filters2.m`

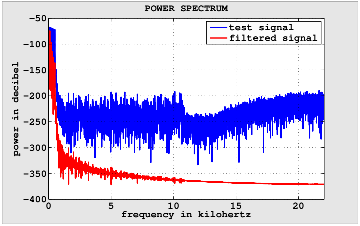


- many **filter families** with different characteristics
- **normalized frequency**

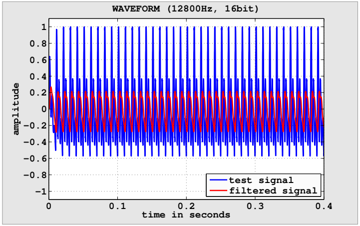
$$\tilde{f}_k = \frac{f_k}{f_{\mathrm{Ny}}} = \frac{2f_k}{f_{\mathrm{S}}} \in [0, 1] \quad \text{with} \quad k \in \{1, \ldots, N\}$$

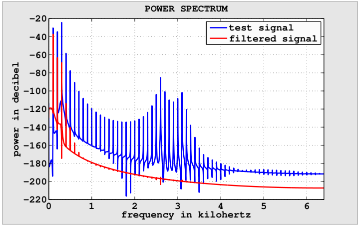
- **example:** `matlab/spectrum.m` (`matlab/sound.wav`)





- **example:** `matlab/spectrum.m` (`matlab/ivowel.wav`)





- **exercise:**
  - observe the occurrence of **filter delay**

- **Butterworth filter** (high-pass, second-order, 100 Hz cutoff)

```
>> m = 2; % filter order
>> cutoff = 100; % cutoff frequency
>> [b, a] = butter( m, cutoff / (fS/2), 'high' );
```

- **Chebyshev filter** (high-pass, 1 dB ripple, 40 dB attenuation, 100 Hz cutoff)

```
>> cutoff = 100; % cutoff frequency
>> stopband = 90; % stopband frequency
>> ripple = 1; % passband ripple
>> attenuation = 40; % stopband attenuation
>> m = cheb2ord( cutoff / (fS/2), stopband / (fS/2), ripple, attenuation );
>> [b, a] = cheby2( m, attenuation, stopband / (fS/2) );
```

- **apply any filter**

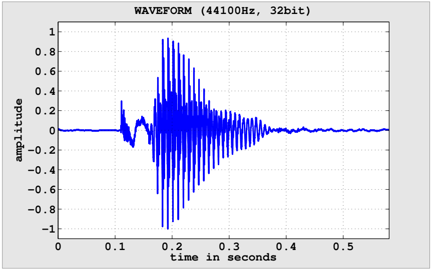
```
>> y = filter( b, a, x ); % filter signal x using coefficients a, b
```

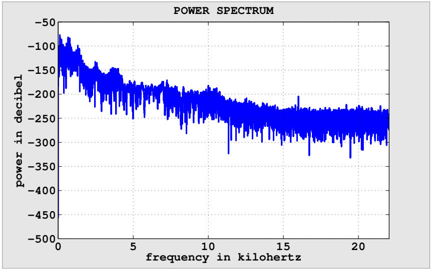
- or in zero-phase version (without filter delay)

```
>> y = filtfilt( b, a, x ); % zero-phase filtering
```

Acoustic signals/Short-time analysis

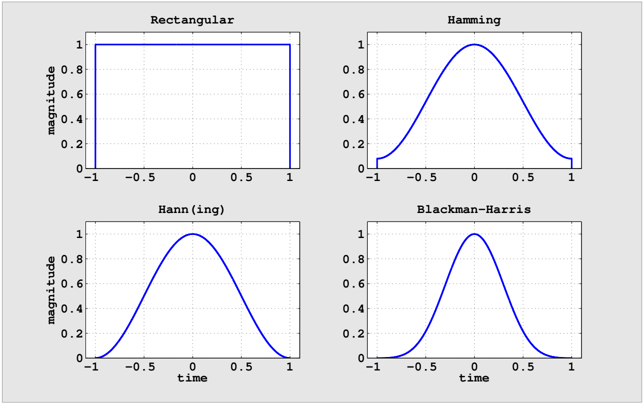
- **spectral analysis** is essential for speech acoustics
- **power spectrum** has no temporal information anymore (`matlab/tam.wav`)





- choosing **short overlapping segments** (windows) at different time points
- length of the segments (**window size**) is crucial
- **overlap** and **window function** control spectral leakage

- **example**: `matlab/windows.m`



- Fourier transforms of these segments lead to **spectrograms**

Acoustic signals/Spectrograms

Acoustic signals/Activity detection

Activity detection

Acoustic signals/Landmarks detection

Acoustic signals/Formants detection

Formants detection