Speech signal processing using MATLAB

Basics and applications

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

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

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

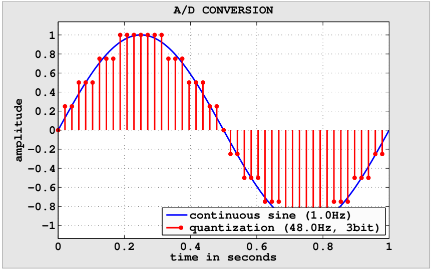
```
>> N = floor( L * fS ); % number of samples
>> ti = (0:N-1) / fS; % quantized time values
```

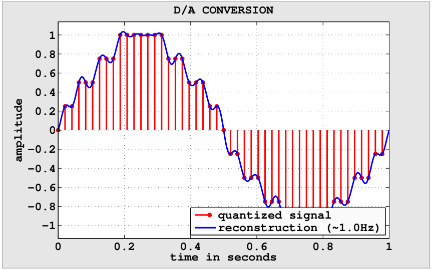
- **bits per sample** n_S, quantization of amplitude

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

```
>> 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

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.

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

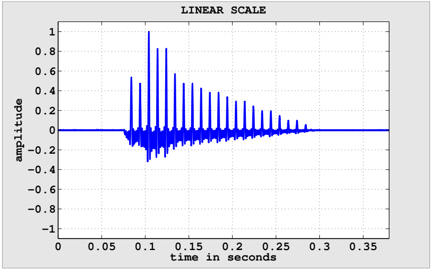
```
>> PdB = 10 * log10( P ); % power-like
>> RMSdB = 20 * log10( RMS ); % magnitude-like
```

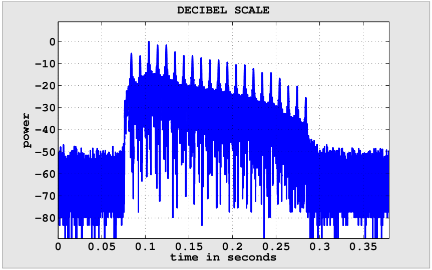
- **zero-crossings rate**

```
>> 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)

$$k \to f_k = \frac{k-1}{N} f_S$$

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

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

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

```
>> 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 $f_{k>1}$

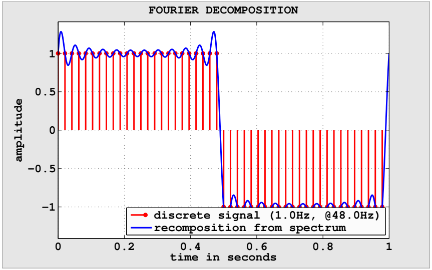
$$x(t) = \sqrt{P_1} + \sqrt{2} \sum_{k>1} \sqrt{P_k} \sin(2\pi f_k t)$$

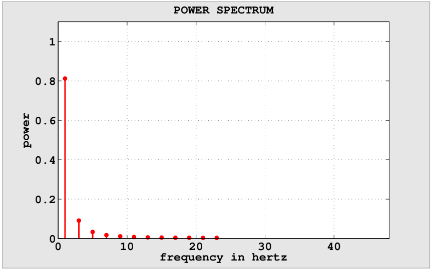
Frequency domain

- 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`



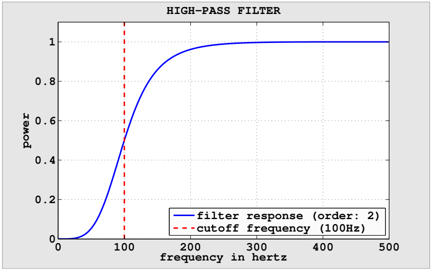


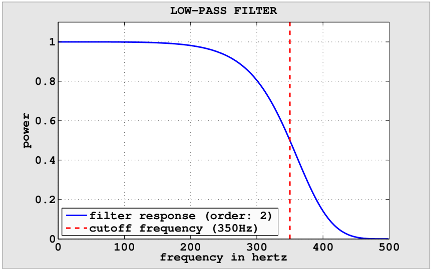
- **exercise:**
  - examine spectra of different wave forms (sines, square, sawtooth, …)
  - examine spectral **frequency range**
  - verify loss of **phase information** in (real valued) power spectra

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`

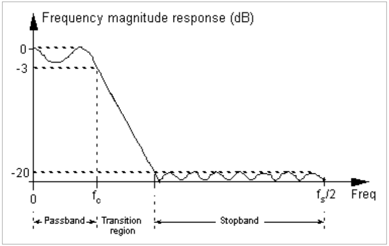


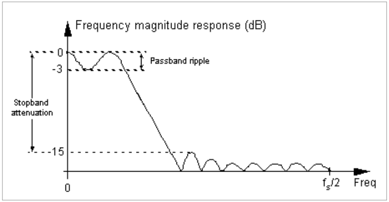


- filters are represented by **filter coefficients** b_j (feedforward) and a_j (feedback)
- high **filter order** m increases computational complexity but thereby quality

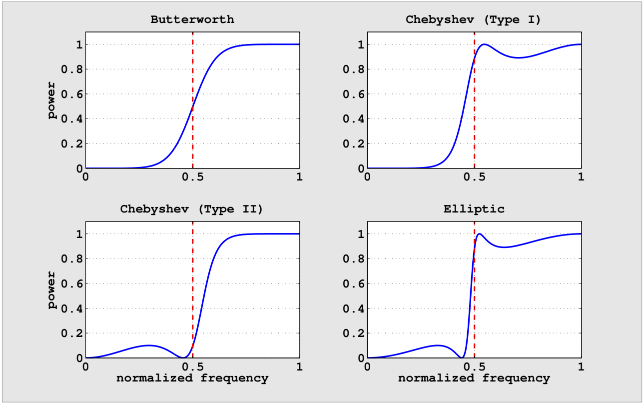
$$y_i = \overbrace{\frac{1}{a_1}\left(\underbrace{\sum_{j=0}^{m} b_{j+1} x_{i-j}}_{\text{FIR}} - \sum_{j=1}^{m} a_{j+1} y_{i-j}\right)}^{\text{IIR}} \quad \text{with} \quad i \in \{1,\ldots,N\}$$

- **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`)





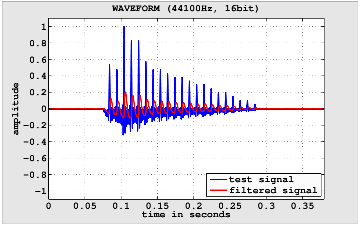
- **example**: `matlab/filters2.m`

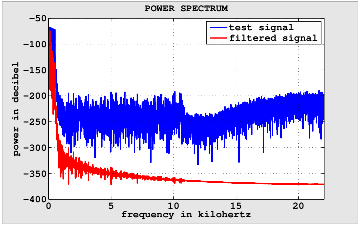


- many **filter families** with different characteristics
- **normalized frequency**

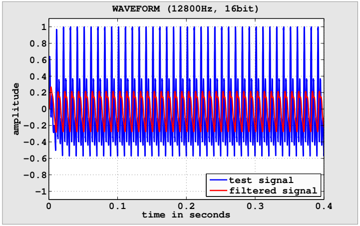
$$\tilde{f}_k = \frac{f_k}{f_{\mathrm{Ny}}} = \frac{2f_k}{f_{\mathrm{S}}} \in [0,1] \quad \text{with} \quad k \in \{1,\ldots,N\}$$

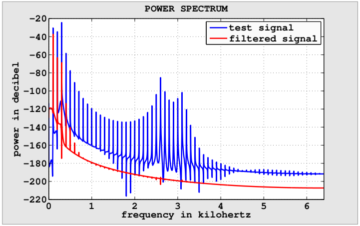
- **example:** `matlab/spectrum.m` (`matlab/sound.wav`)





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





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

Filters

- **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**

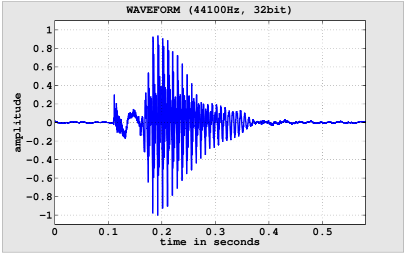
```
>> y = filter( b, a, x ); % filter signal x using coefficients a, b
```

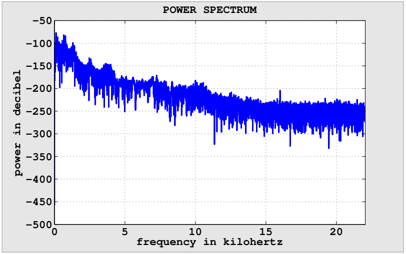
- or in zero-phase version (without filter delay)

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

Acoustic signals/Short-time analysis

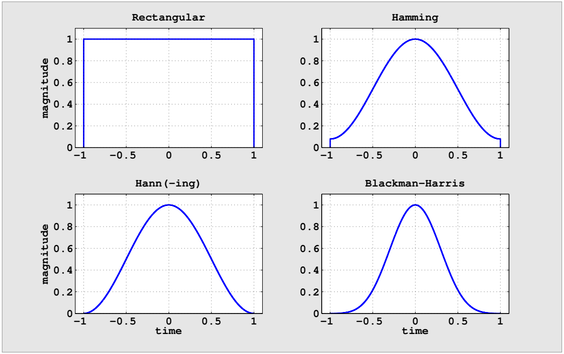
- **spectral and temporal analysis** is essential for speech acoustics
- problem:
  - **power spectrum** has no temporal information anymore (`matlab/tam.wav`)





- solution:
  - choose **short overlapping segments** (windows) at different time points
  - length of the segments (**window size**) is crucial
  - **overlap** and **window function** control spectral leakage
  - aligning Fourier transforms of these (altered) segments leads to **spectrograms**

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



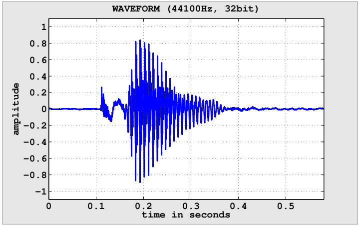
- **optimal overlapping** for minimal spectral leakage

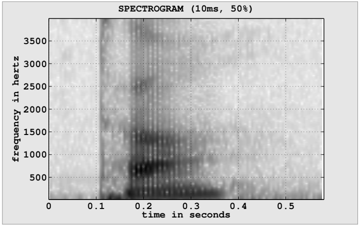
Rectangular:	any value
Hamming:	50%
Hann(-ing):	50%
Blackman-Harris:	66.1%

- other commonly used window functions: **Welch, Kaiser, Gaussian,** …

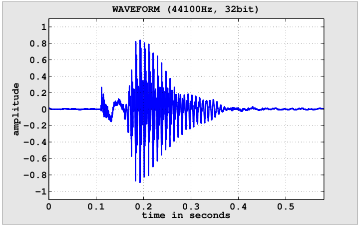
Acoustic signals/Spectrograms

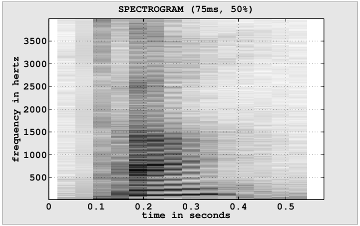
- **example:** `matlab/specgram.m` (`matlab/tam.wav`)





- **example:** `matlab/specgram.m` (`matlab/tam.wav`)





- **exercise:**
  - impact of **window size** → **broad-band** vs. **narrow-band** spectrogram

- **broad-band spectrograms** have good temporal but poor spectral resolution
- **narrow-band spectrograms** have poor temporal but good spectral resolution

spectrogram:	**broad-band**	**narrow-band**
window size:	< 20 ms	> 20 ms
structures:	**formants**	**harmonics**

- set up windowing

```
>> wsize = 10; % window size in milliseconds
>> woverlap = 66; % window overlap in percent
>> wfunc = @blackmanharris; % window function
```

- compute the spectrogram

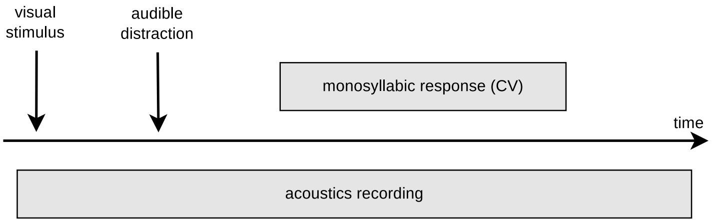
```
>> [Xk, fk, ti] = spectrogram( xi, ... % signal
       wfunc( ceil( wsize/1000 * fS ) ), ... % window function values
       ceil( woverlap/100 * wsize/1000 * fS ), ... % window overlap samples
       4096, fS ); % fourier transform
```

- plot the spectrogram

```
>> colormap( flipud( colormap( 'gray' ) ) );  % set color coding
>> imagesc( ti, fk, Pk .^ 0.1 );  % plot spectral powers
```

Acoustic signals/Activity detection

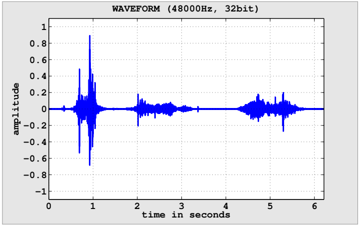
- **experimental data** often contain a lot of noise and little of information
- for **automatic processing** restriction to important parts is essential
- consider the following experiment:

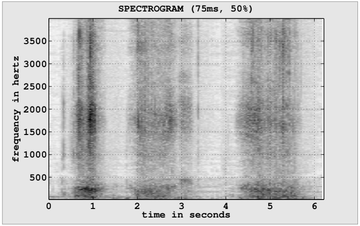


- with features of interest:
  - responded syllable (out of a specific set → classifcation task)
  - voice onset time (→ **landmarks detection**)
  - formants onsets (frequency and time → **formants tracking/detection**)
- all of these require (human) **activity detection** as an initial processing pass

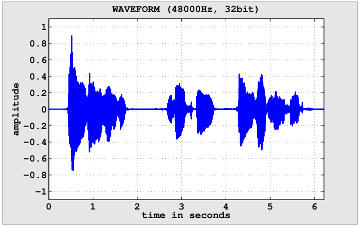
Activity detection

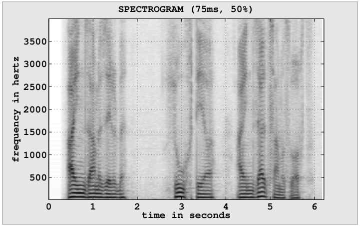
- in literature usually called **voice activity detection** (VAD)
- exploiting **spectral differences** in human speech and ambient sound/noise
- **example:** `matlab/specgram.m` (`matlab/chair.wav`)



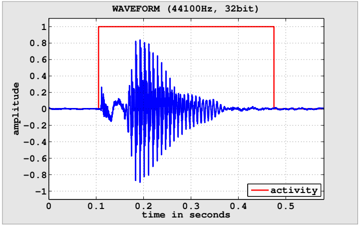


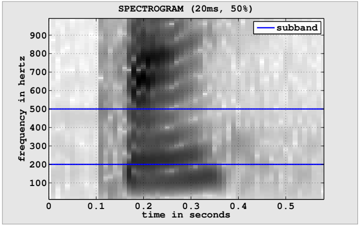
- **example:** `matlab/specgram.m` (`matlab/haiku.wav`)



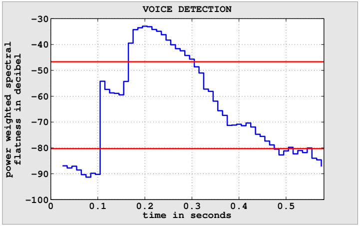


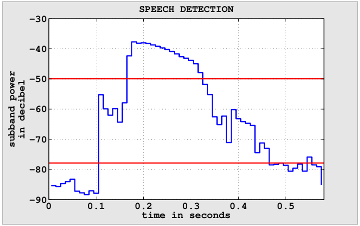
- **example:** `matlab/activity.m` (`matlab/tam.wav`)
- applying an **equal loudness filter** and limiting to **frequency band** 0 . . . 1000 Hz





- **adaptive thresholds** for power-weighted **spectral flatness** and **subband power**





- combining **activity states** based on thresholds gives overall **voice activity**

- D. Robinson. *Equal loudness filter.* 2001.
- D. Burileanu, L. Pascalin, C. Burileanu, M. Puchiu. *An adaptive and fast speech detection algorithm.* Springer, 2000.
- M. Prcin, L. Müller. *Heuristic and statistical methods for speech/non-speech detector design.* Springer, 2002.
- Y. Ma, A. Nishihara. *Efficient voice activity detection algorithm using long-term spectral flatness measure.* Springer, 2013.

Acoustic signals/Landmarks detection

Acoustic signals/Formants detection

Formants detection