SPEAKER CHANGE DETECTION USING FUNDAMENTAL FREQUENCY WITH APPLICATION TO MULTI-TALKER SEGMENTATION

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What is speaker diarization?

Answers the question “who spoke when?” in an audio recording.

Is diarization really that useful?

- Speaker indexing and rich transcription
- Speaker segmentation and clustering helping Automatic Speech Recognition (ASR) systems
- Preprocessing modules for single speaker-based algorithms
DIARIZATION METHOD
Is good segmentation really that useful?

Why not just segment the audio stream into small uniform segments and cluster with realignment?

If the speech segments are small then each segment only contains a small amount of information that can be used for clustering.
SPEAKER PITCH TRACKS
Multi-modal data set consisting of 100 hours of meeting recordings.

Recorded in English using three different rooms with different acoustic properties and includes mostly non-native speakers.
Speaker Pitch Tracks

SPEAKER PITCH TRACKS FROM ‘ES2004B’

Speaker Pitch Tracks

Estimated pitch (Hz)

Time (s)

Speaker A
Speaker B
Speaker C
Speaker D
SPEAKER PITCH TRACKS FROM ‘TS3003B’

Speaker Pitch Tracks

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PITCH SEGMENTATION
Assumption: If the speaker’s pitch only varies in a smooth manner due to physiological constraints (Xu, 2002) it should be possible to estimate the future pitch of the speaker based on their current pitch.

Main Idea: Use a Kalman filter to carry out this future pitch estimation. If the pitch can’t be estimated then the speaker has potentially changed.
Proposed pitch segmentation system
The pitch $x(n)$ for a given frame $n$ can be written in the following way:

$$x(n + 1) = x(n) + w,$$

$$w \in \mathcal{N}(0, \sigma_w^2).$$

The measurement $z(n)$ of the true pitch $x(n)$ can be modelled according to:

$$z(n) = x(n) + v,$$

$$v \in \mathcal{N}(0, \sigma_v^2).$$
Performed on every frame

Predicted pitch estimate:

\[ \hat{x}_{n|n-1} = \hat{x}_{n-1|n-1}. \]

Predicted estimate variance:

\[ P_{n|n-1} = P_{n-1|n-1} + \sigma_w^2. \]
Performed if the frame is considered to be voiced

Updated pitch estimate and updated estimate variance:

\[
\hat{x}_{n|n} = \hat{x}_{n|n-1} + K_n(z_n - \hat{x}_{n|n-1})
\]

\[
P_{n|n} = (1 - K_n)^2P_{n|n-1} + K_n^2\sigma_v^2.
\]

If the Kalman gain is \(K_n = 1\):
\[
\hat{x}_{n|n} = z_n \quad (\text{just the measurement})
\]

If the Kalman gain is \(K_n = 0\):
\[
\hat{x}_{n|n} = \hat{x}_{n|n-1} \quad (\text{just the prediction})
\]

Optimal Kalman gain:

\[
K_n = \frac{P_{n|n-1}}{S_n}.
\]
VARiance ‘P’

Pitch Segmentation

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A Kalman filter is initialised and tracks first speaker.

If the error between measurement and prediction becomes larger than a threshold (10 Hz) then all previously generated Kalman tracks are checked.

- If the closest previous Kalman pitch track is below a threshold (50 Hz) then this Kalman filter is continued.

- If on the other hand, the closest Kalman filter to the measurement does not satisfy this threshold then a new Kalman filter is generated.
GROUND TRUTH
### Ground Truth

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<th>Meeting</th>
<th>SC</th>
<th>PC</th>
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<tr>
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<tr>
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**SC | PC** The probability that there is a ‘speaker change’ given that there is a ‘pitch change’

**PC | SC** The probability that there is a ‘pitch change’ given that there is a ‘speaker change’
EVALUATION
MFCC VS PITCH SEGMENTATION

EVALUATION

Benchmark system (‘Sidekit’)

https://projets-lium.univ-lemans.fr/s4d/

Proposed system
500 ms collar around each speaker change boundary (250 ms before and after)
500 ms collar around each speaker change boundary (250 ms before and after)
EVALUATION COMPARISON

500 ms collar around each speaker change boundary (250 ms before and after)
The proposed Kalman filter prediction error-based approach performed well when compared against a previous MFCC-based method.

An evaluation on the AMI corpus showed a speaker changed detection increase from 43.3% to 70.5%.
In this paper we have...

...carried out a study of meetings in the AMI corpus that has shown that a pitch change is a strong indicator of a speaker change.

...highlighted that an individual’s pitch is smoothly varying and, therefore, can be predicted by using a Kalman filter.

...proposed a Kalman filtering approach to identify speaker change boundaries based on a model of the temporal variation of pitch.
Questions?