

Unsupervised Learning with Speech

M.Tech. Artificial Intelligence, Second Year, NMIMS

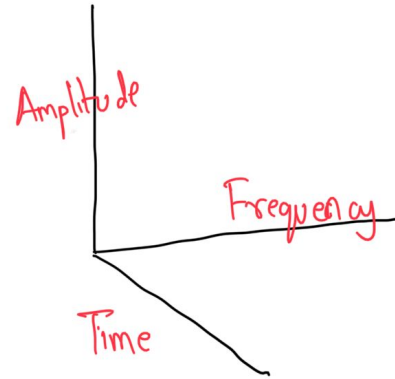
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Sound Conversion:



→ Amplitude
Frequency
Time



Helps for applying
various transformation

→ Frequency are key
characteristics which composed sounds

→ Chunk of information are encoded
in the generation of frequency and their
amplitude over time

Time and Frequency = Spectrogram
Representation

Time Normalization: Template Matching

→ Deterministic Sequence Recognition

Mean Sequence of 12 MFCCs for two words

$$x = [x_1, x_2, x_3, \dots, x_{12}]$$

$$y = [y_1, y_2, y_3, \dots, y_{12}]$$

$$D(x, y) = \sqrt{\sum_i (x_i - y_i)^2} \Rightarrow \text{Euclidean Distance}$$

→ When the words are uttering, the length of the utterance will change due to natural variations.

→ This makes the sequences have different lengths.

→ In order to handle time variation, we need two time normalization technique

→ Linear Time Warping

→ Dynamic Time Warping

Linear Time Warping

→ Linear Function used for twisting two template

→ Words such as

S p e e c h can be uttered as

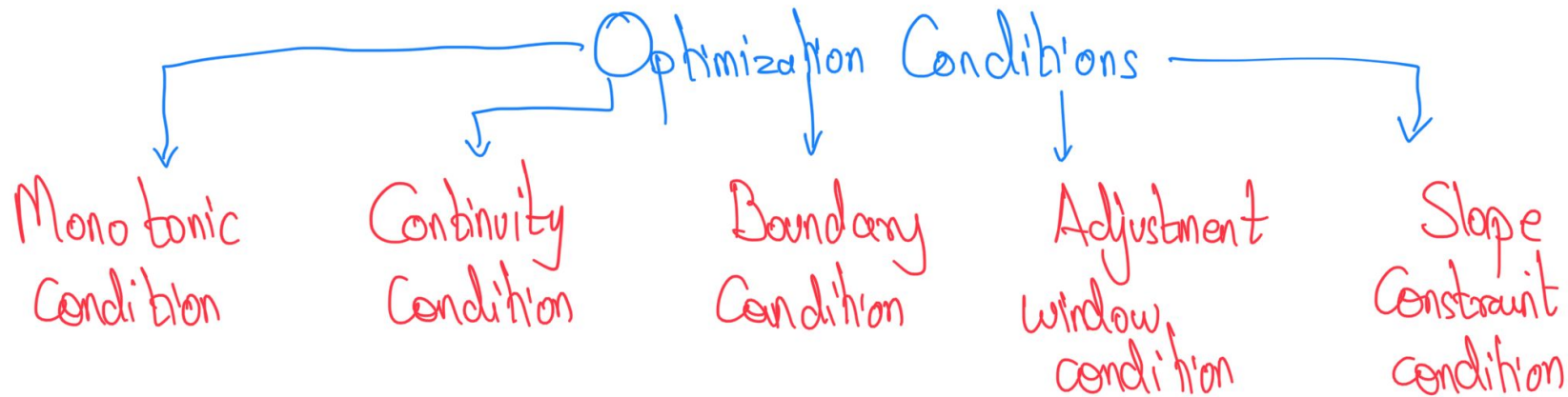
S s p e e h h

As this is a linear function LTW will not help
because the length of speech utterance will not be same.

→ Thus dynamic warping is required

Dynamic Time Warping

- Helps to compare words with different sequences
- The goal is to optimize the dynamic function and minimize the global distance error



Algorithm for DTW:

- Start with the first point in each template, compute the distance (this will be the cost function)
- Move on to the next point in a shorter template and find its cost function with the next point in the longer template
- Go on traversing in the forward direction for each point in the shorter template till both end points meet

$$D(i, j) = \min(D(i+1, j+1), D(i+1, j), D(i, j+1) + d(i, j))$$

