# CMPT 825 Natural Language Processing

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### Automatic Speech Recognition

- Acoustic observations: signal processing to extract energy levels at each frequency level
- Observation sequence **o** is composed of acoustic features extracted from the waveform at regular (10msec) intervals
- ASR is the task of converting the observation sequence o into a transcription

# Noisy Channel Model

• Finding the best transcription w\* given an observation sequence o

$$\mathbf{w}^* = \frac{\arg \max}{\mathbf{w}} P(\mathbf{w} \mid \mathbf{o}) = \frac{\arg \max}{\mathbf{w}} \frac{P(\mathbf{o} \mid \mathbf{w})P(\mathbf{w})}{P(\mathbf{o})}$$

$$= \frac{\arg \max}{\mathbf{w}} \underbrace{P(\mathbf{o} \mid \mathbf{w})P(\mathbf{w})}_{\text{generative language}}$$

$$= \max_{\mathbf{w}} \underbrace{P(\mathbf{o} \mid \mathbf{w})P(\mathbf{w})}_{\text{generative language}}$$

#### Generative Models of Speech

- Typical decomposition of P(w | o) into a cascade of generative models:
  - Acoustic Model:
    - P(o | p) predict observation sequence o given phone sequence p
  - Pronunciation Model:
    - P(w | p) predict phone sequence p given a word sequence w
  - Language Model:
    - P(w) predict word sequence w

### Generative Models of Speech

- Further decomposition of the acoustic model: P(o | p)
  - P(o | d) observation vectors given distribution sequences (quantitative given symbolic)
  - P(d | m) distribution sequences given model sequences (model dependent phone sequences)
  - P(m | p) model sequences given phone sequences

- 1920s: Radio Rex
  - 500 Hz of energy in the word "Rex" caused the toy dog to move
- 1950s: Digit Recognition (Bell Labs)
- 1960s: Advances in Signal Processing and Neural Nets (not much progress in ASR)
- 1970s: Despite large ARPA funding, not much success

- 1980s: Discrete ASR, Language Models, corpus collection efforts
  - TIMIT corpus (phonetics)
  - ATIS corpus (Air Travel Information System)
  - Focus on language understanding dialog systems

- 1990s: Large Vocabulary Continuous ASR
  - Dynamic Time Warping (edit distance)
  - Better phonetic models using classifiers (decision trees and neural nets)
  - Better language models using smoothing
  - Larger corpora: 10<sup>7</sup> and 10<sup>9</sup> in size

- Current Work
  - Other languages and dialects
  - Multiple speakers, Speaker adaptation
  - Speaker identification
  - Noise resistant (telephone speech)
  - Open source software: HTK, Sphinx, CMU LM toolkit, SRI LM toolkit