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Natural Speech Technology Programme Overview

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http://www.natural-speech-technology.org

NST Programme Grant



- At the outset of NST we identified several weaknesses with speech technology systems
 - Fragile operation across domains
 - Synthesis and recognition developed independently
 - Reliance on supervised approaches, manually transcribed training data
 - Models for synthesis and recognition include relatively little speech knowledge
 - Models only weakly factor the underlying sources of variability
 - Systems react crudely (if at all) to the context / environment
- These weaknesses still drive our objectives

NST Technical Objectives



• Learning and Adaptation

 learning to compactly represent speech and to adapt to new scenarios and speaking styles

Natural Speech Transcription

• Speech recognition systems that operate seamlessly across domain and acoustic environment

• Natural Speech Synthesis

• Controllable synthesisers that learn from data, and can generate expressive conversational speech

• Exemplar Applications

 prototype deployment in applications, focusing on health/social domain, media, and the needs of User Group stakeholders

NST Highlights 2014-15



- Best paper awards at IEEE SLT-2015, IEEE ICASSP-2014
- Open source software HTK, Kaldi, HTS
- Speech Recognition applications BBC (NewsHack and MGB Challenge), Ericsson (Just-in-time ASR), MediaEval, Browsing oral history (English Heritage)
- Voice banking and reconstruction
- homeService
- Challenges and Evaluations Spoofing challenge at



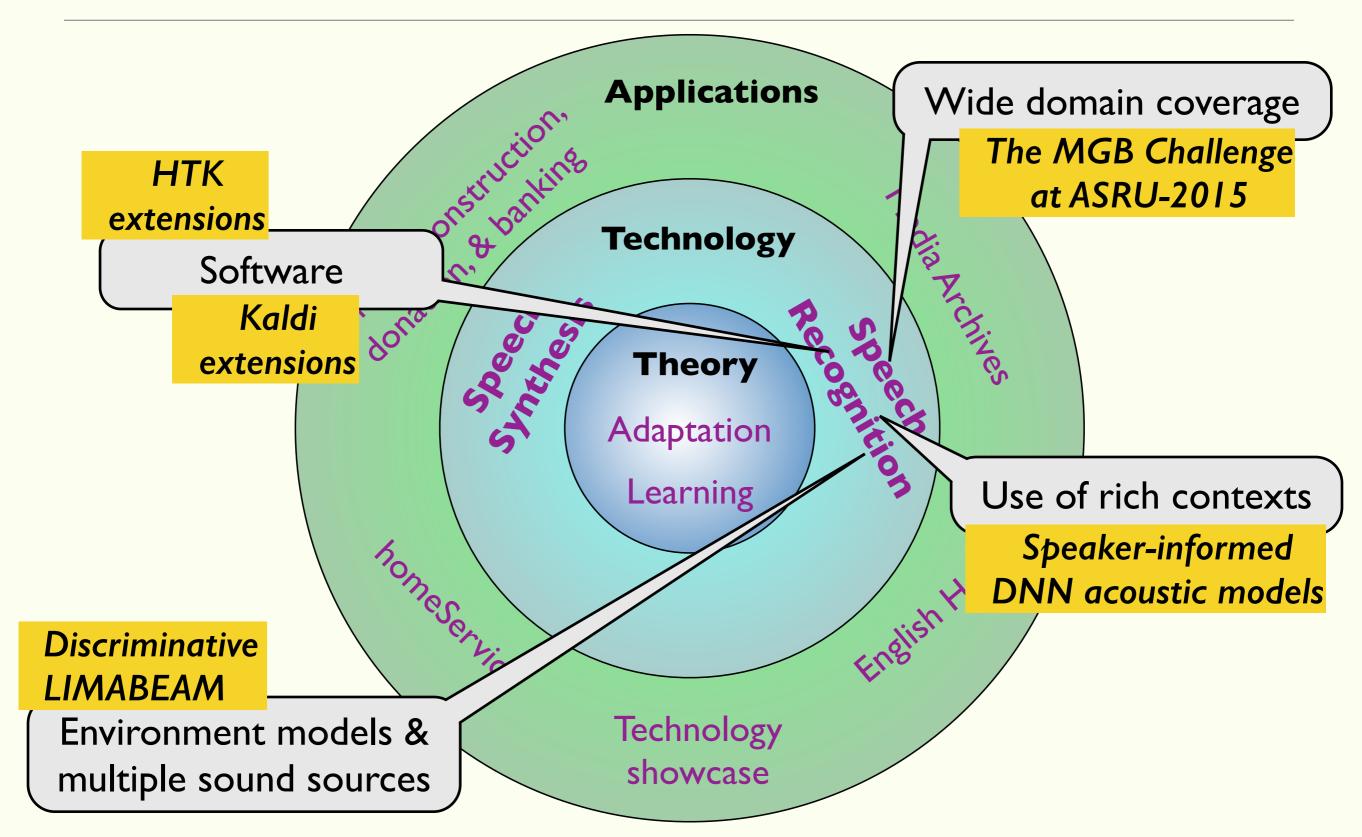
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Learning and adaptation

Applications Canonical acoustic Structuring diverse data Iction. models& adaptation nking **Corpora: new collections** Adaptation of DNN Voice reconst & structuring diverse data acoustic models adia Archives Technology Speech 22 Sold **Cheory** Adaptation Learning Canonical language models Training, adaptus. decoding using RNNLMs Service **Bottleneck features** teritz for DNN synthesis English Learning speech representations **RNN** encoder-decoder Technology for large vocabulary ASR showcase

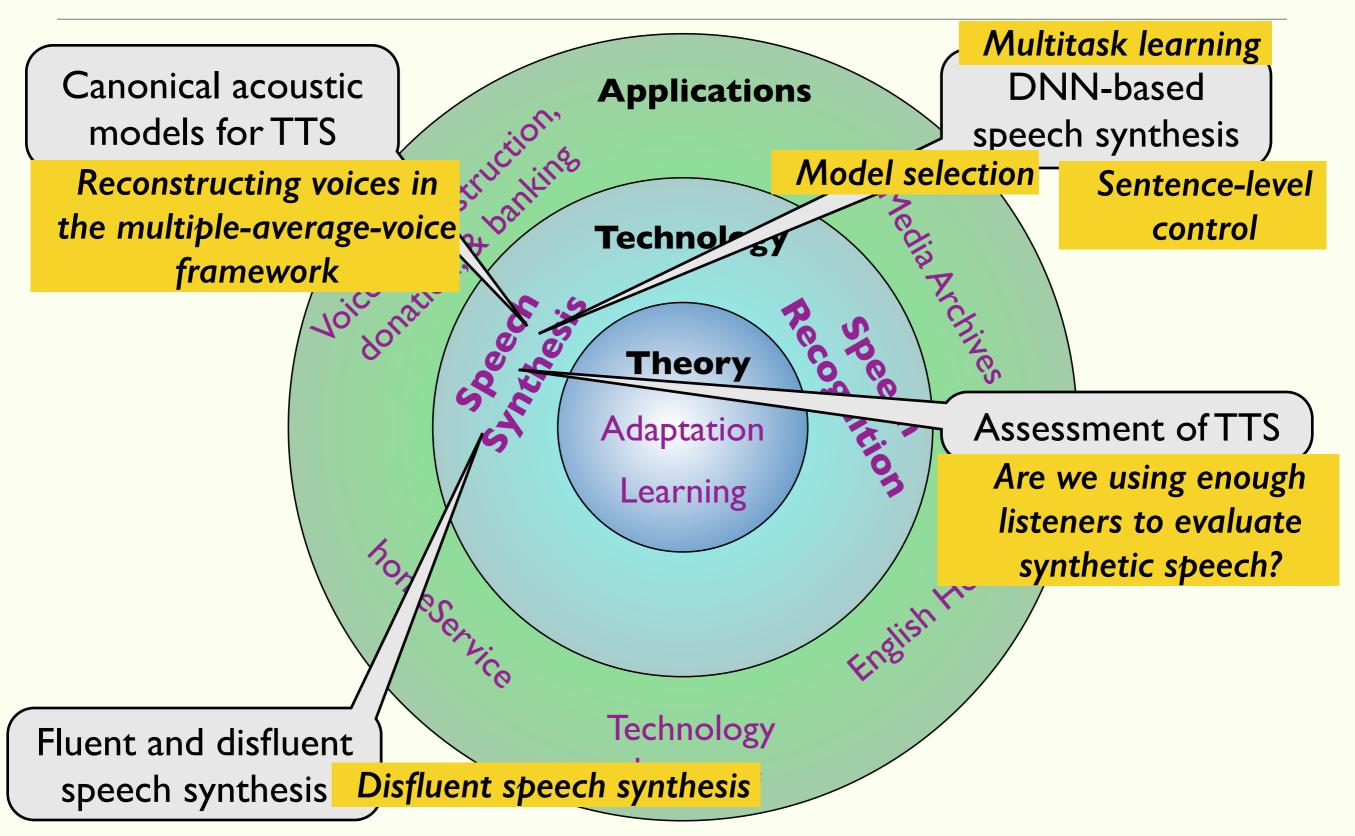


Natural transcription





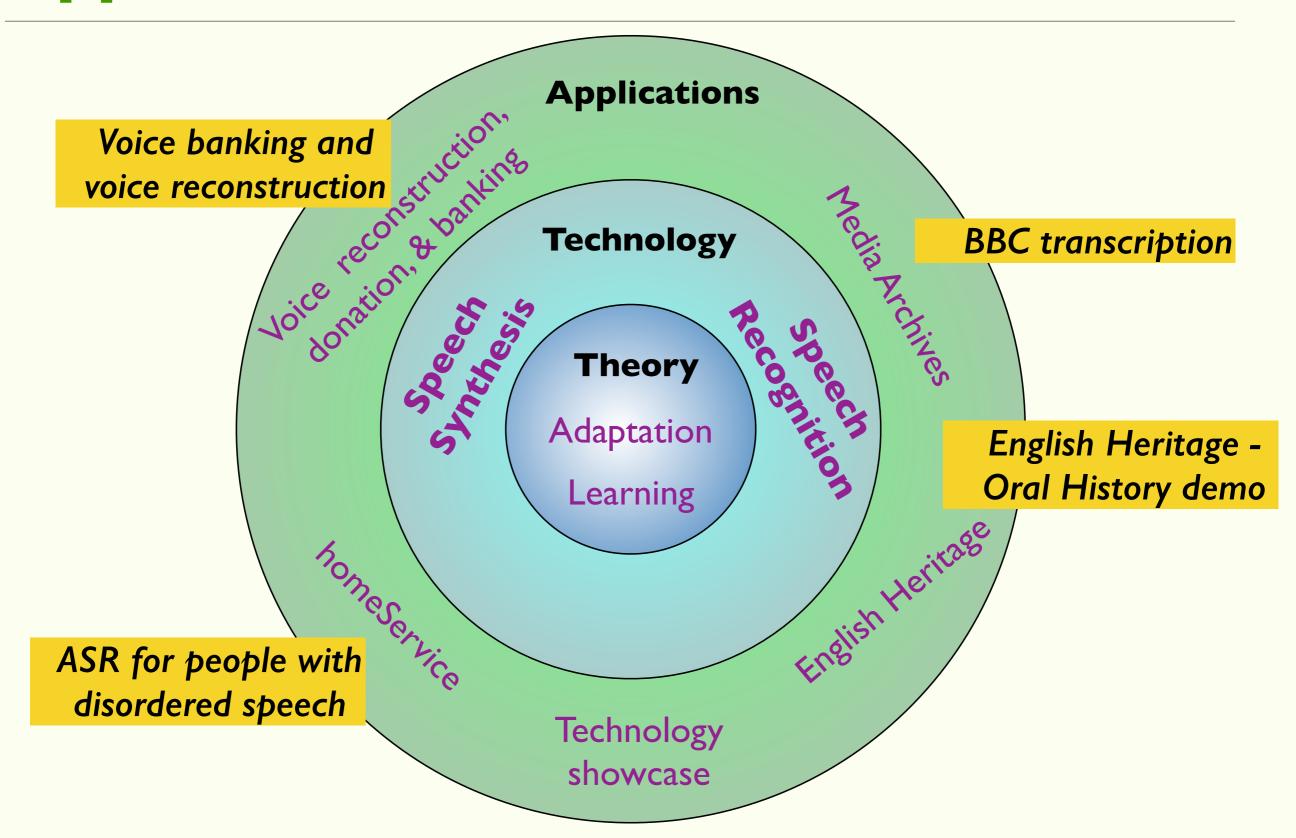
Natural Synthesis





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Applications



HTK ANN Extensions



- HTK 3.5 will support ANNs, maintaining compatibility with most existing functions.
 - Minimises the effort to reuse previous source code and tool
 - Allows transfer of e.g. SI/SD input transforms, MPE/MMI sequence training
 - 64-bit compatible
- Generic extensions
 - Flexible input feature configurations
 - ANN structures can be any directed acyclic graph
 - Stochastic gradient descent supporting frame/sequence training
 - CPU/GPU math kernels for ANNs
- Decoders extended to support tandem/hybrid systems, system combination

HTK Language Model extensions

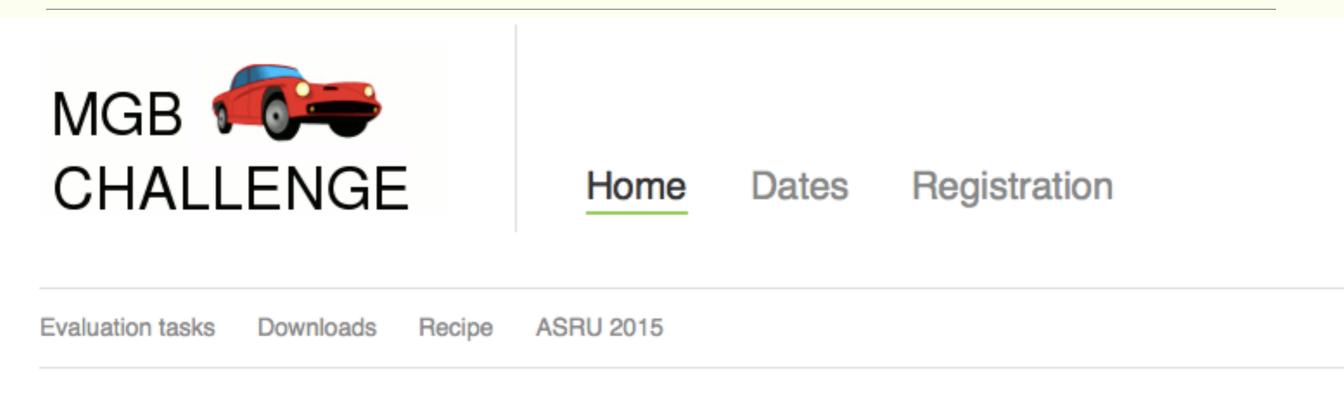


- HTK v3.5 support for decoding RNN language models
 - Lattice rescoring using RNNLMs
 - Class / Full word outputs, interpolation with n-grams
 - Similar functionality for feed-forward NN LMs
- RNNLM estimation enhancements
 - bunch mode GPU training
 - full/class output RNN LMs
 - NCE training
 - variance regularised training

MGB Challenge



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The challenge

The Multi-Genre Broadcast (MGB) Challenge is a new evaluation of speech recognition, speaker diarization, and lightly supervised aligner the British Broadcasting Corporation (BBC). It is an official challenge of the 2015 IEEE Automatic Speech Recognition and Understanding Wor

The speech data is broad and multi-genre, spanning the whole range of BBC TV output, and represents a challenging task for speech technology

The challenge will use a fixed training set of about 1,600 hours of broadcast audio, together with several hundred million words of subtitle text is provided to challenge participants, subject to signing a licence agreement with the BBC. The challenge will explore speech recognition and speaker longitudinal setting - i.e. transcription and speaker diarization and linking of several episodes of the same programme. All tasks will also offer the make use of supplied metadata including programme title, genre tag, and date/time of transmission, enabling novel approaches for domain and applied.

Spoofing Challenge



ASVspoof 2015: Automatic Speaker Verification Spoofing and Countermeasures Challenge

Introduction

Do you have a method/algorithm to discriminate between human and synthetic speech (generated from speech synthesis or voice conversion systems)? If so, you are invited to take part in the Automatic Speaker Verification Spoofing and Countermeasures (ASVspoof) Challenge.

Previously, both spoofing attacks and countermeasures have been developed with full knowledge of a particular speaker verification system used for vulnerability assessments. Similarly, countermeasures have been developed with full knowledge of the spoofing attack which they are designed to detect. This is clearly unrepresentative on the real use case scenario in which the specific attack, much less the specific algorithm, can never been known a priori. It is thus likely that the prior work has as much over-exaggerated the threat of spoofing as it has the performance of countermeasures.

The ASVspoof challenge has been designed to help break this mould and to support, for the first time, independent assessments of vulnerabilities to spoofing and of countermeasure performance. While preventing as much as possible the inappropriate use of prior knowledge, the challenge aims to stimulate the development of generalised countermeasures with potential to detect varying and unforeseen spoofing attacks.

The first evaluation, ASVspoof 2015, is being held within the scope of a special session at INTERSPEECH 2015 and with a focus on spoofing detection. Participants are invited to submit spoofing detection results. You will be provide with a spoofing database along with a protocol for experiments. The spoofing database is generated from



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The Rise of Neural Nets

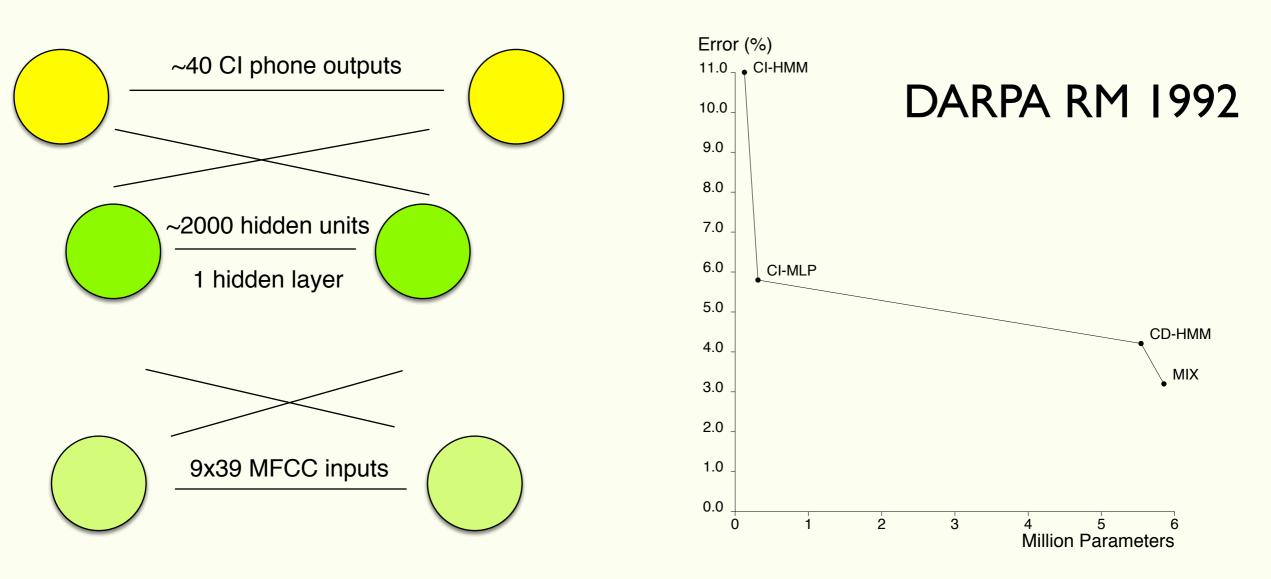


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The Rise and Fall and Rise of Neural Nets

Neural network acoustic models (1990s)



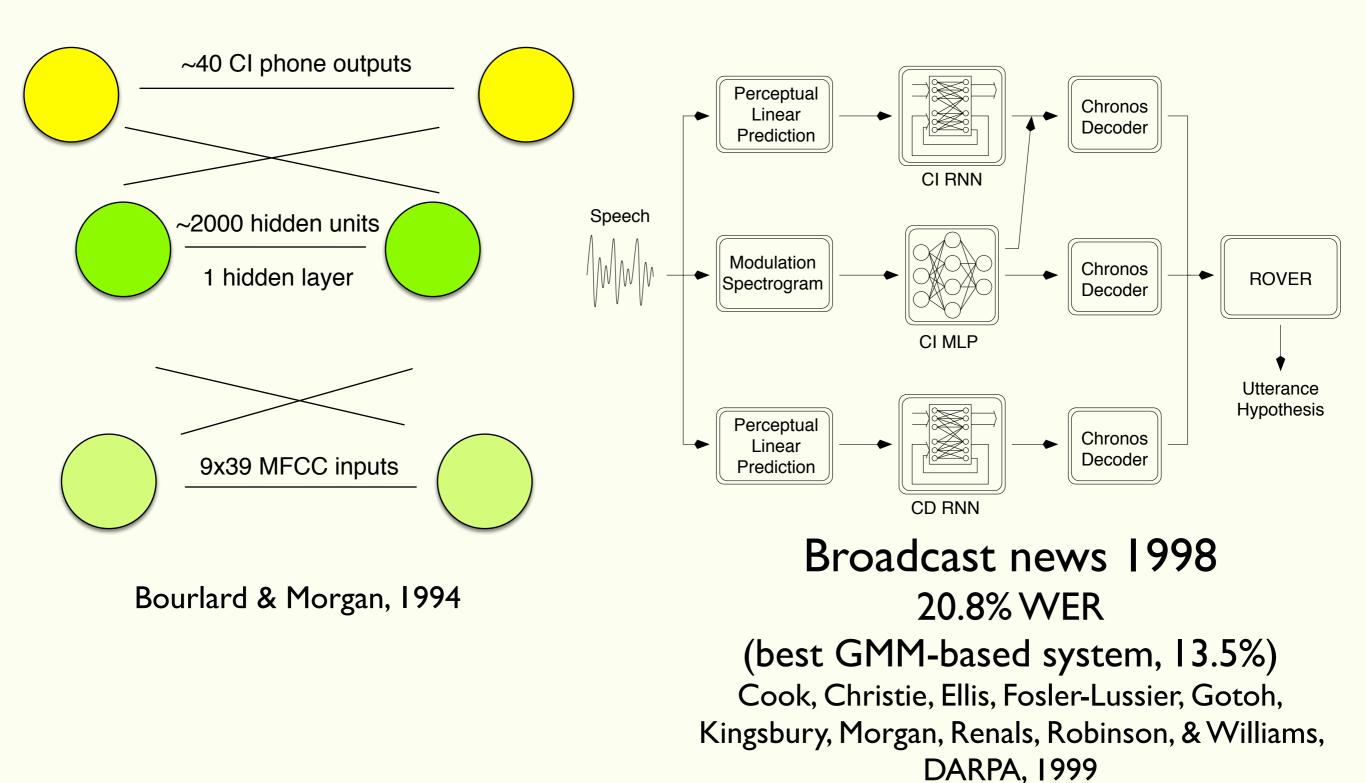


Bourlard & Morgan, 1994

Renals, Morgan, Cohen & Franco, ICASSP 1992

Neural network acoustic models (1990s)





NN acoustic models Limitations vs GMMs



- Computationally restricted to monophone outputs
 - CD-RNN factored over multiple networks limited within-word context
- Training not easily parallelisable
 - experimental turnaround slower
 - systems less complex (fewer parameters)
 - RNN <100k parameters
 - MLP ~IM parameters
- Rapid adaptation hard (cf MLLR)

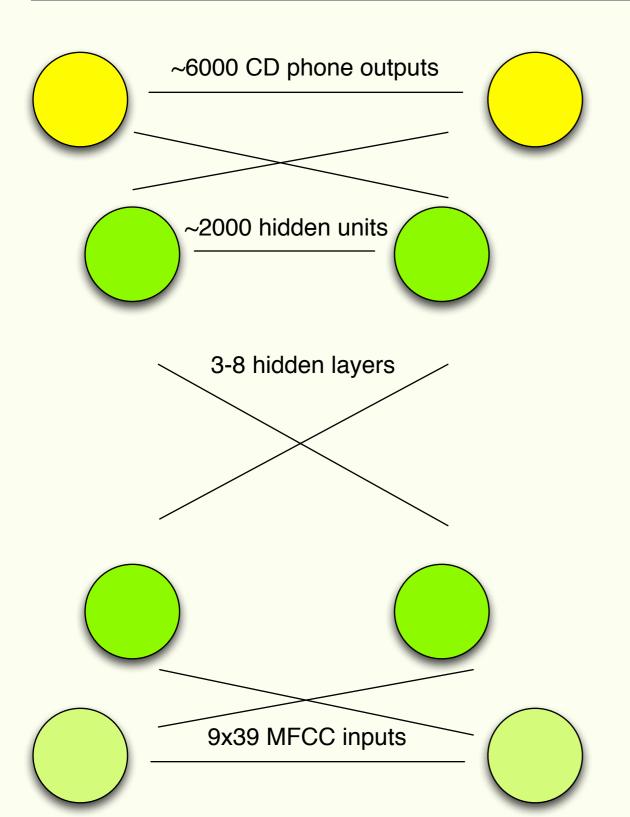
NN acoustic models Benefits



- Fewer limitations on inputs
 - Correlated features
 - Multi-frame windows
- Discriminative training criteria (frame level and sequence level)
- Can be used to generate 'higher-level' features
 - tandem, posteriorgrams
 - bottleneck features



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Dahl, Yu, Deng & Acero, IEEE TASLP 2012 Hinton, Deng, Yu, Dahl, Mohamed, Jaitly, Senior, Vanhoucke, Nguyen, Sainath & Kingsbury, IEEE SP Mag 2012

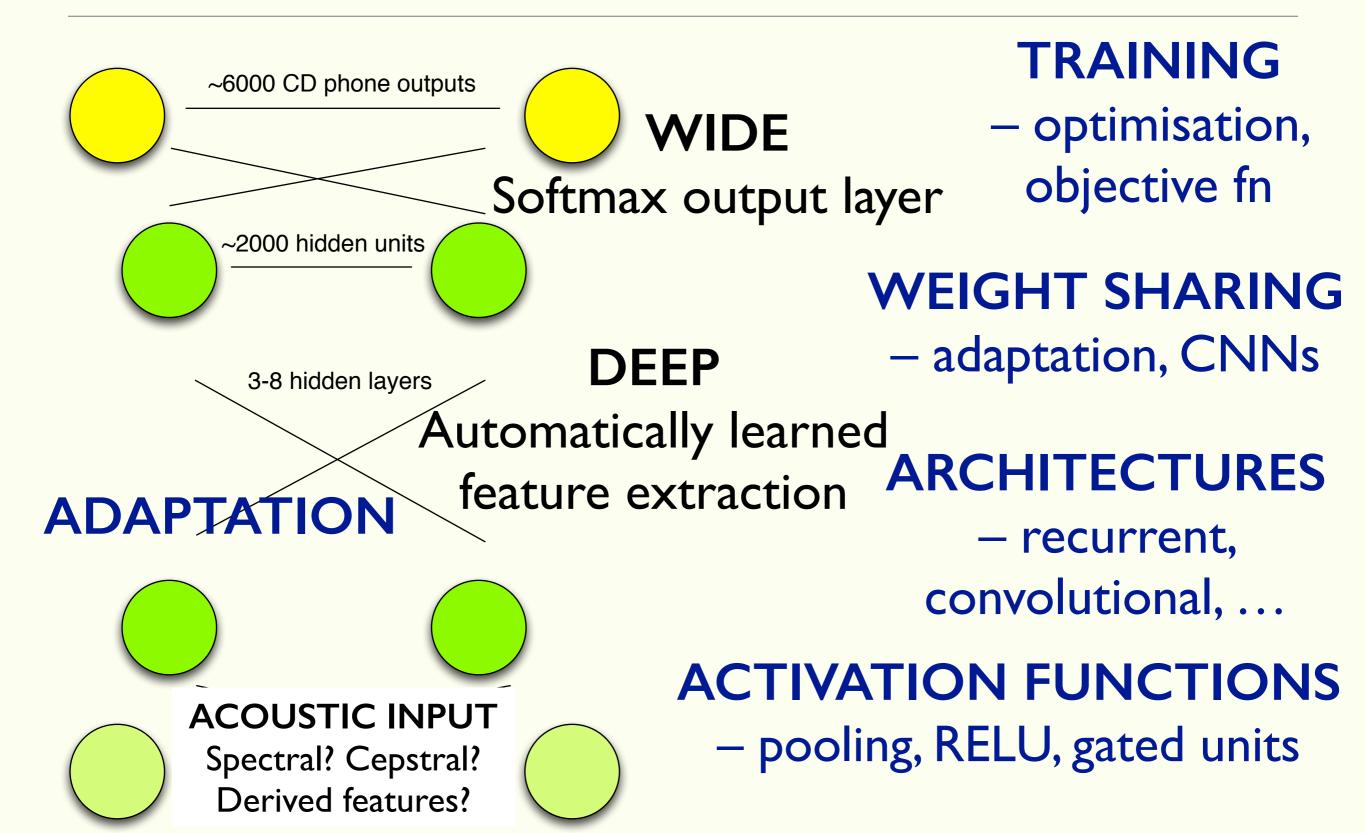


~6000 CD phone outputs WIDE Softmax output layer ~2000 hidden units DEEP 3-8 hidden layers **Automatically learned** feature extraction **ACOUSTIC INPUT**

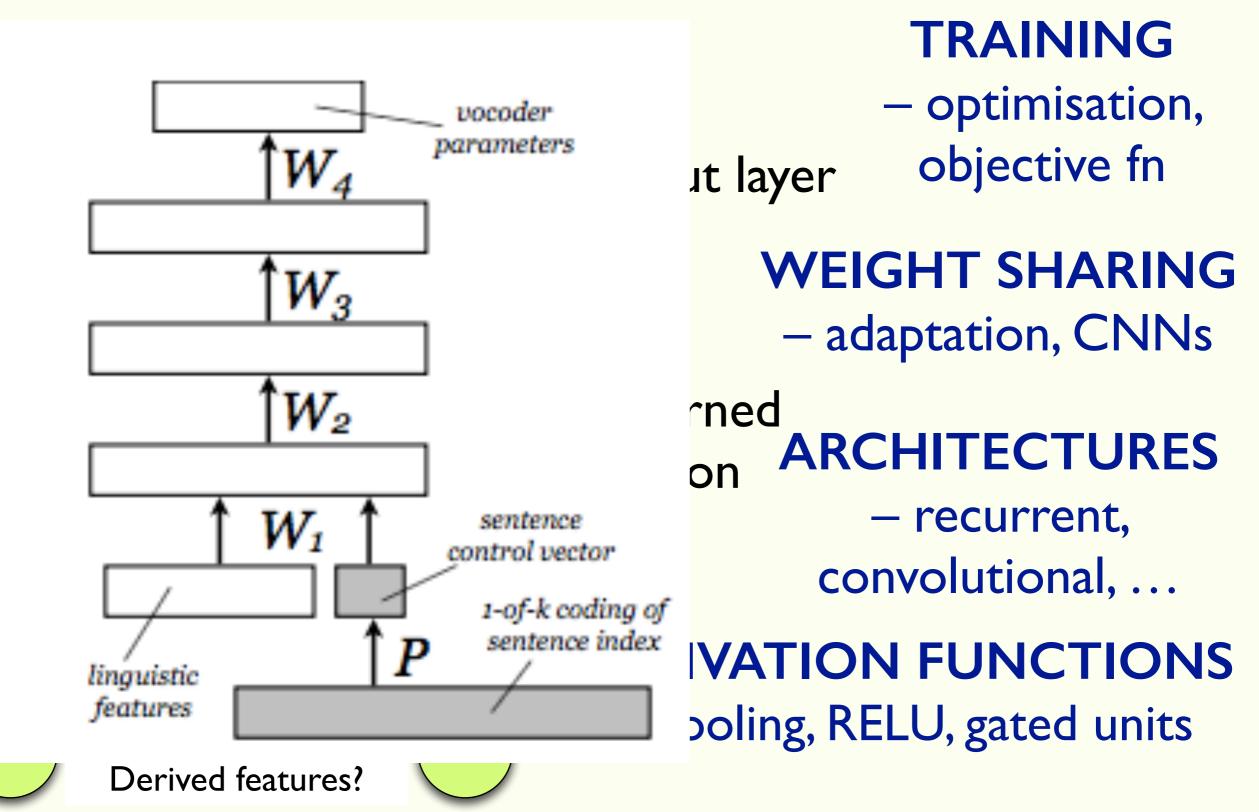
Spectral? Cepstral? Derived features?

Dahl, Yu, Deng & Acero, IEEE TASLP 2012 Hinton, Deng, Yu, Dahl, Mohamed, Jaitly, Senior, Vanhoucke, Nguyen, Sainath & Kingsbury, IEEE SP Mag 2012



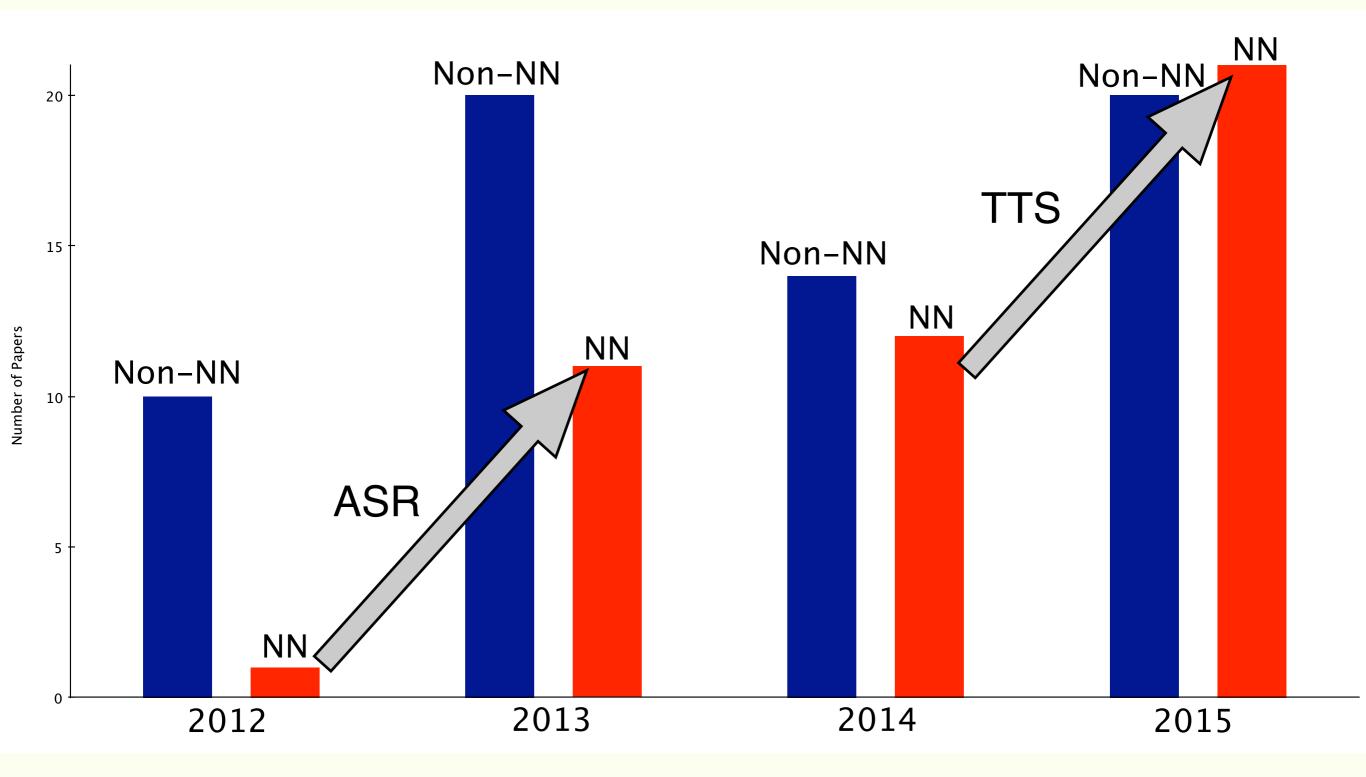






Neural networks & NST





Today's agenda



- 9:30 11:20 Intro, 4 talks, poster spotlights
- 11:20 13:00 Coffee + demos/posters [LR4, ground floor]
- 13:00 14:15 Lunch
- 14:15 15:15 3 talks
- 15:15 15:45 Coffee
- 15:45 16:45 Discussion: Clinical, Media, Future Challenges
- 16:45 17:00 Wrap-up
- 17:00 18:30 Advisory board meeting
- 19:00 Dinner at Emmanuel College

