UT-Scope: Towards LVCSR Under Lombard Effect Induced by Varying Types and Levels of Noisy Background

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- Modified RASTA Filter for ASR
- QCN_RASTA Normalization
- LVCSR Evaluation
- Conclusions

Introduction

What is Lombard Effect?

- Communication in noisy environments → speakers adjust their speech production in effort to maintain intelligible communication (= Lombard effect, LE)
- LE is represented by increase of vocal effort, increase of pitch, shifts of low formants, formant bandwidth reduction, spectral slope flattening, ...
- ASR acoustic models trained typically on neutral speech → ASR deterioration in LE (mismatch between acoustic models and LE speech parameters)

Objective

- Previous ASR studies mostly focused on LE in small vocabulary tasks
  → Focus on LE in large vocabulary continuous speech data
- Analysis of LE speech production in UT-Scope database
- Proposal of temporal filtering strategy derived from RASTA
- Evaluation of state-of-the-art front-end compensations in LVCSR under LE

UT-Scope Database

- UT-Scope: Speech produced under cognitive and physical stress, emotions, and LE
- Lombard portion: 58 subjects (31 native speakers of US English – 25 F, 6 M)
- Neutral (clean) and simulated noisy conditions
- Noisy conditions: background noise samples produced through open-air headphones
- Three types of noise – car (65 mph on highway, windows half open), large crowd, pink
- Noises produced to subjects at 70, 80, and 90 dB SPL (car, crowd) and 65, 75, 85 dB SPL (pink)
- Recording in ASHA certified sound booth
- 3 microphone channels – throat mic, close-talk, and far-field mic

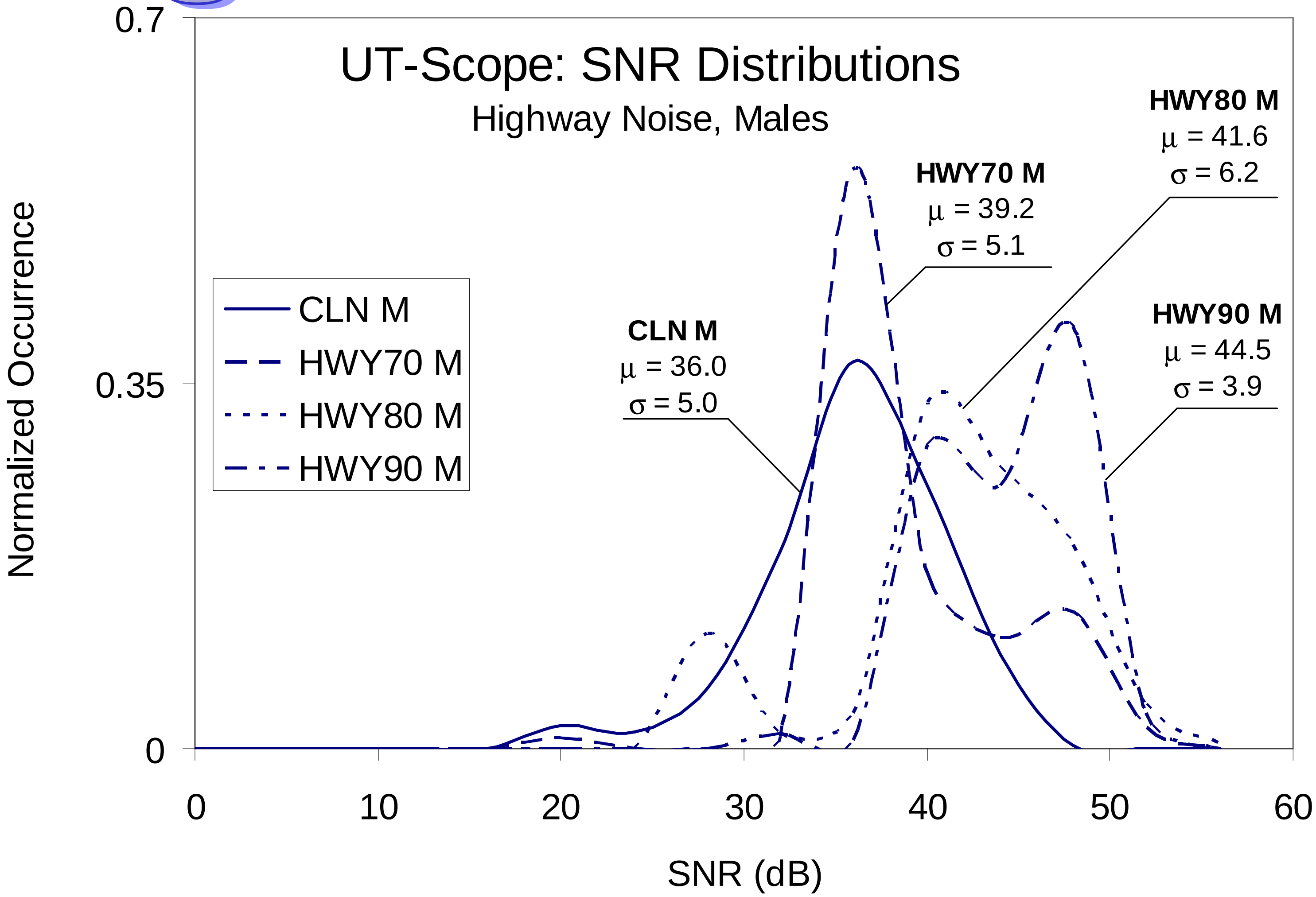
Content

- 100 phonetically balanced read sentences from TIMIT – neutral (clean) conditions
- 20 TIMIT sentences read per each of 9 noise type/level conditions
- Digit strings – 5 repetitions of 10-digit strings per each condition
- Spontaneous speech: ~1 minute per condition – describing content of a picture

Speech Production Under Lombard Effect

- Focus on TIMIT-like sentences recorded by close-talk channel
- Parameters analyzed:
  - Signal-to-noise ratio (SNR) – related to vocal intensity
  - Mean fundamental frequency (F0)
  - Vowel formant frequencies
  - Vowel durations
  - Cepstral distributions
- Extraction tools:
  - WaveSurfer (F0, formants)
  - Segmental SNR estimation tool (CTU in Prague)
  - HTK – forced alignment (vowel boundaries in formant analysis, vowel durations)
  - CTU Copy – extraction of cepstral features

Signal-to-Noise Ratio



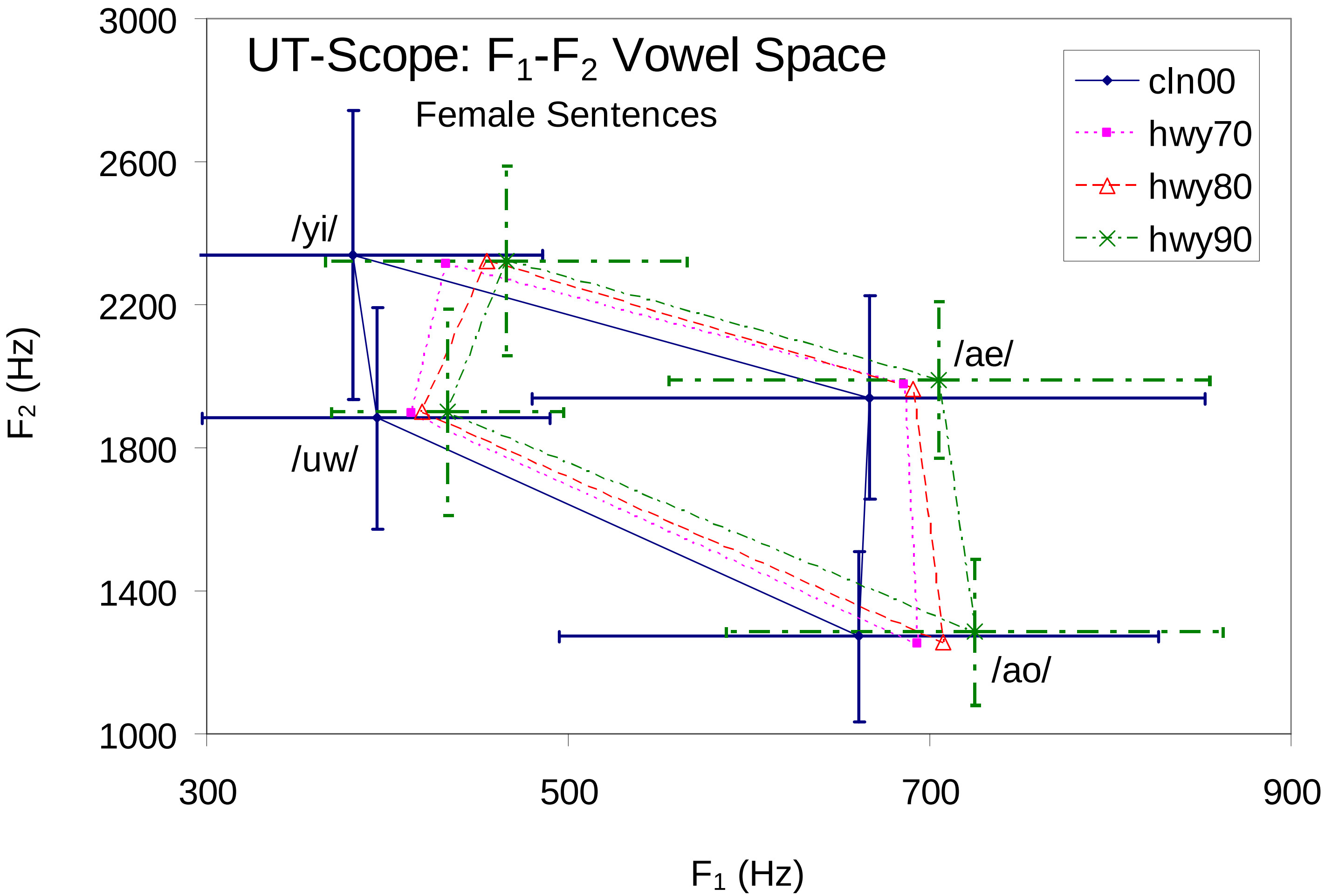
- **Lombard function** (LF) – relation between noise level and speech intensity
- Subjects increase vocal effort with the level of noise; **observed LF slopes** here **0–0.3**

Mean Fundamental Frequency (F0)

	HWY (dB)			CRD (dB)			PNK (dB)		
Gend	70	80	90	70	80	90	65	75	85
F	a=0.938, R^2=0.999 MSE=0.068			a=0.808, R^2=0.998 MSE=0.083			a=0.596, R^2=0.984 MSE=0.380		
M	a=1.195, R^2=1.000 MSE=0.039			a=1.073, R^2=1.000 MSE=0.011			a=0.786, R^2=0.962 MSE=1.634		

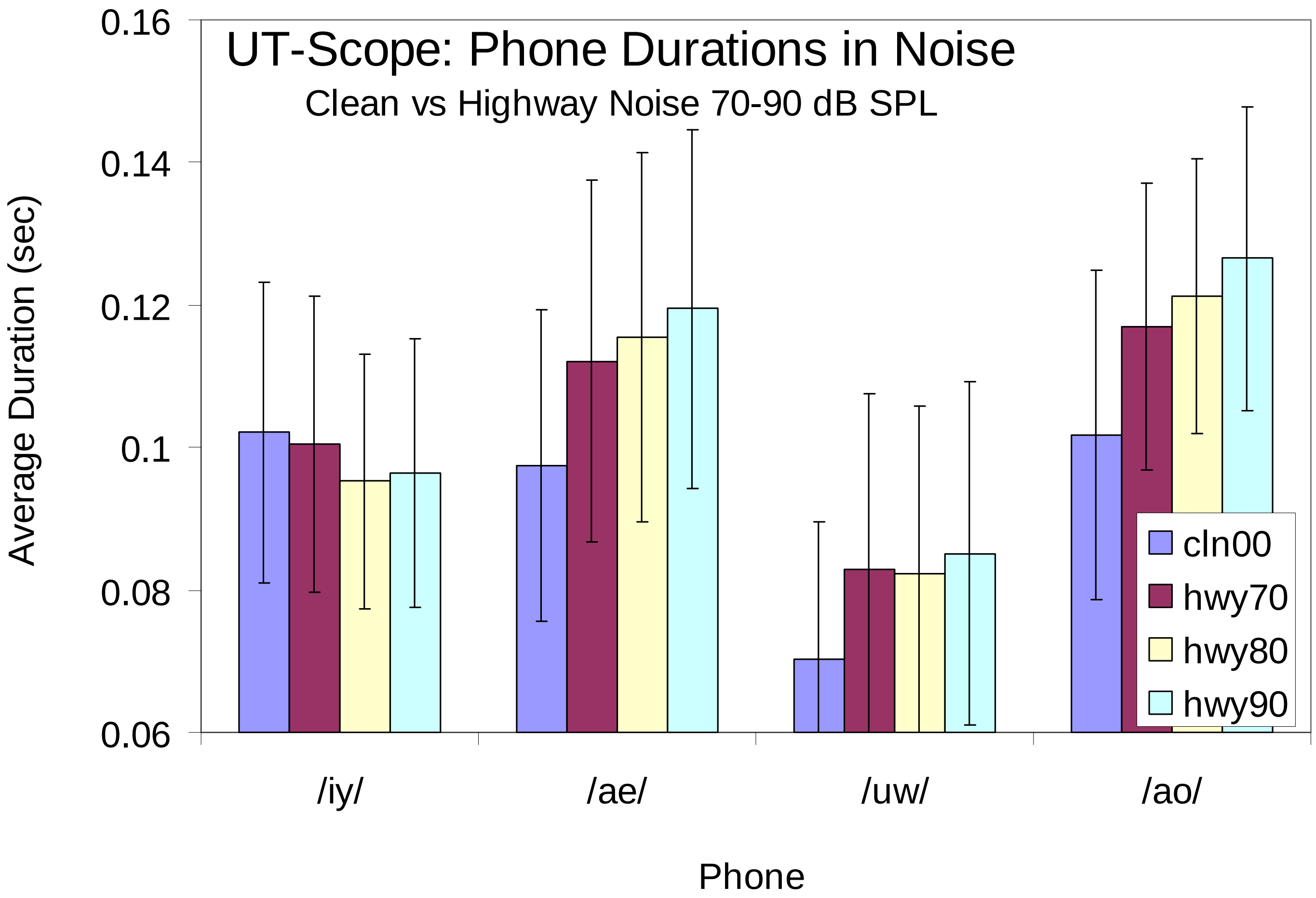
- **Correlation** analysis **between noise presentation level** (in dB) **and mean F0** across all recordings in that noise level
- a – slope of the regression line in the noise level/F0 plane; R^2 – correlation coefficient; MSE – mean square error
- Consistent F0 increase with the level of noise; steepest for car noise
- R^2, MSE – strong correlation between the level of noise and speech intensity (hwy, pnk)

Vowel Formant Frequencies



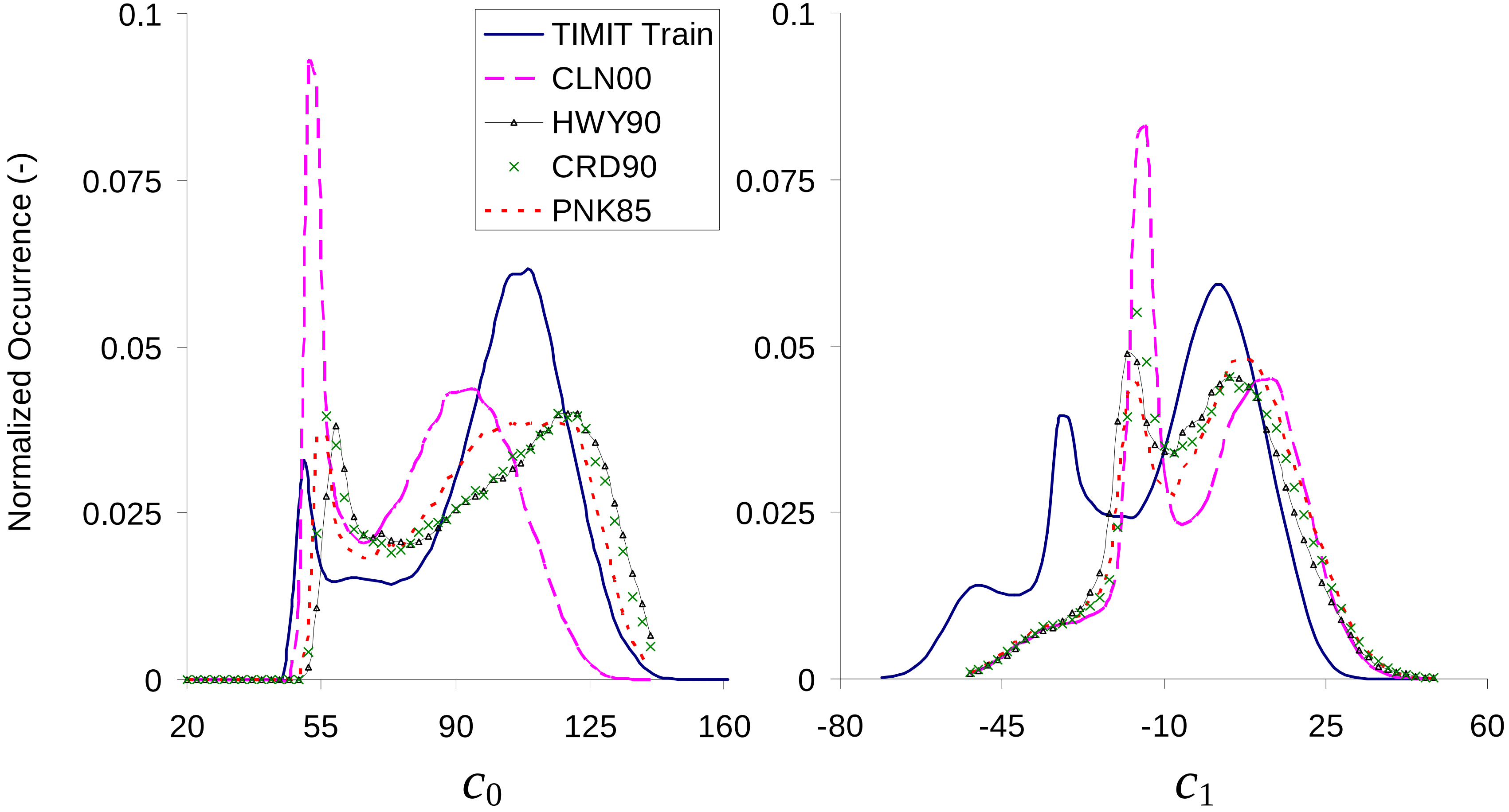
- **Vowel** segment **boundaries** estimated through **forced alignment**
- Systematic shift of vowels in F1-F2 space with increasing noise level

Vowel Durations



- Vowel segments estimated through forced alignment
- **Increasing trend** in some vowel durations, not statistically significant (95% CI's)

Cepstral Distributions



- **Speech production variations in LE** – direct impact on ASR features (here c0, c1 in MFCC) – these plots are for **clean speech signal** (high SNR)
- **Channel differences** – another source of mismatch – compare **TIMIT and CLN00**

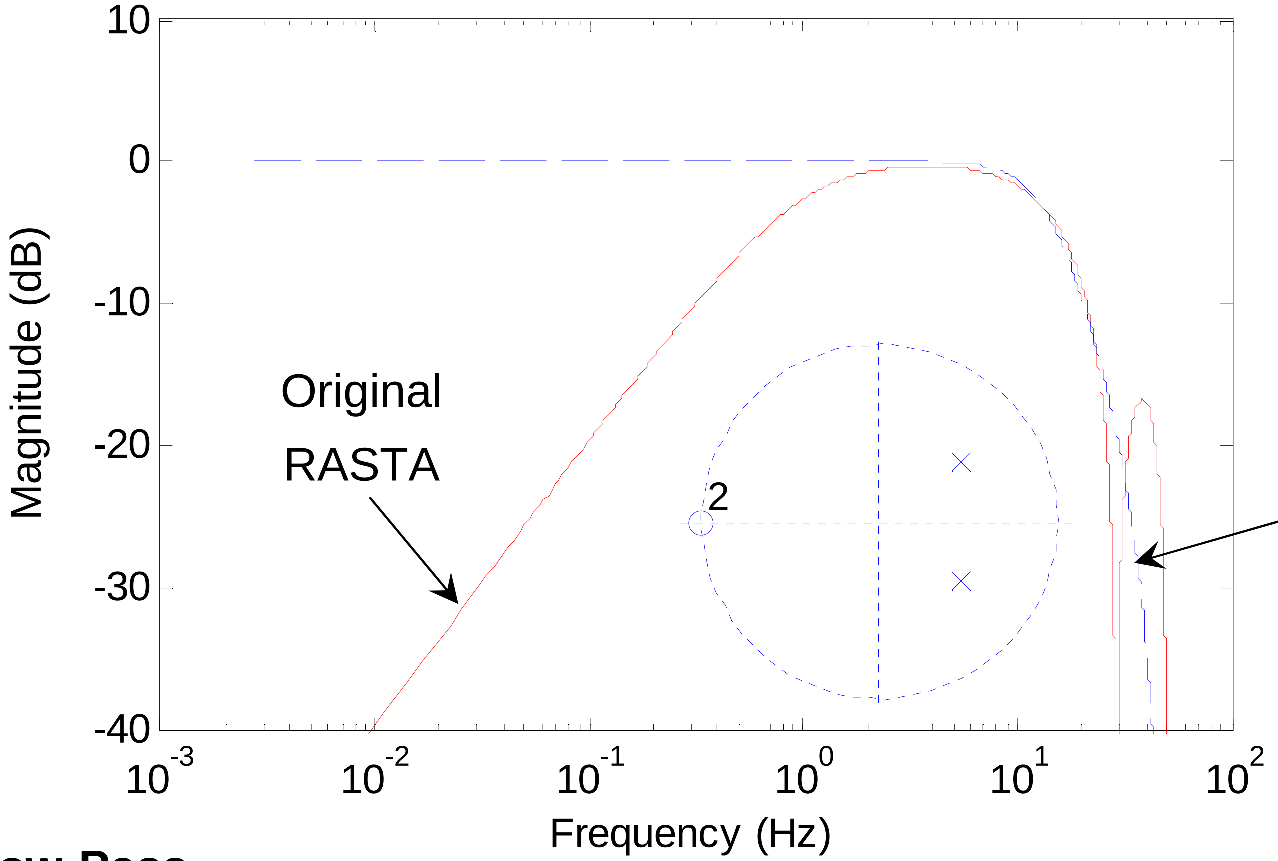
Modified RASTA Filter for ASR

- RASTA– band-pass filtering in log-spectral or cepstral domain; elimination of slow-varying components (including DC) and components varying faster than expected for speech
- RASTA is popular in ASR and speaker ID as it increases robustness to channel variations, reverberation, and noise
- Original RASTA filter - high order IIR band-pass filter – introduces transient distortions in the feature tracks

Proposed Modification

- RASTA can be approximated by a combination of cepstral mean normalization (CMN) and a low-pass filter, i.e., by **distribution normalization** & **temporal filtering**
  → decomposition of RASTA into two blocks
  → low-pass – requires lower order filter → reduced transient effects
  → allows for replacement of first block (CMN) by more powerful normalizations

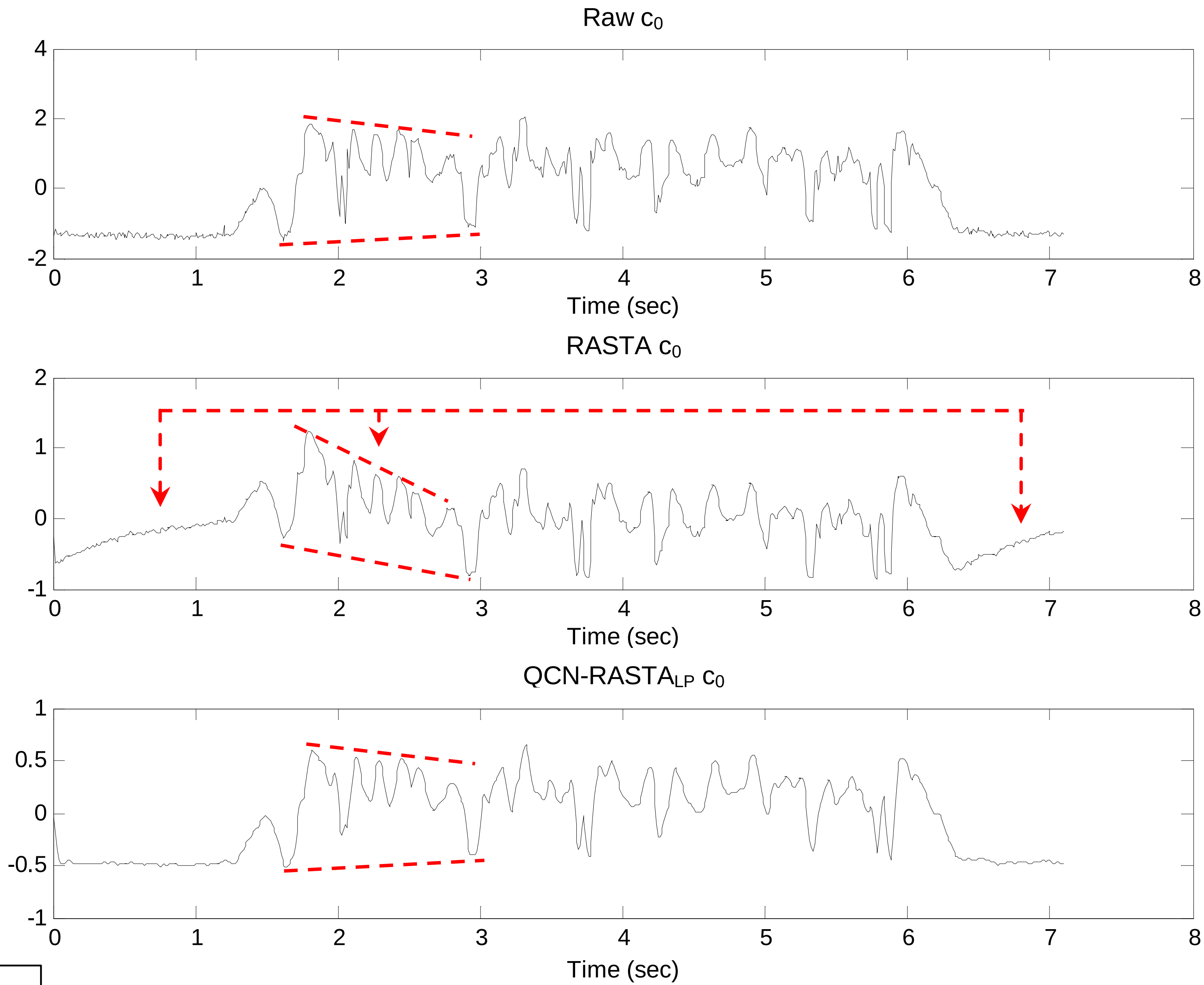
Modified RASTA Filter for ASR



Proposed Low-Pass

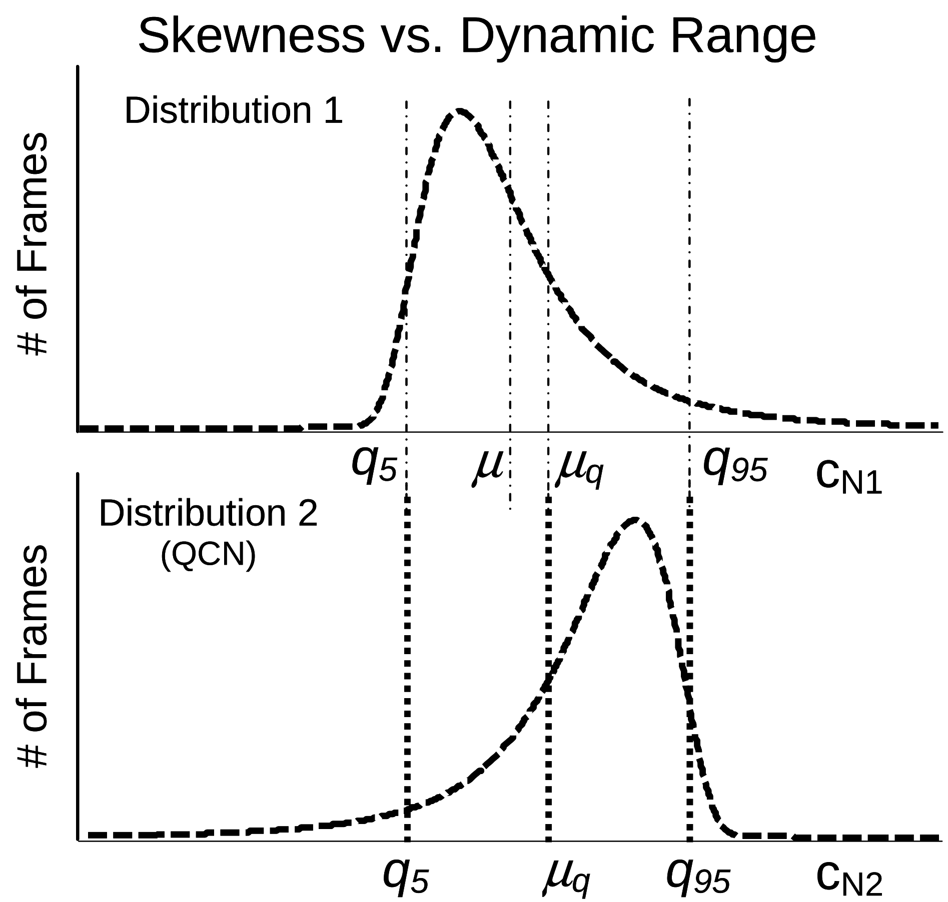
- 2nd order low-pass IIR filter (Butterworth approximation)
- Transfer function is smooth – **eliminates the residual side lobe of original RASTA**

Modified RASTA Filter for ASR



QCN_RASTA Normalization

- QCN – quantile-based cepstral dynamics normalization – introduced at ICASSP'09
- QCN aligns dynamic ranges rather than means of cepstral distributions – found to provide better normalization of distributions with **different skewness due to noise & LE**
- QCN_RASTA – QCN (replacing CMN) + proposed low-pass filter

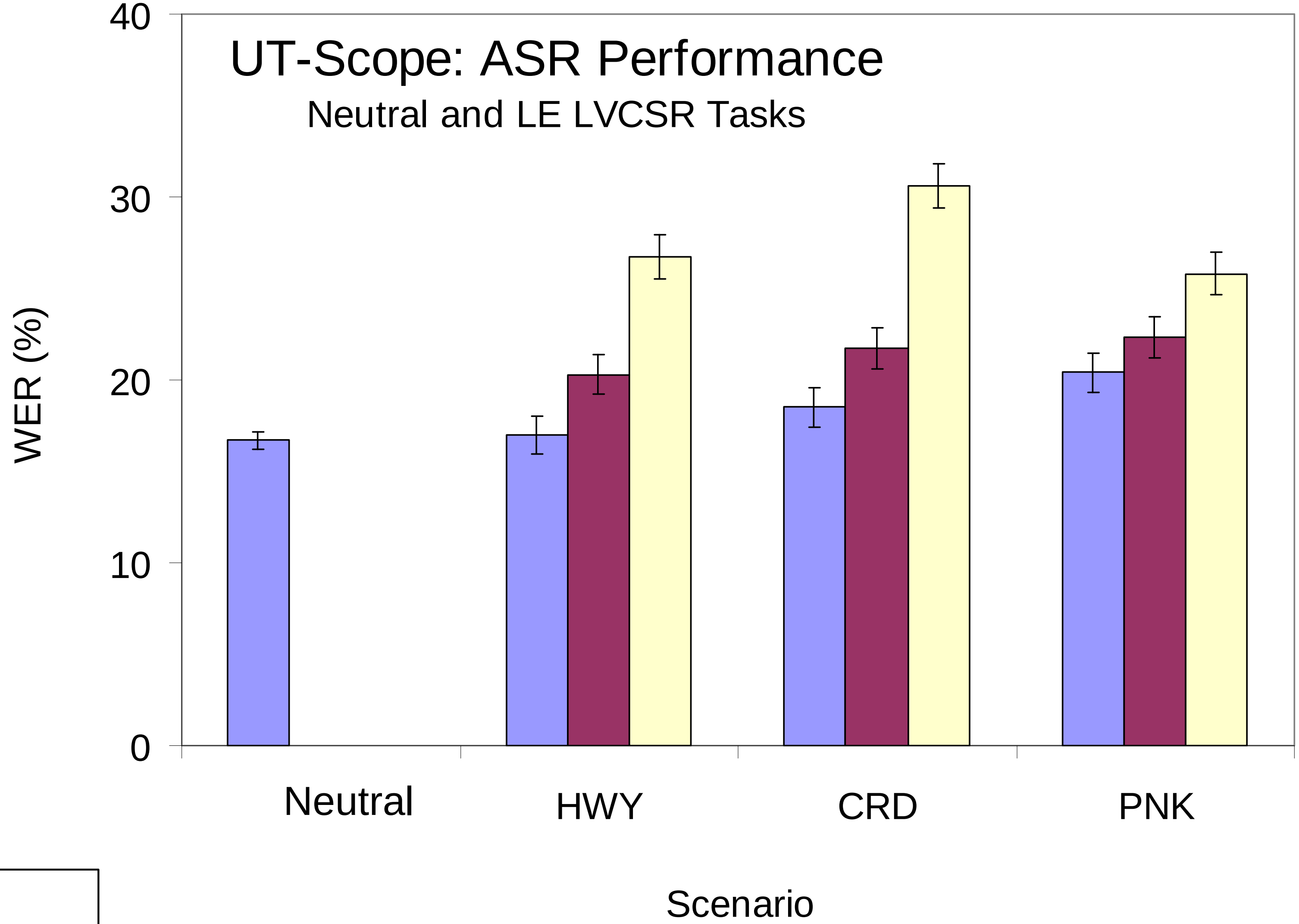


LVCSR Evaluation

- ASR system:
  - acoustic model – triphone HMM's, 32 mixtures (HTK); trained on clean TIMIT
  - language model – SRI LM Toolkit
  - TIMIT acoustic models adapted towards UT-Scope using MLLR and MAP; adapt set - 9 UT-Scope sessions (excluded from open test set)
  - clean test set – 3 male and 9 female subjects, 1 neutral and 9 simulated noisy conditions per subject
  - noisy test set – neutral speech and speech produced in 90 dB of highway noise – both mixed with NOISEX'92 Volvo noise at 5 dB and 15 dB SNR (3 M, 9 F)
- Baseline ASR performance on clean neutral test set
  - MFCC-CVN front-end: 8.3% WER
  - PLP-CVN front-end: 8.9% WER
  - All following results reported for **MFCC**-based systems with **LM off**

LVCSR Evaluation

- Impact of LE on LVCSR: clean recordings (no noise added); MFCC-CVN front-end; no LM
- WER systematically increases with the level of LE



LVCSR Evaluation

- Evaluation of selected cepstral compensation strategies:
  - Cepstral mean normalization (CMN)
  - Cepstral mean-variance normalization (CVN)
  - Cepstral gain normalization (CGN)
  - RASTA filtering in cepstral domain
  - Feature warping (Gaussianization on the utterance level)
  - Histogram equalization (TIMIT training data → reference distributions)
  - Quantile-based cepstral dynamics normalization (QCN); QCN4 – 4% and 96% quantiles bound the dynamic range; QCN9 – utilizes 9% and 91% quantiles
  - QCN_RASTA – QCN + proposed low-pass filter

LVCSR Evaluation

Clean Recordings	
Cepstral Comp.	Across Cond.
none	62.0
RASTA	60.0
warp	55.7
CMN	54.3
QCN4	54.3
HistEq	**53.9**
CVN	**53.3**
CGN	**52.8**
QCN4_RASTA	**52.6**
QCN9	**51.1**

Noisy Recordings	
Cepstral Comp.	Across Cond.
none	77.8
QCN9	69.2
CVN	68.5
QCN4_RASTA	68.4
CGN	67.0
HistEq	64.4

- Proposed QCN_RASTA improves performance of QCN; QCN-normalized features outperform other considered setups on clean neutral and LE recordings
- The ranking of front-ends changes in noisy conditions (recordings mixed with car noise); QCN_RASTA still outperforms 'raw' QCN normalization

Conclusions

- **Analyzed** impact of **LE** on **speech parameters** in UT-Scope database
- A number of speech production **parameters** found to **vary** with the type and level of noise inducing LE
- Strong **linear relationship** between **noise presentation level** (dB) and **mean pitch** (Hz) was observed for large crowd and highway noises
- A **modified** version of **RASTA** filtering scheme was proposed and shown to **reduce transient effects** of original RASTA
- **Combination of QCN** and newly designed **low-pass filter** (**QCN_RASTA**) improved QCN performance in both clean signal and noisy signal conditions (on a mixture of neutral and LE speech)
- A number of **cepstral normalizations were compared** in the task of talking style (neutral/LE) and noisy background mismatch
- CGN, histogram equalization, QCN, and newly proposed QCN_RASTA provided **significant performance gains** in talking style/noise mismatched conditions
- None of the normalizations managed to provide superior performance across all tasks

Thank you!