

Vector Taylor Series Expansion with Auditory Masking for Noise Robust Speech Recognition

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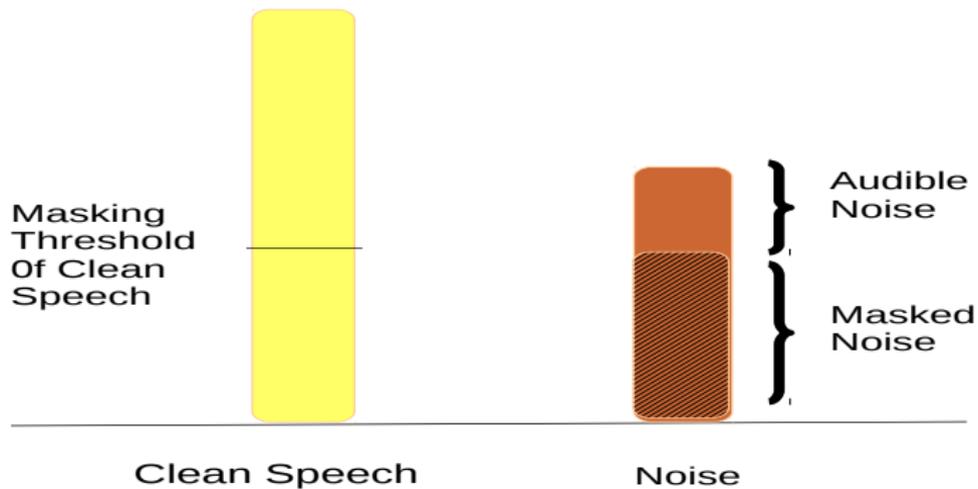
20 Oct, 2016

- Introduction
- Proposed method
- Algorithm
- Experimental Setup
- Experimental Results
- Conclusion

- Existing Methods
 - Vector Taylor Series (VTS) expansion for Gaussian Mixture Model (GMM) and Hidden Markov Model (HMM) (J. Li et al., 2007)
 - Psychoacoustic Compensation (Psy-Comp) technique for GMM-HMM model (B Das and A Panda, 2015).
 - VTS technique for feature enhancement for Deep Neural Network (DNN) (B Li, 2013).

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 - Vector Taylor Series (VTS) expansion for Gaussian Mixture Model (GMM) and Hidden Markov Model (HMM) (J. Li et al., 2007)
 - Psychoacoustic Compensation (Psy-Comp) technique for GMM-HMM model (B Das and A Panda, 2015).
 - VTS technique for feature enhancement for Deep Neural Network (DNN) (B Li, 2013).
- Proposed method
 - Incorporation of auditory masking into the Vector Taylor Series for corrupting clean model parameters.
 - Use of Minimum Mean Square Error (MMSE) to extract clean features.

Speech Masking Noise



- Degraded speech model in spectral domain

$$Y = XH + N, \quad (1)$$

where Y , X , H and N are degraded speech, clean speech, channel factor, and additive noise respectively in spectral domain.

Traditional vector Taylor series

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- VTS Corruption function

$$\vec{y}^s = \vec{x}^s + \vec{h}^s + C \log(1 + \exp(C^{-1}(\vec{n}^s - \vec{x}^s - \vec{h}^s))), \quad (2)$$

where s indicates the static part of all variables. \vec{y} , \vec{x} , \vec{h} and \vec{n} are distorted speech, clean speech, channel factor and additive noise respectively. C and C^{-1} are the discrete cosine transform matrix and its inverse respectively.

Proposed Method

- Proposed degraded speech model in spectral domain

$$Y_f = W_f X_f H_f + N_f \quad (3)$$

where H_f is the channel factor and N_f is the additive noise. The masking factor W_f can be defined as follows:

$$W_f = \frac{X_f - 10^{\frac{T_{mf}}{20}}}{X_f}. \quad (4)$$

T_{mf} is the masking threshold of the clean speech X_f .

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- Masking threshold is calculated as follows:

$$T_{xf} = 20 \log_{10}(\mu_{xf}) - 0.275 \cdot h_f - 6.025 \quad (\text{dB}) \quad (5)$$

where h_f is central frequency of mel-filter in Bark scale.

Proposed Method

- Proposed Corruption function

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- The Jacobian of the mismatch function with respect to clean speech parameter

$$G = C \bullet \text{diag} \left(\frac{1}{1 + \exp(C^{-1}(\vec{\mu}_n - \vec{\mu}_x - \vec{w} - \vec{h}))} \right) \bullet C^{-1}. \quad (7)$$

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- The model mean corruption

$$\vec{\mu}_y = \vec{\mu}_x + \vec{h} + \vec{w} + C \log(1 + \exp(C^{-1}(\vec{\mu}_n - \vec{\mu}_x - \vec{w} - \vec{h}))) \quad (8)$$

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- The model variance corruption

$$\Sigma_y \approx G \Sigma_x G^T + (I - G) \Sigma_n (I - G)^T. \quad (9)$$

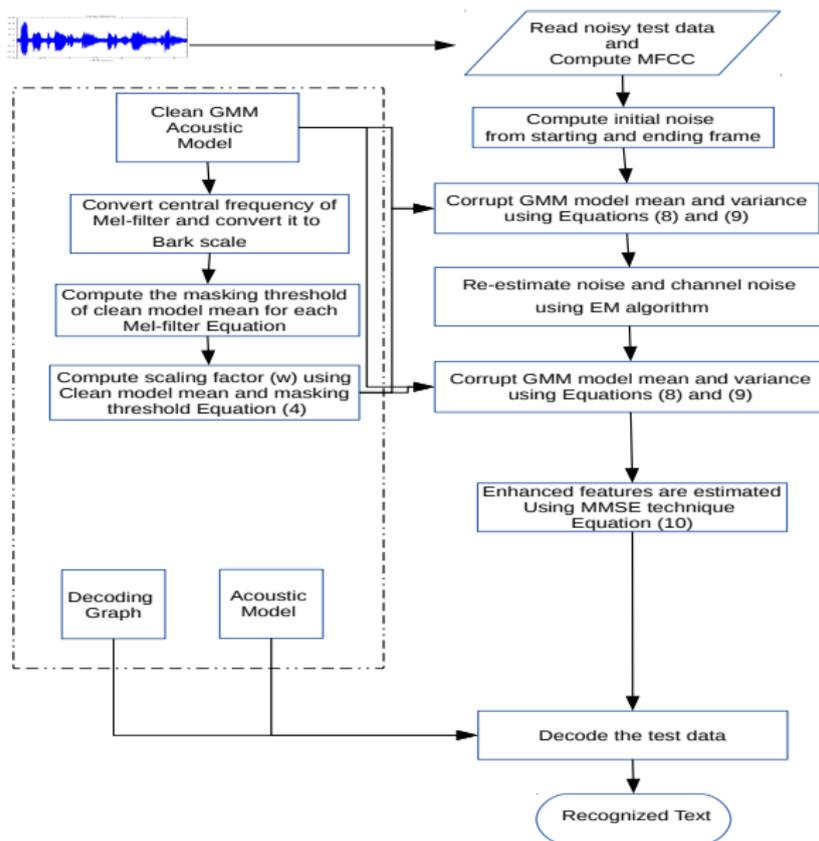
Proposed Method

- We need a GMM model, trained on clean speech utterances
- Estimation of Enhanced Features using MMSE technique

$$\begin{aligned}\vec{x}_{MMSE} &= E(\vec{x}|\vec{o}) = \int \vec{x}p(\vec{x}|\vec{o})dx \\ &= \vec{o} - \sum_{m=0}^{M-1} p(\vec{o}|\lambda_{ym})(\vec{\mu}_{ym} - \vec{\mu}_{xm}),\end{aligned}\tag{10}$$

- \vec{o} : noisy speech features
- $p(\vec{o}|\lambda_{ym})$: posterior probability for the m^{th} Gaussian mixture component of the noise compensated GMM.
- $\vec{\mu}_{ym}$: m^{th} component of the noise compensated model mean.
- $\vec{\mu}_{xm}$: m^{th} component of the clean model mean.

Algorithm for Model Compensation

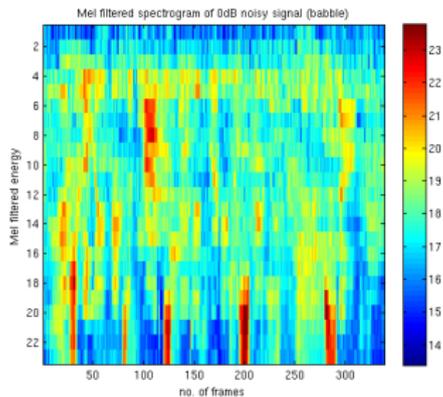
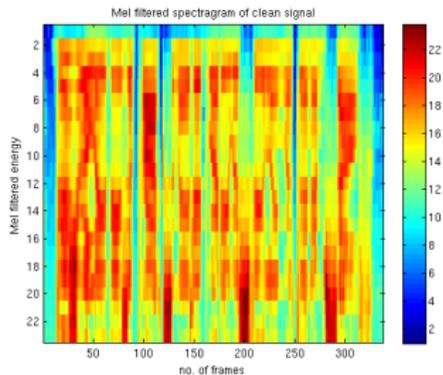


Experimental Setup

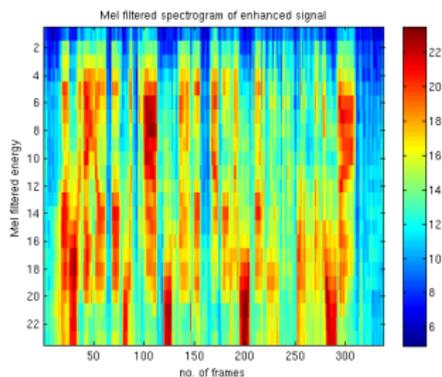
- Speech Corpus: TIMIT
- ASR Toolkit: Kaldi ASR toolkit
- Feature: Mel Filter Cepstral Coefficient (MFCC).
- Model Description (DNN-HMM):
 - Number of hidden layer : 2
- Adding noise: Used Filtering and Noise Adding Tool (FaNT)
- Clean train data and noise corrupted test data.
- Noise Type: Babble, Hfchannel, F-16 from NOISEX-92 database and Street noise (We collected)
- SNR Level: 0dB, 5dB, 10dB and 15dB

- Different acoustic model:
 - ① TRI1: It has been obtained after standard GMM-HMM approach. It is a triphone model.
 - ② TRI2: Applied Linear Discriminative Analysis (LDA) and Maximum Likelihood Linear Transform (MLLT) at the time of training.
 - ③ TRI3: Speaker Adaptive Training (SAT) along with LDA and MLLT for speaker dependent acoustic model.
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 - ③ TRI3: Speaker Adaptive Training (SAT) along with LDA and MLLT for speaker dependent acoustic model.
 - ④ DNN: DNN architecture is used instead of GMM for acoustic modeling.
- Different features enhancement techniques:
 - ① Baseline: No enhancement technique.
 - ② VTS: We have enhanced feature using traditional VTS method.
 - ③ Proposed method : We have introduced masking effect into VTS method to enhance features.



- ① Spectrogram of clean signal
- ② Spectrogram of 0dB noisy signal (Babble) .
- ③ Enhanced spectrogram with proposed method.



Experimental Results

Table: Phoneme error rate for various methods for “hfchannel” with different noise level

	HFCHANNEL												Average		
	0DB			5DB			10DB			15DB			Baseline	VTS	Proposed
	Baseline	VTS	Proposed	Baseline	VTS	Proposed	Baseline	VTS	Proposed	Baseline	VTS	Proposed			
TRI1	81.5	66.1	63.7	72.3	61.6	53.9	62.9	45.9	44.5	50.3	41.3	37.3	66.75	53.73	49.85
TRI2	83.8	65.2	64.3	72.9	61.5	53.8	60.5	45.5	43.3	46.7	41.5	36	65.98	53.43	49.35
TRI3	81.6	64.6	63.6	72.6	61.3	54.1	60.1	45.3	44.5	46.4	39.4	36.4	65.18	52.65	49.65
DNNs	79.6	60.5	59.3	64.2	56	49.8	47.6	41.9	40.8	36.3	37.3	33.1	56.93	48.93	45.75

Table: Phoneme error rate for various methods for “f-16” with different noise level

	F-16												Average		
	0DB			5DB			10DB			15DB			Baseline	VTS	Proposed
	Baseline	VTS	Proposed												
TRI1	87.8	72.3	69.2	78.3	61.6	58.6	66.9	50.8	47.9	53.5	41.3	38.9	71.63	56.5	53.65
TRI2	89.2	71	68.7	82.4	61.5	58.9	70.5	50.9	49.6	55.6	41.5	38.9	74.43	56.23	54.03
TRI3	90.5	71.2	68.2	81	61.3	59.5	70	51.6	49.6	55.1	42	39.6	74.15	56.53	54.23
DNNs	89.5	67.4	64.7	78.2	56	54.3	58.5	46.9	44.2	42.1	37.3	35.3	67.07	51.9	49.62

Experimental Results ..

Table: Phoneme error rate for various methods for “babble” with different noise level

	BABBLE												Average		
	0DB			5DB			10DB			15DB			Baseline	VTS	Proposed
	Baseline	VTS	Proposed												
TRI1	79.5	72.3	69.8	69.5	58.3	57	57.2	47.6	46.6	46.1	40	38.4	63.08	54.55	52.95
TRI2	82.6	72.8	70.2	75	59.4	58	62	48.6	47.8	48.2	41.3	39.8	66.95	55.53	53.95
TRI3	81.7	72.9	70.3	73.2	60.3	59.1	61.2	49.7	48.4	47.8	41.6	40.5	65.98	56.13	54.58
DNNs	82.3	71.6	68.7	68.3	57.4	55.8	52.5	46.2	44.4	40.4	38	36.6	60.88	53.3	51.38

Table: Phoneme error rate for various methods for “street” noise with different noise level

	STREET												Average		
	0DB			5DB			10DB			15DB			Baseline	VTS	Proposed
	Baseline	VTS	Proposed												
TRI1	65.9	52.5	50.7	56.5	45.6	44.7	48.1	39.7	38.6	39.8	34.2	34.6	52.58	43	42.15
TRI2	64	51.3	49.7	54.3	44	43.2	45.1	38.2	37.8	38	33.2	33.2	50.35	41.68	40.98
TRI3	63.2	50.7	49.7	54.4	43.9	43.5	45.6	38.5	37.9	38.3	33.6	33.5	50.38	41.67	41.15
DNNs	59	48.4	47.4	48.1	41.4	40.8	41.5	36.2	35.4	34.7	32.2	31.9	45.83	39.55	38.87

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- The proposed algorithms provide significant performance gain over the traditional VTS technique with little additional computational cost.
- We are currently exploring methods to improve the MMSE estimation by introducing the clean model and compensated model variances into the estimation equation.

THANK YOU