# Cepstrum-Domain Model Combination Based on Decomposition of Speech and Noise Using MMSE-LSA for ASR in Noisy Environments

Author: Hong Kook Kim and Richard C. Rose

Professor: 陳嘉平

Reporter: 吳國豪

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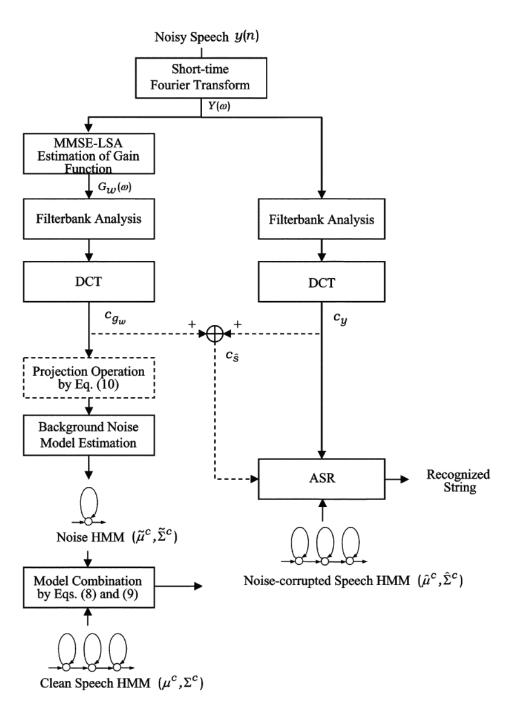
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Abstract—This paper presents an efficient method for combining models of speech and noise for robust speech recognition applications in noisy environments. This method decomposes the cepstrum domain representation of noise-corrupted speech into clean speech cepstrum and background noise cepstrum components using a minimum mean squared error-log spectral amplitude (MMSE-LSA) criterion. Speech recognition is then performed on noisy cepstrum domain observations using a model that is formed by parallel combination of cepstrum domain clean speech distributions and background noise distributions estimated using this MMSE-LSA based noise decomposition. This method is far more efficient than other parallel model combination (PMC) procedures because model combination is performed directly in the cepstrum domain rather than in the linear spectral domain. Whereas background noise model estimation is addressed as a separate issue in existing PMC procedures, this method explicitly incorporates a mechanism to continually update background noise models and signal-to-noise ratio (SNR) estimates over time. The performance of the proposed cepstrum-domain model combination method is compared with a well known implementation of PMC which uses a log-normal approximation when combining speech and background noise model means and variances on a connected digit string recognition task which is subjected to mismatched channel and environment conditions. As a result, it is shown that the proposed model combination technique gives a word error rate that is comparable to PMC when background noise information and SNR are known prior to estimation. The paper will also present the results of experiments where a combination of cepstrum-domain feature compensation and model combination are applied to this task.

# Introduction

- We focus on the category of acoustic model combination which involves combining acoustic models of speech and noise and performing recognition on the noisy acoustic features. This class of approaches is collectively referred to as parallel model combination (PMC).
- Additive model combination (AMC) is introduced as a special case of PMC where speech and noise are decoupled through MMSE-LSA estimation.



• Noisy speech y(n) is represented in the time domain as

$$y(n) = s(n) + w(n)$$

- where s(n) and w(n) are clean speech and additive background noise, respectively
- Using a form of MMSE-LSA estimation, a gain function  $G_{w}(w)$  can be estimated in the linear frequency domain.

$$\hat{S}(w) = G_w(w)Y(w)$$

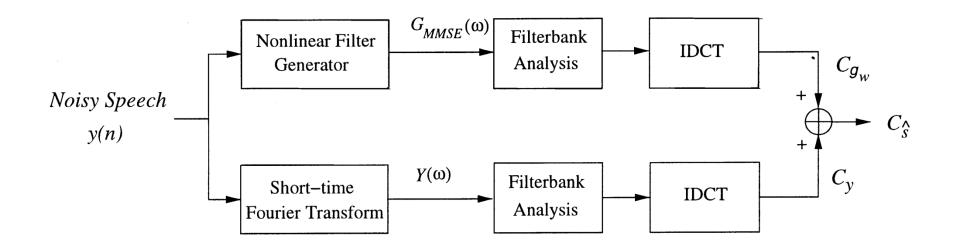
- where Y(w) = S(w) + W(w) with speech spectrum S(w) and background noise spectrum W(w).

• The cepstrum of the gain function  $c_{gw}$  represents the background noise term corresponding to the difference between the noise corrupted speech cepstrum  $c_y$  and clean speech cepstrum  $c_{\hat{s}}$ 

$$c_{\hat{s}} = c_y + c_{gw}$$

• The cepstrum subtraction method (CSM), obtained an estimate of the ith component of the clean speech cepstrum vector at frame  $\tau$  according to the relationship given in

$$c_{\hat{s},i}(\tau) = c_{y,i}(\tau) + c_{gw,i}(\tau)$$



- Let  $S(w) = A(w)e^{j\rho(w)}$ , W(w) and  $Y(w) = R(w)e^{j\vartheta(w)}$  be the Fourier expansions of clean speech s(n), additive noise w(n), and noisy speech y(n), respectively.
- The objective of MMSE-LSA is to find the estimator  $\hat{A}(w)$  that minimizes the distortion measure  $E\{(\log A(w) \log \hat{A}(w))^2\}$  for the given noisy observation spectrum Y(w).

This gain function is decomposed into two components

$$\hat{A}(w) = G_w(w)R(w) = G_M(w)G_{LSA}(w)R(w)$$

- where  $G_{LSA}(w)$  is the gain function associated with the optimal MMSE-LSA estimator and  $G_M(w)$  is a gain modification function.
- The gain  $G_{LSA}(w)$  function is  $G_{LSA}(\omega) = \frac{\xi(\omega)}{1 + \xi(\omega)} \exp(\frac{1}{2} \int_{v(\omega)}^{\infty} \frac{e^{-t}}{t} dt)$
- The gain modification  $G_M(\omega)$  function is  $G_M(\omega) = \frac{\Lambda(\omega)}{1 + \Lambda(\omega)}$

The gain function:

$$G_{LSA}(\omega) = \frac{\xi(\omega)}{1 + \xi(\omega)} \exp(\frac{1}{2} \int_{v(\omega)}^{\infty} \frac{e^{-t}}{t} dt)$$

where

$$v(\omega) = \frac{\xi(\omega)}{1 + \xi(\omega)} \gamma(\omega), \ \gamma(\omega) = \frac{R^{2}(\omega)}{\lambda_{\omega}(\omega)}, \ \xi(\omega) = \frac{\eta(\omega)}{1 - q(\omega)}, \ \eta(\omega) = \frac{\lambda_{s}(\omega)}{\lambda_{\omega}(\omega)}$$
$$\lambda_{s}(\omega) = E\{|S(\omega)|^{2}\} = E\{A^{2}(\omega)\}$$
$$\lambda_{w}(\omega) = E\{|W(\omega)|^{2}\}$$

• The quantity  $\gamma(\omega)$  is the estimate of the a posteriori SNR and  $\eta(\omega)$  is the estimate of the a priori SNR.

• The gain modification function is applied to take into account the probability of the speech presence.

$$G_M(\omega) = \frac{\Lambda(\omega)}{1 + \Lambda(\omega)}$$

•  $\Lambda(\omega)$  is a likelihood ratio between speech presence and speech absence at frequency  $\omega$  and is defined by

$$\Lambda(\omega) = \left. \frac{1 - q(\omega)}{q(\omega)} \frac{\exp(v(\omega))}{1 + \xi(\omega)} \right|_{\xi(\omega) = \frac{\eta(\omega)}{1 - q(\omega)}}$$

This a priori probability is defined as

$$q(w) = \frac{P(H_o(w))}{P(H_o(w)) + P(H_1(w))}$$

- where  $P(H_o(w))$  and  $P(H_1(w))$  are the probabilities associated with speech absence and speech presence hypotheses,  $H_o$  and  $H_1$ , respectively.
- In the regions of speech absence:

$$q(w) \to 1 \Rightarrow \Lambda(\omega) \to 0 \Rightarrow G_M(w) \to 0$$

In the regions of speech presence:

$$q(w) \rightarrow 0 \Rightarrow \Lambda(\omega) >> 1 \Rightarrow G_M(w) \rightarrow 1$$

 It is assumed that an HMM model has been trained from uncorrupted utterances. Given this HMM model, the goal is to update the model means and variances to obtain parameters that describe the noise corrupted observations.

$$\hat{u}_{i}^{c} = E\{c_{y,i}(\tau)\} = E\{c_{\hat{s},i}(\tau) - c_{gw,i}(\tau)\} = u_{i}^{c} - \tilde{u}_{i}^{c}$$

$$\begin{split} \hat{\Sigma}_{ij}^{c} &= E\{(c_{y,i}(\tau) - \hat{u}_{i}^{c})(c_{y,j}(\tau) - \hat{u}_{j}^{c})^{T}\} \\ &= \sum_{ij}^{c} + \tilde{\Sigma}_{ij}^{c} - C_{ij} - C_{ji} \end{split}$$

$$u_{i}^{c} = E\{c_{\hat{s},i}(\tau)\}, \ \tilde{u}_{i}^{c} = E\{c_{gw,i}(\tau)\}\$$

$$\sum_{ij}^{c} = E\{(c_{\hat{s},i}(\tau) - u_{i}^{c})(c_{\hat{s},i}(\tau) - u_{j}^{c})^{T}\}\$$

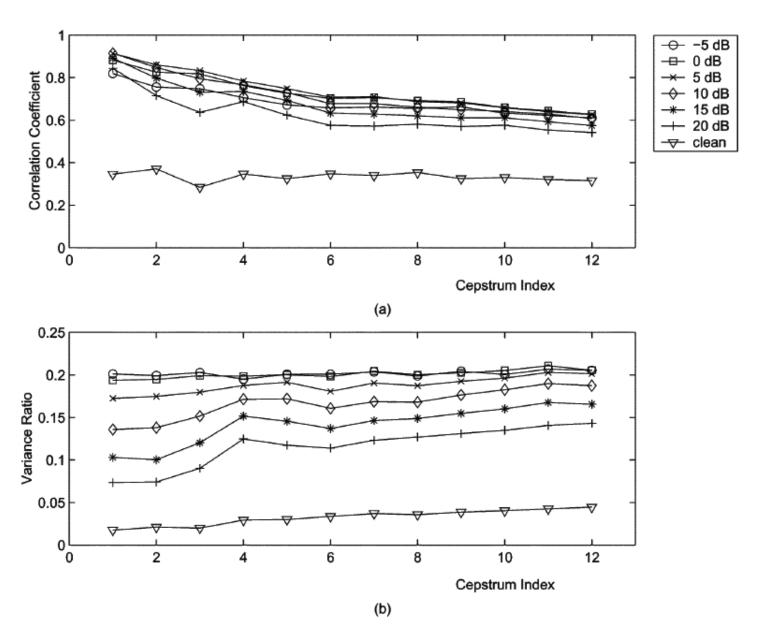
$$\tilde{\Sigma}_{ij}^{c} = E\{(c_{gw,i}(\tau) - \tilde{u}_{i}^{c})(c_{gw,i}(\tau) - \tilde{u}_{j}^{c})^{T}\}\$$

• In addition,  $C_{ij}$  and  $C_{ji}$  are the covariance matrices between the estimates of background noise cepstrum and clean speech cepstrum.

$$C_{ij} = E\{(c_{\hat{s},i}(\tau)c_{gw,j}(\tau)^T\} - u_i^c \tilde{u}_j^c$$

$$C_{ji} = E\{(c_{\hat{s},j}(\tau)c_{gw,i}(\tau)^T\} - u_j^c \tilde{u}_i^c$$

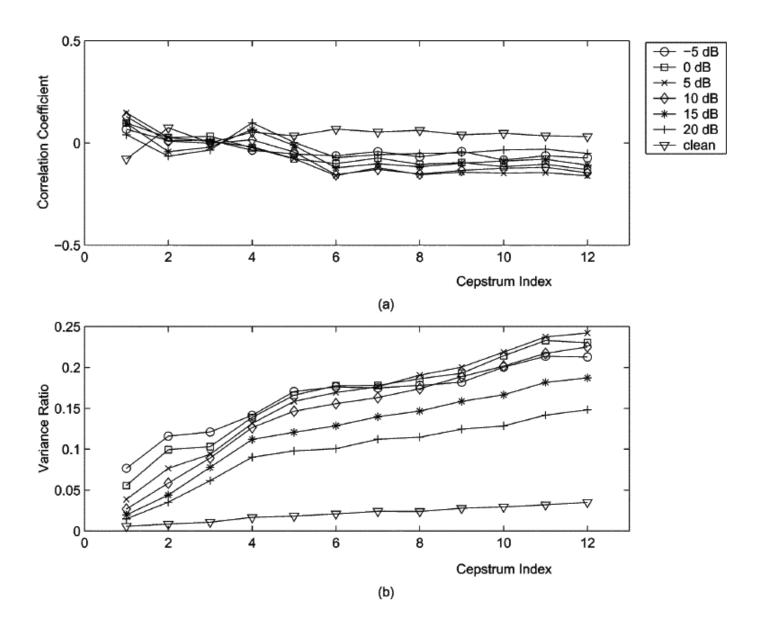
- A simple experiment was performed to demonstrate that the noise cepstrum and the estimate of the clean speech cepstrum are highly correlated.
- Except for the clean condition, the correlation between estimated noise cepstrum and clean speech cepstrum is shown in Fig. 2(a) to be high.
- We computed the average ratio of noise cepstrum variance to clean speech cepstrum variance for each SNR condition and displayed it in Fig. 2(b).
- This suggests that the noise cepstrum variance term can be ignored without having significant impact on the updated covariance  $\hat{\Sigma}_{ii}^c$ .



The projection operation used is

$$\overline{c}_{gw}(\tau) = (c_{gw}(\tau) - \tilde{u}^c) - \frac{\left\langle c_{\hat{s}}(\tau) - u^c, c_{gw}(\tau) - \tilde{u}^c \right\rangle}{\left\langle c_{\hat{s}}(\tau) - u^c, c_{\hat{s}}(\tau) - u^c \right\rangle} (c_{\hat{s}}(\tau) - u^c)$$

- where <x, y> is the inner product of two vectors x and y, and we call  $\overline{c}_{gw}(\tau)$  the projected noise cepstrum.
- We will refer to the approach as projected AMC (PAMC).



# Experiments

- Aurora 2
- we perform ASR experiments in the context of
  - 1) a preprocessing approach that utilizes MMSE-based speech enhancement prior to feature extraction
  - 2) the acoustic feature compensation or cepstrum subtraction method (CSM).
  - 3)we evaluate the performance of AMC by implementing it using either mean-only adaptation or mean-variance adaptation.

	SNR (dB)									
Method	Clean 20 15 10 5 0 -5									
Baseline	0.74	2.85	6.48	16.16	39.45	68.65	83.83	26.72		

	SNR (dB)								WER
Method	Clean	20	15	10	5	0	-5	WER	Reduction (%)
SE	1.00	3.52	6.29	13.55	27.91	53.55	79.83	20.96	21.5
CSM	0.75	2.73	5.09	10.80	23.37	49.42	76.91	18.28	31.6

	Noise HMM	Input MFCCs	Mean	Variance	
Notation	trained with	are	adaptation	adaptation	
AMC(m)	$c_{g_w}$	$c_{m{y}}$	Yes	No	
AMC(m,v)	$c_{g_w}$	$c_y$	Yes	Yes	
CSM+AMC(m)	$c_{g_w}$	$c_{\hat{s}}$	Yes	No	
CSM+AMC(m,v)	$c_{g_w}$	$c_{\hat{s}}$	Yes	Yes	
CSM+PAMC(m)	$ar{c}_{g_w}$	$c_{\hat{s}}$	Yes	No	
CSM+PAMC(m,v)	$ar{c}_{g_w}$	Cŝ	Yes	Yes	

			Avg.	WER					
Method	Clean	20	15	10	5	0	-5	WER	Reduction (%)
AMC(m)	0.89	2.25	4.50	12.93	35.71	65.10	81.40	24.10	9.8
AMC(m,v)	0.72	2.14	4.77	14.68	39.34	67.40	82.13	25.46	4.7
CSM+AMC(m)	0.91	1.96	3.63	8.49	20.17	45.29	75.43	15.90	40.5
CSM+AMC(m,v)	0.74	1.86	3.39	8.05	19.55	45.18	75.81	15.60	41.6
CSM+PAMC(m)	0.93	1.91	3.62	8.51	20.17	45.46	75.59	15.93	40.4
CSM+PAMC(m,v)	0.75	1.85	3.58	8.23	19.90	45.01	75.36	15.71	41.2

			Avg.	WER					
Method	Clean	20	15	10	5	0	-5	WER	Reduction (%)
PMC	0.92	3.65	5.62	11.80	25.78	46.97	74.66	18.76	29.7
AMC(m)	0.89	2.25	4.50	12.93	35.71	65.10	81.40	24.10	9.8
CSM+PAMC(m,v)	0.75	1.85	3.58	8.23	19.90	45.01	75.36	15.71	41.2