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ADAPTIVE DEREVERBERATION METHOD BASED ON COMPLEMENTARY WIENER FILTER AND MODULATION TRANSFER FUNCTION

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INTRODUCTION

Hands-free computer interfaces is rapidly spreading

- Speech quality are degraded by reverberation
- most of the devices have limited computational resources

Low computational dereverberation techniques are required

Previous study [K. Kondo, et al., 2012]

- Proposed a single-channel dereverberation method
- Using a complementary Wiener filter
- Need very little computational resources

Problem of previous study

- Parameters have a large impact on the performance
- Need to selecting the appropriate parameters

In this report

- Estimate the parameters of dereverberation filter
  - Using the modulation transfer function
  - Requiring very little computation

CONVENTIONAL METHOD

Relations between dereverberated speech $Z(k, l)$ and observed speech $X(k, l)$ in frequency domain

$$Z(k,l) = G(k,l)X(k,l)$$

k: frequency bin index
l: frame index

Dereverberation gain : $G(k, l)$

$$G(k,l) = \begin{cases} 1, & \dfrac{P_X(k,l)}{P_X(k)} \geq 1 \\ \dfrac{P_X(k,l)}{P_X(k)}, & \text{otherwise} \end{cases}$$

where

$$P_X(k,l) = |X(k,l)|^2$$
$$P_X(k) = E_l[|X(k,l)|^2] \qquad E_l[\cdot]\text{: expectation for } l$$

Expectation of the power spectrum

- Approximate as an exponential moving average

$$P_X(k) \approx R_2(k,l) = (1-\alpha_2)|X(k,l)|^2 + \alpha_2 R_2(k,l-1)$$
$$\alpha_2 (0 < \alpha_2 < 1)\text{: smoothing constant}$$

Instantaneous speech power

- To avoid the musical noise, using short term moving average

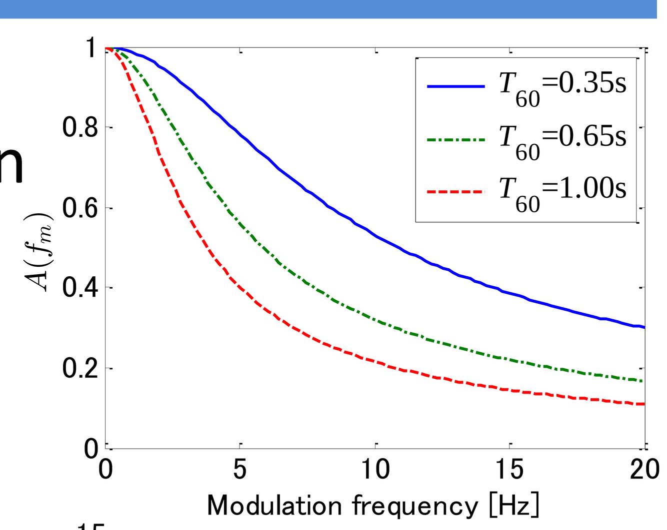
$$P_X(k,l) \approx R_1(k,l) = (1-\alpha_1)|X(k,l)|^2 + \alpha_1 R_1(k,l-1)$$
$$\alpha_1 (0 < \alpha_1 < \alpha_2)$$

PARAMETER ESTIMATION

Modulation Transfer Function (MTF)

- Represent the effect of reverberation
- MTF can be calculated as:

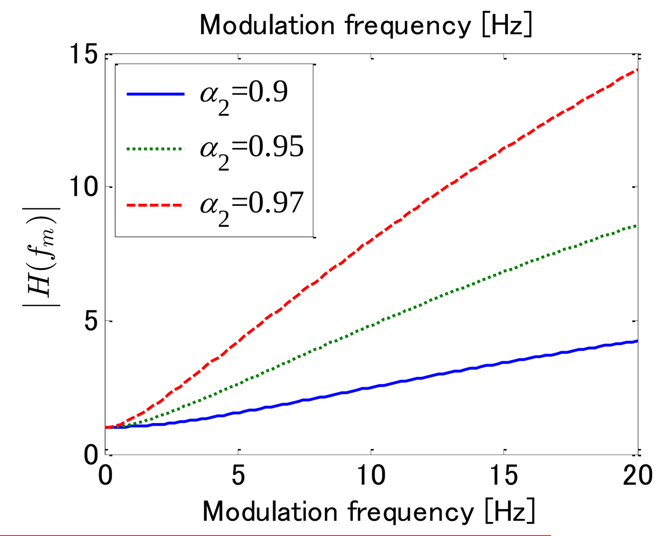
$$A(f_m) = \left\{1 + \left(2\pi f_m \frac{T_{60}}{6 \ln 10}\right)^2\right\}^{-\frac{1}{2}}$$



Amplitude response of the filter

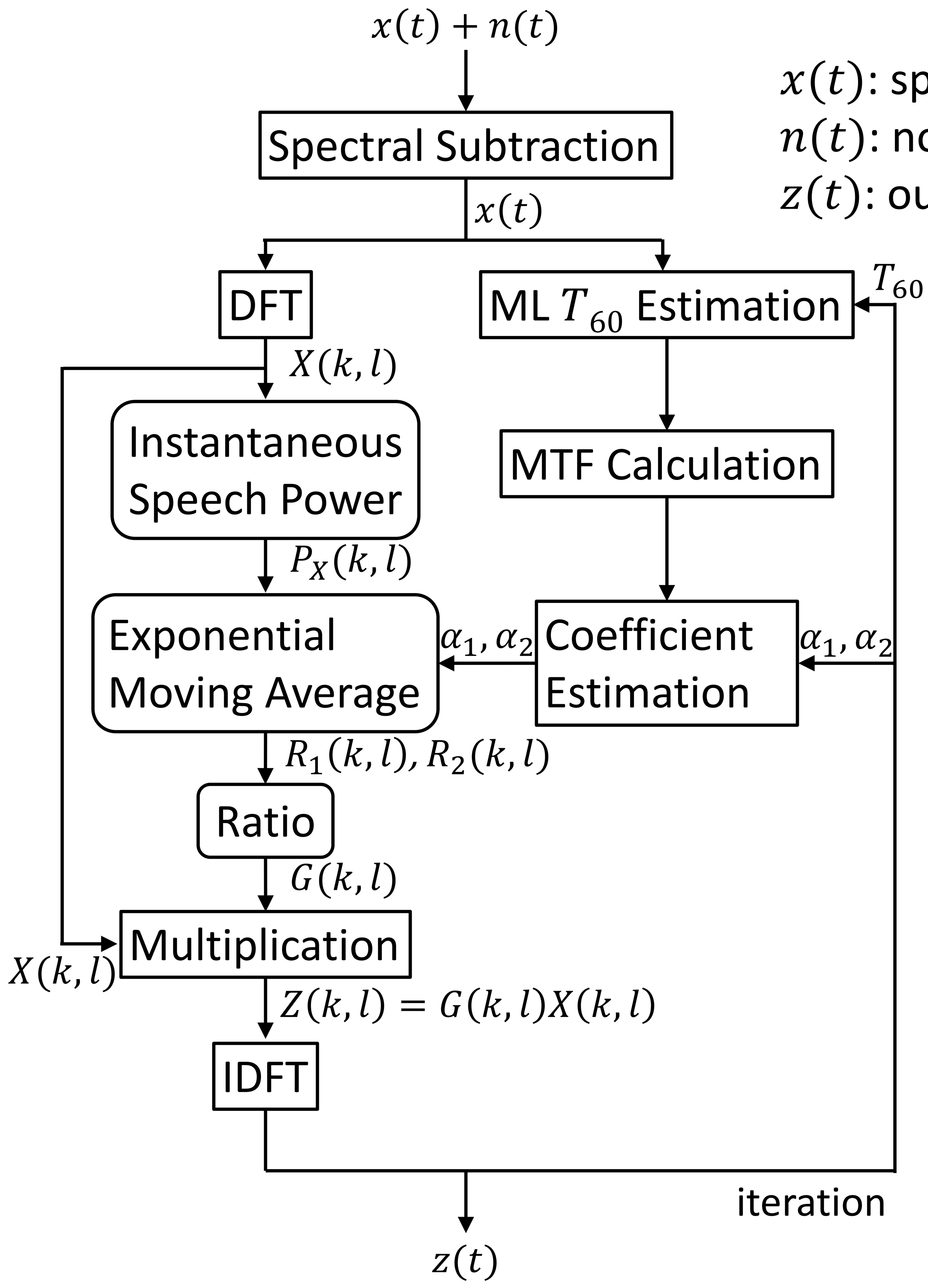
- Inverse characteristics of MTF
- This can be calculated as:

$$H(f_m) = \frac{1-\alpha_2}{1-\alpha_2}\frac{1-\alpha_2 \exp(-j2\pi f_m)}{1-\alpha_1 \exp(-j2\pi f_m)}$$



Using steepest descent method for estimation

OVERVIEW OF SINGLE ITERATION

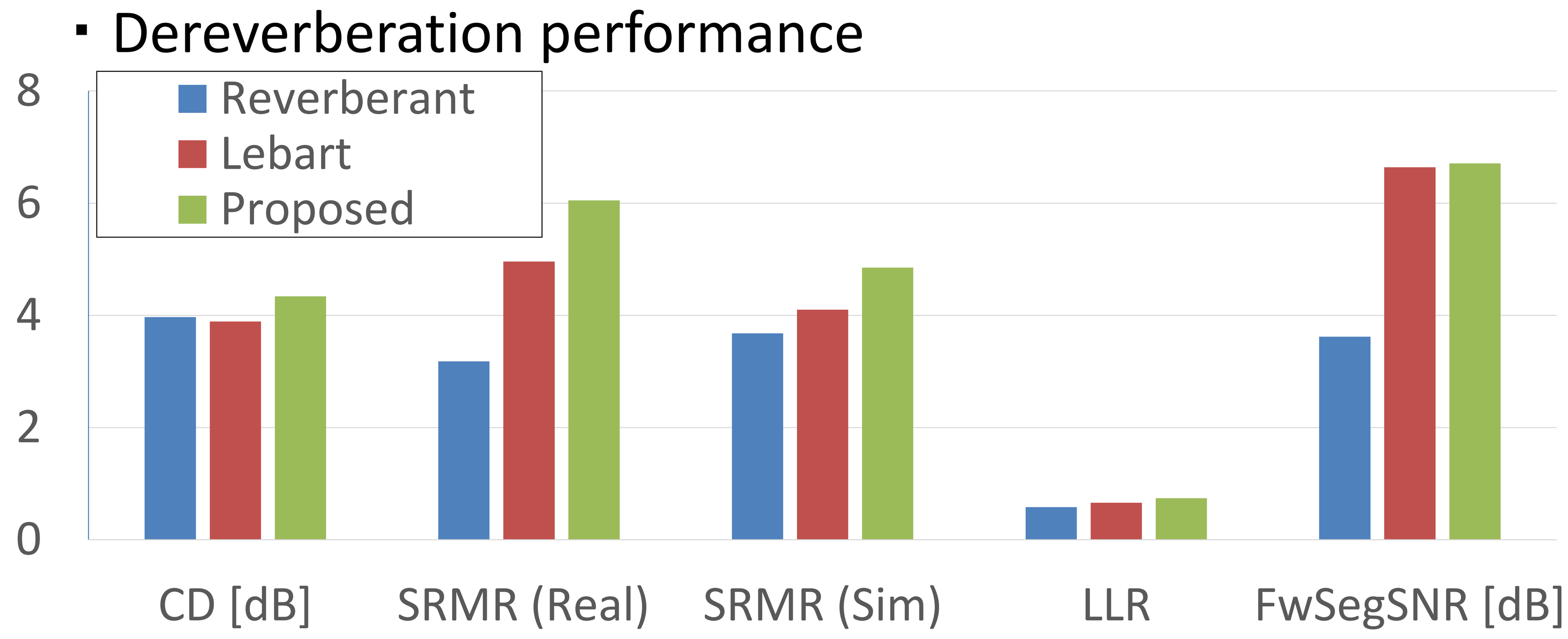


$x(t)$: speech signal
$n(t)$: noise
$z(t)$: output signal

EXPERIMENTAL RESULTS

Compared with Lebart et al., 2001

- Dereverberation performance

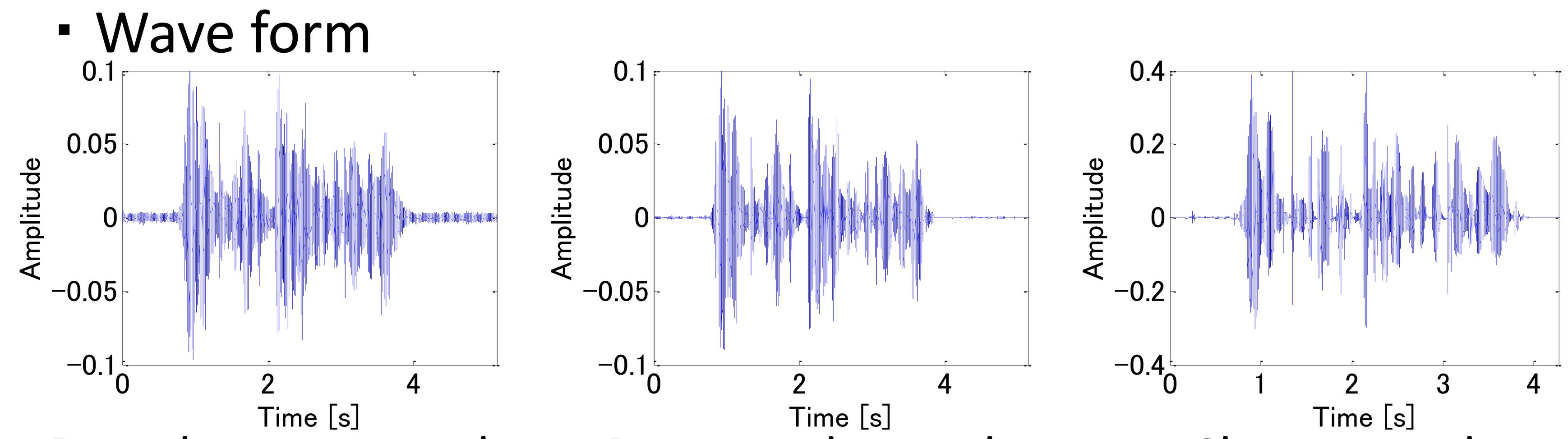


CD: Cepstrum Distance
SRMR: Speech-to-Reverberation Modulation energy Ratio
FwsegSNR: Frequency-weighted Segmental SNR
LLR: Log Likelihood Ratio

- Calculation time

Method	Proposed		Lebart et al.	
Dataset	Simulated	Real	Simulated	Real
Real time factor	0.0743	0.0707	0.3632	0.3594

- Wave form



Reverberant speech　Processed speech　Clean speech

Conclusion

Proposed a parameter estimation method

- Using MTF and amplitude response of the filter
  - They have inverse characteristics
- We can estimate the parameters properly
  - Improvement of SRMR and FwSegSNR
- Proposed method can process effectively