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Band-width Expansion Method Using Spline Codebook-based Spectral Folding

Jihoon Park¹, Seungho Han¹, Heesik Yang¹, Sangbæe Jeong¹, and Minsoo Hahn¹
Speech and Audio Information Lab., School of Engineering
Information and Communications University
119, Munjiro, Yuseong-gu, Daejeon, 305-732, Korea
Tel: +82-42-866-6283, FAX: +82-42-866-6245
E-mail: bath02zn@icu.ac.kr, space0128@icu.ac.kr, sheik@icu.ac.kr, sangbae@icu.ac.kr, mshahn@icu.ac.kr.

Abstract: This paper proposes a band-width expansion method using the spline codebook based spectral folding. The spline codebook consists of the clustered cepstrum and the corresponding splines. The experimental results of the PESQ, the MOS, and preference tests show that the proposed method is superior to the ABE one.

1. Introduction

The narrow-band speech has muffled sound and shows less sufficient intelligibility and naturalness than the wide-band speech because of the deficiency of high-frequency components. The Band-Width Expansion (BWE) method expands band-width of a speech from the narrow-band speech to the wide-band one. The BWE method uses the characteristics such as the spectrum envelope and the excitation signal of narrow-band speech for the estimation of high-band components.

BWE methods generally consist of two processes. One is the estimation of the spectrum envelope in which the VQ (Vector Quantization)[1], the GMM (Gaussian Mixture Model)[2] and the HMM (Hidden Markov Model)[3] are generally used. The other is the generation of an excitation signal. A periodic impulse signal and the mixture excitation signal are used for the generation of the excitation signal. Recently, L. Laaksonen et al. proposed the ABE (Artificial Band-width Expansion) method [4] in which the spectral folding and spline are used. The estimation of the spectrum envelope and generation of the excitation signal are simultaneously performed.

2. Artificial band-width expansion method

The ABE method consists of two processes, i.e., the training part and the restoration part. In the training part, the frame is classified into two categories; voiced and unvoiced sound. Consequently, the five control points are calculated between the original speech and the spectral folding speech in frequency domain. For each category, we calculate the average for each set of five control points. Using a cubic spline method, a spline curve is interpolated among the five control points.

Figure 1 shows the restoration process of the ABE method. The narrow-band speech is expanded using the spectral folding method. And, a frame is classified into either voiced or unvoiced sound. Spectral folding speech modifies high-band components using decoded spline. However, the ABE method doesn’t express a variety of splines because the splines are generated by classification of voiced sounds and unvoiced sounds only.

3. Proposed method using spline codebook

The proposed system also consists of the training and the restoration part.

3.1 Training part

Figure 2 shows the training process. The cepstrum codebook for the spectral folding speech is generated by the VQ of the extracted cepstrum. The spline codebook is made with the cepstrum codebook and the corresponding splines. The splines are generated by the similar method to ABE one. In the proposed method, various splines are expressed as shown in Figure 3. Thus, the proposed method generates the high-band components more correctly.

3.2 Restoration part

In the restoration part, the spline for an input speech is decided using the cepstrum and spline codebooks as shown in Figure 4. Finally, the wide-band speech is constructed by applying the spline to the spectral folding speech.