HIGH FREQUENCY COMPONENTS RECOVERY IN MUSIC SIGNALS

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Abstract

A new technique is presented which improves the subjective quality of band-limited music by recovery of high frequency components. Sequences of harmonics are found in the band-limited signal and these sequences are expanded to the high frequency band to estimate the lost part of spectrum. High frequency signal is generated to match this estimation and is added to the band-limited signal.

Keywords

digital signal processing, signal restoration, music signals, spectrum analysis

1. Introduction

The problem of high frequency components recovery in audio signals have already been solved for speech signals. In [1] and [2] the recovering of wide-band speech from a speech with spectrum limited to telephone range is described. Both algorithms need a trained codebook. Our algorithm is designed for music signals and does not use any codebook.

2. Approach and algorithm

The substance of our algorithm is find sequences of harmonic components in band limited signal that are harmonics of a certain tone and expanding this sequences to high frequency band. This requires a proper segmentation of band limited signal (2.1), detailed spectrum analysis of each segment (2.2), finding out sequences of harmonics (2.3), expanding this sequences into high frequency band (2.4) and generation of high frequency band signal (2.5) that is added to the low frequency band signal to obtain the resultant signal.

2.1 Segmentation

The purpose of signal segmentation is to divide signal into time segments in which no significant spectral changes appears, i.e. the same tones are played. One possible algorithm is to observe changes in spectrogram. The solid line on Fig. 1 is sum of differences of spectrogram values between one and the next time block in all frequency slots within the significant band 600-2000 Hz. The dashed-dot line is a shifted moving average of this line defined correspondingly to (1) with parameter n=3 and q=0.12. Peaks of solid line above the dashed-dot line determine time segment borders.

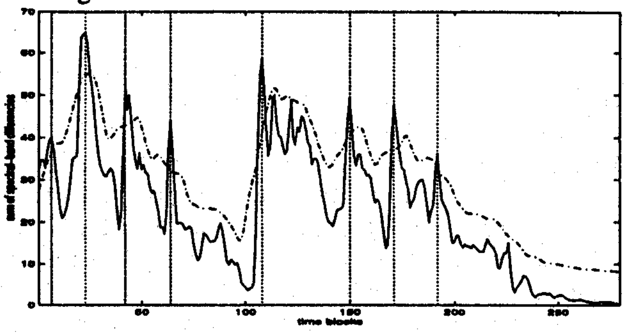

Fig.1 Illustration of signal time segmentation

2.2 Signal spectrum analysis

The purpose of the analysis is to find harmonic components in band-limited signal segment. Fourier transformation with Hanning window gives us a first overview of the spectrum. In the modular spectrum we can find a batches of values which are higher then the moving average defined as

$$\tilde{P}_i = \frac{P_{i-n} + \ldots + P_{i+n}}{2n+1} + qP_{\max}, \tag{1}$$

where n is number of used previous and following values (n=2 in our example), q is a coefficient (q=0.015) and $P_{\max}$ is maximum value of all P_i.

Each batch represents one harmonic component whose frequency f_c and peak value A_c can be computed from the two highest values P_x and P_y in the batch corresponding to frequencies f_x and f_y ([3] chapter 5.5):

$$f_c = f_x \pm k\text{, where } k = \frac{2P_y - P_x}{P_x + P_y}. \qquad (2), (3)$$

$$A_c = \frac{P_x}{\mathrm{sinc}(k) + \frac{1}{2}(\mathrm{sinc}(k-1) + \mathrm{sinc}(k+1))} + \\ + \frac{P_y}{\mathrm{sinc}(1-k) + \frac{1}{2}(\mathrm{sinc}(-k) + \mathrm{sinc}(2-k))}. \tag{4}$$

The sign + in (2) is used when $f_y > f_x$, otherwise use –.

2.3 Finding harmonics

Sequences of harmonics are to be found within a vector **f** of all harmonic component frequencies. Let us take f_i – the frequency of a harmonic component with the highest amplitude A_i – and compute a vector $\mathbf{d} = |f_i - \mathbf{f}|$. Then, let us take d_2 – the first non zero member of **d** – and try whether there exist members of **d** equal to $2d_2$, $3d_2$, etc. with a limited deviation. If more than 2/3 of all possible members in the sequence exist, a sequence of harmonics is found. Then let us take d_3 and try the same and so on. After that let us compute new **d** centred to harmonic component with the second highest amplitude and repeat this algorithm. It is necessary to check whether a found sequence of harmonics has not been found before. This algorithm can continue with next vectors **d** centred to the next harmonic components with the highest amplitudes.

2.4 Spectrum prediction

Each sequence of harmonics can be expanded to the high frequency band. In a simple case, amplitudes of the new harmonic components are obtained from an approximation of sequence of known harmonics amplitudes by an exponential descend function and setting amplitudes of new harmonic components equal to this function values.

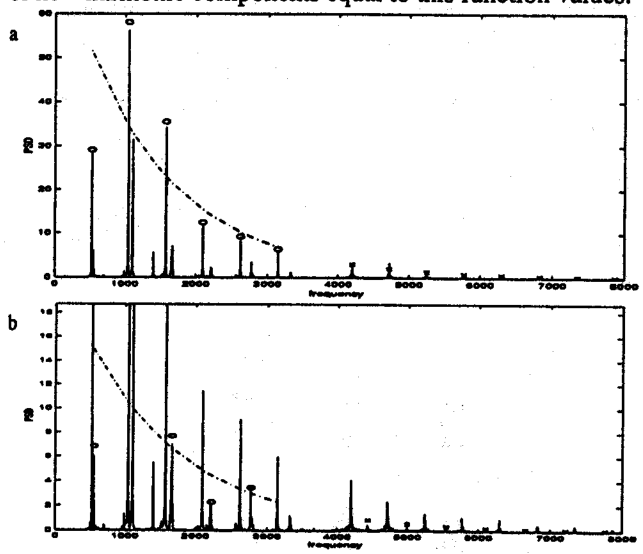

Fig. 2 Spectrum prediction illustration

Fig. 2a displays the spectrum of the original low frequency signal in the band 0 to 4000 Hz. The circles mark computed peaks of harmonic components that form a sequence of harmonics. The dash-dot line is the approximation by an exponential function. Based on expansion of this sequence a high frequency signal is generated whose spectrum is in the band 4000 Hz to 8000 Hz. Fig. 2b is y-axis detail and displays another sequence of harmonics in the same signal, exponential approximation and generated high frequency harmonic components.

2.5 High Frequency signal generation

Pole filtering of white noise can generate each harmonic component of high frequency signal. The filter transfer function in Z-domain is:

$$K(z) = \frac{1}{(z-p)(z-p^*)}, \qquad (5)$$

where pole p is

$$p = 0.999 \cdot \exp\left(j2\pi \tfrac{f_c}{f_s}\right). \qquad (6)$$

f_s is the sampling frequency. The filter output is gained by $A_c\left|\left[\exp\left(j2\pi \frac{f_c}{f_s}\right) - p\right]\left[\exp\left(j2\pi \frac{f_c}{f_s}\right) - p^*\right]\right|$ to set the expected amplitude of the harmonic component.

3. Results

The example music signal starts by solo instrument playing. After all orchestra clap more instruments play together.

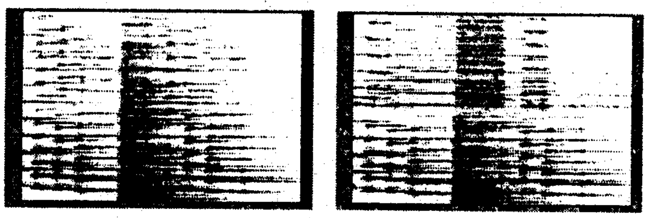

Fig. 3 Spectrograms of the original and the restored signal.

4. Conclusion

The quality of the reconstructed signal is much better then the quality of the frequency limited signal, even though it sounds different than the original signal. The timber of the high tones is changed due to the simple exponential distribution of amplitudes of generated harmonics. This distribution and all principals of the algorithm match up to psycho-acoustic conditions to make the resultant signal sound naturally.

References

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