Removal of Real World Noise in Speech: Comparison of Various Parameters Using Kalman and H-Infinity Filter Algorithms

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Abstract

Background/Objectives: The performance of the speech enhancement techniques is examined by applying them to the speech signals corrupted by real world noise. Methods: An advanced coding Methodology is introduced to reduce the noisy signal using H-infinity filter. Here two compression techniques - Switch Split Vector Quantization (SSVQ) and Multi Stage Vector Quantization (MSVQ) are used. The enhancement techniques like Kalman Filter and Recursive Filter were assessed for similarities and differences with H-Infinity Filter and the outcomes are compared using signal to noise ratio which is likely to be affected on real world noise. Findings: In this paper using kalman filter, recursive filter, h-infinity filter plays vital role for comparing the various parameter characteristics. Kalman filter uses ordered steps that solve a mathematical problem. H-infinity filter differs from the normal changed Kalman filtering as it requires the knowledge of commotion parameters. H-infinity minimizes the estimation errors and thus obtains robustness and obtains the better results in enhancement. Application/Improvements: This study can be helpful for improving the intelligibility in speech signal and the same method can be implemented with different types of hybrid vector quantization techniques and may achieve better SNR and quality of the signal.

Keywords: Commotion, Multi Stage Vector Quantization (MSVQ), Real World Noise, Signal-to-Noise Ratio (SNR), Switch Split Vector Quantization (SSVQ)

1. Introduction

Speech signal is the most important aspect for Communication in humans. There are different types of communications we use like speech, image and video but among them speech plays the vital role to communicate to the distance through telephone and various mediums. In digital communication speech signal representation is the most significant. Recognizing, compressing and Enhancing of noisy speech in reverberant conditions is a difficult task. The goal of speech enhancement is to ameliorate quality of speech using different algorithms. In order to examine the results of the various algorithms in presence of real world constraints, some of the real world noises of machine gun, industry, destroyer, bus, emergency vehicles etc are added to the speech.

2. Methodology

From the process flow diagram i.e Figure 1, compression techniques and enhancement techniques are implemented on the results. Given input signal as a speech, using compression technique the compressed noise signal and in addition with real world noise signal is added. The resultant signal is processed using enhancement technique to compare various noise ratios.

2.1 Compression and its Techniques

Speech transmission and storage and transport of information are the significant fields of research. For compression, the speech signal is done first and then the
The aim of this compression is to demote the rate of transmission.

The uses of speech signal compression is to lower the rate during the data transmission from the transmitter to receiver. MSVQ and SVQ are two different compression techniques used. MSVQ converts the widely spread amplitudes into concentrated residual errors which lies in smaller range near the origin. This conversion increases the quantization efficiency. As the name indicates, in split vector quantization the vectors of larger dimensions are split into vectors of smaller dimensions. During this process the bits allocated to the quantizer are divided among the splits. As a result of splitting, vector dimension decreases and thus it increases the number of training vectors which are used for codebook generation. This results in the improvement of quantization performance and also reduces complexity and memory requirements.

2.2 Speech Enhancement

The main goal of Speech enhancement is to ameliorate the intelligibility, quality and pleasantness of a speech signal. In speech enhancement noise is removed by estimating. The properties and components of noise can be canceled there by retaining the clear speech signal. The problem involved with the speech enhancement techniques is the parts of required speech are also removed along with the noisy portion of the speech signal. During the removal of noise, speech enhancement techniques corrupt the speech signal. Therefore, noise removal cannot be done effectively due to this reason compromising must be done between the effectiveness of noisy speech signal and level of distortion. Present speech processing algorithms are divided into three domains, spectral subtraction, sub-space analysis and filtering algorithms. Using filtering approaches like kalman filter, recursive filter and H-Infinity filters the noise is estimated and removed.

2.2.1 Kalman Filter

Kalman filter developments are limited to its research and application because the algorithm is designed in such a way that it does not have a particular way in order to determine the initial conditions. As to start the recursive process of kalman filter early conditions of mean and variance state vector should be known.

2.2.2 Recursive Filter

The Recursive filter is based on the concepts of Kalman filter. Recursive filter is designed with an estimator which can estimate recursively and updates each frame of the speech signal. Hence it can be concluded that recursive filter can track and estimate non-stationary noise.

2.2.3 H-Infinity Filter

The H-Infinity filter is different from the kalman filter it doesn’t require any information about the previous knowledge on noise statistics rather it only depends up on the finite energy and H-infinity filter minimizes the estimation errors and obtain the robustness with better outcome during enhancement.

3. Results and Discussions

The below results Figure 2 and 3 shows the various comparisons of signal to noise ratio in terms of dB’s, spectral subtraction and kalman filter with 2 dB and 10 dB.
Table 1. Performance Comparison with input SNR=5, dB Filtering Output SNR (dB)

| Algorithm   | White Noise | Helicopter Noise |
|-------------|-------------|------------------|
| Kalman      | 8.7276      | 8.9119           |
| H-∞         | 9.8781      | 10.0693          |

dB and its performances as shown in the Table 1 and 2 comparisons of SNR with 5dB.

4. Conclusions

As speech is non-stationary enhancement is a special case of signal estimation and hence human ear is the ultimate judge and it requires a mathematical error criterion. Compressions of various parameters in speech to estimation and produce quality speech at the end of receiver. From Figure 2 and 3, these techniques explicitly state that the difference between their theoretical background. The results from the Table 1 and 2 shows that H-infinity filter method gave the better noise reduction capability in comparison to other speech enhancement methods presented.

Table 2. Results after compression and enhancement.

| Type of real world noise | Compression using MSVQ | Enhancement using spectral subtraction | Enhancement kalman filter | Enhancement using recursive filter | Using H-Infinity Filter |
|--------------------------|------------------------|--------------------------------------|---------------------------|-----------------------------------|------------------------|
| Factory                  | -21.3076               | -11.0212                             | -1.8995                   | 1.9935                            | 1.8732                 |
| Fire engine              | -22.0378               | -4.9626                              | -1.0830                   | 2.1531                            | 2.0952                 |
| Machine gun              | -17.3602               | -10.7221                             | -2.4530                   | 3.3546                            | 3.1241                 |
| Vehicle                  | -21.9860               | -5.5831                              | -1.1032                   | 2.2501                            | 1.3542                 |
| Volvo bus                | -19.9832               | -11.2567                             | -1.6725                   | 2.3672                            | 1.9234                 |
| Destroyer                | -19.1030               | -9.7234                              | -1.9632                   | 2.0863                            | 1.9243                 |
| Ambulance                | -6.25175               | -2.76305                             | -4.9632                   | 3.5723                            | 2.8192                 |
| Traffic                  | -20.8796               | -11.8905                             | -1.8320                   | 2.7649                            | 2.5315                 |
5. References

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