

# Objective Prediction of the Sound Quality of Music Processed by an Acoustic Echo Canceller

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**ABSTRACT:** Acoustic echo cancellation is a common method of cancelling acoustic feedback in a variety of electro-acoustic systems. Such as digital hearing aids. A new method for the objective assessment and prediction of perceived audio quality is introduced. Reliable objective metrics that predict user perceived distortion could significantly reduce the noise and cost for new hearing aids. The aim of this study was to improve the ability of different objective metrics to predict the subjective ratings of sound quality of music which is processed by acoustic echo canceller. A multivariate network using these perceptual measures was found to have the lowest recorded median generalization error when tested on different musical samples. The median ICC between subjective ratings and ratings predicted by this network architecture is high( $r = 0.97$ ) which compares favourably to results from previous studies of perceptual quality metrics.

**KEYWORDS:** Acoustic Echo canceller, hearing aids, objective quality assessment.

## I. INTRODUCTION

Acoustic Echo cancellation is a common method of cancelling audio feedback in a variety of electro-acoustic systems such as digital hearing aids. The time varying acoustic feedback leakage paths can only be eliminated with adaptive feedback cancellation. When an electro-acoustic system with an adaptive feedback canceller is presented with a correlated input signal, a recurrent distortion artifact, entrainment is generated. There is a difference between the system identification and feedback cancellation [1].

Acoustic echo cancellation has its application in echo cancellation. The error between the desired and the actual output is taken and given as feedback to the adaptive processor for adjusting its coefficients to minimize the error. In hearing aids, feedback arises when a part of the receiver (loudspeaker) signal is captured by the hearing aid microphone(s), gets amplified in the device and starts to loop around through the system. When feedback occurs, it results in a disturbingly loud tonal signal.

Acoustic echo cancellation can increase the maximum closed-loop gain (before instability) by as much as 20 dB [1], enabling the use of less occluding, more comfortable hearing aid ear moulds. Perceptual quality models have been successfully evaluated for predicting subjective quality ratings of audio signals processed by algorithms such as RLS, achieving (Pearson) correlation coefficients with subjective evaluations is of  $r = 0.97$ . Hence by using this approach we improve the interclass correlation and minimize the generalization error.

## II. RELATED WORK

Previously the various univariate and multivariate models are used for calculating the median generalization error and the interclass correlation. Noise loudness, Noise to masking ratio(NMR) and Average block distortion(ABD) are the univariate models but out of these models the noise loudness and the NMR has the lowest median generalization error(i.e.0.252,0.329) [2] than the average block distortion(0.342)[3]. The Rnonlin and PSMt are the correlation based models which have high median generalization error and smaller ICC value.

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Where the multivariate models like AvgModDiff1 and ABD has ( $r=0.88$ ), AvgModDiff1, WinModDiff, Noise loudness Rnonlin, AvgModDiff2 combinely has ( $r=0.94$ ), WinModDiff, Noise Loudness and Rnonlin has ( $r=0.95$ ) and median generalization error is 0.204. By comparing these previous results it is concluded that the best models is that which has the lowest median generalization error and the high interclass correlation i.e ICC value. hence in this paper by using the WinModDiff, Noise loudness, Rnonlin, RLS model the mean generalization error is 0.109 and the interclass correlation i.e. ICC value is ( $r = 0.97$ ). Table shows the comparison between these results.

Table :MedianGeneralization Error And Interclass Correlation.

Pruning stage	Mean Generalization errors (subjective scale units)	Median intraclass correlation
<b>Stage 0</b> : All perceptual measures	0.269	0.91
<b>Stage 1</b> : NMR, WinModDiff, ABD, HSE, AvgModDiff1, Noise Loudness, Rel Dist frames, PSMT, Info-Pstream, Rnonlin, AvgModDiff2	0.246	0.94
<b>Stage 2</b> : WinModDiff, AvgModDiff1, Noise Loudness, Rel Dist frames, Rnonlin, AvgModDiff2	0.226	0.94
<b>Stage 3</b> : WinModDiff, AvgModDiff1, Noise Loudness , Rnonlin	0.214	0.94
<b>Stage 4</b> : WinModDiff, Noise Loudness, Rnonlin	0.204	0.95
<b>Stage 5</b> : WinModDiff, Noise Loudness, Rnonlin, RLS	0.109	0.97

### III. PROPOSED METHOD

1. Perceptual measures based on distortion loudness: it consist of three methods like noise loudness, noise to masking threshold ratio and Relatively disturbed frames. i.e.
  - 1.1. Noise loudness: The noise loudness method is used to increase the loudness of the input signal.
  - 1.2. Noise to masking threshold ratio: This method is used to avoid the interrupt signal.
  - 1.3. Relatively disturbed Frames: This method is used to avoid the feedback canceller frames in excess of 1.5 db.
 These three perceptual measures are used to measure sensitive to distortion loudness or masked distortion levels.
2. WinModDiff, AvgModDiff1 and AvgModDiff2: This perceptual measure method which measures sensitive to modulation effect.
3. Perceptual measure based on the probability of noise detection : it consist of two methods detection probability and average block distortion(ABD).
  - 3.1. Detection probability : This perceptual method gives the probability of audible difference between processed and unprocessed signals.
  - 3.2. Average Block Distortion : By using this perceptual measure we can take the average of these frames which are above the 50% levels.

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4. Perceptual measures sensitive to nonlinear distortion :It contains three methods PSMt (perceptual similarity measures) , Rnonlin and signal to distortion ratio.
  - 4.1.PSMt : this measures is based on the correlation coefficient calculated between output of the modulation filter.
  - 4.2. Rnonlin : this perceptual measure used the different levels of waveforms means it measures the higher level and lower level of signals .where higher level waveforms are speech signals and lower level waveforms are for noise signal. It is different from the PSMt measures.
  - 4.3. Signal to distortion ratio : this perceptual measure is less sensitive to slow moving non linearity such as automatic gain control (AGC). It is used to find the ratio between speech and noise which is useful for audio quality assessment of adaptive feedback canceller.These three method i.e PSMt, Rnonlin and SDR are sensitive to nonlinear distortion.
5. Harmonic structure of Error : Harmonic means real data which is affected due to noise is known as harmonic structure error. This harmonic structure masks the noise which is greater than the speech signal which makes it more noticeable to listener.
6. Acoustic Echo Canceller : The AEC is the popular application of adaptive filter. The role of this is to identify the acoustic path between loudspeaker and microphone. There are different algorithms have been developed and are applicable for acoustic eco cancellation like LMS,NLMS and RLS algorithms. This AEC avoids the different echo produced at the output of hearing aid amplifier of input audio speech signal. Hence we get original speech signal and again it gives to the input side using feedback. As there are various types and causes of Echo with unique characteristic including acoustic Echo (sound from loudspeaker being reflected and recorded by a microphone which can vary substantially over time) and line Echo (electrical impulses caused by, e.g. coupling between the sending & receiving wires, impedance mismatches, electrical reflections etc. which varies much less than acoustic echo).The acoustic echo canceller can cancel line echo as well as acoustic echo."AEC" in particular is commonly used to refer to echo canceller in general, regardless of whether they were intended for acoustic echo, line echo or both.

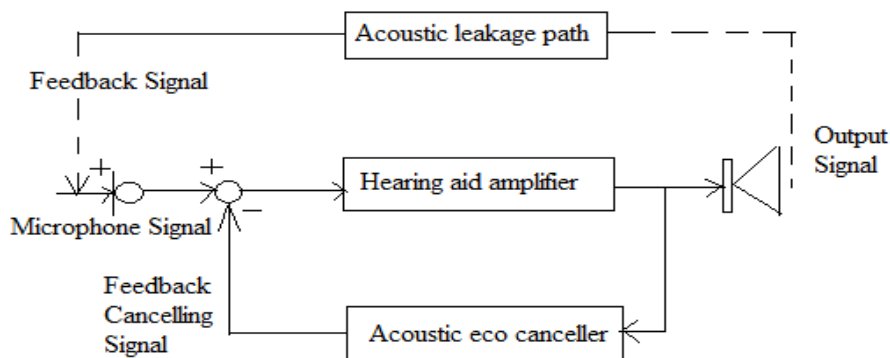


Fig.1.Signal flow diagram of acoustic echo canceller.

As shown in fig.1. when microphone receives the signal its output is then given to the hearing aid amplifier which amplifies the input speech signal. but thereis echo signals present at the output of hearing aid amplifier which must be removed using the feedback canceller that is acoustic echo canceller. which is used to avoids these echo signals. Hence acoustic echo canceller takes these echo signals along with the original signal passes through the hearing aid. In that process using negative feedback only error signals passes throughhearing aid amplifier.At the same time the remaining signals are given to the microphone using the acoustic leakage path. Using this procedure we can minimise the noise in the input speech signal and hence we get the original speech signal with improved sound quality.

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## IV. EXPERIMENTAL RESULTS

Figure shows the results of the input audio samples using acoustic echo canceller and RLS algorithm. Fig.2. Shows the input audio signal. Which is then divided into two channels i.e. left channel and right channel fig.3 and fig. 4 shows the output of input audio signal after they divided into these two channels. Whereas fig.5. shows the comparison between the filtered signal to reference signal. After eliminating the noise from the input signal the sound quality of music sample is improved which is shown in fig.6. where fig.7 shows that the regression plot of results from the validation tests which indicates the objective prediction of the audio samples and also shows the median interclass correlation factor  $r$  is much better than the previous method.

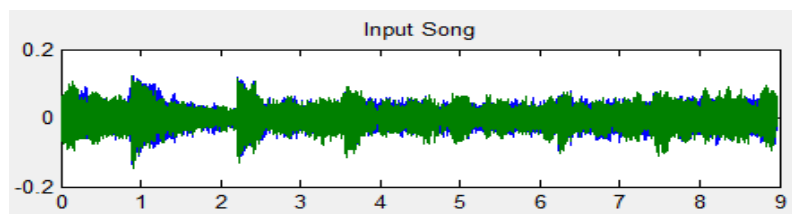


Fig.2. Input Audio Signal.

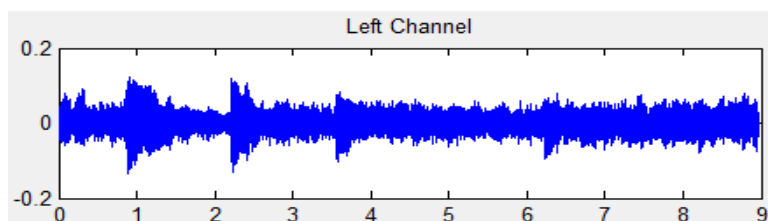


Fig.3. Audio Signal is Divided into Left Channel.

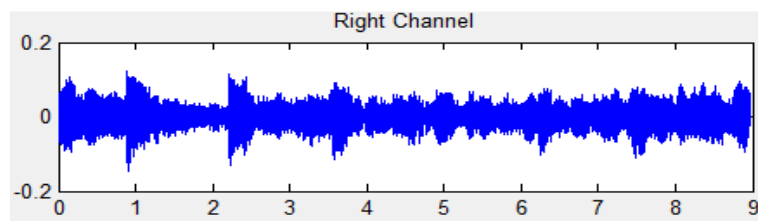


Fig.4. Audio Signal is Divided into Right Channel.

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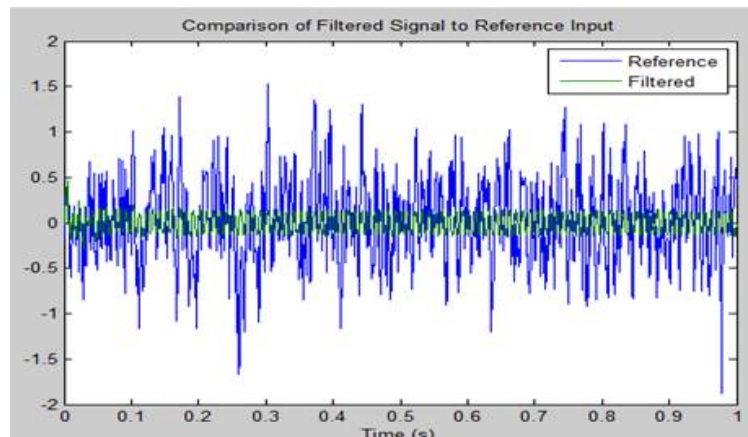


Fig.5. Comparison of filtered signal to reference input.

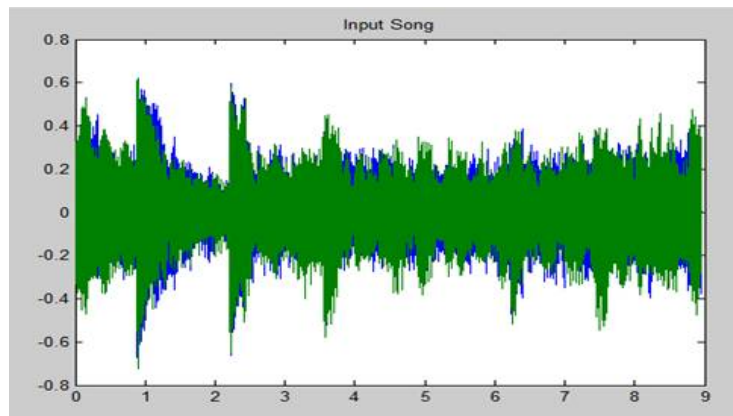


Fig.6. Final Output Of Audio Signal With Improved Sound Quality.

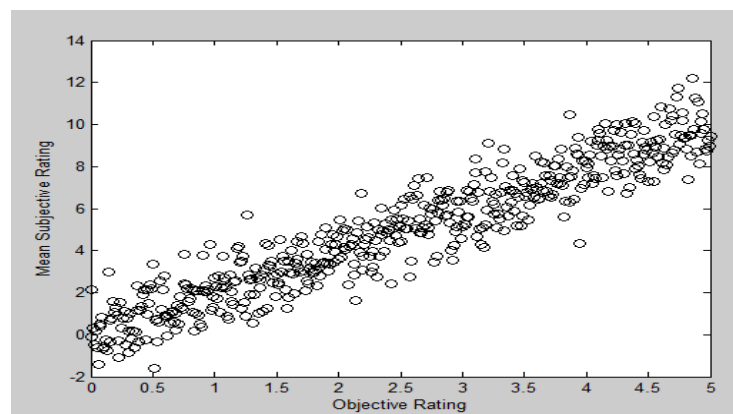


Fig.7. Regression plot of the results from the validation test.

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## V. CONCLUSION

In this model using these perceptual measures was found that they have the lowest median generalization error of the various audio samples that are tested. The median interclass correlation(ICC) rating predicted by this architecture is high (i.e  $r=0.97$ ) which is much better than the previous methods. As previously used models like noise loudness had lowest median generalization error and the other perceptual measures like WinModiff & Rnonlin are sensitive for the modulation difference between clean and processed audio signals. By using this results and analysis it is concluded that this system has more accurate prediction of subjective quality ratings of audio samples. This model suppresses the acoustic feedback and makes the hearing aid comply with a number of additional requirements like allow fit to target, withstand tonal sound sources ,withstand changes to feedback path and also it reduces the development cost of new hearing aid. Because of these features this model is applicable in hearing aid, video and audio signal processing.

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