CLICK REMOVAL IN DEGRADED AUDIO

Report for Module EEP55C22 Computational Methods

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This report is submitted in part fulfilment for the assessment required in EEP55C22 Computational Methods. I have read and I understand the plagiarism provisions in the General Regulations of the University Calendar for the current year. These are found in Parts II and III at http://www.tcd.ie/calendar.

This report describes about removal of clicks by using Auto-regressive models. The clean signal was degraded, the degraded signal y_k has artificial clicks. We create frames, so that we can analyse for each blocks. After finding the co-efficient, the residue is found. By interpolation, the clicks are replaced and the restored signal is found. At the end, all the frames are added together to form restored signal.
The signal is analysed by plotting various graphs and best value of parameters are chosen.

1 Creating degraded signal

Let us define the observed degraded signal as y_k and the original, clean signal as c_k. We can model the degradation as follows.

$$y_k = \begin{cases} r & \text{for every 1000 value of } c_k \\ c_k & \text{Otherwise} \end{cases} \tag{1}$$

where r =+1 or -1 is a random corruption applied in sampled data and k is total number of sampled data obtained while sampling "clean.wav"

Our problem is to find clicks geneared by degraded signal y_k and remove the generated clicks to reconstruct original signal s_k.

2 Autoregressive method process

To replace the clicks with approximate value in degraded signal, we utilise Auto regressive model algorithm.

2.1 Estimating AR coefficients

To obtain AR coefficents, we normalise the signal y_k to remove dc components.
Consider a signal $\hat{y_k}$

$$\hat{y_k} = \sum_{p=1}^{P} a_p y_{k-p}$$

The error can be predicted by

$$e_k = y_k - \hat{y_k}$$

$$e_k = y_k - \sum_{p=1}^{P} a_p y_{k-p}$$

To estimate coefficients by least square method, we use

$$e_k^2 = y_k - \sum_{p=1}^{P} a_p y_{k-p}$$

We denote the least square by E

$$E = \frac{1}{N} \sum_{k=0}^{N-1} e_k^2$$

On substituting, e_k^2 in E and making E = 0 , we get

$$\sum_{k=0}^{N-1} y_{k-p} y_{k-j} \sum_{p=1}^{P} a_p \sum_{k=0}^{N-1} y_k y_{k-j}$$

This is expressed in matrix form and can be summarised as

$$Ra = r$$

The coefficients a is found using

$$a = R^{-1} r$$

2.2 Residual

The prediction error is also known as residual

$$e_k = y_k - \hat{y_k}$$

$$e_k = y_k - \sum_{p=1}^{P} a_p y_{k-p}$$

For model order P= 5,

$$e_k = y_k - a_1 y_{k-1} - a_2 y_{k-2} - a_3 y_{k-3} - a_4 y_{k-4} - a_5 y_{k-5}$$

In Matlab, this operation is performed by using an FIR filter. The residue is convolution of the degraded signal y_k with taps $[1, -a_1, -a_2, -a_3, -a_4, -a_5]$. The FIR filter with taps $h_0 = 1, h_1 = -a_1, h_2 = -a_2, h_3 = -a_3, h_4 = -a_4, h_5 = -a_5$

2.3 Interpolation

The fundamental principle guiding the usage of an AR model for the interpolation of missing samples is to use the model to synthesize data in the missing patches that agree with the model itself. This idea was refined by Sayeed Vaseghi, which was proposed by Raymond Veldhuis. The estimation of error can be written in matrix form as

$$e = Ay$$

where A is the AR coefficients matrix arranged to perform the convolution easily.

Splitting the data vector into the known data y_k and unknown samples y_u. A matrix can be written as

$$e = A_k y_k + A_u y_u$$

where A_u, A_k are coefficient matrices corresponding to linear operations on the unknown data y_u and known data y_k respectively. To estimate the values we make e = 0

$$A_k y_k + A_u y_u = 0$$

Multiply by $A_u^{'}$ on both sides to find the y_u, we get

$$A_u^{'} A_k y_k + A_u^{'} A_u y_u = 0$$

On solving, we get

$$y_u = [A_u^{'} A_u]^{-1} A_u^{'} A_k y_k$$

3 The algorithm

The algorithm is valid only for statistically homogeneous signal. Real signals are not, hence we process the signals in frames. Our algorithm is shown below.

1. The signal is processed into data and first column of data is used.
2. The block is processed into frames depending on the frame duration value, model order.
3. The co-coefficient's for each frame is found
4. The residual for each frame is found and joined together to form a residual signal and plotted as shown in Figure(4).
5. The error signal is generated from residual signal and shown in Figure(4).

$$e_k = \begin{cases} 1 & \text{residual value} > \text{threshold} \\ 0 & \text{Otherwise} \end{cases} \tag{2}$$

6. The restored signal is formed after all frames are joined together. The graph is as shown in Figure(5).

4 Analysis and results

To test our algorithm, first we degraded a signal, the signal was degraded by equation (1) and the diagram is shown in Figure(1)

4.1 Analysis

To analyse the data, we experimented the degraded signal with varying model number, threshold and frame size. Some of the experiments are:

1. Model number is varied, the MSE is calculated.

   The MSE gives an high value for a particular model number. This shows that, the quality of restored signal can be optimised by varying model number.
   For this Figure(2a), we see that, there is an high value of MSE, when model order = 9.

2. Threshold is varied, the Time executed is calculated.

   When threshold is increased, the executed time decreases. As per the Figure(2b), the threshold decreases spontaneously till 0.3 and declines slightly till 1.

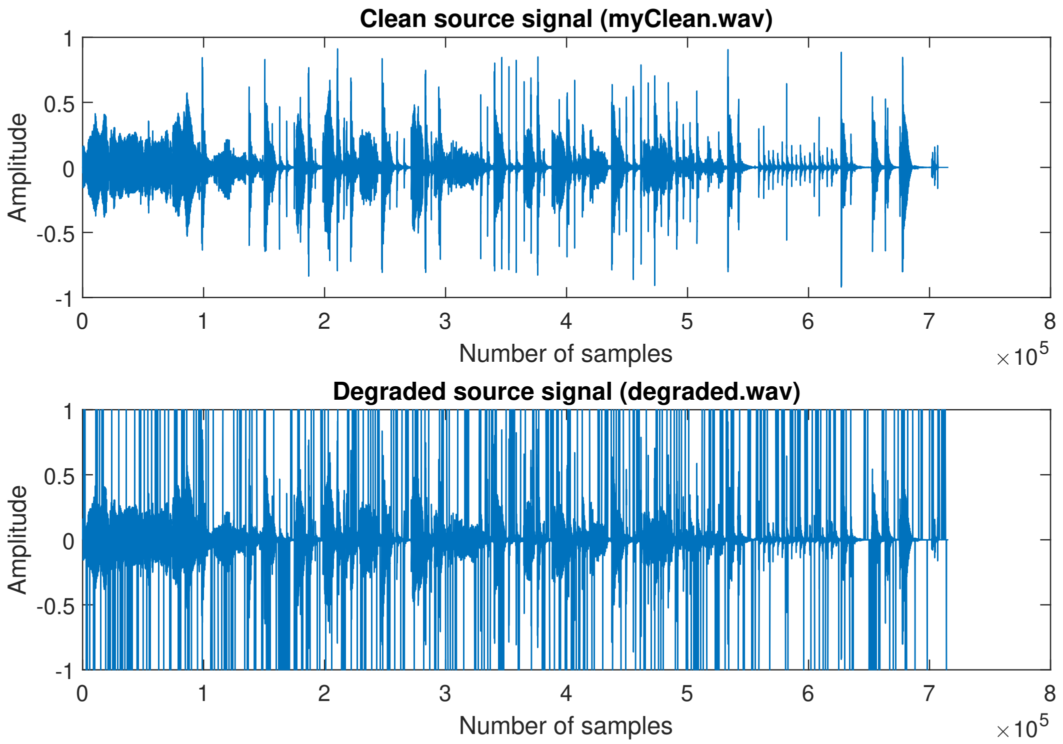

Figure 1: *Top : Original Input Signal. Bottom : Degraded signal.*

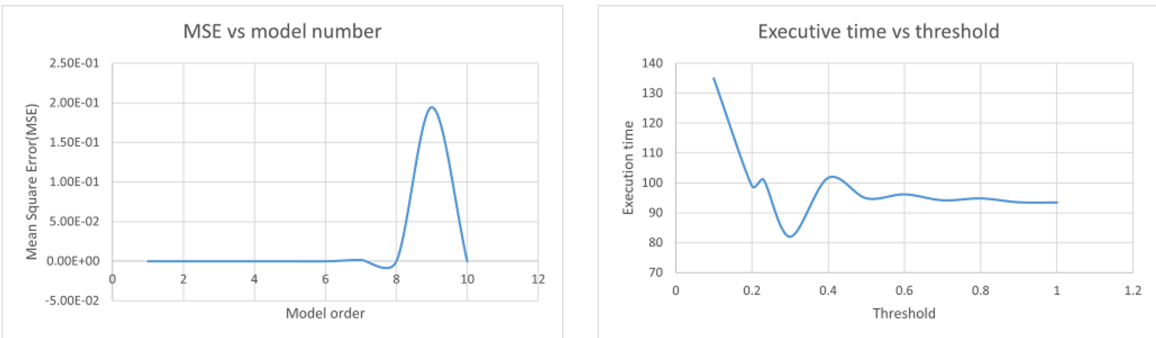

Figure 2: *Left : Figure 2a (MSE vs Model order). Right : Figure 2b (Execution time vs threshold)*

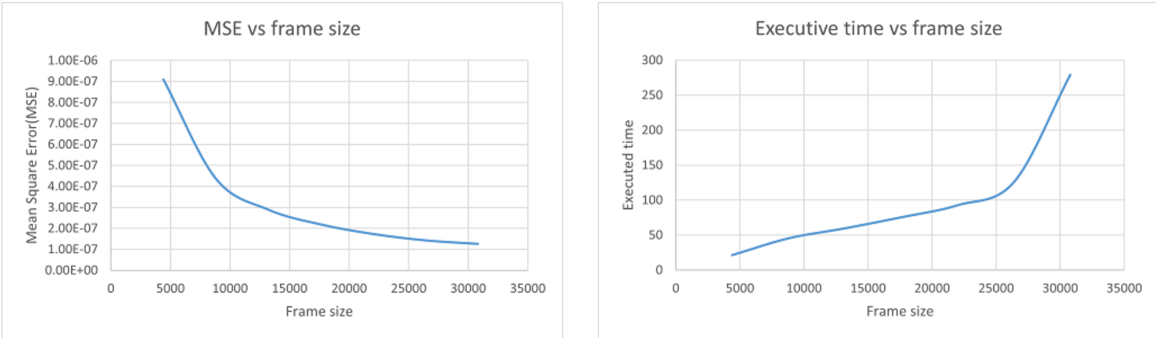

Figure 3: *Left : Figure(3a) (MSE vs frame size). Right : Figure(3b) (Executed time vs frame size)*

3. Frame size is varied, the MSE and Time executed is calculated.

   Frame size is varied by frame duration. As, frame size increases, the MSE keeps on decreasing. However, executed time decreases with increase in frame size

5 Conclusions

Analysing the experiments, we conclude that, we can restore the degraded signal by using Auto Regressive model algorithm. The Algorithm has three parts:

1. Finding AR coefficients : This helps to create A matrix, which is used to find the known and unknown values of the data and their corresponding matrix respectively.

2. Residual : This shows us the clicks in the signal

3. Interpolation : This helps us to replace the clicks by approximation, which is in-turn calculated by seeing the previous data value.

By doing the analysis, we found that the proper input value for the parameters are:
Model order = 6
Threshold = 0.23
Frame duration = 0.5
Frame size = 28,000

After seeing the figure(5), we can conclude that the restored signal is same as the original signal.

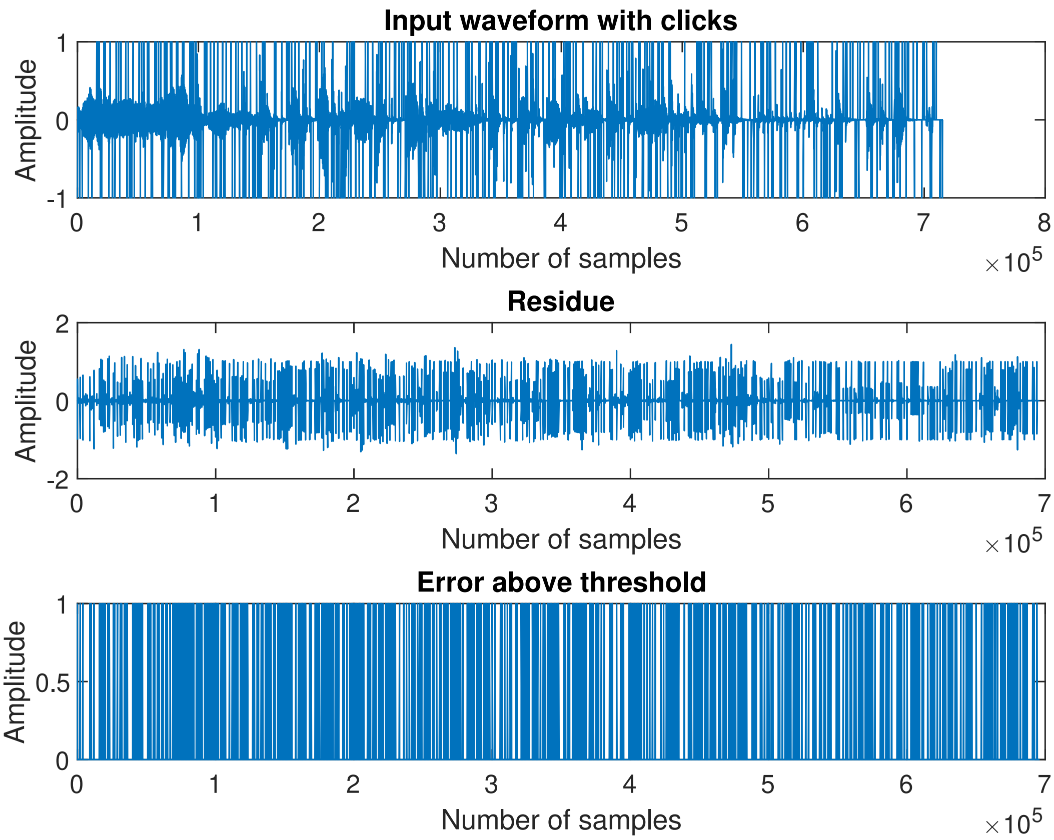

Figure 4: *Top : Degraded signal y_k, Middle: Residual signal Bottom : Error above threshold .*

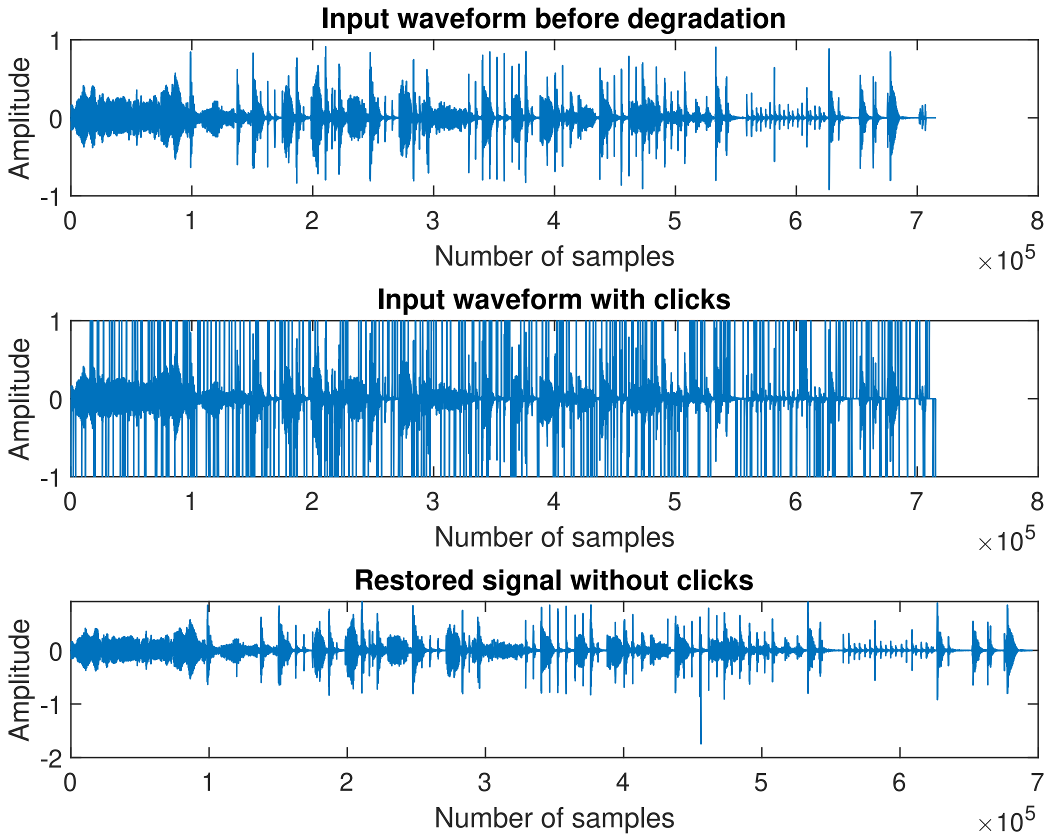

Figure 5: *Top : Original Input Signal. Middle : Degraded signal. Last : Restored signal* s_k

Bibliography

1. Detection and suppression of impulsive noise in speech communication systems by S.V. Vaseghi, PhD and P.J.W. Rayner, PhD.
2. Resources on TCD blackboard by Dr. Angeliki Katsenou.