CS571 Project Report

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Summary

The title of our project is “Linear Predictive Coding of Speech Signals”.

LPC is a very powerful speech analysis technique used to estimate basic speech parameters. It provides extremely accurate estimation of speech parameters.

In LPC, Speech Signals can be approximated as linear combination of past samples by minimizing the error. This has been done by minimizing the sum of squared differences between the actual speech samples and the predicted ones.

Here, in our project we have a speech signal should.wav. At first we have plotted the speech signal for very large samples, then we have divided that speech signal into Hamming window frames. Now for each frames we have performed LPC analysis and plotted DFT and LPC spectrum of it. After this we have created an inverse filter which is an all pole filter which is used for speech synthesis such as the effect of mouth, throat and nasal passage of the speaker on the generated sound.

Using this inverse filter, LPC residual signal is generated. From this residual we have excited the filter and produced a speech frame.

Atlast we have detected the envelope of DFT spectrum and generated that envelope on the spectrum.

Introduction

The main problem of this project is that we have a speech signal should.wav and we have to encode the speech signal using LPC analysis and later decode the same speech signal using the same decoding technique of LPC.

It is useful and most powerful for audio and speech signal processing and also it reduces the bit rate. It is used as a form of voice compression by phone companies for example GSM standard.

Solution

Firstly we have plotted the speech signal for very large samples, then we have divided that speech signal into Hamming window frames. Now for each frames we have performed LPC analysis and plotted DFT and LPC spectrum of it. After this we have created an inverse filter which is an all pole filter which is used for speech synthesis such as the effect of mouth, throat and nasal passage of the speaker on the generated sound.

Using this inverse filter, LPC residual signal is generated. From this residual we have excited the filter and produced a speech frame. Atlast we have detected the envelope of DFT spectrum and generated that envelope on the spectrum.

Assumptions

- Speech signal taken is a linear time-varying filter by some random noise.
- The order of linear filter is an integer value and is greater than 1.
- Speech should not be in ill-condition.

Algorithms used

1) Divide the speech signals into Hamming windowed frames.

2)The following operations are to be done framewise:

3) Perform LPC analysis. Plot the DFT spectrum and LPC spectrum on the same plot.

4) Obtain the inverse filter.

5) Using the inverse filter, obtain the LPC residual signal.

6) Use the LPC residual to excite the LPC filter and produce a (synthetic) speech frame.

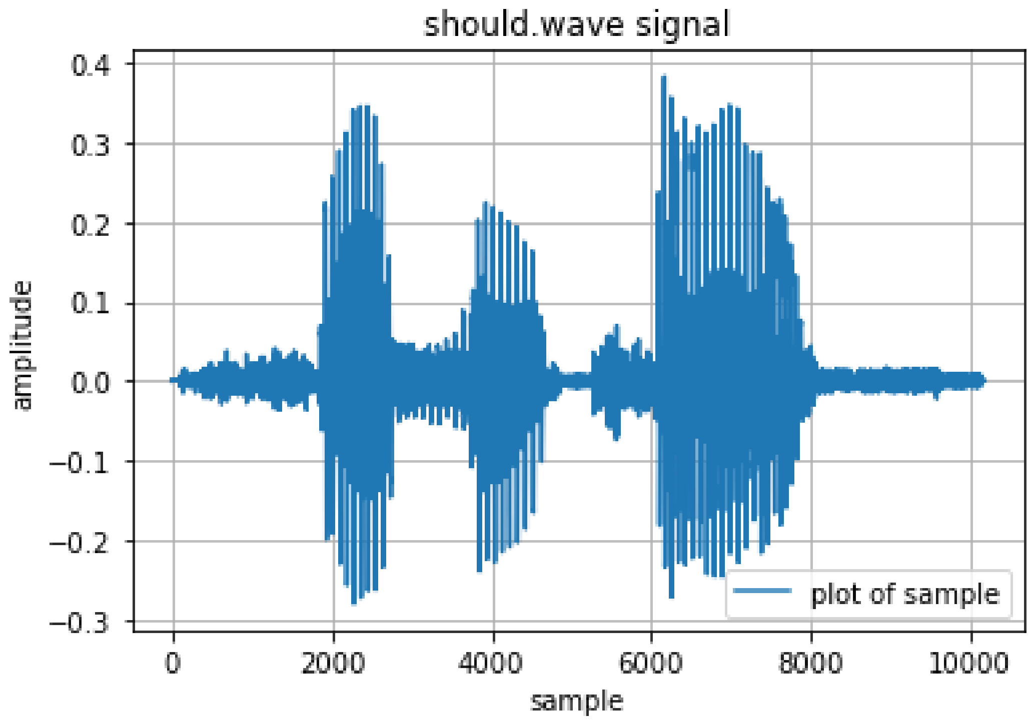
7) Perform overlap-add to obtain back the speech signal. Play back the speech signal.

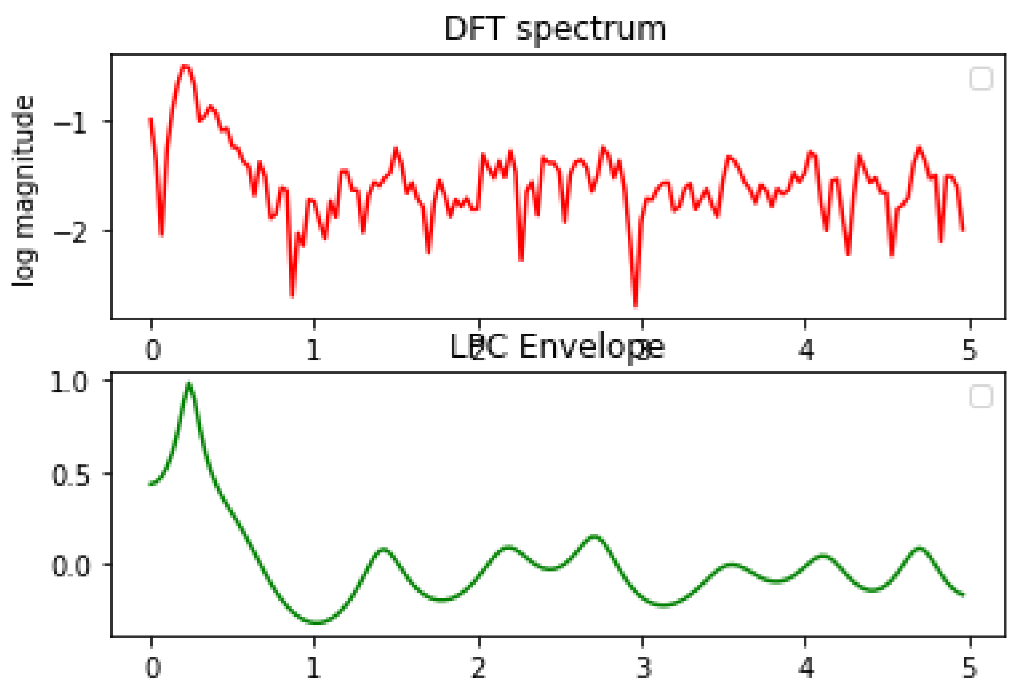
Results and Analysis

Following are the outputs of

1. speech waveform of should.wav signal

2.Plot of DFT and LPC spectrum.

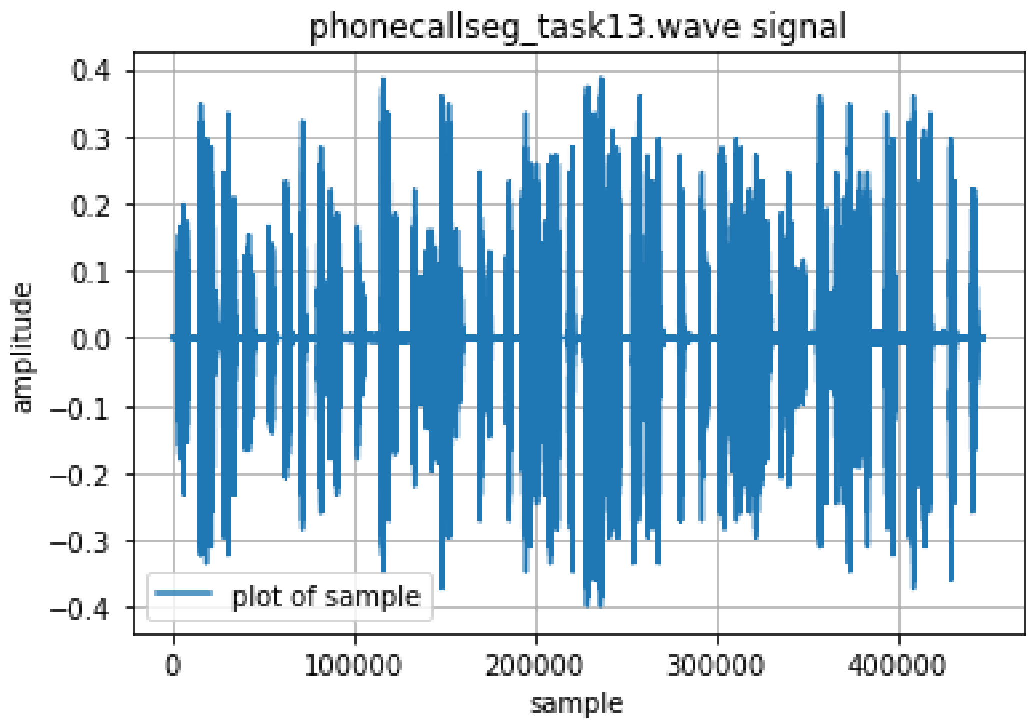

should.wave signal
amplitude
sample
plot of sample

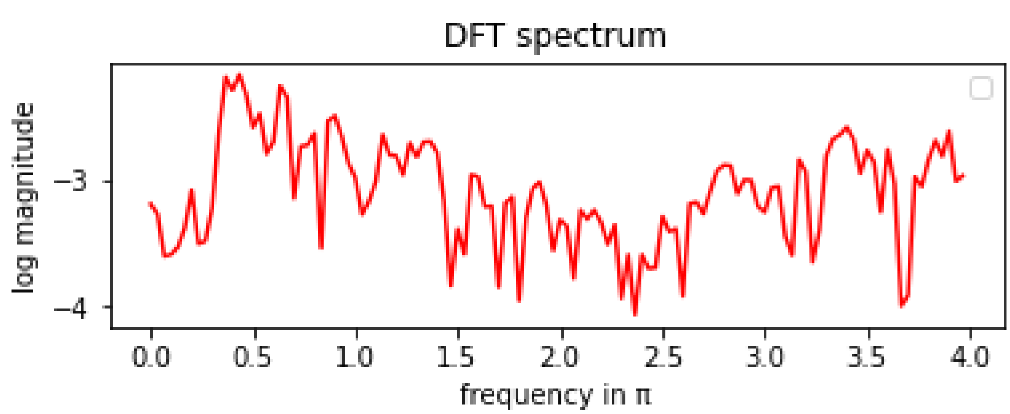

DFT spectrum
log magnitude
LPC Envelope

Following are the outputs of

1. Speech waveform of phonecallseg_task13.wave signal

2 .Plot of DFT Spectrum

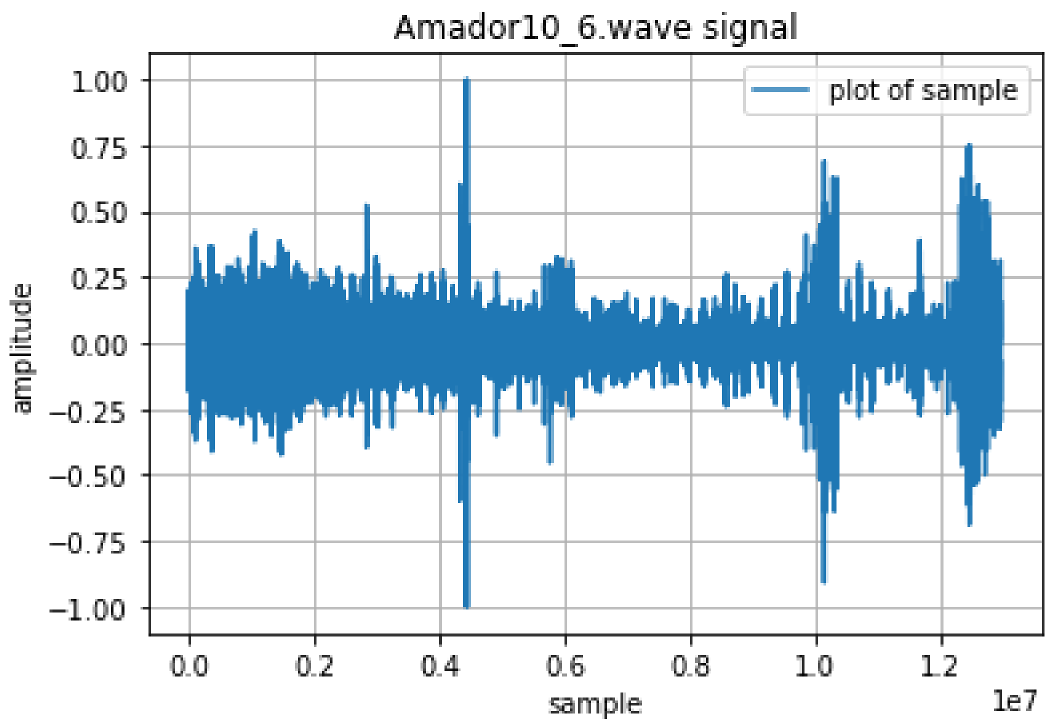


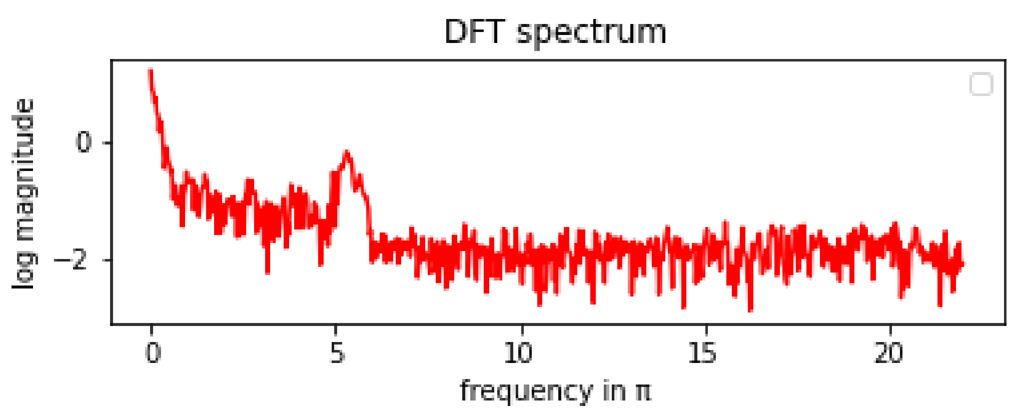


Following are the outputs of

1. Speech waveform of Amador10_6.wav signal

2. Plot of DFT Spectrum





This algorithm fails to execute envelope of speech signals of larger size i.e. higher duration.

Conclusion

- Speech signal is linear time varying.
- Order of the filter is greater than 1.
- Speech is in good condition.
- LPC envelope was generated smoothly for should.wav signal
- Challenge was to plot the LPC envelope of different speech signals.
- Limitation is that we have to take order of filter to be not very large value.

Project GitHub Page

https://github.com/amitkr18/LPC-of-Speech-Signal

References

Textbook:

Theory and Applications of Digital Speech Processing by Rabiner and Schafer, Prentice Hall

YouTube link:

https://www.youtube.com/watch?v=sow15KACJso

Wikipedia:

https://en.wikipedia.org/wiki/LPC