DSP FINAL PROJECT

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Problem statement : causal-anticausal decomposition of speech using complex cepstrum

Intuition:

During the production of voiced sounds, the airflow evicted by the lungs arises in the trachea and causes a quasi-periodic vibration of the vocal folds. This is called glottal flow.Speech signal is basically convolution of signal from Glottal flow i.e. source and an LTI system i.e. the vocal tract system.

We want to separate source and system properties from the speech signal. As convolution involves multiplication in time domain it is not possible to separate simply by looking at the waveforms. So, we convert the multiplication into addition by taking logarithm. This method is called complex cepstrum.

The cepstrum is defined as the inverse DFT of the log magnitude of the DFT of a signal.

Mathematics Involved

★ It is generally accepted that voiced speech results from the excitation of a linear time-invariant system with impulse response h(n) by a periodic pulse train p(n).

★ According to the mechanism of voice production, speech is considered as the result of a glottal flow signal filtered by the vocal tract cavities and radiated by the lips. The system transfer function H(z) then consists of the three following contributions:

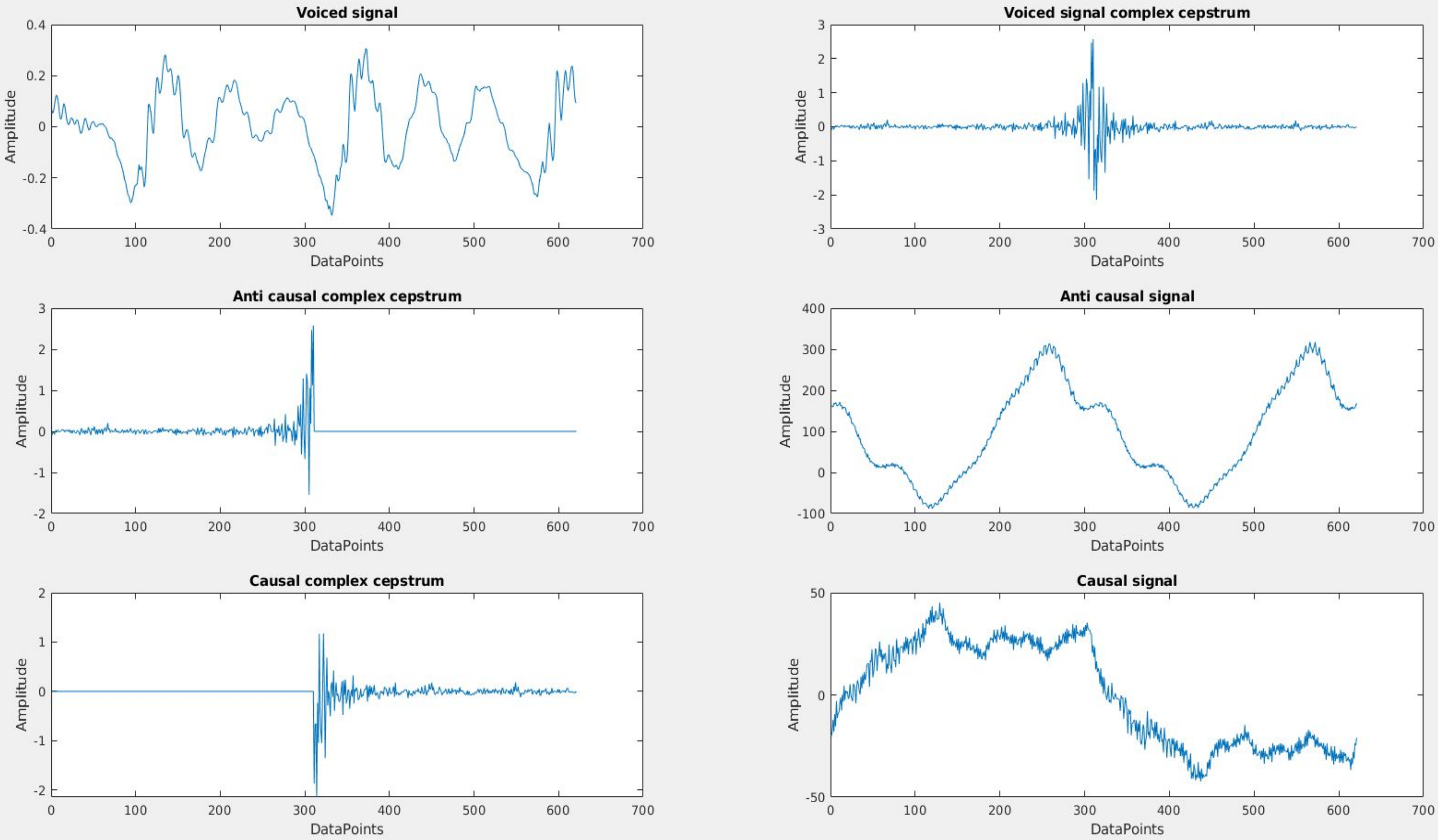
H(z)= A. G(z). V(z). R(z)

where A is the source gain, G(z) the glottal flow over a single cycle, V(z) the vocal tract transmittance and R(z) the radiation load. The resonant vocal tract contribution is generally represented for “pure” vowels by a set of minimum-phase poles.The glottal open phase can be modeled by a pair of maximum-phase (i.e anticausal) poles.

★ Algorithm involves:

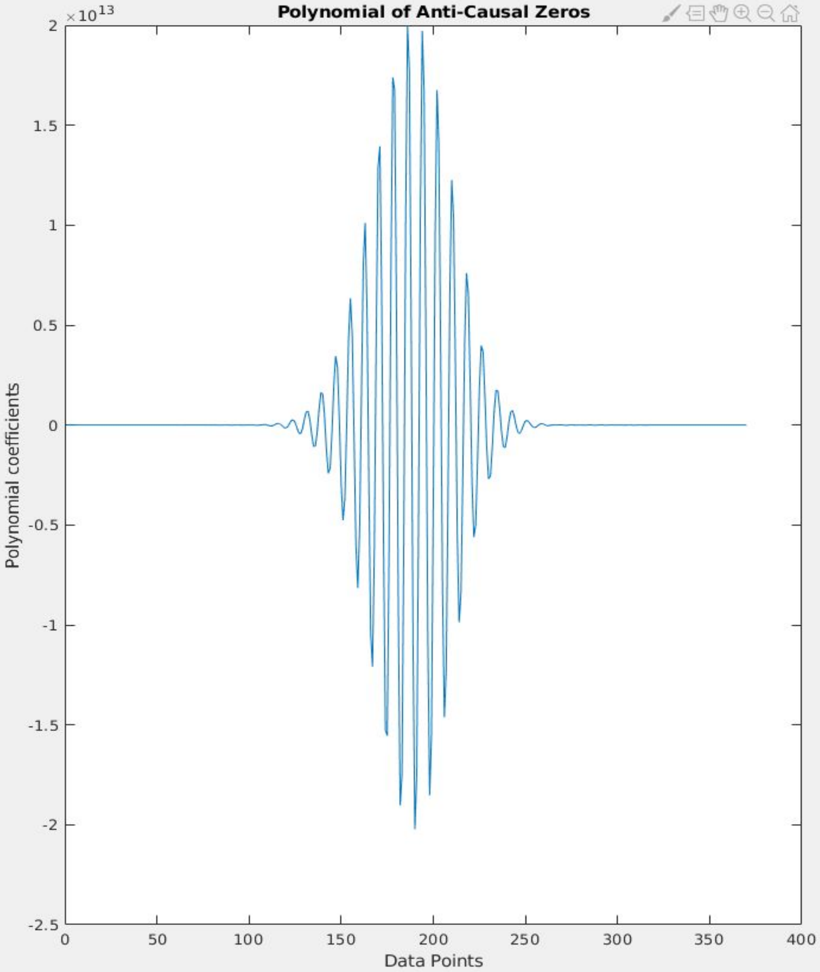
➔ Compute the complex cepstrum of the speech signal.

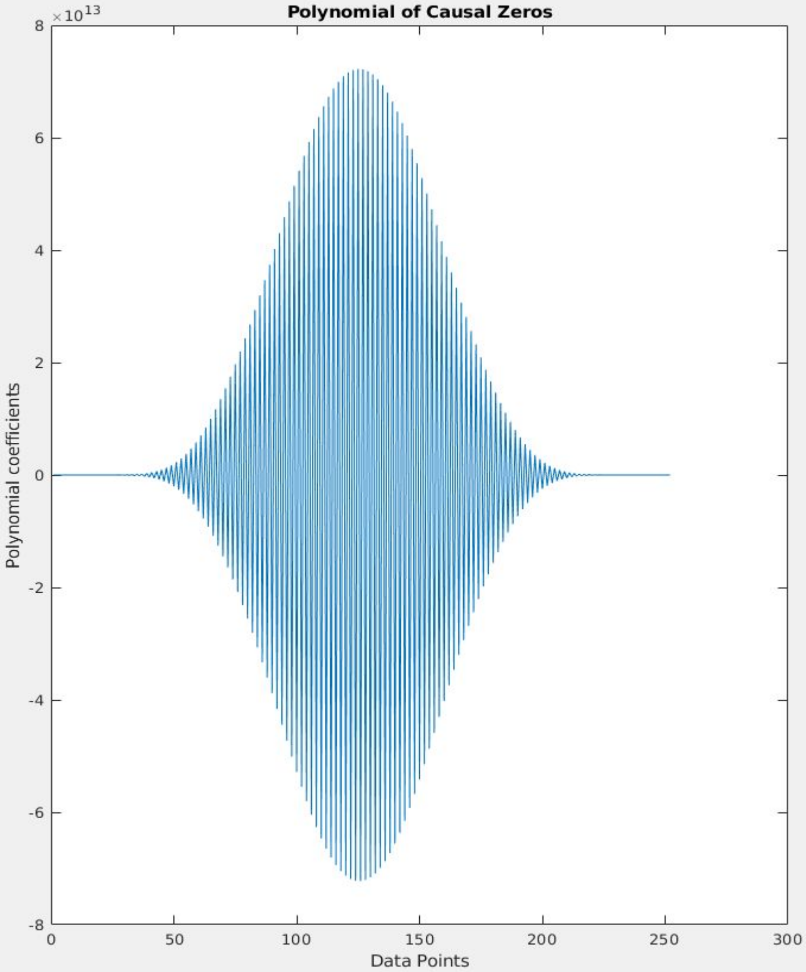
➔ Set the values for n (quefrency) <0 to be zero for causal part and n > 0 to be zero for anti-causal part and then compute inverse complex cepstrum.


Voiced signal
Voiced signal complex cepstrum
Anti causal complex cepstrum
Anti causal signal
Causal complex cepstrum
Causal signal
Amplitude
DataPoints

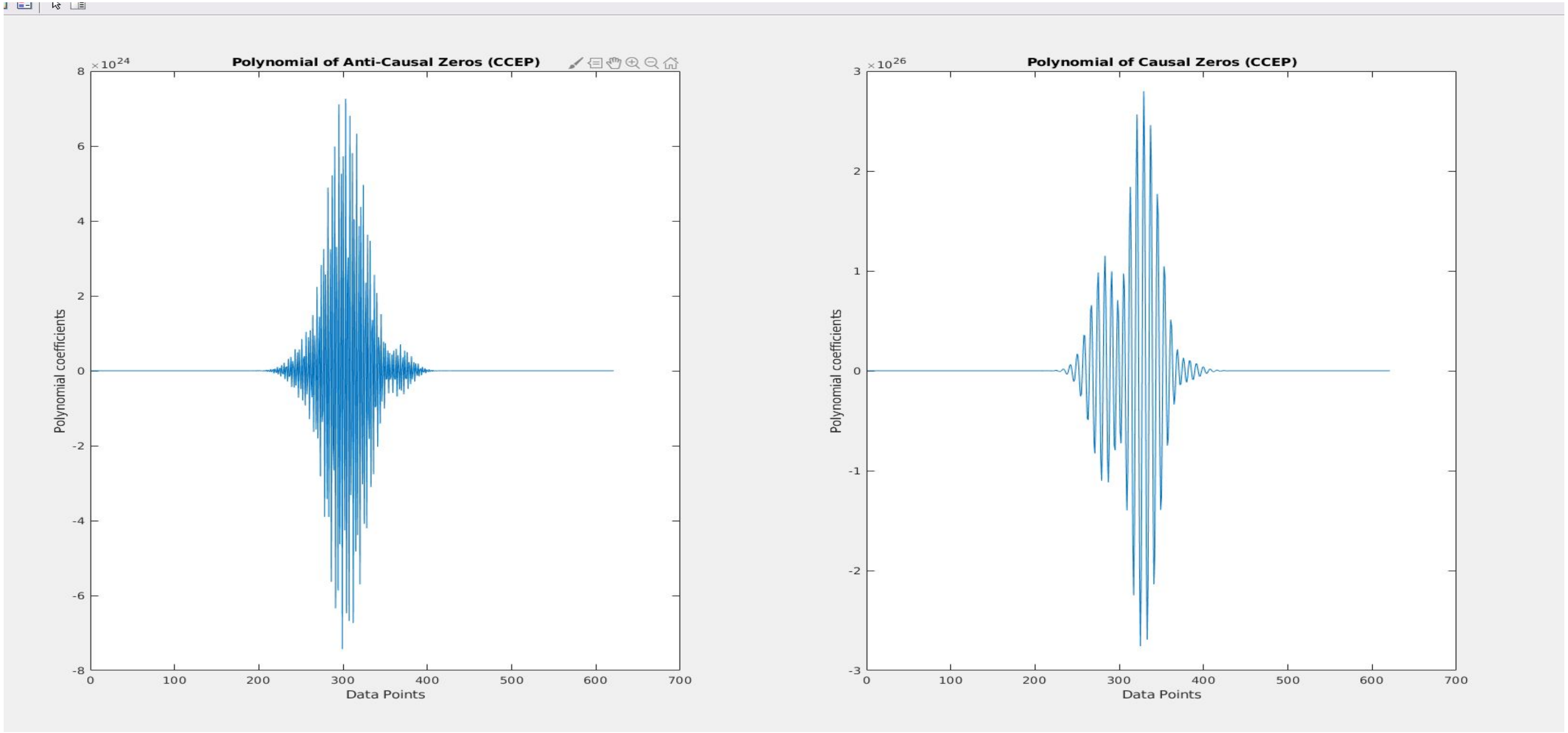
In Final Evaluation : zeros of z transform

- ★ A MATLAB simulation for the above given algorithm has to be done to check it’s validity.
- ★ One more method can be employed to do this decomposition- Zeros of Z Transform. It involves the following steps:
- ➔ Compute the zeros of the speech signal.
- ➔ Isolate the roots with modulus greater than 1 (Maximum phase or anti-causal system).
- ➔ Compute the inverse fourier transform from this.
- ★ However, the ZZT method suffers from high computational load due to the necessity of factorizing large degree polynomials.
- ★ Our aim is to implement this method in MATLAB as well.


Polynomial of Anti-Causal Zeros
×10^13
Polynomial coefficients
Data Points
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
-2.5
0
50
100
150
200
250
300
350
400


Polynomial of Causal Zeros
×10^13
Polynomial coefficients
Data Points
8
6
4
2
0
-2
-4
-6
-8
0
50
100
150
200
250
300

Zeros from complex cepstrum output



Observation

We can see the separated and anticausal part.

Anti Casual part is periodic with period ≅ 10 ms

We have computed zeros of causal and anticausal part using Complex Cepstrum method and Zeros of Z transform method. We found them almost same.

References

★ Causal-anticausal decomposition of speech using complex cepstrum for glottal source estimation by Thomas Drugman, Baris Bozkurt and Thierry Dutoit.