

Linear predictive analysis and synthesis

Consider the speech signal in “machali.wav”, sampled at 8 kHz. Consider the following signal segments in the final word “pani”: (1) /a/ (first half); (2) /n/; (3) /l/; and (4) /s/ in the word “uska”.

Use PRAAT to extract the above segments to separate .wav files for further analyses as below.

1. Compute the narrowband spectrum using a Hamming window of duration = 30 ms before and after pre-emphasis.
2. Using a 30 ms Hamming window centered in the pre-emphasized waveform:
 - (a) Compute the autocorrelation coefficients required for LPC calculation at various $p = 4, 6, 8, 10, 12, 20$. Use the Levinson algorithm to compute the LP coefficients from the autocorrelation coefficients. Show the pole-zero plots of the estimated all-pole filter for $p=6, 10$.
 - (b) Compute the gain and plot the LPC spectrum magnitude (i.e. the dB magnitude frequency response of the estimated all-pole filter) for each order “p”. Superimpose each plot on the narrowband dB magnitude spectrum of part 1. Comment on the characteristics of the spectra.
 - (c) Plot error signal energy (i.e. square of gain) vs p.
3. Based on the 10th-order LPCs, carry out the inverse filtering of one of the vowel segments and of the unvoiced sound. Obtain the residual error signal. Can you measure the pitch period of the voiced sound from the residual waveform? Observe the magnitude spectrum of the residual signal.
4. Next, we wish to resynthesize the phone sounds from the parameters of the source-filter model obtained above for $p=10$: pitch, gain, LP coefficients. Use the LP filter estimated above, and an ideal impulse train input as source excitation (for the voiced sounds). Carry out de-emphasis. For the unvoiced sound, use a white noise signal as source excitation. Set the duration of the synthesized sound to be 300 ms at 8 kHz sampling frequency, and view/listen to your created sound.

Make a single document presenting your method with relevant code fragments, results and discussion.

