Department of Communications and Computer Engineering
CCE5223 - Speech Processing and Coding

Tutorial 2
Acoustic Parameters

1. Speech over telephone lines is limited to the 300 – 3300 Hz frequency band. What phonemes are distorted most? Explain, giving examples of confusions that would be expected among words over the telephone.
2. Consider filtering speech with a bandpass filter, eliminating all energy below X Hz and above Y Hz,
   (a) What is the smallest range of frequencies, (X,Y) Hz that would allow all (English) phonemes to be distinguished. Explain.
   (b) If X = 1 kHz and Y = 2 kHz, explain which phonemes will be most confused with one another.

3. Consider a time window for speech analysis.
   (a) What are the advantages and disadvantages of short and long windows?
   (b) To what type of filter should the spectrum of a window correspond?
   (c) Explain how the bandwidth of an analysis window affects spectrographic estimation of formants and F0.

4. Consider a pitch detection scheme that lowpass filters the speech to 900Hz and then calculates an autocorrelation function.
   (a) Why is the speech first lowpass filtered?
   (b) How is the autocorrelation function used to generate a pitch estimate.
   (c) For as sampling rate of 16,000 samples per minute suggest a suitable range of values of displacement τ in samples to obtain the pitch value.

5. Consider a steady vowel with formants at 500, 1500, 2500, .. Hz lowpass filtered to 4000Hz., and then sampled at the Nyquist rate.
   (a) Draw a detailed block diagram of a system to generate a good version of this already sampled signal at 10,000 samples per second.
   (b) Within the range $|\omega| < \pi$, at which "digital frequencies" ω_k should the formants be for the 10,000 samples /sec signal.

6. (a) Contrast the average magnitude difference method of calculating the pitch with the autocorrelation method.
   (b) Suggest a method of obtaining the pitch using the frequency domain