Answer the following questions:

Question 1: (Answer two points only)   (5 Marks for each point)

a- Explain a speech watermarking algorithm. What is the importance of the speech watermarking process?

b- Derive a mathematical expression for the adaptive Wiener filter applied for speech enhancement.

c- Derive the digital filter transfer functions for the Haar wavelet transform. Explain the wavelet denoising process of speech signals.

Question 2: (Answer two points only)   (5 Marks for each point)

a- Explain the basic idea of the neural classifier used for automatic speaker identification.

b- Explain how the convergence parameter of an adaptive filter is estimated. What is the effect of this parameter on the convergence speech of the adaptive algorithm?

c- Explain the basic idea of chaotic encryption.

Question 3: (Answer two points only)   (5 Marks for each point)

a- Is it possible to use the comb filter instead of the mel-frequency filter bank for speaker identification? Why?

b- Explain how the multi-level security concept can be applied on speech signals.

c- What is the difference between the adaptive Wiener filter used for speech enhancement and the adaptive LMS filter?

Question 4: (Answer two points only)   (5 Marks for each point)

a- Write the mathematical formulation of the singular value decomposition of a matrix. Explain why this decomposition is suitable for speech watermarking.

b- Explain a speech encryption algorithm. What is the importance of the speech encryption process?

c- Explain the basic idea of speech deconvolution. Compare between the different speech denconvolution algorithms.

Question 5: (Answer two points only)   (5 Marks for each point)

a- Is it possible to use signal separation instead of enhancement algorithms in the cases of noisy speech signals? Why?

b- Compare between spectral subtraction and Wiener filtering techniques for speech enhancement.

c- Explain a strategy to enhance the detectability of watermarks in speech watermarking algorithms.

Question 6: (Answer two points only)   (5 Marks for each point)

a- Show mathematically how the MFCCs and polynomial coefficients are estimated.

b- Show mathematically how the blind signal separation is applied to speech signals for quality enhancement.

c- What is the objective of estimating cepstral coefficients from transform domains for speaker identification?

Question 7:   (Answer two points only)   (5 Marks for each point)

a- Discuss the symmetry conditions required to build linear phase filters.
b- Define the comb filter. For order 8 comb filter, sketch the magnitude and phase response of the filter. What is the important applications of this filter?
c- Explain the basic idea of adaptive LMS filters. Give two different applications for these filters.

Best Regards
Fathi E. Abd El-Samie