

Linear Prediction Analysis of Speech

The slide features a white background with the title 'Linear Prediction Analysis of Speech' in a blue sans-serif font. At the bottom, there are decorative blue elements: a horizontal bar on the left with a wavy, layered effect, and a large, solid blue shape on the right that resembles a stylized wave or a speech bubble tail.

Linear Prediction Analysis of Speech

- **LPC provides good model of speech**
 - ✓ All pole model of LPC provides good approximation to vocal-tract spectral envelope for quasi-steady state voiced regions of speech.
 - ✓ Less effective for unvoiced and transition regions of speech
- **LPC analysis leads source & vocal tract separation**
 - ✓ LP synthesis model (Generative model)
 - ✓ LP analysis model (inverse filter model)
- **LPC is analytically tractable model**
 - ✓ Mathematically precise
 - ✓ Straightforward to implement either S/W or H/W
 - ✓ Computationally efficient over bank of filters
- **LPC model works well in recognition applications**

The LPC Model

$$S(n) \approx a_1 s(n-1) + a_2 s(n-2) + \dots + a_p s(n-p) \quad (1)$$

$$S(n) = \sum_{i=1}^p a_i s(n-i) + G u(n) \quad (2)$$

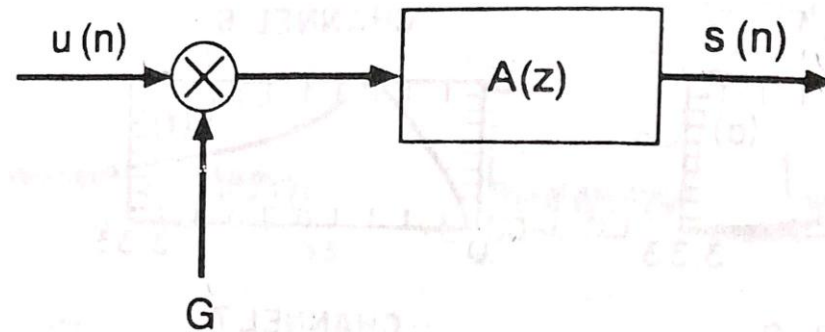
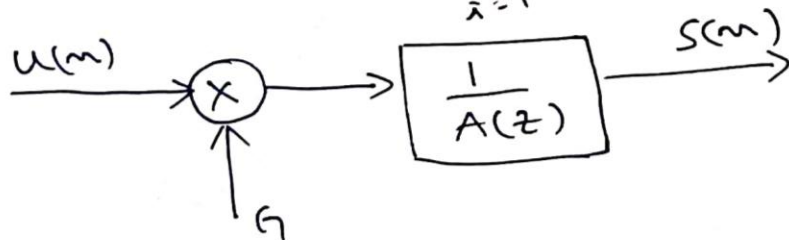
$u(n)$ — Normalized Excitation

G — Gain of the Excitation

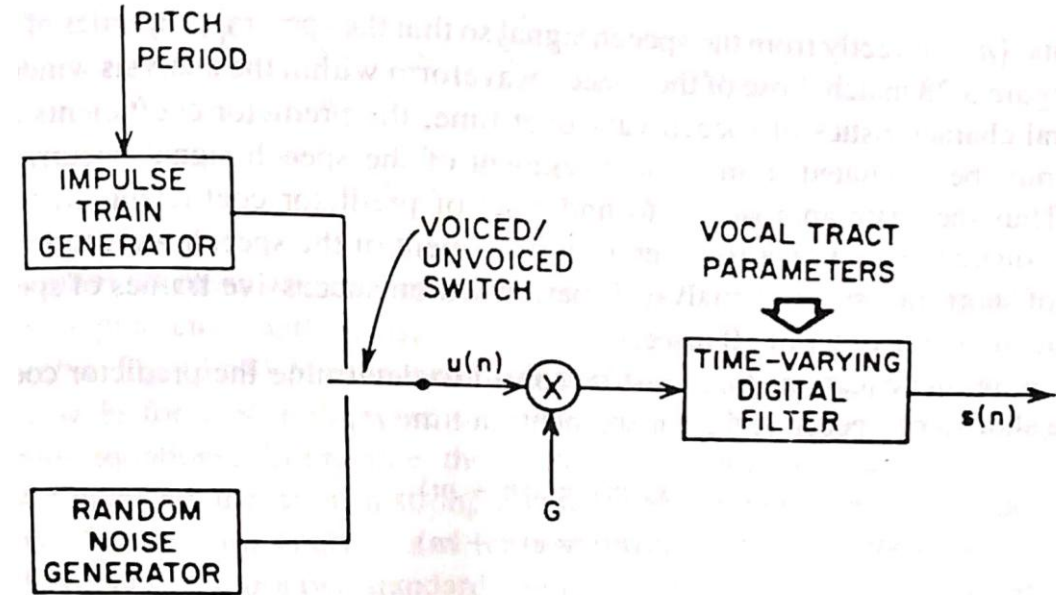
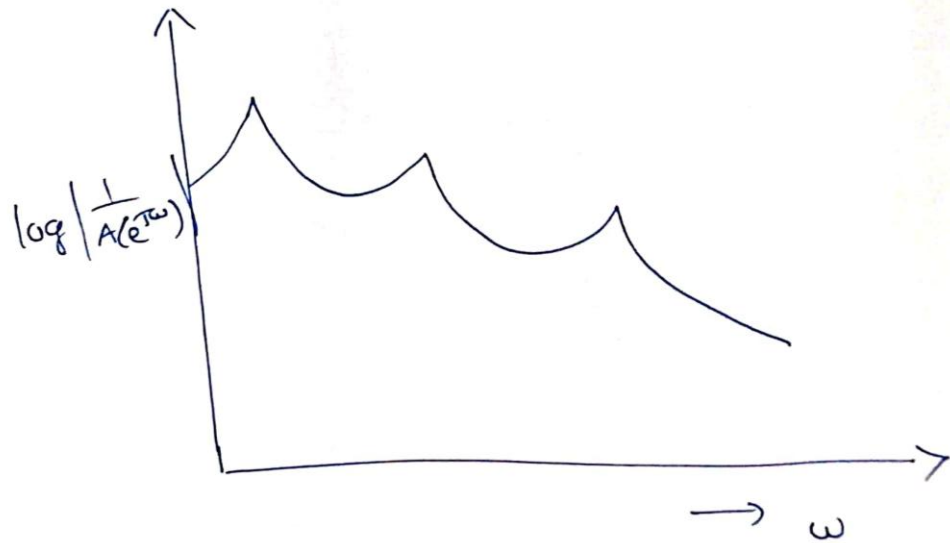
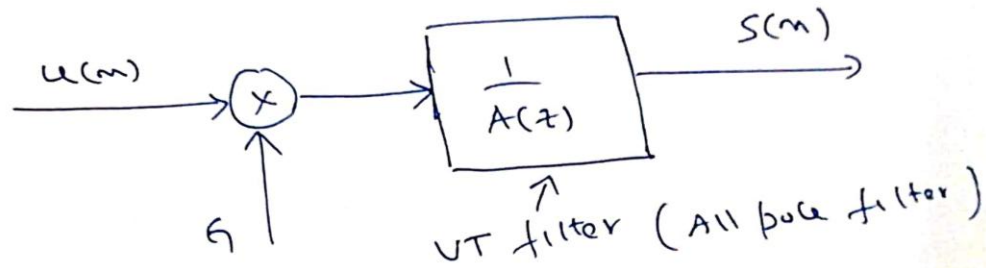
Applying z -transform on eq - (2)

$$S(z) = \sum_{i=1}^p a_i z^{-i} S(z) + G u(z)$$

$$H(z) = \frac{S(z)}{G u(z)} = \frac{1}{1 - \sum_{i=1}^p a_i z^{-i}} = \frac{1}{A(z)}$$



Speech Synthesis Model Based on LPC Model



Speech Analysis Model Based on LPC Model

Speech Analysis Model $\hat{S}(m) = \sum_{i=1}^p a_i s(m-i)$

$\hat{S}(m)$ — predicted signal

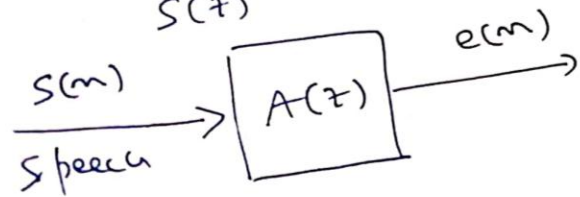
$S(m)$ — Actual signal

$e(m)$ — Error in prediction

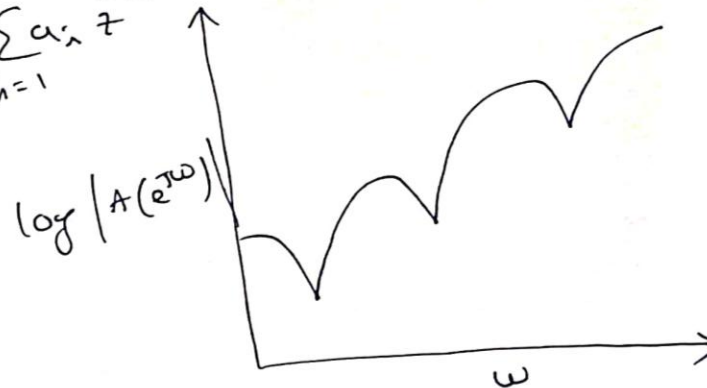
$$e(m) = S(m) - \hat{S}(m) = S(m) - \sum_{i=1}^p a_i s(m-i)$$

$$E(z) = S(z) \left[1 - \sum_{i=1}^p a_i z^{-i} \right]$$

$$\frac{E(z)}{S(z)} = A(z) = 1 - \sum_{i=1}^p a_i z^{-i}$$



Inverse Filter formulation



LP Analysis of Speech : Computation of LPCs

LP analysis of Speech (computation of LPCs)

$$\hat{S}(m) \text{ — predicted speech signal} = \sum_{k=1}^p a_k s(m-k)$$

$S(m)$ — Actual speech signal

$e(m)$ — Error in prediction of m^{th} sample

$$e(m) = S(m) - \hat{S}(m) = S(m) - \sum_{k=1}^p a_k s(m-k)$$

$$E = \sum_{m=1}^N e^2(m) = \text{Squared error for one frame of } N \text{ samples}$$
$$= \sum_{m=1}^N \left[S(m) - \sum_{k=1}^p a_k s(m-k) \right]^2$$

For optimal values of a_k , $\frac{\partial E}{\partial a_k} = 0 \quad (k=1, 2, \dots, p)$

$$\sum_{k=1}^p a_k \sum_n s(m-k) s(m-i) = - \sum_n s(m) s(m-i) \quad 1 \leq i \leq p$$

LP Analysis of Speech : Computation of LPCs

$$\sum_{k=1}^p a_k R(i-k) = R(i) \quad i = 1, 2, \dots, p$$

$$\begin{bmatrix} R(0) & R(1) & R(2) & \dots & R(p-1) \\ R(1) & R(0) & R(1) & \dots & R(p-2) \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ R(p-1) & R(p-2) & R(p-3) & \dots & R(0) \end{bmatrix} \begin{bmatrix} a_1 \\ a_2 \\ \vdots \\ a_p \end{bmatrix} = \begin{bmatrix} R(1) \\ R(2) \\ \vdots \\ R(p) \end{bmatrix}$$

Gauss Reduction Method $\begin{cases} \frac{p^3}{3} + o(p^2) & \text{— operations} \\ p^2 & \text{— storage} \end{cases}$

Cholesky decomposition $\begin{cases} \frac{p^3}{6} + o(p^2) \\ p^2/2 \end{cases}$

Levinson & Robinson ~~Method~~ Recursion

Durbin Recursion $\begin{cases} p^2 + o(p) \\ 2p \end{cases}$

Error in LP Analysis of Speech

Total Error

$$E = \sum_n \tilde{e}^2(m) = \sum_n [s(m) - \hat{s}(m)]^2$$

$$E = \sum_n \left[s(m) - \sum_{k=1}^p a_k s(m-k) \right]^2$$

$$= \sum_n \left[s^2(m) + \left(\sum_{k=1}^p a_k s(m-k) \right)^2 - 2 \sum_{k=1}^p a_k s(m-k) s(m) \right]$$

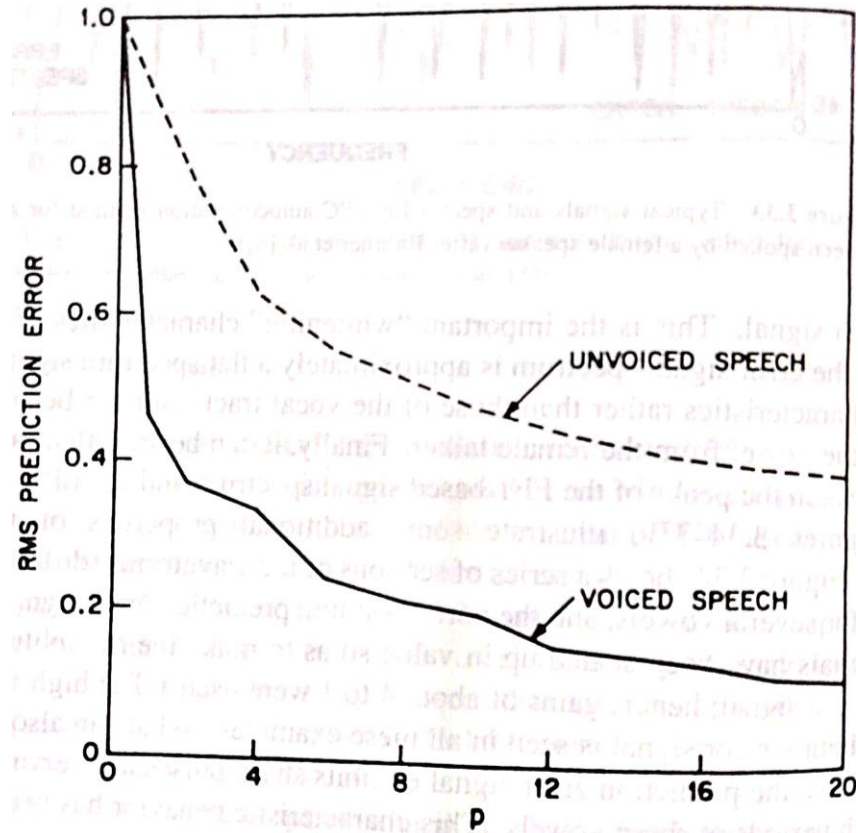
$$= \sum_n s^2(m) + \sum_n \sum_{k=1}^p (a_k s(m-k))^2 - 2 \sum_{k=1}^p a_k \sum_n s(m) \cdot s(m-k)$$

Substituting the result of p normal equations

$$\sum_{k=1}^p a_k \sum_n s(m+k) s(m-k) = - \sum_n s(m) s(m-k) \quad 1 \leq k \leq p$$

$$E = \sum_n s^2(m) + \sum_{k=1}^p a_k \sum_n s(m) s(m-k)$$

$$= R(0) - \sum_{k=1}^p a_k R(k)$$



LP Analysis of Vowel 'I'

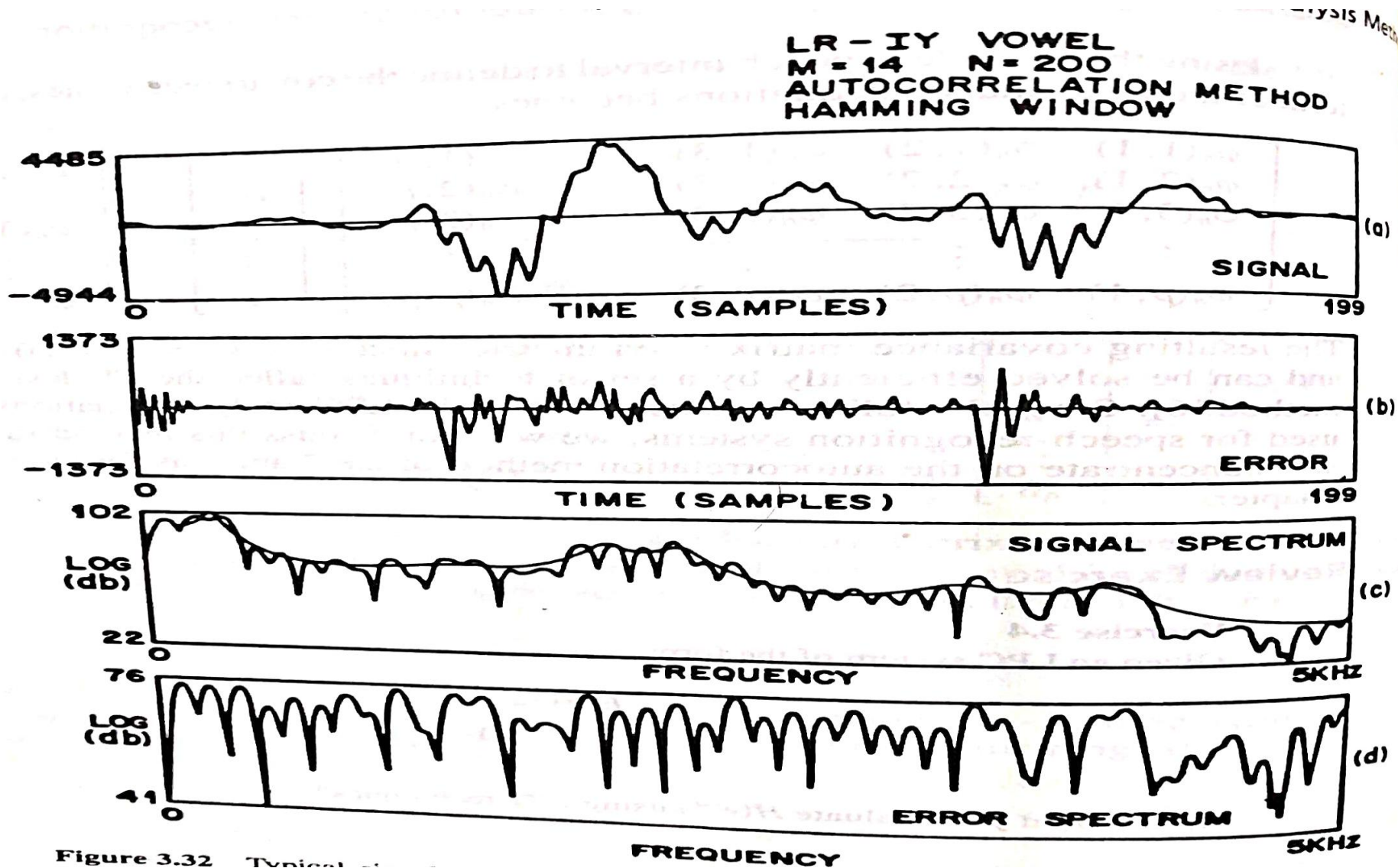
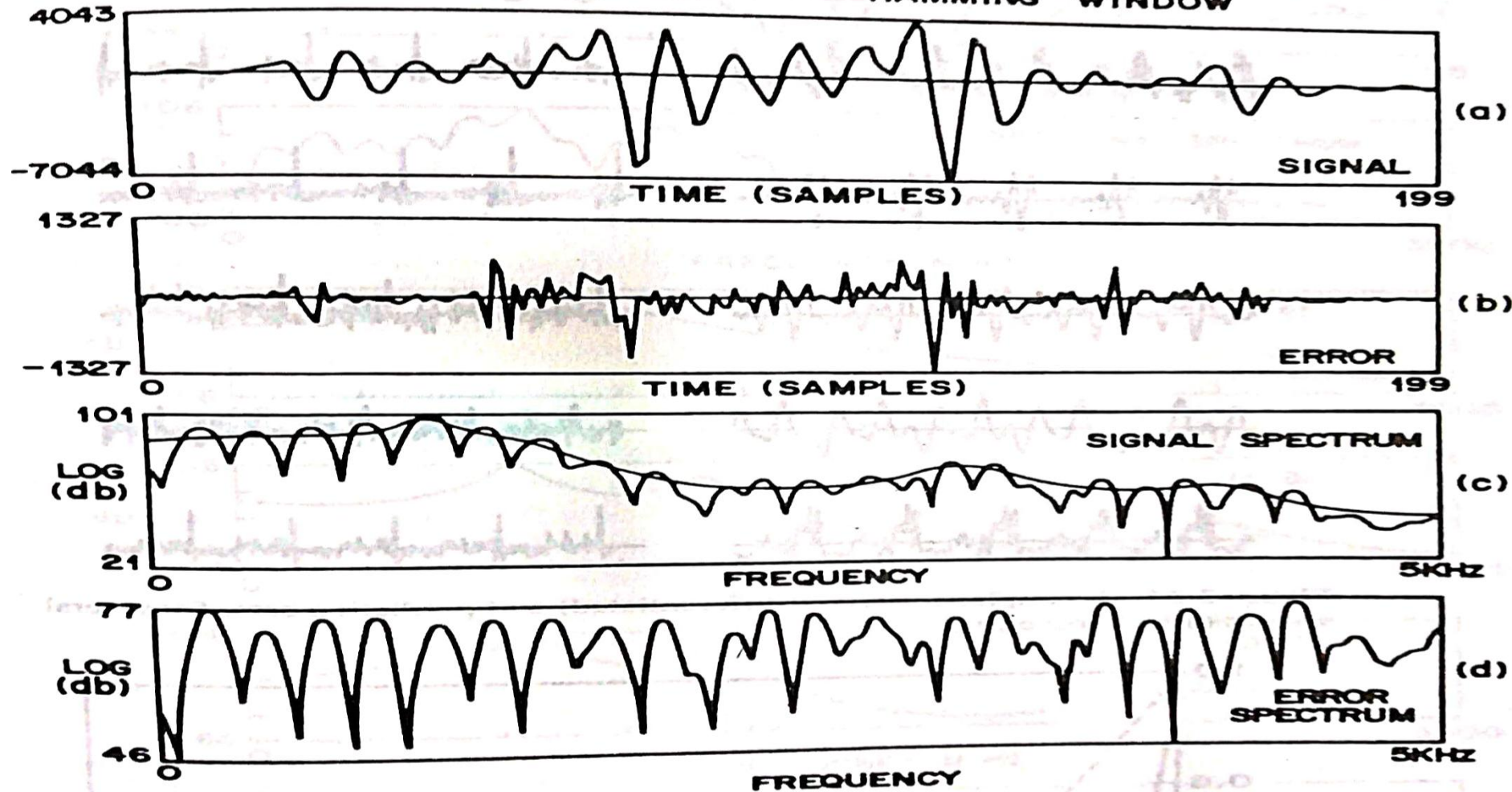


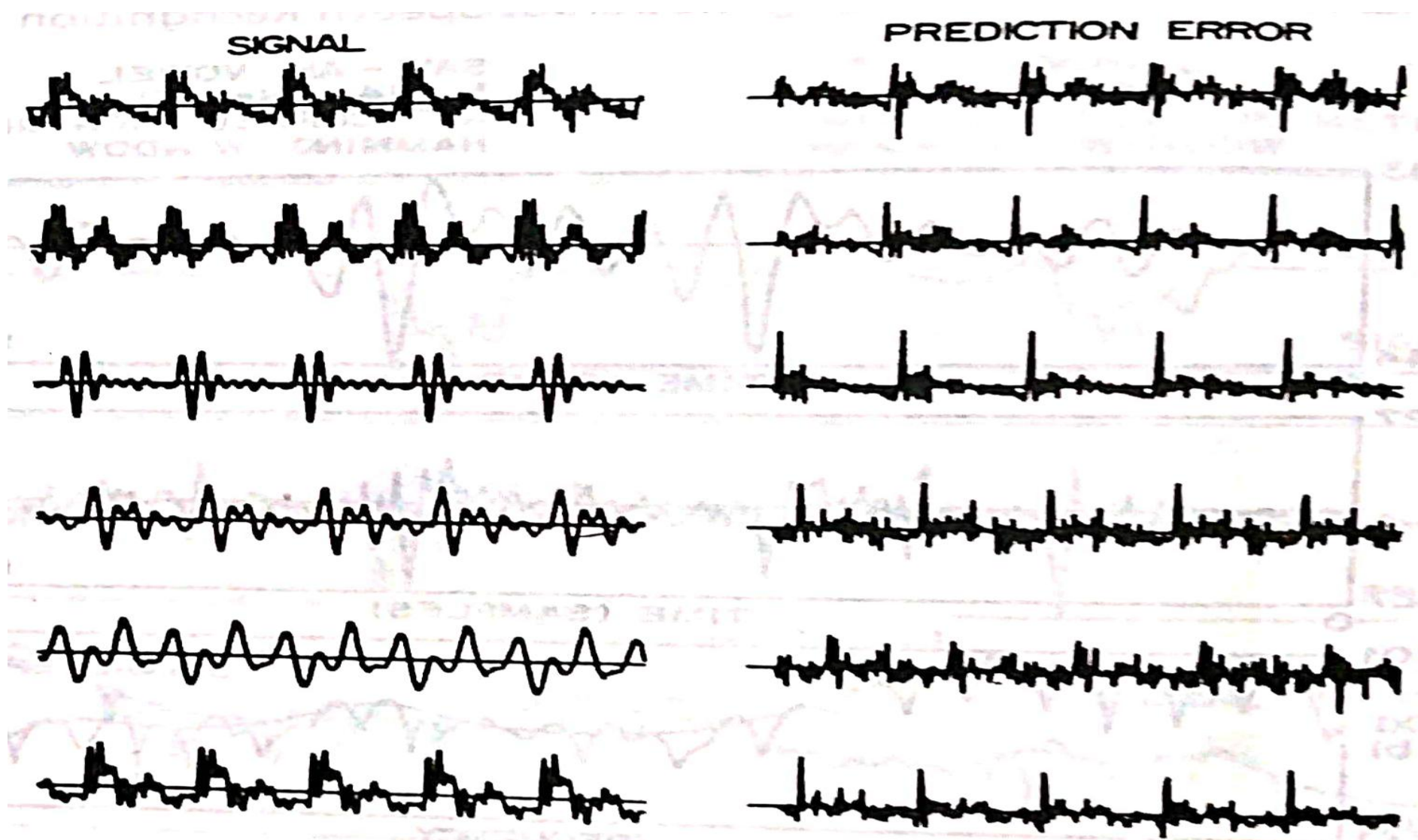
Figure 3.32 Typical LP analysis results for the vowel 'I'.

LP Analysis of Vowel 'A'

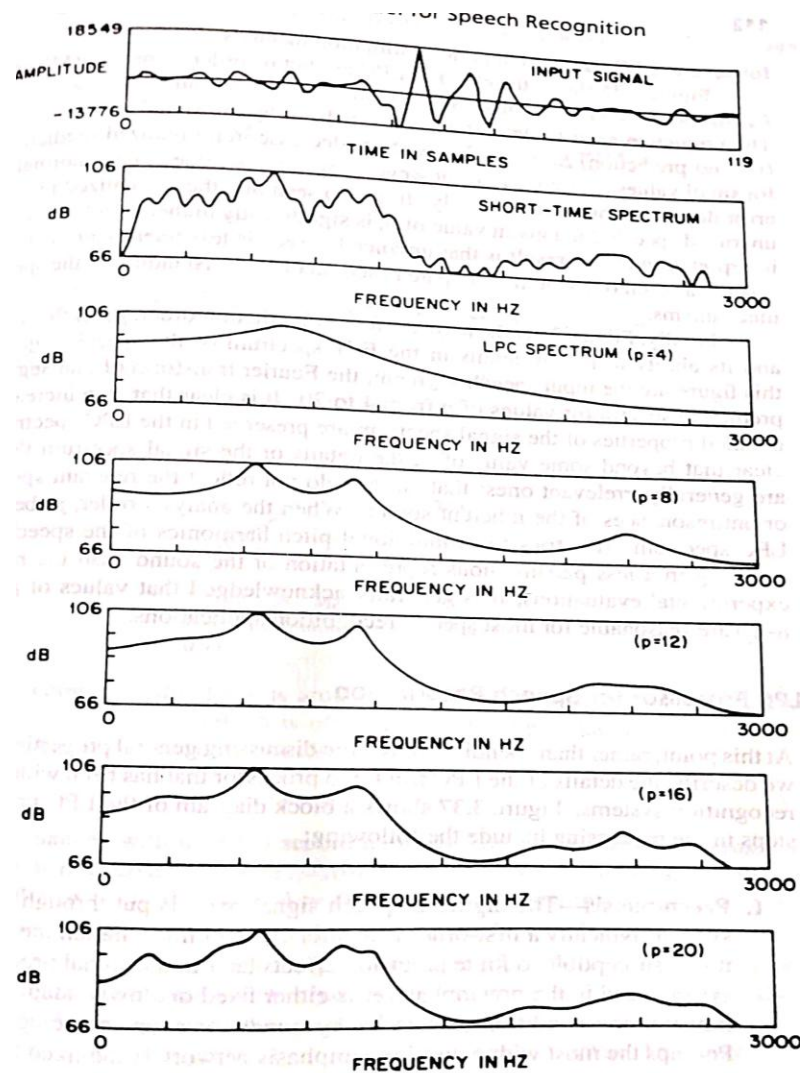
SAW - AH VOWEL
M = 14 N = 200
AUTOCORRELATION METHOD
HAMMING WINDOW



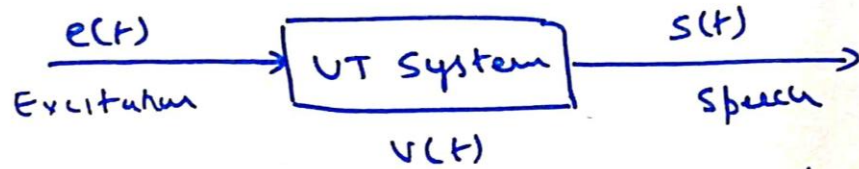
LP Analysis of Vowels



LP Analysis of a Vowel for different P values



Cepstral Analysis of Speech



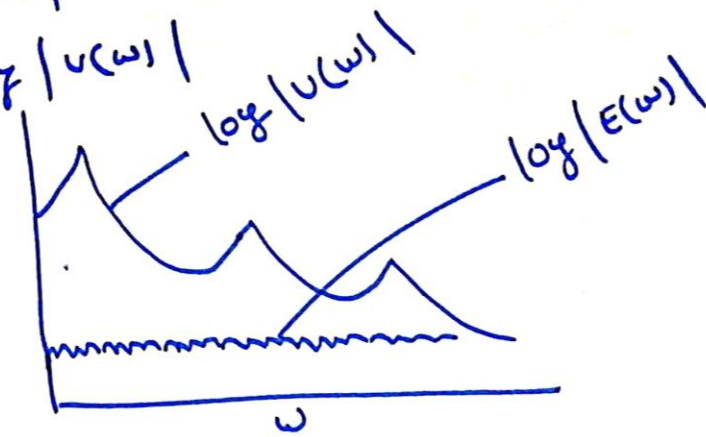
$$s(t) = e(t) * v(t); \quad S(\omega) = |S(\omega)| e^{j\phi_s(\omega)}$$

$$S(\omega) = E(\omega) V(\omega); \quad E(\omega) = |E(\omega)| e^{j\phi_e(\omega)}$$

$$\log S(\omega) = \log E(\omega) + \log V(\omega) \quad V(\omega) = |V(\omega)| e^{j\phi_v(\omega)}$$

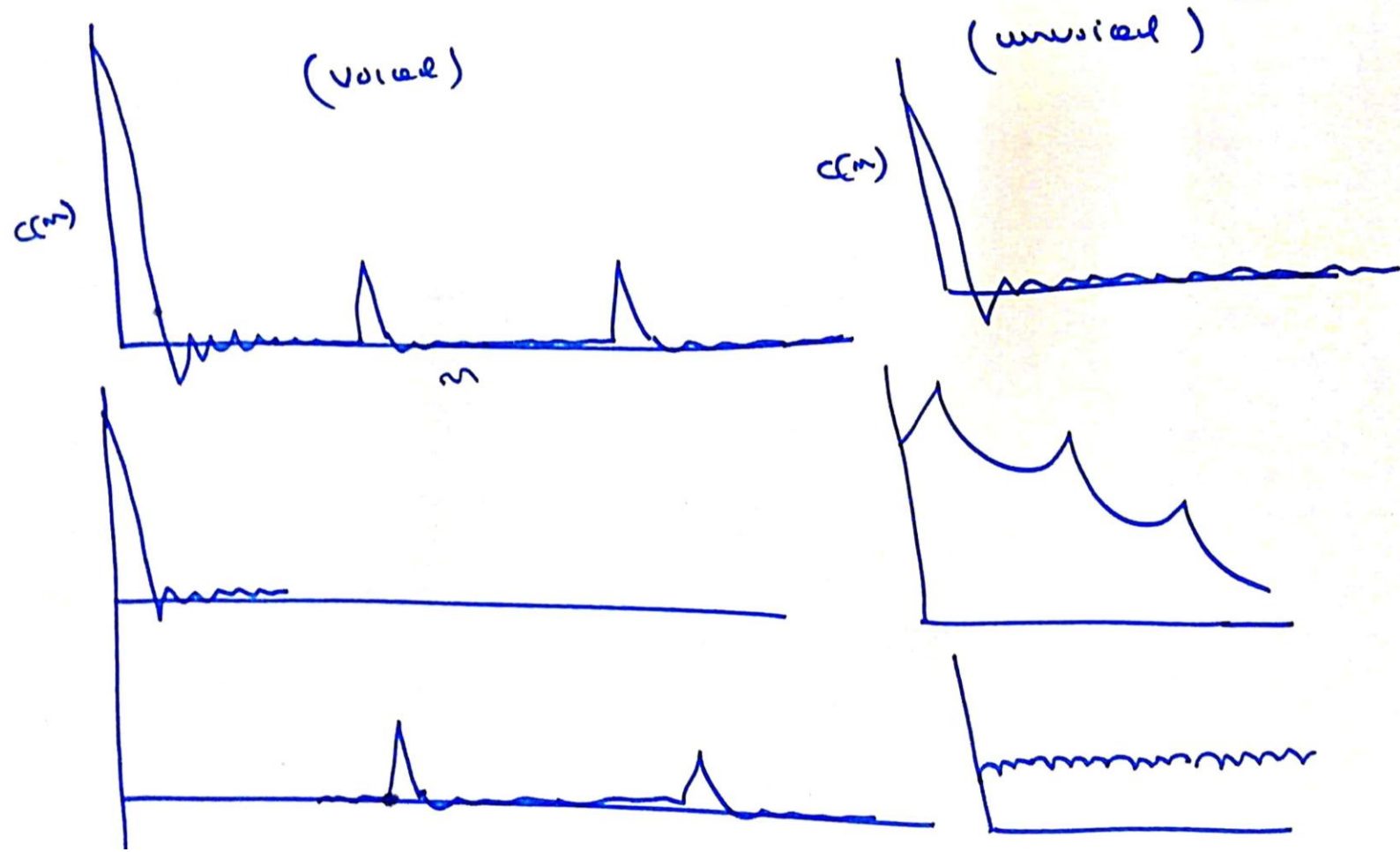
$$\log |S(\omega)| + \phi_s(\omega) = \log |E(\omega)| + \phi_e(\omega) + \log |V(\omega)| + \phi_v(\omega)$$

$$\log |S(\omega)| = \log |E(\omega)| + \log |V(\omega)|$$



Cepstral Analysis of Speech

$$\text{IFT} [\log |S(\omega)|] = \text{IFT} [\log |E(\omega)|] + \text{IFT} [\log |U(\omega)|]$$



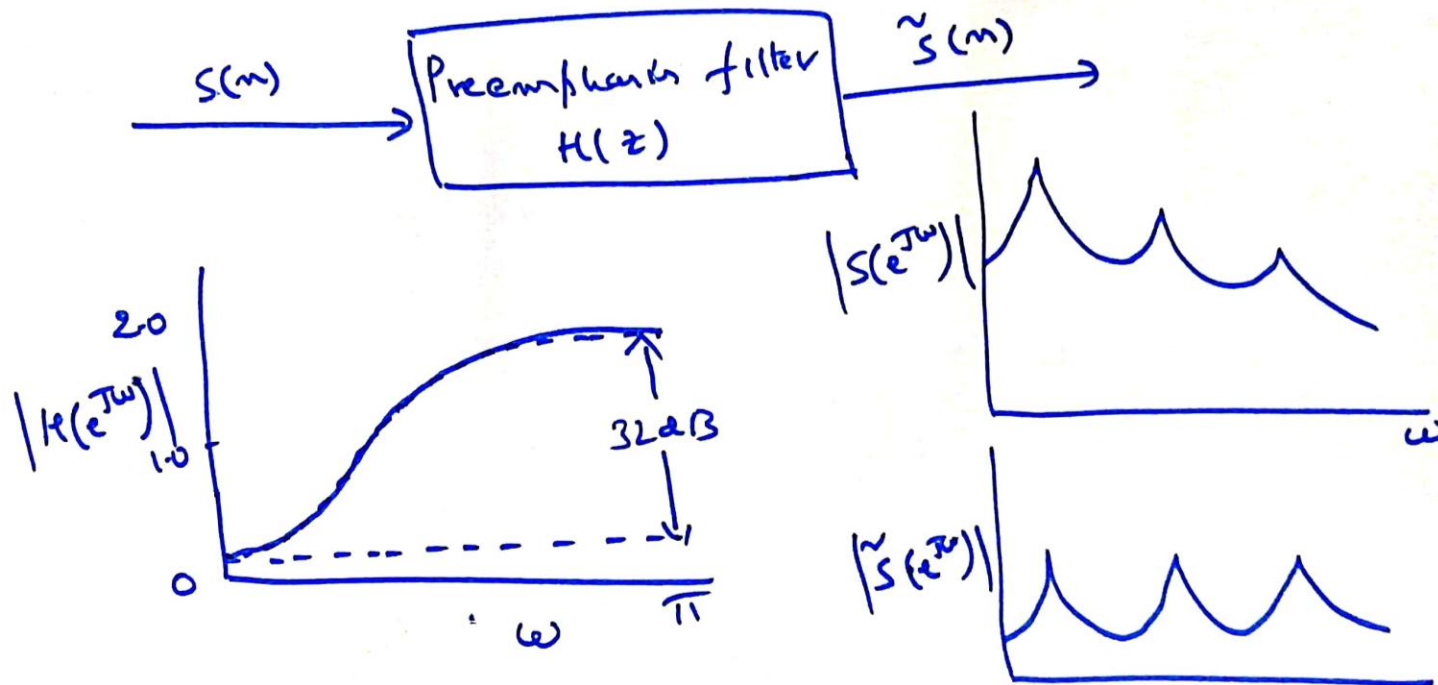
Extraction of Cepstral Features from LP Analysis

Features from LPC analysis of speech

1. Preemphasis (First order FIR filter)

$$H(z) = 1 - az^{-1} \quad 0.9 \leq a \leq 1.0$$

$$\tilde{s}(m) = s(m) - as(m-1); \quad a = 0.95$$

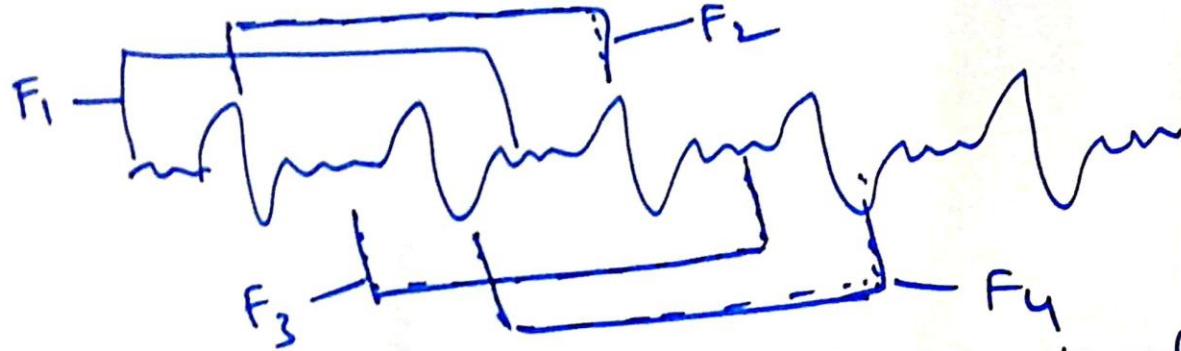


Extraction of Cepstral Features from LP Analysis

2. Frame Blocking

Frame Size = N samples

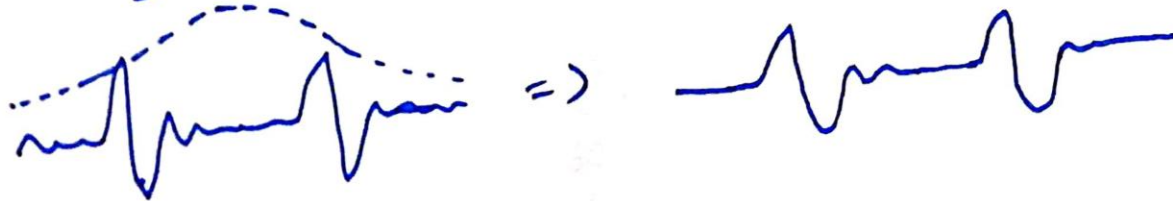
Adjacent frame duration = M samples $\approx \frac{1}{3}N$



Correlation between LPC spectral estimates

3. windowing: Avoids signal discontinuities at begin and end

Hanning window $w(m) = 0.54 - 0.46 \cos\left(\frac{2\pi m}{N-1}\right)$



Extraction of Cepstral Features from LP Analysis

4. Auto correlation Analysis

$$R(k) = \sum_{m=0}^{N-1-k} s(m)s(m+k) \quad k = 0, 1, \dots, p$$

5. LPC Analysis ($p = 8$ to 16)
(Durbin's Recursion)

LPC coefficients

Reflection coefficients

Log area ratio coefficients

Cepstral coefficients

6. LPC \rightarrow LPCCs

$$\# \text{ LPCCs} \approx \frac{3}{2}(p)$$

Extraction of Cepstral Features from LP Analysis

7. Parameter weighting (WLPCs)

Low-order cepstral coeffs — Spectral slope

High-order cepstral coeffs — Noise

Weighting the cepstral coeffs with tapered window

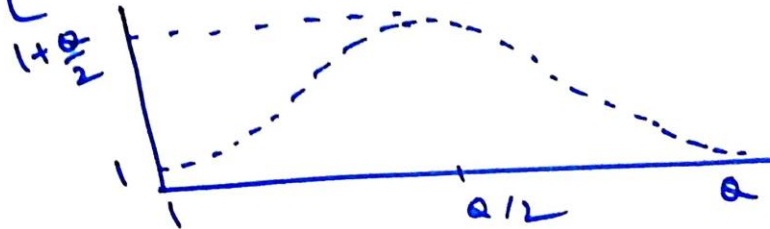
→ Differential log-mag spectrum

— fixed spectral slope → constant

— prominent spectral peaks are preserved

Band pass filter in the cepstral domain

$$W_m = \left[1 + \frac{Q}{2} \sin\left(\frac{\pi m}{Q}\right) \right] \quad 1 \leq m \leq Q$$



Extraction of Cepstral Features from LP Analysis

8. Temporal Cepstral Derivative

Cepstral Representation — Local spectral prop

$$C_x = [c_1 \ c_2 \ c_3 \ \dots \ c_q] ; \text{ cepstral coeffs of } i^{\text{th}} \text{ frame}$$

$$\Delta C_i(m) = \frac{\sum_{k=-N}^N k c_i(m+k)}{\sum_{k=-N}^N k^2} ; N=3$$

$$\{c_i\} \text{ — } q$$

$$\{c_i, \Delta c_i\} \text{ — } 2q$$

$$\{c_i, \Delta c_i, \Delta \Delta c_i\} \text{ — } 3q$$

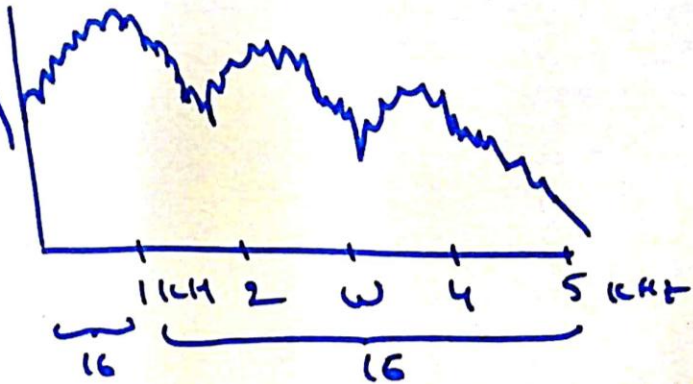
Extraction of Mel-Frequency Cepstral Coefficients

Spectral coefficients

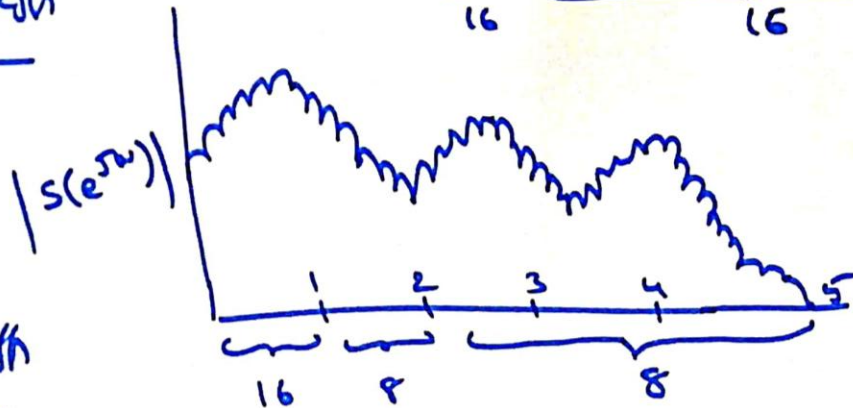
Speech frame $S(m) \mid_{m=0 \text{ to } N-1}$

$$S(\omega) = \text{FFT}(S(m))$$

Spectral coeffs $\begin{cases} 0-15 (0-7\text{K}) \\ 16-31 (1-5\text{K}) \end{cases}$



Mel-freq spectral coeffs



Mel-freq cepstral coeffs

MFCCs

IFT [Mel-spectral coeffs]

Linear Prediction Analysis of Speech (Hands-on)

The slide features a white background with a large blue shape on the right side that resembles a stylized profile of a human head or a speech bubble. At the bottom, there are decorative blue elements: a horizontal bar on the left and a series of vertical lines on the right.