## SIO 209: Signal Processing for Ocean Sciences Class 17

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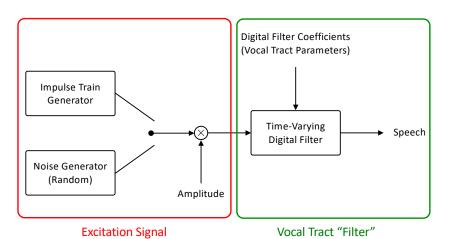
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## **Recall Speech Production Model**

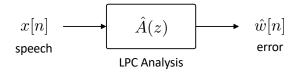


A vocoder extracts parameters of excitation signal and the vocal tract filter; transmitting these parameters (instead of the sampled speech signal) can reduce data rate significantly

1

## **Recall Speech Transmission**

• Direct digitization of speech results in a data rate of 64 kB/s since typically 8 Bits/sample are used and  $f_{\rm s}=8{\rm kHz}$  (speech is bandlimited to  $4-5{\rm kHz}$ )



- We take segments ("frames") that are 25 ms long, i.e., there are 40 segments/sec
- For each segment we need to obtain
  - the  $\hat{A}$  vector (10–14 coefficients)
  - voiced/unvoiced decision
  - pitch period
  - amplitude

 $\longrightarrow$  Results in a data rate of  $1.0-2.4 \mathrm{kb/s}$ 

2

## Recall High Resolution Spectral Analysis

 $w[n] \xrightarrow{\text{All-Poles} \\ \text{Filter}} x[n]$  white noise  $H(z) = \frac{1}{A(z)}$ 

Auto-Regressive (AR) Model
(IIR filter with no zeros)

Terms in A(z) have the form

Analysis Goal: Estimate  $P_{xx}(\omega) = |H(\omega)|^2$ All-Zeros Filter  $e_P(n) = \hat{w}(n)$   $\hat{A}(z)$   $E_P = \mathrm{E}\big[e_P^2[n]\big]$  prediction error filter "lpc" and "levinson"

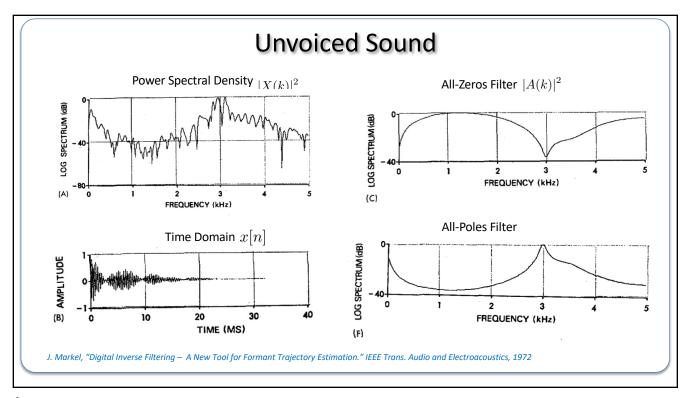
The polynomial coefficients of  $A(z) \, \mathrm{can} \ \mathrm{be} \ \mathrm{obtained} \ \mathrm{from} \ \mathrm{"poly"}$ 

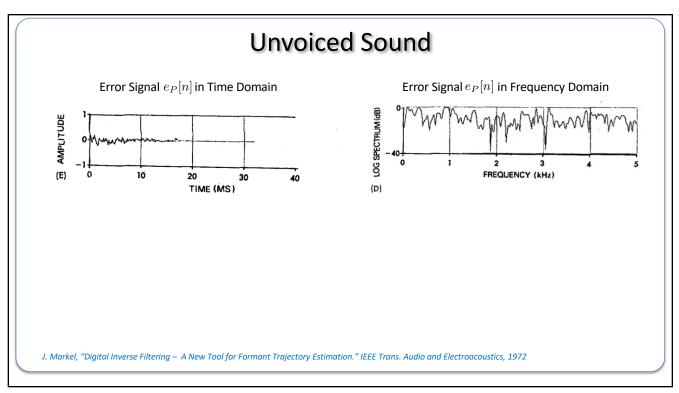
The all-pole filters can be implemented using "filter"

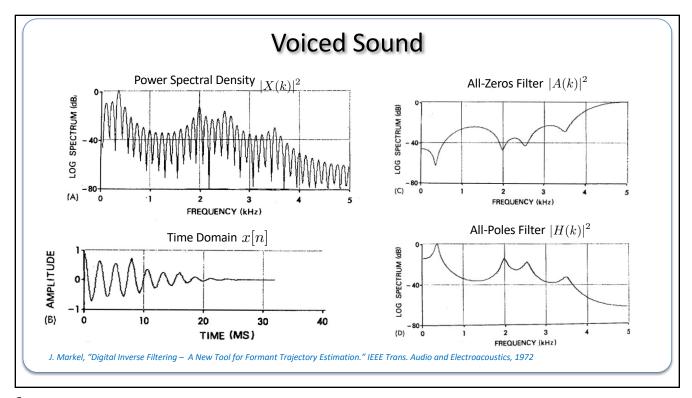
$$\frac{(z-z_k)(z-z_k^*)}{z^2}$$

 $k = 1 \dots K/2$ 

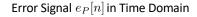
Note that there is also a ARMA (AR-moving-average) model which is based on a general IIR filter (poles & zeros)

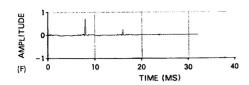




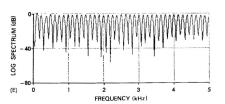






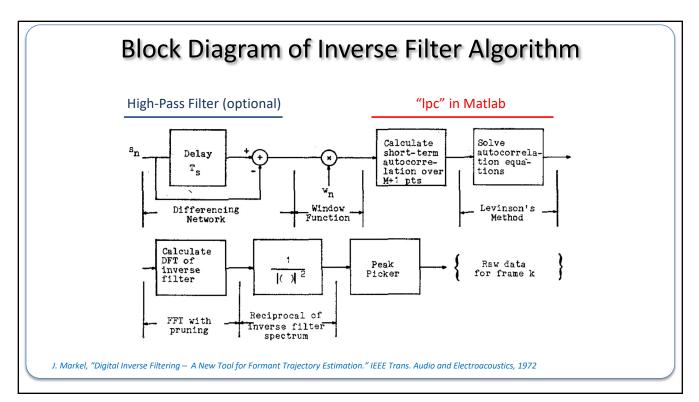


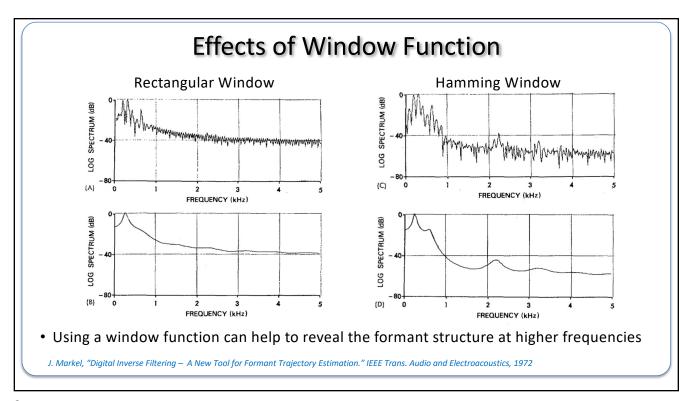
Error Signal  $e_P[n]$  in Frequency Domain

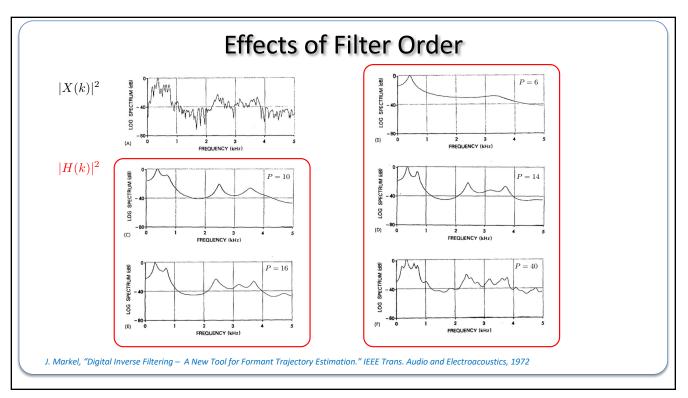


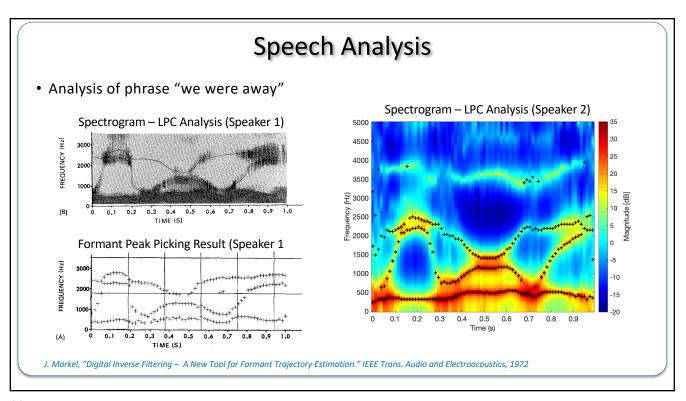
- The autocorrelation function of the error signal can be used to make a voiced/unvoiced decision by looking for peaks between 8 ms and 12 ms
- If a peak is detected, the sound can be declared as voiced, and the delay related to the peak is the pitch period

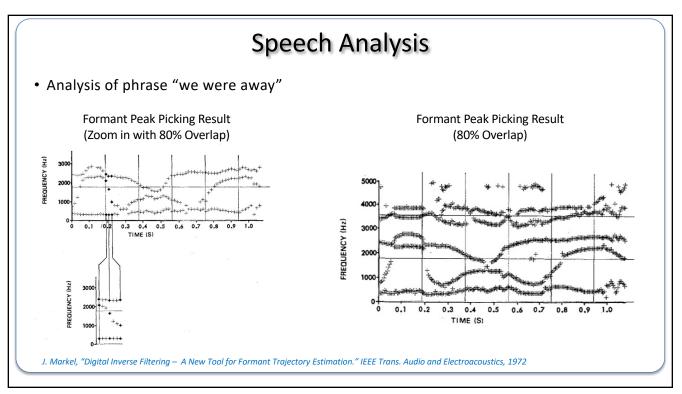
J. Markel, "Digital Inverse Filtering – A New Tool for Formant Trajectory Estimation." IEEE Trans. Audio and Electroacoustics, 1972





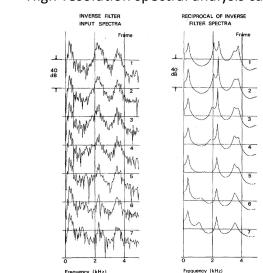








• High-resolution spectral analysis can reveal small changes in formant structure



J. Markel, "Digital Inverse Filtering — A New Tool for Formant Trajectory Estimation." IEEE Trans. Audio and Electroacoustics, 1972