

# Frequency-Domain Pitch Estimation

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Source: Discrete-Time Speech Signal Processing \ Quatieri, chapter 10

# Introduction

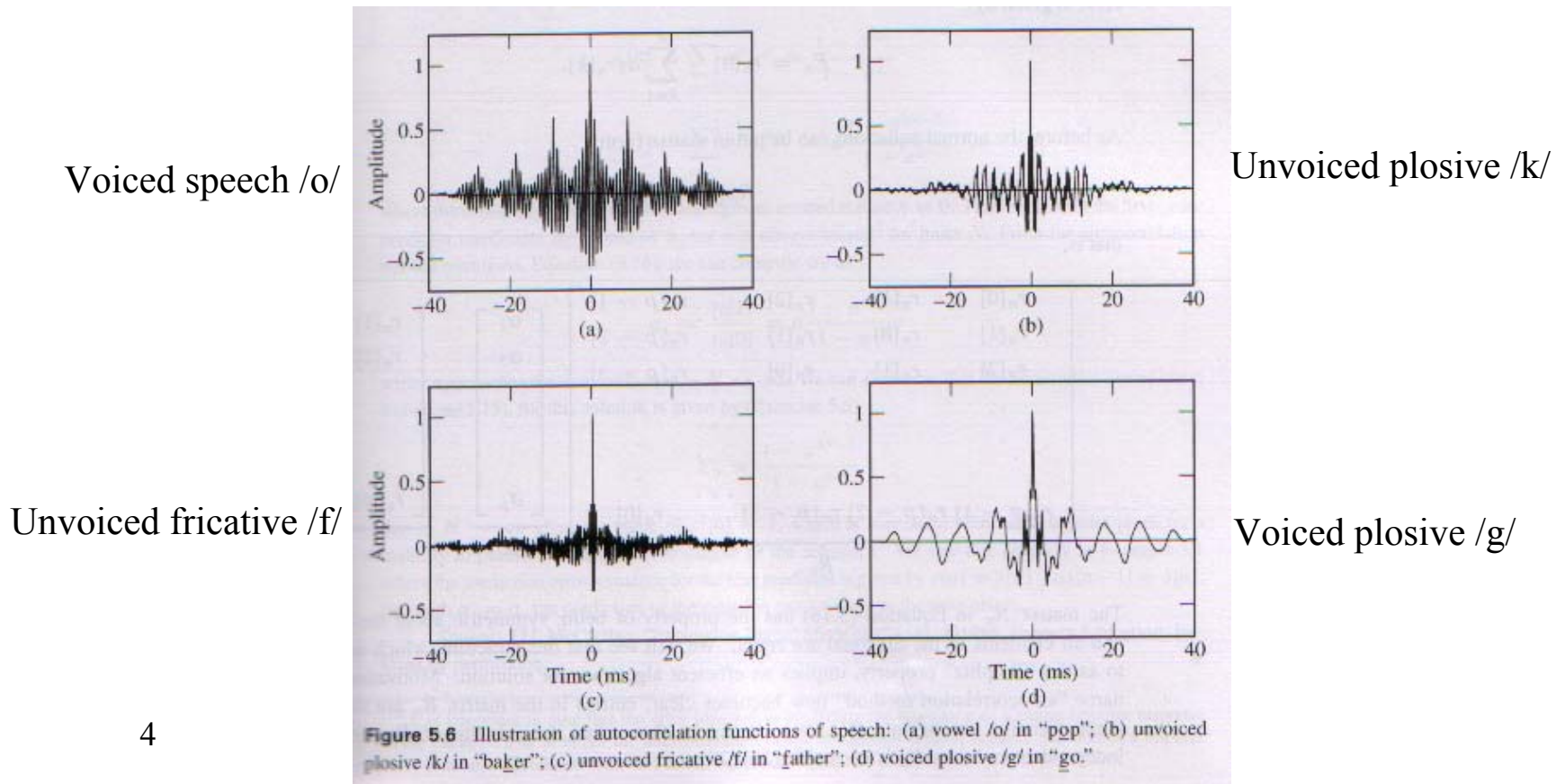
- Reliable estimation of the fundamental frequency is of high importance and has many applications to speech coding, speech synthesis and speech recognition.
- Different approaches to speech analysis/synthesis naturally lead to different methods for pitch and voicing estimation.
- Pitch and voicing estimation algorithms based on the sinusoidal model will be introduced.

# Presentation outline

- Background:
  - Common pitch estimation methods.
  - Sinusoidal Speech model.
- Pitch estimation based on sinusoidal model.
- Voicing detection.
- Multi band pitch and voicing estimation.

# The pitch

- Duration: 2.5-16msec (20-128 samples at 8Khz).
- Frequency: 62.5-400Hz.



# Common pitch estimation methods

- Autocorrelation method:

Define the short-time sequence  $s_n[m]$

$$s_n[m] \triangleq s[m]w[n-m]$$

$$\left( S_n[m] = 0 \quad \forall m \notin \left[ n - \frac{N_w-1}{2}, n + \frac{N_w-1}{2} \right] \right)$$

$w[n]$  – *Analysis window, length  $N_w$  (odd)*

$s[m]$  – *Speech signal*

Define the short-time Autocorrelation  $r_n[\tau]$

$$r_n[\tau] = s_n[\tau] * s_n[-\tau] = \sum_{m=-\infty}^{\infty} s_n[m] s_n[m + \tau]$$

# Autocorrelation method

In order to minimize the MSE:

$$E[P] = \sum_{m=-\infty}^{\infty} \left( s_n[m] - s_n[m+P] \right)^2$$

We will choose:

$$\hat{P} = \arg \max_p \sum_{m=-\infty}^{\infty} s_n[m] s_n[m+P]$$

$$\hat{P} > \varepsilon$$

$$r_n[\tau] > \textit{Threshold}$$

# Common pitch estimation methods

- AMDF (Average Magnitude difference Function)

$$A_n[P] = \sum_{k=n-N_w+P}^n |s[k] - s[k-P]|, \quad 20 \leq P \leq 128$$

$$\hat{P} = \arg \min_P A_n[P], \quad (A_n[P] < \text{Threshold})$$

- Cepstral Method (CEP)

$$c[n] \triangleq \text{IDFT} \left\{ \log \left| \text{DFT} \{s[n]\} \right| \right\}$$

$$\hat{P} = \arg \max_P \left( \text{Re} \{c[P]\} \right), \quad c[P] > \text{Threshold}; P_{\min} < P < P_{\max}$$

# Common pitch estimation methods

- LPC based pitch detector (SIFT-Simplified Inverse Filtering Technique)
  1. Calculate the All-pole filter  $\frac{1}{A(Z)}$ .
  2. Filter the speech with  $A(z)$  to produce the residual signal (=excitation)  $u[n]$ .
  3.  $\hat{P}$  = lag between peaks of  $u[n]$ .



# Typical drawbacks of pitch estimators

- Pitch doubling.
- Pitch halving.
- Mixed V/UV segments.
- Adaptive threshold.
- First formant.
- Speech corruption by noise.

## **Solutions:**

- Non-linear transformation of original speech signal.
- Pitch tracking from frame to frame (recursive smoothing, median filter, limit change rate...).

# Sinusoidal Speech model

$$s(t) = (\text{Re}) \sum_{k=1}^K A_k e^{j(\omega_k t + \theta_k)}$$

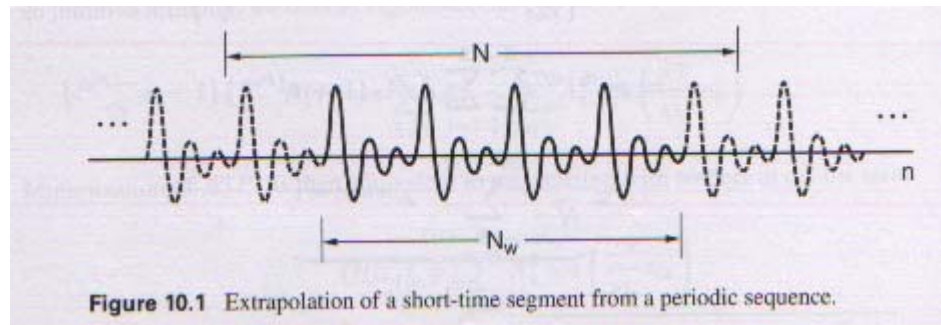
$K(t)$  – Number of frequencies.

$A_k(t), \omega_k(t), \theta_k(t)$  – Amp., Freq. phases.

- Model parameters are calculated using STFT.
- Analysis window length must be at least 2.5 times the average pitch.

# Pitch estimation based on sinusoidal model

- The sinusoidal model representation is used to extrapolate the short-time sequence  $s_n[m]$ .



- The same MSE criterion as in the autocorrelation method is used, but this time with the signal:

$$\tilde{s}[m] = \sum_{k=1}^K A_k e^{j(\omega_k m + \theta_k)} \quad \text{instead of } s_n[m].$$

The MSE:

$$E[P] = \frac{1}{N} \sum_{m=-(N-1)/2}^{(N-1)/2} \left| \sum_{k=1}^K A_k \left[ e^{j(mw_k + \theta_k)} - e^{j((m+P)w_k + \theta_k)} \right] \right|^2$$

Substituting  $\left| \sum_{k=1}^K [\cdot] \right|^2$  by  $\sum_{k=1}^K [\cdot] \sum_{l=1}^K [\cdot]$  and letting the extrapolation interval  $N$  go to infinity yields:

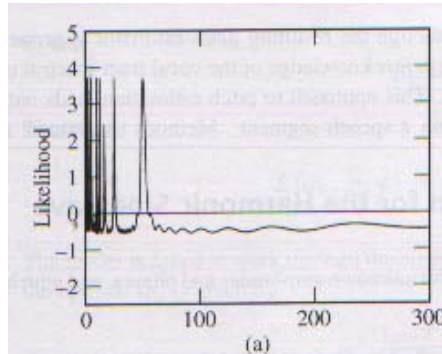
$$E[P] \approx \sum_{k=1}^K A_k^2 (1 - \cos(P\omega_k))$$

Define:  $P \triangleq \frac{2\pi}{\omega_0}$  and  $Q(\omega_0) \triangleq \sum_{k=1}^K A_k^2 \cos\left(\frac{2\pi}{\omega_0} \omega_k\right)$

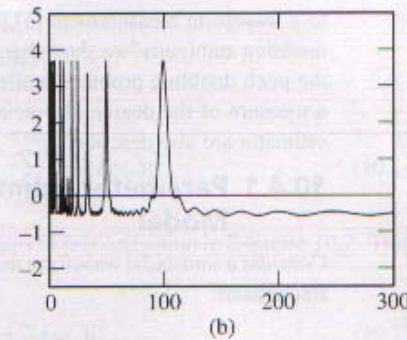
$Q(\omega_0)$  is the *likelihood function* of  $\omega_0$  to be the true pitch.

- Algorithm:  $\hat{\omega}_0 = \arg \max Q(\omega_0)$
- If the  $\omega_k$ 's are multiples of a fundamental frequency  $\tilde{\omega}_0$ , and if  $\omega_0 = \tilde{\omega}_0$ , then  $E[P]=0$

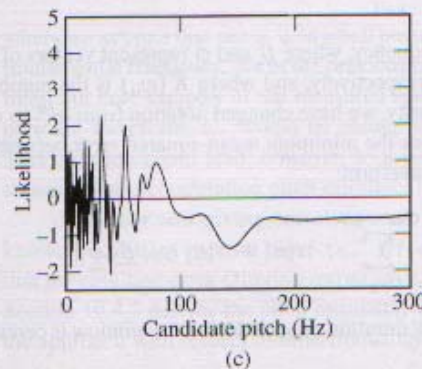
Pitch freq = 50Hz



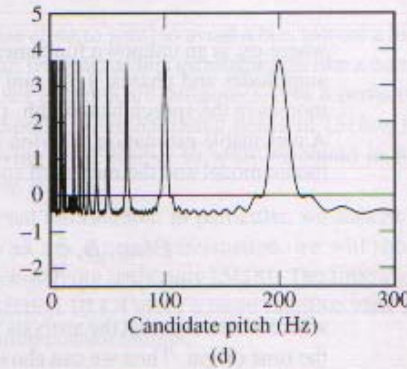
Pitch freq = 100Hz



Pitch freq = 50Hz  
With noise



Pitch freq = 200Hz



**Figure 10.3** Pitch likelihood function  $Q(\omega_0)$  for (a) 50, (b) 100, and (d) 200 Hz. The effect of noise on (a) is shown in (c).

# **Pitch estimation based on sinusoidal model**

- Pitch doubling problem is solved.
- Pitch halving problem still exists.

# Pitch Estimation Based on a Harmonic Sinewave Model

- Harmonic sinewave model:

$$s[n; \omega_0, \underline{\phi}] = \sum_{k=1}^{K(\omega_0)} \bar{A}(k\omega_0) e^{j(nk\omega_0 + \phi_k)}$$

$K(\omega_0)$  – Number of harmonics in the speech bandwidth.

$\bar{A}(\omega) = |H(\omega)|$  – Vocal tract envelope

(for unity excitation magnitude).

$\underline{\phi}$  – Phases of the harmonics

- Our goal is to fit the best harmonic model to the speech.

The MSE:

$$E(\omega_0, \underline{\phi}) = \frac{1}{N_w} \sum_{m=-(N_w-1)/2}^{(N_w-1)/2} \left| s[n] - \hat{s}[n; \omega_0, \underline{\phi}] \right|^2$$

Using the harmonic model representation leads to:

$$E(\omega_0) = P_s - \underbrace{2 \left( \sum_{k=1}^{K(\omega_0)} \bar{A}(k\omega_0) |S(k\omega_0)| - \frac{1}{2} \sum_{k=1}^{K(\omega_0)} \bar{A}^2(k\omega_0) \right)}_{\rho(\omega_0)}$$

$P_s$  – signal average energy

$S(k\omega_0)$  – STFT at frequency  $k\omega_0$

Algorithm:  $\hat{\omega}_0 = \arg \max_{\omega_0} \rho(\omega_0)$



# Pitch estimation based on harmonic sinewave model

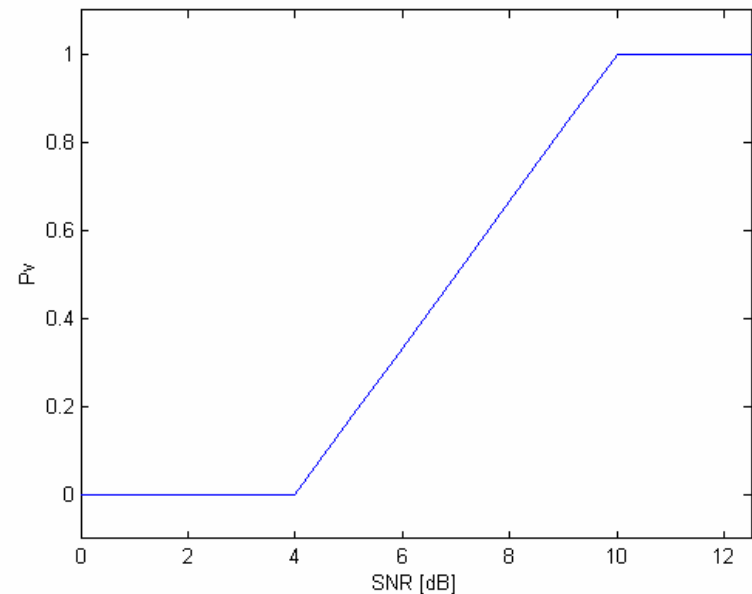
- Pitch doubling problem is solved.
- Pitch halving problem is solved.
- Enhancements:
  - Eliminate formant interaction problem.
  - Adaptive pitch resolution.

# Voicing detection

$$SNR = \frac{\sum_{n=-(N-1)/2}^{(N-1)/2} |s[n]|^2}{\sum_{n=-(N-1)/2}^{(N-1)/2} |s[n] - \hat{s}(n; \omega_0)|^2}$$

$$P_v = \begin{cases} 1 & SNR > 10dB \\ \frac{1}{6}(SNR - 4) & 4dB < SNR \leq 10dB \\ 0 & SNR \leq 4dB \end{cases}$$

$P_v$  – probability that speech is voiced



# Multi band pitch and voicing estimation

- MBE -Multiband Excitation Model,(Griffin & Lim, 1988).
- Speech model:

$$\hat{S}_{\omega}(\omega) = H_{\omega}(\omega) |E_{\omega}(\omega)|$$

$H_{\omega}(\omega)$  – *spectral envelope*

$|E_{\omega}(\omega)|$  – *excitation spectrum*

- Pitch period and Spectral envelope are calculated.
- The bandwidth is divided into ~20 subbands around the pitch harmonics.

# Multi band pitch and voicing estimation

- The normalized MSE for subband  $m$  is defined by:

$$E_m(\hat{\omega}_0) = \frac{\int_{\gamma_m} |S_\omega(\omega) - \hat{S}_\omega(\omega; \hat{\omega}_0)|^2 d\omega}{\int_{\gamma_m} |S_\omega(\omega)|^2 d\omega}$$

$$\gamma_m = \{\omega : \omega_{m-1} \leq \omega \leq \omega_m\}, \quad m=1,2,\dots,M$$

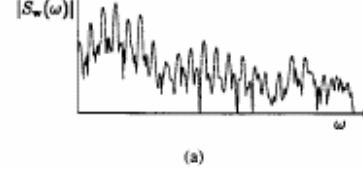
$$\hat{S}_\omega(\omega; \omega_0) - \text{Harmonic Model}$$

*if  $E_m(\hat{\omega}_0) < \text{Threshold} \Rightarrow \text{Band is voiced}$*

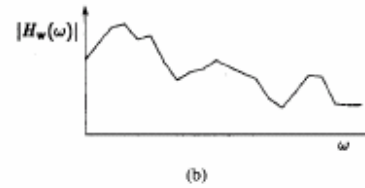
# MBE

## example:

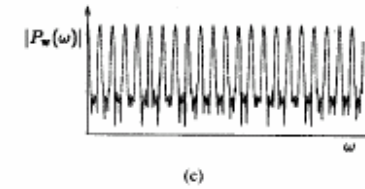
Original spectrum



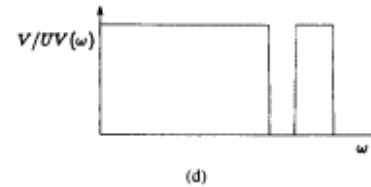
Spectral envelope



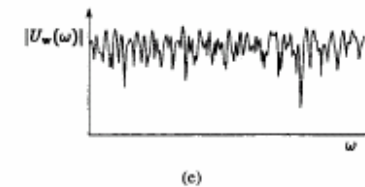
Periodic spectrum



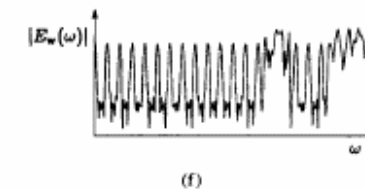
V/UV information



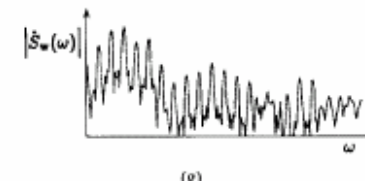
Noise spectrum



Excitation spectrum



Synthetic spectrum



# Advantages of the MBE model

- Quality for mixed V+UV speech.
- Robustness for additive acoustic noise.
- Lower bitrate for noiselike subbands.

# Summary

- Pitch estimation can be calculated by fitting an harmonic sinewave model to the speech signal.
- The pitch estimate is robust to pitch doubling and pitch halving problems.
- The harmonic model naturally leads to a voicing detection based on the SNR.
- Implementation of voicing detection in sub bands was introduced.
- Pitch estimation under noise conditions remains an open problem.

# Bibliography

1. **Discrete-Time Speech Signal Processing** \ Quatieri, chapters 9,10
2. **A comparative Performance Study of Several Pitch Detection Algorithms**\ Rabiner, Cheng, IEEE tran. on acoustics, speech and sig. Proc., 1976
3. **Multiband Excitation Vocider**\Griffin, Lim, IEEE tran. on acoustics, speech and sig. Proc., 1988