ΕΛΛΗΝΙΚΗ ΔΗΜΟΚΡΑΤΙΑ
ΠΑΝΕΠΙΣΤΗΜΙΟ ΚΡΗΤΗΣ

Ψηφιακή Επεξεργασία Φωνής

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On the Inverse Filtering of Speech

George Kafentzis

CS578-November 2012

Outline

Human Speech Production System

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- The vocal tract, which is the path from the vocal folds to the lips.
- The lips, which is the final bound before speech output.

Source-Filter model

- Based on this simplification, voiced speech can be modeled as a linear filtering operation:

$$s(t) = g(t) \star v(t) \star l(t) \leftrightarrow S(z) = G(z)V(z)L(z)$$

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- $s(t)$ the output speech waveform.

(Glottal) Inverse Filtering - IF

What is inverse filtering?

- Inverse Filtering is a technique for obtaining the source of voiced speech: the glottal airflow velocity waveform.
- How does the glottal flow look like?

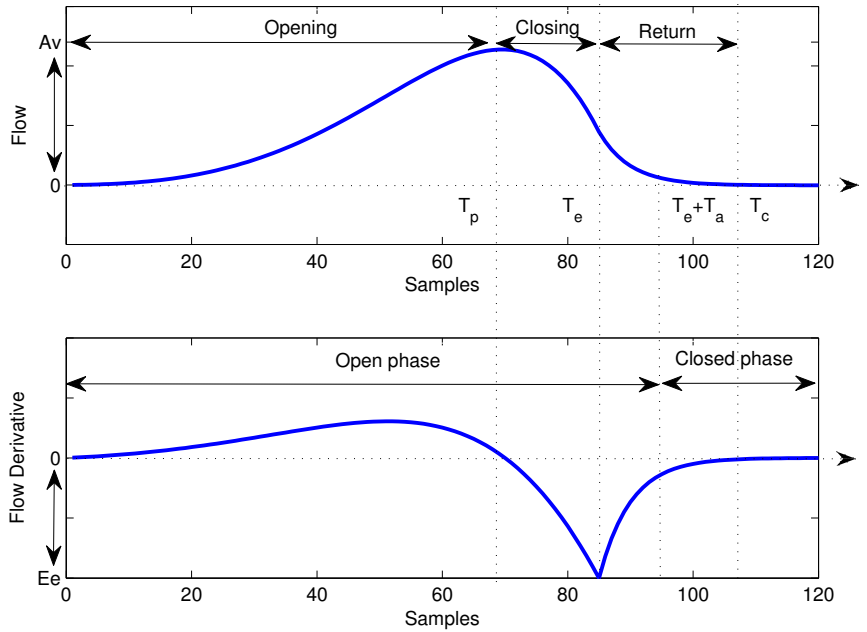

Figure: Phases of the glottal flow and its derivative.

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- Form a computational model for the vocal tract filter, $\hat{V}(z)$.

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Basic idea:

- Form a computational model for the vocal tract filter, $\hat{V}(z)$.
- Cancel its effect from the speech waveform by filtering the speech signal through the inverse of the model.
- It is obvious that the heart of an IF system is the vocal tract filter estimation.

Evaluation of IF techniques

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- Approaches:
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- None of the previous approaches is truly objective.

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- In this case, both the speech pressure waveform and the glottal flow are available.

Database of physically modeled speech

- In detail, self sustained vocal fold vibration was simulated with three masses coupled to one another through stiffness and damping elements.

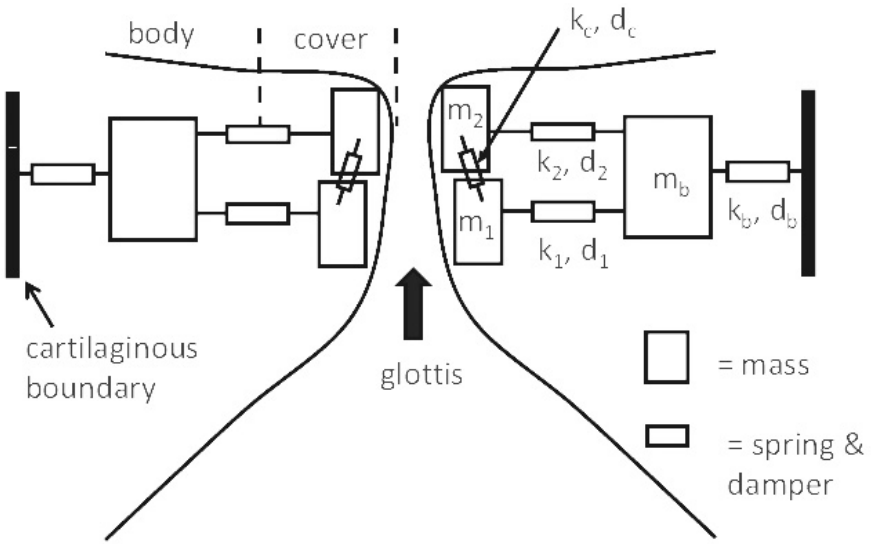

Figure: Schematic diagram of the lumped-element vocal fold model. The cover-body structure of each vocal fold is represented by three masses that are coupled to each other by spring and damping elements. Bilateral symmetry was assumed for all simulations.

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- These parameters were transformed into mechanical parameters according to Titze and Story[3].

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- It generates both the signal on which inverse filtering is typically performed (microphone signal) and the signal that is seeked to be determined (glottal flow).

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- In summary, the model is a simplified but physically motivated representation of a speaker.
- It generates both the signal on which inverse filtering is typically performed (microphone signal) and the signal that is seeked to be determined (glottal flow).
- This provides an idealized test case for inverse filtering algorithms.

Example

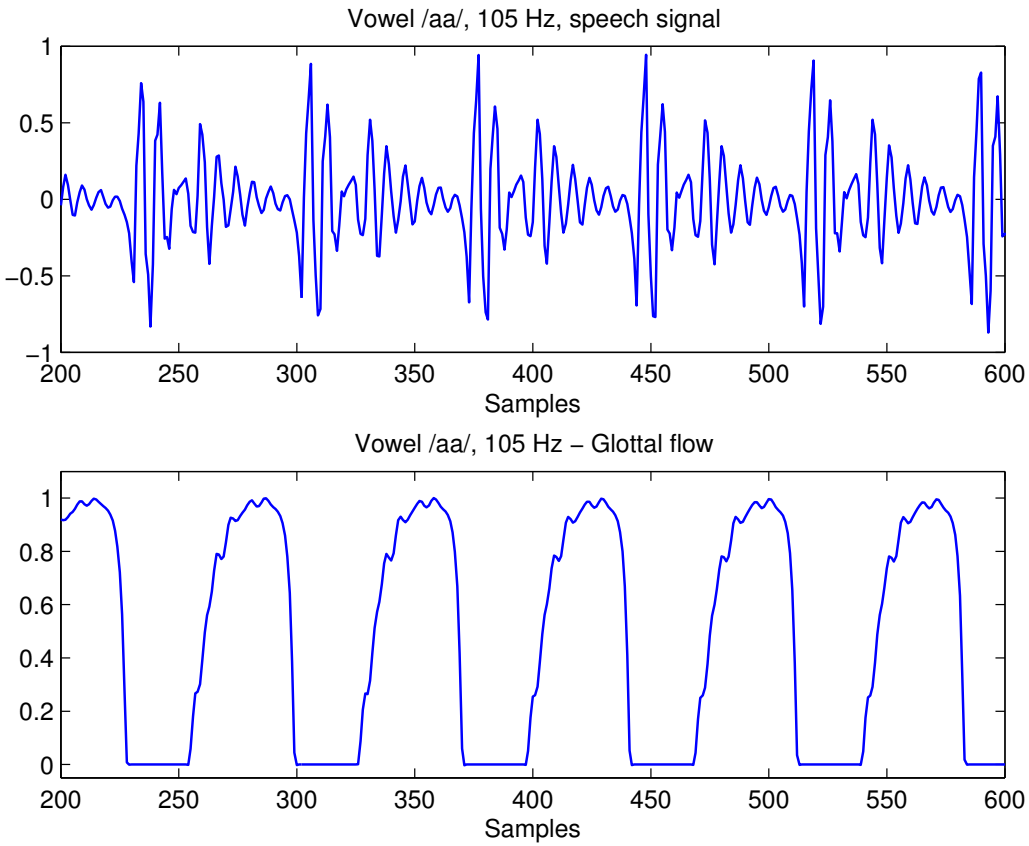

Figure: Upper panel: Speech waveform.
Lower panel: Glottal flow waveform.

Outline

Vocal tract estimation techniques

A number of techniques have been developed to robustly estimate the vocal tract filter.

- Most of them are based on *Linear Prediction* (LP).
- LP is used to produce an all-pole model of the system filter, $H(z)$, which turns out to be a model of the vocal tract and its resonances or formants.

$$V(z) = \frac{1}{\sum_{k=1}^{p}(\alpha_k z^{-k})},$$

- In general, LP minimizes the mean squared error over a region R

$$E = \sum_{R} e^2[n], \text{ where } e[n] = s[n] - \sum_{k=1}^{p} a_k s[n-k],$$

where p is the prediction order and a_k are the prediction coefficients.

- The selection of this region leads to different approaches.

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  - Closed Phase Covariance-based Linear Prediction,
  - Stabilised Weighted Linear Prediction,
  - Closed Phase (CP) Covariance-based Linear Prediction with Mathematical Constraints.

Linear Prediction - Autocorrelation method

- Assuming that the speech signal $s[n]$ is zero outside an interval $0 \leq n \leq N-1$
- the total error minimization leads to the matrix equation

$$\mathbf{\Phi}\vec{a} = \vec{r}, \tag{1}$$

- where the matrix Φ is called the *autocorrelation matrix* and its elements are given by $\Phi_{i,j} = R(|i-j|) = \sum_{n=i}^{N-1} s[n]s[n-|i-j|], \quad 0 \leq i \leq p$, and the other two vectors are given by

$$\vec{a} = [a_1, a_2, a_3, ..., a_p]^T \quad , \vec{r} = [R(1), R(2), R(3), ..., R(p)]^T. \tag{2}$$

- A tapered window (e.g. Hanning) is often used to eliminate beggining and end effects.

Linear Prediction - CP Covariance method

- Assuming that the speech signal $s[n]$ is zero outside an interval $-p \leq n \leq N-1$
- the total error minimization leads to the matrix equation

$$\mathbf{\Phi}\vec{a} = \vec{\psi}, \tag{3}$$

- where the matrix Φ has the properties of a *covariance matrix* and its elements are given by

$$\phi_{i,j} = \sum_{n=0}^{N-1} s[n-i]s[n-j], \tag{4}$$

where $1 \leq i, j \leq p$, and the other two vectors are given by

$$\vec{a} = [a_1, a_2, a_3, ..., a_p]^T \quad , \vec{\psi} = [\phi_{0,1}, \phi_{0,2}, \phi_{0,3}, ..., \phi_{0,p}]^T. \tag{5}$$

- However, in CP analysis, the total error is minimized over a region where the glottis is closed.

Stabilised Weighted Linear Prediction

Stabilized Weighted Linear Prediction (SWLP)[1], is an all-pole modeling method based on Weighted Linear Prediction (WLP).

- SWLP uses time domain weighting of the square of the prediction error signal.
- Short Time Energy (STE) Weighting function: $w_n = \sum_{i=0}^{M-1} x^2[n-i-1]$, where $x[n]$ is the signal and M is the duration of the STE window.
- The prediction error energy E in the SWLP method is

$$E = \sum_{n=1}^{N+p} (e_n)^2 w_n = \mathbf{a}^T \Big(\sum_{n=1}^{N+p} w_n \mathbf{x}[n] \mathbf{x}^T[n] \Big) \mathbf{a} = \mathbf{a}^T \mathbf{R} \mathbf{a}, \tag{6}$$

where w_n is the weight imposed on sample n, N is the length of the signal $x[n]$, and

$$\mathbf{R} = \sum_{n=1}^{N+p} w_n \mathbf{x}[n] \mathbf{x}^T[n].$$

Stabilised Weighted Linear Prediction

- Constrained minimization problem:

$$\text{minimize } E \text{ subject to } \mathbf{a}^T\mathbf{u} = 1,$$

where $\mathbf{u}$ is the vector defined as $\mathbf{u} = [1\,0\,\ldots\,0]^T$.

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- It can be shown that $\mathbf{a}$ satisfies the linear equation

$$\mathbf{R}\mathbf{a} = \sigma^2\mathbf{u}, \tag{7}$$

where $\sigma^2 = \mathbf{a}^T\mathbf{R}\mathbf{a}$ is the error energy.

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  where $\sigma^2 = \mathbf{a}^T\mathbf{R}\mathbf{a}$ is the error energy.
- Finally, the SWLP all-pole model is obtained as $H(z) = 1/A(z)$, where $A(z)$ is the z-transform of vector $\mathbf{a}$.

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- By emphasizing on these samples that occur during the glottal closed phase, it is likely to yield more robust acoustical cues for the formants.
- A high value of M increases the sharpness of the resonances of the spectrum, whereas a low value of M increases the smoothness of the spectrum.

Stabilised Weighted Linear Prediction

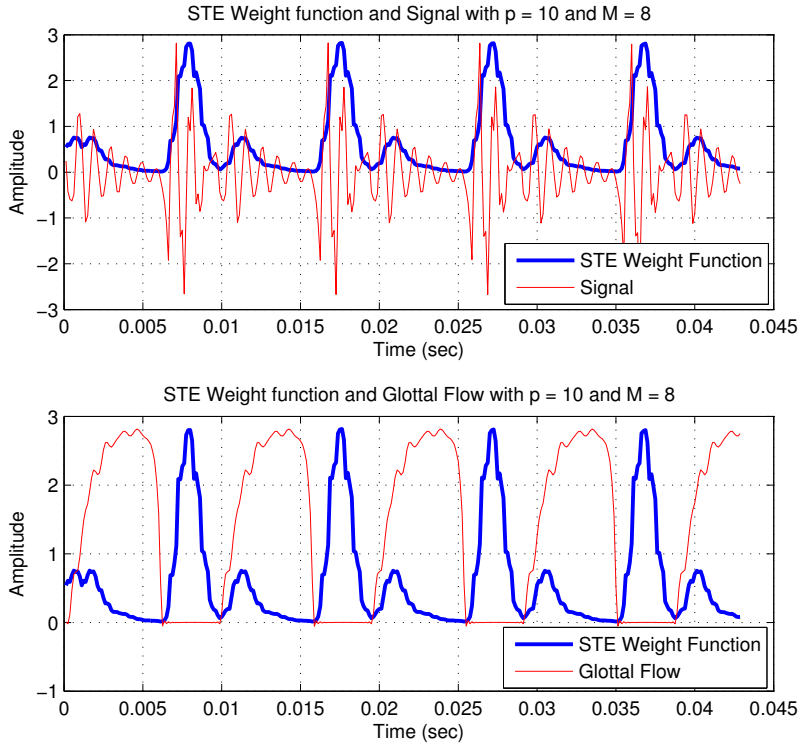

Figure: Upper panel: time domain waveforms of speech (vowel /a/ produced by male speaker) and short-time energy (STE) weight function (M=8).
Lower panel: Glottal flow waveform of the vowel /a/ together with the STE weight function (M=8).

Stabilised Weighted Linear Prediction

- Stability is ensured by the following formula:
- $$\mathbf{R} = \mathbf{Y}^T \mathbf{Y}, \tag{8}$$

  where

  $$\mathbf{Y} = [\mathbf{y_0}\ \mathbf{y_1} \cdots \mathbf{y_p}] \in \Re^{(N+p)x(p+1)}$$

  and

  $$\mathbf{y}_0 = [\sqrt{w_1}x[1] \cdots \sqrt{w_N}x[N]\ \ 0 \cdots 0]^T.$$

- The column vectors are given by

  $$\mathbf{y}_{k+1} = \mathbf{B}\mathbf{y}_k,\ \ k = 0, 1, \cdots, p-1, \tag{9}$$

  where

  $$\mathbf{B} = \begin{bmatrix} 0 & 0 & \cdots & 0 & 0 \\ \sqrt{w_2/w_1} & 0 & 0 & \cdots & 0 \\ 0 & \sqrt{w_3/w_2} & 0 & \cdots & 0 \\ \vdots & \ddots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & \sqrt{w_{N+p}/w_{N+p-1}} & 0 \end{bmatrix}$$

Stabilised Weighted Linear Prediction

- Before forming the matrix $\mathbf{Y}$, the elements of the secondary diagonal of the matrix $\mathbf{B}$ are defined for all $i = 1, \cdots, N + p - 1$ as

$$\mathbf{B}_{i+1,i} = \begin{cases} \sqrt{w_{i+1}/w_i}, & \text{if } w_i \leq w_{i+1} \\ 1, & \text{if } w_i > w_{i+1} \end{cases}$$

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- This method of computing matrix R is called the *Stabilized Weighted Linear Prediction* model, and the stability of the all-pole filter is ensured.

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Sensitivity of the covariance frame position:

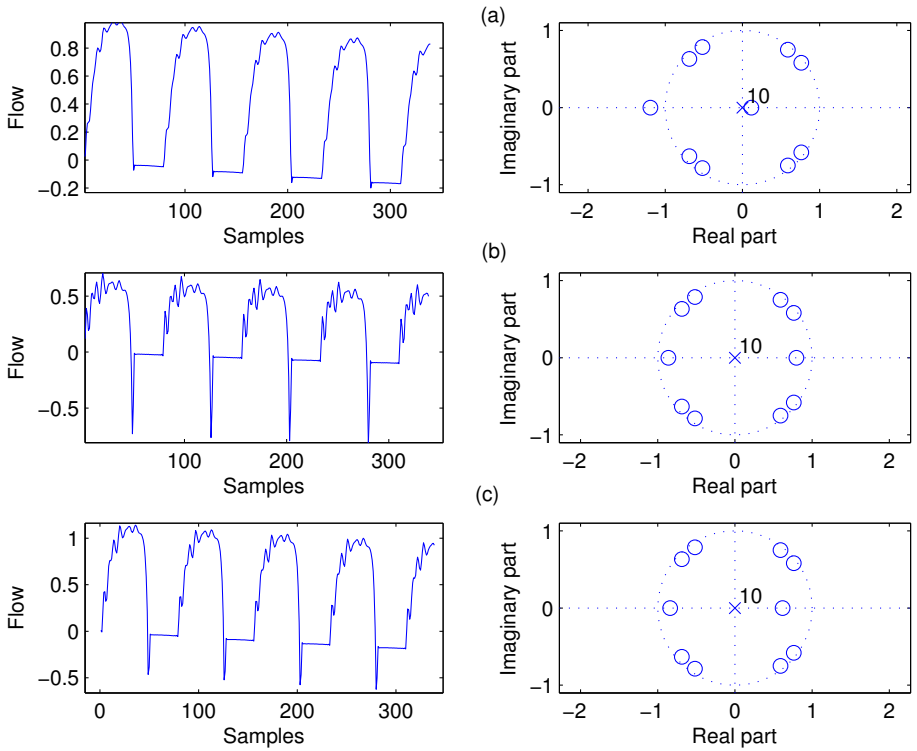

Figure: Covariance Frame Misalignment and Glottal Flow Distortion.

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- The removal of such roots results in less dependency on the covariance frame location.

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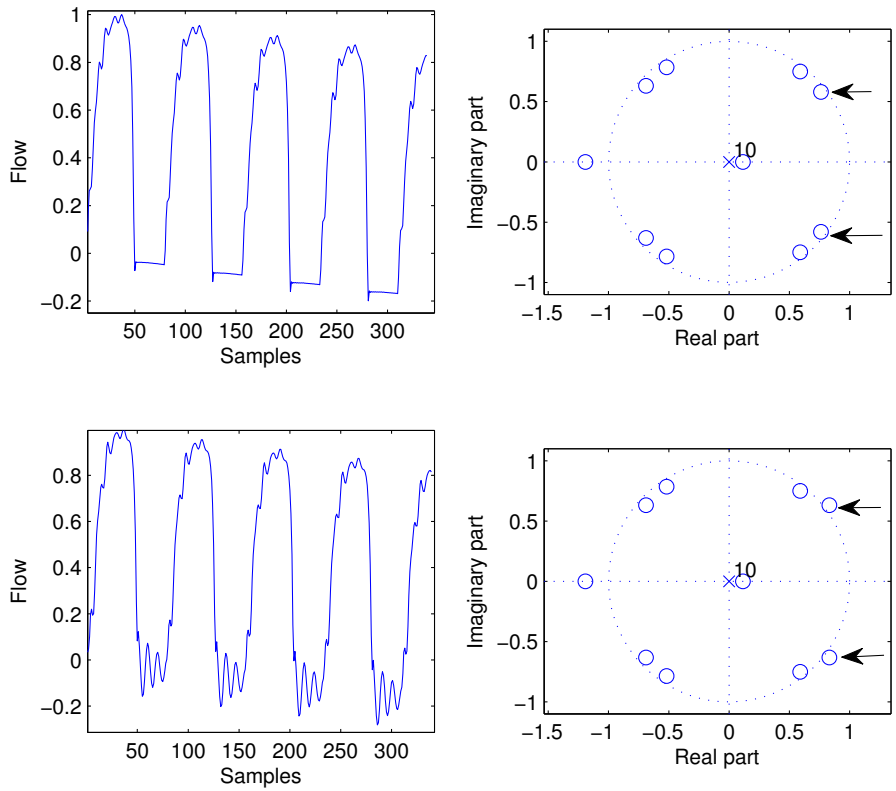

Figure: Distortion caused by non-minimum phase filter.

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- That leaves the amplitude spectrum unchanged.
- The phase characteristics change, though.

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- How?
  - Not allow mean square error to locate the roots freely on the z-plane.
  - Impose mathematical restrictions in a form of concise mathematical equations.
  - Two suggestions: DC-constraint and/or π-constraint.

DC-Constrained CP Covariance Linear Prediction

- DC-constraint:

$$V(e^{j0}) = \sum_{k=0}^{p} \alpha_k e^{-j0n} = \sum_{k=0}^{p} \alpha_k = I_{DC}. \qquad (10)$$

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- Why DC-constraint?
  - Amplitude response of voiced sounds approaches unity at zero frequency[5]

DC-Constrained CP Covariance Linear Prediction

- DC-constraint:

$$V(e^{j0}) = \sum_{k=0}^{p} \alpha_k e^{-j0n} = \sum_{k=0}^{p} \alpha_k = I_{DC}. \qquad (10)$$

- Why DC-constraint?
  - Amplitude response of voiced sounds approaches unity at zero frequency[5]
  - A short and misplaced covariance frame might lead to a response with higher gain at DC than at formants.

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- Why DC-constraint?
  - Amplitude response of voiced sounds approaches unity at zero frequency[5]
  - A short and misplaced covariance frame might lead to a response with higher gain at DC than at formants.
  - With such a constraint, one might expect a better match of the amplitude response to Fant's source-filter theory.

DC-Constrained CP Covariance Linear Prediction

Formulation:

- minimize $\mathbf{a}^T \mathbf{\Phi} \mathbf{a}$ subject to $\mathbf{\Gamma}^T \mathbf{a} = \mathbf{b}$,
- where $\mathbf{a} = [a_0, \cdots, a_p]^T$ is the filter coefficient vector with $a_0 = 1$, $\mathbf{\Phi}$ is the covariance matrix, $\mathbf{b} = [1,\ l_{DC}]^T$, and $\mathbf{\Gamma}$ is a $(p+1)$ by 2 constraint matrix defined as

$$\mathbf{\Gamma} = \begin{bmatrix} 1 & 1 \\ 0 & 1 \\ \vdots & \vdots \\ 0 & 1 \end{bmatrix} \tag{11}$$

- Using Lagrange multipliers (convex problem), we have the filter coefficients:
-

$$\mathbf{a} = \mathbf{\Phi}^{-1}\mathbf{\Gamma}(\mathbf{\Gamma}^T\mathbf{\Phi}^{-1}\mathbf{\Gamma})^{-1}\mathbf{b} \tag{12}$$

DC-π-Constrained CP Covariance Linear Prediction

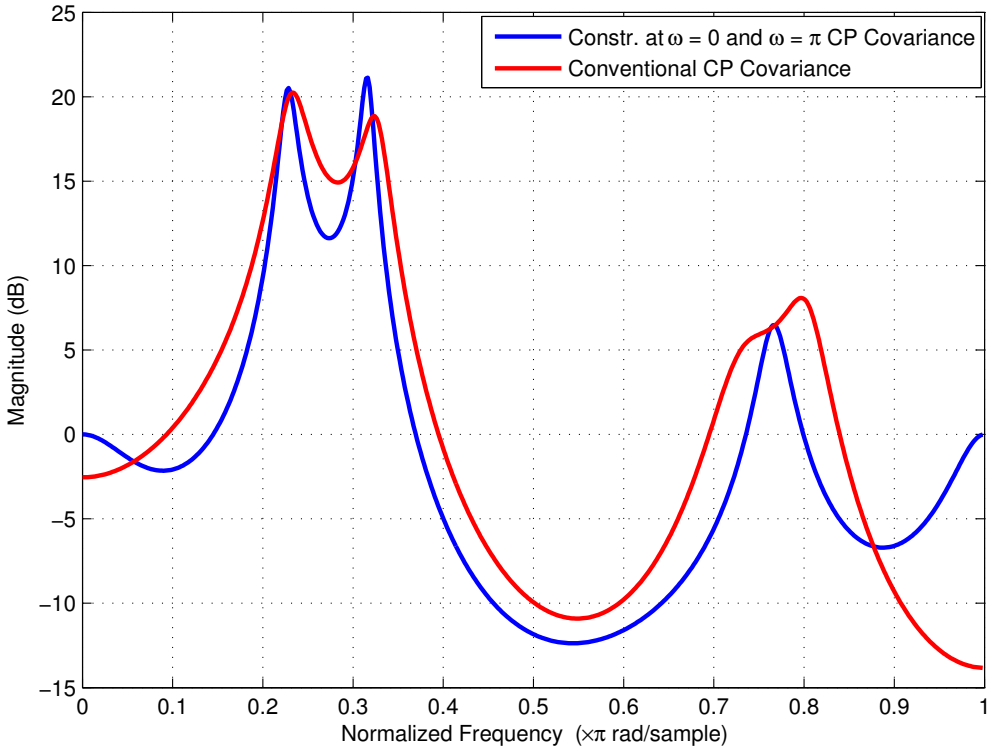
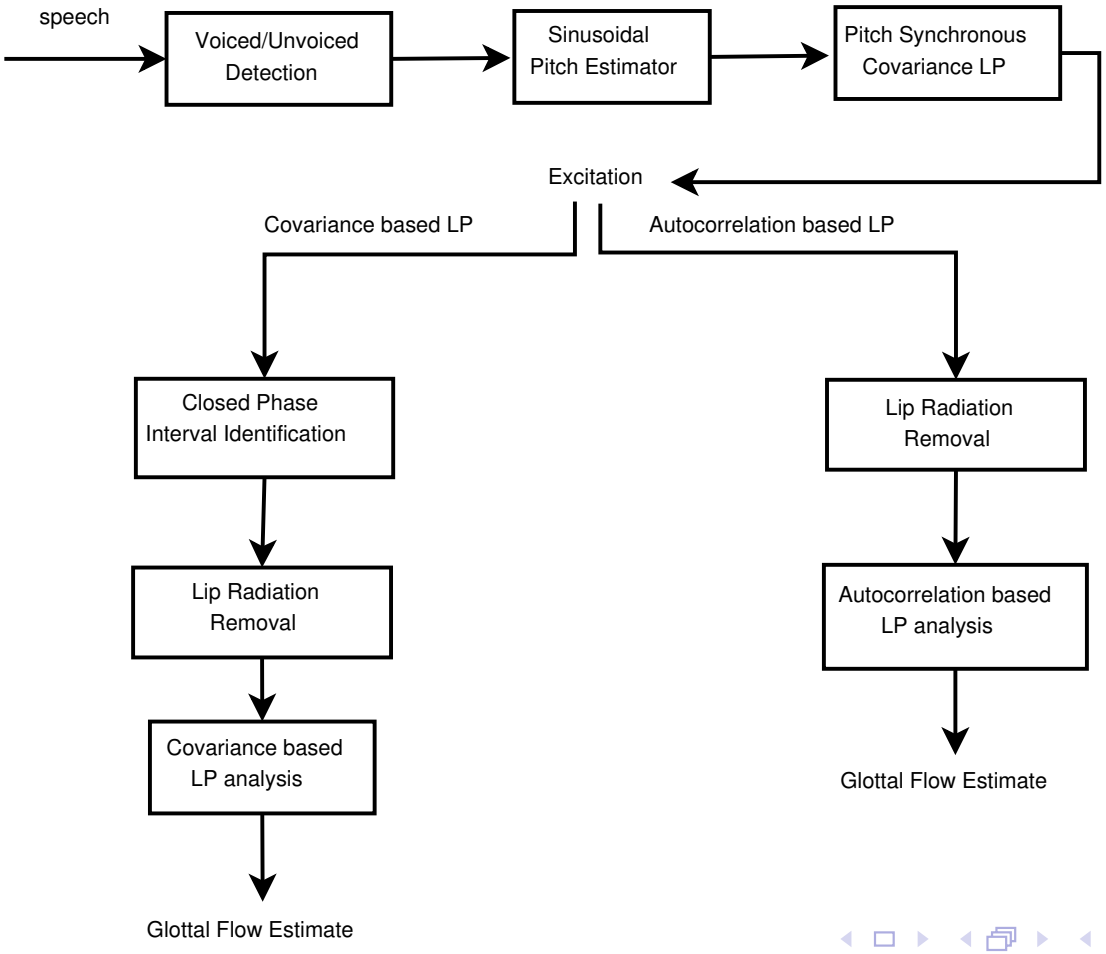


Figure: Examples of all-pole spectra computed in the closed phase covariance analysis by the conventional LP and by the DC-π constrained LP.

Outline

Inverse Filtering Procedure - Flowchart



Inverse Filtering Procedure - Details

- Sampling Frequency: $f_s = 8$kHz.
- Order of LP Analysis: $p = 10$.
- The lip radiation effect was canceled by a first order all-pole filter with its pole at $z = 0.999$.
- Autocorrelation approaches:
  - Analysis window: hanning type, 250 ms duration.
  - M-parameter for SWLP: $M = 8$, $M = 24$.
- Covariance approaches:
  - Analysis window: rectangular type,
  - Duration determined by the detected closed phase interval[6].
  - Analysis frame rate: one pitch period
- Inverse filtered speech signals were computed in a frame by frame basis.
- Two local pitch periods and a frame rate of one pitch period was applied.
- The overall glottal flow was synthesized using the Overlap-Add (OLA) method.

Outline

Metrics for evaluating the IF techniques

- Signal to Reconstruction Error Ratio - SRER
  - SRER is a standard index for measuring the effectiveness of modeling a waveform and is defined as:

$$SRER = 20log10\Big(\frac{\sigma_{s[n]}}{\sigma_{e[n]}}\Big) \tag{13}$$

    where $s[n]$ is the original (or true) glottal flow signal in our case, $e[n]$ is the modeling (or reconstruction) error, $e[n] = s[n] - \hat{s}[n]$, and σ denotes the corresponding standard deviation.
  - SRER was computed from the overall glottal flow waveforms.
- Difference between the first two harmonics - H1-H2
  - H1-H2 is an index of the spectral decay (or spectral tilt) of the glottal spectrum.
  -

$$ER_{H1H2} = \Big|Ref_{H1H2} - Est_{H1H2}\Big| \tag{14}$$

    where Ref_{H1H2} and Est_{H1H2} denote the H1-H2 metric for the true (or reference) and the estimated glottal airflow, respectively.
  - For a good estimation ER_{H1H2} should be close to zero.

Examples

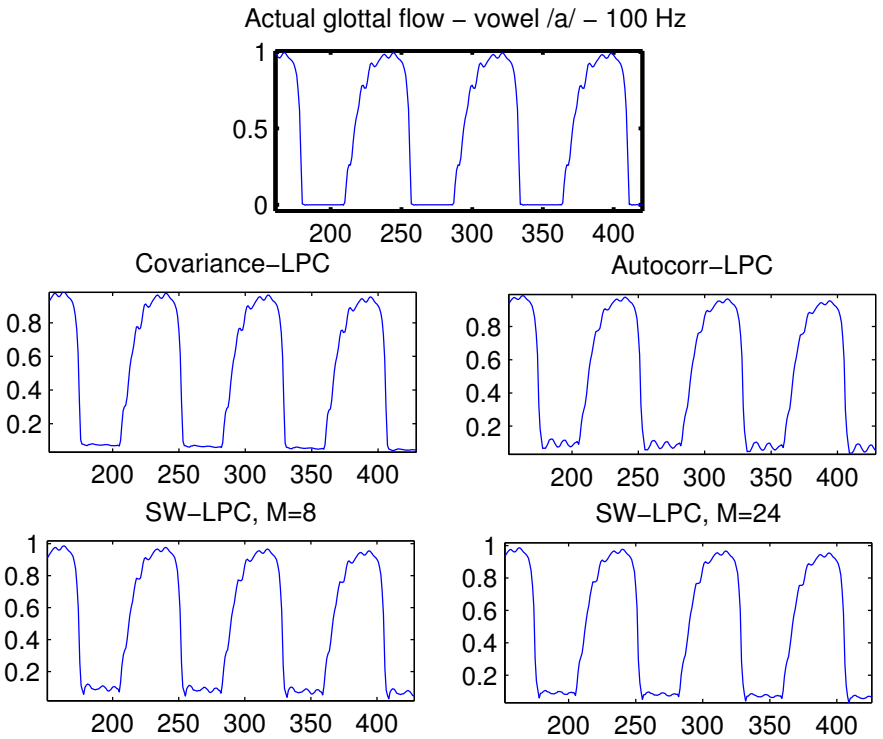


Figure: Glottal flow estimates for vowel /aa/ of $f_0 = 105$ Hz. Upper panel: Original glottal flow. Middle panel: Covariance (left) and Autocorrelation (right) based glottal flow estimates. Lower panel: SWLP with $M = 8$ and $M = 24$ glottal flow estimates. In all panels, time is indicated in samples.

Examples

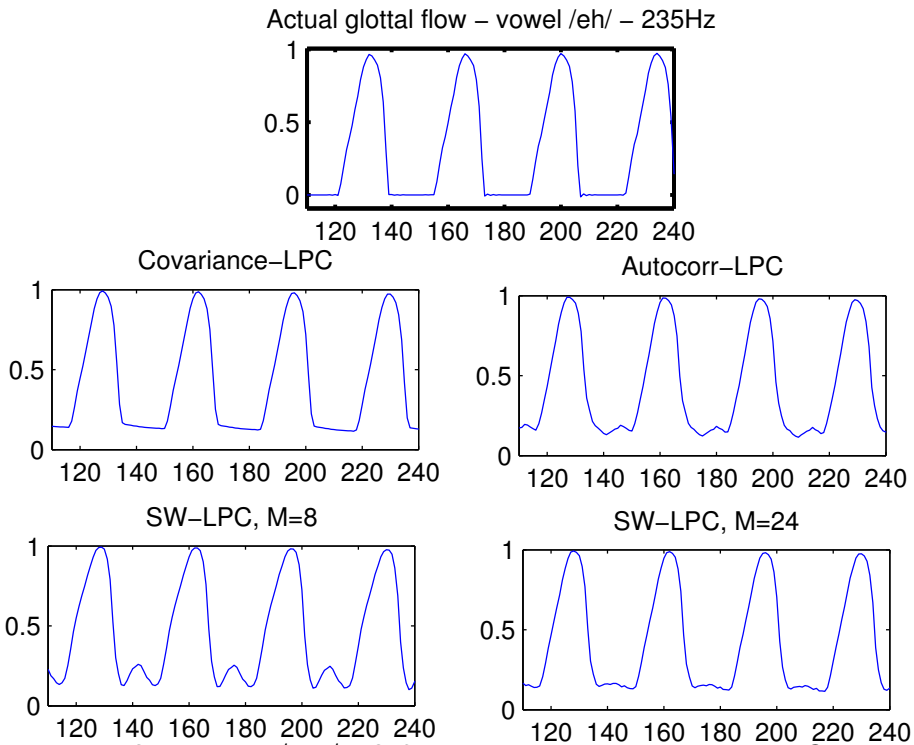


Figure: Glottal flow estimates for vowel /eh/ of $f_0 = 230$ Hz. Upper panel: Original glottal flow. Middle panel: Covariance (left) and Autocorrelation (right) based glottal flow estimates. Lower panel: SWLP with $M = 8$ and $M = 24$ glottal flow estimates. In all panels, time is indicated in samples.

Results - SRER

SRER					
Vowel	$SWLP_8$	$SWLP_{24}$	LPC	CovLPC	CLPC
/aa/	33.5 ($\pm$2.0)	39.7 ($\pm$4.5)	36.2 ($\pm$5.7)	41.9 ($\pm$6.3)	41.5 ($\pm$6.6)
/ae/	32.7 ($\pm$4.4)	35.2 ($\pm$2.9)	37.8 ($\pm$3.0)	40.4 ($\pm$6.4)	40.9 ($\pm$6.9)
/eh/	34.0 ($\pm$1.9)	38.4 ($\pm$4.2)	33.9 ($\pm$4.0)	40.5 ($\pm$5.2)	41.1 ($\pm$7.4)
/ih/	32.3 ($\pm$1.5)	37.6 ($\pm$3.1)	35.3 ($\pm$4.6)	39.2 ($\pm$5.6)	40.8 ($\pm$8.7)

Table: Mean and standard deviation of the SRER value for each vowel (all 8 frequencies) and method is illustrated. LPC stands for autocorrelation LP, CovLPC for Closed Phase covariance LP, CLPC for Constrained Closed Phase covariance LP, $SWLP_8$ for SWLP with $M = 8$, and $SWLP_{24}$ for SWLP with $M = 24$

Results - H1-H2

ER_{H1H2}					
Vowel	$SWLP_8$	$SWLP_{24}$	LPC	CovLPC	CLPC
/aa/	0.68 ($\pm$0.10)	0.23 ($\pm$0.09)	0.75 ($\pm$0.09)	0.20 ($\pm$0.20)	0.08($\pm$0.28)
/ae/	0.15 ($\pm$0.12)	0.15 ($\pm$0.05)	0.55 ($\pm$0.05)	0.18 ($\pm$0.13)	0.07($\pm$0.30)
/eh/	0.34 ($\pm$0.09)	0.30 ($\pm$0.07)	0.54 ($\pm$0.08)	0.38 ($\pm$0.17)	0.16($\pm$0.30)
/ih/	0.72 ($\pm$0.14)	0.39 ($\pm$0.11)	0.85 ($\pm$0.12)	0.35 ($\pm$0.24)	0.34($\pm$0.50)

Table: Mean and the standard deviation of ER_{H1H2} for each vowel (all 8 frequencies) and each method is illustrated. LPC stands for autocorrelation LP, CovLPC for CP covariance LP, CLPC for Constrained Closed Phase covariance LP, $SWLP_8$ for SWLP with $M = 8$, and $SWLP_{24}$ for SWLP with $M = 24$

Outline

Conclusions

- For both metrics, it is obvious that $SWLP_{24}$ outperforms conventional autocorrelation LP.

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- As expected, covariance methods prevails for both metrics.

Future Work

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- Real time system approach to IF.

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Acknowledgments (say thanks to everyone!)

Time for Questions!

Any questions? (Hope not...)

Τέλος Ενότητας

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Σημείωμα Αναφοράς