

On the (Glottal) Inverse Filtering of Speech Signals

AN INTRODUCTION

CS578-DIGITAL SPEECH SIGNAL PROCESSING

INVITED LECTURE

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On the (Glottal) Inverse Filtering of Speech Signals

- Introduction
- Inverse Filtering Techniques
- Conclusions

Introduction

On the (Glottal) Inverse Filtering of Speech Signals

- The human speech production system is a complicated system
- From an engineering point of view, it can be roughly divided into three parts [1]
 - The vocal folds, which is the source of the system
 - The vocal tract filter, which is the path from the vocal folds to the lips
 - The lip radiation, which is the final bound before system output

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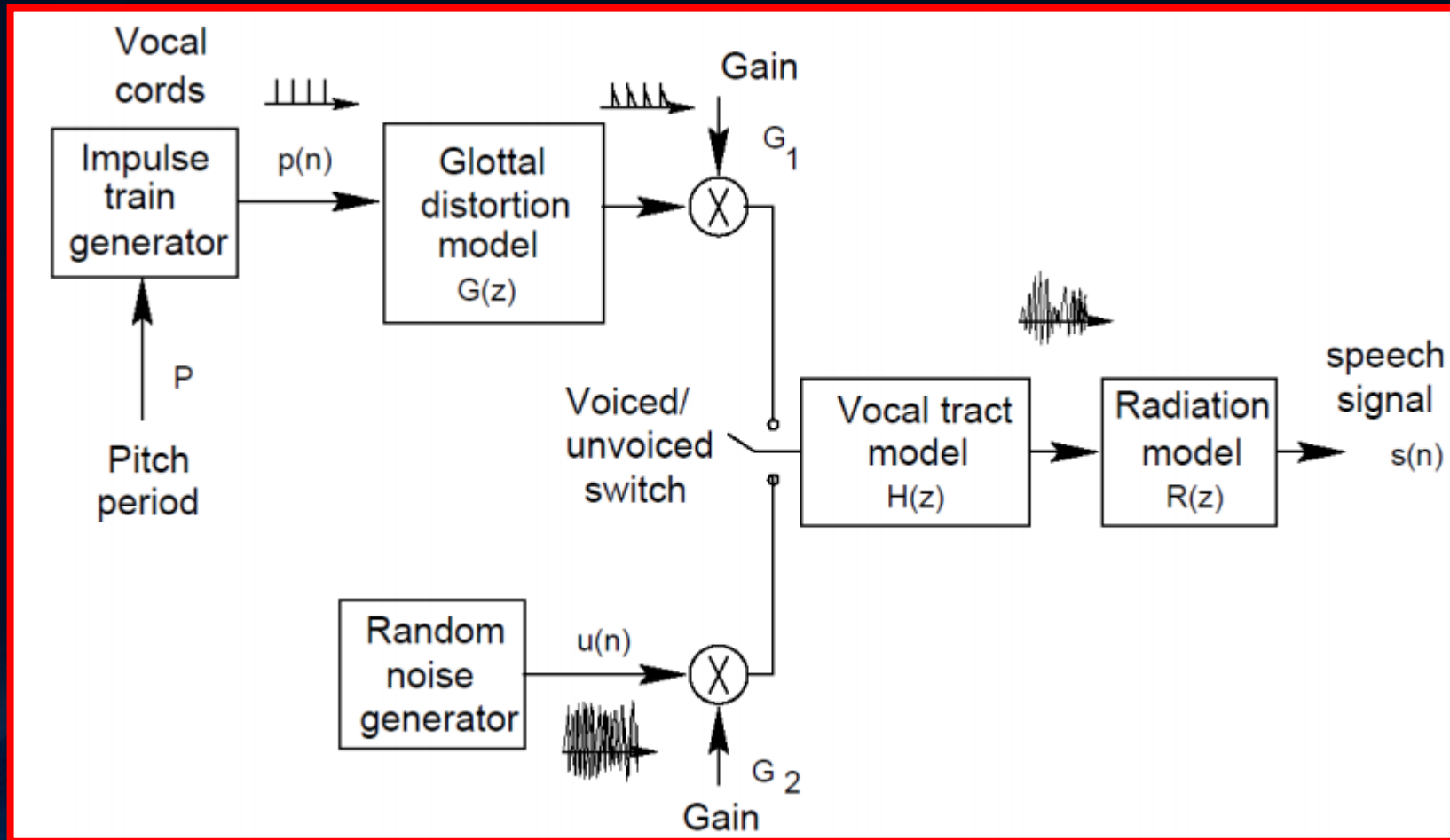
- Based on this simplification, voiced speech can be modeled as a linear filtering operation:

$$s(t) = g(t) * h(t) * r(t) \leftrightarrow S(z) = G(z)H(z)R(z)$$

where $*$ denotes convolution and

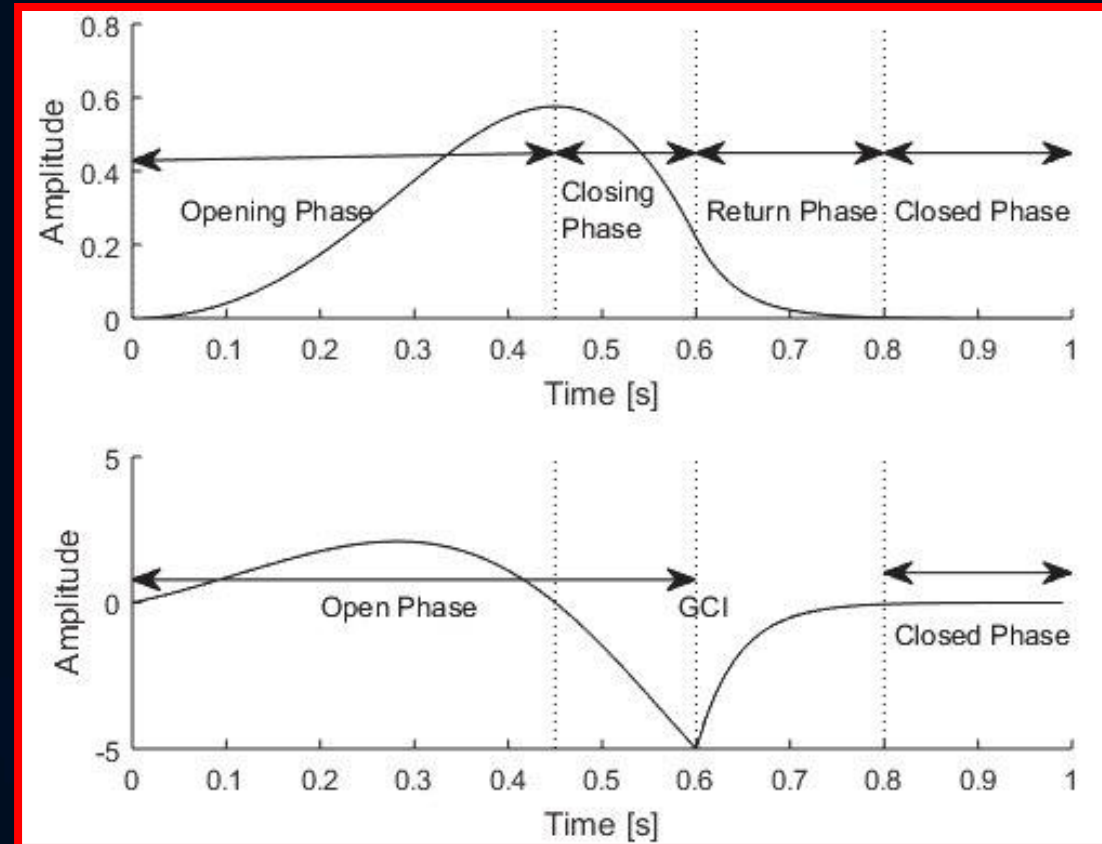
- $g(t)$ is the glottal airflow velocity waveform
- $h(t)$ is the vocal tract filter
- $r(t)$ is the lip radiation filter

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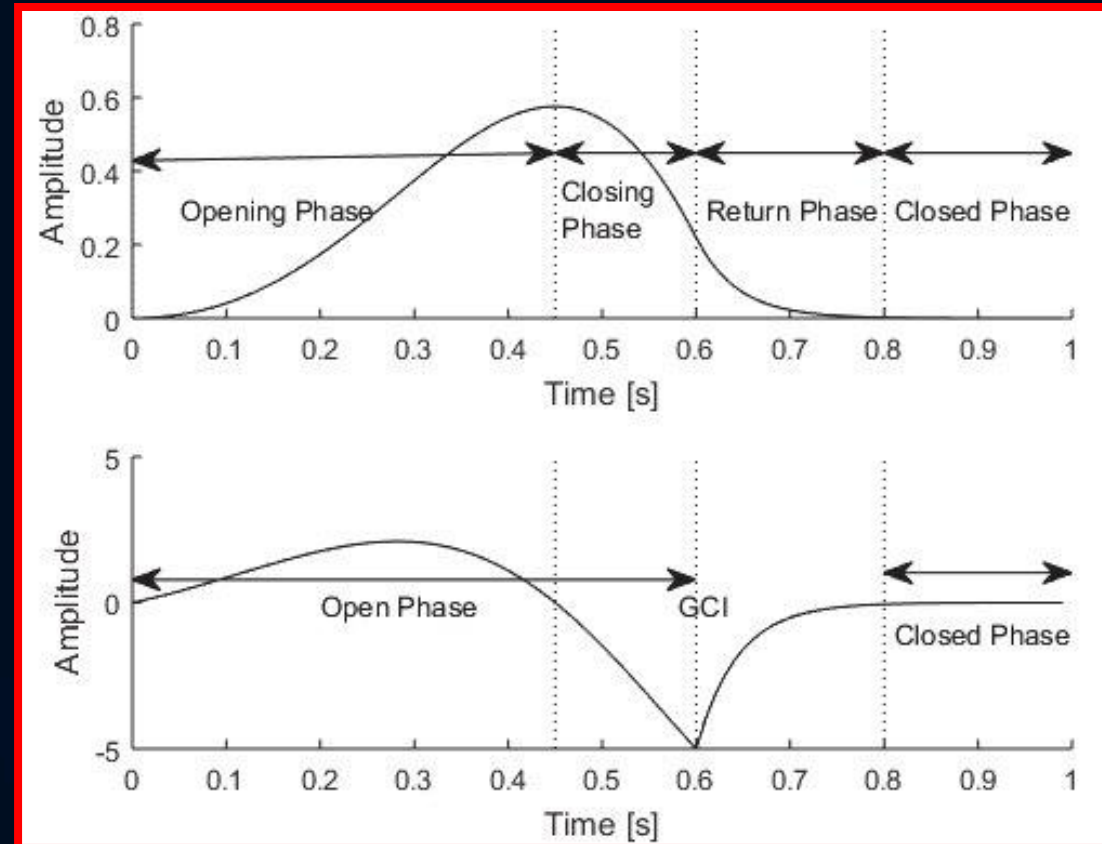
On the (Glottal) Inverse Filtering of Speech Signals

- What is Glottal Inverse Filtering (GIF)?
 - GIF refers to techniques for obtaining the source of voiced speech, the glottal airflow velocity waveform, from voiced speech itself [10]
 - How does this signal look like?
 - **Open phase:** air flows through the glottis
 - **Return phase:** vocal folds are snapping shut
 - **Closed phase:** glottis is shut and airflow velocity is zero



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- While radiation occurs after the vocal tract filter, we often combine $G(z)$ and $R(z)$ into a single expression
 - This applies the radiation effect to the glottal source before it enters the vocal tract
 - Effect of differentiation on the source
- The resulting signal is the so-called **glottal flow derivative**
- Very commonly used in literature



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- Why bother?
 - Basic research of speech production
 - Applications to speech analysis, synthesis, and modification
 - Environmental voice care
 - Voice pathology detection
 - Analysis of the emotional content of speech
 - Voice source modeling for TTS

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- Basic idea:
 - Form a computational model for the vocal tract filter, $H(z)$
 - Cancel its effect from the speech waveform by filtering the speech signal through the inverse of the model, $\frac{1}{H(z)}$

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- **Problem:**

- The actual glottal flow waveform IS NOT AVAILABLE!
- ...at least in a non-invasive manner [18]

- **Approaches:**

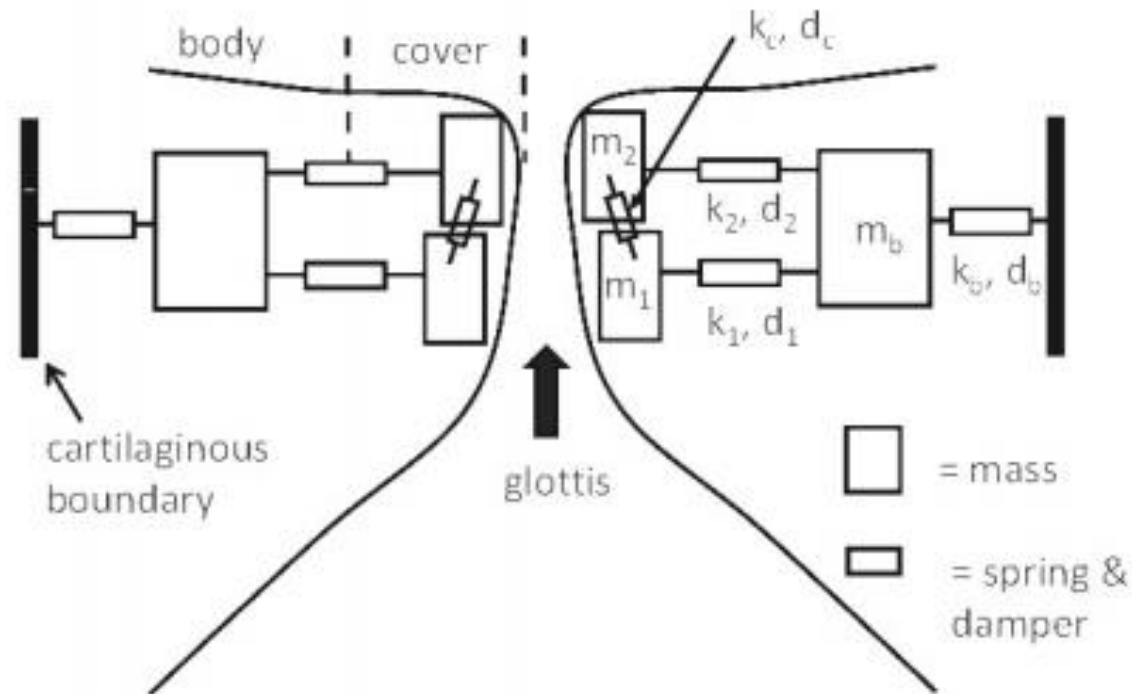
- “Visual” inspection of the resulting glottal flow waveform
 - Use of synthetic speech signal produced by a known artificial excitation
 - Compare the results of different GIF algorithms
-
- None of the previous approaches is truly objective

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- One solution is to build a physical model of the speech production mechanism
 - Generate waveforms from this model
- Time-varying waveforms are simulated
- Such waveforms are expected to provide a more firm and realistic test of GIF methods
- **Both** the speech output **and** the source are available
- A well known dataset of such signals is described in [2,6]

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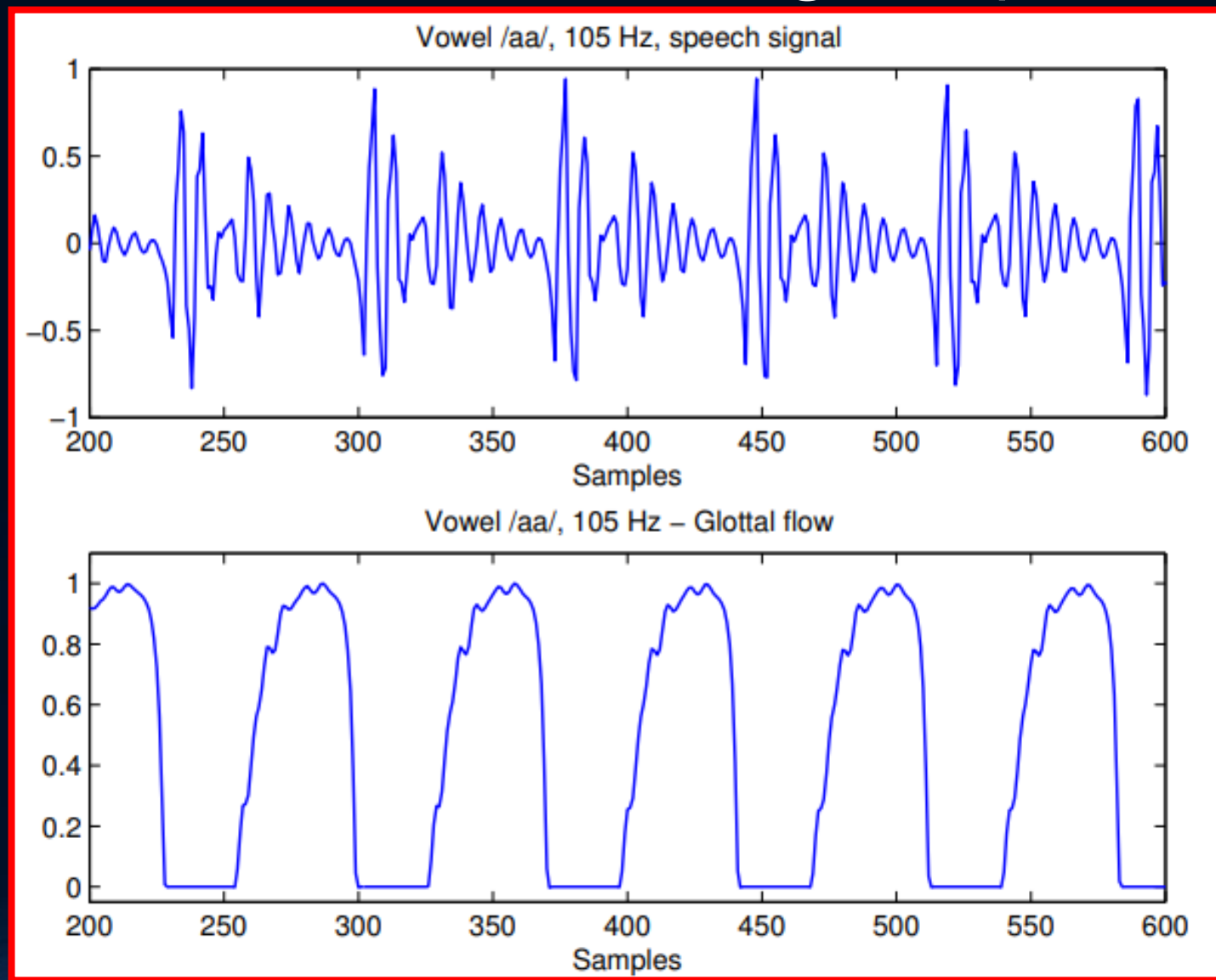
- ▶ In detail, self sustained vocal fold vibration was simulated with three masses coupled to one another through stiffness and damping elements.



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- The model has a parametrized input such as
 - Lung pressure
 - Prephonatory glottal half-width (adduction)
 - Vocal fold length and thickness
 - Activation levels of the cricothyroid and thyroarytenoid muscles

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GIF techniques

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- Since we already know about Linear Prediction (LP), we will discuss GIF methods based only on that
- You already know two methods for estimating LP coefficients
 - Autocorrelation method: zero samples outside prediction error interval – minimize MSE everywhere
 - Covariance method: non-zero samples outside prediction error interval – minimize MSE inside prediction error interval

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- LP is used to produce all-pole models of the vocal tract filter

$$H(z) = \frac{1}{\sum_{k=1}^p a_k z^{-k}}$$

where p is the filter order and a_k are the LP coefficients

- In general, LP minimizes the MSE over a region R

$$E = \sum_R e^2[n]$$

where $e[n] = s[n] - \sum_{k=1}^p a_k s[n-k]$

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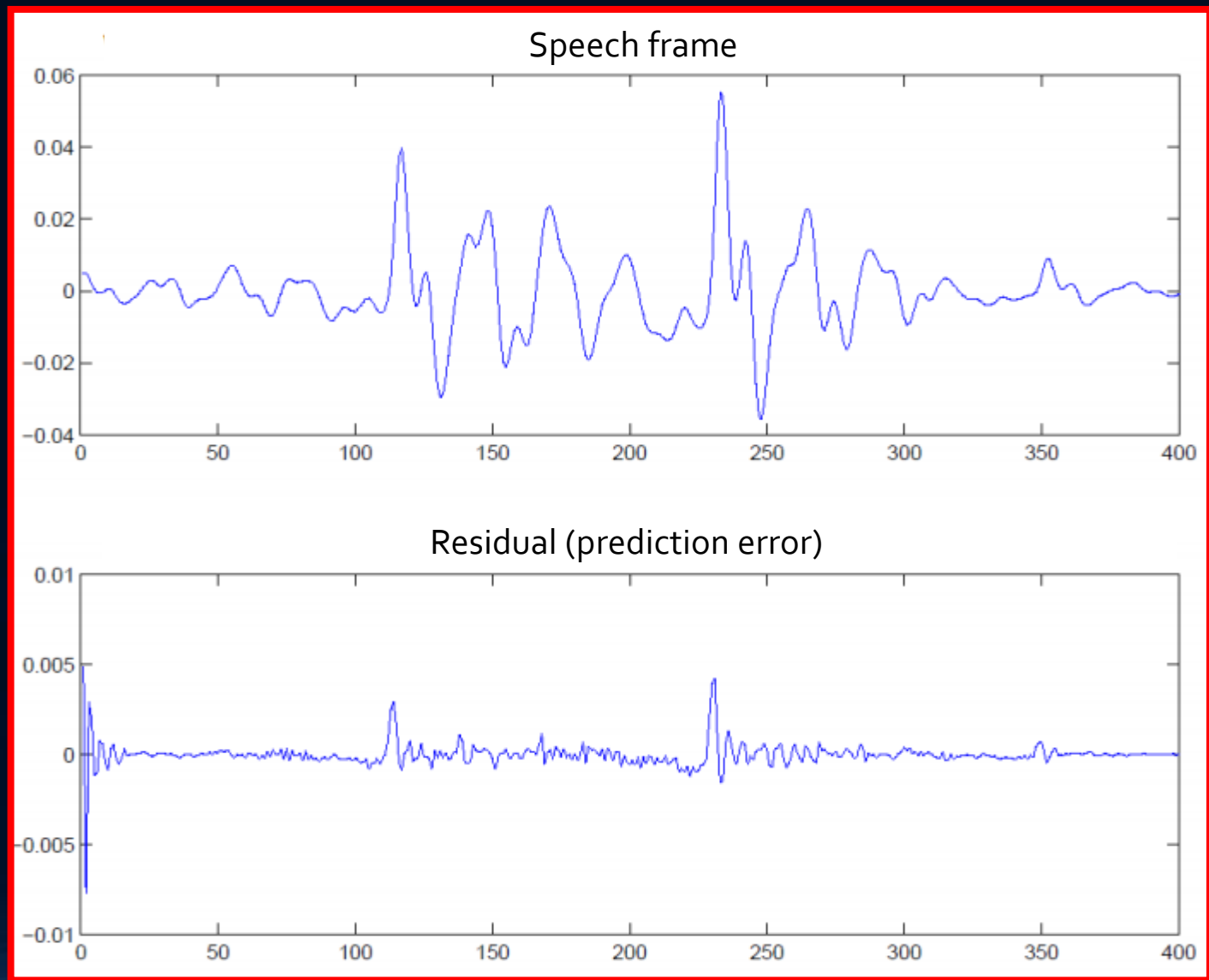
- How can we find the source excitation through LP analysis?
- If we consider speech as an AR process

$$s[n] = Ag[n] + \sum_{k=1}^p a_k s[n-k] \Rightarrow Ag[n] = s[n] - \sum_{k=1}^p a_k s[n-k]$$

then minimization of the MSE leads to $e[n] \approx Ag[n]$

- Thus, the **prediction error** (or **residual**) can be thought of an estimation of the source excitation
- But how are glottal source and residual related?

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- But how are glottal source and residual **related**?
- As you've seen, the two signals do not quite match
- The reason is that the Z transform of speech is a combined transfer function

$$Y(z) = G(z)R(z)H(z)$$

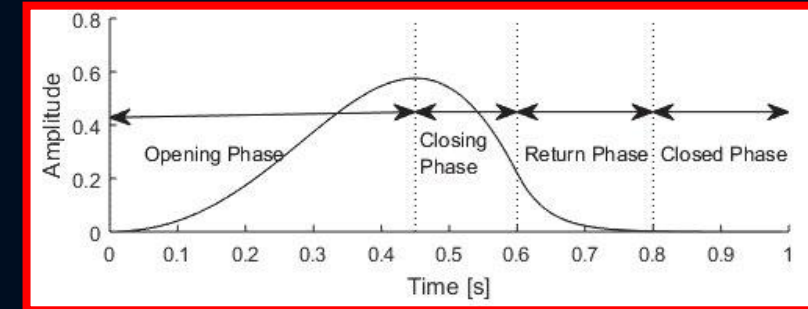
- $G(z)$ can be further decomposed as an impulse sequence passed through a glottal filter: $G(z) = I(z)U(z)$, where $I(z)$ is the impulse sequence
- Thus $Y(z)$ contains
 - ❑ Zeros from the glottal source and lip radiation
 - ❑ Poles from the glottal filter and the vocal tract filter

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- LP analysis provides an overall transfer function $\hat{H}(z)$ where all these contributions are combined!
 - ...not to mention that we're using an all-pole method for a pole-zero signal...
- So what we are cancelling via simple LP-based GIF is this overall estimation
 - ...resulting into something that looks like a series of impulses!
- So how can we work this out?
- Identify instants where there is no interaction between the source and the filter!

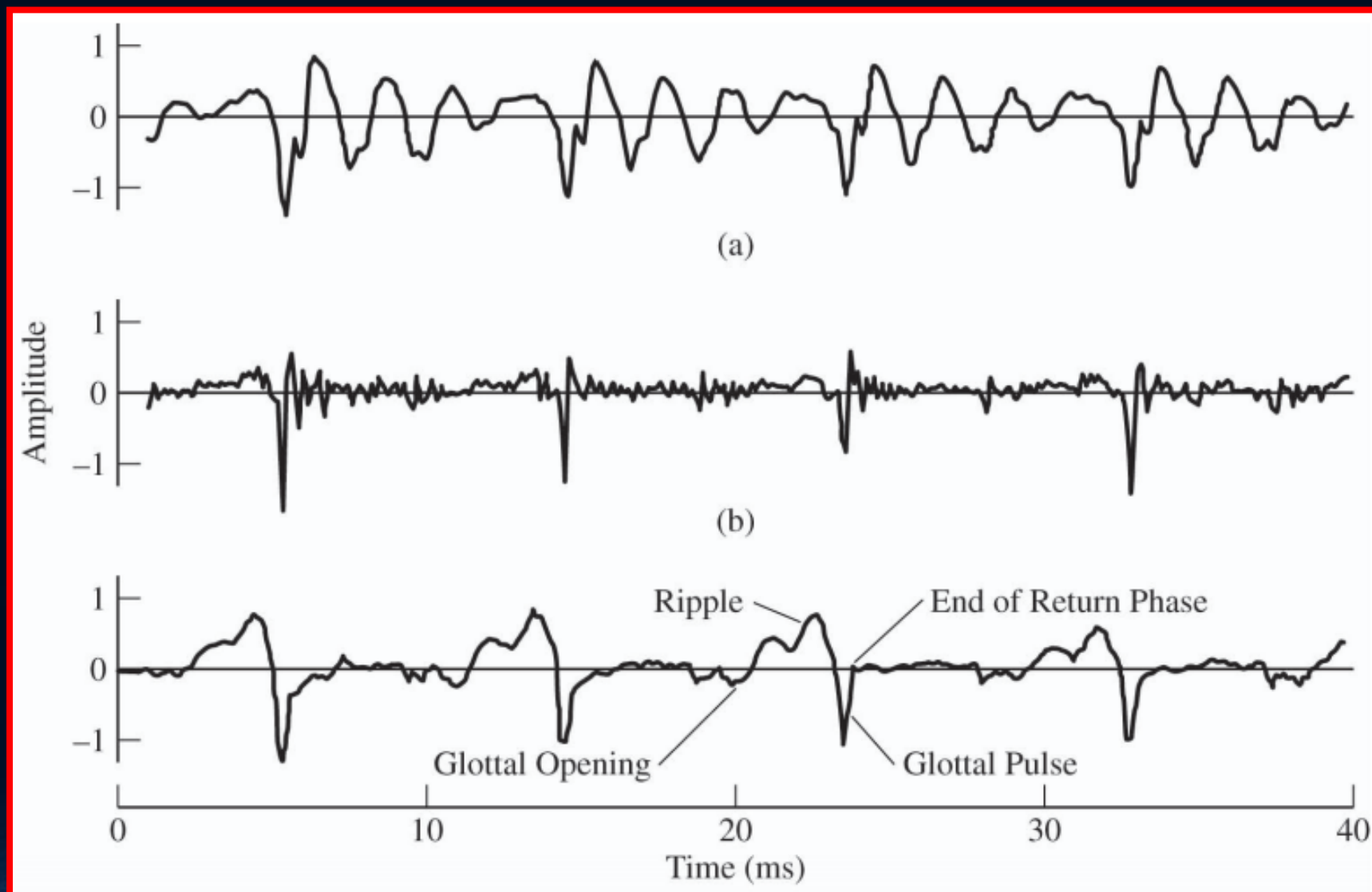
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- **Closed-phase analysis**
- Identify regions where the vocal folds are closed
 - No contribution from $G(z)$, the speech signal should contain vocal tract and radiation factors $H(z)R(z)$
 - $R(z)$ can be modeled as a differentiator (single-zero FIR filter), so it can be cancelled by a simple integrator
- Vocal tract estimation in the closed phase region leads to a more precise result
- Estimation in the closed phase → cancelling vocal tract via GIF over the whole pitch period
- Use of covariance-based LP on the closed-phase [9]



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- Closed-phase analysis

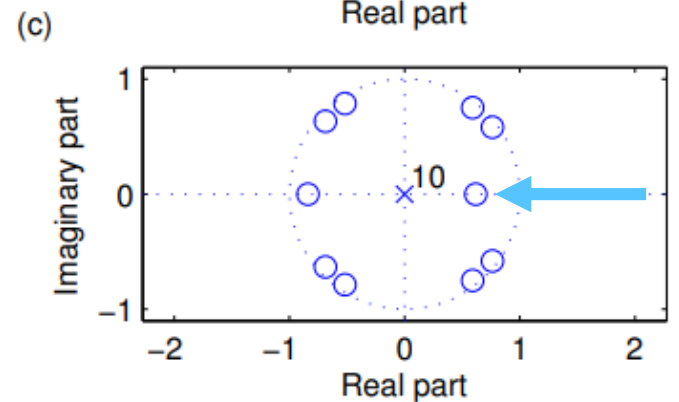
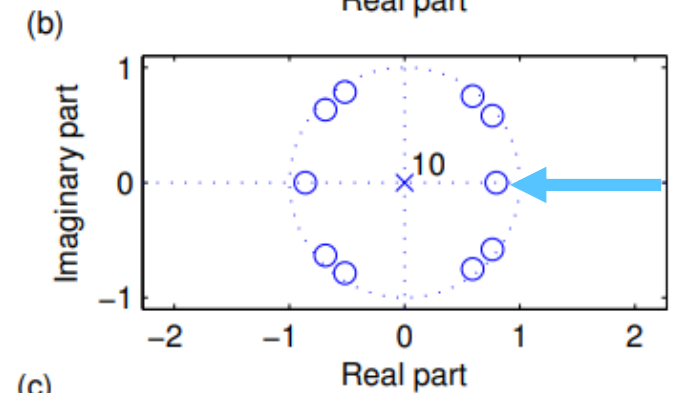
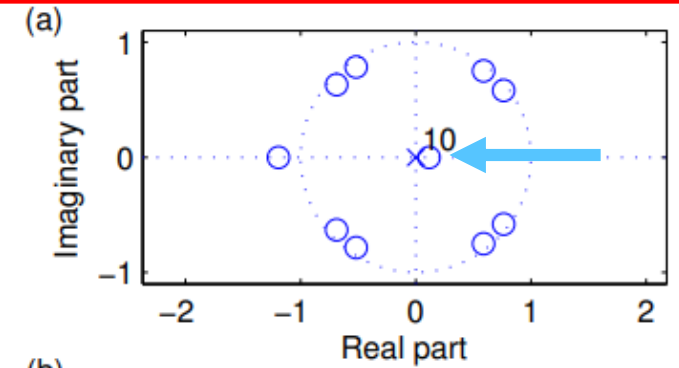
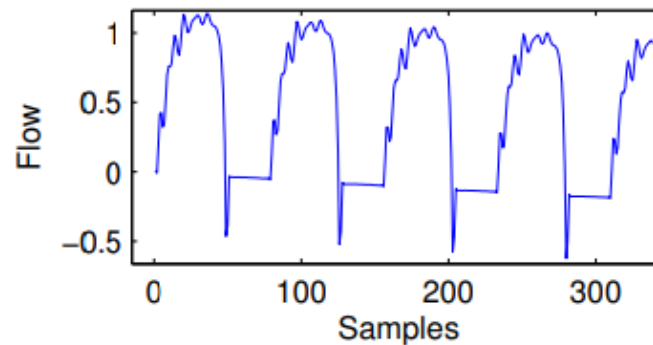
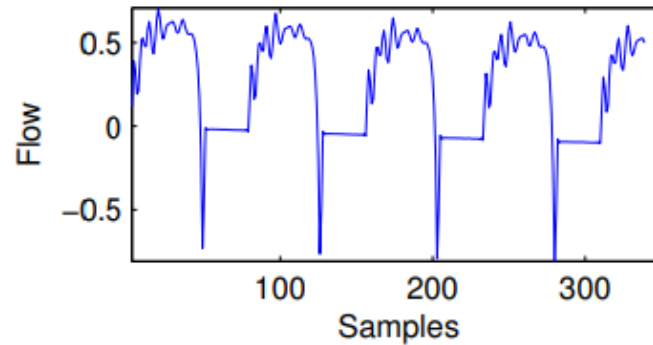
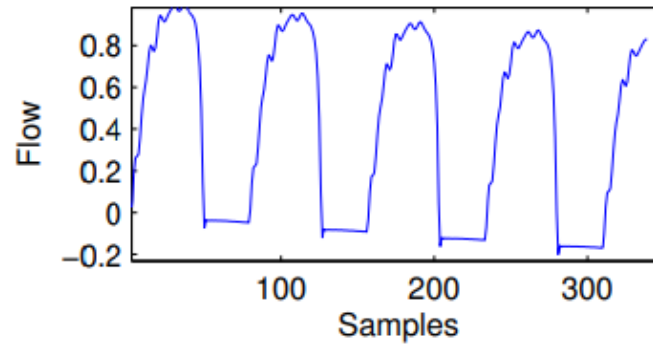


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- **However**, standard closed-phase covariance LP suffers from certain shortcomings
 - ❑ **Short closed phase duration (especially for high pitched speakers)**
 - Too few samples to obtain a good estimation
 - ❑ **Sensitivity to the exact position of the covariance frame**
 - Small variation from the exact closed phase interval produces artifacts
 - ❑ **Vocal tract filter instability**
 - Covariance-based LP does not guarantee a stable filter
 - Inverse filter might not be minimum phase

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- Frame position sensitivity

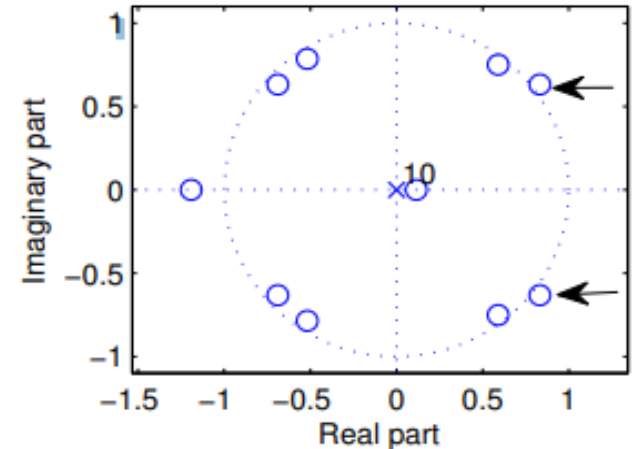
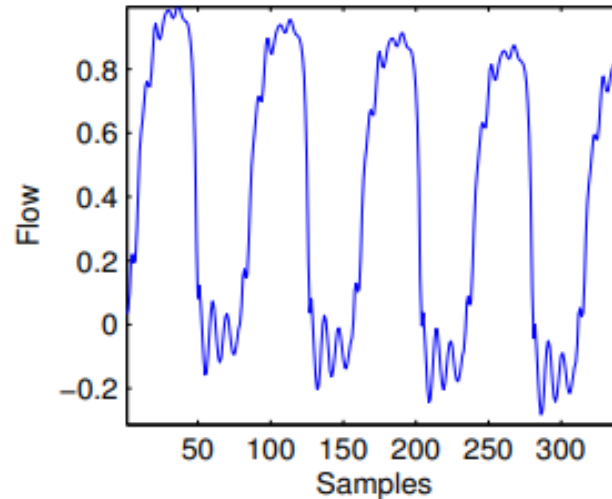
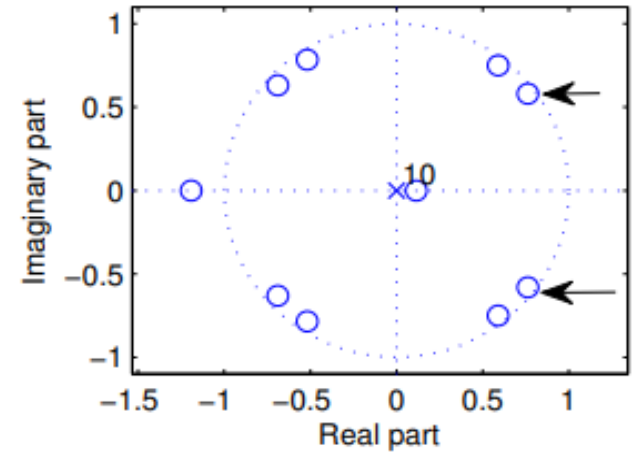
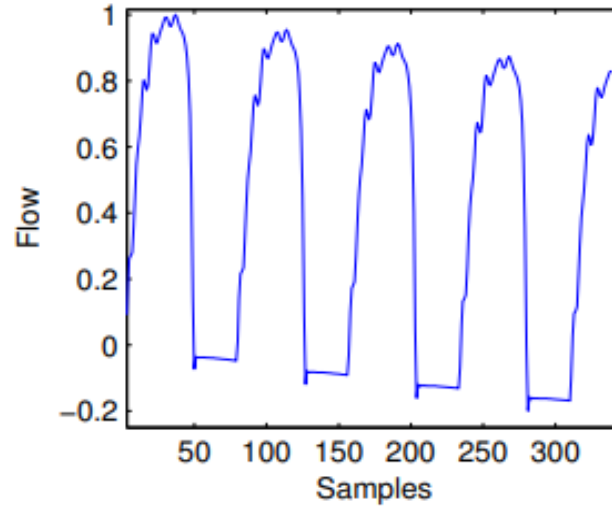


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- The effect of an inverse filter root which is located on the positive real axis has the properties of a first order differentiator, when the root approaches the unit circle
- A similar effect is also produced by a pair of complex conjugate roots at low frequencies
- This distortion is more apparent at the time instants where the glottal flow changes more rapidly, that is, near glottal closure
- The presence of such roots are in contrast to the source-filter suggested theory
- The removal of such roots results in less dependency on the covariance frame location

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- Non-minimum phase inverse filter



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- The inverse filter $1/H(z)$ might **not** be minimum phase
- As we know from basic DSP, it can become minimum phase by replacing each zero by its mirror image partner
- That leaves the magnitude spectrum unchanged
- The phase characteristics change, though

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- **Constrained Covariance-based Closed-Phase LP [3]**
- Idea: modification of the conventional CP covariance analysis in order to provide more realistic root locations, in the acoustic sense
- 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
 - DC-constraint

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- **Constrained Covariance-based Closed-Phase LP [3]**
- DC-constraint:

$$H(e^{j0}) = \sum_{k=0}^p a_k e^{-j0n} = \sum_{k=0}^p a_k = l_{DC}$$

- Why?
 - Magnitude response of voiced sounds approaches unity at zero frequency [1]
 - 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 magnitude response to the source-filter theory

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- **Constrained Covariance-based Closed-Phase LP [3]**
- Constrained convex minimization problem
- Minimize $a^T \Phi a$ subject to $\Gamma^T a = b$

$$a = [1, a_1, \dots, a_p]^T \quad \Phi = [\Phi_{ij}], \Phi_{ij} = \sum_{n=0}^{N-1} s[n-i]s[n-j], \quad 1 \leq i, j \leq p$$

$$b = [1, l_{DC}]^T \quad \Gamma = \begin{bmatrix} 1 & 1 \\ 0 & 1 \\ \vdots & \vdots \\ 0 & 1 \end{bmatrix}^T$$

Solution:

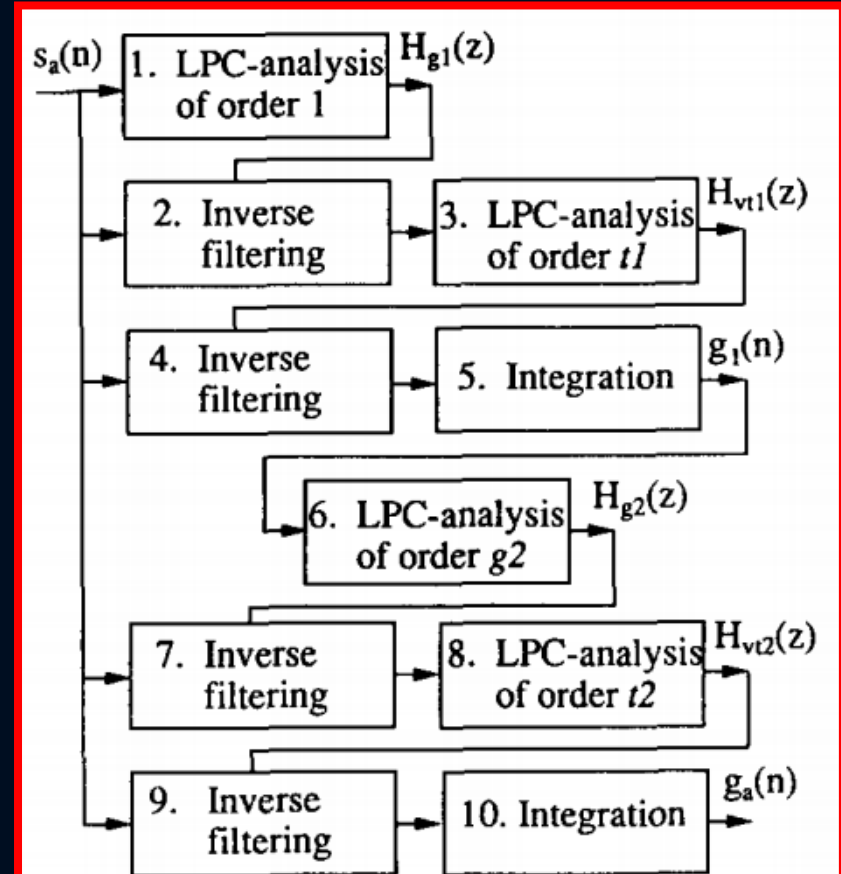
$$a = \Phi^{-1} \Gamma (\Gamma^T \Phi^{-1} \Gamma)^{-1} b$$

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- **Still**, the computational load of covariance-based LP along with its shortcomings (cases of very small CP, frame dependent, CP identification) might make the method not appropriate
- Idea: use autocorrelation method with “enhancements”
 - Fast & stable
 - Not optimal but good enough
 - Try to introduce “enhancements”
 - Try to approach performance of CP analysis without detecting CP

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- **Iterative Adaptive Inverse Filtering [4]**
- An iterative method for obtaining the glottal source
- Motivation:
 - A priori knowledge of the overall shape of the vocal tract
 - Cancel the tilting effect of the glottal source
 - Estimate vocal tract filter



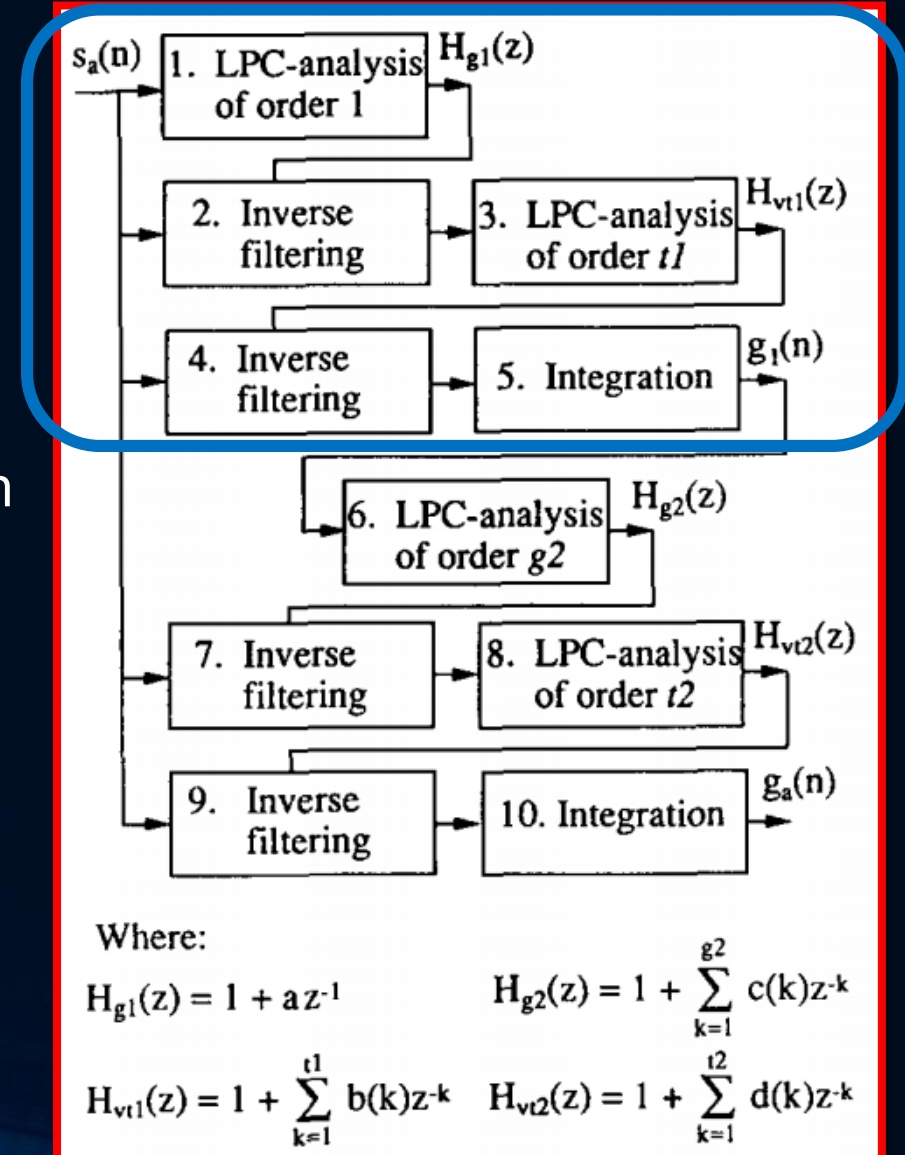
Where:

$$H_{g1}(z) = 1 + az^{-1} \quad H_{g2}(z) = 1 + \sum_{k=1}^{g2} c(k)z^{-k}$$

$$H_{vt1}(z) = 1 + \sum_{k=1}^{t1} b(k)z^{-k} \quad H_{vt2}(z) = 1 + \sum_{k=1}^{t2} d(k)z^{-k}$$

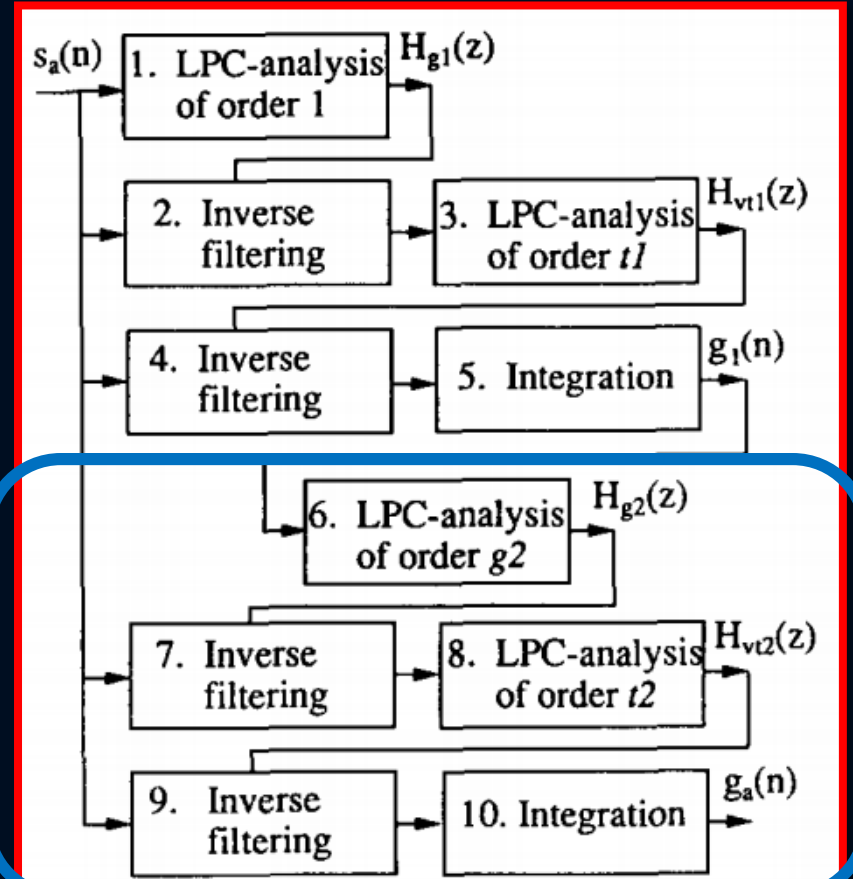
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- Iterative Adaptive Inverse Filtering [4]
- First iteration
- 1. LPC of order 1 to model the effect of the glottal source on the speech spectrum
- 2. Cancel 1.
- 3. LPC of high order to model the vocal tract
- 4&5. Cancel vocal tract and lip radiation



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- Iterative Adaptive Inverse Filtering [4]
- Second iteration
- 6. LPC of order 2-4 to more accurately model the effect of the glottal source on the speech spectrum
- 7. Cancel 6.
- 8. LPC of high order to model the vocal tract
- 9&10. Cancel vocal tract and lip radiation



Where:

$$H_{g1}(z) = 1 + az^{-1} \quad H_{g2}(z) = 1 + \sum_{k=1}^{g2} c(k)z^{-k}$$

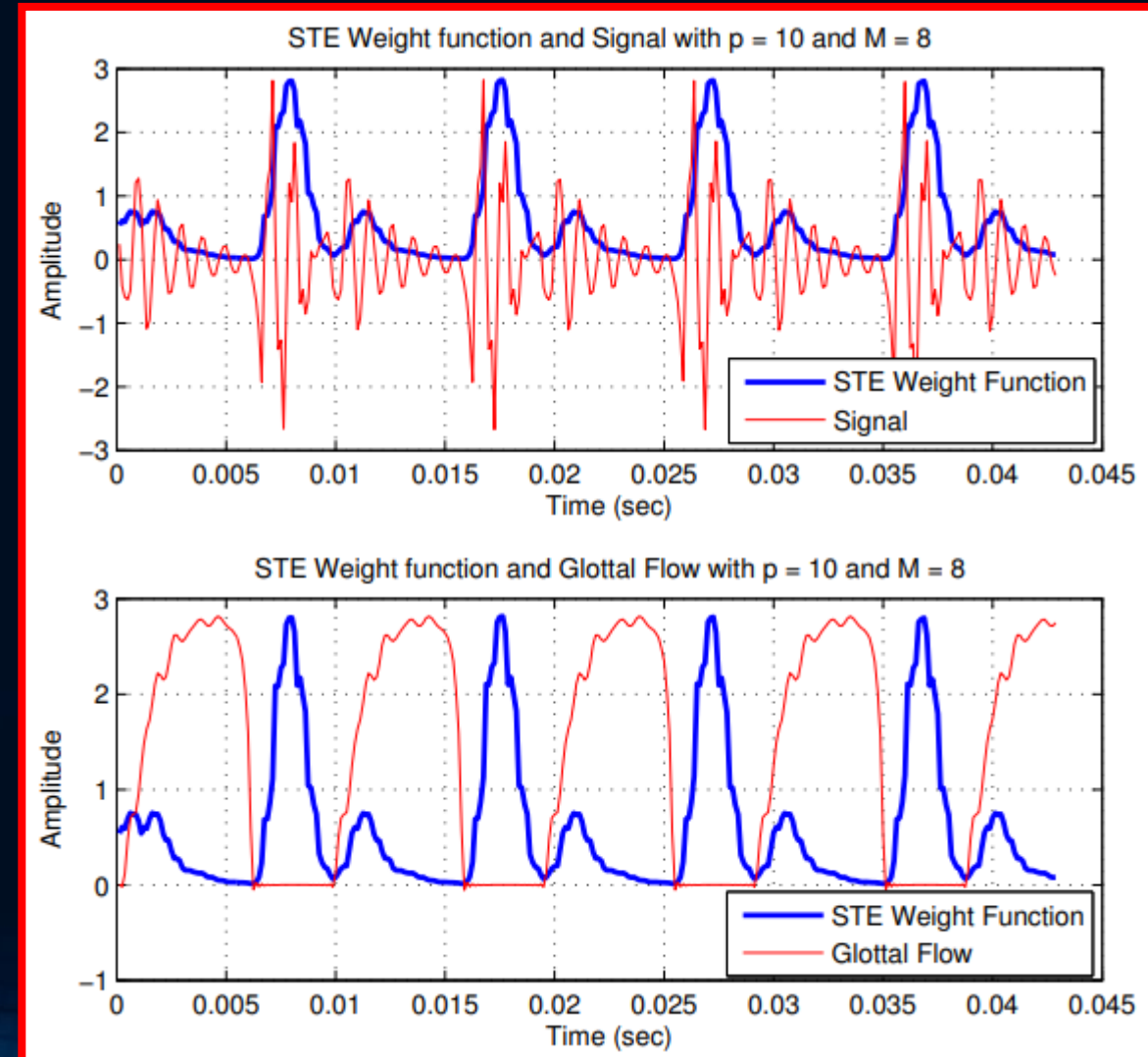
$$H_{vt1}(z) = 1 + \sum_{k=1}^{t1} b(k)z^{-k} \quad H_{vt2}(z) = 1 + \sum_{k=1}^{t2} d(k)z^{-k}$$

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- **Stabilized Weighted Linear Prediction [5,7]**
- An all-pole method based on Weighted Linear Prediction (WLP)
- Idea: use standard autocorrelation method but give more weight to some samples of the autocorrelation matrix compared to others
- How to give more weight?

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- Stabilized Weighted Linear Prediction
- Compute *the short time energy (STE)* of the signal
- High energy samples fall in the closed phase region!



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- **Stabilized Weighted Linear Prediction**
- STE function emphasizes the speech samples of large amplitude, which typically occur during the closed phase interval
- By emphasizing on these samples that occur during the glottal closed phase, it is likely to yield more robust acoustical cues for the formants
- The method depends on a parameter M , the energy window length
- 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

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- Stabilized Weighted Linear Prediction

- STE:

$$w_n = \sum_{i=0}^{M-1} x^2[n - i - 1]$$

- Prediction error energy:

$$E = \sum_{n=1}^{N+p} e^2[n] w_n = a^T \left(\sum_{n=1}^{N+p} w_n x[n] x^T[n] \right) a = a^T R a$$

where

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

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- **Stabilized Weighted Linear Prediction**
- Constrained minimization problem (again 😊)
- Minimize E subject to $a^T u = 1$, where $u = [1, 0, 0, \dots, 0]^T$
- It can be shown that a satisfies the linear equation

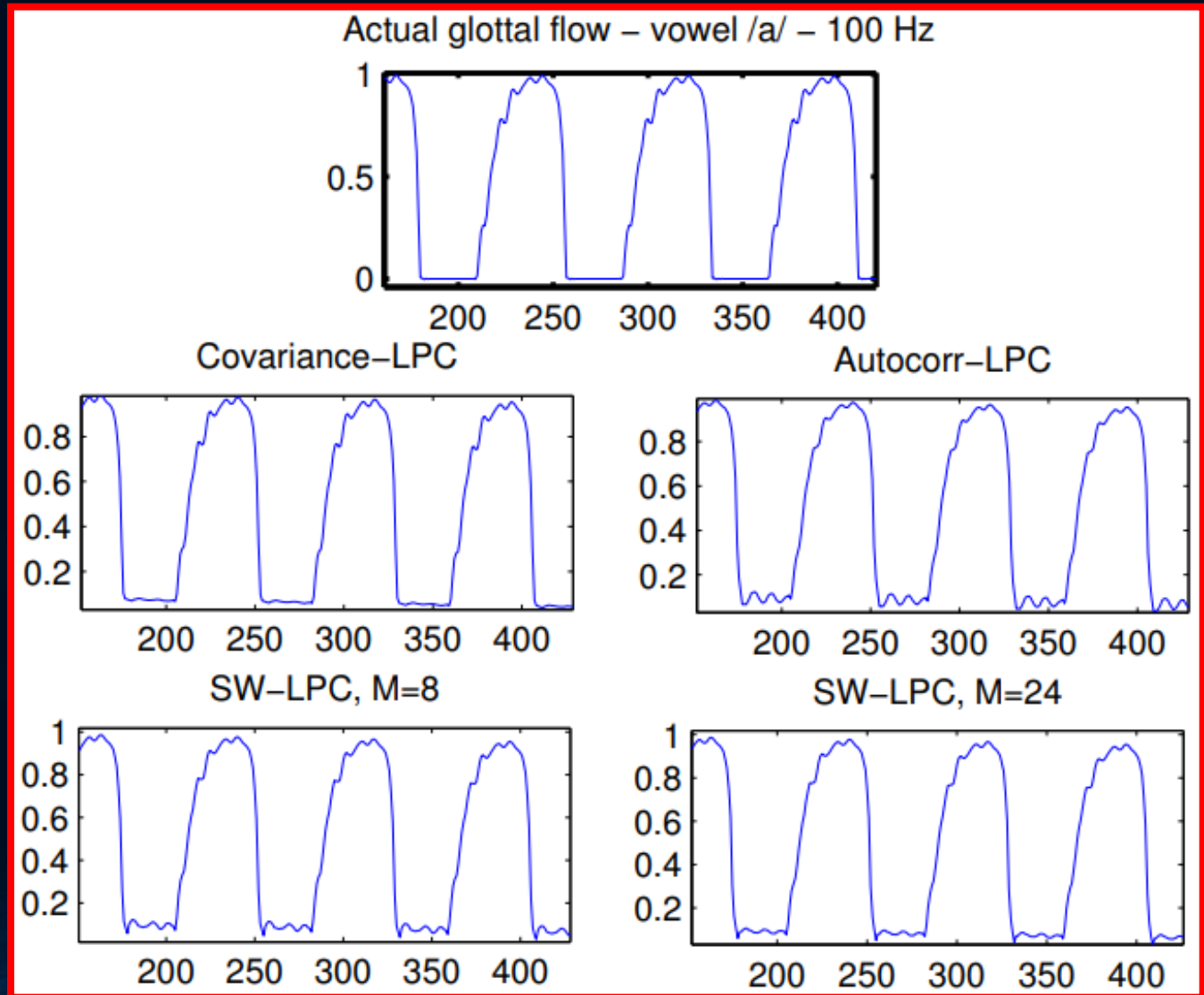
$$Ra = \sigma^2 u$$

where σ^2 is the error energy

- Stability is ensured by a specific algorithm

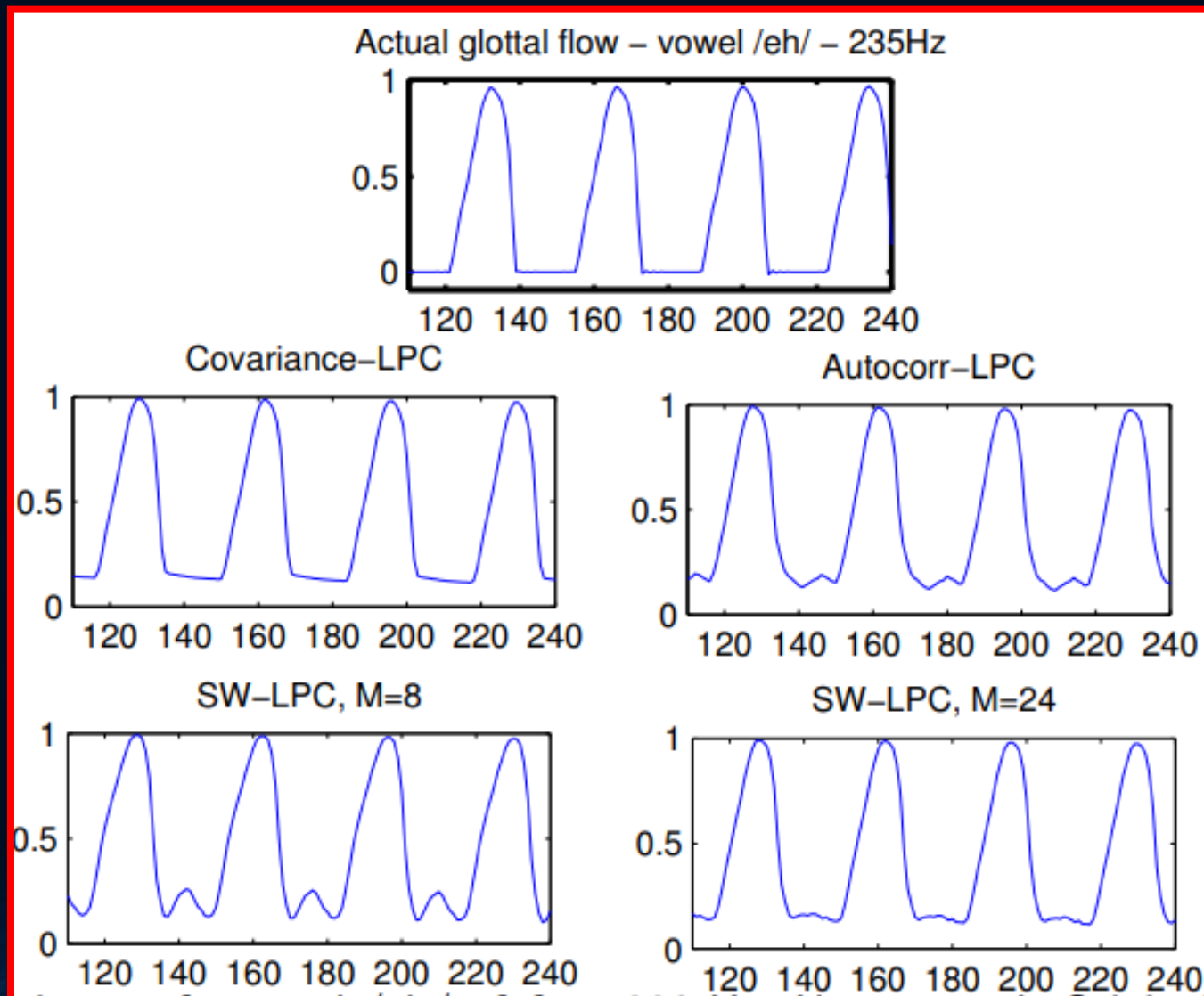
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- Results



On the (Glottal) Inverse Filtering of Speech Signals

- Results



Conclusions

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- GIF has been around for more than five decades
- **Attractive analysis method**
 - Non-invasive
 - Using only speech signal
 - Mostly automatic
 - Applications in many speech technologies
 - Still improving! (QCP Analysis [16])
- **Software: OPENGlott, Aparat, etc**

On the (Glottal) Inverse Filtering of Speech Signals

- GIF has been around for more than five decades
- Shortcomings
 - Recording should be made with caution
 - Introducing non-linearities that distort GIF result
 - “Ground truth” is very rarely available
 - Synthetic speech or physiologically modeled data is used
 - Unreliable analysis of certain voice types [11,12]
 - High-pitch speech, low F_1 , vulnerability of best method (closed-phase CP)
 - Based on all-pole methods → speech is pole-zero (nasal sounds)
 - Fixed filter coefficients over successive periods

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