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Iterative Estimation of Phase Using Complex Cepstrum Representation

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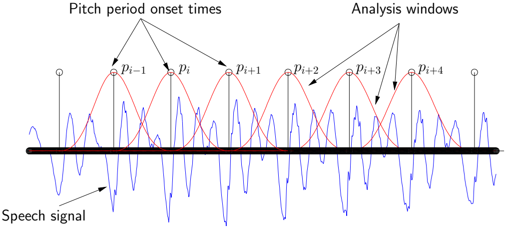
Introduction

- Goal: accurate estimation of phase from speech
- Importance of phase
  1. Speech parameterization for TTS
  2. Features for ASR
  3. Detection of speech pathologies
- Estimation of continuous phase spectrum
  1. Detection of glottal closure instants (GCI)
  2. Phase unwrapping
- Minimum MSE-based complex cepstrum analysis [Maia et al., 2013a]
  - Complex cepstrum analysis with no phase unwrapping
  - GCI are iteratively optimized in the process
- Phase estimation can be performed using the same concept!

Typical phase estimation issues

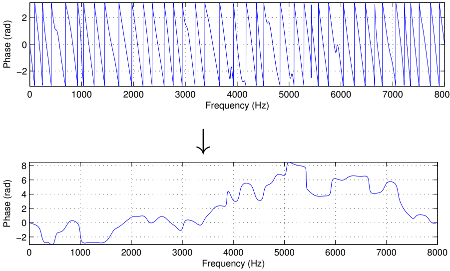
Speech segmentation

- Detection of glottal closure instants (GCIs)
- Influence of the shape and length of the windows



Phase unwrapping

- Discrete Fourier transform (DFT) gives phase modulo 2π
- Phase must be *unwrapped*



Complex cepstrum-based speech analysis and synthesis

Analysis

1. Determine pitch period onset times (or GCI): $\{p_0, \ldots, p_{Z-1}\}$
2. Complex cepstrum analysis

$$\hat{h}(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} \ln\left|S\left(e^{\jmath\omega}\right)\right| e^{\jmath\omega n} d\omega + \frac{\jmath}{2\pi}\int_{-\pi}^{\pi} \theta\left(\omega\right) e^{\jmath\omega n} d\omega$$

$\hat{h}(n)$ ↓ *Cepstrum at* p_z; $\frac{1}{2\pi}\int_{-\pi}^{\pi} \ln\left|S\left(e^{\jmath\omega}\right)\right| e^{\jmath\omega n} d\omega$ ↓ *Amplitude response at* p_z; $\frac{\jmath}{2\pi}\int_{-\pi}^{\pi} \theta\left(\omega\right) e^{\jmath\omega n} d\omega$ ↓ *Phase response at* p_z

Synthesis

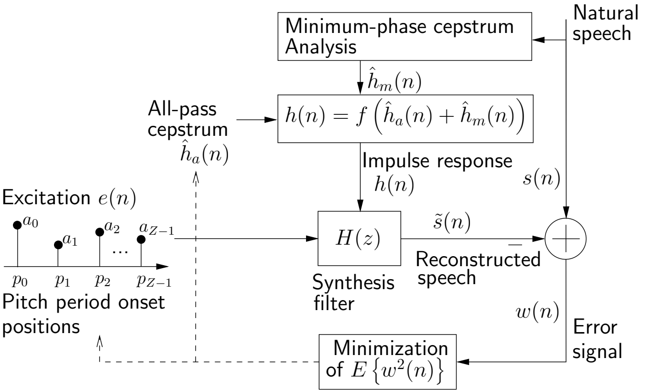
1. Derive non-causal impulse responses

$$h(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} \exp\left\{\sum_{p=-C}^{C} \hat{h}(p)e^{-\jmath\omega p} + \jmath\omega n\right\} d\omega$$

$h(n)$ ↓ *Impulse response at* p_z; $\sum_{p=-C}^{C} \hat{h}(p)$ ↓ *Cepstrum at* p_z

2. Make excitation $e(n)$ with pulses located at p_z
3. Synthesize speech by making $\tilde{s}(n) = h(n) * e(n)$

Proposed phase estimation approach



- Phase iteratively estimated by minimizing the error between natural and reconstructed speech in the time domain
- Pitch period onsets jointly optimized
- No windowing: frame-based time-varying filtering
- No phase unwrapping: cepstral domain

Iterative estimation of phase: requirement

- Pulse positions $\{p_0, \ldots, p_{Z-1}\}$ **must correctly indicate pitch periods** but not necessarily GCI

- Because
  1. Smooth speech spectral envelope at p_z

$$|H_z(e^{\jmath\omega})| = \left| \sum_{n=p_{z-1}}^{p_{z+1}} k(n - p_{z-1})\, s(n) e^{-\jmath\omega n} \right|, \qquad k(n): \text{ window}$$

  2. Real cepstrum

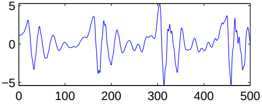
$$\hat{h}_r(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} \ln |H(e^{\jmath\omega})| \, e^{\jmath\omega n} d\omega, \quad n = -C, \ldots, C$$

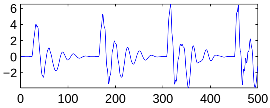
  3. Minimum-phase cepstrum

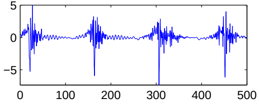
$$\hat{h}_m(n) = \begin{cases} 0 & n < 0 \\ \hat{h}_r(n) & n = 0 \\ 2\hat{h}_r(n) & n = 1, \ldots, C \end{cases}$$

All-pass/minimum-phase speech decomposition

$$s(n) \quad = \quad s_m(n) \quad * \quad s_a(n)$$







$$h(n) = h_m(n) * h_a(n) \Rightarrow \hat{h}(n) = \hat{h}_m(n) + \hat{h}_a(n)$$

Minimum-phase cepstrum	*All pass cepstrum*
$\hat{h}_m(n) = \begin{cases} 0, & -C \leq n \leq -1 \\ \hat{h}(0), & n = 0 \\ \hat{h}(n) + \hat{h}(-n), & 1 \leq n \leq C \end{cases}$	$\hat{h}_a(n) = \begin{cases} \hat{h}(n), & -C \leq n \leq -1 \\ 0, & n = 0 \\ -\hat{h}(-n), & 1 \leq n \leq C \end{cases}$

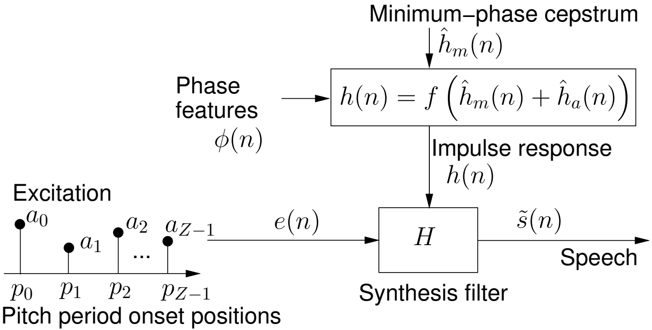
$$\hat{\boldsymbol{h}}_m = \left[\hat{h}_m(0) \quad \cdots \quad \hat{h}_m(C)\right]^\top$$

$$\boldsymbol{\phi} = \left[\hat{h}_a(1) \quad \cdots \quad \hat{h}_a(C)\right]$$

Min.-phase cepstrum: amplitude and min. phase!

Causal all pass cepstrum: residual phase!

Parameters of the model



- Excitation parameters: *hidden variables*
  - $\boldsymbol{a} = \{a_0, \ldots, a_{Z-1}\}$: pulse amplitudes
  - $\boldsymbol{p} = \{p_0, \ldots, p_{Z-1}\}$: pulse locations
- Non-causal synthesis filter parameters: variables to be determined
  - $\{\phi_0, \ldots, \phi_{T-1}\}$: phase features at every frame

A two-step optimization process at the utterance level

Step 1 Estimation of the locations and amplitudes of the excitation signal

- Keep $\hat{h}(n)$ fixed
- Optimize $\{p_0, \ldots, p_{Z-1}\}$ and $\{a_0, \ldots, a_{Z-1}\}$

Step 2 Phase estimation given the new pulse positions and amplitudes

- Keep $e(n)$ fixed
- Re-estimate $\phi(n)$ using a gradient method
  - ➔ Non-linear relationship between $h(n)$ and $\phi(n)$

$$h(n) = f_1\left(\phi(n)\right)$$

$$\downarrow$$

$$h(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} \exp\left\{ \sum_{p=0}^{C} \hat{h}_m(p) e^{-\jmath\omega p} - 2\jmath \sum_{p=1}^{C} \phi(p-1) \sin(\omega p) + \jmath\omega n \right\} d\omega$$

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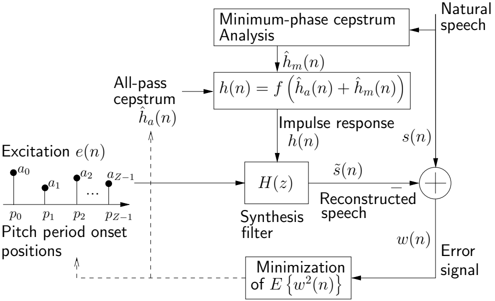
Step 1: estimation of $\{a_0, p_0, \ldots, a_{Z-1}, p_{Z-1}\}$

$H(z)$ fixed, $e(n)$ to be optimized!



Error vector

$$\boldsymbol{w} = \boldsymbol{s} - \tilde{\boldsymbol{s}} = \boldsymbol{s} - \sum_{z=0}^{Z-1} a_z \boldsymbol{g}_{p_z}$$

where

$$\boldsymbol{g}_n = \Big[\underbrace{0 \ \cdots \ 0}_{n} \quad \boldsymbol{h}_n^\top \quad \underbrace{0 \ \cdots \ 0}_{N-n-1}\Big]^\top$$

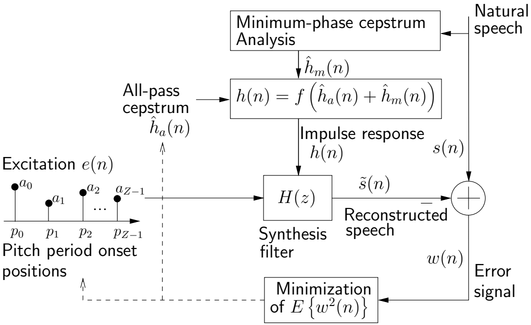
$$\boldsymbol{h}_n = \left[h_n\left(-\frac{M}{2}\right) \quad \cdots \quad h_n\left(\frac{M}{2}\right)\right]^\top$$

Cost function and new positions and amplitudes

$$\varepsilon(\boldsymbol{p}, \boldsymbol{a}) = \frac{1}{N}\left[\boldsymbol{s} - \sum_{z=0}^{Z-1} a_z \boldsymbol{g}_{p_z}\right]^\top \left[\boldsymbol{s} - \sum_{z=0}^{Z-1} a_z \boldsymbol{g}_{p_z}\right] \Rightarrow \begin{cases} \hat{p}_z = \underset{p_z-\Delta p, \ldots, p_z+\Delta p}{\arg\max} \dfrac{\left\{\boldsymbol{g}_{p_z}^\top \left[\boldsymbol{s} - \sum_{i=0, i\neq z}^{Z-1} a_i \boldsymbol{g}_{p_i}\right]\right\}^2}{\boldsymbol{g}_{p_z}^\top \boldsymbol{g}_{p_z}} \\ \hat{a}_z = \dfrac{\boldsymbol{g}_{p_z}^\top \left[\boldsymbol{s} - \sum_{i=0, i\neq z}^{Z-1} a_i \boldsymbol{g}_{p_i}\right]}{\boldsymbol{g}_{p_z}^\top \boldsymbol{g}_{p_z}} \end{cases}$$

Step 2: estimation of $\{\phi_0, \ldots, \phi_{T-1}\}$

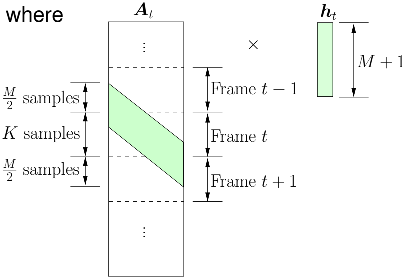
$e(n)$ fixed, $H(z)$ to be optimized!



Error vector

$$\boldsymbol{w} = \boldsymbol{s} - \tilde{\boldsymbol{s}} = \boldsymbol{s} - \sum_{t=0}^{T-1} \boldsymbol{A}_t \boldsymbol{h}_t$$

where



- Matrix $\boldsymbol{A}_t$ contains samples of $e(n)$ at frame t
- Impulse response vector at frame t: $\boldsymbol{h}_t = \left[h_t\left(-\frac{M}{2}\right) \quad \cdots \quad h_t\left(\frac{M}{2}\right)\right]^\top$

Step 2: estimation of $\{\phi_0, \ldots, \phi_{T-1}\}$

- MSE

$$\varepsilon = \frac{1}{N}\left[\boldsymbol{s} - \sum_{t=0}^{T-1} \boldsymbol{A}_t \boldsymbol{h}_t\right]^\top \left[\boldsymbol{s} - \sum_{t=0}^{T-1} \boldsymbol{A}_t \boldsymbol{h}_t\right]$$

- Cost function

$$\varepsilon(\boldsymbol{\phi}_t) = \frac{1}{N}\left[\boldsymbol{r}_t^\top \boldsymbol{r}_t - 2\boldsymbol{r}_t \boldsymbol{A}_t f_1(\boldsymbol{\phi}_t) + \{f_1(\boldsymbol{\phi}_t)\}^\top \boldsymbol{U}_t f_1(\boldsymbol{\phi}_t)\right]$$

$$\begin{cases} \boldsymbol{r}_t = \boldsymbol{s} - \sum_{j=0, j\neq t}^{T-1} \boldsymbol{A}_j f_1(\boldsymbol{\phi}_j) \\ \boldsymbol{U}_t = \boldsymbol{A}_t^\top \boldsymbol{A}_t \end{cases}$$

- Relationship between impulse response and residual phase

$$\boldsymbol{h}_t = f_1(\boldsymbol{\phi}_t) = \frac{1}{2L}\boldsymbol{D}_2 \exp\left(\boldsymbol{D}_{m,1}\hat{\boldsymbol{h}}_{m,t} + \boldsymbol{D}_{a,1}\boldsymbol{\phi}_t\right)$$

$$\begin{cases} D_{m,1}(i,j) = e^{-\jmath\omega_i j} & -L+1 \leq i \leq L, 0 \leq j \leq C \\ D_{a,1}(i,j) = -2\jmath \sin(\omega_i j) & -L+1 \leq i \leq L, 0 \leq j \leq C \\ D_2(i,j) = e^{\jmath\omega_j i} & -\frac{M}{2} \leq i \leq \frac{M}{2}, -L+1 \leq j \leq L \end{cases}$$

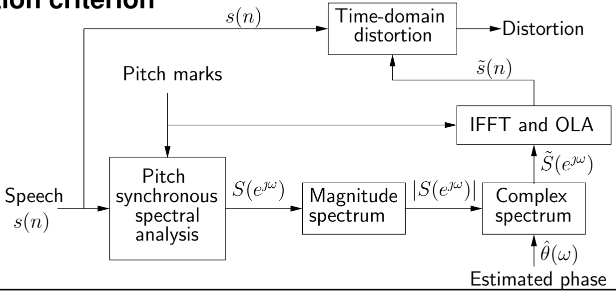
ϕ_t is determined by a gradient descent method!

Experiment

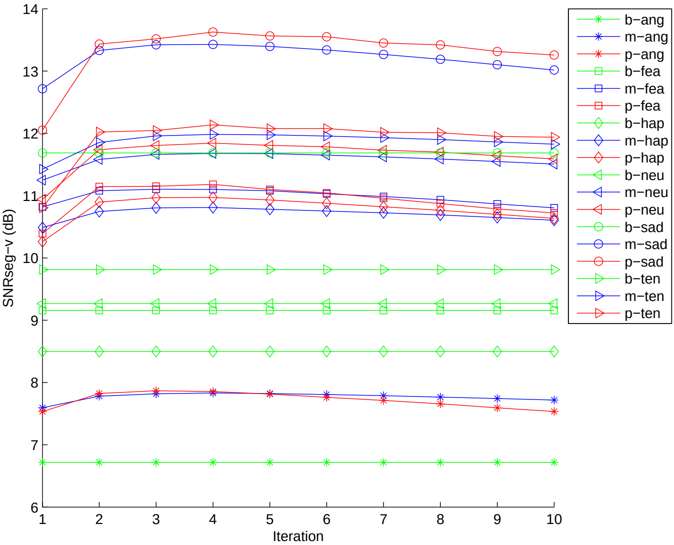
Conditions

- Female UK English speaker, 22.05 kHz, 50 sentences in each of the following styles: angry, fear, happy, neutral, sad and tender
- Methods evaluated
  1. GCI detection using DYPSA [Naylor et al., 2007] + phase unwrapping with a 8192-point DFT (g)
  2. MSE cepstrum analysis (MSE-CCEP) as in [Maia et al., 2013b] (b)
  3. Proposed (r)

Evaluation criterion



Results



- Proposed method performs better than GCI detection + unwrapping for all speech styles
- Proposed is similar to MSE-CCEP for all styles

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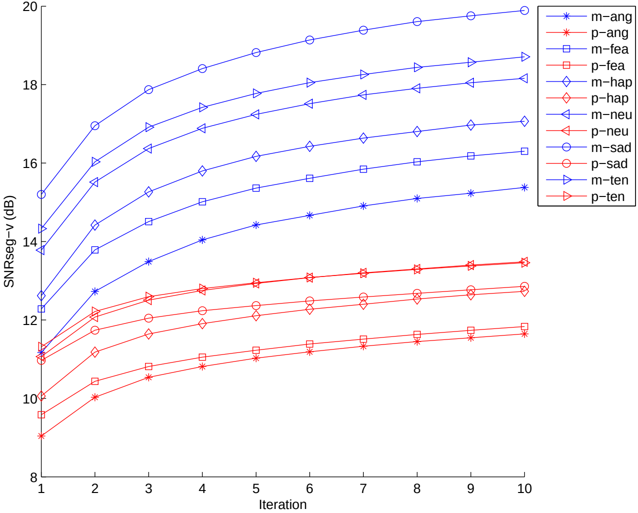
Analysis-synthesis comparison with MSE-CCEP



- MSE-CCEP performs better in terms of analysis-synthesis
- MSE-CCEP optimizes not only phase but also amplitude
- In terms of phase estimation both perform the same
- The proposed method runs in average 3 times faster

Conclusions

- Method to estimate short-term residual phase using complex cepstrum-based analysis and synthesis of speech
- Accurate markings of the pitch periods are necessary
- Better performance than GCI detection followed by multi-resolution phase unwrapping
- Similar performance to the more computationally expensive MSE complex cepstrum analysis
- An automatic way to extract phase information from pitch marks, with no need of
- Future work: application to TTS and ASR exact GCI information

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