SPEECH ANALYSIS BASED ON SINUSOIDAL MODEL WITH TIME-VARYING PARAMETERS

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1. Introduction

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The paper presents some techniques for extracting pitch and spectral envelope of a signal using sinusoidal model with instantaneous parameters.

The main features of the proposed algorithms are:

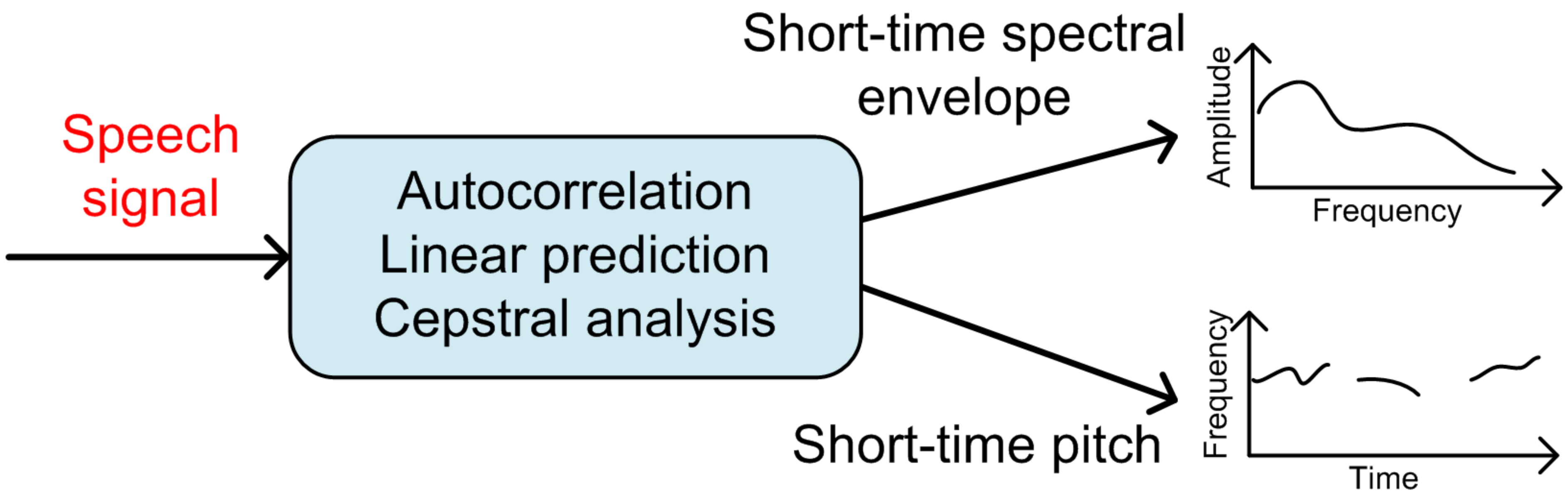
- model-based analysis that provides high time-frequency resolution;
- model-based estimation of instantaneous pitch
- model-based estimation of instantaneous spectral envelope;
- accurate envelope representation by linear predictors of high order.

Two-step analysis:

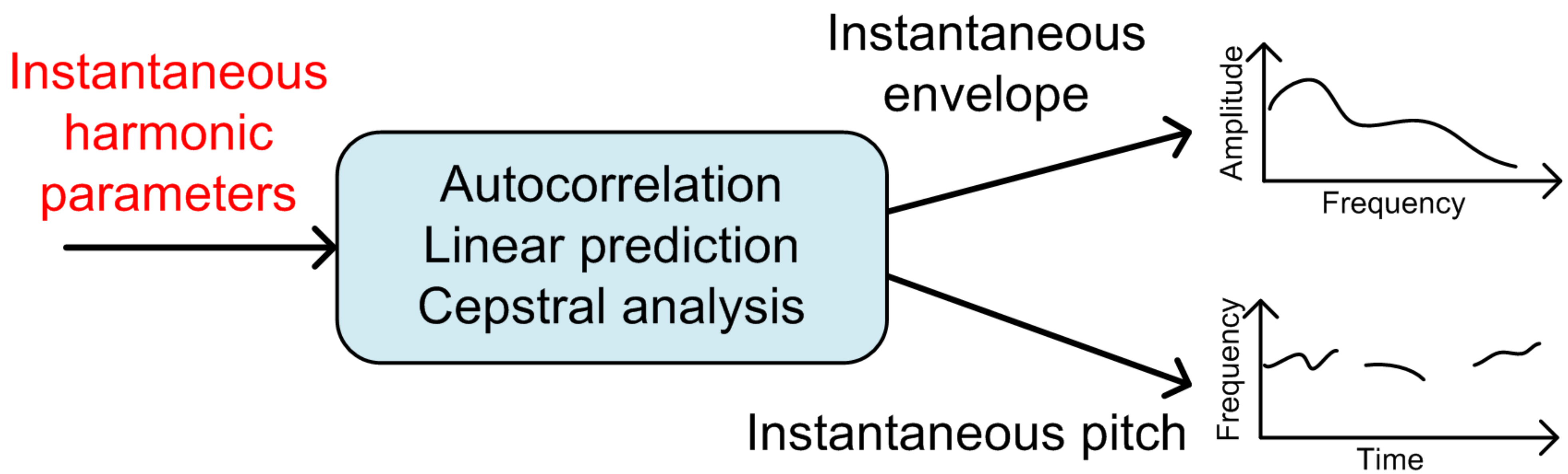
1. extraction of sinusoidal parameters;
2. parameters transformation to required characteristics

2. Model-based extraction of speech characteristics

Classical extraction techniques



Model-based extraction techniques



3. Estimation of instantaneous harmonic parameters

The signal $s(m)$ is decomposed into overlapping bandlimited analytical signals $S_{F_\Delta,F_c^i}(m)$:

$$S_{F_\Delta,F_c^i}(m) = \sum_{n=-\infty}^{\infty} \frac{\sin(F_\Delta n)}{n\pi} w(n)s(m-n)e^{-jF_c^i n} = A_{F_\Delta,F_c^i}(m)\cos\left(\varphi_{F_\Delta,F_c^i}(m)\right),$$

where $2F_\Delta$ - bandwidth and F_c^i - center frequency of the *i*-th band and $w(n)$ – an even window function. Then instantaneous parameters are evaluated as

instantaneous amplitude $\longrightarrow$ $A_{F_\Delta,F_c^i}(m) = \sqrt{R^2(m) + I^2(m)},$

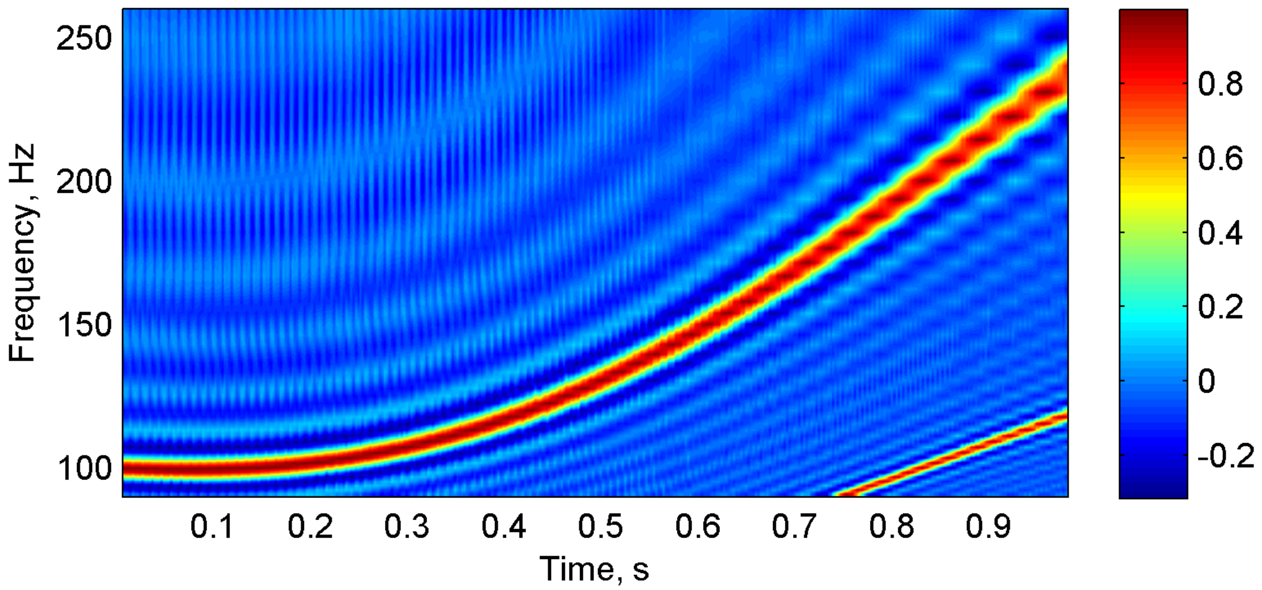
instantaneous phase $\longrightarrow$ $\varphi_{F_\Delta,F_c^i}(m) = \arctan\left(\frac{-I(m)}{R(m)}\right),$

instantaneous frequency $\longrightarrow$ $F_{F_\Delta,F_c^i}(m) = \varphi'_{F_\Delta,F_c^i}(m),$

where $R(m)$ and $I(m)$ are real and imaginary parts of $S_{F_\Delta,F_c^i}(m)$ respectively.

4. Normalized cross-correlation function (NCCF)

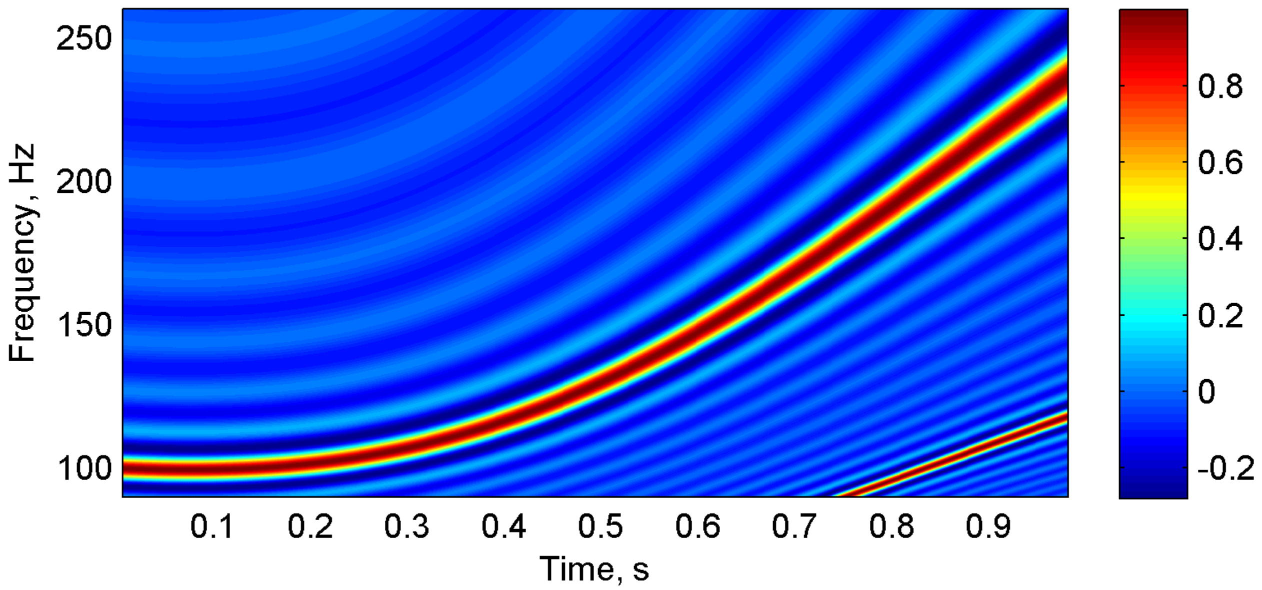
Time domain



$$\phi(m,k) = \frac{\sum_{i=m}^{m+n-1} s(i)s(i+k)}{\sqrt{e_m e_{m+k}}},$$

where $e_i = \sum_{l=i}^{i+n-1} s_l^2$ and n – window size

Model-based estimation



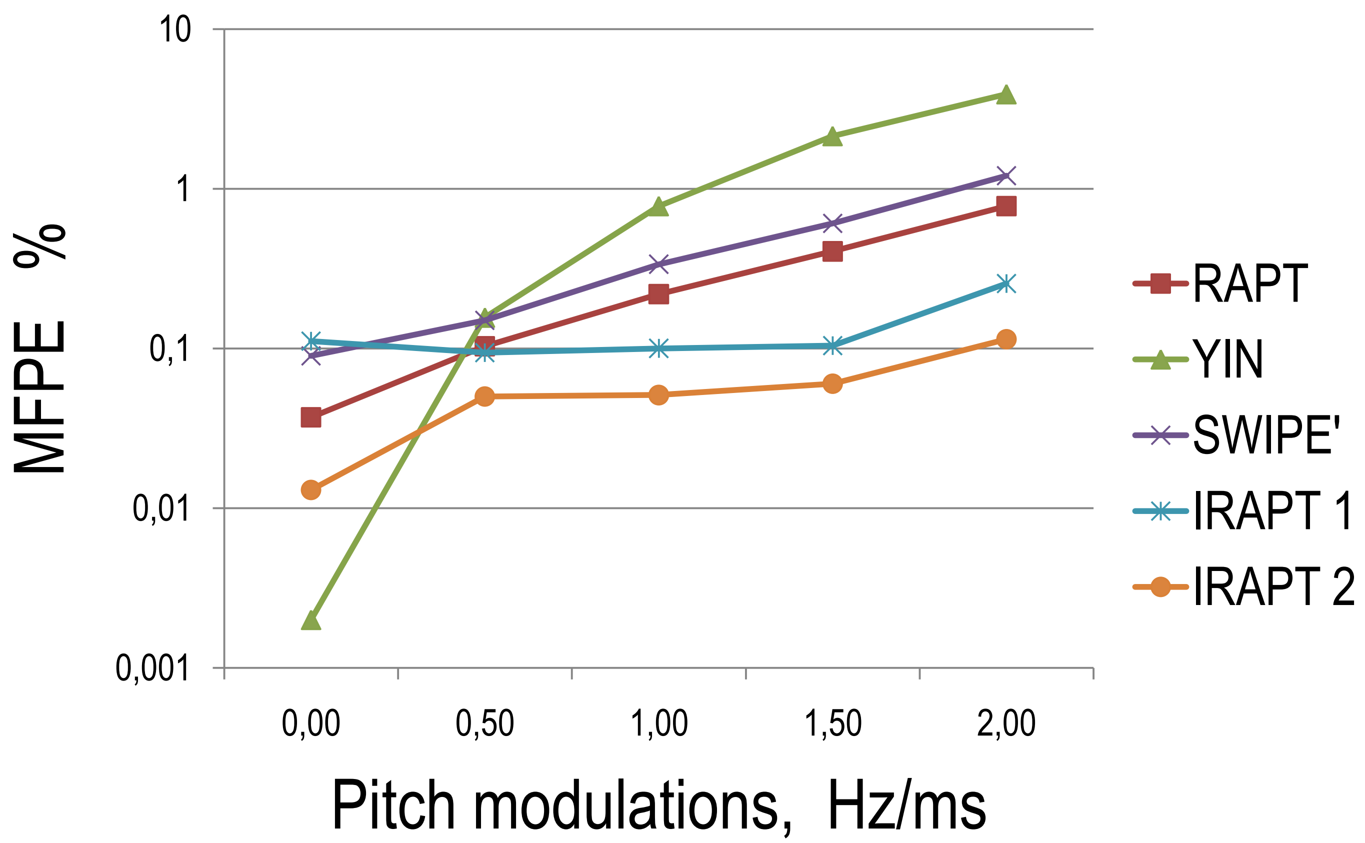
$$\phi_{inst}(m,k) = \frac{\sum_{p=1}^{P} A_p^2(m)\cos(F_p(m)k)}{\sum_{p=1}^{P} A_p^2(m)}$$

P – number of bandlimited analytical signals

5. Experimental results of pitch extraction

The proposed technique is compared with other pitch estimation algorithms in terms of gross pitch error (GPE, %) and mean fine pitch error (MFPE, %).

Artificial signals



Natural speech

	Male		Female	
	GPE	MFPE	GPE	MFPE
RAPT	3.69	1.74	6.07	1.18
YIN[3]	3.18	**1.39**	3.96	0.84
SWIPE'[4]	**0.78**	1.51	4.27	**0.80**
IRAPT 1	1.63	1.61	**3.78**	0.98
IRAPT 2	1.57	1.57	**3.78**	1.05

[3] A. Cheveigné and H. Kawahara "YIN, a fundamental frequency estimator for speech and music", *Journal Acoust. Soc. Am.*, vol. 111, no. 4, pp 1917-1930, Apr. 2002.

[4] A. Camacho and J. G. Harris, "A sawtooth waveform inspired pitch estimator for speech and music", *Journal Acoust. Soc. Am.*, vol. 123, no. 4, pp 1638-1652, Sep. 2008.

6. High-order linear prediction

The coefficients are evaluated using the following system:

$$Q = \begin{bmatrix} q(0) & \cdots & q(p-1) \\ \vdots & \ddots & \vdots \\ q(p-1) & \cdots & q(0) \end{bmatrix} \begin{bmatrix} a_1 \\ \vdots \\ a_p \end{bmatrix} = -\begin{bmatrix} q(1) \\ \vdots \\ q(p) \end{bmatrix} \qquad q(l) = \sum_{i=1}^{K-1} D(l,i)$$

where $a_1, \dots, a_p$ – prediction coefficients, *p* – prediction order.

Each segment of the spectral envelope $f_i \le \omega \le f_{i+1}$, $1 \le i \le K-1$ is defined by a linear equation $A(\omega) = b_i\omega + c_i$

$$D(l,i) = \begin{cases} \frac{b_i}{l^2}[\cos(f_{i+1}l) + f_{i+1}l\sin(f_{i+1}l)] + \frac{c_i}{l}\sin(f_{i+1}l) - \\ \quad -\frac{b_i}{l^2}[\cos(f_i l) + f_i l\sin(f_i l)] - \frac{c_i}{l}\sin(f_i l) & l \ne 0 \\ \quad \frac{1}{2}b_i f_{i+1}^2 + c_i f_{i+1} - \frac{1}{2}b_i f_i^2 - c_i f_i & l = 0 \end{cases}$$

7. Model-based linear prediction of speech