A LOW–DELAY ALGORITHM FOR INSTANTANEOUS PITCH ESTIMATION

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1. Introduction

The paper presents an algorithm for instantaneous pitch estimation specifically designed for real-time applications.

The main features of the proposed algorithms are:

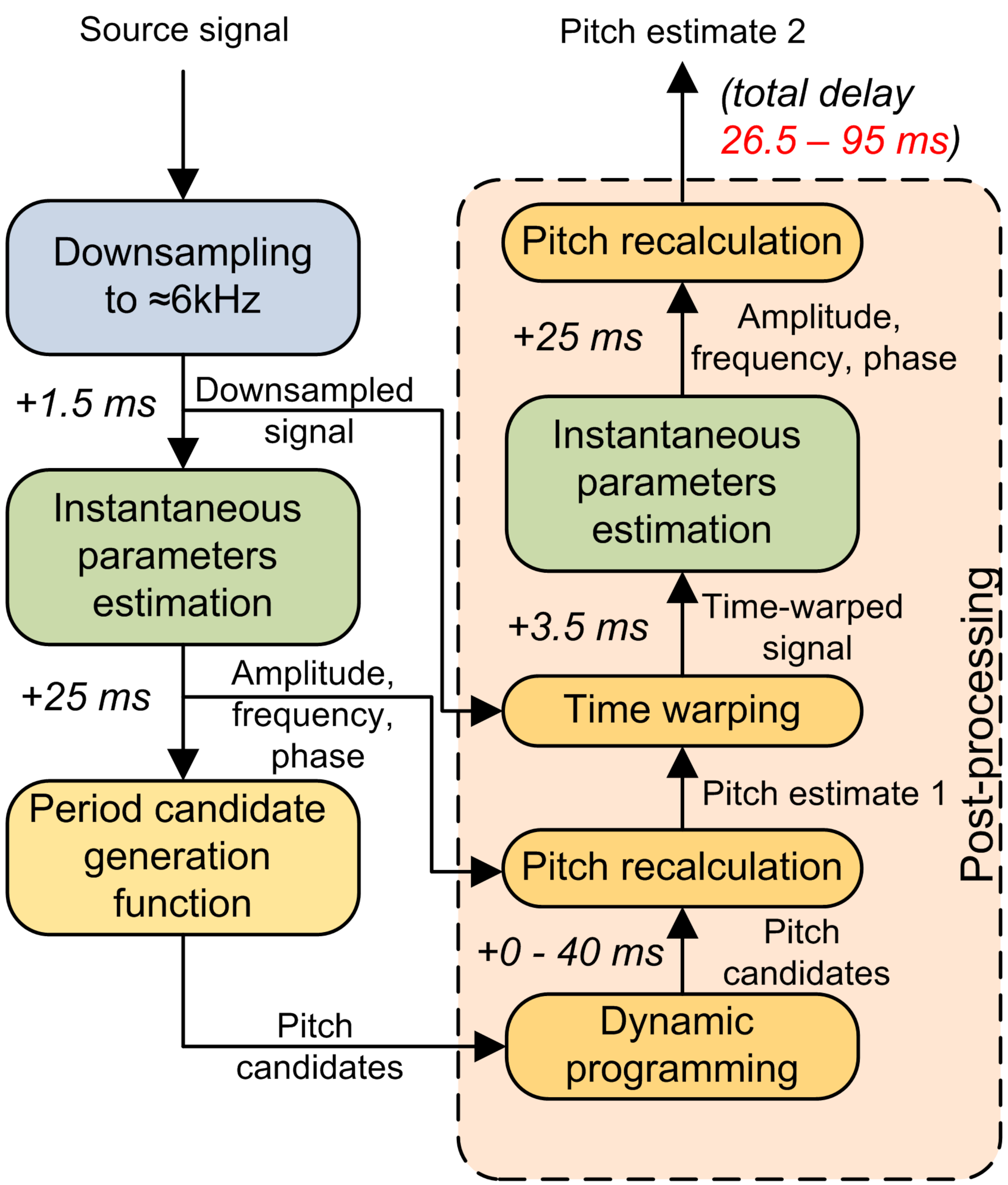
- high estimation accuracy and time resolution;
- low inherent delay;
- efficient processing scheme;
- good performance in noisy conditions;

The general idea:

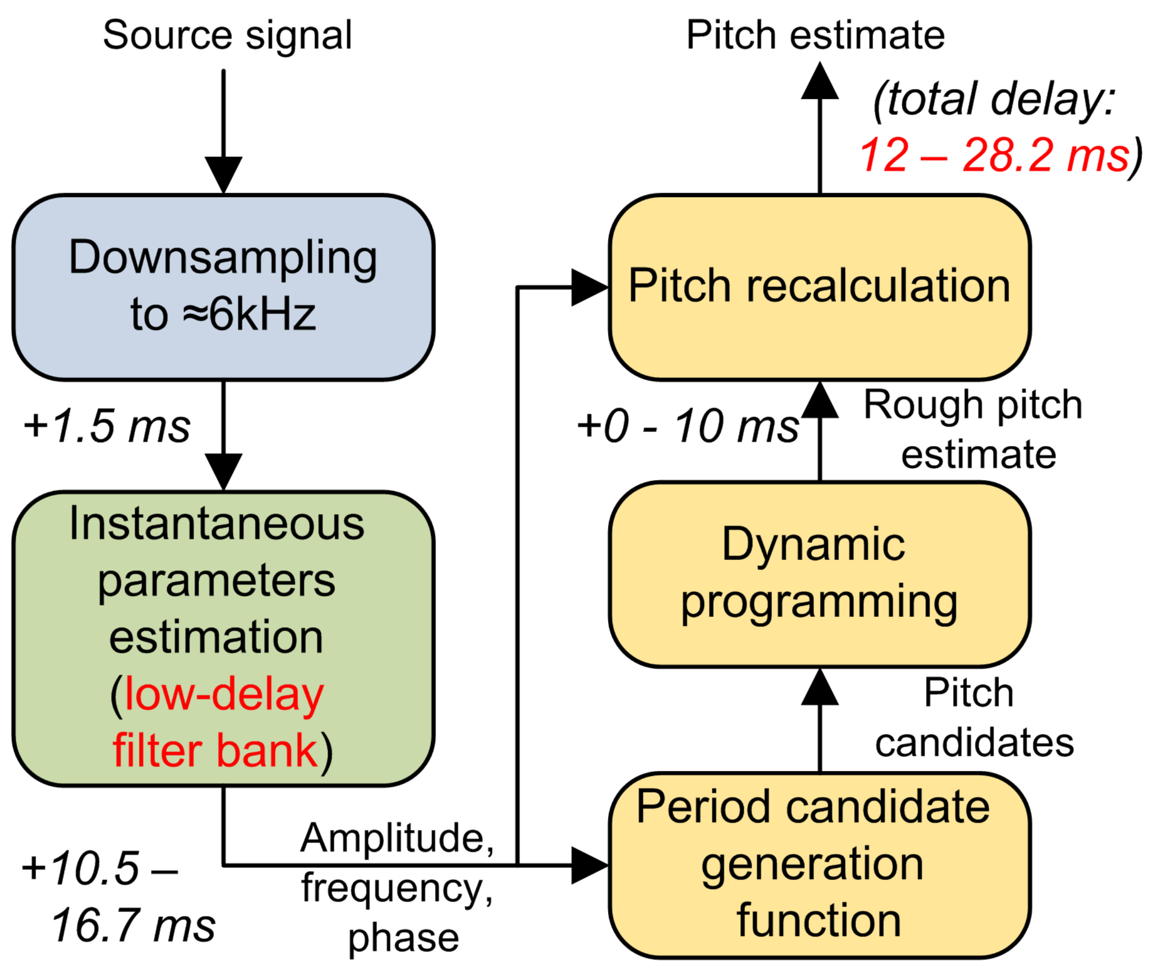
1. the analysis scheme is based on the robust algorithm for instantaneous pitch tracking (IRAPT);
2. a non-linear phase analysis filter bank is applied to get a shorter inherent delay;

2. Pitch estimation schemes

IRAPT scheme



Proposed low-delay scheme



3. Analysis filter bank

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

$$S_{F_\Delta,F_c^i}(m) = \sum_{n=-\infty}^{\infty} \overbrace{\frac{\sin(F_\Delta n)}{n\pi} w(n)}^{\text{linear-phase low-pass filter}} \underbrace{e^{-jF_c^i n}}_{\text{modulation function}} s(m-n) = A_{F_\Delta,F_c^i}(m)\cos\left(\varphi_{F_\Delta,F_c^i}(m)\right),$$

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. Filter prototype 1 – minimal-phase design

1) Calculate frequency response of the filter: $H_{lp}(k, F_\Delta) = FFT(h_{lp}(n, F_\Delta))$;

2) Calculate cepstrum $C_{lp}(c, F_\Delta) = IFFT\left(\log\left|H_{lp}(k, F_\Delta)\right|\right)$;

3) Set zero for all $c > \frac{N}{2}$ and multiply by 2 except the first:

$$\bar{C}_{lp}(c, F_\Delta) = \begin{cases} C_{lp}(c, F_\Delta), & c = 1 \\ 2C_{lp}(c, F_\Delta), & 1 < c \le \frac{N}{2} \\ 0, & c > \frac{N}{2} \end{cases}$$

4) Calculate minimum-phase freq. response $\bar{H}_{lp}(k, F_\Delta) = e^{FFT(C_{lp}(c, F_\Delta))}$;

5) Calculate impulse response of minimum phase filter form its frequency response $\bar{h}_{lp}(n, F_\Delta) = IFFT\left(\bar{H}_{lp}(k, F_\Delta)\right)$.

5. Filter prototype 2 – IIR approximation

1) Set $\omega_c = 2.64\pi/M$, where $M = 360$, $L = 3$

2) Using low-pass Chebychev filter of type 1 obtain first *N* samples of the impulse response

$$|H(j\omega)|^2 = \frac{1}{1 + \varepsilon^2 T_L^2(j\omega/\omega_c)}$$

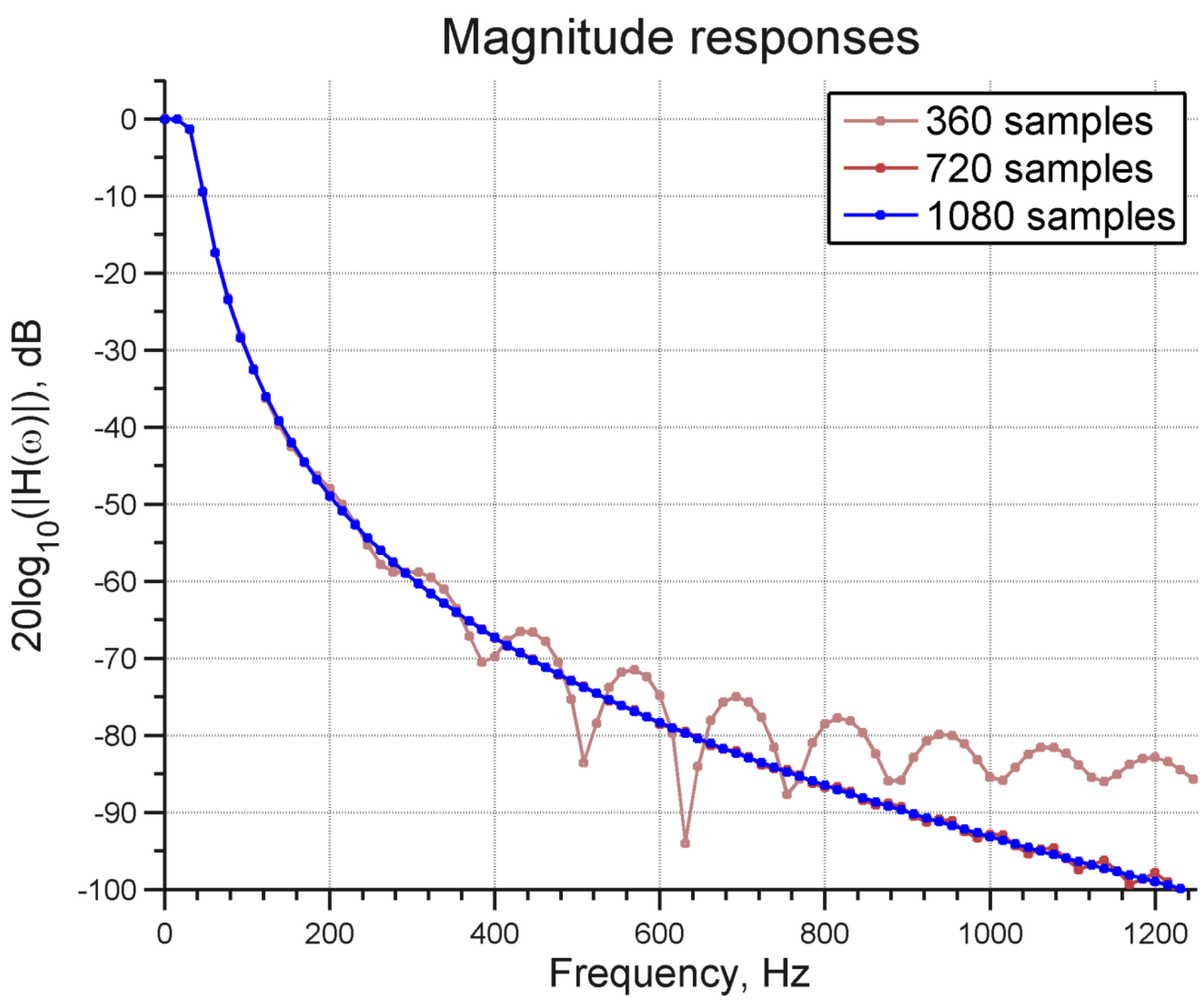
where

$T_L(x)$ – the Chebyshev polynomial of order *L*,

ω_c – the cutoff frequency,

ε – passband ripple parameter.

$N = 1080$ produces a filter with no ripple in the stoppband



6. Comparing filter prototypes

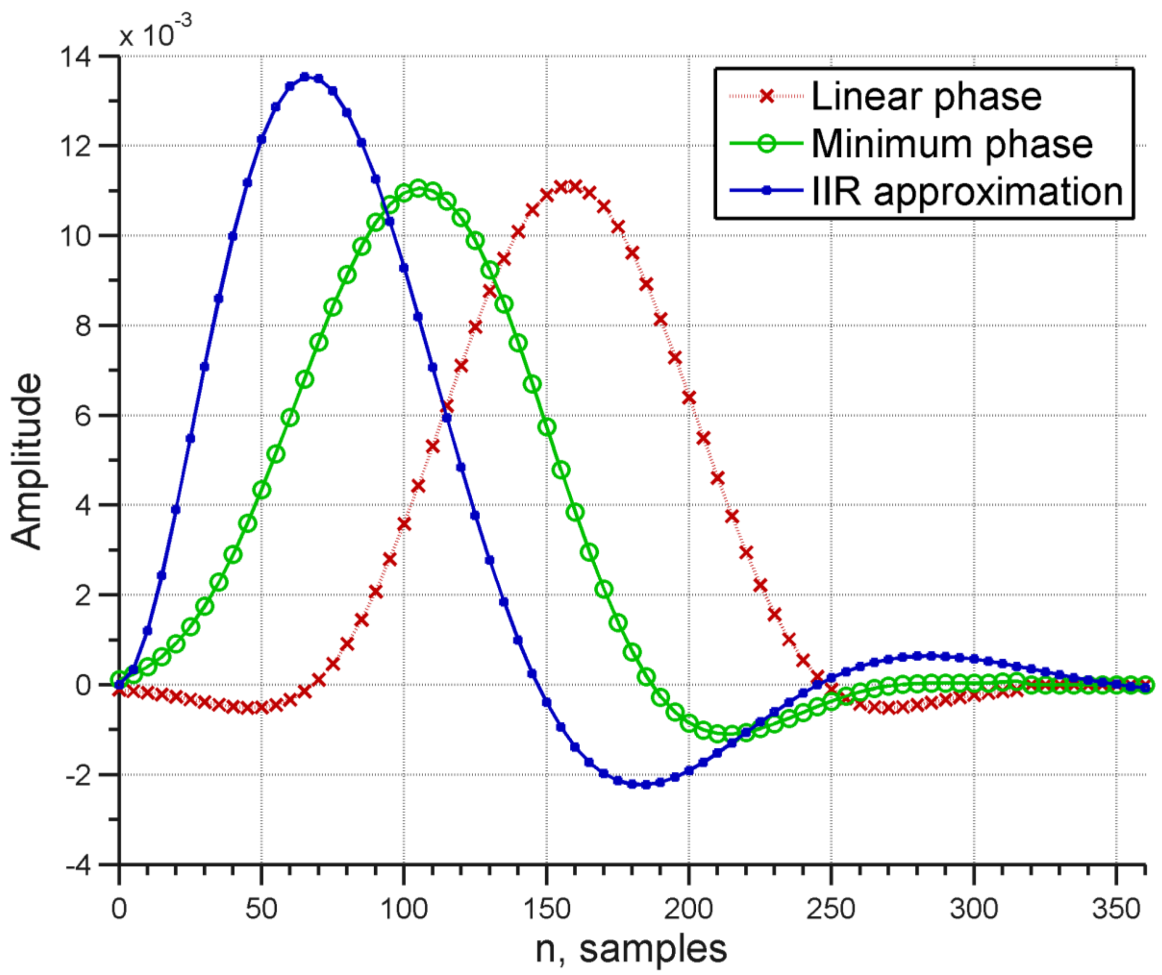
IIR approximation has the shortest response time compared to all competitors.

Linear phase – a constant group delay 25ms

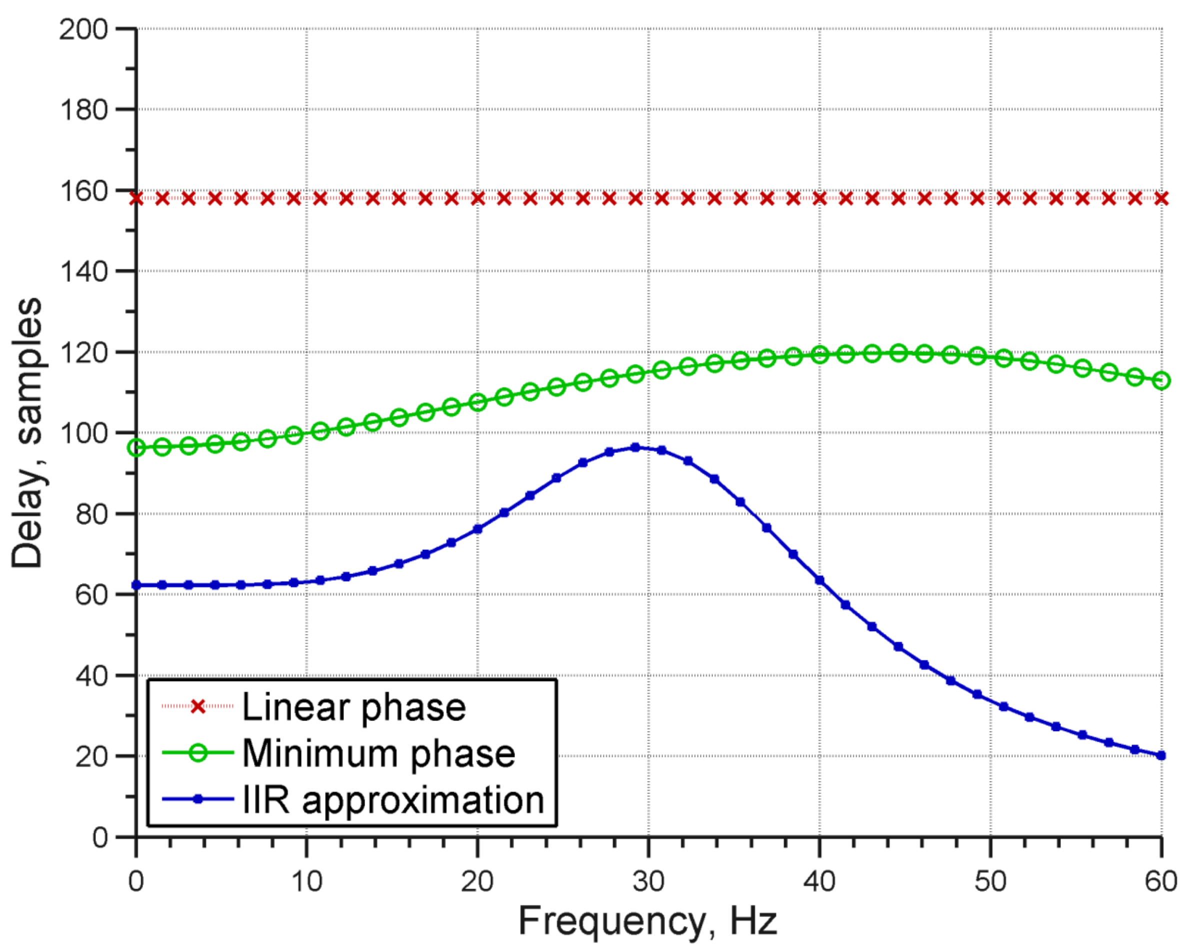
Minimal phase – variable group delay from 16 to 19ms

IIR approximation – variable group delay from 5 to 15 ms

Impulse responses



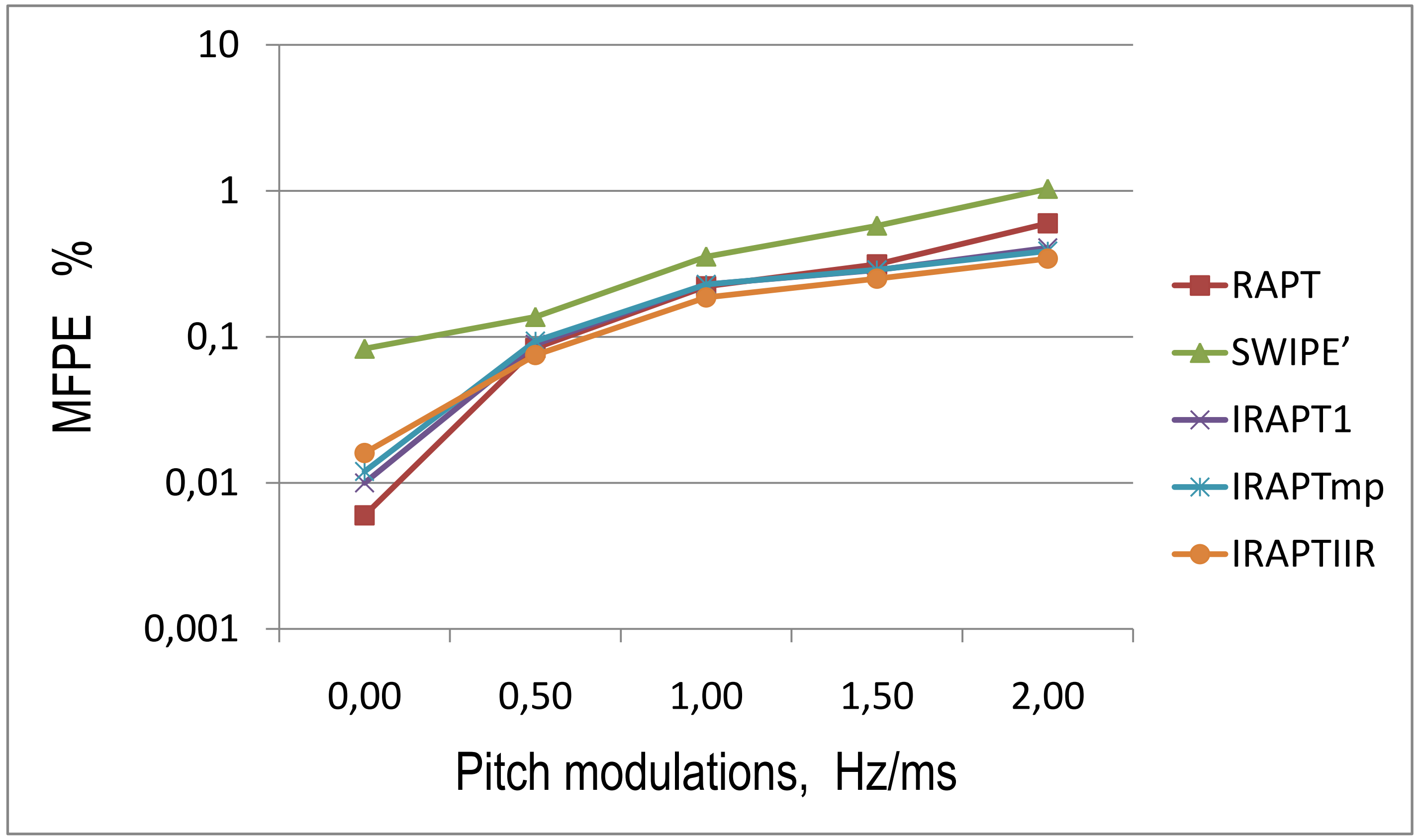
Group delays



7. Experimental results

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
SWIPE'[4]	0.78	1.51	4.27	0.80
IRAPT 1	1.63	1.61	3.78	0.98
$IRAPT_{mp}$	1.73	1.36	5.82	0.90
$IRAPT_{IIR}$	4.2	1.49	4.83	0.93

[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.