SCALABLE PARAMETRIC AUDIO CODER USING SPARSE APPROXIMATION WITH FRAME-TO-FRAME PERCEPTUALLY OPTIMIZED WAVELET PACKETE BASED DICTIONARY

Convention paper 9264

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1. Introduction

Presented work describes a new algorithm of parametric audio coding based on sparse approximation that used matching pursuit (MP) algorithm with optimized wavelet packet (WP) dictionary.

Main features are:

- High quality of reconstructed audio signal;
- Low speed rate of audio data transmission;
- Algorithm scalability to audio data coding;
- Universality for different nature of audio signals.

Main ideas of research are:

- Using MP algorithm as a core of encoder;
- Psychoacoustic optimized of time-frequency functions dictionary;
- WP – single transform domain.

2. MP Using WP Dictionary

Common MP procedure[1]

signal approximation ⟵ $$x(t) = \sum_{n=0}^{\infty} a_n g_{\gamma_n}(t)$$

window function ⟵ $$g_\gamma(t) = \frac{1}{\sqrt{s}} g\left(\frac{t-u}{s}\right) e^{i\xi t}$$

where s – scale, ξ – frequency modulation, u – translation.

WP based dictionary of time-frequency functions

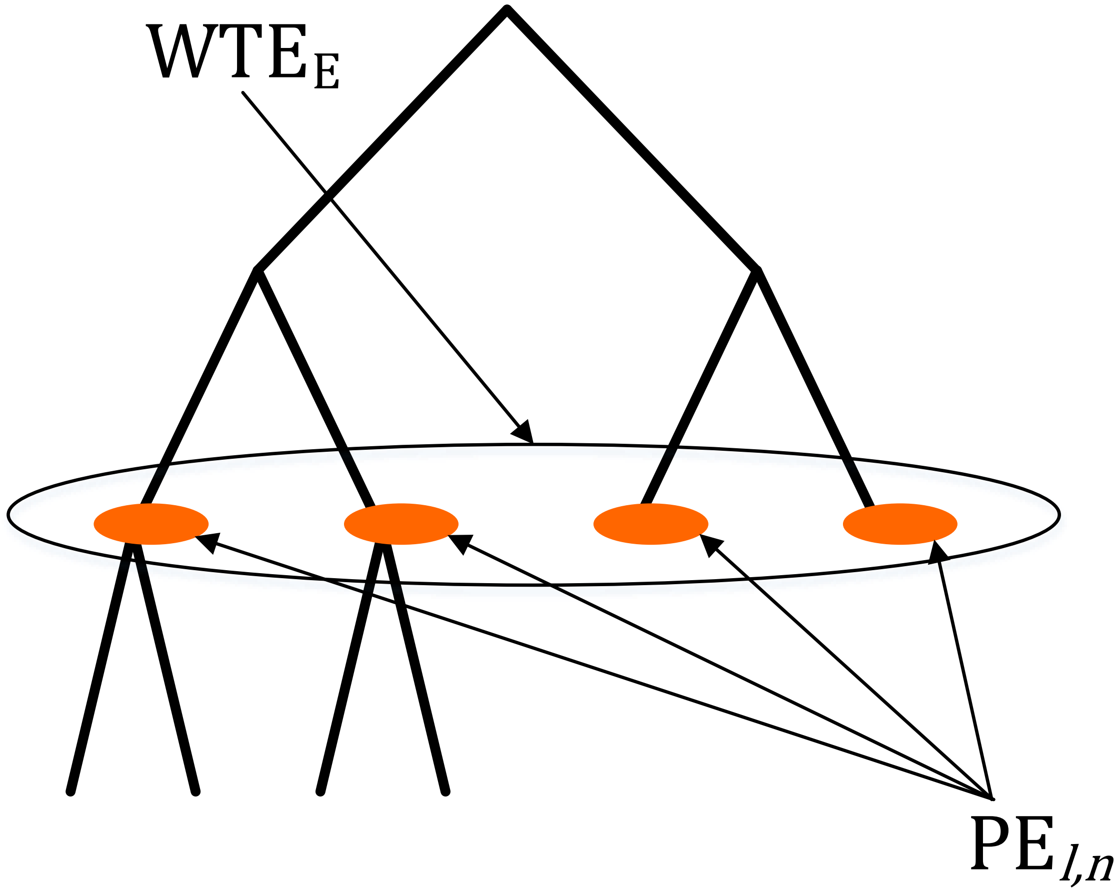
WP-based dictionary ⟵ $$g_\gamma \in D, \gamma = (l, n, k)$$

WP tree structure ⟵ $$E \in \{(l, n): 0 \le l \le L, 0 \le n \le 2^l\}$$

where l – WP tree level number, n – tree node number.

[1] S. Mallat, Z. Zang, "Matching Pursuits with Time-Frequency Dictionaries", IEEE Transactions on signal processing, vol. 41, pp. 3397-3415 (1993 December).

3. Adaptive WP Decomposition



Cost functions

Adaptation cost functions:

Wavelet time entropy estimation

$$WTE_{E_i} = -\sum_{\forall(l,n)\in E_i} \sum_{k} \frac{|X_{l,n,k}|}{\sum_{\forall(l,n)\in E_i}|X_{l,n,k}|} ln\left(\frac{|X_{l,n,k}|}{\sum_{\forall(l,n)\in E_i}|X_{l,n,k}|}\right)$$

Perceptual entropy estimation

$$PE_{l,n} = \sum_{k=1}^{K_{l,n}-1} \log_2(2[nint(SMR_{l,n,k})] + 1)$$

where $SMR_{l,n,k} = |X_{l,n,k}| \big/ \sqrt{12 \cdot T_{l,n}/K_{l,n}}$

Tree optimization procedure

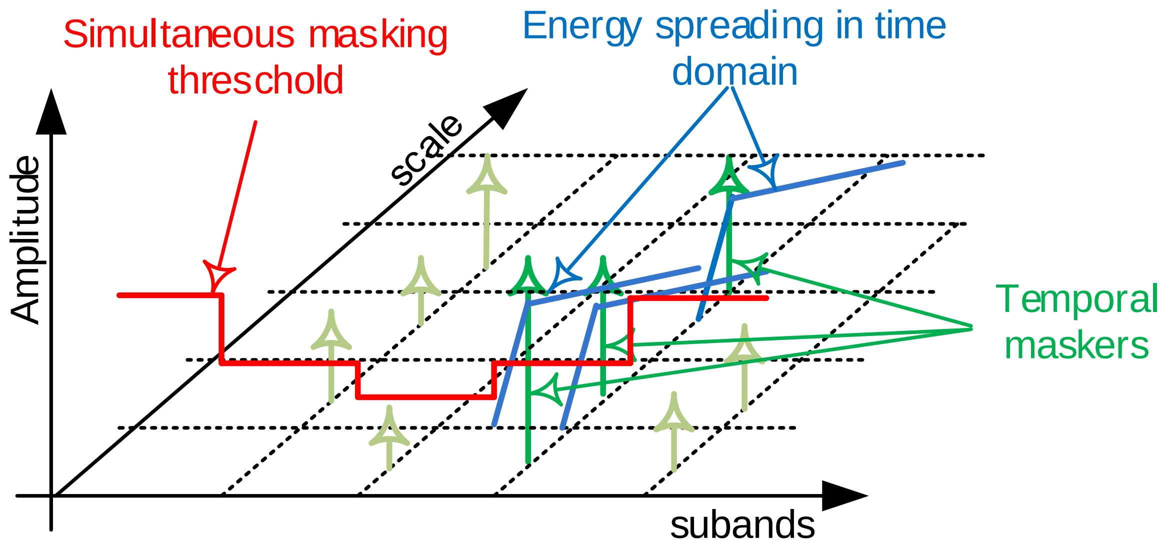
IF $WTE_{E_i} \leq WTE_{E_{i-1}}$ and $PE_{l,n} \geq PE_{l+1,2n} + PE_{l+1,2n+1}$,

THEN perform decomposition of the current level $l = l + 1$ and corresponding nodes $(l, n) \in E_j$ and transfer to new tree structure $E_j = E_{j+1}$,

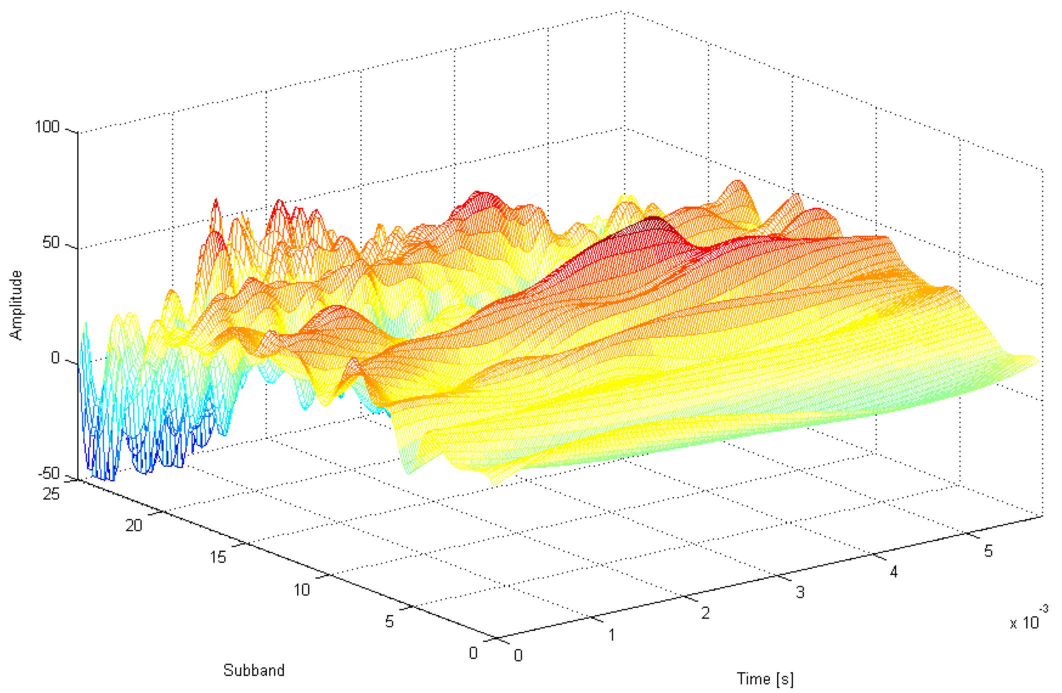
and repeat optimization procedure for next new tree structure E_j.

4. Excitation Scalogram Creation

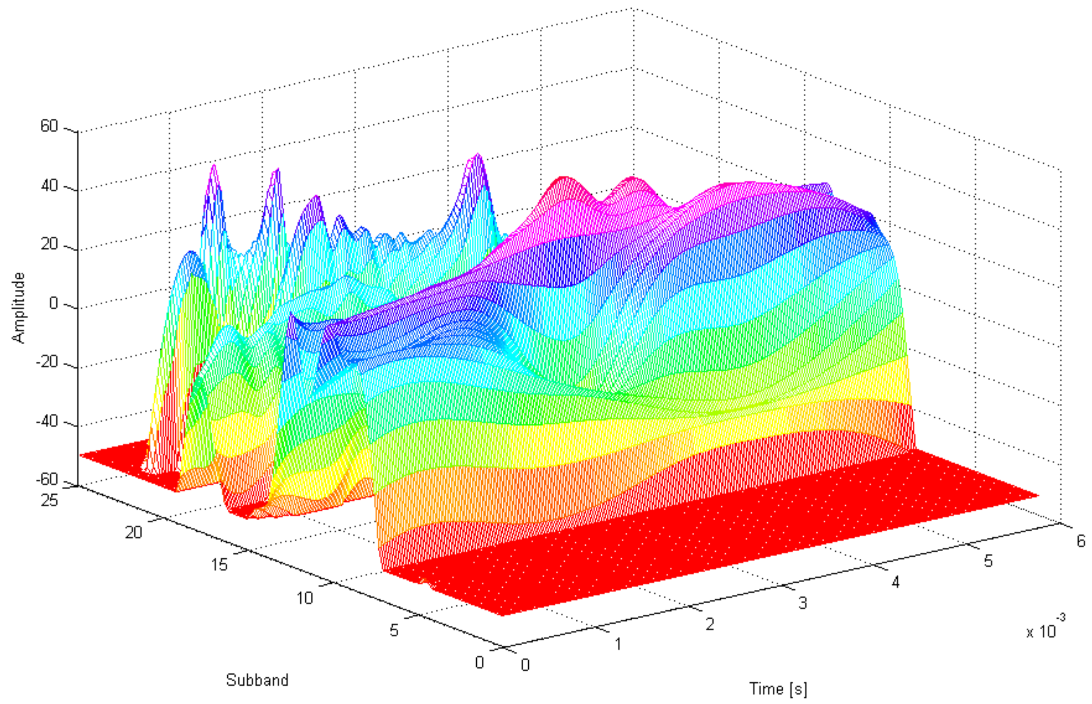
Masking thresholds[2] $T_{l,n}$ and temporal maskers[3] $F_{l,n}$ are used for excitation scalogram estimation;



Applying spreading energy functions in two domains



Excitation scalogram associated with original signal



Excitation scalogram associated with modeled signal with 5 atoms

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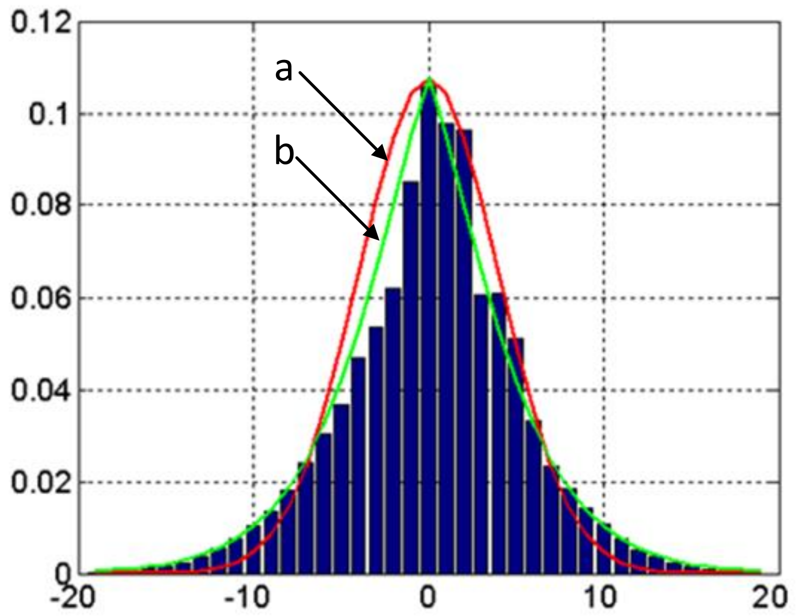
[2] A. Petrovsky, D. Krahe, A.A. Petrovsky, “Real-Time Wavelet Packet-based Low Bit Rate Audio Coding on a Dynamic Reconfigurable System”, presented at the AES 114th Convention, Amsterdam, The Netherlands, 2003 March 22-25.

[3] Al. Petrovsky, E. Azarov, A., Petrovsky, “Hybrid signal decomposition based on instantaneous harmonic parameters and perceptually motivated wavelet packets for scalable audio coding”, Elsiver, Signal Processing, Special Issue “Fourier Related Transforms for Non-Stationary Signals”, vol. 91, pp. 1489-1504 (2011, June).

5. Parameters Quantization & Coding

Example of wavelet coefficients histogram and Gauss (a) and Laplace (b) probability distribution functions.



Laplace function parameters

level, l	α	β
1	0.00001	0.12
2	0.0008	0.2
3	0.15	0.36
4	0.13	0.7
5	0.27	1.25
6	0.26	1.3
7	0.35	1.8
8	0.6	1.8

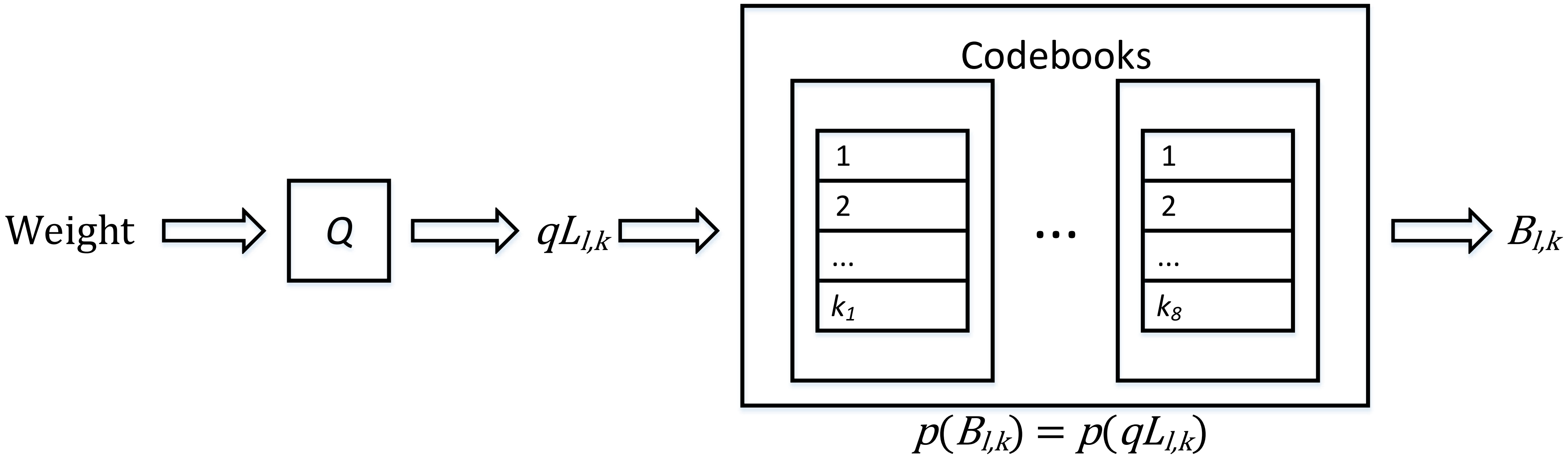
Weight quantization:

$$qL_{l,n,k} = 2\left| nint\left(\frac{|X_{l,n,k}|}{\Delta_{l,n}}\right)\right| + 1$$

$\Delta_{l,n} = \sqrt{12T_{l,n}/K_{l,n}}$ – quantization step

Quantized parameters coding:

$qL_{l,n,k}$ encoded using Huffman algorithm.



$$B_{l,k} = \left(b_{k,1}, b_{k,2}, b_{k,3}, \dots, b_{k,w_k}\right), b_{k,j} \in \{0,1\}, j = \overline{1, w_k}$$

6. WP Tree Structure Coding

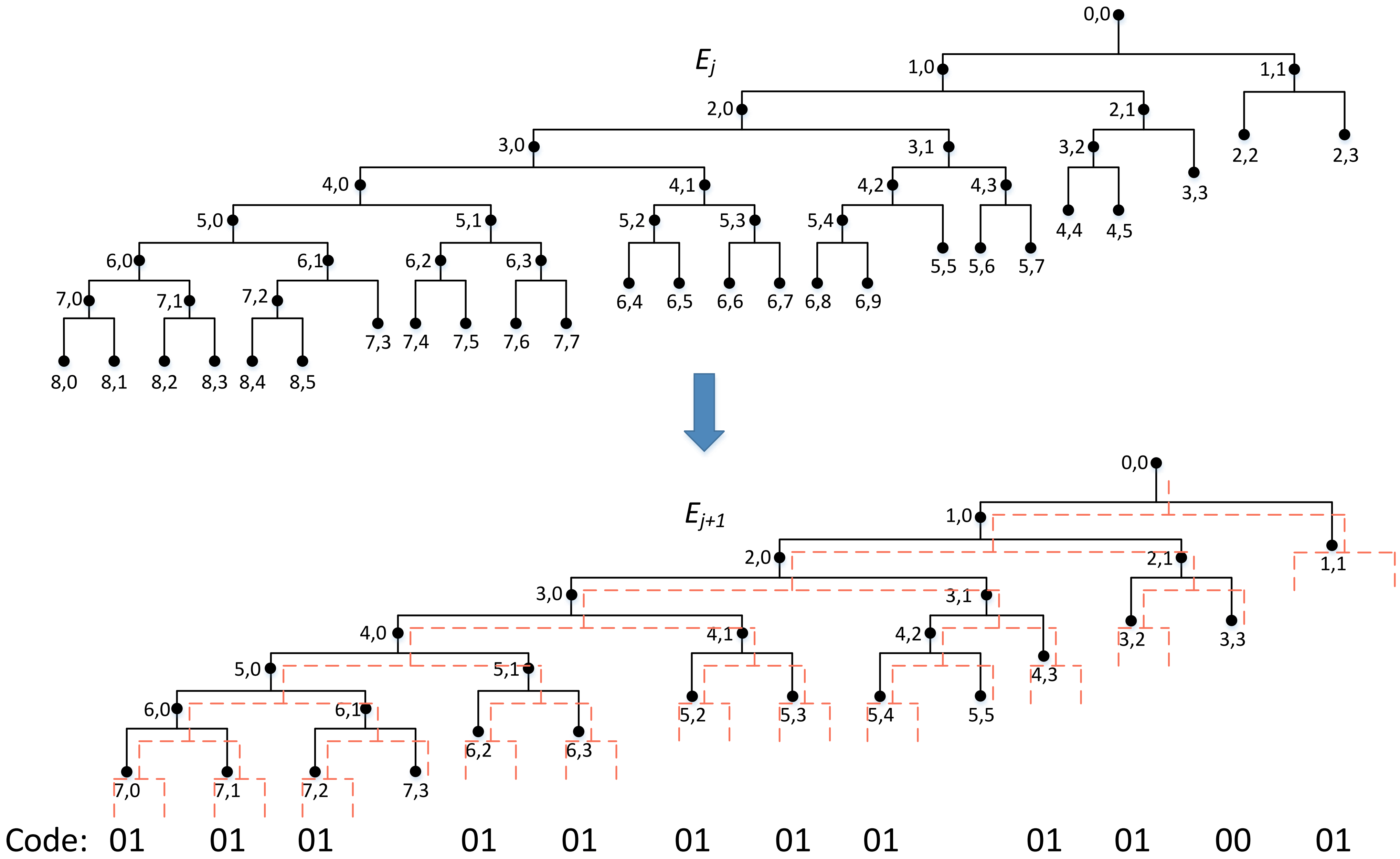
- 121 bits for straight *CB-WPD* tree coding – too many;
- Frame dependent WP tree structure coding;
- UP-DOWN one level WP tree structure grows.

#	Action	Code
1	no changes	00
2	delete node	01
3	node grows	10
4	double “no changes”	11

<u>Example:</u>

One level up of terminal nodes from E_j – *CB-WPD*, to E_{j+1}.

Up to 22 bits required.



7. An Objective Assessment[4] of the Audio Quality

	250 atoms	450 atoms	AAC
Bitrate, *kbps*	**45**	**80**	100
Compression rate	**15.6**	**8.8**	7.0

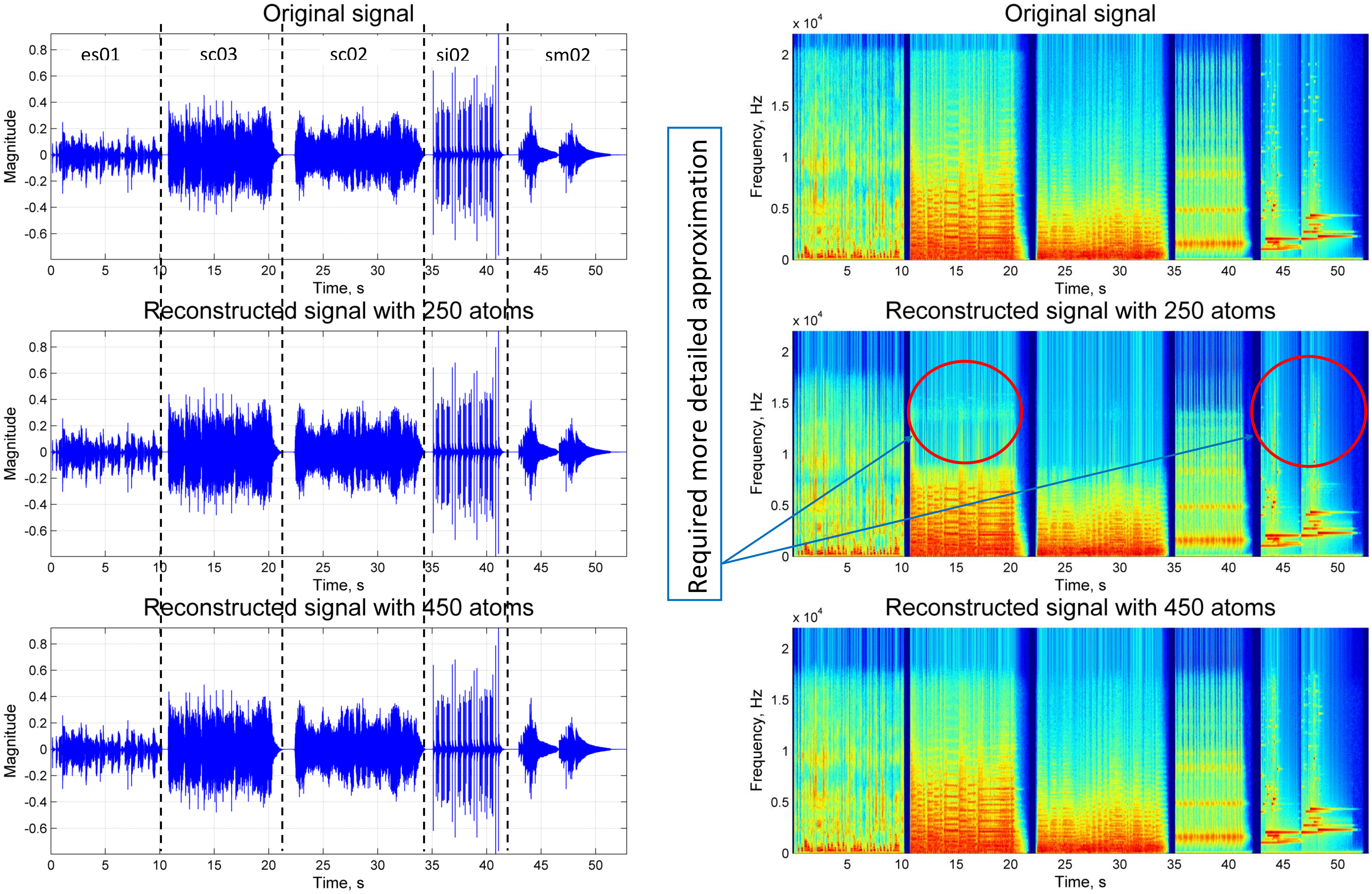
Impairment description	ODG
Imperceptible	0.0
Perceptible, but not annoying	-1.0
Slightly annoying	-2.0
Annoying	-3.0
Very annoying	-4.0

Description (44,1 *kHz*, 16 *bit*, mono)	**Proposed coder**					AAC
	250 atoms	300 atoms	350 atoms	400 atoms	450 atoms	
es01 – Vocal (Suzan Vega)	-2.1254	-1.8778	-1.4908	-1.1021	-0.8544	-0.218
es02 – German speech	-0.8315	-0.6183	-0.4641	-0.4494	-0.3854	-0.100
es03 – English speech	-1.5974	-1.4694	-1.3776	-0.8741	-0.5970	-0.132
sc01 – Trumpet solo and orchestra	-0.2116	-0.1738	-0.1706	-0.1656	-0.1624	-0.085
sc02 – Orchestra piece	-0.8659	-0.7642	-0.3608	-0.2509	-0.2165	-0.154
sc03 – Contemporary pop music	-2.5894	-1.6039	-0.8315	-0.3985	-0.3001	-0.236
si01 – Harpsichord	-2.4671	-1.8598	-1.0398	-1.0578	-0.9873	-0.483
si02 – Castanets	-3.0789	-2.7877	-1.7663	-1.4170	-1.0545	-0.918
si03 – Pitch pipe	-1.0545	-0.9036	-0.7675	-0.6380	-0.6380	-0.542
sm01 – Bagpipes	-3.3451	-2.6495	-1.1496	-0.7118	-0.6265	-0.485
sm02 – Glockenspiel	-3.2723	-2.7661	-2.3840	-2.1697	-2.1484	-0.269
sm03 – Plucked strings	-1.4563	-0.8216	-0.4543	-0.2952	-0.2099	-0.151

Additional 50 atoms add 8.6 *kbps* for the bitrate.

[4] R. Huber, B. Kollmeier, "PEMO-Q – A New Method for Objective Audio Quality Assessment Using a Model of Auditory Perception", IEEE Transactions on audio, speech, and language processing, vol. 14, pp. 1902-1911 (2006 November).

8. Encoded Audio Samples Example Results



9. Conclusions & Future Research

Conclusion:

- Perceptually optimized WP dictionary approach allows to adapt psychoacoustically WP tree structure to each signal frame and provides WP dictionary with less number of functions;
- The nonlinear nature of the algorithm leads to compact signal representation;
- Proposed scalable parametric audio encoder using sparse approximation as a core provides:
  - more than twice increase compression ratio (for some sequences 250 atoms variant provides comparable results with AAC).
  - decreasing bitrate depending on signal type.

    (For example, sc01 - 250 atoms, sc02 - 300 atoms (~53 kbps, CR – ~13), si01 - 350 atoms (~62 kbps, CR – ~11) and so on.

Future Research:

- Further parameters selection optimization and quantization algorithm improvement to increase quality of reconstructed audio signal;
- Hardware implementation of scalable parametric audio coder using sparse approximation with frame-to frame perceptually optimized wavelet packet based dictionary as a field programmable system-on-chip (FPSoC).