GENERAL-PURPOSE LISTENING ENHANCEMENT BASED ON SUBBAND NON-LINEAR AMPLIFICATION WITH PSYCHOACOUSTIC CRITERION

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1. Introduction

This paper proposes an algorithm for near end listening enhancement in noisy conditions designed for processing both speech and music.

The main features of the proposed algorithms are:

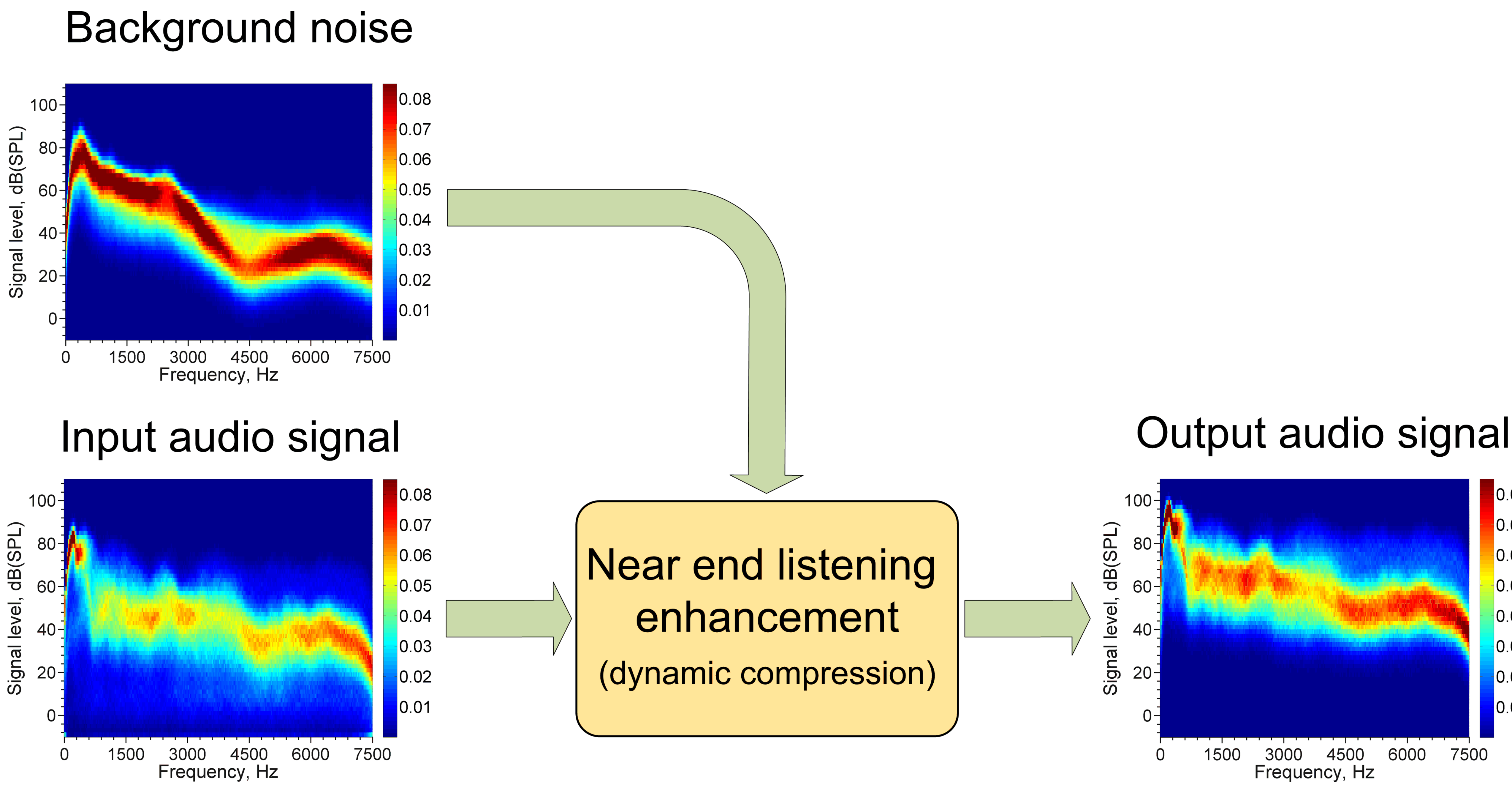
- automatic processing of sound in accordance with current spectrum of the background noise;
- correction of the output sound according to the individual hearing of the listener;
- processing algorithm is experimentally implemented as a smartphone application.

The main idea of the algorithm:

1. subband decomposition and adaptive compression of the signal;
2. audiometry and hearing compensation using frequency-dependent amplification of the signal;

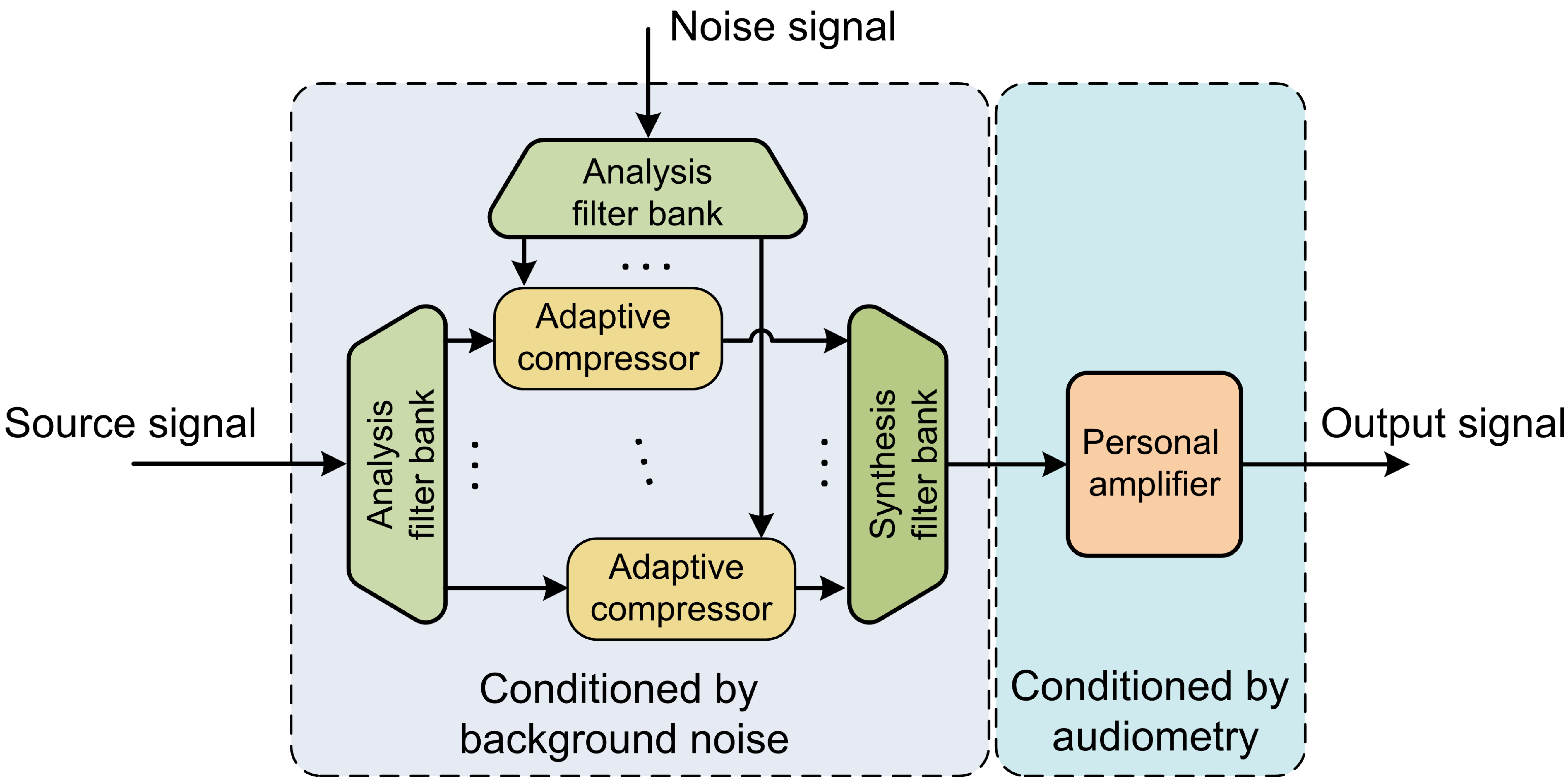
2. Masking cancelation via dynamic compression

Using dynamic compression it is possible to place all frequency components of the signal above noise masking threshold without excessive amplification of loud sounds.



3. Processing scheme

The implemented listening enhancement scheme is shown below.



4. Non-uniform cosine-modulated filter bank

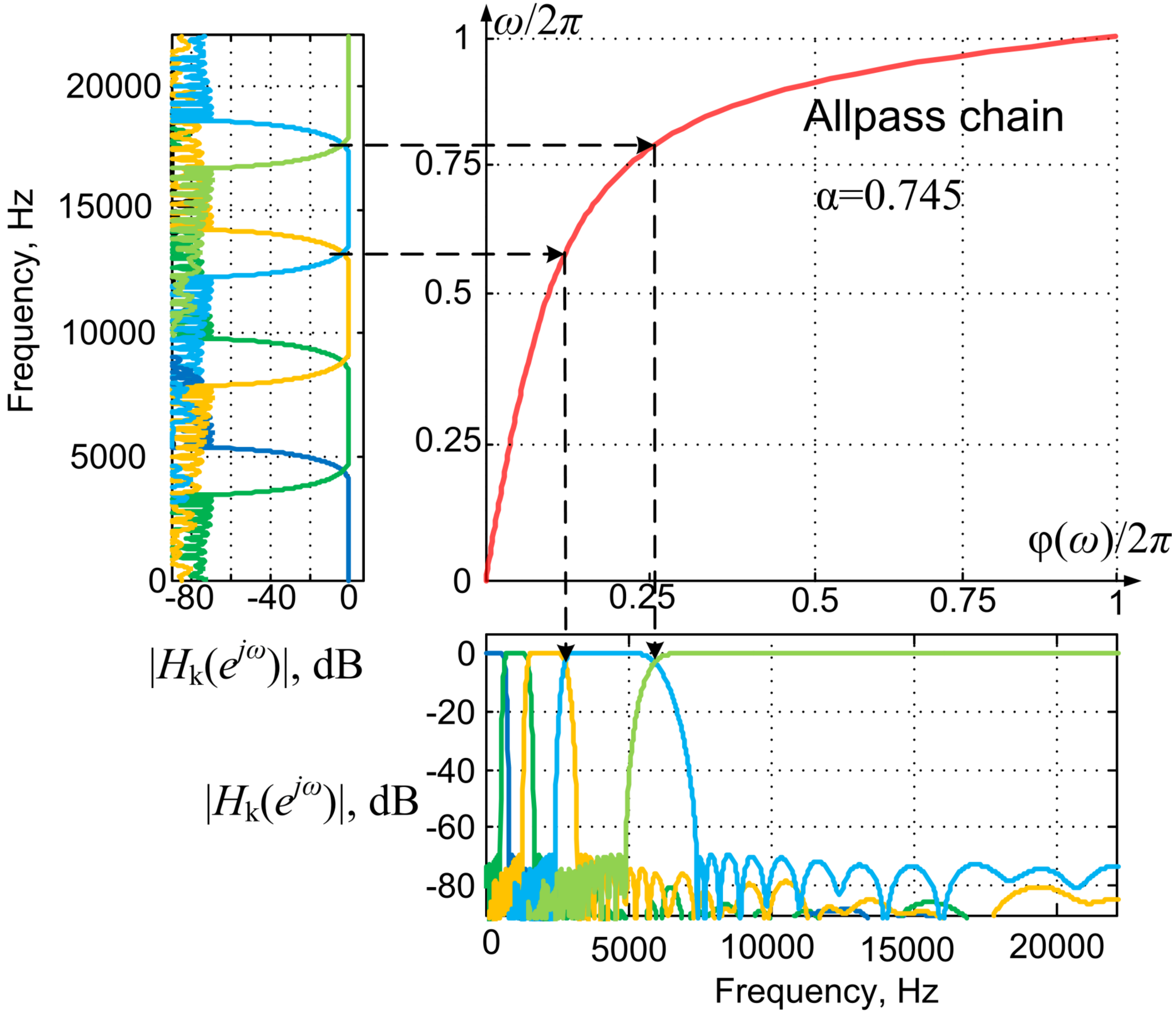
$$z^{-1} \to A(z)$$

$$A(z) = \frac{\alpha + z^{-1}}{1 + \alpha z^{-1}}$$

$$A(e^{j\omega}) = e^{j\varphi(\omega)}$$

$$\varphi(\omega) = -\omega + 2\mathrm{arctg}\left(\frac{\alpha\sin\omega}{\alpha\cos\omega - 1}\right)$$

$\alpha = 0.745$ gives an approximation of the Bark scale

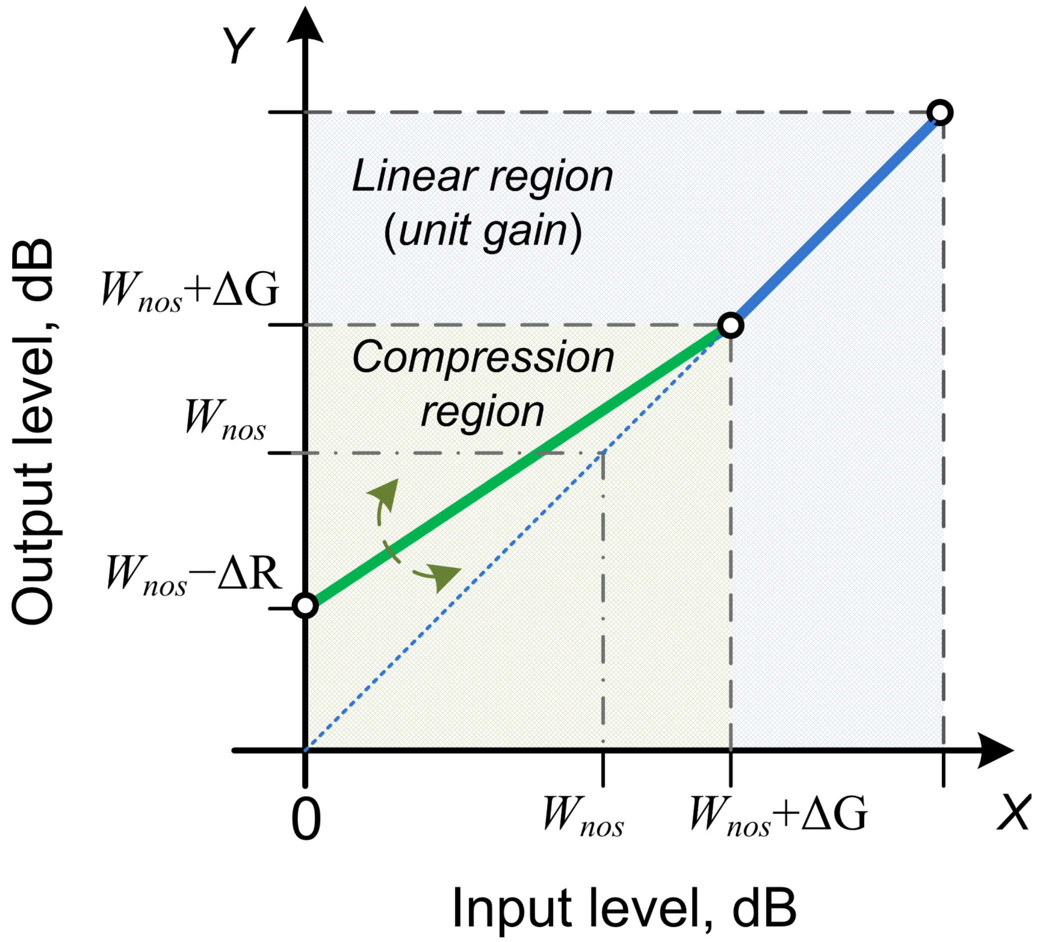


5. Dynamic compression with adaptive response curve

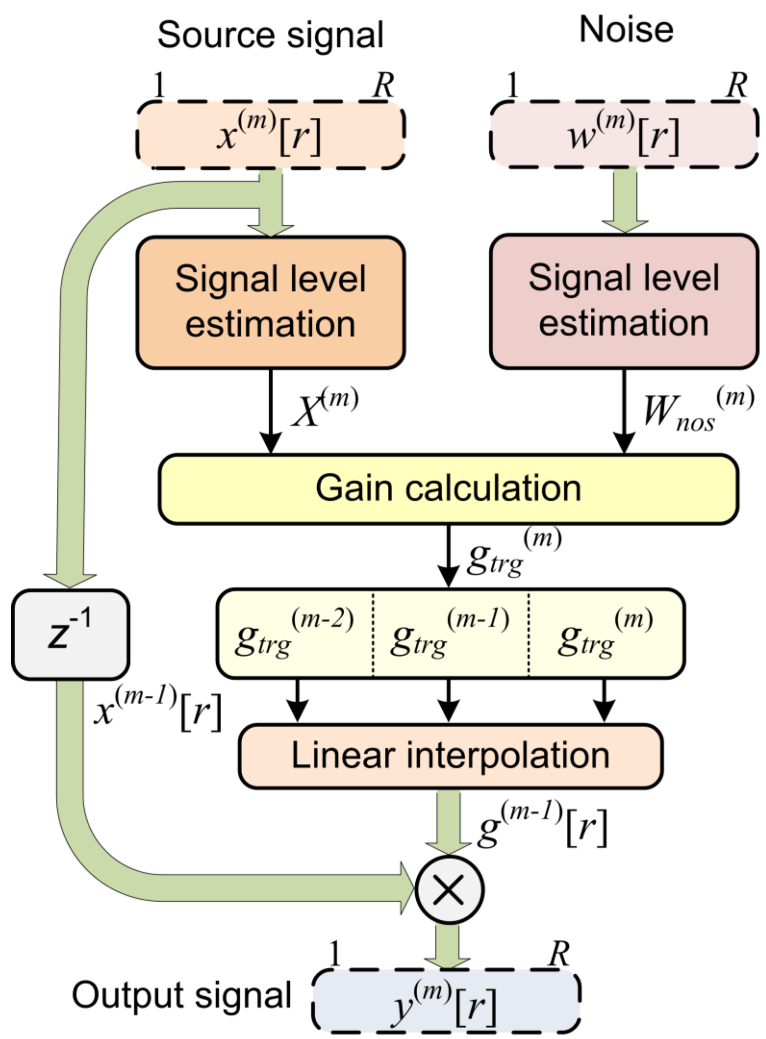
The curve is specified by three parameters:

1) noise level W_{nos} (dB) which is proportional to current noise masking threshold;
2) threshold offset from current noise level ΔG (dB);
3) gain increase offset from current noise level ΔR (dB).

Parametrical response curve



Processing algorithm



6. Personal amplification

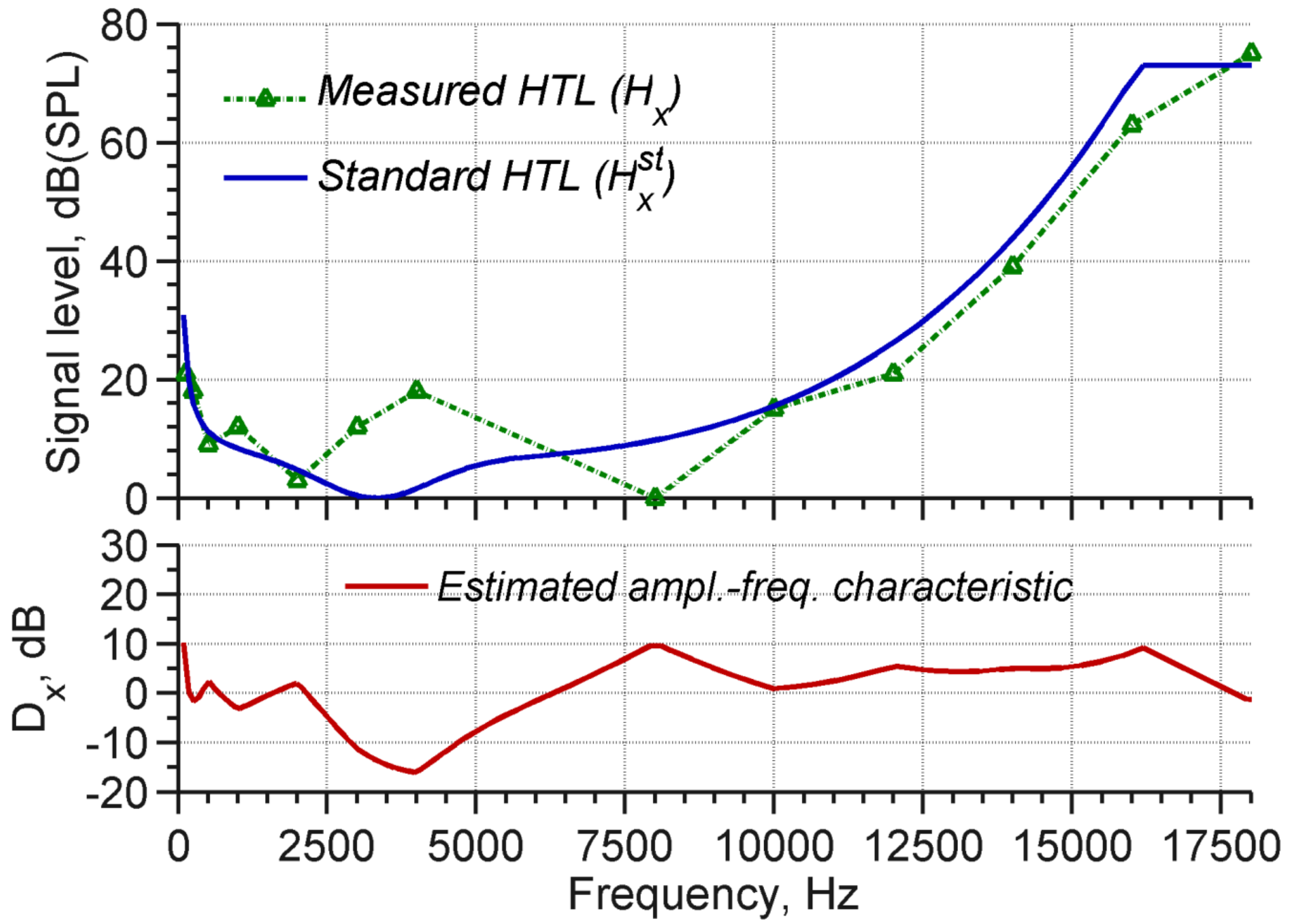
We use audiometry to measure hearing threshold levels (HTL). During playback the sound is equalized according to subjective sensitivity of the listener using accepted standards in digital hearing aids.

Assuming that hearing thresholds of the listener is normal it is possible to estimate actual amplitude-frequency response of the device.

The amplitude-frequency response D_x can be estimated from measured HTL values as

$$D_x = H_x^{st} - H_x,$$

where H_x^{st} – standard hearing threshold in quiet and H_x – measured hearing threshold for frequency x. Место для формулы.

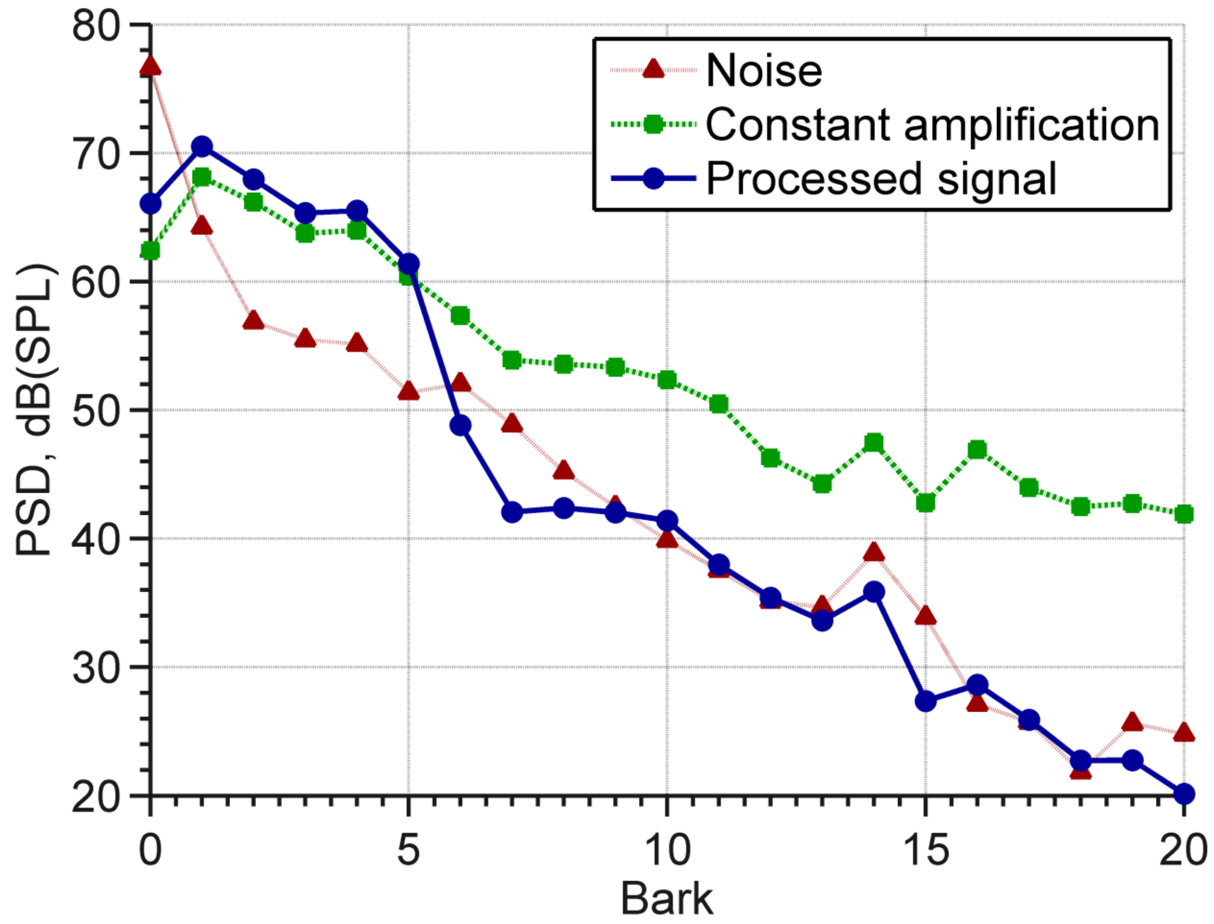


7. Experimental results

We compared sound pressure levels (SPL) for constantly amplified and enhanced signals in noisy conditions (car noise). At roughly the same intelligibility point enhanced signal has significantly attenuated high-frequency components.

We compared speech intelligibility as well using SII (speech intelligibility index).

Sound pressure level



Speech intelligibility