Instantaneous compression – a model based hearing aid algorithm

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Introduction

One main issue of hearing impairment is the so called 'Recruitment'-phenomenon, a frequency-specific elevation of the hearing threshold level and a reduced dynamic range between threshold level and uncomfortable loudness level. It is generally thought to be mainly a consequence of a loss of outer haircells in the cochlea [1]. To compensate for Recruitment single- or multi-frequency-band amplification and dynamic compression schemes are commonly used in hearing aids. Established methods use time-constants close to or shorter than the phoneme rate. The prevailing view is that this deteriorates hearing in many everyday listening conditions, because fast amplitude compression performed independently in a large number of frequency bands introduces nonlinear distortions (see [2]). In contrast to the detrimental effects found in fast multi-band systems, the healthy auditory system provides a large number of overlapping frequency bands and almost instantaneous amplitude compression due to the function of the outer haircells (see, e.g., [1]). This suggests that fast or even instantaneous compression should in principle be possible without degrading speech intelligibility. A model of this instantaneous spectro-temporal compressive auditory processing is introduced, which avoids the detrimental effects of fast multiband compression andmight help to improve compression schemes in hearing instruments in the future. An effective implementation with signal-processing in real time is possible using the master hearing aid [3], which will allow a further evaluation of the algorithm regarding speech intelligibility.

Model of instantaneous spectro-temporal compressive processing

The cochlea acts as a non-linear spectro-temporal analyzer that codes the frequency-distribution of the incoming acoustic energy (frequency-to-place transformation). The nonlinearity in the cochlear response to sound is mediated by the outer haircells, which provide a compressive saturating gain characteristics, i.e., a high gain at low input levels that decreases with level ('cochlear amplifier'). Both physiological (e.g., [4]) and psychophysical (e.g., [5]) measurements indicate a compressive response to tones at a fixed place along the basilar membrane only when stimulated with the characteristic (or: best) frequency of that place ('on-frequency tone'). The compression was found to be almost instantaneous [6]. Gain and compression decrease with increasing deviation of the stimulus frequency from the best frequency towards low frequencies, and gain decreases with increasing deviation towards high frequencies. The response to 'off-frequency tones' linearizes and the frequency resolution of the cochlear amplifier is higher than that of the passive cochlear resonant response.

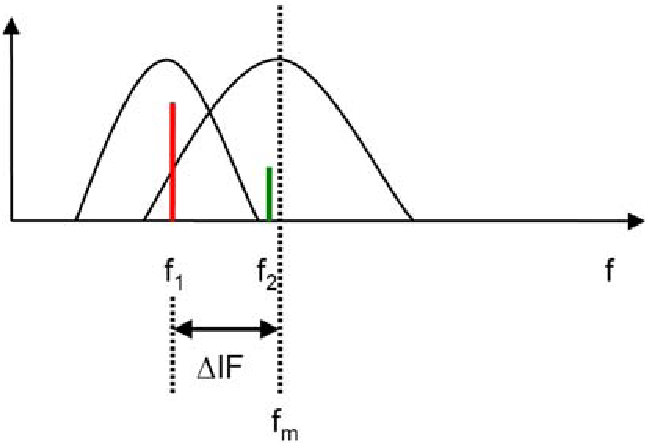

Figure 1: Sketch of the frequency response of an auditory filter with center frequency f_m (solid line, right filter) and spectrum of an on-frequency component with frequency f_2 (right bar) and an off-frequency component with frequency f_1 (left bar). On- and off-frequency components can be distinguished by measuring the instantaneous frequency at the output of the auditory filter and calculating its deviation ΔIF from the center frequency fm. Measuring the level of a component at the output of the filter does not allow for this distinction. Note that the instantaneous frequency can be measured within a time period corresponding to only a few waveforms of the narrowband signal at the filter output.

In order to simulate this selection process, Hohmann and Kollmeier [7] proposed to measure the instantaneous frequency in each frequency band of an auditory filterbank. The deviation of the inst. frequency from the bands respective center frequency is then used as a control signal for the amount of compression and gain: The higher the instantaneous frequency deviation, the less gain and compression is applied. In this way, a direct measure of the relative prominence of off-frequency components relative to the on-frequency components in each filter-band is revealed with a very high time and frequency-resolution. Figure 1 shows the idea of using the deviation in instantaneous frequency as an indicator for the presence of off-frequency components. Figure 2 shows a block diagram of the nonlinear auditory filterbank that implements a compression stage in each filterband that is steered by the deviation in instantaneous frequency.

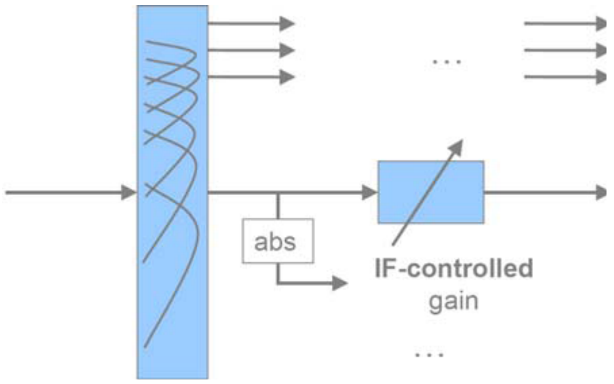

Figure 2: Block diagram of the auditory filterbank that simulates nonlinear spectro-temporal processing in the healthy cochlea (Hohmann and Kollmeier, 2007). At the output of a linear filterbank (complex-valued Gammatone filterbank (Hohmann, 2002)) that simulates the passive resonant properties of the basilar membrane, a compression stage provides instantaneous gain and compression in each filterband (simulation of the outer haircell function). For this, the level is measured ('abs'-block), from which a gain is derived and applied to the signal. The amount of gain and compression is steered by the deviation of the inst. frequency from the respective bands center frequency ('IF-controlled gain').

Results and discussion

Figure 3 shows the excitation pattern of the model (described in legend). As expected, the model with full gain and compression in all filter-bands smears out the spectro-temporal pattern of the signal and destroys the spectro-temporal contrasts (right panel). On the other hand, the model with instantaneous frequency control (middle panel) provides gain for the prominent peaks in the spectro-temporal pattern only and thus sharpens the pattern. At a certain instant of time, only a few of the auditory filters get gain and thus compression. Most of the bands do not receive full gain and compression because the energy found in these bands is more determined by off-frequency components. The selection process is very fast, i.e., gain and compression is switched on within a few waveforms of the signal after on-frequency components become prominent in a certain frequency band.

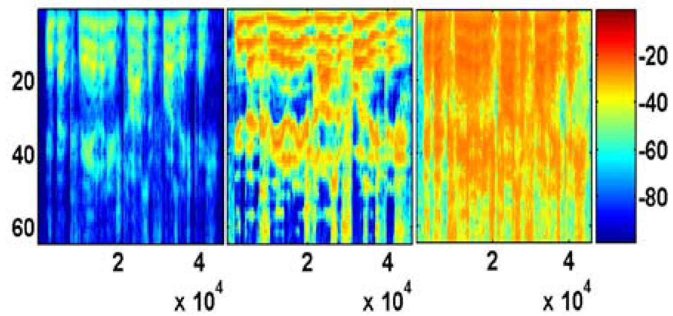

Figure 3: Excitation pattern, i.e., response of the nonlinear auditory filterbank of Fig. 2 to a speech signal (one sentence). Plotted is the magnitude of the response in dB (color-coded) re. full scale as a function of time in samples and frequency (number of channel from low (top row) to high frequencies (bottom row). Left panel: no gain and compression (passive filterbank); right panel: full gain and compression; middle: gain and compression with instantaneous frequency control.

Table 1: Measured SRT in dB SNR from 6 subjects using the Oldenburger Satztest in modulated noise with linear hearing aid simulation and instantaneous compression. Line 3 shows the benefit of the new approach.

Linear	-7,0	-13,1	-8,4	-9,8	1,7	-8,2
Inst. compression	-3,7	-15,7	-7,3	-13,0	2,6	-12,9
benefit	-3,3	2,6	-1,1	3,2	-0,9	4,7

Table 1 shows the values of measured SRT (speech reception threshold) in dB SNR from 6 hearing impaired subjects using the Oldenburger Satztest (sentence test) in modulated noise (ICRA-5-250) [8]. The new model algorithm is compared to a hearing aid simulation with a linear (in time) gain which is calculated as the mean gain from the compressed signal. The benefit values indicate that is possible to have an advantage in speech intelligibillity: 3 subjects show a positive value which means that they had a higher speech intelligibility in this situation. Further measurements are indicated to explain the differences between the subjects and to get more detailed results.

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