# Auditory Adaptation Effects described by a Signal Dependent Compression

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**Abstract**

A nonlinear signal processing method, termed Signal Dependent Compression (SDC), is introduced, which is suitable to describe a number of adaptation effects inherent in the auditory system. The SDC has been incorporated into a computational auditory model, which in this way can be adjusted to describe common simultaneous and non simultaneous masking effects. The performance of the proposed method is analyzed, by comparing model predictions to psychophysical masking data taken from the literature.

## 1 Introduction:

The principle of the SDC method is based on the generally accepted assumption that the auditory system performs some kind of signal compression. Here, it is further assumed that the signal compression is not a static function, but also to be dependent on the input signal's evolution. By adapting this dynamic compression concept, it is possible to describe a number of the effects commonly related to auditory adaptation. The term adaptation refers to a decrease in responsiveness, which occurs during and following the presentation of a stimulus [1]. Adaptation effects can be demonstrated in various psychophysical and physiological measurements, whereas the specific adaptation sources in the auditory system have not yet been established. Physiological adaptation effects can be found throughout the entire auditory nervous system, but not in the auditory periphery [1]. Psychophysical adaptation appears for example as loudness adaptation, "overshoot" in simultaneous masking and forward masking [1], whereas only forward masking is further considered here. Hence, a SDC-based masking model is introduced, which is based on accepted psychophysical studies related to simultaneous and non simultaneous masking. However, the application range of the SDC concept is not restricted to describe only masking and auditory adaptation effects. Due to the flexibility of the SDC concept and the computational very efficient implementation, it also appears to be very suitable for applications in the areas of dynamic range control (especially hearing aids), audio coding and speech recognition.

## 2 Computational auditory model

The general structure of the proposed auditory model is represented in Fig. 1. The input signal is analyzed by a preprocessing stage; the resulting signal is processed by the SDC module and then convolved with a lowpass filter w(n) (1st order LP, $f_g$ = 4Hz). The preprocessing stage is composed of gammatone bandpass filtering, half wave rectification, lowpass filtering (5th order LP, $f_g$ = 1kHz) and squaring.

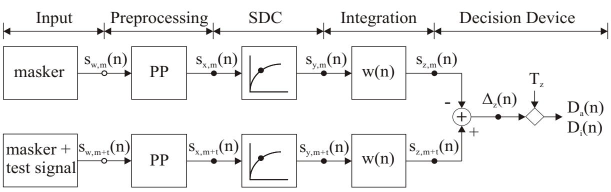


**Figure 1:** Block diagram of the proposed auditory model.

Regarding to the proposed decision device, it is based on comparison between the masker alone $s_{x,m}(n)$ and the masker plus the test signal $s_{x,m+t}(n)$, processed by the computational auditory model. The difference $\Delta_z(n)$ of the two separate output signals is calculated and compared with a predefined static threshold $T_z$. In the case, the maximum of the difference signal $\Delta_z(n)$ is above the threshold $T_z$, the test signal is considered to be audible and otherwise to be inaudible. The masked threshold ($\max\{\Delta_z(n)\} = T_z$) is determined by adjusting the level of the test signal by a one-up one-down procedure. Given that the principal signal processing is equal for all frequency bands, only one frequency band will is considered. Due to the fact that the SDC module is the main part of the auditory model and given that the other modules are based on already existing realizations, only the SDC module will be described in detail.

## 3 The SDC concept

The SDC can be seen as an abstraction of a discrete nonlinear device applied in electronics (e.g. Transistor), which is realized by a digital signal-processing methodology (Fig. 1). Such a discrete device is characterized by its input-output function and an operating point, which is usually adjusted by a DC-offset. Following the transistor analogue, the static operating point is replaced by a dynamic operating point signal, which is directly derived by a nonlinear lowpass filtering of the input signal itself. Obviously, the value of the operating point signal directly determines the rate of compression. However, the SDC structure can in principal be split up into a static component (the function Ψ) and a dynamic component, which is introduced by the operating point signal. It is suggested that the static function Ψ in combination with the squaring included in the preprocessing (Fig. 2) compose the basilar membrane nonlinearity, which for example is described by [2], and the dynamic component describes adaptation processes.

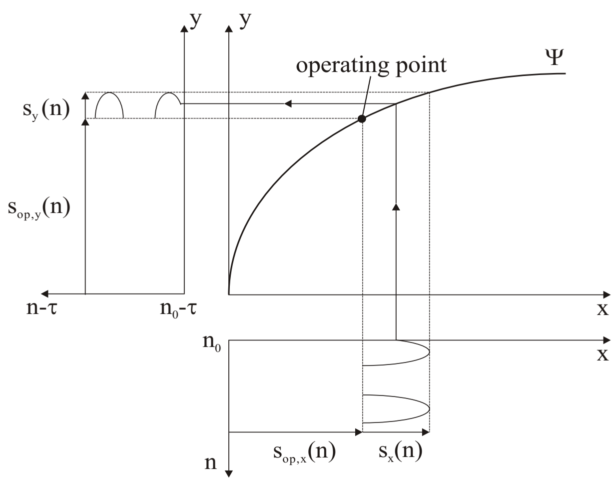


**Figure 2:** Illustration of the operating principle of the SDC.

The basic mathematical formulation of the proposed SDC structure shown in Fig. 2 is given by

$$s_y(n) = \Psi\left\{s_x(n) + s_{op,x}(n)\right\} - \Psi\left\{s_{op,x}(n)\right\} \qquad (1)$$

with $s_x(n)$ the input signal, $s_y(n)$ the output signal and $s_{op,x}(n)$ the input related operating point signal.

### 3.1 SDC implementation

The block diagram of the proposed SDC structure realized on a signal-processing platform (software or hardware) is presented in Fig. 3.

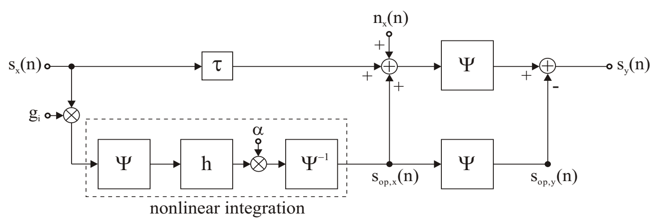


**Figure 3:** Block diagram of the proposed signal processing realization of the SDC.

Two main signal paths can be observed, one path for the signal $s_x(n)$ and one for the operating point signal $s_{op,x}(n)$, converging in two summation points. The linear integrator h(n) approximates a hyperbolic function and is realized by a summation of three first order IIR lowpass filters. The nonlinear function $\Psi$ is given by

$$\Psi\{x\} = 10 \cdot \lg(x+1) \qquad (2)$$

Jestead et al [3] described the masker level $L_m$ and test signal delay $d_t$ dependency of forward masking for moderate masker level and test tone delays by

$$M_{dB} = a \cdot \left[b - \lg\left(d_t\right)\right] \cdot \left[L_m - c\right] \qquad (3)$$

with a, b and c some constants. The present SDC structure is based on a mathematical concept, which approximates this function for forward masking and in this way, the introduction of the constants $g_i$ and $\alpha$, the filter h(n) and the nonlinear function $\Psi$ was motivated.

**4 Model simulations**

The reference data for the simulated masked thresholds (MT) are based on the psychophysical data measured by [4]. Given that only broad band noise masking brief test tones are considered, off-frequency listening effects can be neglected and only the single frequency band centered around the test tone has to be regarded. Throughout all simulations, the same parameter-set has been employed.

In Fig. 4 the reference data and the model simulations of the masker level and test tone delay dependency of the forward masked threshold (FMT) are represented. The higher the level of the masker, the faster the temporal decay of the FMT. Furthermore, all FMTs converge into the threshold in quiet after almost the same test tone delay. Obviously the model simulations are in good agreement with the reference data and show the expected behavior described by equation 3, which is indicated by the dotted lines.

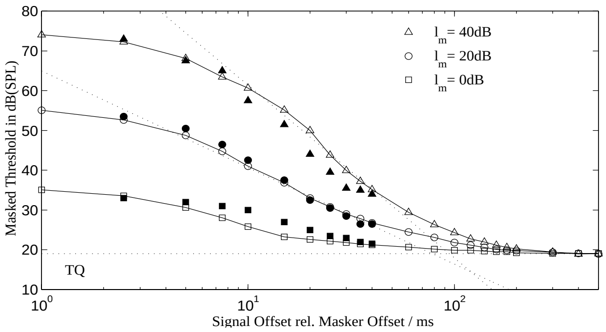


**Figure 4:** Masker spectrum level $l_m$ and test tone delay $d_t$ dependency of the FMT. ($D_m$ = 200ms, $f_0$ = 1kHz, $D_t$ = 10ms)

In Fig. 5 the influence of the masker duration on the FMT for both the model simulations and the reference data is represented. For shorter masker, the masked threshold decays faster with time and thus generates less masking. Additionally the artificial FMT simulations for $s_{op,x}(n) = 1$, representing the case the adaptation is absent, is shown. The model simulations are in good agreement with the reference data and show the expected behavior reported in the literature [5].

In Fig. 6 the reference data and the model simulations for the masker level dependency of the simultaneous masked threshold (SMT) are represented. For low masker level the SMT converges into the threshold in quiet (TQ) and for masker level considerably above the TQ, the SMT increases linearly with increasing masker level. The model simulations and the reference data are in good agreement and as expected from the relevant literature [5] for masker level considerably above the TQ, they comply with Webber's law.

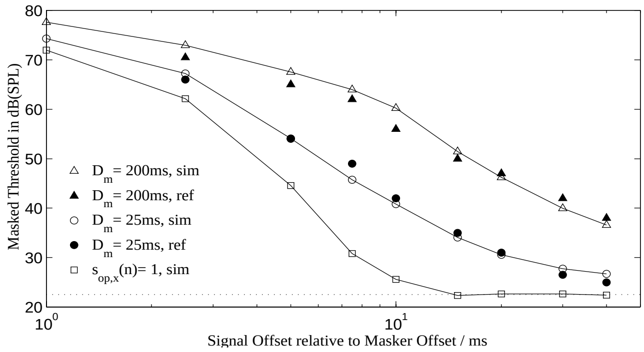


**Figure 5:** Effect of the Masker duration $D_m$ on the FMT. ($l_m$ = 40dB, $f_0$ = 2kHz, $D_t$ = 5ms)

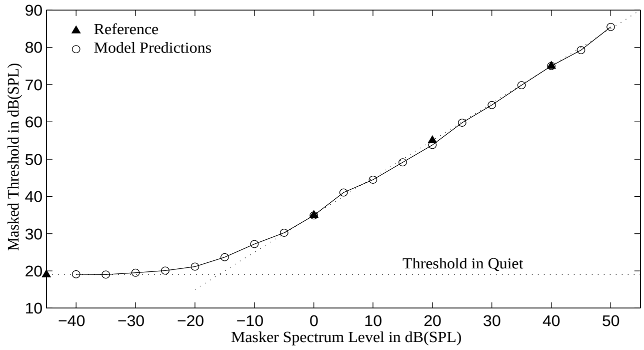


**Figure 6:** Masker level $l_m$ dependency of the SMT. ($l_m$ = 40dB, $D_m$ = 200ms, $D_t$ = 10ms, $f_0$ = 1kHz, $d_t$ = -100ms).

**5 Summary**

A novel computational model has been introduced, which is based on the mathematical concept of a signal dependent compression. The approach is motivated by widely accepted psychophysical results on simultaneous and non simultaneous masking experiments. Furthermore a very efficient computational realization of the SDC structure on a signal processing platform (hardware or software) has been described.

**6 Acknowledgement**

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**7 Literature**

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