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Analysis of the IHC Adaptation for the Anthropomorphic Speech Processing Systems

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We analyse the properties of the physiological model of the adaptive behaviour of the chemical synapse between inner hair cells (IHC) and auditory neurons. On the basis of the performed analysis, we propose equivalent structures of the model for implementation in the digital domain. The main conclusion of the analysis is that the synapse reservoir model is equivalent in its properties to the signal-dependent automatic gain-control mechanism. We plot guidelines for creation of artificial anthropomorphic algorithms, which exploit properties of the original synapse model. This paper also presents a concise description of the experiments, which prove the presence of the positive effect from the introduction of the depicted anthropomorphic algorithm into feature extraction of the automated speech recognition engine.

Keywords and phrases: inner hair cell (IHC), Meddis IHC model, IHC adaptation, auditory models, modulation spectrum filtering.

1. INTRODUCTION

1.1. *Anthropomorphism, psychoacoustics, and auditory physiology*

Many contemporary speech processing techniques tend to reflect properties of the human auditory apparatus. As a rule, most of the information about the way human beings process acoustic data comes into artificial applications from the field of psychoacoustics (for classical psychoacoustics work, refer to [1]).

Apart from the experiments with subjects that have reliably diagnosed and anatomically localised auditory pathology, psychoacoustics treats the whole human auditory system as a "black box" and tries to infer its properties without particular interest to its internal structure. Most of the psychoacoustical experiments include analysis of the responses to "simple" sounds, like pure tones, wideband noise, coloured noises, clicks, and so forth. But a lot of evidence (simultaneous and nonsimultaneous masking, pitch perception, etc.) points to the fact that the auditory system is essentially a nonlinear system.

From the system identification theory, it is known that the response of the linear system to an arbitrary excitation can be derived from the study of responses of such system to simple sounds, for example, tones, noises, and clicks. There is no need to study the internal structure of the linear black box as far as responses to the simple input signals are known. Strictly speaking, for the case of nonlinear systems, this black box approach is not applicable. There are mainly two possibilities to model a nonlinear system: either to construct a semiparametric statistical learning machine, a "neural-network-like" structure, and let it adapt through a kind of learning algorithm, or follow the parametric approach and somehow infer the internal structure of the nonlinear system to be modelled, parse it into smaller and, hopefully, simpler building blocks, then tune parameters of those blocks, so that model response matches that of the original system.

The first alternative suffers from the problems in creating the representative training set, as well as from the absence of a priori information regarding the required model complexity. The mentioned difficulties virtually prohibit application of this approach to the auditory modelling. The second of the mentioned approaches corresponds to the physiologically grounded studies of the auditory apparatus.

Among the solutions, which could benefit most from the employment of the physiological models, one can name the development of cochlear implants, the objective and quantitative quality assessment of the coded audio reconstruction, anthropomorphic audio coding, and automated speech recognition applications. While the first two mentioned branches are concentrated on the closest possible literal reproduction of the auditory apparatus properties in the artificial device, the latter imply a computationally efficient way to implement the "biological" audio processing algorithm with a certain predefined precision.

In spite of being precise and objective, the physiological hearing models neither provide a clear signal processing interpretation of those phenomena, nor give a ready answer regarding the relevance of the modelled phenomena to the hearing process in general. Thus, straightforward application of the physiological models to the fields of audio coding and speech recognition may not easily gain advantage over the conventional algorithms [2]. Before the employment of a certain physiological model into the mentioned applications, one should answer the questions of why it is important (i.e., what result is expected from it) and what is the most efficient way of its implementation. This reasoning leads to a conclusion that the further analysis of the available physiological models with the aim of finding their algorithmical interpretation is needed. This paper is further devoted to such kind of analysis.

Particularly, we are aiming at analysing the adaptation of the chemical "inner-hair-cell auditory nerve" (IHC-AN) synapse, and trying to infer its importance to the artificial anthropomorphic audio signal (and particularly speech) processing systems in adverse environments. Indeed, strong onset responses of the auditory nerve (AN) fibers to the presented stimulus are followed by the "adaptation", that is, gradual decrease of the response amplitude over time while the stimulus amplitude remains constant. This "adaptive strategy" at first glance seems to be advantageous since it allows an emphasis of nonstationarities within the incoming signal.

2. RESERVOIR MODEL OF IHC-AN CHEMICAL SYNAPSE

Physiological research into the way the inner ear converts an acoustical stimulation into a response of the auditory nerve fibers (for a brief summary and review, refer to [3]) among many other findings led to the conclusions that

(i) inner hair cells are mechanical vibrations sensory cells;
(ii) each IHC makes chemical synapses with approximately 10–30 peripheral axons of primary bipolar neurons which cell bodies contained in the spiral ganglion and modiolar axons forming the auditory (VIIIth) nerve;
(iii) one can distinguish three groups of afferent neurons based on the level of their spontaneous activity: low-spontaneous rate, medium-spontaneous rate, and high-spontaneous rate fibers. The level of spontaneous activity of the fiber is closely related to the form and the size of the synapse it formed with IHC;
(iv) chemical nature determines the following properties of IHC-AN synapses: adaptive responses, synaptic delays, quantised response amplitudes.

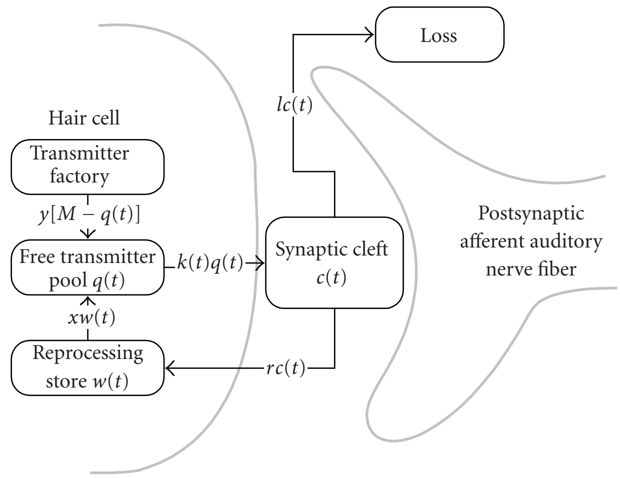

Figure 1: Schematic representation of the Meddis reservoir model.

Properties of the chemical IHC-AN synapse are successfully captured by the so-called "reservoir models," in which neurotransmitter is produced and stored in the IHC to be released in accordance with IHC transmitter release probability that changes with mechanical vibrations in the inner ear. First reservoir models for IHC-AN synapses were proposed as early as [4, 5].

Meddis has put forward [6] and further developed [7, 8, 9, 10, 11] a model of IHC, which includes a version of reservoir model of chemical synapse. The latest model [10, 11] allows for a nice fit between experimental and model data for all thee groups of IHC-AN synapses (low-, medium-, and high-spontaneous rate fibers) with only calcium conductance parameters being changed.

It must be noted here that in reality neurotransmitter release into synaptic cleft is a probabilistic and quantal process. However, to a certain degree, the dynamical properties of the synapse may be reflected by the model that assumes that neurotransmitter flow is deterministic and continuous. From the practical point of view, this assumption corresponds to the averaging of the synapse response over many identical stimulations. Latest Meddis models [9, 10, 11] depart from this assumption offering better correspondence to the data recordings of individual experiments. For the purpose of the analysis of the core properties of IHC-AN synapse and construction of the anthropomorphic artificial algorithms, we further narrow our consideration to the deterministic and continuous case.

Meddis version of the reservoir model is represented by schematic drawing in Figure 1, and is described by the set of (1). "Free transmitter pool" is the main storage facility for the transmitter that is immediately ready to be released from the cell to the "synaptic cleft." It is filled with neurotransmitter coming from the "transmitter factory" as well as that recycled at the "reprocessing store." Neurotransmitter is being released into "synaptic cleft" with a certain rate, dependent upon IHC stimulation, as well as instantaneous quantity of the stored transmitter. From the "synaptic cleft," transmitter is either being returned to the cell for reprocessing or lost by diffusion.

We assume that the pool capacity equals M. The quantity of the transmitter stored in the pool at a certain time instant will be denoted by $q(t)$. The rate, at which the factory produces new transmitter, is proportional to the free volume of the pool $y[M - q(t)]$, here operation $[\cdots]$ constitutes the choice of the biggest value between zero and the value inside square brackets. Alternatively we may put that coefficient y becomes zero at the moment the pool is filled to the limit. We denote the instantaneous amount of the transmitter in the reprocessing by $w(t)$. The recirculation rate is proportional to the amount of the transmitter in the reprocessing $xw(t)$. The rate, at which transmitter is sent to the cleft, is equal to the product of membrane permeability $k(t)$ and the quantity of the transmitter in the pool $q(t)$. The quantity of the neurotransmitter in the cleft at certain instant will be denoted by $c(t)$. Rates of neurotransmitter loss and return for reprocessing are proportional to the amount of the transmitter in the clefts $lc(t)$ and $rc(t)$, respectively.

As it follows from the above-presented description, Meddis version of the reservoir model is described by the following set of differential equations:

$$\begin{aligned} \frac{dq(t)}{dt} &= xw(t) + y[M - q(t)] - k(t)q(t), \\ \frac{dc}{dt} &= k(t)q(t) - (l+r)c(t), \\ \frac{dw}{dt} &= rc(t) - xw(t). \end{aligned} \tag{1}$$

Initial conditions of the model are taken in accordance with the assumption that at a certain instant t_0 the system is in an equilibrium state:

$$\begin{aligned} xw(t_0) + y[M - q(t_0)] &= k(t_0)q(t_0), \\ k(t_0)q(t_0) &= (l+r)c(t_0), \\ rc(t_0) &= xw(t_0). \end{aligned} \tag{2}$$

3. ADAPTATION PROPERTY OF THE RESERVOIR MODEL OF IHC-AN CHEMICAL SYNAPSE

Figure 2 presents a typical response of the Meddis model to the excitation. Signal $k(t)$ is an input to the reservoir model and is computed by earlier stages of cochlear model (the cochlear filter bank [12] in combination with the first part of IHC model [10]) when the test tone of 6 kHz is presented. IHC medium-spontaneous rate fiber model gets its input from the cochlear filter bank section with the closest to 6 kHz centre frequency. It is running at the sampling frequency of 16 kHz.

Typical values of the model coefficients were taken from the works of Meddis [6, 7, 8, 9] and are as follows:

$$\begin{gathered} M = 10, \qquad x = 66.3, \qquad y = 10, \\ l = 2580, \qquad r = 6580. \end{gathered} \tag{3}$$

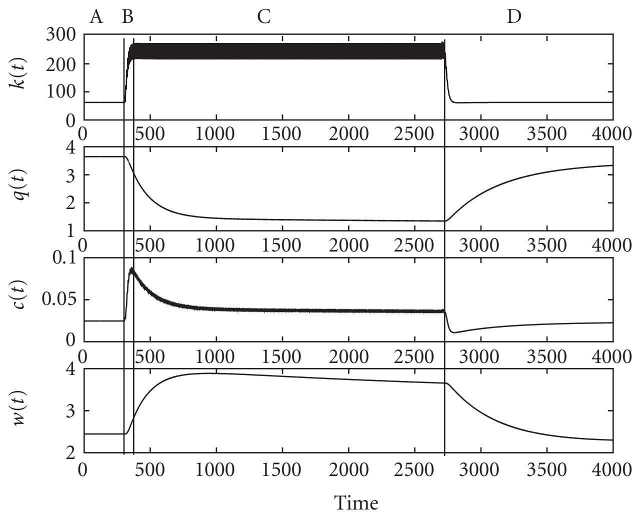

Figure 2: Reservoir model response to the excitation with the 6 kHz tone, CF ~ 6 kHz, $F_s = 48$ kHz, medium-spontaneous rate fiber. A: steady state, B: onset, C: adaptation, D: offset.

In order to perform this digital simulation (depicted in Figure 2) of the synapse model, the forward difference approximation of the set of differential equations (1) was used, as it is advised in [8].

As it can be seen from the above figure, there are four distinct regions in the model response signal $c(t)$: steady-state response to a long-term absence of stimulation (denoted as region A); onset response (region B)—brief rise of the response level to higher values; subsequent adaptation of the response level to a much lower activity (region C); offset region (region D), when synapse recovers from the stimulation and response level slowly converges to a steady-state level.

For a detailed review of adaptation properties of IHC, please refer to [11].

4. ANALYSIS OF THE RESERVOIR MODEL OF IHC-AN CHEMICAL SYNAPSE

Looking at the equation set (1), one can easily notice that functions $c(t)$ and $f(t) = k(t)q(t)$ are linked with the linear constant-coefficient differential equation of the first order with zero-free member:

$$\frac{dc(t)}{dt} + (l+r)c(t) = f(t). \tag{4}$$

Thus, (4) describes a linear time invariant system, which performs transformation of $f(t)$ into $c(t)$. Taking forward difference approximation of the differential problem and assuming that both functions take discrete values at discrete-time instances, it is possible to approximate this system with

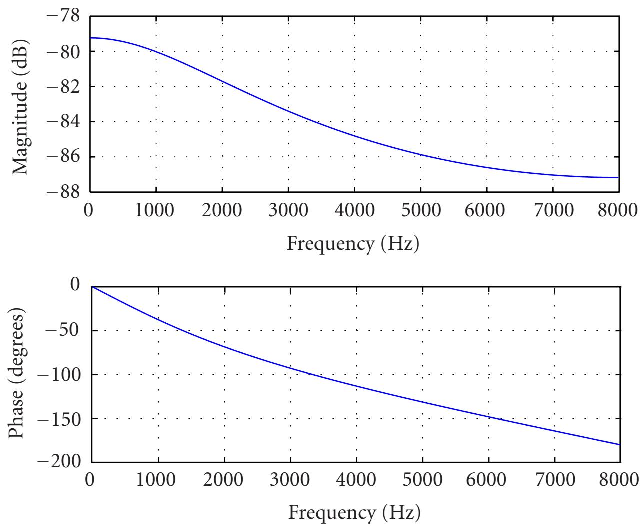

Figure 3: Frequency characteristic of filter A ($F_s = 16\,\text{kHz}$).

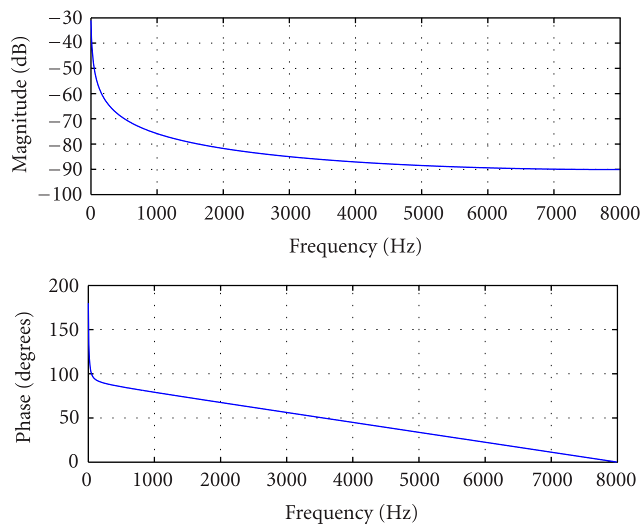

Figure 4: Frequency characteristic of filter B ($F_s = 16\,\text{kHz}$).

a digital filter:

$$c(n) = \frac{1}{F_s} f(n-1) - \frac{(l+r-F_s)}{F_s} c(n-1), \tag{5}$$

$$H_A = \frac{(1/F_s)z^{-1}}{1-\left(\left(1-(l+r)/F_s\right)\right)z^{-1}}. \tag{6}$$

Here F_s denotes the sampling frequency. We will further refer to this filter as "filter A." With the typical values of parameters l and r, this filter is a lowpass filter, which has rather smooth slope response characteristic that is presented in Figure 3.

Further analysis of the equation set (1) leads to a conclusion that functions $s(t) = M - q(t)$ and $f(t) = k(t)q(t)$ are also linked with the linear constant-coefficient differential equation of the first order with zero-free member:

$$\begin{aligned}\frac{d^3s(t)}{dt^3} &+ (x+y+l+r)\frac{d^2s(t)}{dt^2} + \left((x+y)(l+r)+xy\right)\frac{ds(t)}{dt} \\ &+ xy(l+r)s(t) = \frac{d^2f(t)}{dt^2} + (x+l+r)\frac{df(t)}{dt} + xlf(t).\end{aligned} \tag{7}$$

We note that this equation is valid for all such values $s(t) = M - q(t) \geq 0$. If $s(t) = M - q(t) \leq 0$, then it must be substituted with the following equation, which is obtained from (7) by letting $y = 0$:

$$\begin{aligned}\frac{d^3s(t)}{dt^3} &+ (x+l+r)\frac{d^2s(t)}{dt^2} + x(l+r)\frac{ds(t)}{dt} \\ &= \frac{d^2f(t)}{dt^2} + (x+l+r)\frac{df(t)}{dt} + xlf(t).\end{aligned} \tag{8}$$

The performed digital simulations show that for realistic input signals and reasonably high sampling frequency, it is enough to use (7) only.

Again it is possible to approximate the system described by (7) with a digital filter:

$$\begin{aligned}s(n) &= \frac{b_1}{a_0}f(n-1) + \frac{b_2}{a_0}f(n-2) + \frac{b_3}{a_0}f(n-3) \\ &\quad - \frac{a_1}{a_0}s(n-1) - \frac{a_2}{a_0}s(n-2) - \frac{a_3}{a_0}s(n-3), \\ a_0 &= F_s^3, \\ a_1 &= -3F_s^3 + F_s^2(x+y+l+r), \\ a_2 &= 3F_s^3 - 2F_s^2(x+y+l+r) + F_s\left((x+y)(l+r)+xy\right), \\ a_3 &= -F_s^3 + F_s^2(x+y+l+r) \\ &\quad - F_s\left((x+y)(l+r)+xy\right) + xy(l+r), \\ b_1 &= F_s^2, \\ b_2 &= -2F_s^2 + F_s(x+l+r), \\ b_3 &= F_s^2 - F_s(x+l+r) + xl.\end{aligned} \tag{9}$$

We denote this filter as "filter B." This is a lowpass filter with rather sharp frequency response characteristic (see Figure 4) for typical values of parameters x, y, l, and r.

Filter B has two real zeros and three real poles:

$$n_{B1,2} = 1 - \frac{1}{2F_s}\left((x+l+r) \pm \sqrt{(x+l+r)^2 - 4xl}\right), \tag{10}$$

$$p_{B1} = 1 - \frac{l+r}{F_s}, \qquad p_{B2} = 1 - \frac{x}{F_s}, \qquad p_{B3} = 1 - \frac{y}{F_s}. \tag{11}$$

The above conclusions imply that there must be a link between functions $c(t)$ and $s(t) = M - q(t)$ in the form

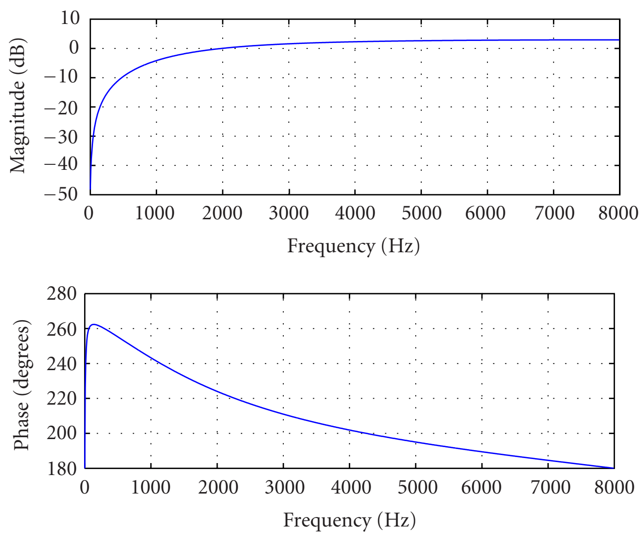

Figure 5: Frequency characteristic of filter C ($F_s = 16$ kHz).

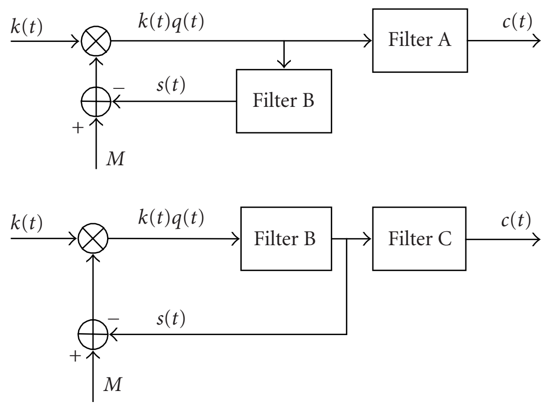

Figure 6: Reservoir model equivalent structures.

of the linear constant-coefficient differential equation of the first order with zero-free member. Indeed, it is the case

$$\begin{aligned}\frac{d^2c(t)}{dt^2} &+ (x+l+r)\frac{ds(t)}{dt} + xlc(t) \\ &= -\frac{d^2s(t)}{dt^2} - (x+y)\frac{ds(t)}{dt} + xys(t).\end{aligned} \tag{12}$$

As in the case of (7), this equation is valid for $s(t) = M - q(t) \geq 0$, if it is less than zero, then y in (12) should be put to zero.

The digital filter, which is equivalent to the system (12), is defined as follows:

$$\begin{aligned}s(n) &= \frac{d_0}{c_0}f(n) + \frac{d_1}{c_0}f(n-1) + \frac{d_2}{c_0}f(n-2) \\ &\quad - \frac{c_1}{c_0}s(n-1) - \frac{c_2}{c_0}s(n-2), \\ c_0 &= F_s^2, \\ c_1 &= -2F_s^2 + F_s(x+l+r), \\ c_2 &= F_s^2 - F_s(x+l+r) + xl, \\ d_0 &= -F_s^2, \\ d_1 &= 2F_s^2 - F_s(x+y), \\ d_2 &= -F_s^2 + F_s(x+y) - xy.\end{aligned} \tag{13}$$

We will further denote this filter as "filter C." It is a high-pass filter with rather sharp frequency response characteristic (see Figure 5) for typical values of its parameters.

We also note that a cascade connection of filters B and C should be equivalent to filter A. This is true and can be immediately proved by looking at (9), (13), and (5).

5. EQUIVALENT DIGITAL STRUCTURES FOR THE RESERVOIR MODEL

Analysis of the Meddis reservoir model allows us to plot its equivalent structures for realisation in the digital form (see Figure 6). The realisation with the help of filter A is more preferable since it is more computationally efficient.

Apart from the linear digital filters, the developed equivalent representations include an operation of multiplication of the signals in the time domain. It should be noted that, in general, multiplication of time-varying signals does not comply with the superposition principle, thus the reservoir model equivalent structure performs a nonlinear signal transformation. The signal $q(t) = M - s(t)$, which is multiplied by $k(t)$ is confined in the interval $[0, M]$ in accordance with the reservoir model definition. It consists mainly of low-frequency components of signal $k(t)q(t)$ in accordance with the properties of filter B.

Operation of the multiplication in the equivalent structure may be viewed as an automatic gain-control (AGC) operation. The gain $q(t)$ is a parameter, which slowly varies through time between M in the case of weak input signal and zero in the case of strong one.

Our equivalent structure of the Meddis reservoir model has similarities with that plotted in the works of Perdigao [13, 14].

6. LINEAR APPROXIMATION OF THE SIGNAL MULTIPLICATION OPERATION IN THE EQUIVALENT STRUCTURE OF THE RESERVOIR MODEL

It is possible to build a linear digital filter, which approximates the effect of the AGC mechanism for the case of small deviations of the system from the equilibrium state. Particular form of such filter is dependent on initial conditions, namely, the steady-state input signal value k_0.

A method we are going to use is thoroughly investigated in [15]. Similar methods of differential equation linearisation (which lead to the identical results) are widely known and used in the classical literature on theoretical mechanics.

Indeed, we assume that the system depicted in Figure 6, at a certain time instant, resides in the equilibrium. For such case, we may write

$$\begin{aligned} f_0 &= k_0 q_0, \\ q_0 &= M - s_0, \\ y(l+r)s_0 &= l f_0. \end{aligned} \tag{14}$$

Any deviations from the steady state are sufficiently small:

$$\begin{aligned} k(n) &= k_0 + \delta k(n), \\ f(n) &= f_0 + \delta f(n), \\ q(n) &= q_0 + \delta q(n), \\ s(n) &= s_0 + \delta s(n). \end{aligned} \tag{15}$$

Thus, for such system at an arbitrary time instant, we may write the following set of equations (see Figure 6):

$$\begin{gathered} f_0 + \delta f(n) = (k_0 + \delta k(n))(q_0 + \delta q(n)), \\ q_0 + \delta q(n) = M - (s_0 + \delta s(n)), \\ (a_0 + a_1 + a_2 + a_3)s_0 + a_0\delta s(n) + a_1\delta s(n-1) + a_2\delta s(n-2) \\ + a_3\delta s(n-3) = (b_1 + b_2 + b_3) f_0 + b_1\delta f(n-1) \\ + b_2\delta f(n-2) + b_3\delta f(n-3). \end{gathered} \tag{16}$$

Coefficients in the third equation of the set are those of filter B. Comparing sets (15) and (16), we may conclude that the following set of equations holds for deviations:

$$\begin{gathered} \delta f(n) = k_0\delta q(n) + q_0\delta k(n), \\ \delta q(n) = -\delta s(n), \\ a_0\delta s(n) + a_1\delta s(n-1) + a_2\delta s(n-2) + a_3\delta s(n-3) \\ = b_1\delta f(n-1) + b_2\delta f(n-2) + b_3\delta f(n-3). \end{gathered} \tag{17}$$

A solution of the equation set (17) with respect to variables $\delta k(n)$ and $\delta f(n)$ is represented as

$$\begin{gathered} q_0 = \frac{My(l+r)}{y(l+r) + lk_0}, \\ q_0(a_0\delta k(n) + a_1\delta k(n-1) + a_2\delta k(n-2) + a_3\delta k(n-3)) \\ = a_0\delta f(n) + (b_1k_0 + a_1)\delta f(n-1) \\ + (b_2k_0 + a_2)\delta f(n-2) + (b_3k_0 + a_3)\delta f(n-3). \end{gathered} \tag{18}$$

This equation represents a desired linear digital filter, which linearly approximates AGC of the equivalent structure. This filter is capable of transforming the signal $\delta k(t) = k(t) - k_0$ into $\delta f(t) = \delta(k(t)q(t)) = f(t) - f_0$ under the condition that these deviations are sufficiently small. The transfer function of this filter is expressed as

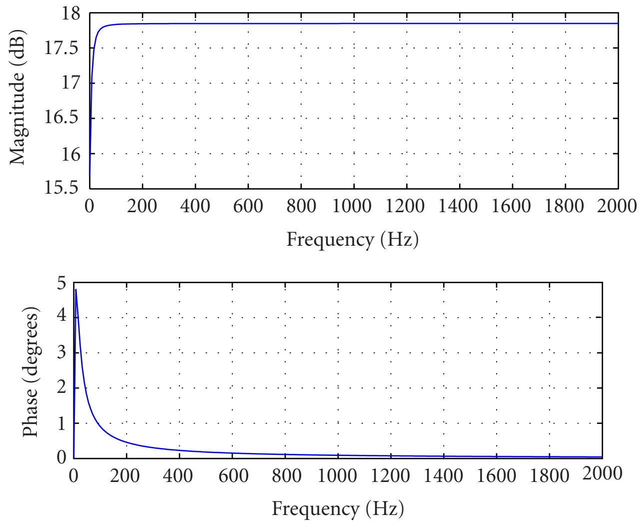

Figure 7: Frequency characteristic of filter D ($F_s = 16$ kHz, $k_0 = 10$).

$$\begin{aligned} &H_D(z, k_0) \\ &= \frac{My(l+r)}{(y(l+r) + lk_0)} \\ &\quad\cdot \frac{a_0 + a_1z^{-1} + a_2z^{-2} + a_3z^{-3}}{a_0 + (b_1k_0 + a_1)z^{-1} + (b_2k_0 + a_2)z^{-2} + (b_3k_0 + a_3)z^{-3}}. \end{aligned} \tag{19}$$

Note the explicit dependency of the form of this transfer function on the value of k_0. We will further denote this filter as "filter D."

The steady-state output $f_0(k_0)$ of the system is derived from the equilibrium set of (14) and is expressed as

$$f_0(k_0) = q_0k_0 = \frac{My(l+r)}{y(l+r) + lk_0}k_0. \tag{20}$$

Filter D is a highpass filter with quite sharp frequency response characteristic (see Figure 7) for a typical value of $k_0 = 10$.

In order to illustrate the dependence of the properties of filter D upon the value of k_0, Figure 8 depicts frequency characteristic of that filter with $k_0 = 1000$. As it can be seen from the comparison of Figures 7 and 8, apart from the change of the gain, the cut-off frequency of the filter is getting bigger with the increase of k_0.

From the digital filter theory it is known that the linear digital filter is "bounded-input bounded-output" (BIBO) stable if all of its poles lay inside the unit circle in z-plane. Filter D has three real poles. Analytical derivation of their values is rather complex in general. To perform such derivation, one could take advantage of Cardano formula for the roots of cubic equation.

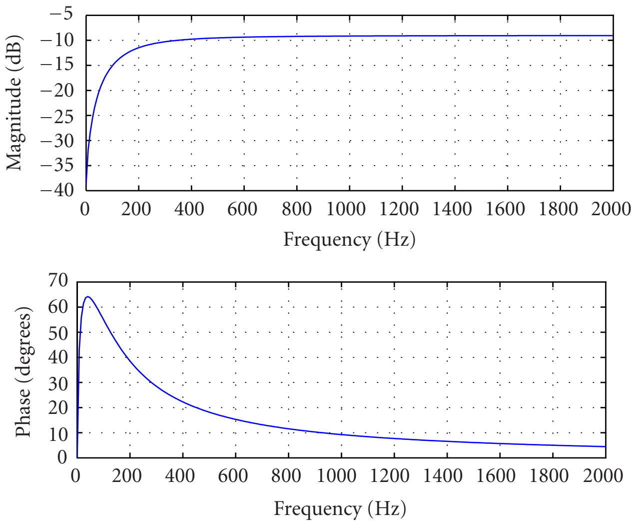

Figure 8: Frequency characteristic of filter D ($F_s = 16$ kHz, $k_0 = 1000$).

An alternative way is to estimate positions of the filter poles. Indeed, for realistic values of $k_0 \sim 10^1$–10^2 with quite good precision, filter D poles lay in the vicinity of its zeros. Zeros of filter D coincide with poles of filter B (11), and approximately we may put

$$p_{D1} \approx n_{D1} = 1 - \frac{l+r}{F_s}, \qquad p_{D2} \approx n_{D2} = 1 - \frac{x}{F_s}, \qquad p_{D3} \approx n_{D3} = 1 - \frac{y}{F_s}. \tag{21}$$

It must be noted also that if $k_0 \to 0$, then $p_{DN} \to n_{DN}$.

Pole p_{D1} first leaves the unit circle while sampling frequency is being decreased, indeed the realistic values of $l+r$ are significantly larger than the values of x and y. Consequently, approximation of the position of the first pole gives us a condition of filter D stability, while $k_0 \to 0$:

$$F_s > \frac{l+r}{2}. \tag{22}$$

Pole p_{D1} moves to the right on the real axis if the value of k_0 is being increased. This allows for filter D to become stable with increased k_0 even if it was unstable with the smaller values of k_0. This leads us to a conclusion that (22) represents sufficient condition for filter D to be stable with arbitrary realistic values of k_0.

In the work [8], it is required that the sampling frequency must be sufficiently large for a successful digital implementation of the reservoir model. Our finding of stability condition (22) puts a quantitative restriction on the sampling frequency for the linearised approximation.

Under the same assumption of small deviations from the equilibrium state, it is possible to construct an equivalent linear filter, which would serve as linear approximation of relation of the signals $\delta k(t)$ and $\delta c(t) = c(t) - c_0$, that is, the input and the output signals of the reservoir model measured relatively to their corresponding equilibrium values.

Such filter (further denoted as filter E) corresponds to the cascade of the filters D and A. Its frequency response characteristic is presented in Figure 9. Filter E transfer function is defined as

$$H_E(z, k_0) = \frac{1}{F_S} \cdot \frac{My(l+r)}{(y(l+r) + lk_0)} \cdot \frac{(1 - n_{D2}z^{-1})(1 - n_{D3}z^{-1})z^{-1}}{1 + ((b_1k_0 + a_1)/a_0)z^{-1} + ((b_2k_0 + a_2)/a_0)z^{-2} + ((b_3k_0 + a_3)/a_0)z^{-3}}. \tag{23}$$

It should be noted that the pole of filter A and the first zero of filter D are equal, thus they are removed from (23).

Response magnitude in the equilibrium state is derived from (2) and (20) and it looks like

$$c_0(k_0) = \frac{1}{(l+r)} f_0(k_0) = \frac{My}{y(l+r) + lk_0} k_0. \tag{24}$$

The notion that poles of filter E coincide with those of filter D leads to a conclusion that condition of the stability of the filter is identical to that of filter D.

7. PRACTICAL OUTCOME OF THE PRESENTED RESERVOIR MODEL ANALYSIS

As it can be seen from Figure 6, the reservoir model is equivalent to a kind of signal-dependent gain-control mechanism. The presented equivalent structure may be perceived as the interpretation of the IHC adaptation mechanism from the algorithmical signal processing point of view. In the equivalent structure, filters A, B, and C are all linear time-invariant structures, the only nonlinear element here is the multiplication of the signals. Implementation of the equivalent structure via a combination of filters A and B seems more preferable among the alternatives, presented in Figure 6, since it requires less computational effort.

A brief look at the poles of filter A (6) and B (11) gives an indication that their stability conditions are identical to that of filter D (22). This fact is a direct result of employment of forward difference approximation of the differential problem in the filter synthesis. All known digital implementations of the IHC reservoir model [16, 17, 18] share this method of differential approximation. However, this limitation seems impractical from the technological point of view, since it prevents implementation of the described equivalent structure, as well as other implementations mentioned above, for signals with sampling frequency below $\sim$ 4,6 kHz using the

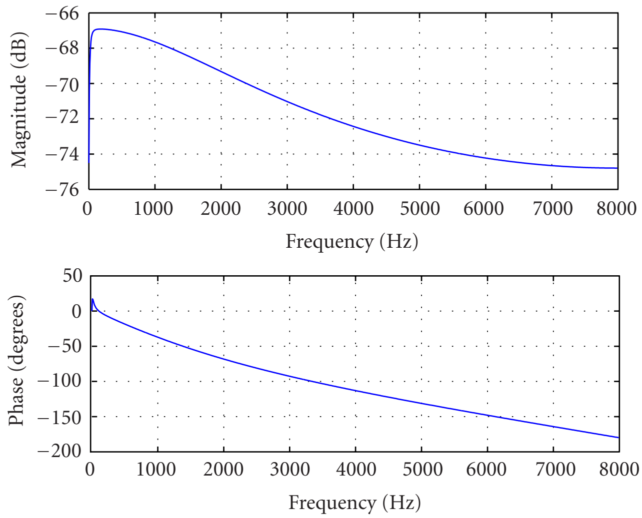

Figure 9: Frequency characteristic of filter E ($F_s = 16\,\text{kHz}$, $k_0 = 50$).

realistic values of the model parameters. Indeed, such limitation of the lowest possible sampling frequency makes efficient combination of the model with multirate cochlear filter banks impossible.

Fortunately, there exist other methods of approximation of the differential problem in the digital domain, for example, bilinear transformation. In accordance with its properties, any stable analog linear time-invariant filter, described by the corresponding differential equation, is converted into a stable digital filter. With the help of bilinear transformation, it is possible to construct universally stable digital filters A and B regardless of the sampling frequency. This procedure as well as its combination with computationally efficient implementation of the multirate cochlear filter bank is described in detail in [19].

However, in the case of bilinear transformation, unlike the situation with difference approximation, the coefficient b_0 of the filter B is not equal to zero:

$$
\begin{aligned}
H_B(z) &= \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + b_3 z^{-3}}{a_0 + a_1 z^{-1} + a_2 z^{-2} + a_3 z^{-3}},\\
a_0 &= 8F_s^3 + 4F_s^2(x+y+l+r)\\
&\quad + 2F_s\big((x+y)(l+r)+xy\big) + xy(l+r),\\
a_1 &= -24F_s^3 - 4F_s^2(x+y+l+r)\\
&\quad + 2F_s\big((x+y)(l+r)+xy\big) + 3xy(l+r),\\
a_2 &= 24F_s^3 - 4F_s^2(x+y+l+r)\\
&\quad - 2F_s\big((x+y)(l+r)+xy\big) + 3xy(l+r),\\
a_3 &= -8F_s^3 + 4F_s^2(x+y+l+r)\\
&\quad - 2F_s\big((x+y)(l+r)+xy\big) + xy(l+r),\\
b_0 &= 4F_s^2 + 2F_s(x+l+r) + xl,\\
b_1 &= -4F_s^2 + 2F_s(x+l+r) + 3xl,\\
b_2 &= -4F_s^2 - 2F_s(x+l+r) + 3xl,\\
b_3 &= 4F_s^2 - 2F_s(x+l+r) + xl.
\end{aligned}
\tag{25}
$$

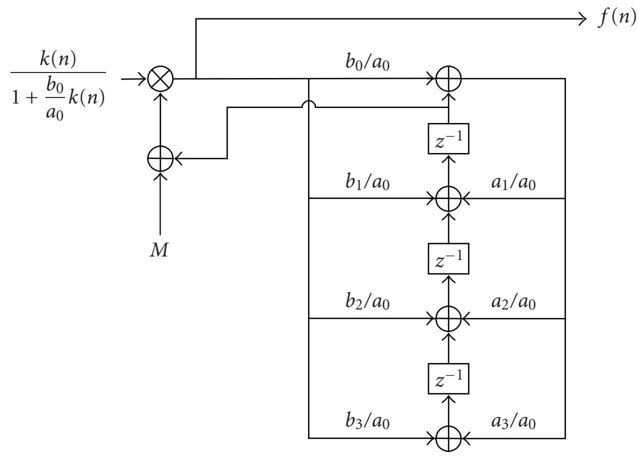

Figure 10: Transposed direct form II realization.

This fact leads to the additional operations at the implementation of the signal flow of Figure 6. Indeed, writing a set of equations describing the signal flow over the feedback loop of Figure 6 results in the following relations:

$$
\begin{aligned}
f(n) &= k(n)q(n) = k(n)\big(M - s(n)\big),\\
\sum_{i=0}^{3} b_i f(n-i) &= \sum_{i=0}^{3} a_i s(n-i).
\end{aligned}
\tag{26}
$$

It is evident that simple substitution of the second equation into the first does not lead to the expression of the output signal $f(n)$ through the current value of the input signal $k(n)$ and previous values of the signals f and s. The current value of the output is present on the both sides of the equation. Separation of the variables leads to the following expression for the output signal:

$$
f(n) = \frac{k(n)}{1 + (b_0/a_0)k(n)} \cdot \left(M - \sum_{i=1}^{3}\frac{b_i}{a_0} f(n-i) + \sum_{i=1}^{3}\frac{a_i}{a_0} s(n-i)\right).
\tag{27}
$$

It appears that the most computationally effective way to implement filter B with its signal feedback is a transposed direct form II structure (Figure 10). This realisation minimises the number of delay units.

For the sake of completeness of the picture, the following formula presents a version of the digital filter A, which is obtained with the help of bilinear transformation:

$$
H_A(z) = \frac{1}{l + r + 2F_s} \cdot \frac{1 + z^{-1}}{1 + \big((l + r - 2F_s)/(l + r + 2F_s)\big)z^{-1}}.
\tag{28}
$$

The formulae (25), (27), and (28), as well as the Figures 6 and 10, contain exact instructions for the implementation of the reservoir IHC model, which remains stable at any sampling frequency. As it was noted above, this property saves computational load and is desirable for efficient incorporation of the model into multirate cochlear filter bank.

Linear approximation (23) of the reservoir model might be viewed as a computationally effective way to implement the model when input signal does not significantly deviate from a certain fixed stationary value. It might also serve as the linear time-variant filter, which simulates the reservoir model, when the slowly varying stationary value of the signal k_0 is known in advance or is estimated through a long-term moving average procedure.

This linear approximation is also important because of its link to the RASTA filtering technique [20, 21], a well-established channel normalisation and speech augmentation means in ASR. Although the nature of this link needs further investigation, both techniques represent low-passband filters, running in separate frequency channels, which are converted with the help of nonlinearity. In the case of RASTA, each frequency channel is decimated to represent one frequency bin of the short time Fourier transform spectrogram and converted into modulation-frequency domain by Jah-log transformation [16]. In the case of reservoir model there is no explicit decimation and the passband signal is transformed by “BM vibration—membrane permeability” transformation [6], which somewhat resembles Jah-log transform.

8. EXPERIMENTS

Several experiments were run in order to validate the original assumption that the anthropomorphic auditory modelling in general and IHC adaptation model in particular may indeed augment performance of the ASR systems. A comparison involved three experimental setups, which are described in more detailed fashion in [22].

(i) BASELINE: an ASR feature extraction (FE) algorithm, which is based on linear time-invariant perceptually aligned filters.

(ii) A-MORPHIC: anthropomorphic feature extraction algorithm [22], which combined linear time-variant cochlear filters to model auditory suppression and the above-described IHC reservoir model implementation. However, results mainly reflect effect of the IHC reservoir model since speech recordings in the experiment had approximately the same loudness level (∼ 40 dB SPL).

(iii) RASTA: the conventional RASTA algorithm-based feature extraction [16].

In order to be effective, ASR FE algorithms should convey as much information about the speech source as possible. The measure of the amount of conveyed information, that is, the mutual information between a speech source S, which at any instant of time resides in one of the possible states C_i, $i = 1, 2, \ldots, N$, and a measured feature vector component X is defined as follows:

$$
\begin{aligned}
I(S,X) &= H(S) - H(S \mid X) \\
&= -\sum_{\forall C_i \in \{c\}} P(C_i)\log_2 P(C_i) \\
&\quad + \sum_{\forall C_i \in \{c\}} \int_{G(X)} P(C_i, X)\log_2 P(C_i \mid X)\, dX.
\end{aligned}
\tag{29}
$$

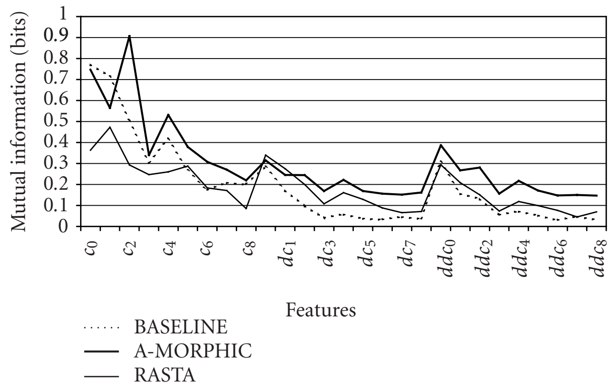

FIGURE 11: Mutual information of feature components ($_\Delta X = 0.01$).

Estimation of the mutual information has been performed with the help of the following procedure [22]:

$$
\begin{aligned}
I_{\Delta X}(S,X) \approx \log_2 N + \frac{1}{N}\Bigg(&\sum_{\forall i}\sum_{\forall j} N(C_i, {}_\Delta X_j)\log_2 N(C_i, {}_\Delta X_j) \\
&- \sum_{\forall i} N(C_i)\log_2 N(C_i) \\
&- \sum_{\forall j} N({}_\Delta X_j)\log_2 N({}_\Delta X_j)\Bigg).
\end{aligned}
\tag{30}
$$

Here N denotes a total number of feature frames in the measurement; $N({}_\Delta X_j)$—a number of frames when the feature value falls into the interval $[\min(X) + (j - 1)_\Delta X, \min(X) + j_\Delta X]$; $N(C_i)$—a number of frames which were generated in the state C_i; $N(C_i, {}_\Delta X_j)$—a number of frames, belonging to the certain feature interval, which were generated by the source in the state C_i.

Phonetically labelled TIMIT speech corpus was used in this experiment. Probability distributions were approximated with histograms that had a step size $_\Delta X = 0.01$. Results, which are presented in Figure 11, show that A-MORPHIC features are generally the most informative.

Another experiment was performed to estimate a degree of invariance of the feature vectors to different kinds of adverse interference. To provide estimates of the feature invariance degree a simple Euclidian distance between feature vectors was used. Exact experiment description may be found in [22]. Results of the experiment, which are presented in Table 1, reflect a mean distance of the feature vectors in adverse conditions to those perceived in a “clean” environment. As it can be seen from the table, A-MORPHIC features are less invariant to the adverse interference than RASTA. Anyway, a distance between “clean” and severely noisy (SNR 0 dB) features in the case of A-MORPHIC FE matches that between “clean” and mildly-noisy (SNR 30 dB) features in the BASELINE case.

Results of the depicted experiments are also supported by the reported in [22] comparison of the speech recogniser

Table 1: Expected mean distance between the feature vectors in adverse conditions and clean environment.

Feature extraction algorithm	Interference			
	Noise 30 dB	Noise 10 dB	Noise 0 dB	Convol. channel
BASELINE	0.41597	0.78894	1.05047	0.49298
RASTA FE	0.09842	0.17563	0.22338	0.05300
A-MORPHIC	0.26853	0.44951	0.42615	0.16665

performances (refer to [23] for a description of the recogniser). It's main result is that in adverse environments the recogniser with A-MORPHIC FE performs at least as good as the one with RASTA FE. These facts support the conjecture of the present paper that application of the anthropomorphic algorithms in technical devices, namely, ASR engines, is fruitful.

9. CONCLUSIONS

Analysis of the physiological model of the chemical IHC-AN synapse creates an opportunity to implement it in the form of the anthropomorphic algorithm, which is computationally efficient and thus may be used in technical devices. The equivalent digital and linearised equivalent representations create alternatives for a traditional direct difference approximation of the original set of differential equations. These representations allow for a multiple "accuracy versus computational load" tradeoffs at the implementation stage. Within the described framework, it is possible to create implementations, which remain stable regardless to the signal sampling frequency.

It was found that effect of the IHC adaptation model is equivalent to the action of signal-dependent automatic gain control mechanism. It is also conjectured that effect of the linearised equivalent representation resembles that of RASTA, an algorithm engineered with the aim of alleviating the influence of additive and convolutive noises. This interpretation of the IHC-AN synapse model gives us reasons to believe that it is important as a mean of increasing ASR robustness to the real-world environments (e.g., "too slow" and "too fast" varying additive and convolutive noises) and also as a mean of enhancement of the useful signal in the speech coding applications. Presented and referenced experiments confirm viability of the application of the discussed anthropomorphic algorithm to the ASR field. However, the exact form of the relation between the IHC-AN synapse model and RASTA should be investigated further.

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Search and Retrieval of 3D Content and Associated Knowledge Extraction and Propagation

Call for Papers

With the general availability of 3D digitizers, scanners, and the technology innovation in 3D graphics and computational equipment, large collections of 3D graphical models can be readily built up for different applications (e.g., in CAD/CAM, games design, computer animations, manufacturing and molecular biology). For such large databases, the method whereby 3D models are sought merits careful consideration. The simple and efficient query-by-content approach has, up to now, been almost universally adopted in the literature. Any such method, however, must first deal with the proper positioning of the 3D models. The two prevalent-in-the-literature methods for the solution to this problem seek either

- Pose Normalization: Models are first placed into a canonical coordinate frame (normalizing for translation, scaling, and rotation). Then, the best measure of similarity is found by comparing the extracted feature vectors, or
- Descriptor Invariance: Models are described in a transformation invariant manner, so that any transformation of a model will be described in the same way, and the best measure of similarity is obtained at any transformation.

The existing 3D retrieval systems allow the user to perform queries by example. The queried 3D model is then processed, low-level geometrical features are extracted, and similar objects are retrieved from a local database. A shortcoming of the methods that have been proposed so far regarding the 3D object retrieval, is that neither is the semantic information (high-level features) attached to the (low-level) geometric features of the 3D content, nor are the personalization options taken into account, which would significantly improve the retrieved results. Moreover, few systems exist so far to take into account *annotation* and *relevance feedback* techniques, which are very popular among the corresponding content-based image retrieval systems (CBIR).

Most existing CBIR systems using knowledge either annotate all the objects in the database (full annotation) or annotate a subset of the database manually selected (partial annotation). As the database becomes larger, full annotation is increasingly difficult because of the manual effort needed. Partial annotation is relatively affordable and trims down the heavy manual labor. Once the database is partially annotated, traditional image analysis methods are used to derive semantics of the objects not yet annotated. However, it is not clear "how much" annotation is sufficient for a specific database and what the best subset of objects to annotate is. In other words how the knowledge *will be propagated*. Such techniques have not been presented so far regarding the 3D case.

Relevance feedback was first proposed as an interactive tool in text-based retrieval. Since then it has been proven to be a powerful tool and has become a major focus of research in the area of content-based search and retrieval. In the traditional computer centric approaches, which have been proposed so far, the "best" representations and weights are fixed and they cannot effectively model high-level concepts and user's perception subjectivity. In order to overcome these limitations of the computer centric approach, techniques based on *relevant feedback*, in which the human and computer interact to refine high-level queries to representations based on low-level features, should be developed.

The aim of this special issue is to focus on recent developments in this expanding research area. The special issue will focus on novel approaches in 3D object retrieval, transforms and methods for efficient geometric feature extraction, annotation and relevance feedback techniques, knowledge propagation (e.g., using Bayesian networks), and their combinations so as to produce a single, powerful, and dominant solution.

Topics of interest include (but are not limited to):

- 3D content-based search and retrieval methods (volume/surface-based)
- Partial matching of 3D objects
- Rotation invariant feature extraction methods for 3D objects

- Graph-based and topology-based methods
- 3D data and knowledge representation
- Semantic and knowledge propagation over heterogeneous metadata types
- Annotation and relevance feedback techniques for 3D objects

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Special Issue on Robust Speech Recognition

Call for Papers

Robustness can be defined as the ability of a system to maintain performance or degrade gracefully when exposed to conditions not well represented in the data used to develop the system. In automatic speech recognition (ASR), systems must be robust to many forms of signal degradation, including speaker characteristics (e.g., dialect and accent), ambient environment (e.g., cellular telephony), transmission channel (e.g., voice over IP), and language (e.g., new words, dialect switching). Robust ASR systems, which have been under development for the past 35 years, have made great progress over the years closing the gap between performance on pristine research tasks and noisy operational data.

However, in recent years, demand is emerging for a new class of systems that tolerate extreme and unpredictable variations in operating conditions. For example, in a cellular telephony environment, there are many nonstationary forms of noise (e.g., multiple speakers) and significant variations in microphone type, position, and placement. Harsh ambient conditions typical in automotive and mobile applications pose similar challenges. Development of systems in a language or dialect for which there is limited or no training data in a target language has become a critical issue for a new generation of voice mining applications. The existence of multiple conditions in a single stream, a situation common to broadcast news applications, and that often involves unpredictable changes in speaker, topic, dialect, or language, is another form of robustness that has gained attention in recent years.

Statistical methods have dominated the field since the early 1980s. Such systems tend to excel at learning the characteristics of large databases that represent good models of the operational conditions and do not generalize well to new environments.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Channel and microphone normalization
- Stationary and nonstationary noise modeling, compensation, and/or rejection
- Localization and separation of sound sources (including speaker segregation)
- Signal processing and feature extraction for applications involving hands-free microphones
- Noise robust speech modeling
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Signal Processing Technologies for Ambient Intelligence in Home-Care Applications

Call for Papers

The possibility of allowing elderly people with different kinds of disabilities to conduct a normal life at home and achieve a more effective inclusion in the society is attracting more and more interest from both industrial and governmental bodies (hospitals, healthcare institutions, and social institutions). Ambient intelligence technologies, supported by adequate networks of sensors and actuators, as well as by suitable processing and communication technologies, could enable such an ambitious objective.

Recent researches demonstrated the possibility of providing constant monitoring of environmental and biomedical parameters, and the possibility to autonomously originate alarms, provide primary healthcare services, activate emergency calls, and rescue operations through distributed assistance infrastructures. Nevertheless, several technological challenges are still connected with these applications, ranging from the development of enabling technologies (hardware and software), to the standardization of interfaces, the development of intuitive and ergonomic human-machine interfaces, and the integration of complex systems in a highly multidisciplinary environment.

The objective of this special issue is to collect the most significant contributions and visions coming from both academic and applied research bodies working in this stimulating research field. This is a highly interdisciplinary field comprising many areas, such as signal processing, image processing, computer vision, sensor fusion, machine learning, pattern recognition, biomedical signal processing, multimedia, human-computer interfaces, and networking.

The focus will be primarily on the presentation of original and unpublished works dealing with ambient intelligence and domotic technologies that can enable the provision of advanced homecare services.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Video-based monitoring of domestic environments and users
- Continuous versus event-driven monitoring
- Distributed information processing
- Data fusion techniques for event association and automatic alarm generation
- Modeling, detection, and learning of user habits for automatic detection of anomalous behaviors
- Integration of biomedical and behavioral data
- Posture and gait recognition and classification
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Special Issue on Spatial Sound and Virtual Acoustics

Call for Papers

Spatial sound reproduction has become widespread in the form of multichannel audio, particularly through home theater systems. Reproduction systems from binaural (by headphones) to hundreds of loudspeaker channels (such as wave field synthesis) are entering practical use. The application potential of spatial sound is much wider than multichannel sound, however, and research in the field is active. Spatial sound covers for example the capturing, analysis, coding, synthesis, reproduction, and perception of spatial aspects in audio and acoustics.

In addition to the topics mentioned above, research in virtual acoustics broadens the field. Virtual acoustics includes techniques and methods to create realistic percepts of sound sources and acoustic environments that do not exist naturally but are rendered by advanced reproduction systems using loudspeakers or headphones. Augmented acoustic and audio environments contain both real and virtual acoustic components.

Spatial sound and virtual acoustics are among the major research and application areas in audio signal processing. Topics of active study range from new basic research ideas to improvement of existing applications. Understanding of spatial sound perception by humans is also an important area, in fact a prerequisite to advanced forms of spatial sound and virtual acoustics technology.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Multichannel reproduction
- Wave field synthesis
- Binaural reproduction
- Format conversion and enhancement of spatial sound
- Spatial sound recording
- Analysis, synthesis, and coding of spatial sound
- Spatial sound perception and auditory modeling
- Simulation and modeling of room acoustics
- Auralization techniques
- Beamforming and sound source localization
- Acoustic and auditory scene analysis
- Augmented reality audio
- Virtual acoustics (sound environments and sources)
- Intelligent audio environments
- Loudspeaker-room interaction and equalization
- Applications

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Special Issue on

Advances in Electrocardiogram Signal Processing and Analysis

Call for Papers

Since its invention in the 19th century when it was little more than a scientific curiosity, the electrocardiogram (ECG) has developed into one of the most important and widely used quantitative diagnostic tools in medicine. It is essential for the identification of disorders of the cardiac rhythm, extremely useful for the diagnosis and management of heart abnormalities such as myocardial infarction (heart attack), and offers helpful clues to the presence of generalised disorders that affect the rest of the body, such as electrolyte disturbances and drug intoxication.

Recording and analysis of the ECG now involves a considerable amount of signal processing for S/N enhancement, beat detection, automated classification, and compression. These involve a whole variety of innovative signal processing methods, including adaptive techniques, time-frequency and time-scale procedures, artificial neural networks and fuzzy logic, higher-order statistics and nonlinear schemes, fractals, hierarchical trees, Bayesian approaches, and parametric models, amongst others.

This special issue will review the current status of ECG signal processing and analysis, with particular regard to recent innovations. It will report major achievements of academic and commercial research institutions and individuals, and provide an insight into future developments within this exciting and challenging area.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Beat (QRS complex) detection
- ECG compression
- Denoising of ECG signals
- Morphological studies and classification
- ECG modeling techniques
- Expert systems and automated diagnosis
- QT interval measurement and heart-rate variability
- Arrhythmia and ischemia detection and analysis
- Interaction between cardiovascular signals (ECG, blood pressure, respiration, etc.)
- Intracardiac ECG analysis (implantable cardiovascular devices, and pacemakers)
- ECGs and sleep apnoea
- Real-time processing and instrumentation
- ECG telemedicine and e-medicine
- Fetal ECG detection and analysis
- Computational tools and databases for ECG education and research

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Special Issue on

Emerging Signal Processing Techniques for Power Quality Applications

Call for Papers

Recently, end users and utility companies are increasingly concerned with perturbations originated from electrical power quality variations. Investigations are being carried out to completely characterize not only the old traditional type of problems, but also new ones that have arisen as a result of massive use of nonlinear loads and electronics-based equipment in residences, commercial centers, and industrial plants. These nonlinear load effects are aggravated by massive power system interconnections, increasing number of different power sources, and climatic changes.

In order to improve the capability of equipments applied to monitoring the power quality of transmission and distribution power lines, power systems have been facing new analysis and synthesis paradigms, mostly supported by signal processing techniques. The analysis and synthesis of emerging power quality and power system problems led to new research frontiers for the signal processing community, focused on the development and combination of computational intelligence, source coding, pattern recognition, multirate systems, statistical estimation, adaptive signal processing, and other digital processing techniques, implemented in either DSP-based, PC-based, or FPGA-based solutions.

The goal of this proposal is to introduce powerful and efficient real-time or almost-real-time signal processing tools for dealing with the emerging power quality problems. These techniques take into account power-line signals and complementary information, such as climatic changes.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Detection of transients
- Classification of multiple events
- Identification of isolated and multiple disturbance sources
- Compression of voltage and current data signals
- Location of disturbance sources
- Prediction of transmission and distribution systems failures
- Demand forecasting
- Parameters estimation for fundamental, harmonics, and interharmonics

Digital signal processing techniques applied to power quality applications are a very attractive and stimulating area of research. Its results will provide, in the near future, new standards for the decentralized and real-time monitoring of transmission and distribution systems, allowing to closely follow and predict power system performance. As a result, the power systems will be more easily planned, expanded, controlled, managed, and supervised.

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Special Issue on
Super-resolution Enhancement of Digital Video

Call for Papers

When designing a system for image acquisition, there is generally a desire for high spatial resolution and a wide field-of-view. To achieve this, a camera system must typically employ small f-number optics. This produces an image with very high spatial-frequency bandwidth at the focal plane. To avoid aliasing caused by undersampling, the corresponding focal plane array (FPA) must be sufficiently dense. However, cost and fabrication complexities may make this impractical. More fundamentally, smaller detectors capture fewer photons, which can lead to potentially severe noise levels in the acquired imagery. Considering these factors, one may choose to accept a certain level of undersampling or to sacrifice some optical resolution and/or field-of-view.

In image super-resolution (SR), postprocessing is used to obtain images with resolutions that go beyond the conventional limits of the uncompensated imaging system. In some systems, the primary limiting factor is the optical resolution of the image in the focal plane as defined by the cut-off frequency of the optics. We use the term "optical SR" to refer to SR methods that aim to create an image with valid spatial-frequency content that goes beyond the cut-off frequency of the optics. Such techniques typically must rely on extensive a priori information. In other image acquisition systems, the limiting factor may be the density of the FPA, subsequent postprocessing requirements, or transmission bitrate constraints that require data compression. We refer to the process of overcoming the limitations of the FPA in order to obtain the full resolution afforded by the selected optics as "detector SR." Note that some methods may seek to perform both optical and detector SR.

Detector SR algorithms generally process a set of low-resolution aliased frames from a video sequence to produce a high-resolution frame. When subpixel relative motion is present between the objects in the scene and the detector array, a unique set of scene samples are acquired for each frame. This provides the mechanism for effectively increasing the spatial sampling rate of the imaging system without reducing the physical size of the detectors.

With increasing interest in surveillance and the proliferation of digital imaging and video, SR has become a rapidly growing field. Recent advances in SR include innovative algorithms, generalized methods, real-time implementations, and novel applications. The purpose of this special issue is to present leading research and development in the area of super-resolution for digital video. Topics of interest for this special issue include but are not limited to:

- Detector and optical SR algorithms for video
- Real-time or near-real-time SR implementations
- Innovative color SR processing
- Novel SR applications such as improved object detection, recognition, and tracking
- Super-resolution from compressed video
- Subpixel image registration and optical flow

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Advanced Signal Processing and Computational Intelligence Techniques for Power Line Communications

Call for Papers

In recent years, increased demand for fast Internet access and new multimedia services, the development of new and feasible signal processing techniques associated with faster and low-cost digital signal processors, as well as the deregulation of the telecommunications market have placed major emphasis on the value of investigating hostile media, such as powerline (PL) channels for high-rate data transmissions.

Nowadays, some companies are offering powerline communications (PLC) modems with mean and peak bit-rates around 100 Mbps and 200 Mbps, respectively. However, advanced broadband powerline communications (BPLC) modems will surpass this performance. For accomplishing it, some special schemes or solutions for coping with the following issues should be addressed: (i) considerable differences between powerline network topologies; (ii) hostile properties of PL channels, such as attenuation proportional to high frequencies and long distances, high-power impulse noise occurrences, time-varying behavior, and strong inter-symbol interference (ISI) effects; (iv) electromagnetic compatibility with other well-established communication systems working in the same spectrum, (v) climatic conditions in different parts of the world; (vii) reliability and QoS guarantee for video and voice transmissions; and (vi) different demands and needs from developed, developing, and poor countries.

These issues can lead to exciting research frontiers with very promising results if signal processing, digital communication, and computational intelligence techniques are effectively and efficiently combined.

The goal of this special issue is to introduce signal processing, digital communication, and computational intelligence tools either individually or in combined form for advancing reliable and powerful future generations of powerline communication solutions that can be suited with for applications in developed, developing, and poor countries.

Topics of interest include (but are not limited to)

- Multicarrier, spread spectrum, and single carrier techniques
- Channel modeling
- Channel coding and equalization techniques
- Multiuser detection and multiple access techniques
- Synchronization techniques
- Impulse noise cancellation techniques
- FPGA, ASIC, and DSP implementation issues of PLC modems
- Error resilience, error concealment, and Joint source-channel design methods for video transmission through PL channels

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Special Issue on
Video Adaptation for Heterogeneous Environments

Call for Papers

The explosive growth of compressed video streams and repositories accessible worldwide, the recent addition of new video-related standards such as H.264/AVC, MPEG-7, and MPEG-21, and the ever-increasing prevalence of heterogeneous, video-enabled terminals such as computer, TV, mobile phones, and personal digital assistants have escalated the need for efficient and effective techniques for adapting compressed videos to better suit the different capabilities, constraints, and requirements of various transmission networks, applications, and end users. For instance, Universal Multimedia Access (UMA) advocates the provision and adaptation of the same multimedia content for different networks, terminals, and user preferences.

Video adaptation is an emerging field that offers a rich body of knowledge and techniques for handling the huge variation of resource constraints (e.g., bandwidth, display capability, processing speed, and power consumption) and the large diversity of user tasks in pervasive media applications. Considerable amounts of research and development activities in industry and academia have been devoted to answering the many challenges in making better use of video content across systems and applications of various kinds.

Video adaptation may apply to individual or multiple video streams and may call for different means depending on the objectives and requirements of adaptation. Transcoding, transmoding (cross-modality transcoding), scalable content representation, content abstraction and summarization are popular means for video adaptation. In addition, video content analysis and understanding, including low-level feature analysis and high-level semantics understanding, play an important role in video adaptation as essential video content can be better preserved.

The aim of this special issue is to present state-of-the-art developments in this flourishing and important research field. Contributions in theoretical study, architecture design, performance analysis, complexity reduction, and real-world applications are all welcome.

Topics of interest include (but are not limited to):

- Heterogeneous video transcoding
- Scalable video coding
- Dynamic bitstream switching for video adaptation
- Signal, structural, and semantic-level video adaptation
- Content analysis and understanding for video adaptation
- Video summarization and abstraction
- Copyright protection for video adaptation
- Crossmedia techniques for video adaptation
- Testing, field trials, and applications of video adaptation services
- International standard activities for video adaptation

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Transforming Signal Processing Applications into Parallel Implementations

Call for Papers

There is an increasing need to develop efficient "system-level" models, methods, and tools to support designers to quickly transform signal processing application specification to heterogeneous hardware and software architectures such as arrays of DSPs, heterogeneous platforms involving microprocessors, DSPs and FPGAs, and other evolving multiprocessor SoC architectures. Typically, the design process involves aspects of application and architecture modeling as well as transformations to translate the application models to architecture models for subsequent performance analysis and design space exploration. Accurate predictions are indispensable because next generation signal processing applications, for example, audio, video, and array signal processing impose high throughput, real-time and energy constraints that can no longer be served by a single DSP.

There are a number of key issues in transforming application models into parallel implementations that are not addressed in current approaches. These are engineering the application specification, transforming application specification, or representation of the architecture specification as well as communication models such as data transfer and synchronization primitives in both models.

The purpose of this call for papers is to address approaches that include application transformations in the performance, analysis, and design space exploration efforts when taking signal processing applications to concurrent and parallel implementations. The Guest Editors are soliciting contributions in joint application and architecture space exploration that outperform the current architecture-only design space exploration methods and tools.

Topics of interest for this special issue include but are not limited to:

- modeling applications in terms of (abstract) control-dataflow graph, dataflow graph, and process network models of computation (MoC)
- transforming application models or algorithmic engineering
- transforming application MoCs to architecture MoCs
- joint application and architecture space exploration
- joint application and architecture performance analysis
- extending the concept of algorithmic engineering to architecture engineering
- design cases and applications mapped on multiprocessor, homogeneous, or heterogeneous SOCs, showing joint optimization of application and architecture

Authors should follow the EURASIP JASP manuscript format described at http://www.hindawi.com/journals/asp/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JASP manuscript tracking system at http://www.mstracking.com/asp/, according to the following timetable:

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Institute of Acoustics

NEWS RELEASE

Nominations Invited for the Institute of Acoustics 2006 A B Wood Medal

The Institute of Acoustics, the UK's leading professional body for those working in acoustics, noise and vibration, is inviting nominations for its prestigious A B Wood Medal for the year 2006.

The A B Wood Medal and prize is presented to an individual, usually under the age of 35, for distinguished contributions to the application of underwater acoustics. The award is made annually, in even numbered years to a person from Europe and in odd numbered years to someone from the USA/Canada. The 2005 Medal was awarded to Dr A Thode from the USA for his innovative, interdisciplinary research in ocean and marine mammal acoustics.

Nominations should consist of the candidate's CV, clearly identifying peer reviewed publications, and a letter of endorsement from the nominator identifying the contribution the candidate has made to underwater acoustics. In addition, there should be a further reference from a person involved in underwater acoustics and not closely associated with the candidate. Nominees should be citizens of a European Union country for the 2006 Medal. Nominations should be marked confidential and addressed to the President of the Institute of Acoustics at 77A St Peter's Street, St. Albans, Herts, AL1 3BN. The deadline for receipt of nominations is **15 October 2005**.

Dr Tony Jones, President of the Institute of Acoustics, comments, "A B Wood was a modest man who took delight in helping his younger colleagues. It is therefore appropriate that this prestigious award should be designed to recognise the contributions of young acousticians."

Further information and an nomination form can be found on the Institute's website at www.ioa.org.uk.

A B Wood

Albert Beaumont Wood was born in Yorkshire in 1890 and graduated from Manchester University in 1912. He became one of the first two research scientists at the Admiralty to work on antisubmarine defence. He designed the first directional hydrophone and was well known for the many contributions he made to the science of underwater acoustics and for the help he gave to younger colleagues. The medal was instituted after his death by his many friends on both sides of the Atlantic and was administered by the Institute of Physics until the formation of the Institute of Acoustics in 1974.

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EDITORS NOTES

The Institute of Acoustics is the UK's professional body for those working in acoustics, noise and vibration. It was formed in 1974 from the amalgamation of the Acoustics Group of the Institute of Physics and the British Acoustical Society (a daughter society of the Institution of Mechanical Engineers). The Institute of Acoustics is a nominated body of the Engineering Council, offering registration at Chartered and Incorporated Engineer levels.

The Institute has some 2500 members from a rich diversity of backgrounds, with engineers, scientists, educators, lawyers, occupational hygienists, architects and environmental health officers among their number. This multidisciplinary culture provides a productive environment for cross-fertilisation of ideas and initiatives. The range of interests of members within the world of acoustics is equally wide, embracing such aspects as aerodynamics, architectural acoustics, building acoustics, electroacoustics, engineering dynamics, noise and vibration, hearing, speech, underwater acoustics, together with a variety of environmental aspects. The lively nature of the Institute is demonstrated by the breadth of its learned society programmes.

For more information please visit our site at www.ioa.org.uk.

CALL FOR PAPERS

8th ACM Multimedia Security Workshop

September 26–27, 2006

ACM Multimedia and Security Workshop
September 26–27, 2006

Organizing Committee

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M. Kivanc Mihcak
Pierre Moulin
Fernando Perez-Gonzalez
Thierry Pun
Claus Vielhauer
Min Wu

Time Schedule
Submissions start: February 1
Submissions end: April 20
Authors notified: July 20
Camera ready by: August 10
Workshop: September 26-27, 2006

The objective of this 8th ACM Multimedia and Security Workshop is to identify the key issues to be addressed by future research in the areas of media manipulation recognition, media data authentication, and detection of hidden communication channels. We expect the workshop to help motivate this research and to establish fruitful relationships with the key actors from academia, industry, and government in the US and European and Asian countries. It will consist of invited papers, full papers, short papers, and possibly a rump or a panel session. This event continues a successful series of workshops started in 1998.

OBJECTIVES

• Discussion of emerging technologies in digital multi-media authentication, identification, fingerprinting, and steganalysis
• Identification of key research problems with the biggest impact on specified deficiencies in the field of secure multimedia distribution
• Formulation of target applications of identified technologies (in both the commercial and military sectors)
• Discussion of legal issues connected to multimedia security, digital watermarking, and steganography

SCOPE AND PAPERS

Papers should address theoretical and practical issues of multimedia watermarking and steganography. We will consider both theoretical papers dealing with fundamental issues and application oriented contributions (software and hardware demos are highly encouraged). Topics include but are not limited to:
• Robust watermarking of multimedia
• Authentication of multimedia
• Informational-theoretical aspects of data hiding
• Steganography and steganalysis
• Forensic analysis of digital multimedia
• Practical systems with aspects of data hiding
• Watermarking quality evaluation and benchmarks
• New applications, security issues, and legal aspects
• Data hiding applications in biometrics: document security and person authentication
In particular, the call for papers includes request for full papers with high degree of innovations as well as short papers with interesting improvements of existing art or new ideas and research directions. Full papers should be 6–12 pages long, short papers 4–6 pages long (ACM format http://www.acm.org/sigmm/). Accepted papers will be published in ACM workshop proceedings.

Submission Deadline April 20, 2006

Authors are invited to submit, by email, full papers or short papers by indicating the type of the paper (full/short) in electronic format (PDF or PostScript) to https://msrcmt.research.microsoft.com/ACM2006. Create a new user account, login, and follow the submission instructions.

4th International Workshop on Adaptive Multimedia Retrieval
July 27-28, 2006 - University of Geneva, Switzerland

http://viper.unige.ch/amr2006

The workshop focuses especially on researchers that are working on feature extraction techniques for multimedia, computer linguistic approaches, (dynamic) data analysis methods, and visualization methods as well as user interface design. Therefore, contributions to the workshop should focus on, but are not limited to:

- Multimedia retrieval systems (for text, image, audio, video and mixed-media)
- Theoretical foundations of multimedia retrieval and mining
- Intelligent multimedia data modelling, indexing and structure extraction
- Adaptive Hypermedia and web based systems
- Metadata for multimedia retrieval
- Multimedia and multi-modal mining
- Semantic content analysis for multimedia
- Semantic web and ontologies
- Adaptive query languages
- Similarity measures (especially user adaptive measures)
- User and preference modelling (including feedback models)
- Methods for adaptive data visualization and user interfaces

General Chair:
- **Stéphane Marchand-Maillet**
  University of Geneva

Program Chairs:
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  University of Geneva
- **Andreas Nürnberger**
  University of Magdeburg
- **Marcin Detyniecki**
  LIP6, CNRS Paris

Local Chair:
- **Nicolas Moënne-Loccoz**
  University of Geneva

Deadline for paper submission:	**March 17, 2006**
Notification of acceptance/rejection:	May 19, 2006
Deadline for final paper submission:	June 2, 2006
Workshop starts:	July 27, 2006

Submissions should be formatted according to **Springer LNCS style.**
Papers should have about 10 pages but must not exceed **15 pages.**

Support:

http://www.im2.ch
http://www.unige.ch
http://cui.unige.ch
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