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First English language issue
July

We can be proud of the results achieved in Hungary in the areas of telecommunications, research and development. Valuable research work is being done in factories, by service providers and primarily in universities and the produced results are put to practice not only within Hungary. People who use new methods generally do not know by whom and where were these modern solutions developed. This way intellectual property rights get lost although it would be important both for companies and individuals to make their professional results recognised not only in Hungary. Knowledge is a significant value that can bring benefit.

That is why it is necessary to present major professional results on the international arena. A precondition thereto is to publish results in English language to be accessible to everybody. That is why the editorial board decided to publish from time to time, two or three times a year, an English language issue of Híradástechnika (Communications).

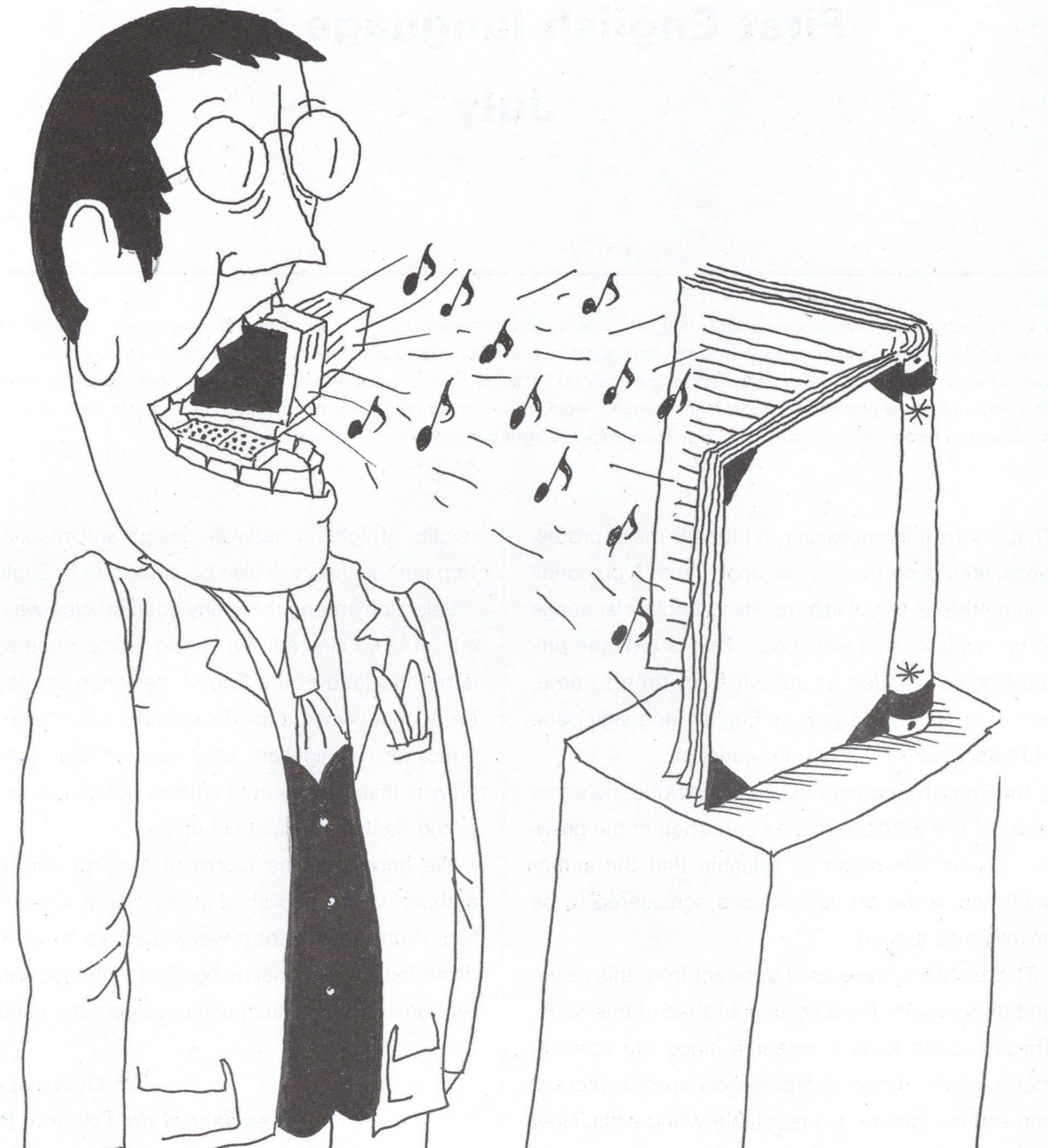
The English language issues will include the translation of the valuable articles published in the previous period. We mean by valuable that the author published in the article solutions considered to be novelty also abroad.

These criteria were used to select from this year's first three issues the articles published in this issue. These include speech research since the achievements of Hungarian computerised speech production and recognition are reputable world-wide. New results of lighting network design and mobile system introduction will also be published in English.

Selection among the many good articles was difficult. It is no discredit when the article of an author is not included in the English selection. It may still be a good work, provide valuable information for education, or highlight new aspects. Our common view is that the selected articles will be real novelty to the readers of most countries.

We hope that the works of a rising number of authors will be published in the future also in English. With these efforts we would like to bring the deserved international recognition to the journal, professionals and research and development institutes.

Dr. György Lajtha
President of the Editorial Board



Nonparametric Bayesian Estimation by Feedfoward Neural Networks

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The paper is concerned with developing novel nonparametric detectors implemented by neural networks. These detection algorithms are of great importance in point-to-point microwave digital communication and mobile communication, as well. It is proven that asymptotically optimal detection performance can be achieved by the proposed methods. The minimal algorithmic complexity of the newly developed algorithms are ensured by different coding techniques. Extensive numerical results demonstrate the optimal performance of the new detection schemes in the case different channel models. This makes them able to perform real-time, optimal detection in practical digital communication systems.

Introduction – The model

In digital communication systems there are digital patterns to be recognized based on noisy and distorted observations. The transmitter transmits binary vectors which represent coded messages (e.g. digitized voice in mobile communication). This binary vectors, however, are subject to distortion and noise due to the narrow band communicatin channel. The distortions results from the multipath propagation characteristics of radio channels. Thus, the received signal is an imperfect version of the transmitted code vector, which can give rise to detection errors. As a result, detection algorithms, being robust enough to counter the channel distortion and noise, are of great importance. In order to derive such kinds of detection scheme, let us assume that the following quantities are given:

- a set of binary codewords ***(messages)***, $Y = \{-r,r\}^M$ where $|Y| = N = 2^M < \infty$ and the amplitude r^2 is related to the transmission power;
- a $P(y)$ probability distribution of messages (in most cases it is supposed to be uniform);
- a continuous random variable ***(received sequence)*** $\mathbf{x}$ which is observable and is a random function of $\mathbf{y}$ (e.g., $\mathbf{x} = \mathbf{Hy} + \upsilon$ where $\mathbf{H}$ is an unknown linear transformation and υ is an additive noise, as depicted by figure 1);
- a training set
  $\tau^{(n)} = \{(\mathbf{x}_k, \mathbf{y}_k), k = 1, \ldots, n)\}$,
  in which for each received $\mathbf{x_k}$ the corresponding transmitted $\mathbf{y_k}$ is identified;
- a mapping $\mathbf{y} = \mathbf{f(x,w)}$, called detector, which performs the classification, were $\mathbf{w}$ represents the free parameters of the detector structure which are subject to optimization in order to optimize the detection performance. Note that the notation "bold $\mathbf{f}$" refers to the fact that the output of this function is a vector thus we keep block coding/decoding in mind;
- the $\mathbf{y}$ messages are given, Bernoulli distributed probability variables $y_i \in \{-1,1\}$;
- $\mathbf{s}$ is the codevector, where the goal of coding is to achieve the minimal numerical complexity;
- the vector $\mathbf{s}$ is transmitted by the channel, which transfrms them by the following linear transformation: $\mathbf{x} = \mathbf{Hs} + \eta$,

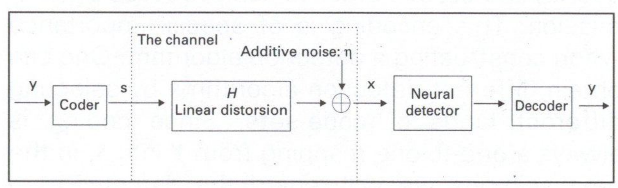

Figure 1 Digital communication system applying teh neuron based AOBDR

- where the matrix $\mathbf{H}$ is a Toeplitz-matrix derived from the discrete impulse response function of the channel: $H_{ij} = h_{i-j}$; η is an N dimensional normally distributed white noise vector with covariance matrix $\mathbf{K}$ and the following density function:

$$f(\boldsymbol{x}) = \frac{1}{\sqrt{(2\pi)^N \det \mathbf{K}}} exp\left(-\frac{\mathbf{x}^T\mathbf{K}^{-1}\mathbf{x}}{2}\right);$$

on the receiver side of the channel one can observ the vector $\mathbf{x}$. We use a neural network based detector to detect the original $\mathbf{s}'$ vector from $\mathbf{x}$.

In the case of given noise distribution and channel distortion, it is well known [1, 2] that the optimal detection algorithm denoted by $\mathbf{f}_{\mathbf{opt}}(\mathbf{x})$, which minimizes the error probability is the Bayesian rule. Namely if the error probability is defined as

$$P_E(\mathbf{f}) := \sum_{\mathbf{y}} P(\mathbf{f}(\mathbf{x}) \neq \mathbf{y} | \mathbf{y}) p(\mathbf{y}),$$

which is to be minimized over all possible decision rules F, then the optimal detector must perform the following mapping:

$$\mathbf{y} = \mathbf{f}_{\mathbf{opt}}(\mathbf{x}) = \mathrm{argmax}_{\mathbf{y} \in Y}\, p(\mathbf{y}|\mathbf{x}) = \mathrm{argmax}_{\mathbf{y} \in Y}\, p(\mathbf{x}|\mathbf{y}) P(\mathbf{y}) \tag{1}$$

The objective of nonparametric detection is to approximate $\mathbf{f}_{\mathbf{opt}}: X \rightarrow Y$ based on the training set $\tau^{(n)} = \{(\mathbf{x}_k, \mathbf{y}_k, k = 1, \ldots, n)\}$. More precisely, an estimation of the Bayesian rule is to be given in a recursive fashion:

$$\mathbf{f}_{n+1}(\mathbf{x}, \mathbf{w}, \tau^{(n+1)}) = \Psi(f_n(\mathbf{x}, \mathbf{w}, \tau^{(n)}), \mathbf{x}_{n+1}, \mathbf{y}_{n+1}), \tag{2}$$

for which

$$\lim_{n \to \infty} \mathbf{E} \| \mathbf{f}_n(\mathbf{x}, \mathbf{w}, \tau^{(n)}) - \mathbf{f}_{\mathbf{opt}} \|^2 = 0.$$

We call such an $\mathbf{f}_n(\mathbf{x}, \mathbf{w}, \tau^{(n)})$ estimator as *Asymptotically Optimal Recursive Bayesian Detector* (AORBD).

When implementing such a nonparametric detection algorithm in a communication system, the designer has the liberty to encode the original messages $\mathbf{y}^{\mathbf{i}} \in Y$ into a new set $\mathbf{s}^{\mathbf{i}} \in S$. This is frequently the case in order to achieve reliable transmission. This encoding is of special importance when constructing a detection algorithm. One can obtain different detection algorithms by selecting different kinds of code-sets. Since coding is always a one-to-one mapping from Y into S, in the next sections we will regard the detector as a mapping $\mathbf{s} = \mathbf{f}(\mathbf{x}, \mathbf{w})$ and are concerned with its computational complexity. Our objective is to find optimal encoding scheme $\mathbf{s} = \mathbf{f}_{\mathbf{opt}}(\mathbf{y})$ which allows for a simple neural network and a good approximation of the optimal Bayesian decision rule and yields a fast learning algorithm.

The reader should note that the overall number of patterns is N which is usually encoded into $M = \log_2 N$-bit length binary messages (assuming uniform message distribution). Nevertheless, to implement a detector which can effectively combat against noise and distortion, we are better off having extended codewords with dimension larger than M. Thus, when we talk about detection complexity, we basically bear in mind how close we can get with the dimension of the codewords $\mathbf{s}$ to $\log_2 N$. This is a good measure of complexity, taking into account that in the case of dim $(\mathbf{s}) >> \log_2 N$ the corresponding detector complexity increases in an exponential manner.

Some results from neural network theory

As our estimates will be implemented by means of Feedforward Neural Networks (FFNNs), this section gives a brief summary of the results we need.

A k layer feedforward neural network is given in the for of

$$F_j = \psi_j^{(k)} \left(\sum_{i_{k-1}=1}^{l_{k-1}} \omega_{i_{k-1} j}^{(k)} \psi_{i_{k-1}}^{(k-1)} \left(\ldots \sum_{i_1=1}^{l_1} \omega_{i_1 i_2}^{(2)} \psi_{i_1}^{(1)} \left(\sum_{i_0=1}^{N} \omega_{i_0 i_1}^{(1)} x_{i_0} \right) \right) \right), \tag{3}$$

which mapping is depicted in Figure 2. For short, we will refer to this mapping as $\mathbf{F}(\mathbf{x}, \mathbf{w})$, where

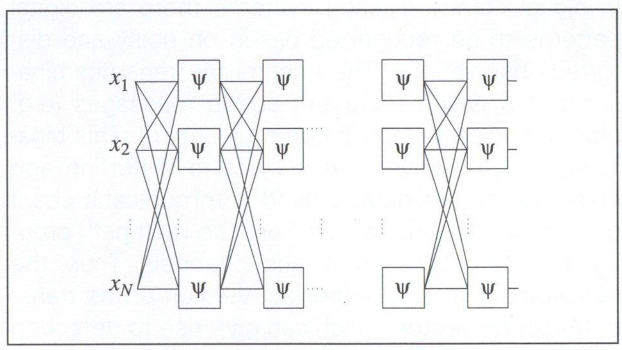

Figure 2 A feedforward neural network with k + 1 layers

the bold letters represent vectors (the network may have many outputs) and the weight vector is denoted as

$$\mathbf{w} = (w_{11}, \ldots, w_{1l_1}, w_{2l}, \ldots, w_{2l_2}, \ldots, w_{k1}, \ldots, w_{kl_k}).$$

In our detection scheme, the input vector $\mathbf{x}$ represents the observable variable, while the output is a vector which (after quantization) provides the detected codeword. Therefore, the detector is constructed as $\mathbf{s} = \mathbf{f}(\mathbf{x}, \mathbf{w}) = D\mathbf{F}(\mathbf{x}, \mathbf{w})$, where D denotes an appropriate quantization operation specified later.

The approximation and learning capabilities of FFNNs

The approximation capabilities of this type of network in L^p spaces have been demonstrated in numerous papers [5, 6, 7, 8] and are based on the following result:

Theorem 1:
A one-layer neural network (setting $k = 1$) is a universal approximator in L^p, in the sense that $\forall \mathbf{G(x)} \in L^p$ and given $\epsilon > 0$ there exists a F(x,w), such as:

$$\left(\int \cdots \int_X \| \mathbf{G(x)} - \mathbf{F(x,w)} \|^p dx_1 \ldots dx_N \right)^{\frac{1}{p}} \le \epsilon \quad (4)$$

For a detailed proff see [8, 7].

According to the statistical nature of the input vectors, one can consider the FFNN as performing a nonlinear regression by minimizing the following quadratic error function

$$\min_{\mathbf{w}} \frac{1}{n} \sum_{k=1}^{n} \| \mathbf{F(x_k,w)} - \mathbf{y_k} \|^2.$$

This, however, gives rise to the following result:

Theorem 2:
Having a training set $\tau^{(n)} = \{(\mathbf{x}_k, \mathbf{s}_k),\ k = 1, \ldots, n\}$ of size n and optimizing the weights as follows

$$\mathbf{w_{opt}} = \lim_{n \to \infty} \mathbf{w_{opt}^{(n)}},$$

where

$$\mathbf{w_{opt}^{(n)}} : \min_{\mathbf{w}} \frac{1}{n} \sum_{k=1}^{n} \| \mathbf{F(x_k,w)} - \mathbf{y_k} \|^2,$$

then

$$\mathbf{F(x,w_{opt})} = E(\mathbf{s}|\mathbf{x}). \quad (5)$$

The proof can be found in [10].

A frequently used technique for finding $\mathbf{w_{opt}}$ is the Back Propagation (BP) algorithm, which often fails to achieve the global minimum, though. Therefore, statistical optimization techniques, such as simulated annealing or diffusion can be used to determine the corresponding $\mathbf{w_{opt}}$. For a more detailed review on these theories and their application to neural network learning see [9].

Nonparametric detection by FFNNs

In this section, we will demonstrate that the task of nonparametric detection can be carried out by FFNNs. We will also assess the complexity of the proposed algorithms with special regard to the number of outputs.

In this section we deal with the optimal detection as an algebraic problem related to a set of linear equations. Our objective is to get a better insight of different coding schemes by using some techniques of linear algebra. As was seen before, when estimating the Bayesian rule by a feedforward neural network the following conditional expected value vector is estimated:

$$\mathbf{e} = \mathbf{F(x,w)} = E(\mathbf{s}|\mathbf{x}). \quad (6)$$

where $\mathbf{e}$ represents the vectorial output of the network. Then (13) can be rewritten into the form

$$e_i = \sum_{j=1}^{N} s_j^{(i)} p\,(\mathbf{s^{(i)}} | \mathbf{x})\ j = 1, \ldots, N\ i = 1, \ldots, L, \quad (7)$$

where $\mathbf{x}$ is the input vector presented to the neural network and $\mathbf{s^{(i)}}$ $i = 1, \ldots, N$ are the codewords dedicated to the patterns to be detected. Equation (14) above can be viewed as a set of linear equations $\mathbf{e} = \mathbf{Sp}$, where the ijth element of matrix $\mathbf{S}$ is defined as the *j*th component of the *i*th codeword $S_{ij} := s_j^i$ and the *i*th element of vector $\mathbf{p}$ is $p_i := p\,(\mathbf{s^{(i)}} \mid \mathbf{x})$. Note that $\mathbf{S}$ is of type LxN where $N = 2^M$ is fixed by the number of patterns to be detected. Nevertheless the parameter L which only determines the dimension of the codewords can be smaller than N. This parameter greatly influences the complexity of the underlying neural network (it equals to the number of outputs). Throughout this paper we have been concerned with minimizing L. Of course L cannot be smaller than $\log_2 N$ in order to differentiate among the patterns. This gives rise to the following algebraic problem:

Problem:
Given the set of linear equations $\mathbf{e} = \mathbf{Sp}$, let us assume that one can observe only vector $\mathbf{e}$, vector $\mathbf{p}$ is not subject to observation, whereas matrix $\mathbf{S}$ can be freely constructed. Then how to construct matrix $\mathbf{S}$ (how to select the codewords) that the index of the maximum component of the unknown vector $\mathbf{p}$ can be identified (optimal detection) based on observing $\mathbf{e}$? Furthermore, we would like to have a construction where $L << N$ which allows a low complexity neural network to perform the detection task.

We can distinguish two cases:

1. $\mathbf{S}$ *is a square matrix:*
   When $\mathbf{S}$ is of tye NxN, then $\mathbf{p}$ can be calculated as $\mathbf{p} = \mathbf{S^{-1}e}$ and the detection rule is $i = \mathrm{argmax}_j p_j$. In this case, one can choose $\mathbf{E}$ (which is the unit matrix) to be $\mathbf{S}$, in order to avoid matrix inversion. Hence, $s_j^i := \delta_{ij}$.

2. $\mathbf{S}$ *is a non-square matrix:*
   When $\mathbf{S}$ is of tye NxL, where $L << N$, then some other calculations are needed to get $i = \mathrm{argmax}_j p_j$ based on observing $\mathbf{e}$, if possible at all. These cal-

culations are represented by the transformation $i = \Psi(\mathbf{e})$. In the next section we will develop such a transformation, the complexity of which is much smaller than matrix inversion. Since $L = \log_2 N$ theoretical lowerbound can be achieved, the complexity of the associated neural network will be minimal, as well.

To obtain a code construction by which means we can achieve the theoretically possible lowerbound $L = \log_2 N$, we assume that the maximal component of vector p which is denoted by pl is dominant of the vector, meanin that $p_r > \sum_{j,j\neq r} pj$. Based on this property one can easily establish the following lemma:

Lemma 1:
Let assume that vector $\mathbf{p}$ is a probability mass vector and $r = \text{argmax}_i p_i$. If vector $\mathbf{p}$ is multiplied with any binary vector with components $b_i \in \{-1, 1\}$ then the sign of $\mathbf{b}^T\mathbf{p}$ equals to the sign of b_r.

Proof:
Let us write the inner product into the following form

$$\sum_j b_j p_j = b_r p_r + \sum_{j,j\neq r} b_j p_j \leq \sum_{j,j\neq r} |b_j p_j| = \sum_{j,j\neq r} p_j .$$

Using the property $p_r > \sum_{j,j\neq r} pj$ it is clear that $sign(b_r) = sign(\mathbf{p}^T\mathbf{b})$.
Q.E.D.

Theorem 3:
If
(i) the condition $p_r > \sum_{j,j\neq r} pj$ holds;
(ii) one constructs matrix $\mathbf{S}$ as $S_{ij} = 2\left(\left[\frac{i}{2^j}\right] mod\, 2\right) - 1$,
(iii) we observe vector e = Sp,
then the index of the maximal component of $\mathbf{p}$ (detoned by r in the previous discussion) is indicated by the binary string obtained as $sign(e_1), sign(e_2), \ldots, sign(e_{\log 2N})$.

Proof:
First it is noteworthy that according to the construction of $\mathbf{S}$ the row vectors are binary vectors having +1 or -1 values with increasing frequency (i.e., in the first row of $\mathbf{S}$ the first *M/2* components are +1 and the rest is -1; in the second row of $\mathbf{S}$ the first M/4 components are +1, the next M/4 are -1 ...etc.). Performing the matrix vector multiplication for the first component

$$e_1 = \sum_{j=1}^{M} S_{1j} p_j$$

it is easy to see that

$$sign(e_i) = 2\left(\left[\frac{i}{2^r}\right] mod\, 2\right) - 1.$$

Consequently, the number

$$\sum_{i=1}^{N} \frac{sign(e_i)+1}{2} 2^i$$

constructed form the binary string of $sign(e_i)$ $i = 1, \ldots, N$ will exactly yield *r*. Therefore, the maximum conditional probability can be reconstructed by observing only $sign(e_i)$ $i = 1, \ldots, N$. A more verbal elaboration on this statement is as follows:
If index r falls into the first M/2 then $e_1 > 0$, otherwise $e_1 < 0$. Investigating

$$e_2 = \sum_{j=1}^{M} S_{2j} p_j ,$$

one can establish that if $e_2 > 0$ then index r should fall either in the interval $1, \ldots, M/4$ or $M/2, \ldots, 3M/4$. With the knowledge of the sign of e_1 however, one can fully ascertain which of the two cases is true (e.g., if $e_1 > 0$ then $r \in (1, \ldots, M/4)$ ...etc.) As a result since there are $\log_2 M$ component of $\mathbf{e}$, the binary vector obtained as $(sign(e_1), sign(e_2), \ldots, sign(e_{log2N}))$ will uniquely identify the component *r* where the maximal p_r is (in fact the vector $(sign(e_1), sign(e_2), \ldots, sign(e_{log2N}))$ the binary representation of the number *r).*
Q.E.D.
With this method, low complexity optimal Bayesian detection can be achieved.

Performance analysis – estimating the error probability of the nonparametric detection algorithm

When calculating the performance of nonparametric detection schemes there are two effects to be taken into account:

1. the conditions on $p(\mathbf{s} \mid \mathbf{x})$, which were introduced by Theorem 3;
2. the approximation error with which the neural network represent the conditional expectation (which is partly due to the finite number of neurons and partly to the finite duration of learning).

Therefore in the rest of the paper we are going to be engaged to evaluate how this factor affect the detection performance. The performance is measured by the error probability, thus our concern is to come up with a bound like

$$P_{Bayes\ opterror} \leq P_{error} \leq P_{Bayes\ opterror} + P_{appr} , \qquad (8)$$

where the term P_{appr} is responsible for the secondary effects mentioned above. In this evaluation the following lemma will be of use:

Lemma 4:
Let us assume that there are two conditions imposed on $\mathbf{x}$ *under which the optimal Bayesian performance is achieved. In general, these two conditions are denoted by* $\mathbf{x} \in \mathrm{A}$ *and* $\mathbf{x} \in \mathrm{B}$*, respectively. (E.g., set* A *represents the conditions imposed on the density, whereas* B *results from the conditions imposed on the neural network complexity and the length of the learning set in order to obtain an accurate approximation.) Then*

$$P_{error}(\mathbf{d}) \le P_{Bayes\ opterror} + (1 - P(A)) + (1 - P(B)) + (1 - P(A))(1 - P(B)), \tag{9}$$

where

$$P(A) := \int \cdots \int_{X \in A} p(x_1, \ldots, x_k) dx_1 \ldots dx_k,$$

$$P(B) := \int \cdots \int_{X \in B} p(x_1, \ldots, x_k) dx_1 \ldots dx_k$$

and $P_{Bayes\ opterror}$ *is the theoretically achievable minimum error probability given by the Bayesian decision.*

In our interpretation *P(A)* will denote the probability of fulfilling the conditions imposed on the conditional densities, whereas *P(B)* stands for the probability that the neural network fulfills some criteria needed for the accurate approximation of the conditional expected value. The forthcoming sections deal with assessing these probabilities.

The effect of restrictions imposed on the conditional density

First let us recall that $\mathbf{x} = \mathbf{Hs} + \nu$, where ν is now supposed to be a multivariate Gaussian random variable with zero mean and covariance matrix $\mathbf{K}$. Hence, $p(\mathbf{s^i} \mid \mathbf{x})$ can be characterized by the following distribution:

$$p(\mathbf{s^i} | \mathbf{x}) = \frac{p(\mathbf{x} | \mathbf{s^i}) p(\mathbf{s^i})}{p(\mathbf{x})} = \frac{e^{\frac{1}{2}(\mathbf{x} - \mathbf{Hs^i})^T \mathbf{K}^{-1} (\mathbf{x} - \mathbf{Hs^i})}}{\sum_{m=1}^{N} e^{\frac{1}{2}(\mathbf{x} - \mathbf{Hs^m})^T \mathbf{K}^{-1} (\mathbf{x} - \mathbf{Hs^m})}}. \tag{10}$$

When investigating FFNNs as AOBDR, the applicability of them according to Theorem 3 boiled down to:

Condition 1:

$$p(\mathbf{s^1} | \mathbf{x}) > \sum_{j, j \ne 1} p(\mathbf{s^j} | \mathbf{x}) \ \ \forall\ \mathbf{x} \in X \ \text{if}\ p(\mathbf{s^1} | \mathbf{x}) = \max_i p(\mathbf{s^i} | \mathbf{x})$$

Therefore now we are embarking the evaluation of *P(A)* determined by Condition 1.

Calculating P(A) in case of Condition 1

Now we embark on evaluating *P(A)* when Condition 1 is our concern. Then the following theorem can be obtained.

Theorem 4:
The probability of Condition 1, in the case of Gaussian noise model is given as

$$P(A) = \int \cdots \int_{X \in A} p(\mathbf{x}) dx_1, \ldots, dx_M, \tag{11}$$

where the set A *is defined as* $A = \bigcup_{r=1}^{M} A_r$ *and*

$$A_r = \left\{ \mathbf{x} : \exp\left((\mathbf{x} - \mathbf{Hs}^{(r)})^T K^{-1} (\mathbf{x} - \mathbf{Hs}^{(r)}) \right) > \right. \left. > \sum_{i, i \ne r} \exp\left((\mathbf{x} - \mathbf{Hs}^{(i)})^T K^{-1} (\mathbf{x} - \mathbf{Hs}^{(i)}) \right) \right\}.$$

In the case of white Gaussian noise one obtains

$$A_r = \left\{ \mathbf{x} : \exp\left(\sum_j \frac{(x_j - s_j^{(r)})^2}{N_0} \right) > \sum_{i, i \ne r} \exp\left(sum_j \frac{(x_j - s_j^{(i)})^2}{N_0} \right) \right\}$$

where N_0 *is the spectral density of the white Gaussian noise.*

One can expect that if the N_0 is small then the measure of A characterized by

$$P(X \in A) = \int \cdots \int_{X \in A} p(\mathbf{x}) dx_1, \ldots, dx_M$$

will also be small yielding a small departure from the Bayesian decision rule (a slight increase in the error probability).

Feedforward neural networks based on the Moore-Penrose inverse

In case one codes the messages such a way that the codewords correspond to the lines of the unit matrix, then the length of the codewords (and consequently the complexity of the neural network) increases exponentially with respect to the number of messages. To overcome this problem in this section we introduce another method that is capable of reducing the complexity of the network in a logarithmic manner. The underlying idea is to determine the maximum of the conditional distribution $p(\mathbf{x}, \mathbf{s}^i)$ in order to perform the Bayes decision. Let the output vector of *FFNN* be denoted by $\mathbf{r}$. The coding is carried out by the matrix $\mathbf{S}$ ($\mathbf{S}$ is arbitrary but is must realize a bijective mapping), in such a way that the *i*-th column of $\mathbf{S}$ corresponds to the *i*-th codeword.

One must note that **S** is not necessarily a square matrix (the number of rows and columns can differ from each other). If **p** is the conditional distribution vector, then

$$\mathbf{Sp} = \mathbf{r}$$

and

$$\mathbf{p} = \mathbf{S}^{-1}\mathbf{r} \qquad (12)$$

it **S** is a square matrix. Because **S** is generally not square (otherwise we could not shorten the length of the codewords with respect to the messages), we have to generate the Moore-Pensore inverse to solve 12. More precisely, we can approximate **p** in the following way:

$$\mathbf{p^*}: \min_p \|\mathbf{Sp}-\mathbf{r}\|^2 = \min_p (\mathbf{Sp}-\mathbf{r})^T(\mathbf{Sp}-\mathbf{r}) = \\ = \min_p \mathbf{p}^T\mathbf{S}^T\mathbf{Sp} - 2\mathbf{r}^T\mathbf{Sp} + \mathbf{r}^T\mathbf{r},$$

which can be written as

$$\mathbf{p^*}: \min_p, \mathbf{p}^T\mathbf{S}^T\mathbf{Sp} - 2\mathbf{r}^T\mathbf{Sp}$$

where $\mathbf{W} = \mathbf{S}^T\mathbf{S}$ és $\mathbf{b} = \mathbf{r}^T\mathbf{S}$. This expression has a gradient, so the global minimum is

$$\mathbf{Wp^*} - \mathbf{b} = 0 \rightarrow \mathbf{p^*} = \mathbf{W}^{-1}\mathbf{b},$$

substituting **W**, we get

$$\mathbf{p^*} = (\mathbf{S}^T\mathbf{S})^{-1}\mathbf{Sr},$$

where $(\mathbf{S}^T\mathbf{S})^{-1}\mathbf{S}$ is often referred to as the Moore-Penrose inverse. As it can be seen the numerical complexity of this is still relatively high ($O(N^3)$), which may hinder real-time applications. Therefore one can use an iterative version to calculate the Moore-Penrose inverse, namely

$$\mathbf{p}(k+1) = \mathbf{p}(k) - \Delta\{\mathbf{Wp}(k) - \mathbf{b}\}.$$

It is easy to prove that choosing a proper Δ, the recursion converges to **p***.

$$(P^* = p(k+1) = p(k) = Wp(k) = b$$

It can also be proven, that the optimal value for Δ is

$$\Delta_{opt} = \frac{2}{(\lambda_{min} + \lambda_{max})},$$

where λ_{min} and λ_{max} are the minimal and maximal eigenvalues of **W,** accordingly. As practical examples show the calculation of the exact value of parameter Δ is not necessary, an approximation also suffices, e.g. one increases Δ from zero until a predefined level. Besides the methods introduced above there are other methods to minimize the complexity as well (e.g. rectangular grid coding, triangular grid coding). These methods require different restrictions on the distribution. A detailed description of these methods and the corresponding restrictions can be found in [14] and [15].

Numerical results and applications

Based on the results listed above a high performance nonparametric detector can be constructed which can detect digital messages with error probability close to the minimal one. The detector performance proves to be more or less robust to the statistical nature of the noise and the channel. In this section we are providing some numerical results which support these claims by simulations. Our primary concern will be to highlight the relationship between transmission power, error probability (performance), an detector complexity. The numerical evaluation of these relationships yields useful data for practical engineering design.

	h_0	h_1	h_2
Channel 1	0.818	0.407	0.0
Channel 2	0.688	0.0460	0.277

Table 1 Discrete channel characteristics

The communication scheme under investigation is depicted by figure 1. Two typical channel models which frequently occur in digital communication systems ([2]) (especially in communication over radio channel corrupted by multipath propagation and Gaussian noise) were put under invesstigation. The channel characteristics are given by ***Table 1.*** The simulations first dealt with calculating the probabilities $P(A)$ and $P(B)$, needed to evaluate the performance. Then the calculated error probability is compared with the error rate obtained by simulations over one million amount of data. The performance will also be numerically analyzed in the case of non-Gaussian noise.

Performance analysis by calculating the bounds on the error probability

As was detailed in *Section 5* the error probability of AOBDR can be upperboundes as

$$P_{error} \leq P_{Bayes\ opterror} + (1 - P(A)) + (1 - P(B)).$$

From this expression it is clear that the larger $P(A)$ and $P(B)$ are, the better the performance is. One must also remember that the different coding schemes yielded differenc $P(A)$-s. Figure 3 shows the dependence of $P(A)$ on the transmission power

r, in the case of different coding and channel distortions. It is noteworthy, that the figure 3 says what transmission power is paid for achieving a certain $P(A) \geq 1 - \epsilon$, which is of direct use for engineering design. One, also must notice, smaller r resulting better $P(a)$ (e.g. in the case of geometrical coding) will yield a more complicated neural network (the number of out puts are not the logarithm of N but only a corresponding root).

Next, we turn our attention to the effect of the approximation error of **FFNN** expressed by $P(B)$ (see *Secton 5.2*). For Channel 1 the necessary training set size was 11,491.

For Channel 2 the necessary training set size was 55,424.

As a result the overall bound on the performance is pictured by the next tables, where the intricate relationship between the coding schemes (detection complexity), performance, and transmission power can be analyzed.

Performance analysis by simulations

In this part we validate the performance by simulations, measuring the error rate. If this error rate is

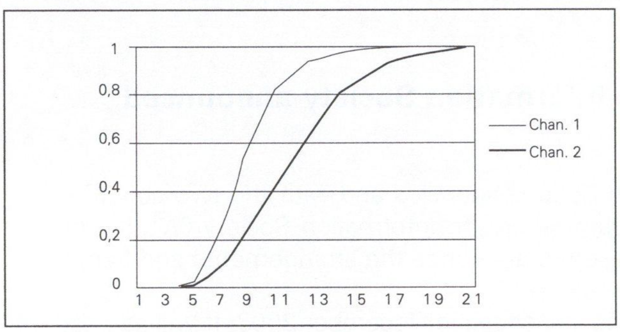

Figure 3 P(A) versus signal to noise ratio in the case of Channel 1 and 2

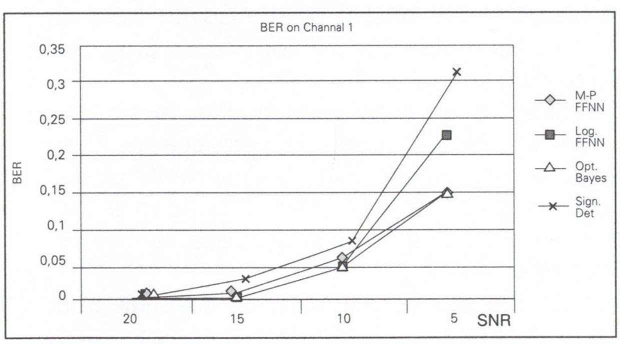

Figure 4 BER simulation results on Channel 1

more or less the same or smaller than the analytical performance (which was calculated as an upper bound) then the simulations validate our bounds.

In order to compare the obtained results, we also indicate the performance of the traditional ***signum detection*** (Sign.Det.) algorithm and the theoretically optimal Bayes decision rule.

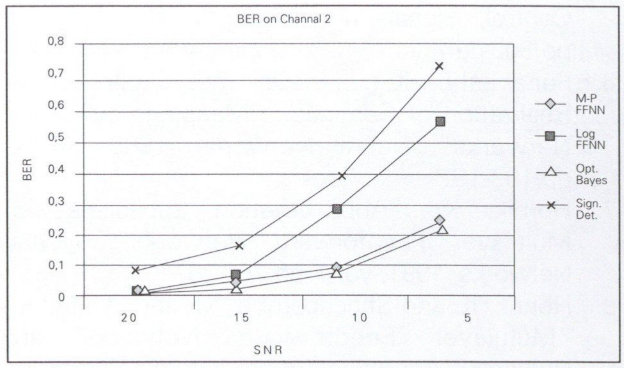

Figure 5 BER simulation results on Channel 2

Conclusions

Novel nonparametric detection algorithms were developed for digital communication systems using feedforward neural network architectures. Different coding schemes were proposed to obtain the optimal Bayesian performance under certain conditions. These conditions shed light on the intricate relationship among detector complexity, performance and transmission power. The analytical results and the detailed numerical analysis yield quantitative tools for engineering design, when the designer is to strike a good balance among the detector complexity, performance and transmission power. The algorithms proposed in the paper can be applied in mobile and point-to-point microwave digitial communication when signals are to be detected under varying distortion and noise.

Acknowledgement

The work reported here was supported by the project COP579 sponsored by the EU and by the project T025964 sponsored by the Hungarian National Scientific Foundation.

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News

Framework of the World Summit on the Information Society announced

Geneva – After extensive consultations with United Nations agencies and with the two countries having submitted candidatures to host the World Summit on the Information Society (WSIS), the Secretary-General of the ITU, Yoshio Utsumi, is pleased to announce the arrangements and framework for WSIS.

The first phase of the World Summit will take place in Geneva in December 2003. It will address the broad range of themes concerning the Information Society and adopt a Declaration of Principles and Action Plan, addressing the whole range of issues related to the Information Society.

The second phase of the World Summit will take place in Tunis hosted by the Government of Tunisia, in 2005. Development themes will a key focus in this meeting and it will access progress that has been made and adopt any further action Plan to be taken.

The Information Society promises a fundamental change in all aspects of human existence, including knowledge dissemination, social interaction, economic and business practices, political engagement, media, education and health, leisure and entertainment. But the inherently global nature of this new environment, and the uneven pace of its development – often referred to as the Digital Divide, make it vital that international dialogue take place and that world-wide approaches be considered to facilitate the successful adaptation to this new reality.

The Summit will provide a unique opportunity to gather the world community to assemble at a high level and to develop a better understanding of this revolution and its impact. It aims to bring together Heads of State, Executive Heads of United Nations Agencies, industry leaders, non-governmental organizations, media representatives and civil society.

The proposed themes raised by the Information Society and will likely include:
- Building the infrastrucure
- Opening the gates: universal and equitable access to information society
- The needs of the User
- Developing a Framework
- ICT and Education

Why and How of the Digital Television in Hungary

GYÖRGY ÁGOSTON

Technical Director, Hungarian Television Co.

Hungary is a medium size European country with developing economy. Its capability for implementing new technologies are differing from the more developed countries. However, the Hungarian broadcast professionals have recognised the advantages of the new digital services in several aspects. They have come together and worked out a strategic plan for tempting the officials to make the necessary and appropriate decisions for the as early as possible implementation of digital terrestrial television in Hungary. Below, the major issues of this plan are discussed, and the "why and how" questions are partly answered.

Introduction

Hungary is a medium size country in Central Europe. This country is surrounded by as many as seven neighbouring countries, including such a giant, as Ukraine. Of course, to be small, nevertheless having so many neighbours around is sometimes an advantageous position, on the other hand, sometimes less beneficial. In this context, to coordinate and allocate frequencies to all of the digital services which are to be established and developed in the foreseeable future is, not an easy task for our frequency authorities.

Another interesting fact according to relevant statistics, Hungarian people are sitting in front of their TV-sets every day the longest time in Europe watching at least three (one public and two commercial) nation-wide terrestrial channels and several regional and local programmes. Additionally, a lot of free DTH analogue and digital satellite TV programmes (and also some pay TV-channels), mostly from the Astra and Hot Bird satellites are available for those approximately 45 percent programme-hungry Hungarians, who are connected to cable networks and another 20 percent who have analogue or digital satellite receivers at home.

Probably the great interest of the people for the media and also the strong and strict political influence during the socialistic era had generated a desperate and lengthy battle on the media among the political forces immediately after the political changes in 1990, which lasted until 1996. Then the political parties came to an agreement and accepted the first media law in Hungary. The law proved to be not without any problems, nevertheless it opened a wide gate to the commercial media, which had no roots in Hungary earlier. The main idea was to provide competition not only between the public and the new commercial media, but, at the same time, to make the commercials also compete with each other. Therefore, one year later, the newly established National Radio and Television Commission awarded two licences to the new nation-wide commercial broadcasters: the TV2 and the Hungarian RTL-Club.

As only three nation-wide terrestrial frequency-sets were available, one of the sets was taken away from Hungarian Television, the national broadcaster, who had two nation-wide terrestrial programme-channels until 1997. Since then, the second public channel has been transmitted by one of the Hot Bird satellites.

The digital dawn...

In the mid of the 90's some ingenious business people in Hungary came to the conclusion: if the people are so keen watching TV, why not to satisfy them as much as possible with more and more programmes. The emerging DVB-S digital satellite standard was the key to satisfy the people and to make money at the same time (what is the first, of course, for the businessmen).

So, in 1997, Antenna Hungaria, the Hungarian programme transmission company, in cooperation with a specialised company from Israel, launched the first Hungarian digital programme-bouquet through the AMOS satellite. There were four Hungarian-speaking digital programmes in the bouquet and most of the cable network operators were pleased to carry them (after converting into analogue format) to their costumers. As the cable penetration was as high as 40 % at that time, more than 3 million people were able to enjoy this new and unique service. It was really unique, because lots of Hungarian people did not speak any foreign languages, so it was a real joy for the people to understand what was going on on the screen.

Since then, the content of the bouquet has been changed several times, but the popularity of the programmes has not fallen of, moreover, nowadays, an increasing number of digital satellite set-top-box owners are able to receive directly these and other digital DTH programmes.

The Hungarian Digital Platform

Later, when the DVB-T digital terrestrial standard appeared as a possible supplement to the digital satellite services, some experts from different Hungarian institutions somehow related to the television broadcasting (including Antenna Hungaria, the transmission company, Hungarian Television, the national broadcaster, the Technical University in Budapest, the Ministry of Communication, the Communication Authority, the National Radio and Television Commission, and others) came together and started lengthy and vivid discussions about the possible future of DVB-T in Europe and particularly in Hungary. On the basis of the warm welcome of digital satellite, the successful future of DVB-T was inevitable for them. They believed that the digital terrestrial broadcasting would carry with it extraordinary possibilities, but it also would come with significant challenges.

In the beginning of 1998, within the frame of the Hungarian Scientific Association for Info-communications, these experts established the Hungarian DVB-platform, aiming at thoroughly studying the technical, economic, financial and legal conditions of the introduction of digital terrestrial television services in Hungary. The major goal of the platform was to answer the questions in the title of this presentation: Why and How of the Digital Television in Hungary? And also: When?

The members of the platform intensively followed what was going on over Europe on this matter, studied the different approaches to introducing digital terrestrial television, in particular the UK-model, and tried to find out how to adapt these experiences in Hungary. Soon, they understood that a number of great difficulties could emerge in the process of introducing terrestrial TV-broadcasting in Hungary. Some of them were generic for all countries, but there were several others too, which seemed to be specific to Hungary, with its specific geographic and economic conditions.

During the first phase of its work (by the end of 1999), the Platform had identified a number of specific technical, financial and political issues concerning the introduction of digital terrestrial television in Hungary:

The technical issues

- Hungarian Television, the Hungarian national broadcaster was one of the first of the broadcasters over the world, which introduced digital production in the studio in 1990. All the significant commercial TV-stations, which have been established since the launch of the commercial media in Hungary have also been equipped with digital technology. Considering the technical quality of the programmes, and also the value added capabilities of the digital transmission, all the major content providers in Hungary would be pleased if their digitally created signals would be delivered to their costumers in digital format.
- As the receiver market is relatively small in Hungary, nobody is interested in any proprietary, non-standardised solution, quite the contrary all the parties, including the few receiver manufacturers are interested in the adaptation of the standardised DVB-MHP initiative as a common platform.
- The frequency allocation is one of the major issues. At present, as it was mentioned earlier, there are three nation-wide terrestrial analogue television channels in Hungary, and several local services are also running. As a result, there are no free frequencies at all in the Band III, and the available frequencies in the Bands IV-V (470-862 MHz) allow to form only 3 multiplexes. Some of the frequencies necessary to complete these 3 multiplexes are still being used by military services (TV channels 61-69). Recently, Hungary has become a member of NATO, so it is expected that some frequencies presently used by the military services soon will be available for digital television. However, further difficulties are foreseen in the coordination with some neighbours, where the military facilities are not going to be changed. It is also questionable, how successful the coordination will be with Yugoslavia, which has not signed the Chester'97 Agreement.
- The Hungarian frequency authorities have recognised that there is a common interest with the neighbouring countries to use the frequency spectrum more efficiently. On the other hand, it has also been realised that any delay in starting the planning and coordination processes for the digital

frequency networks can cause a significantly disadvantageous position for Hungary in relation to its neighbours.

- The difficulties in the coordination, and the structure of the available frequencies allow only MFNs (Multi-Frequency Networks) to be envisaged for the first three multiplexes. According to the calculations of our frequency regulators, it would be favourable to have some nation-wide and also regional SFNs (Single-Frequency Networks) in the future, however, due to the above mentioned problems, SFNs can only be established in Hungary after the analogue services will be ceased, and, considering the geographical situation, if all the countries in this area will plan SFNs in a carefully coordinated manner.
- Important technical (and also financial) issue that to what extent and how to re-use the existing analogue infrastructure at both the sending and the receiving sites.
- Due to the geographical location, Hungary plays an important role in the European transport. Therefore, it is highly desirable to consider in the network planning the mobile receivers as well.

Financial issues

- Now, Hungary has a rapidly growing economy, nevertheless the GDP and the average standard of living are still lower then in the EU countries. As a consequence, the advertising market, which plays an extraordinary role in financing the media is still relatively limited.
- Naturally, all the new commercial TV-s and radios, and the public media, which, according to the media law, can also have a little portion from the limited market are being in a difficult financial position. (Note, the licence fee for the public programmes is so low in Hungary that it is simply not enough for supporting the public media.)
- In summary, the whole media sector (both, the commercial and the public) is short of money at the moment, so it is a big question that despite the keen interest in exploiting the new opportunities, who will be able to finance the content for the new digital services.
- The analogue-digital transition apparently could also be a good business for the set-top-box manufacturers in Hungary, and also for the small and medium-size suppliers of the electronic components. The success of the whole story, however, depends on a very important factor: will anybody subsidise the STBs or not? If not, this could postpone the entire introductory process, because the mass of consumers simply would hardly be able to afford 400 GBP for a STB (Set-Top-Box).
- The Government seems to be the only player at the moment, who could take the task to push the process forward even financially until the "critical mass" of the digital services has been reached and the market takes over the control. The Government should finance at least partially for example the setting up of the digital networks, the free licences of the public media in the multiplexes, and the subsidy of the STBs. This is still a question, whether the Government is intending or able to take this not easy task.
- The Government strongly supports development of technology and infrastructure of the coming Information Society. It is the experts' responsibility to make clear that the terrestrial digital television can be an important part of this infrastructure.

Legal and political issues

- The media law was accepted by the Parliament in 1996 without any consideration or reference to the new digital services. For the smooth introduction of digital services the appropriate paragraphs of the law should be changed. Now it cannot be forecasted, when the 2/3 majority of the votes could be collected for executing any (even this technical kind of) changes in the law.
- Fortunately, in Hungary there is a new Telecommunication Law under preparation. The Platform has reached (by active lobbying) that the new Law will consider some of the most important aspects of digital television.
- The participants of the Platform have come to mutual understanding that all the present terrestrial analogue broadcasting players should be granted free licences for simultaneous digital terrestrial transmission. It could prevent their resistance, and probably would raise the interest of the audience of analogue programmes for the digital services.

Practical measures

Taking into account the European situation and the above mentioned specific issues, the participants of the platform have prepared a document, in which they have summarized all the findings and conclusions of their study, and listed all the issues, which need to be further studied. The document has been sent over to all of the interested parties: the decision-makers, the broadcasters and the service-providers.

The reflection was rather diverse. The broadcasters were quite reserved (probably because of their financial situation), while the service providers, for example Antenna Hungaria were greatly enthusiastic about the launch of new digital services. The decision-makers: the Ministry of Communication and the

National Radio and Television Commission expressed their concern in establishing digital terrestrial services as soon as possible, as part of the development of information society.

In the mid of 1999 a DVB-T pilot project has been launched by Antenna Hungaria. Three digital TV-channels (the two public programmes of Hungarian Television and the satellite public programme of Duna-TV) have been transmitted in one multiplex by an 80 W power DVB-T transmitter. The coverage of the experimental transmission includes the centre and the south part of Budapest. There is a plan for the near future to increase the power of the existing transmitter, and also to put into operation a second transmitter. The two transmitters will form a Single Frequency Network, and feed all the territory of Budapest and its surroundings. The number of channels in the multiplex will also be expanded: one additional channel will serve test purposes, and another one will carry data.

The National Radio and Television Commission (in agreement with the Ministry) has requested the Communication Authority to start the planning and afterwards the international coordination of at least two nation-wide digital multiplexes. Now, the plans are ready, and the coordination of 2 multiplexes has already been started.

An expert-group was set up to prepare a proposal for the Government to provide the legal and administrative framework to the introduction of digital terrestrial television in Hungary, and consider a solution for the financial issues. The expert group has considered the following points to be included into the proposal:

The goal: The DVB-T must be introduced as soon as possible, because it will:
- lead to more efficient usage of the frequency-spectrum;
- promote new type of TV- and multimedia services;
- promote competition on the media market;
- make possible to cease the analogue services;
- make possible to utilize a part of the frequency band for new type of services.

The deadlines to be defined:
- When to announce the bid for the multiplexes and the digital programme-licenses (probably 2002 is the right date);
- When to launch the digital terrestrial services (the proposed date: first half of 2004);
- The final deadline to stop issuing analogue licences (possibly immediately!);
- The deadline to free up frequencies from military usage necessary to form the third digital MFN;
- The final date to stop analogue services (this is still a question).

Who will coordinate the introductory process?
- There is a proposal to establish a non-profit organisation for this purpose.

The multiplexes:
- How many of them, at the start, and afterwards (the proposal: minimum 3 at the start, afterwards according to the possibilities);
- Who will be (how to select) the multiplex providers?
- The content structure of the multiplexes (free and pay-TV channels, the policy in connection with the existing terrestrial channels).

Financial matters
- Who will finance the set up of the networks?
- Who should finance the transition period (simulcast cost)?
- What about the subsidising of STBs?

Technical matters
- transmission system (2k-8k, MFN-SFN),
- coverage of the digital networks;
- mobile reception;
- set-top-boxes (MHP, common interface);
- frequency re-use after the analogue switch over.

Of course, not all the questions which should be answered are listed above, and not all the questions listed above should be answered by the Government. The experts hope that all the important questions will be soon answered by competent institutions, otherwise the delay is unavoidable, and our position is getting worse, as it was noted earlier.

Conclusions

The possibilities, the sizes, the ranges and the deadlines can be different. Nevertheless, we in Hungary have come to the conclusion that we have to make efforts to keep up with the digital trend. And that is for the reason that we have understood the advantages of being able to participate in the most up-to-date technology exchange.

Finally, we, the fifteen million Hungarians speak a beautiful, but unique language. The media can be an excellent tool for protecting and developing our also unique cultural heritage and bringing the people closer to each other. If the digital terrestrial broadcasting can help us – and it does – open the door for a variety of new services, new tools, we have to grasp this opportunity, and to exploit it as much as possible.

22/02/2001 Budapest

MPEG4-like codec scheme for advanced multimedia communications

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The paper proposes a full scheme of encoding-decoding process of objects in video sequences thus following new concept of MPEG4 standard, representing a video sequence as a visual object stream. The codec is based on a spatio-temporal segmentation of a video sequence, which is deprived from our recent research on joint tracking of region based and mesh based models of video objects. At the beginning of the encoding process, the result of segmentation can be interactively selected and defined as a video object of interest. Then it is encoded in intra mode and sent to users at the other end of communication. For the next frames the object is encoded in a differential (inter) manner. The coding is based on a triangulated mesh model. During the encoding process, we deployed some enhanced algorithms together with several tools of MPEG4 to reduce the bit-rate of application. With computer-simulation we show that our complete scenario of codec does work with a good trade-off between bit-rate and image quality.

Introduction

As a successor of the MPEG standard family, MPEG-4 [1] inherits all the good properties of predecessors, offering the possibility of transmitting and/or storing a vast amount of data, which is necessary to represent digital images and video in an efficient and robust way. Going further than that, with MPEG4, an image is no longer viewed in a traditional rectangular frame. Video objects with arbitrary shape can be defined. The instances of the objects at each time called VOPs (Video Object Plans) are then encoded separately so that the private properties of objects are exploited, resulting in an efficient compression before passing them to a communication channel. At the decoder side, objects from multiplexed bit stream are retrieved. After decoded with proper tools, they are manipulated by a compositor. The operation of the compositor is partly controlled by end-users: they can interactively "add" / "delete" objects to / from the current view, fully interact with the received information, thus using advanced functionalities of an object based coding scheme. In MPEG-4 standard – comparing to other family members – a very first step for a new method of representing video objects is also specified, namely mesh-based model. Shape and even motion of general objects can be conveniently represented by a collection of polygonal patches (triangles or quadrangles). The geometry of meshes and the movements of the vertices of their polygonal patches (referred to as nodal optical flow) must be known by decoder, then texture inside each patch in the reference frame (object) is mapped on to the currently being decoded frame (object). Thus at the decoder side, the nodal positions, nodal optical flow and texture from the reference frame constitute the necessary information to construct the current.

Recently, a large amount of work [4-8,10] was devoted to the problem of mesh construction and nodal optical flow estimation. The two large families of approaches proposed for the choice of the node positions are: i) the regular triangulation into a set of equal triangles and ii) the Delaunay [3] triangulation of adaptively chosen set of nodes based on image content. The common property of these two solutions is that the topology of a mesh should not be transmitted to the decoder, as the triangulation is reconstructed in a unique way. As far as the motion estimation is concerned, the typical solutions are: polygonal matching [7], global optimisation of an affine optical flow [5,4], sub-sampling of dense optical flow generated locally or globally [10,13]. Once the nodes are chosen and the nodal optical flow is calculated, efficient coding methods have to be proposed for these two components for VLBR video communications. For encoding of the texture, several tools can be envisaged, DCT based MPEG schemes, polynomial surface fitting, wavelets...

In this paper we propose an efficient encoder-decoder both for the geometry, nodal optical flow and texture which follows the scheme of MPEG4 algorithm in its advanced version.

The novelty of the paper consists in an adequate choice of algorithms and developing of new ones for making an end-to-end solution for encoding of natural complex video objects in video sequences. The paper is organised as follows: section 2 gives a description of the whole coding scheme, section 3 presents the developed solution for a mesh-based encoder. With simulation results shown in section 4 we prove that our scheme offers a good quality of images in relative low bit-rate required. The paper is closed with our conclusion on the obtained results. The perspective of further improvement in the proposed scheme is also exposed in section 5.

Structure of encoder-decoder

The general scheme of the proposed encoder-decoder is shown at the figure 1. The first phase is a spatio-temporal segmentation of the sequence [9, 11]. Then regions with homogenous motion detected from segmentation phase are offered to end user (an author, composer) so that he / she can link them together according to their logical meaning, that is declares several objects possessing one or more regions. Then the triangulated mesh for each separate object is constructed in such a way, that the resulting mesh for each object is a union of meshes of each region of the object. Then the mesh is conveyed to the mesh-based encoder. The tracking phase of the whole segmentation of video frames, which updates regions' shape and their motion, allows updating the mesh along the time. The encoder thus uses the mesh in intra mode (mesh construction block in figure 1) or in inter frame mode (mesh update block in figure 1) to encode the moving object in video sequence and sends it to the decoder. We call the scheme mesh-based because the communication between encoder and decoder uses mesh-based representation of an object.

In current framework, we deal with luminance component of video sequences. The chrominance values can be processed in the same manner.

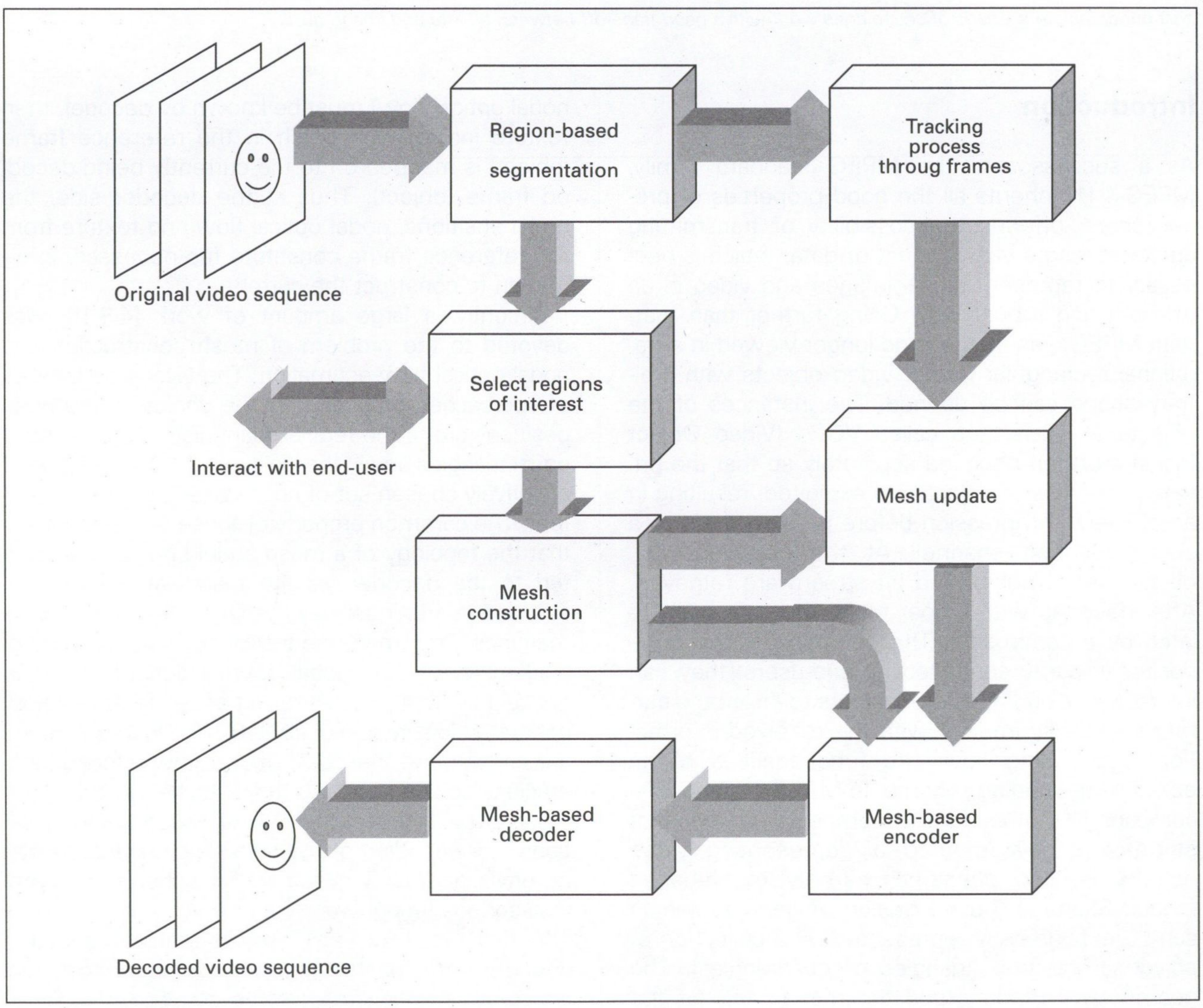


Figure 1 General structure of encoder decoder

In this paper we focus on the mesh-based encoder-decoder, as the segmentation, tracking and mesh construction and update methods were developed in previous work and can be found from [9,11,13]. Nevertheless, the region-based model issued from the segmentation process and the mesh construction process will be briefly described bellow as they are necessary for the comprehension of the mesh encoding method we develop in this paper.

Mesh-based codec

The mesh-based encoder processes the data resulted from the mesh construction and tracking phase so it uses the mesh model operated by these two phases. We refer the reader to [10, 13] for the detailed description of these two phases and briefly introduce the mesh model here.

Triangulated mesh model and construction for articulated video objects

In [9] we proposed a method for a spatio-temporal segmentation of video sequences. This segmentation is a partition of a video frame into a set of polygonal regions, each of them being characterised by an affine 4 parameter motion model. Thus in each pixel of a given region the elementary displacement vector can be presented as

$$\mathbf{d}(x,y)=\begin{pmatrix} dx(x,y) \\ dy(x,y) \end{pmatrix}=\begin{pmatrix} tx \\ ty \end{pmatrix}+\begin{pmatrix} k & \theta \\ -\theta & k \end{pmatrix}\begin{pmatrix} x-x_g \\ y-y_g \end{pmatrix}$$

Here k is a zoom factor, q is a rotation angle and tx and ty are translation parameters; xg,yg are the co-ordinates of the gravity center of a region. The method presented in [9] allows to obtain several motion models for one region characterised by a decreasing quality of a global motion compensation per region.

A set of regions selected by the user at the beginning of the sequence as a VOP is used for mesh construction. This articulated structure of the VOP is maintained in the mesh model of it: the mesh of a VOP is a set of Delaunay constrained meshes (one for each region) with the continuity of these partial meshes on the region borders.

The whole mesh construction process includes three steps: simplification of polygonal borders of regions, Delaunay triangulation, computation of the nodal optical flow. The illustration of this process is given in figure 2.

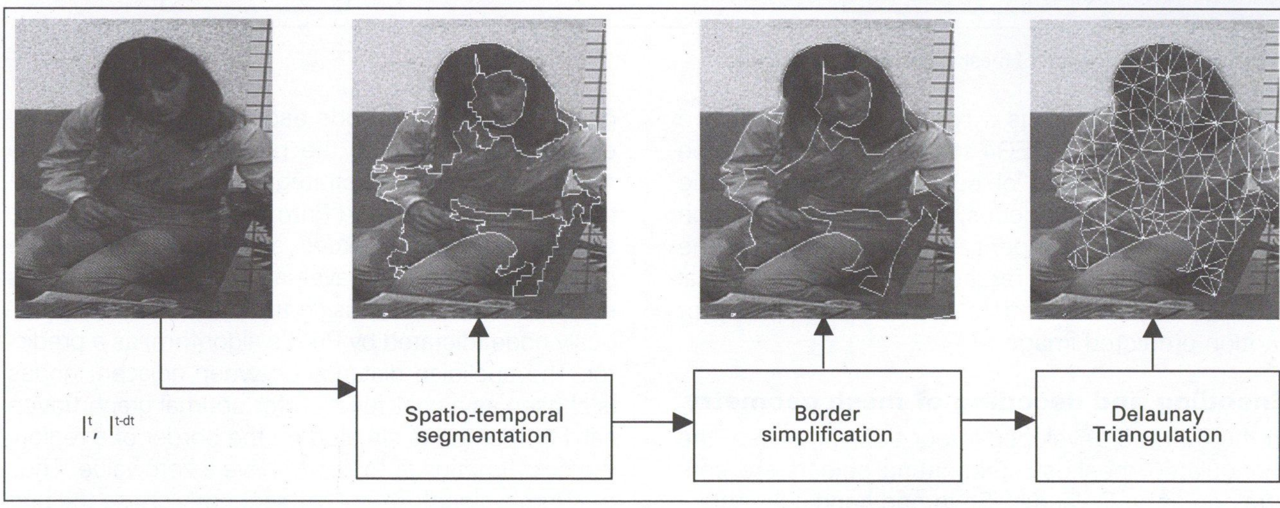


Figure 2 General structure of encoder decoder

Border simplification is needed to avoid too small triangles yielding aliasing effects. It consists of iterative removal of closest border points according to the minimum distance threshold.

Delaunay triangulation is computed for each polygon obtained after the border simplification, making use of the reference software proposed in [12]. The triangulation is constrained which means that all polygonal vertices of a region border are included into a final set of triangulation nodes. Furthermore, the border of a region is also a border of the triangulation. The basic algorithm inserts internal nodes in each region to satisfy the Delaunay properties, that is a circle circumscribed around each triangle does not contain vertices of other triangles.

The computation of nodal optical flow uses the whole set of affine models associated with each region. The nodal displacement vector for a given node at the position (x,y) is computed by the formula (1). If several motion models are available for the same region, then the optimal nodal vector is retained minimising local motion compensation error (see details in [11, 13]). For the nodes situated on the borders of the regions, the nodal motion vectors are computed as a liner combination of nodal vectors obtained from motion models of adjacent regions sharing the given node.

The mesh constructed for a video object in the first frame, is tracked along the time. During the tracking process for each node two motion vectors are computed; a "backward" motion vector and a "forward" motion vector. The backward nodal optical flow serves to predict a video frame at a current time instant from a reference frame. The forward optical flow allows to obtain the position of mesh nodes in the next frame, knowing their position in the current frame. It has to be mentioned that in the actual work the number of nodes per region in a VOP is constant along the time.

General scheme of mesh coding

For each frame, three kinds of information are sent to decoder as seen in Figure 3. They are nodal positions (or their differences), nodal motion vectors and spatial-temporal textures of the mesh.

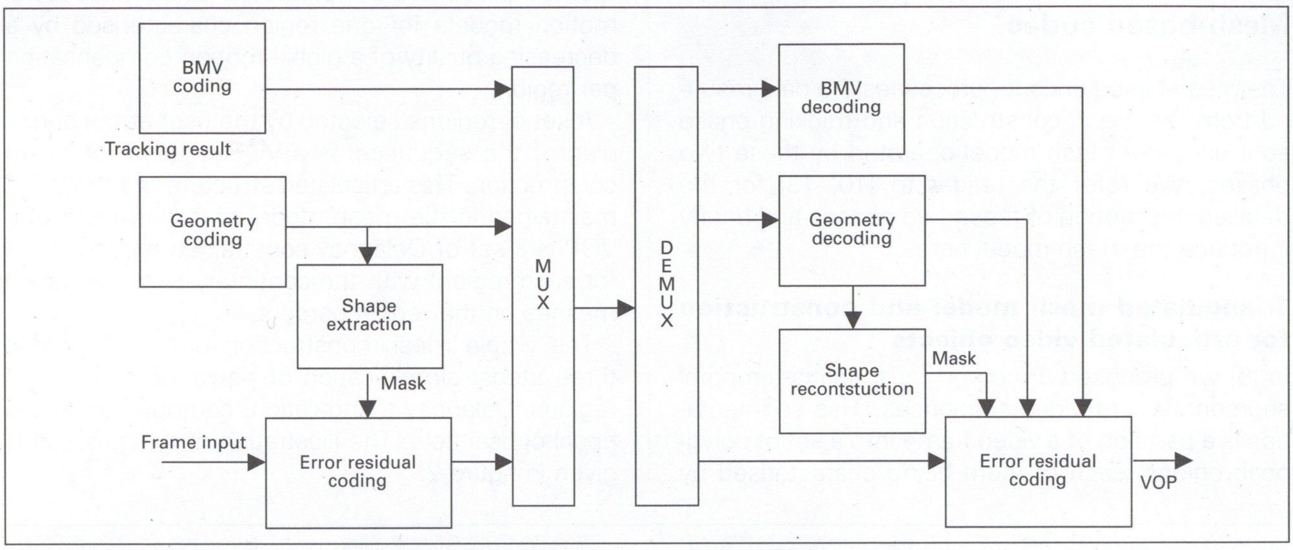


Figure 3 Block diagram of mesh-based encoder decoder

This block diagram is a typical for the predictive coding scheme in MPEG4 standard. Here the shape information for video objects is conveyed by the geometry of mesh nodes, the backward motion vectors serve to compensate current frame at the decoder from the reference frame, the residual error coding is needed to improve the quality of motion predicted image.

Encoding and decoding of mesh geometry

In the method developed in our tracking phase, the triangulated meshes representing objects are constrained to be Delaunay. So in one hand it inherits a good property of Delaunay triangulation, i.e. the mesh is unique and can be computed by both encoder and decoder, given the positions of node points; therefore the topology of the mesh does not need to be encoded. In another hand transmitting the geometry of constrained Delaunay mesh, we implicitly convey the shape information to decoder. The alpha channel becomes redundant in out scheme. Detailed process of geometry codec is depicted in Figure 5 and Figure 6.

In the encoding of the geometry, two distinct steps are needed: intra-frame coding of the nodal positions in the first triangulated frame and the inter-frame coding. For the intra-mode we propose to use an optimal graph traversal to encode each node position differentially with regard to the previous one. Here the minimal spanning tree strategy, precisely Prim's algorithm [14] is deployed ("Prim reordering" box at the encoder and "Prim order extraction" box at the decoder). During this traversal, each node position is encoded differentially using the position of the previously node (dictated by Prim's algorithm) as a predictor. The euclidian distance between adjacent nodes is chosen as a cost function for optimal graph traversal. For the nodes situated on the border of a region, the cost function is forced to have a zero-value. Thus the Prim's algorithm will consequently chose the border nodes as the first path in the spanning tree (see Figure 4, bold line)

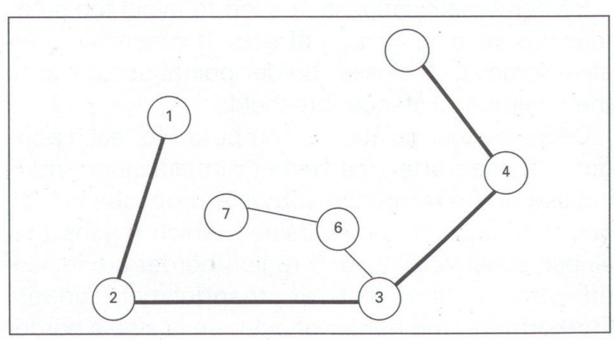


Figure 4 Prim's travelsal with forced border nodes

The encoder is creating the bit-stream for the geometry following the constructed tree and LIFO ordering. Thus when a node with a degree higher than 2 is met, it will insert a specific flag T. When the end of the branch is encountered than it will insert the end-flag F. The end of the bit-stream is shown with a flag B. The co-ordinates of the first mesh-node in the polygon border will be encoded in an absolute manner.

The bit-stream for the example shown in the Figure 4 will be as follows:

(X1,Y1),(DX2,DY2),(DX3,DY3),T,(DX4,DY4),
(DX5,DY5),F,(DX6,DY6),(DX7,DY7)FB.

Here as (DX6,DY6) follows the F flag the co-ordinates are encoded differentially with regard to the last point which is encoded just before the farthest T flag from the current F flag (LIFO ordering). This intra-frame coding of the geometry is performed only once for the first frame.

In inter VOP the geometry temporal differences are encoded. They are treated as forward motion vectors (FMV) of mesh nodes and are encoded in the same manner as backward motion vectors (BMV), i.e. the "FMV encoder" box in Figure 5 and the "FMV demultiplex, reconstruction" box in Figure 6 are actually the BMV encoder and decoder in Figure 7 and Figure 8. The temporal geometry differences – FMV – are deprived as the following before being sent to geometry encoder:

$$\mathrm{FMV}_X^i(t) = N_X^i(t+1) - N_X^i(t)$$
$$\mathrm{FMV}_Y^i(t) = N_Y^i(t+1) - N_Y^i(t) \qquad (2)$$

where i, X, Y and t stand for index of node, horizontal, vertical component and point of time respectively.

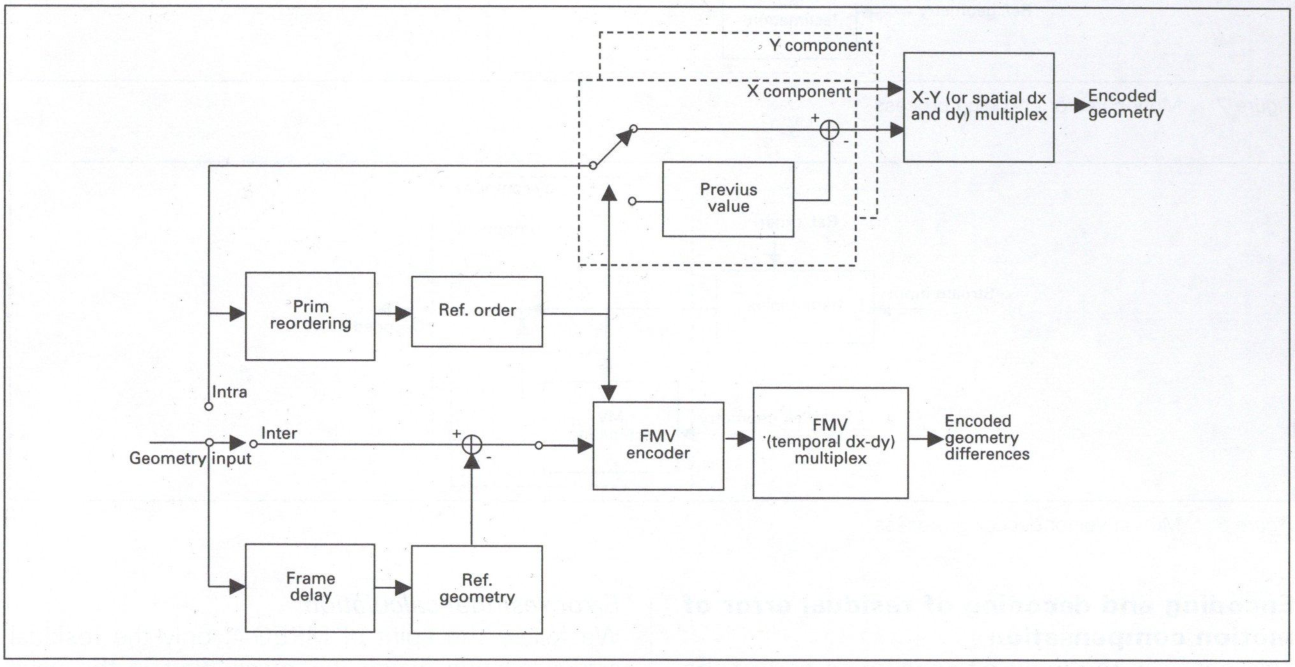


Figure 5 Geometry encoding process

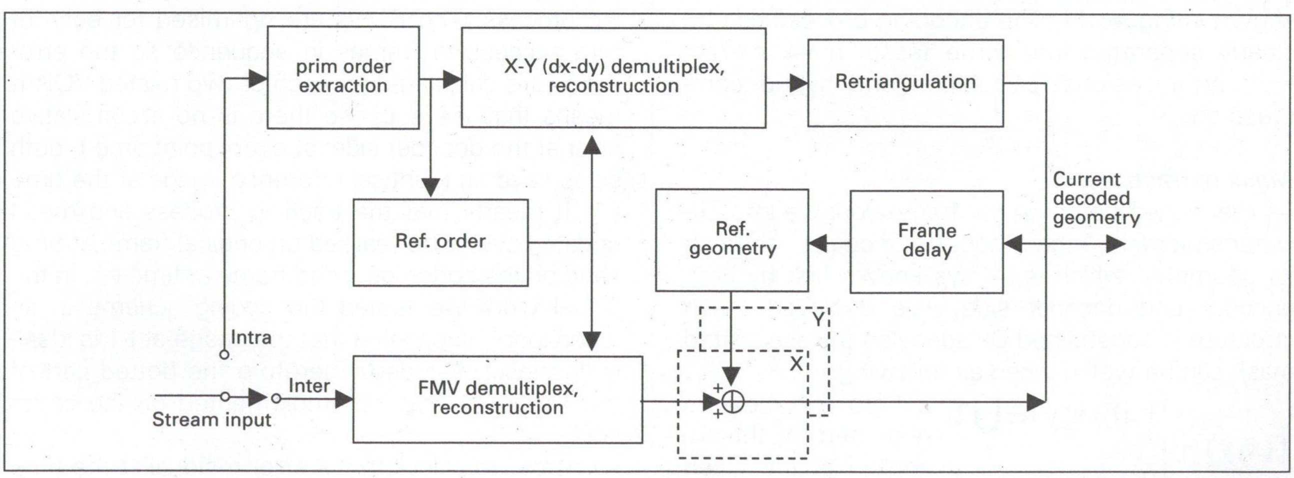


Figure 6 Geometry decoding process

Encoding and decoding of backward motion vectors

Every node of the mesh has its own BMV obtained as a result from tracking process. To reduce the amount of transmitted data for BMV, they are encoded in predictive manner. For a given triangle, knowing motion vector V_1, V_2 of its two nodes, the motion vector (MV) V_3 of the third node can be predicted as [6]

$$\tilde{V}_3 = \left(\frac{V_1}{D_{13}} + \frac{V_2}{D_{23}}\right) / \left(\frac{1}{D_{13}} + \frac{1}{D_{23}}\right) \quad (3)$$

with D_{ij} is the distances between node i and j.

Then only the prediction error is transmitted to decoder. In order not to transmit any information about "which MV correspond to which node", the order obtained from the geometry encoding phase is reused (it is referred as "Ref. Order" signal at both encoder and decoder side in Figure 7 and Figure 8)

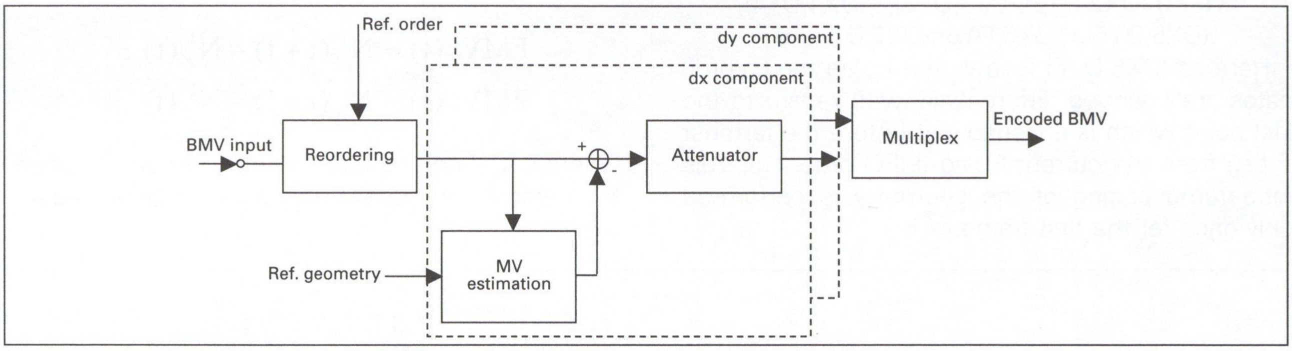


Figure 7 Motion vector encoding process

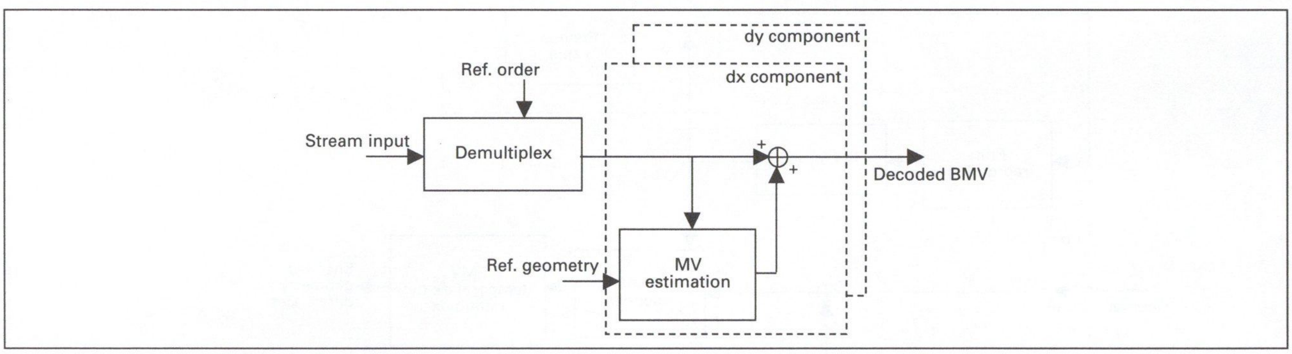


Figure 8 Motion vector decoding process

Encoding and decoding of residual error of motion compensation

This phase is the most bit-consuming part of the whole transmission. Its detailed implementation is shown in Figure 11. The encoding process can be clearly separated into three tasks: mask extraction, error residual calculation and hybrid compression.

Mask extraction

As mentioned before in our framework we implicitly transmit mask – the silhouette of object – through its geometry, which is always known first by both encoder and decoder side. The deployed mesh structure is constrained Delaunay so the associated mask can be well defined as following:

$$f(x,y) = \begin{cases} 1 \text{ if } (x,y) \in \bigcup_i T_i \\ 0 \text{ otherwise} \end{cases}$$

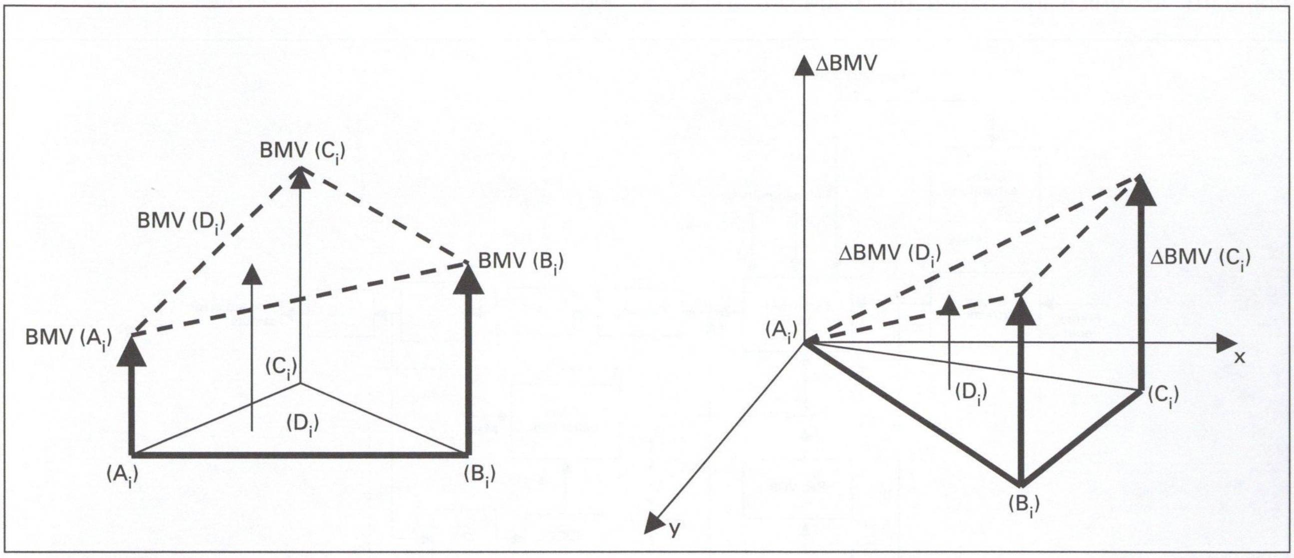
,where Ti is the area restricted by the triangle i of the given mesh.

Error residual calculation

We follow the spirit of MPEG-4: only the residual errors locating inside the mask (that is the pixel belongs to the object) become the targets of a coding process. The BMVs are optimised for each of two successive frames in sequence so the error residual is calculated for each of two related VOP. It means that we suppose there is no accumulative error at the decoder side: at every point time t, both sides have an identical reference image at the time t-1. It means that the tracking process and mesh update have to be realised on original frame at time t and on the coded-decoded frame at time t-1. In the actual work we tested the coding scheme in an "open loop", supposing that the image at t-1 is ideally (lossless) decoded. Therefore the dotted part of the Figure 11 was not implemented on the coder side.

In order to calculate the error residual at the time t, BMV for each pixel of the object is deprived from interpolating the known nodal BMVs, based on available triangle geometry at the time t-1 in bilinear manner. Referring to Figure 9, every general case of BMVs for arbitrary triangle in (A) can be reduced to (B), in which the formula for interpolated DBMV(D_i) can be written shortly as below:

$$\Delta BMV(D_i) = \frac{\Delta BMV(B_i)\cdot[x(D_i)y(C_i) - x(C_i)y(D_i)] + \Delta BMV(C_i)\cdot[x(B_i)y(D_i) - x(D_i)y(B_i)]}{[x(B_i)y(C_i) - x(C_i)y(B_i)]} \quad (4)$$



(A) General case: calculate unknown BMV for pel D_i, given three BVMs of pixels A_i, B_i and C_i located three summits of the support triangle

(B) Special case: A_i is positioned at the center of the coordinate system. $\Delta BMV(X_i) = BMV(X_i) - BMV(A_i)$ where $X \in \{B,C,D\}$

Figure 9 Bilinear interpolation for every BMVs inside triangle i of the mesh

Applying (4) to horizontal and vertical component of BMV, the interpolated BMV is obtained. With just calculated BMV, a pixel from time t can refer to a non-integer pixel at the time t-1. Therefore another interpolation is necessary.

In the second interpolation, for each pixel in the current VOP at time t, the bilinear interpolation is applied again to calculate its estimated intensity value. At this time the interpolated value can be calculated as the following:

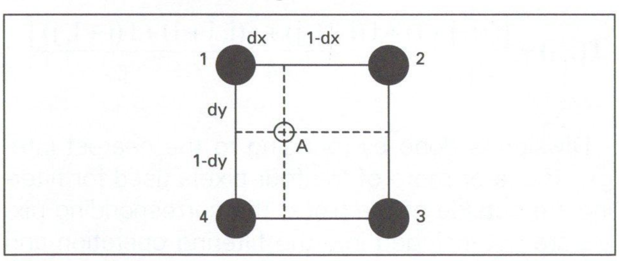


Figure 10 Bilinear interpolation for pixel's intensity with non integer coordinates

$$I_{Inter.}(t) = dx \cdot dy \cdot I_1(t-1) + (1-dx)\cdot dy \cdot I_2(t-1)$$
$$+ dx \cdot (1-dy)\cdot I_3(t-1) + (1-dx)\cdot(1-dy)\cdot I_4(t-1) \quad (5)$$

where I_1, I_2, I_3, I_4 are the intensity of the closest four pixels to the position A referred to by the BMV of the pixel whose interpolated value is being calculated (see Figure 10). Then the residual error for every pixel is defined by

$$E^{x,y}(t) = I^{x,y}(t) - I^{x,y}_{Inter.}(t)$$

where the superscript x and y are the pixel's horizontal and vertical coordinates respectively.

This error signal inside a VOP will now be encoded using DCT-based method (DCT-block in the scheme), quantised (Q-block) and lossless coded with a Variable Length Coder (in actual work we do not implement it and limit ourselves with entropy assessment of the bit-rate). On the blocks which cover the borders of a VOP padding techniques will be applied as described bellow (Padding block in Figure 11).

At the decoder side, the geometry information is decoded first. If the current VOP is inter one, the results of geometry decoder process – they are FMV – are used to adjust to position of nodes in the reference geometry; otherwise – in intra VOP – the nodal positions for current frame themselves are obtained. Then two interpolations are processed in the same way as at the encoder side with the exception of the last operation in the second interpolation: the intensity of current pixel not the error residual is computed as the following:

$$\hat{I}^{x,y}(t) = E^{x,y}(t) + I^{x,y}_{Inter.}(t)$$

Hybrid compression

Compression task is necessary for a low bit consuming transmission of error residual. Here we use the hybrid compression method proposed by MPEG standard, in which intensities of image are transformed into spectral domain by using DCT transformation, the resulting coefficients are then quantised with weighting matrix and scale factor before being sent to decoder. The object of hybrid compression task is not the conventional rectangular frame but an arbitrary shaped VOP, therefore it necessitates the Padding function proposed by MPEG-4 [2] (see Figure 11)

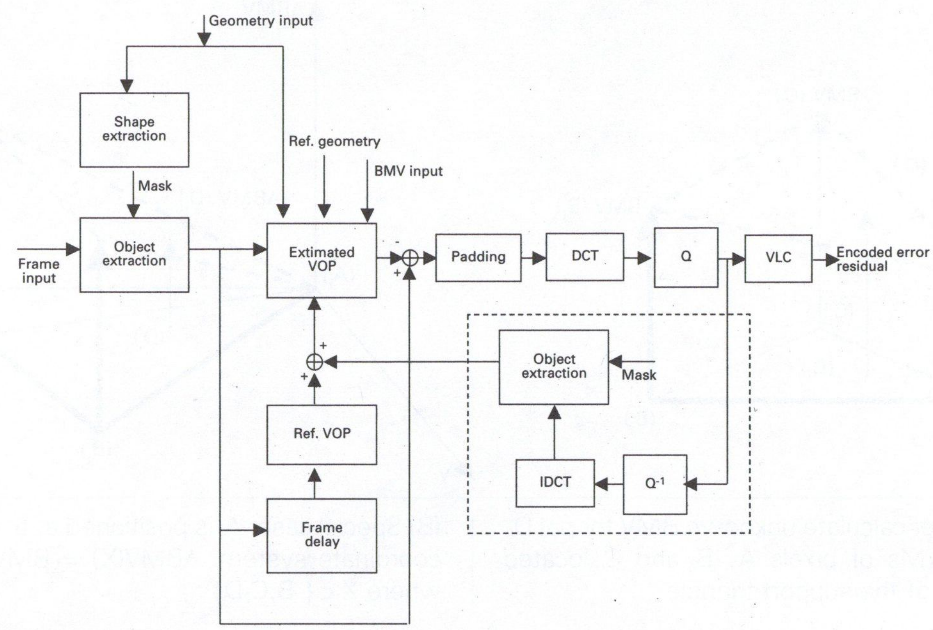


Figure 11 Error residual encoding process

In the present work we test the efficiency of three methods that MPEG-4 proposed for texture coding inside an arbitrary shape and applying to bounding blocks: padding with zero, low pass extrapolation padding technique and shape adaptive DCT (SA-DCT)

● Padding with zero: it is a simplest way mostly applied to error residual encoding. The values of those pixels in bounding blocks, which do not belong to the object, are set to zero. Then these blocks are coded in a manner identical to the interior blocks.

● Low pass extrapolation padding technique: the method is applied to each intra block that has at least one transparent and one non-transparent pixel in its associated shape (alpha) information. The padding is performed in three steps:

1. Calculate the arithmetic mean value m of all block pixels f(i, j) situated within the object region R

$$m = (1/N) \sum_{(i,j)\in R} f(i,j)$$

where N is the number of pixels situated within the object region R, f(i, j) is the value of pixel at the position (i, j). Division by N is done by rounding to the nearest integer.

2. Assign m to each block pixel situated outside of the object region R, i.e. f(i, j) = m for all (i, j) $\notin$ R.

3. Apply the following filtering operation to each block pixel f(i, j) outside of the object region R, starting from the top left corner of the block and proceeding row by row to the bottom right pixel

$$f(i,j) = \frac{[f(i,j-1) + f(i-1,j) + f(i,j+1) + f(i+1,j)]}{4}$$

Division is done by rounding to the nearest integer. If one or more of the four pixels used for filtering are outside of the block, the corresponding pixels are not included into the filtering operation and the divisor 4 is reduced accordingly.

● SA-DCT: the algorithm [2] is based on the predefined orthonormal sets of DCT basis functions. The basic concept of the proposed method is outlined in Figure 12 for coding an arbitrary shaped image foreground segment contained within an 8x8 reference block. Figure 10-A shows an example of an image block segmented into two regions, foreground (grey) and background (light). To perform the vertical transform of the foreground, the length (vector size N, 0<N<9) of each column j (0<j<9) of the foreground segment is calculated, and the columns are shifted and aligned to the upper border of the 8x8 reference block (Figure 10-B). Dependent on the vector size N of each particular column of the segment, a one-dimensional DCT with a transform kernel $\underline{\mathrm{DCT\text{-}N}}$ containing a set of N basis vectors is selected for each particular column and applied to the first N pixels of the column. Thus, the N vertical DCT-coefficients cj for each segment column data $\underline{x}_j$ are calculated according to the formula:

$$\underline{c}_j = \sqrt{\frac{2}{N}} \cdot \underline{\mathrm{DCT-N}} \cdot \underline{x}_j$$

$$\text{with} \quad \underline{\mathrm{DCT-N}}(p,k) = c_0 \cdot \cos\left[p \cdot (k+0.5) \cdot \frac{\pi}{N}\right] \quad \text{and} \quad c_0 = \sqrt{1/2} \quad \text{if} \quad p=0; \quad c_0 = 1$$

$$\text{otherwise} \quad \text{for} \quad 0 \le p,k \le N-1$$

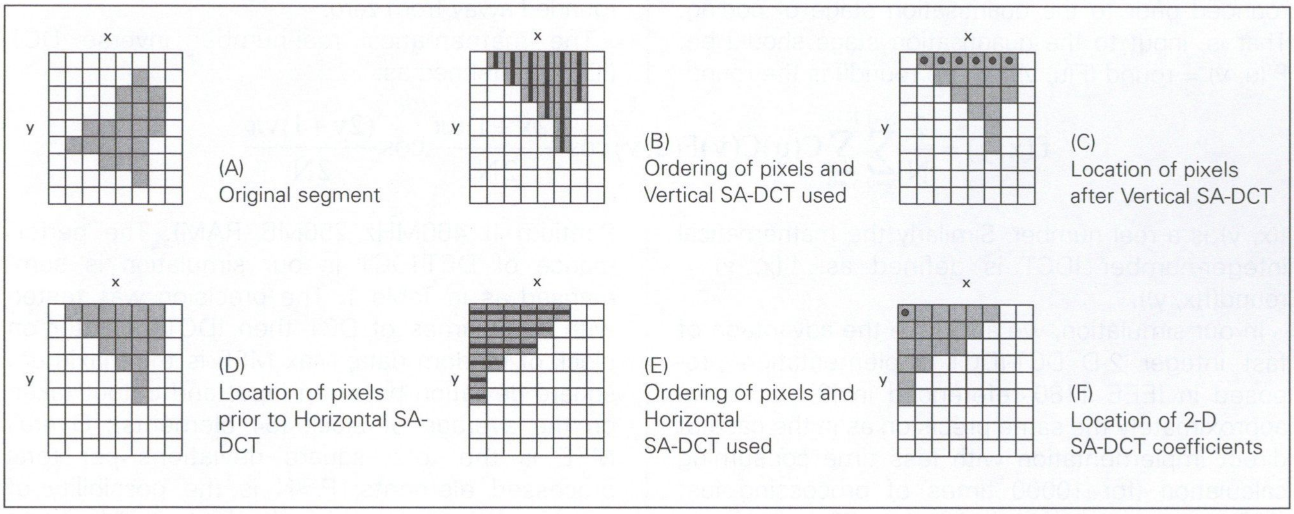


Figure 12 Successive steps involved for performing a SA-DCT forward transform on a block of arbitrary shape

For example, in Figure 10-B the right most column is transformed using DCT-3 basis vectors. After SA-DCT in vertical direction, the lowest DCT-coefficients (DC values •) for the segment columns are found along the upper border of the 8x8 reference block (Figure 12-C). To perform the horizontal DCT transformation (Figure 12-E), the length of each row is calculated, and the rows are shifted to the left border of the 8x8 reference block, and a horizontal DCT adapted to the size of each row is then calculated using the above formulas. Note that horizontal SA-DCT-transformation is performed along vertical SA-DCT-coefficients with the same index (e.g. all vertical DC coefficients (•) are grouped together and are SA-DCT-transformed in horizontal dimension). Figure 12-F shows the final location of the resulting DCT coefficients within the 8x8-image block.

In this algorithm, the final number of DCT-coefficients is identical to the number of pixels contained in the image segment. Also, the coefficients are located in comparable positions as in a standard 8x8 block. The DC coefficient (•) is located in the upper left border of the reference block and, dependent on the actual shape of the segment, the remaining coefficients are concentrated around the DC coefficient.

Since the nodal geometry and therefore the shape is decoded prior to this process, the decoder can perform the shape-adapted inverse DCT as the reverse operation in both horizontal and vertical segment direction on basis of decoded binary alpha block (BAB) data:

$$\underline{x}_j = \sqrt{\frac{2}{N}} \cdot \underline{\mathrm{DCT-N}}^{T} \cdot \underline{c}_j$$

Here $\underline{c}_j$ denotes a horizontal or vertical coefficient vector of size N.

In our simulation, default weighting matrix proposed for intra and inter frame in both MPEG-2 and MPEG-4 standard are involved [2, 15]. The quantiser-scales are restricted to the predefined values in these standard. Referring to MPEG-4 for non-intra block, the quantising and dequantising operators are based on the following relation:

$$F(x,y) = \frac{\{2 \cdot QF(x,y) + \mathrm{Sign}[QF(x,y)]\} \cdot W(x,y) \cdot \mathrm{quantiser\_scale}}{32}$$

where QF(x, y) is the quantised value of F(x, y) at the position (x, y) in block pixel; W(x, y) is the element at the position (x, y) in weighting matrix.

For DCT calculation, a separable 2-dimensional Discrete Cosine Transform is used. The NxN two-dimensional DCT is defined as:

$$F(u,v)=\frac{2}{N}C(u)C(v)\sum_{x=0}^{N-1}\sum_{y=0}^{N-1}f(x,y)\cos\frac{(2x+1)u\pi}{2N}\cos\frac{(2y+1)v\pi}{2N}$$

with u, v, x, y = 0, 1, 2, ... N-1; where x, y are spatial coordinates in the sample domain; u, v are coordinates in the transform domain

$$C(u),C(v)=\begin{cases}\frac{1}{\sqrt{2}} & \text{for } u,v=0\\ 1 & \text{otherwise}\end{cases}$$

The results of the forward DCT calculation is rounded prior to the quantisation stage of coding. That is, input to the quantisation stage should be: F'(u, v) = round (F(u, v)) where round() is the rounding to the nearest integer, with half-integer values rounded away from zero.

The mathematical real-number inverse DCT (IDCT) is defined as:

$$f(x,y)=\frac{2}{N}\sum_{x=0}^{N-1}\sum_{y=0}^{N-1}C(u)C(v)F(u,v)\cos\frac{(2x+1)u\pi}{2N}\cos\frac{(2y+1)v\pi}{2N}$$

f(x, y) is a real number. Similarly the mathematical integer-number IDCT is defined as: f'(x, y) = round(f(x, y)).

In our simulation, we also take the advantage of fast integer 2-D DCT-IDCT implementation proposed in IEEE 1180 referenced in [2]. It shows approximately the same precision as in the case of direct implementation with less time consuming calculation (for 10000 times of processing fast DCT on a random block, it takes 200ms against 3400μs of direct DCT in average, the test run on Pentium II 466MHz 256MB RAM). The performance of DCT-IDCT in our simulation is summarised as in Table 1. The precision was tested with 1000 times of DCT then IDCT operator on block of random data; Max MSE is the maximum square deviation between input and output taken on the average of block (64 elements); Overall MSE is the total square deviations per total processed elements; P_e>N is the possibility of error occurrence, in which absolute error deviation is greater than N.

	Max MSE	Overall MSE	Pe>0 Pe>1,2,3
Direct 2D-DCT	0,171875	0,083406	0,0834063 0
Fast 2D-DCT	0,203125	0,083672	0,0836719 0
1D-DCT with 1 element	0	0	0 0
1D-DCT with 2 elements	0,5	0,081	0,081 0
1D-DCT with 3 elements	0,333333	0,083333	0,0833333 0
1D-DCT with 4 elements	0,25	0,128	0,128 0
1D-DCT with 5 elements	0,4	0,0784	0,0784 0
1D-DCT with 6 elements	0,5	0,089833	0,089833 0
1D-DCT with 7 elements	0,428571	0,085	0,085 0
1D-DCT with 8 elements	0,375	0,082375	0,082375 0

Table 1 Precision of DCT and IDCT in our simulation

Simulation results

The whole coding-decoding method we proposed in this paper was tested on a video sequence "Interview" (see Figure 2), which represents cut frames of size 256x256 pixels in CCIR601 format at 25 ips rate. The results of geometry coding by the Prim's traversal are shown in Figure 13. To show the efficiency of optimal Prim traversal with regard to an arbitrary traversal ("depth search") we supposed all frames to be intra-coded, so the geometry of the mesh is retransmitted for each frame. The entropy gain is of 0.01 in average.

In coding of MVs, our experiments showed that even with the roughest precision (with value 1), FMVs are losslessly encoded. It is due to the fact that FMVs with integer value in nature are still coarser than the coarsest precision guaranteed by our MV encoder. In order to select a relative good precision for BMVs (they are double valued), we

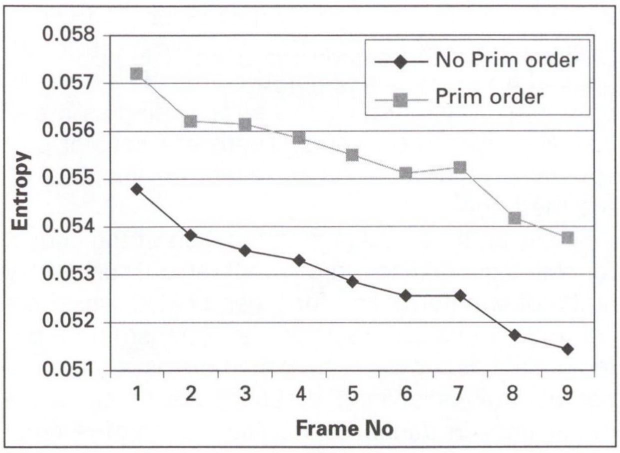


Figure 13 Effect of Prim algorithm on geometry compression

measured the PSNR right after the motion estimation phase at the decoder side, therefore the loss in error residual transmission is completely excluded, the PSNR exclusively reflects the impact of precision choice (the error residuals also contribute to this PSNR value, but they are constant during the selection of precision, i.e. they no longer play role in the change of the measured PSNR). Several precisions were tested in Figure 14. We consider precision = 10 as a proper value for our case.

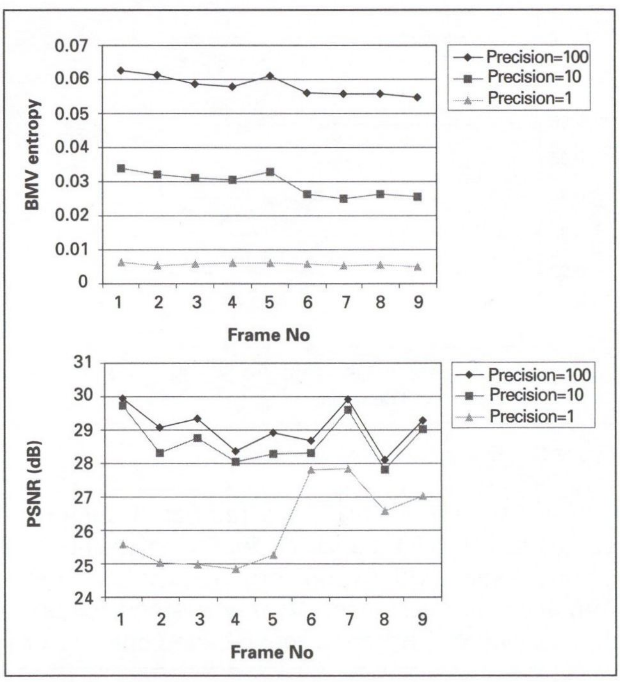


Figure 14 Transmission BMVs with different precisions, PSNR is computed after compensation by motion

In coding of residual motion-compensation error, we tested several padding methods proposed in MPEG-4 for coding blocks situated at objects' boundary. LPE padding for intra bounding block is also involved. Their performance is shown in PSNR and entropy of error residual, which will be send to receiver. In Figure 15 the entropy of error residual after processing DCT with selected padding method is shown. The calculated coefficients are then quantised and transmitted to receiver. In the quantising phase, we applied the default MPEG-4 inter weighting matrix with the quantiser-scale = 1 (quantiser scale then has less effect on overall PSNR). At the decoder side, receiving lossless BMV and geometry, we measure the PSNR of the reconstructed VOP to see the effect of the given padding method. Figure 15 shows that SA-DCT method gives the best compression of error residual with even slightly higher PSNR. We choose this method in our final scheme.

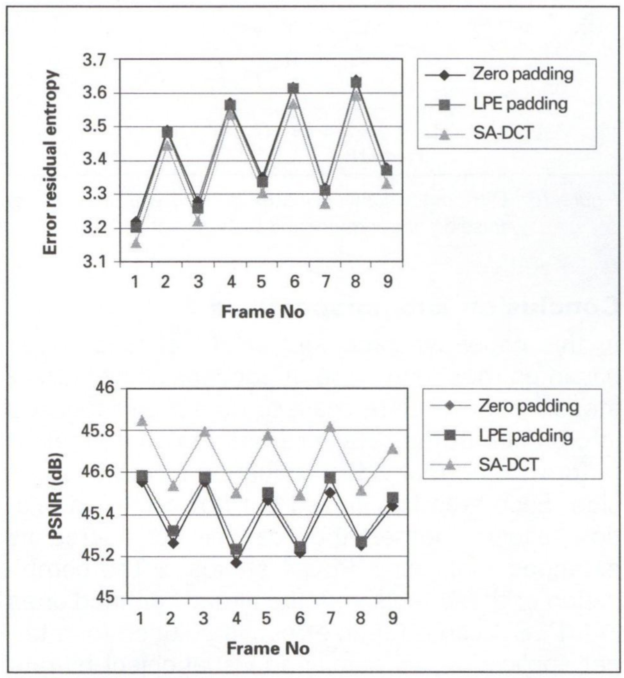


Figure 15 Padding methods with ideal transmission of geometry, error residual is encoded with MPEG-4 inter weighting matrix, quantiser scale = 1

Next step is to select weighting matrix and quantiser-scale for compression. Several default weighting matrix proposed in MPEG standard were deployed with quantiser-scale=1 see Figure 16. Here we used the same method as before to exclude other factors but weighting matrix to the final PSNR calculated at the decoder side: motion vectors are identical at encoder and decoder.

Finally, selection of quantiser scale is processed. The values of 4, 5 and 6 were tested. As seen in Figure 17, with quantiser scale = 5 a good compromise between entropy and PSNR is obtained. This value was finally retained for the quantising.

Figure 18 and Figure 19 show the overall performance of the whole scheme with the previous selections in parameters: transmission with BMV precision=10, SA-DCT, geometry transmission with prim algorithm, weighting matrix is MPEG-4-inter and quantiser scale is 5.

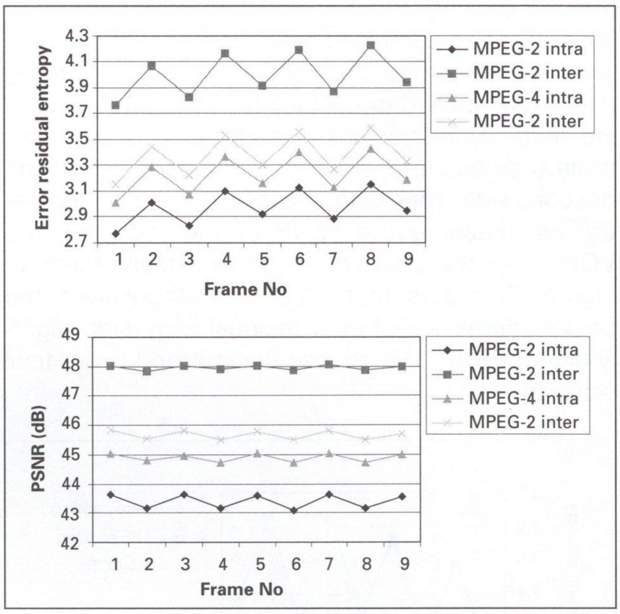


Figure 16 Different weighting matrix is used with ideal transmission of geometry and BMV, quantiser scale = 1

Conclusion and perspectives

In this paper we proposed a full scenario codec based on mesh structure. It contains all necessary steps for a complete chain to obtain object-based information at the receiver from a raw information in form of normal video sequence at the encoder side. Each step has closed but not exclusive relation to one another; they can be considered as extended tools for MPEG-4 standard. The combination of these tools with the already defined ones in MPEG-4 can offer an effective solution for a target application. The obtained visual object bit-rate

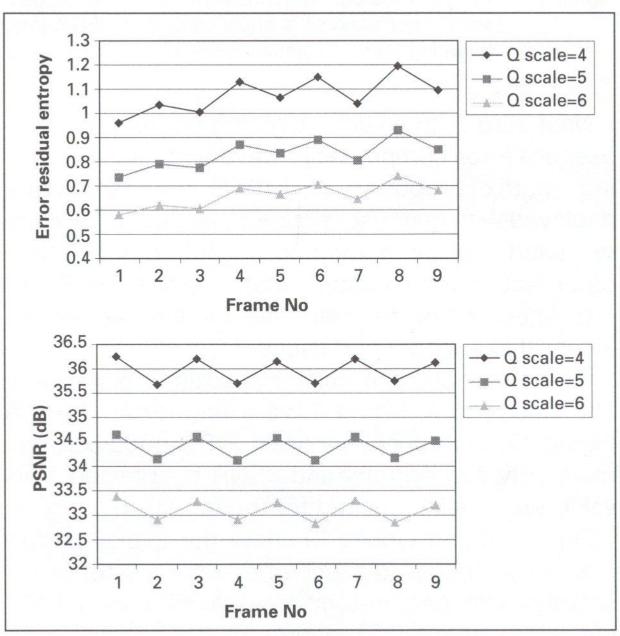


Figure 17 MPEG-4 inter weighting matrix with several quantiser scale, ideal transmission of geometry and BMV are used

is now closed to 0.6 Mbit/sec for CCIR 601 sequence at the full PAL/SECAM frame rate of 25 ips and a complex articulated object. The most of bit-rate is consumed by the error coding as it can be seen from the results. There are several perspectives to reduce the total bit-rate supplied by the method.

Firstly, in the actual implementation of the codec, for object containing more than 2 regions, geometry and motion information for those points, which are common between regions, is retransmitted per region, that is repeated as many time as is the number of regions sharing these points. It causes a redundancy in transmission. The reason of retransmission is that our tracking algorithm is based strictly on moving region, and for the simplicity we use triangulating method applied to single region when generating mesh for objects. The improvement is under development.

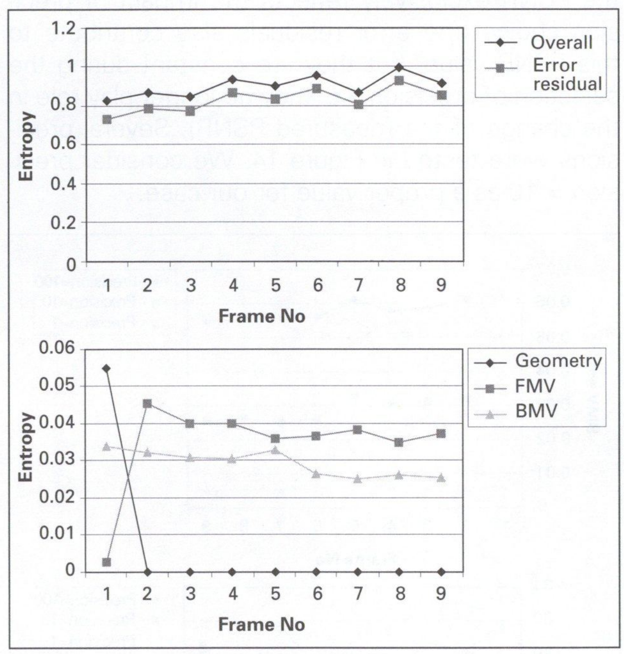


Figure 18 Entropy performance of codec

In error compensation, we restricted ourselves strictly to MPEG-4 standard: deploying pixel-based compensation with hybrid compression DCT transformation and quantiser. A new method for error compensation – we call it mesh-based one – is also

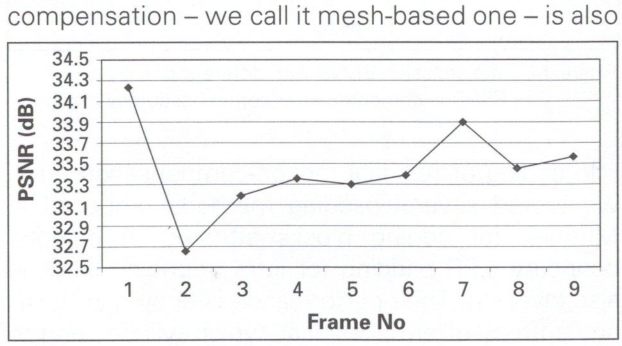


Figure 19 PSNR performance of codec

being developing in our laboratory with prospective results. The coding of error can be adapted per each triangle in a mesh with a finer coding in the triangles situated in critical areas, such as region junctions, and a rougher coding in flat areas inside regions.

Acknowledgement

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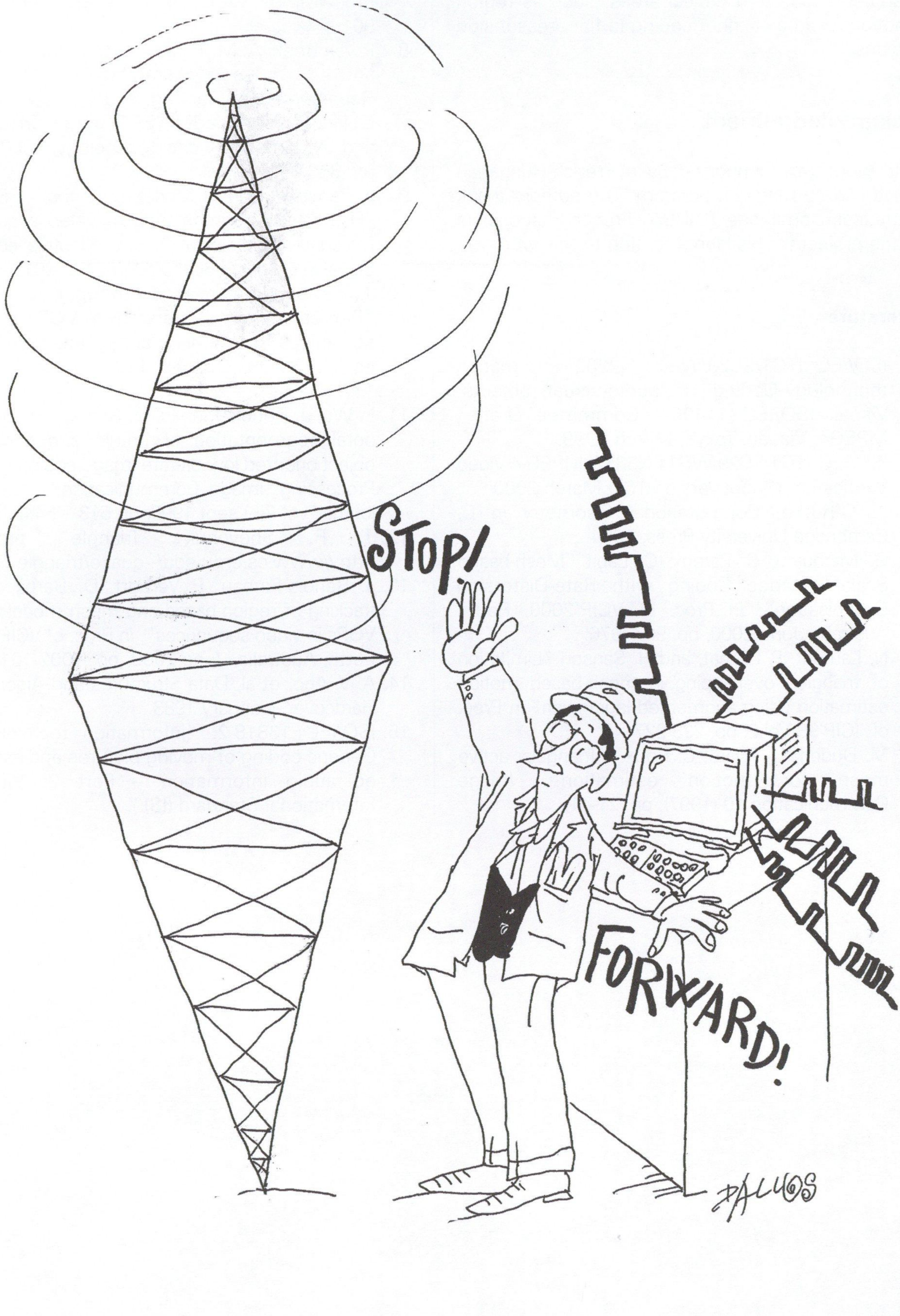
STOP!
FORWARD!
DALLOS

Statistical analyses used for e-mail reader development and enhancement

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Introduction

As a consequence of the information bang, people want to access various information sources by using all devices available to them. There is a demand for being able to listen to e-mail messages (traditionally read on a computer screen) by phone as well. The first such systems were developed in the early nineties in the USA for the English language. Their roll-out to the European languages is still in progress, as there are several problems to be solved which do not exist in English (e.g. recovery of accents, accentuation); on the other hand, the control surface of the systems is to be fitted to the habits prevailing in the given society.

This method has the advantage of enabling the use of e-mail through a simple wired or mobile phone without any computer. At present, most of the actually operating systems use a DTMF control method, while speech recognition based navigation is currently being developed. The present situation can adequately be characterized by the fact that the development of an e-mail reader for the major languages used in Spain was supported with a significant amount from 5th EU Research Framework Programme (E-MATTER, the project was launched in January 2001).

The development of a Hungarian-speaking e-mail reader was started at BME in 1995 and, since December 1999, it operates as a standard service of Westel Rt. The present article describes the language and traffic analyses required to developing and enhancing this solution.

There are several ways for utilizing statistical analyses for the enhancement of a system. We may study the full system or focus only on a part of it. First we shall discuss the language pre-processing module being one of the basic sub-systems of telephone-based e-mail readers; next we shall analyze traffic data obtained in the pilot period and characterizing the functioning of the full system. Users of the standard service "Mailmondó" [Mail Teller] can find its description on the homepage www.westel.hu.

Language analyses

Statistical text analysis conveys information on those characteristics of the investigated written language that can be deducted from the grammatical rules only to a restricted extent or not at all. Relevant studies made in connection with the Hungarian language have targeted some other goals such as the generation of dictionary entries (Pajzs-Váradi 1997, Pajzs-Kiss 2000). We focussed on word-level investigations as this was the primary area we had to study in order to solve the problem of accentuation.

In the course of the analysis we made frequency lists that can be used in computerized speech generation as well. Statistics provide help when various vocabularies used for speech processing are compiled or speech synthesizers are optimized, but they can be used – in case they are available for several languages – for the improvement of speech detection as well.

Hungarian is an agglutinative language enabling the formation of huge amounts of word forms. According to some estimates, these amount to several billions; this means there is no possibility to have all word forms taught and recognized one by one. In addition to the theoretical magnitudes, the number of inflected/non-inflected word forms actually incurring in real speech contexts is also worth investigating.

Corpus collection

The size of the underlying text corpus determines whether the statistics qualify as having a general validity or they merely reflect the characteristics of the given corpus. Statistics characterizing the written Hungarian language cannot be obtained by studying small-volume texts but require corpora containing several millions of words. When reviewing the corpora, the topics of the included texts are also to be considered. In case of a narrow thematical scope we may obtain data characterizing only the given area without being able to draw conclusions about the overall characteristics of the language. However, there may be special applications when it is expedient to investigate the characteristics of just one language domain. In such cases, we must be careful to exclude from this special corpus the text parts which are not related to the given topic.

Only texts available in an electronic format can be used for statistical processing. When collecting text corpora, we used first of all texts available on the Internet as these cover a wide scope of topics and may be accessed in quantities large enough for the purposes of the study.

Factors influencing the quality of statistical analysis

Written texts often contain misprints/errors deteriorating the quality of the statistics. In such cases we may include non-existing Hungarian word forms as well and, depending on the frequency of misprints, we may over-estimate the number of different word forms (types).

Orthographic errors are similar to misprints, but they include also a category for which the writer of the text cannot be blamed: Even in case of recently written texts it happens sometimes that the letters í and Í are not used while typing as these are not available on the given keyboard. A similar problem is that certain computer programs handle accented letters incorrectly so that accents may be lost. Most often such problems occur in connection with the letters o and u.

We see another problem in case of foreign language text sections appearing in the corpus. Coherent foreign language text sections may be filtered out by sentence-level language detection, but sporadic words embedded in a Hungarian text cannot be identified by a program in all cases. The most common examples are foreign proper names within Hungarian texts. These occur usually in the news (especially sport news). When there are too many foreign words, the corpus may miss the characteristics that would be otherwise exclusively typical for the given language.

When printed texts are page-set, make-up editors and text writers use hyphens for a better visual effect and easier readability. However, these hyphens are sometimes retained in the sentences even after re-formatting them; in such cases hyphens are no longer in a line-end position. This results in creating hyphenated words that give erroneous word forms again.

Statistical results are also influenced by the definition used when differentiating words and strings of characters not qualifying as words within the text. For the purposes of the present study, by 'word' we mean strings of characters containing letters only. A part of the words of German origin have been missed by excluding the letters ß and ä. Senseless forms have can be filtered out by such strict restrictions which, however, may also result in loosing some meaningful word forms.

Corpus characteristics

• **nytud:** Magyar Nemzeti Szövegtár [Hungarian National Text Archive] at the Research Institute for Linguistics of the Hungarian Academy of Sciences (version 1999). Here the starting point was not the text itself as the frequency list was already available for us. In such cases, however, word contexts cannot be analyzed. We used this data base mainly for investigating word frequency related characteristics.

The corpus contained altogether 20 805 975 words (tokens), while the quantity of different word forms (types) was 691 159.

• **mek:** We used more than 2000 Hungarian-language documents available in the collections of Hungarian Electronic Library. The selected texts covered a rich variety of topics such as works of literature, scientific articles, legal and everyday texts. Comparing this with the corpus nytud we find that, in spite of its smaller size (being only about one third of the former one), it contains a large number of different word forms (types); this can be explained by its wide scope of topics.

Tokens: 6 799 701 words, types: 522 432

• **mn:** This includes ten thousand articles published in the daily newspaper "Magyar Nemzet" between April and October 2000, available also on the Internet. The topics within this corpus are not of a general character, as it contains mostly documents written in the language of the media. The main point is, however, that the documents included in the corpus are recent ones reflecting the present state of the language. When evaluating the results we have to pay attention to the fact that year 2000 Olympic Games were held at Sydney in the analyzed period. The reports about the Olympic Games contain a large quantity of foreign names and sport-related words that are not commonly used in the everyday speech.

Tokens: 4 373 412 words, types: 345 657

• **hvg:** 4000 articles of the weekly paper "Heti Világgazdaság" published in the past three and half years. Due to their being published in a weekly periodical, these articles are somewhat longer.

Tokens: 4 091 732 words, types: 311 578

• **mh:** Nearly 8000 articles of the daily newspaper "Magyar Hírlap" were also collected from the Internet; they cover the first 4 months of 2000.

Tokens: 2 054 777 words, types: 196 965

Unified corpora

The corpora described above are independent of each other thus it was possible to aggregate their content. The figures relating to the analyzed corpora are shown in Table 1. In the course of aggregation, all tokens were added up without summing up the types, thus the increase in the number of types indicates a shared sub-set of two or more corpora. For example, we added the corpus hvg (containing 306 thousand types) to the unified corpus nytud+mek. The number of types was increased only by 100 thousand i.e. approx. 200 thousand word forms were included in both corpora. The largest corpus we have set up contains 38 million word forms (tokens) out of which more than 1.2 million were different (types).

Abbreviation of words (tokens)	*Number of different words (types)*	*Number*
mh+hvg+mek	*12 946 210*	*761 255*
nytud+mek 27 605 673	*991 546*	
nytud+mek+hvg	*31 697 487*	*1 090 916*
nytud+mek+hvg+mh0	*33 752 182*	*1 129 811*
nytud+mek+hvg+mh+mn	*38 125 594*	*1 225 101*

Table 1 Size of the unified corpora

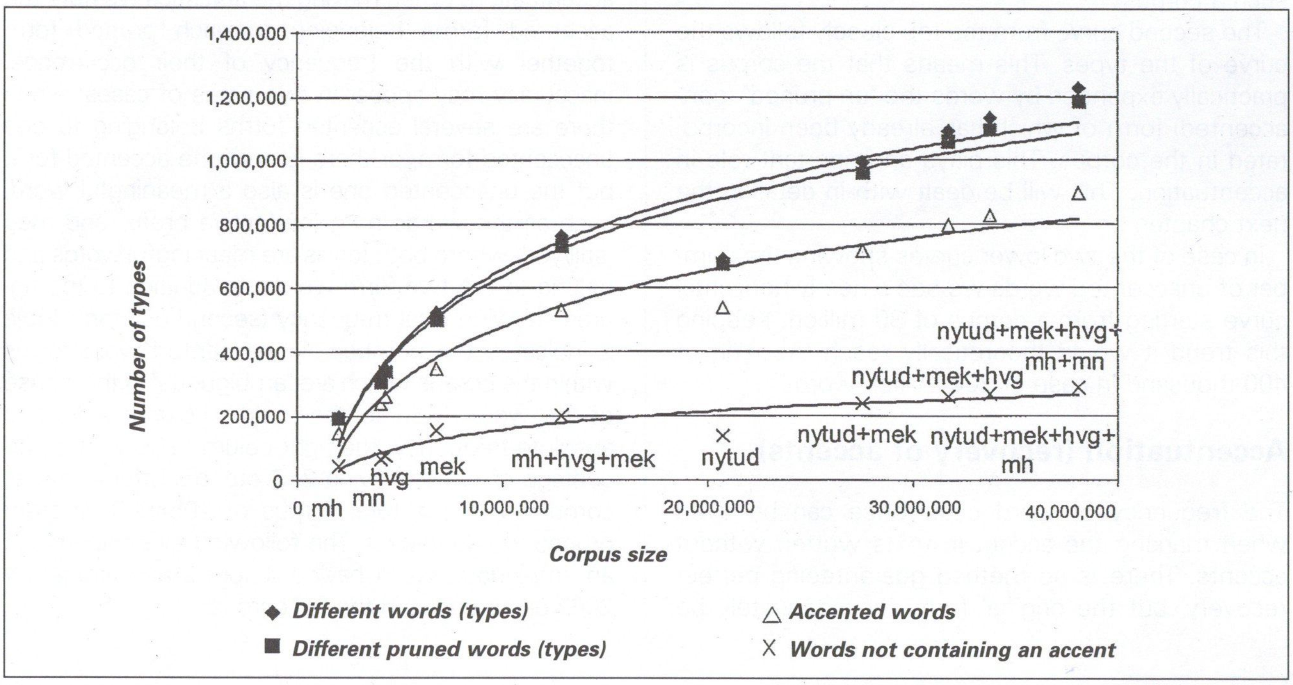


Figure 1 Comparison of individual and unified corpora

Statistics

Figure 1 shows the main characteristics of independent and unified corpora. The number of types is indicated on the vertical axis, while the horizontal one shows the size of the various corpora. The name of the corpora is given at the bottom of the figure in line with the above description and Table 1.

The uppermost line of dots shows the number of different words (tokens) found in the corpora. The squares immediately below them show the number of tokens we get when cutting ('pruning') the accented words and replacing the accented letter by its non-accented counterpart. E.g. by pruning the word ágyat [bed (acc.)] we get agyat [brain (acc.)]. The two lowest series show the quantity of different accented/non-accented words in the corpora. In this manner, all corpora are characterized by four signs under each other, with the abbreviated name of the corpus given below. The four curves (getting increasingly flatter) are logarithmic approximations of the series indicating the trend of the changes. This means, the upper curve indicates the number of different words (types) within the corpora as a function of the corpus size. It can be seen that, in case of a small corpus, the graph section showing the number of types is rather steep, but when we go on analyzing more texts, the more often do we find identical words, thus the increase of types is getting slower. In case of the dotted line belonging to the curve we see a break at nytud. In comparison to the more than 20 million words (tokens) in the corpus, the number of types is only around 700 thousand although the approximation curve would justify a value around 900 thousand. Presumably, the reason for this value's being smaller than expected is that the other corpora included in the figure contain smaller quantities from texts of a wider scope, while nytud contains a larger amount of texts within a narrower scope of topics. This smaller scope, however, ensures less variability, this is the reason for the "recess" in the sequence.

The equation of the approximation curve indicates that the variance speed of 1% (i.e. the point where the size of the corpus is to be increased by 100 words in order to get one new type) is reached at 31 million tokens. On the other hand, the 1‰ limit is at 310 million tokens what implies an extremely large corpus even when compared on an international level; it takes an enormous amount of labor to build such a corpus.

The second curve from the top closely follows the curve of the types. This means that the corpus is practically expanded by words the "un-pruned" (non-accented) form of which has already been incorporated in the corpus. This plays an important role in accentuation. This will be dealt with in detail in the next chapter.

In case of the two lower curves showing the number of unaccented words we see a nearly horizontal curve starting from a corpus of 30 million. Keeping this trend it would theoretically reach the limit of 400 thousand in case of 200 million words.

Accentuation (recovery of accents)

The frequency of word occurrence can be used when marking the accent in texts written without accents. There is no method guaranteeing perfect recovery, but the original form can adequately be guessed by using data provided by the statistics. This type of accentuation can be used first of all in situations where speed is an important factor and there is neither time nor resource for grammatical or other analyses.

2	29076	2,7561%
3	1609	0,1525%
4	285	0,0270%
5	19	0,0018%
6	1	0,0001%
7	1	0,0001%
8	0	0,0000%
9	1	0,0001%

Table 2 Number of words having identical pruned forms

First we remove the accents from the accented form, thus we obtain the form we shall find during the accentuation. When making the statistics we note the accented forms belonging to each pruned form together with the frequency of their occurrence. Inaccuracy may appear in two types of cases: when there are several accented forms belonging to one unaccented form, or there is only one accented form but the unaccented one is also a meaningful word, such as the words meg 'perfective prefix' and még 'still/yet', where both forms are meaningful words and belong to the two-form words. In addition to the figures showing total frequency (second column), Table 2 indicates the number of unaccented word forms within the corpus which are "ambiguous" in this sense of the word. Next to the middle column indicating absolute frequency, the right column shows the percentage of words having 2, 3 etc. doubtful forms as compared to the total corpus of 1054983 different pruned words (types). The following example shows an ambiguous word having 4 forms and altogether 2062 occurrences in the full corpus.

Example:	arat '(he) harvests'	árát 'its price (Acc.)'	árat 'price (Acc.)'	arát 'bride (Acc.)'
	5.19%	52.6%	41.10%	0.05%

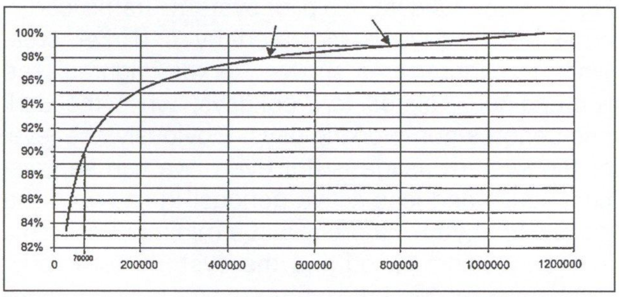
The last line of the table shows that the corpus contains even a word having 9 forms. There is no such word in Hungarian, and its 'virtual' forms are the results of misprints or character set deficiencies described above.

When setting the accents, the unaccented text is broken down to words; next the word to be accentuated is replaced by the most frequent form belonging to the given unaccented one. For example, if we see arat in the text, it will be replaced by árát; based on an analysis of a large amount of texts this gives a good solution in 52.67% of all cases. However, Table 2 indicates that this method is insufficient for 3% of words; this is the occurrence frequency of words with several possible forms. Actually, the situation is even worse as this 3% includes commonly used words as well, thus the safety guaranteed by this solution – in its present form – is not 97% but by some percents less. The enhancement of the procedure described above is subject of a patent procedure (Németh, Zainkó, Olaszy, Gordos).

Such cases (e.g. the most common unambiguous pair meg/még) require further statistical and/or grammatical analyses.

Quality of the statistics

The quality of the obtained results is indicated by the convergence of various data. We have investigated the scope of the corpus covered by the 20 most frequently used words. Our result (22.8%) is close to the value stated on the basis of other sources (Pajzs-Váradi 1997). Out of the 20 most frequently used words, 18 were identical with a list generated in one of our previous studies (Németh et al. 1999). This indicates the extant to which the original corpus is covered by its most frequently used words. The vertical line in the chart marking the most frequently used 70 000 words covers already 90% of all words used in the corpus. This means that by using the 70 000 most frequently used words of the corpus in some procedure (e.g. fixed vocabulary based speech synthesis/recognition or accentuation), corpus text coverage errors will occur only in maximum 10% on the average.



On the other hand, the flatness of the curve indicates that we have to use a constantly increasing amount of words for improving the coverage. In case of the first example, we managed to reach 90% at 70 thousand words, the step from 98% to 99% – indicated by the two arrows in the figure – would require 300 thousand words more. This means, a relatively large coverage can be achieved by few words, while a very accurate coverage requires many times more words.

When we use procedures connected to the most frequently used words, the size of the corpus is adequate, as the few words found ensure an appropriate coverage. But when we want to see how far we are from the practical limit estimated for coming across all imaginable words used, we have to admit the size of our corpuses is extremely small. According to certain estimates, the number of grammatically possible words is between 500 million and 1 billion (Prószéky 1993).

Figure 3 shows the percentage of words occurring in the various corpora only once, twice, three, four, five or more times. This diagram indicates that the number of words included in the corpuses is not large enough, as the different words (types) occur in 55% of all cases only once on the average. However, a corpus can be described as approximating the practical limit when the course of the graph is reversed, and the number of words occurring only once is no longer significant in comparison to the size of the corpus. The distribution is nearly independent from the size of the corpus, as the mh corpus containing about 2 million words is characterized basically by the same distribution as the corpus nytud+mek+hvg (31 million words). Based on this we assume the appropriate corpus size may be around some hundreds of millions. Accordingly, systems of rules relying on language analyses are to be developed in case of most practical large-vocabulary applications.

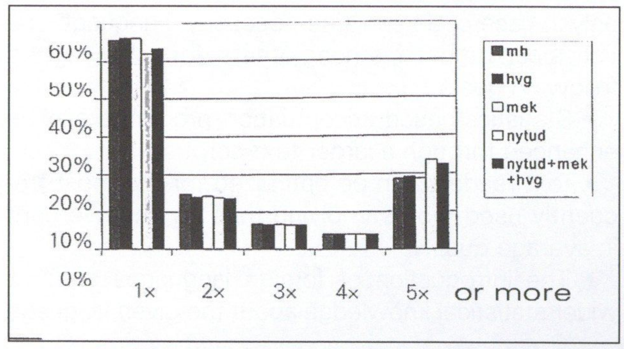


Figure 3 Word frequency distribution

Comparison with other languages

In order to compare our results with other languages, two English corpuses (Morgan) containing both historical and present-day texts have also been analyzed according to our method developed for the Hungarian language. Our results obtained for British and American English indicate that corpuses of about ten million tokens contain approx. 80 thousand different words (types), and shared forms amount to 47 thousand.

The difference between English and Hungarian is adequately indicated by the fact that, according to our investigations, a Hungarian text of ten million words (tokens) contains about 700 thousand different forms (types) as compared to the 80 thousand in English.

British National Corpus (BNC) includes 100 million words, being thus much higher (even in absolute terms) than our corpuses. When we consider also the fact that the English language has about ten times less different words (types), then the Hungarian corpus should contain 800 million – 1 billion words. In this case the frequencies would have an accuracy approximating that of the BNC. It is important to note, however, that the expression 'vocabulary' used in the English terminology relates not to basic forms (dictionary entries) but to the quantity of known word forms. This means, the expressions hajó 'ship', hajók 'ships', hajót 'ship' (acc.) count as 3 words. Our former studies have indicated that a vocabulary of 20 000 words in line with the above definition would cover only the vocabulary of short weather reports.

Reader enhancement

A more detailed study of the Hungarian language as well as a comparison with the most important foreign languages enable us to enhance the language modules and apply additional languages. Constantly growing user requirements cannot be met unless accentuation is improved. Today it is not enough to market a stuttering e-mail reader that can be hardly understood; when it is uncomfortable for the users to listen to such a voice, they will use the service only in case of an extreme necessity. Language statistics can be used, among others, for achieving the following goals:

- Statistics-based accentuation programs can be enhanced through a larger text corpus
- Text readers can be optimized for the most frequently used words implying thus an improvement in average quality
- The introduction of foreign languages requires wider statistical knowledge about the given language

Analysis of traffic data

Systems cannot be enhanced without getting familiar with user habits. Data may be collected through questionnaires or telephone inquiries requesting information on the users' habits. This type of research cannot be substituted by any other method, as this is the best method for obtaining information on the subjective opinion of the users. The problem is that only a relatively small number of users can be questioned in a cost-efficient manner and/or it is not sure whether we have a representative sample.

The set of questions and complaints received by various on-line and off-line customer services may serve as an additional source of information. Various errors and poor features disturbing a large part of the users may presumably be identified in this manner, but customer services are a less probable source for indicating minor faults and suggesting positive developments.

However, there are certain data that we could not get from the users (not even by detailed questionnaires) even when collecting information from all users. These may include traffic data as an average user cannot estimate (not even approximately) the date/time and duration of his/her using the system.

Traffic data

Figure 4 shows data of a pilot system. Traffic load in non-working days is indicated by a darker area, while the lighter area shows workday figures. The lighter figure relates to weekdays, while the dark one denotes non-working days. Obviously, the level of the system usage is not even; we can see distinct active and less active periods. In line with the human biorhythm, there is just a minimum of traffic in the sleeping period and the system is more or less characterized by daytime usage (what is not a surprise).

However, a more close study of the figure reveals that the habits of the users do not fully comply with the classic traffic load distribution typical for tele-

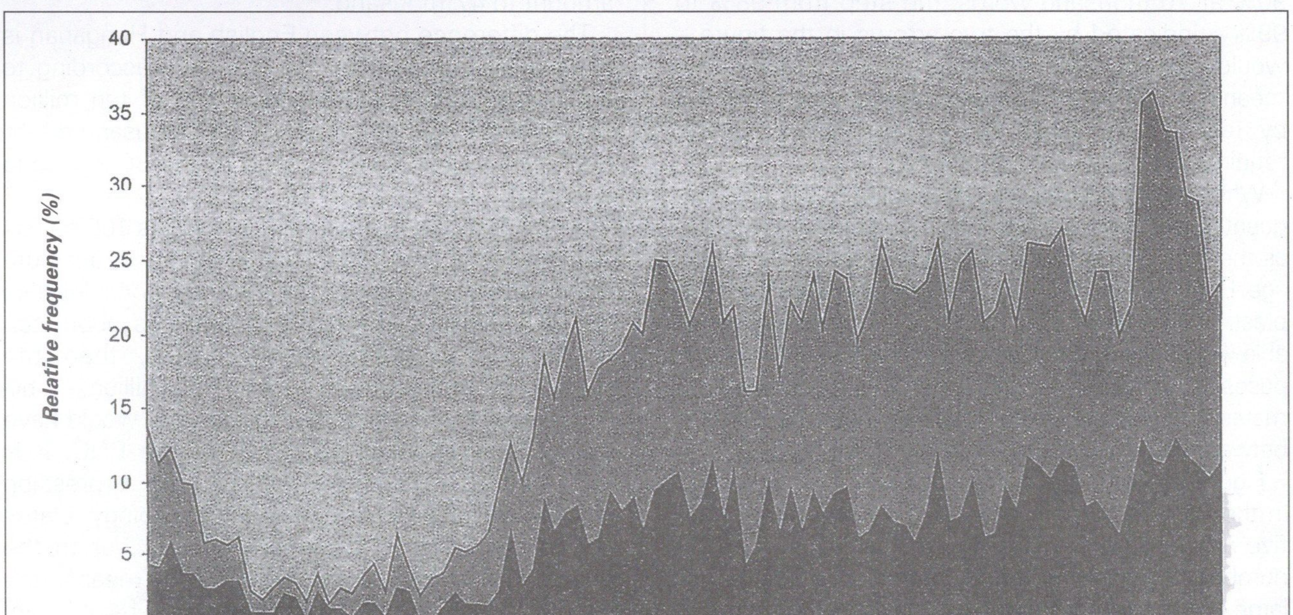


Figure 4 Daily distribution of traffic

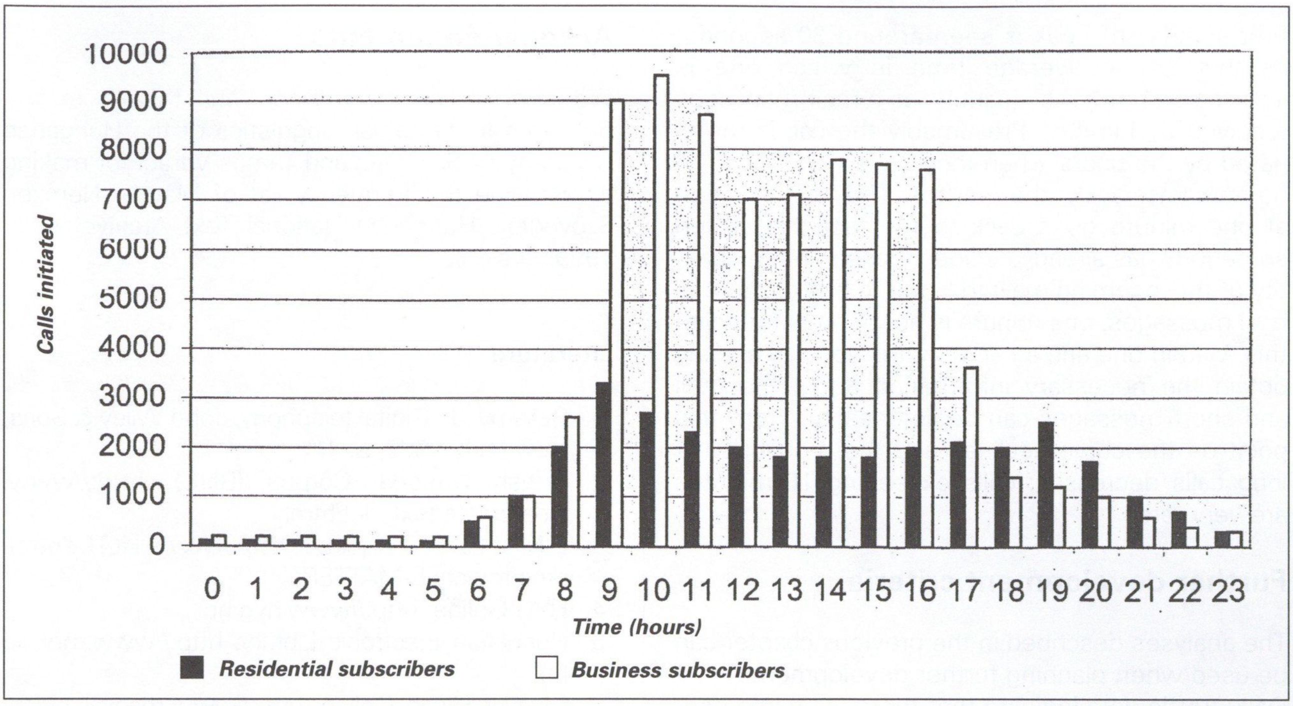


Figure 5 Classical traffic distribution (source: Bellarmy)

phone exchanges (Figure 5). The main difference is that peak traffic appears not at mid-day but in the late evening hours when significant discounts are given. Furthermore we can see that the decrease of the late night traffic is relative slow, and there is a significant traffic till 02:00 in the night. The first peak period of the day comes around 09:30, when a large part of the user listen to their mail for the first time on that day. There is another peak before lunch, and then a strong decrease around 13:00. Most likely there are many people who do not care for their electronic mail during the lunch or immediately after it. In the afternoon period, we see a significant traffic increase around the end of the working hours, when users listen to their e-mails before going home. Traffic is smaller in the period of the evening rest, while maximum traffic is generated in the cheapest period.

Figure 6 shows the holding time of successful incoming calls and their approximation curve. Accordingly, this figure shows only the figures of calls where the user succeeded in identifying himself/herself. The lack of extremely short calls (lasting only for a few seconds) is explained by the time requirements of user identification. The

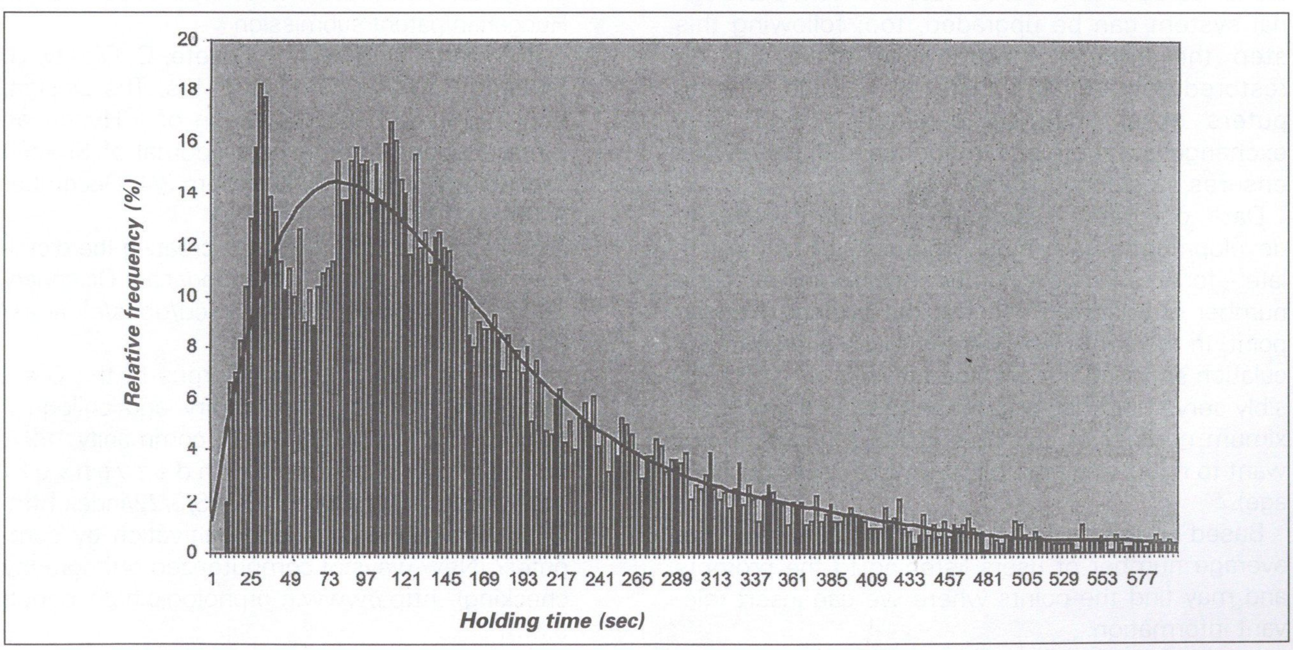


Figure 6 Distribution of holding time

first significant peak is seen around 30 seconds, as this is the average time in which one is informed about the quantity of e-mail messages received by him/her. Presumably the call is terminated by the users when they have no letters – or no new letters – in the mailbox. The "recess" seen at one minute goes back to the fact that, when somebody has already decided (based on the quantity of the incoming mail) to listen to one or more e-mail messages, one minute is not enough for doing this. Within one and a half minutes you can already obtain the necessary information about the mail and short messages can be listened to. From this point on the curve declines and the probability of long calls decreases; calls exceeding 10 minutes are very rare.

Further development criteria

The analyses described in the previous chapter can be used when planning further developments.

Daily traffic data indicate that there is no idle period in the system, we see only some sections with a smaller traffic (between 02:00 and 06:00). This is important from a maintenance point of view; when we want to render an ongoing service (being a basic requirement for up-to-date systems), system development and maintenance can be implemented only in an automated manner or by using a stand-by system. >From a development point of view, the use of stand-by systems is a more advantageous solution. In such cases, the upgraded system can be installed to the stand-by computer, and subsequently – after a pilot period proving the reliability of the system – the upgraded or enhanced service can be launched by a continuous cutover. Next the original system can be upgraded, too; following this step the original allocation of tasks can be restored between the systems. In case of computers of an identical capacity, a final task exchange is a more advantageous solution as this ensures a balanced traffic load.

Data on traffic distribution facilitate capacity development as well by giving a possibility to calculate – for an average n number of calls per day – the number of unserved users at the system overload point. In the same way we can make a reverse calculation showing the average number of users possibly served per day or week in case of a given maximum number of channels (given that we do not want to reject any user because of a capacity shortage).

Based on the holding time we can determine the average number of users listening to the prompts and may find the points where we can insert relevant information.

Acknowledgements

We should like to express our thanks to the Research Institute for Linguistics of the Hungarian Academy of Sciences and Tamás Váradi for making us available the frequency list of Magyar Nemzeti Szövegtár [Hungarian National Text Archive] (year 1999 version).

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Developments in Hungarian grand dictionary and attached word based, computer aided speech recognition

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The increasing integration of computers into everyday life presents a growing need for input devices that are more natural than widely used keyboards. Since speech is one of the most natural way of communications for people, there is a worldwide need for the study of human speech as input technology to computers. This paper presents a short overview of the latest developments achieved in the field of the Hungarian computer-based speech recognition at the Budapest University of Technology and Economics, Telecommunications and Signal Processing Laboratory.

In the Hungarian language it is the size of lexicon that presents the greatest difficulty. To manage this difficulty we introduce the method of stochastic morphologic analysis which has already proved its capabilities in spell checking. This method will then be extended to the case of non-deterministic series of observation. This will be followed by the short introduction of a method which can be used to manage pronounced variants according to the formerly mentioned lexicon representation. Finally two practical developments will be introduced: a speaker-independent attached word based number recognition system and a 2000-word city name recognition system. In connection with these developments the signal representation used for the experiments, the structure of the applied acoustic models as well as the achieved accuracy of recognition will be reviewed.

Morpheme-based lexicon representation

In this section two alternatives will be introduced for the representation of lexicon of the agglutinating languages. Both methods uses morphemes as basic elements but while the first alternative makes use of mainly standard statistic language modeling techniques, the other variant allows also for the explicit setting-in of linguistic knowledge available for the morphology of a particular language. Accordingly, the main advantage of the first method lies in its simple and easy applicability to any language whereas the other method results in a more accurate model, its application in speech recognition systems is expected to reduce the number of recognition errors.

The bigram model of the morpheme

In case of isolating language – such as English – the stochastic linguistic modeling is a proven method to improve the accuracy of recognition. Generally known as n-gram linguistic model, this method supports recognition by assigning probability to each potential word-n statistically (estimated with relative frequency) and weights thereby elements of the strand of words to be recognized. With this method the majority of incorrect word orders can be discarded since they are assigned 0 or a very low probability value. In these cases lexicons are represented by n parallel lexicons where the aforementioned probabilities (elements of the so-called stochastic transition matrix) are assigned to n x b links between adjacent lexicons. In practice the value of n is 2 (bigram) or 3 (trigram) since the appropriate assessment of transition matrices a higher order linguistic model would require an immense textual data base (and immense processing time).

Originally this method was designed for modeling relations between words but it can now be used also for the management of the problems of lexicon representation. The easiest way of representation of a lexicon using morphemes is to create two sub-lists. One list consists the vocabulary and alternative stems of words to be recognized and the other consists all possible suffixes. Now agglutinated forms are represented with stems and the suffix list linked together by the bigram linguistic model. This representation can be seen in Figure 1. In this example there is just one word with all the three possible stem variants on the stem list. The suffix list is formed of all possible variants of the eight different suffixes. (Obviously this suffix list is far from exhaustive since at least 1000 different suffix combinations can be linked to this word.) Since the suffix list is general, it shall contain all variants of all suffixes. The permitted links between individual stems (rather stem variants) and suffixes is determined by the bigram linguistic model (stochastic transition matrix).

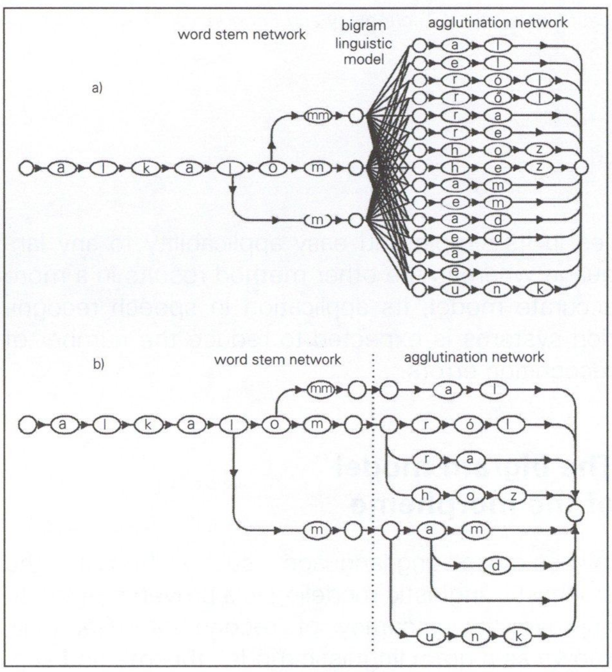

Figure 1 Representation of agglutination information by bigram linguistic model (a) and morphologic analyzer (b)

Obviously this way of representation comprises all agglutinated forms of all words figuring in the word stem list. A great advantage of this method is that each suffix appears once only in the model while in case of full word representation (where words with different suffixes are considered as different words and stored in parallel lexicons) each suffix would appear as many times as many words there are in the stem list. In addition, each word would appear as many times as many different suffix combinations can be added to it. Besides the considerable memory savings, it is also a strength of this method that it requires just the collection of possible word stems and suffixes and the bigram transition matrix assessment can be performed with standard automated processes.

A disadvantage of the method stems also from the automates assessment process. In practice, the accurate estimation of the bigram transition matrix has proved to be extremely difficult. This is due to the fact that in case of using finite size textual databases there are always agglutinated word variants which do not appear in the learning database but can occur during the use of the system. The assessment process would assign zero probability to the missing agglutinated form and therefore the system would be unable to recognize that particular variant. These situation can be handled by the usual "smoothing" techniques. This method assigns a low (but above zero) probability to word combination not contained in the learning database.

Such algorithms are, however, unable to differentiate word variants having suffixes that are missing owing to data deficiency or undetermined grammatical rules. As a consequence, the smoothed linguistic model assigns positive probability to grammatically incorrect suffixed words as well, e.g. not only to the correct form of "alkalm-ad" but also to the incorrect "alkalm-ed".

Use of the stochastic morphologic analysis

An alternative to the method outlined above can be the use of morphologic analysis, a proven means in spell checking [5].

Up to now morphologic analysis has been used in written language systems, such as in lemmatization, translation aids and spell checking. As a consequence, established methods of computer-aided morphology are based on the assumption that words to be analyzed are represented in textual format. With computer-based speech recognition, however, stochastic observation series, generated with the assessment of analogue speech signals, have to be matched to linguistic symbols, i.e. the traditional deterministic approach cannot be used directly. For this reason morphologic analyzers have been proposed to use in the post-processing step [5] of the recognition work, when a roughly accurate series of phonemes are already available following a "pre-processing" job.

This paradigm, however, does not allow for the "pre-processor" to access to linguistic information and, as a consequence, the quality of the series of phonemes determined in the first process can be rather low. This reduces considerably the chance for the morphological analyzer to achieve an acceptable word form during the second step.

Now we propose the extension of traditional – or rather two-level [3] – morphologic analyzers to handle stochastic observation series. The proposed method is called stochastic morphology whereas the analyzer will be cited as stochastic morphological analyzer. In the following sections we outline basic principles of the stochastic morphological analysis and show how it can be used in the analytical speech recognition.

Formalization

The basic function of a morphologic analyzer is to convert the input word into a series of morphemes. Though there exist many alternative implementations, in this article we will consider the morphologic analyzer as a finite translator, according to the model used in two-level morphology [3].

The finite translator reads one by one the input observation series and assigns an output symbol to each observation and the state machine moves to the next state. Both input and output symbols can be blank observations as well. Generally the output is formed by the canonic variants of morphemes of the input word, sometimes accompanied by a label referring to the grammatical role of the given morpheme. If after processing the last input symbol the state machine stops in an accepting state, the analysis of the input series is successful and the output series contains the results of the analysis. If the machine stops in a not accepting state, the analysis of the series is unsuccessful (does not form part of the language processed by the machine). The advantage of this model is its efficiency in the representation of the lexicon of the particular language.

On the other hand, the capabilities of the model is limited by the fact that a state processing job cannot be performed unless exactly the pre-defined symbol is read from the input. This limitation does not allow its use for the analysis of noisy input series which are so frequent in everyday life (and in speech recognition).

The deterministic morphologic analyzer described above van be converted into stochastic with replacing its inherent finite state machine by a hidden Markov model. State transitions depend on input symbols in case of the Markov model as well, however, this dependence is not deterministic but stochastic. In case of an arbitrary input symbol any transition can be performed with certain probability but the probability of the transition depends on the symbol. (With specially selected model parameters the performance of the analyzer can be set to that of a deterministic analyzer.) Thanks to the stochastic operation the analyzer can accept more than one analysis for any input series (depending on the selected state transitions). From point of view of practical implementations the task is now to identify the most probable (or N most probable) analysis that can be done with the use of the well-known Viterbi-algorithm.

The analyzer created in this way provides an efficient representation of the given lexicon. State series belonging to grammatically incorrect word variants are of zero probability since the topology of the state machine has not changed. In case of a stochastic analyzer, however, it is no more necessary that each word be represented by a predefined series at the input. This feature makes analyzers of the stochastic morphology ideal means for the representation of various word forms of agglutinating languages in speech recognition systems.

Figure 1 presents an example for the representation of 8 agglutinated forms of the word 'alkalom' using a stochastic morphologic analyzer.

Modeling of pronunciation

Similar problem to that of the representation of lexicon is how the system can be made capable for accepting different (correct) pronounced variants of a word. For English words the usual method is to put alternative phonetic versions of the given word element under each other. If the system recognizes any of these variant, it will associate to that particular word element and displays it on the screen.

However, this method is memory-wasting in itself since pronounced variants differ generally just in one or two tones, all others are the same and their repeated storage is unnecessary. There is a more serious concern about this method: it was designed for parallel lexicon representation and cannot be used to a stochastic morphologic analyzer at all.

Below we introduce a pronunciation modeling method which can be fit to the stochastic lexicon representation and presents a minimum memory demand as well.

Main features of the algorithm:

1. Canonic phonetic transcriptions of individual vocabulary elements (morphemes) are created automatically [4]. (Since pronunciations vary most frequently at the borders of morphemes, the transcription of individual morphemes is clear and meets specifications, i.e. can be considered as canonic.)

2. Phonetic transcriptions of lexicon elements are grouped into a graph according to the topology of the state machine performing the stochastic morphologic analysis. As a matter of fact a hidden Markov model is produced which contains lexical, morphological and partly pronunciation information but does not make a model of phonetic changes due to morpheme associations.

3. Finally, pronunciation rules of the Hungarian language are formalized an then applied to morpheme borders. To achieve this, first the hidden Markov model is converted into phoneme-level, then poten-

tial pronunciation variants appearing at association points are represented, with the insertion of appropriate phoneme states and edges as well as with the re-generation of the graph, if necessary (Fig. 2).

All this process will result in a recognition network which contains all available information above phoneme-level.

In the next section our recognition experiments will be described, noting also some acoustic and signal processing aspects.

Recognition experiments

There have not been much research work in the field of speaker-independent speech recognition in Hungarian. Publications appeared up to now concerned phoneme and semi-syllable recognition [8, 1, 2]. Main barrier to achieving more practical results was the lack of an appropriate textual database. This difficulty was overcome with the compilation of the BABEL database [9].

Though in the number of speaking persons and the volume of speech this offers far less than other textual databases available in English, it can be used for teaching and testing speech recognition systems.

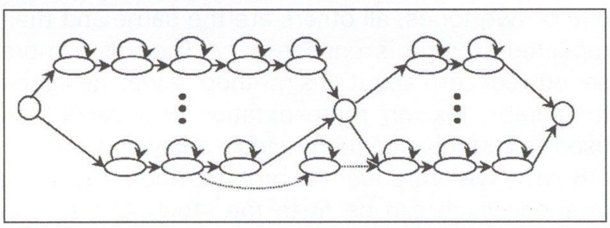

Figure 2 Modeling phonological modifications occurring at morpheme borders with the introduction of new form states and edges (dashed line)

Signal processing

This section describes the setting of the signal processing module. This setting remained unchanged during the whole process of experiment. In course of the signal processing job the sampled speech was converted into a series of vectors, each element of which describes the spectral parameters for the period of a frame. During speech recording the initial sampling frequency was set to 20 kHz. In every 10 millisecond a 10-dimension cepstrum vector was calculated according to the mel-scale, using the actual speech segment weighted with a 20 ms long Hamming-window. The logarithmic value of the energy was also associated to the resulting vector.

In addition to these static coefficients we used so-called delta, or differential coefficients as well. Delta coefficients were calculated using the regression method with data from ±2 frames (e.g. [6]). The so-called "acceleration" coefficients were also taken into account. At the end we have got 33-dimension parameter vectors.

Experiments with BABEL database

At present BABEL is the greatest, high-quality Hungarian database available for research purposes. The database consists of three parts: isolated and attached-word number announcements, CVC (consonant-vowel-consonant) syllables as well as continual, read-up speech [9]. In the continual section certain parts are whispering, but these were not used in the experiments.

Both read sentences and series of numbers were designed to cover tone combinations occurring in the Hungarian language. A small part of the database is segmented into phonemes and labeled. In the sample available for the experiment speech from five speakers contained phoneme-level labels in a 400 seconds long speech. (For the development of a speech recognition system usually several hours long speech, labeled at phoneme-level is used.)

The database comprises altogether the voice of 30 speakers (15 male and 15 female), 2000 sentences and 14 000 attached-word numbers. Unfortunately we were not allowed to use the whole database and this fact reduces the robustness of the estimated phoneme models. The voice of speaker used for teaching was not used in testing.

Teaching the models

Since only a small part of the database comprises phoneme-level labels, the teaching was carried out in two phases. In the first phase initial models were taught using available phoneme labels, then with the use of these models and the FlexiScribe program [7] the complete teaching set was labeled. In the second phase these labels and the complete teaching set was used to teach models. Models were of left-to-right type with three states in each phoneme. Observation probabilities were modeled in each state with a 10-component, diagonal covariance matrix of Gaussian mixing distribution. Out of all the 64 phonemes of the Hungarian language only vowels and short consonants (39 phonemes altogether) were used since short and long consonants differ mainly in their duration and the hidden Markov-models cannot interpret duration.

a) Acoustic models taught on numbers

Phonetic transcription	*Recognition error*	*Relative decrease in errors*
Single (most frequent from)	*0.48%*	*6.3%*
Pronunciation options	*0.45%*	

b) Acoustic models taught on general speech

Phonetic transcription	*Recognition error*	*Relative decrease in errors*
Single (most frequent from)	*2.69%*	*4.1%*
Pronunciation options	*2.58%*	

Table 1

Recognition of numbers as isolated words

These experiments had two purposes. On the one hand we wanted to use them as basis for comparison in future experiments, on the other hand we wanted to find out the impact of alternative pronunciations on the accuracy of recognition.

Testing the number recognition as isolated words were possible because the database comprises 140 different numbers (such as ezerhuszonhárom [1023], kettő [2], négyszázhat [406], etc.) which are announced several times by different speakers. The available part of the database contained numbers between 1 and 10,000, in 9,700 different announcements.

The impact of teaching speech on the accuracy of recognition was also tested. For this reason two different teaching sets were used in the experiments. In one case acoustic models were taught with about two thirds of the number database which were then tested on the remaining 3340 announcements. In the other case continuous speech of general phonetic character was used for teaching and the recognition was tested on 3600 number announcements.

Word models of numbers were built of context-independent phoneme models, associated in the way determined by the automatic phonetic transcriber. In one part of the experiments only the most probable form of the transcription was kept (which was considered as more frequent) while in the other part of experiments all pronunciation options were recorded. For the representation of the vocabulary the easiest parallel lexicon was used. Finding of the experiments are summed up in Table 1.

Table 1 shows that in this special recognition problem the introduction of pronunciation options in itself practically had no impact on the error rate. However, the phonetic composition of the teaching data set has a considerable effect on the accuracy of recognition. This result suggests the necessity of the creation of context-dependent acoustic models.

Recognition of numbers as attached words

The stochastic morphologic analysis based lexicon representation and the associated pronunciation modeling technique described above were tested on this recognition task. The system vocabulary comprised 23 morphemes altogether (such as nulla, egy, két, kettő, három, hárm, ... an, en, van, ven, ... ezer). When associated with the grammar network,

a) teaching acoustic models on numbers

Lexicon representation	*Recognition error*	*Relative decrease in errors*
Stochastic morphological analyzer with canonic phonetic variants	*5.89%*	*41.4%*
Stochastic morphological analyzer with pronunciation modeling	*3.45%*	

b) teaching acoustic models on general speech

Lexicon representation	*Recognition error*	*Relative decrease in errors*
Stochastic morphological analyzer with canonic phonetic variants	*16.28%*	*18.4%*
Stochastic morphological analyzer with pronunciation modeling	*13.28%*	

Table 2

the finite machine accepted every correct form between 0 and 999 999.

Experiments were carried out with two different acoustic models (on numbers and on general speech as described in the previous section). The main concern was to identify whether pronunciation modeling improves considerably the efficiency of the recognition in an attached word job. For this reason part of the experiment was performed with a hidden Markov model graph containing purely canonic morphemes which offered no modeling of pronunciation options at morpheme borders. The other part of the experiment was performed on a graph which contained pronunciation options. (The expression of hidden Markov model graph refers always to a stochastic analyzer which is a morphologic and phonologic analyzer as well in the last example.)

Recognition results are summed up in Table 2. These results suggest that modeling of the pronunciation brought about a significant improvement in both acoustic models. It is remarkable that the relative improvement was considerable higher where also the initial recognition was better. Numbers in the table are relevant to the rate of number series errors but in most cases only one digit of a multi-digit number was mistakenly interpreted.

If recognition errors are compared to isolated-word results we get 5 to 7 fold higher values. This may seem too much at first glance but we should not forget that the amount of number forms accepted by the morphological analyzer augmented to about 10,000-fold (without any increase in memory demand or computing capacity!). With this view in mind the results can be considered positive, noting that with the use of context-depending phoneme models error rates achieved with acoustic models taught by general speech can be further reduced.

Recognition of city names

This experiment was a grand dictionary recognition problem in a general office environment. The task was the recognition of the name of the 2000 greatest Hungarian settlements. The pronunciation vocabulary was prepared automatically with algorithm described in [4]. In this case word models were built up with phoneme models taught on the BABEL speech database as well. Acoustic models were not adapted to different environments. In the testing set there were two male voices and 2550 announcements with breaks between words. The achieved error rate was 14.58%. Works to further decrease error rate is underway with the development of teaching databases and context-dependent phoneme models.

Acknowledgements

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Stereo Echolocation and Audio Techniques to Aid Blind People's Mobility

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This paper introduces a novel navigation aid for the blind designed at our laboratory. The system detects obstacles in front of the user by means of ultrasonic echolocation and indicates the distance and horizontal position of the nearest detected object by spatial stereo sound effects. The instrument is based on a fixed-point digital signal processor. The applied object-localization method as well as the principle of the binaural audio feedback are discussed. Also the operation and structure of the hardware are briefly introduced. Finally some measurement results on an ex-perimental device are reported.

Introduction

The Aim

The potential usefulness of a navigation aid is unquestionable to help the perception of the envi-ronment for visually im-paired people. Bats are well-known examples, which can perfectly navigate by the help of ultrasonic echolocation without vision, and today this technique is successfully ap-plied in mobile robots. Independently from this field, real-time 3D sound generation has been developed to an accessi-ble technology even in home PCs. Our basic idea is to con-nect these two threads using fast and power efficient DSPs (Digital Signal Processor) to produce a small, portable in-strument, which can be a useful navigation aid for the blind. The ultimate goal is to indicate the environmental obstacles by such a stereo sound effect as if it had originated from the obstacle itself.

Preliminaries

In the past three decades several electronic travel aids have been introduced that aimed at improving their blind user's mobility in terms of safety and speed [3]. Many of these de-vices apply ultrasound to detect objects and to inform the user through the auditory system [4], [5], but these instruments have not gained wide acceptance. The reason might be that according to the technology level either the pre-cision of the obstacle localization was not satisfac-tory or the audio feed-back was not natural enough. Clearly, it is difficult to evalu-ate a complex sound signal, (see for example the SonicVisioN [5]) and a considerable learning period is needed before ac-tual use or, in other cases [4], the indication is too poor in information. Therefore, we have decided to pay more atten-tion to the sound indication and applied a digital signal proc-essor in our device in order to provide a flexible and sophisti-cated stereo signaling sound feedback [1], [2].

The Developed System

On the basis of three years experience we have designed a battery operated ultrasonic navigation aid for the blind. The digital signal processor based system is able to determine the distance and the horizontal position of the obstacles in front of the user and to indicate the location of the nearest (most dangerous) one by stereo sound through ear-phones. The DSP is equipped with suitable interface circuits connected to one ultrasound transmitter and two receivers, which can be mounted on the hat of the blind person. The instrument is currently under construction, therefore the operation and algorithms have been tested on an experimental device based on a Starter Kit [11] equipped with external interface circuits but it is functionally equiv-alent to the portable device.

Two-dimensional echolocation

In the followings we introduce one of the key issues of our navigation aid, that is, how to determine the distance and direction of an object by means of ultrasound reflections.

The Basic Method

The principle of the applied echolocation is shown below in Fig. 1. To simplify the discussion some assumptions have been made: 1) There is only one obstacle. 2) The size of the obstacle is relatively small. 3) The surface of the object to be detected is ideally reflecting.

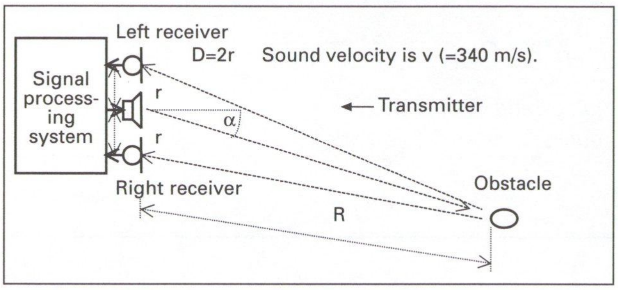

Figure 1 The principle of two-dimensional echolocation

The localization method is as follows: The transmitter emits an ultrasound impulse of given (T_{imp}) length. First a direct wave arrives in the receivers, therefore the inputs should be inhibited for a period of (T_{imp} + r/v). Later, depending on the distance (R) and direction (α) of the obstacle, a reflected impulse arrives first in the nearer receiver and then in the farther one.

At this point, the critical task is to determine – as precisely as possible – the instant when the reflected impulse arrives at the left and the right side, respectively. Let us consider now the simplest way of detection, the "threshold method" illustrated in Fig.2 for the left one of the receivers. When the received signal first exceeds the receiver's threshold, the echo is de-tected and the measured "to-and-fro" propagation time (t_L) is stored. Though this simple method is sensitive to noise, it should be mentioned that it could work without envelope-demodulation of the received signal; so it can provide the basis of more sophisticated methods like distance varying or adaptive threshold methods.

Using the left and right time delay of the echoes (t_L, t_R), the coordinates of the detected object can be calculated. Assum-ing that R >> r, the following approximations can be used:

· The distance is: (1)

$$R = \frac{v \cdot \min\{t_L, t_R\}}{2}$$

· The direction is calculated from the difference of time delays: (2)

$$\alpha = \arcsin\frac{(t_L - t_R)\cdot v}{D}$$

Some Theoretical Limits

Even in the case of ideally precise measurement of the time delays (t_L, t_R), the echolocation system has its limits. First let us consider the range of the object localization. The percepti-ble distance has a theoretical lower bound, that is, (calculating with the parameters of our device; T_{imp} = 1 ms, D = 17.5cm): (3)

$$R_{min} = \frac{v\cdot\left(T_{imp} + \frac{D}{2\cdot v}\right)}{2} = 21{,}4\,cm$$

The maximum range is limited by practical factors, such as the ultrasonic transducers characteristics, the detected object's shape and material, the direction of the object, etc., therefore it cannot theoretically be predicted easily.

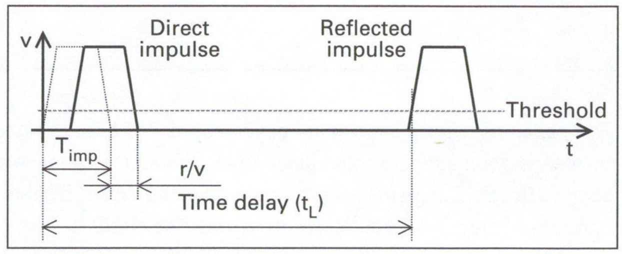

Figure 2 The ideal envelope-signal of the left receiver

The theoretical error of the digital signal processing system due to sampling can be more straightforwardly calculated. In our case the upper bound of the distance measurement error (4) and the minimum of the direction measurement maximum errors (5) are the following (with the parameters of our device – sampling time: T =45.4 μs)

$$\max R_{err} = \frac{v\cdot T}{2} = 7.7\,mm \tag{4}$$

$$\min\max \alpha_{err} = \arcsin\left(\frac{v\cdot T}{D}\right) = 5° \tag{5}$$

The expectations (mean values) of the errors above can be efficiently reduced using common interpolating techniques.

The Effects of More Realistic Conditions

In practice, the three assumptions we made at the beginning of the section are generally not valid. Let us investigate them very briefly one by one.

Firstly, if there is more than one obstacle, the nearest will be detected. This is quite convenient for us, and does not pose any critical problem. If a special situation occurs where two obstacles are at nearly equal distances but in different directions, then it can easily be seen that the system will indicate only one object with the right distance but with a direction somewhere between the two obstacles. That we do not con-sider a severe mistake, since in a dynamically changing envi-ronment such conditions can hold only for a limited time.

If there is only one obstacle but its size is relatively large, then similarly to the previous case, clearly its nearest part will be detected. The extrapolation to more than one or to concave obstacles is now obvious.

Finally, if the surface of the object to be detected is poorly reflecting (because it is too small, or its material is too soft, or the angle of the surface to the receivers is too sharp) then, unfortunately, such an object cannot be detected or only from very short distances. Actually, that is the main limitation of ultrasonic echolocation.

Two-dimensional spatialization of a monophonic sound

In this section we discuss the second vital question of the study, that is, how to make the 2D-location of an obstacle perceptible with stereo audio.

The Principle of Spatialization

Spatialization of a monaural sound means virtually placing the sound source to a given point of the space related to the listener's head. The procedure is a special transformation of the mono source to binaural signals to be transmitted to the ears typically through headphones. Here, we narrow our interest on the horizontal plane, in which it is often called 2D spatialization. The principle of the process is depicted in Fig. 3.

As the figure illustrates, the audio signal of an outer source is filtered at left and right by the head and the pinnaes; this effect is described with the Head Related Transfer Functions (HRTFs). HRTF sets are given as a result of measurements on humans or using artificial head and torso. HRTFs are quite complex functions of the four variables of the space coordi-nates and the frequency, but the dependency can be reduced in the far field (R > 1 m) and in the horizontal plane to the azimuth (α) and frequency [6]. The inverse Fourier trans-formed form of the HRTF – namely the Head Related Impulse Response (HRIR) – is applied generally in synthesis systems as (6) and (7).

$$x_L(t,\alpha)=\int HRIR_L(\alpha,\tau)x(t-\tau)d\tau \qquad (6)$$

$$x_R(t,\alpha)=\int HRIR_R(\alpha,\tau)x(t-\tau)d\tau \qquad (7)$$

If the desired direction (α) is given, selecting the appropriate pair of HRIR real-time spatialization can be achieved by convolving the source signal at the left and right side. Moving sound sources can also be simulated in this way; the loudness (intensity) should be changed in accordance with the distance (no additional spectral shaping is required).

As the transfer functions are related to the head, usually some kind of head tracking is necessary in spatialization systems, otherwise the sound space would move together with the moving head.

Lateralization vs. Spatialization

The FIR filter representation of a HRIR used to synthesis has about a length of 3 ms (128 tap) [8], which means at 44.1 kHz sampling frequency more than 10 million multiplications and 10 million additions per second per simulated sound source.

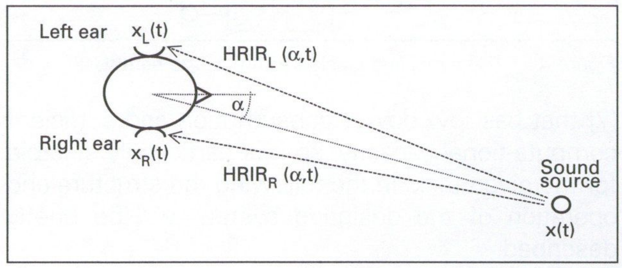

Figure 3 The principle of spatialization

These considerable computational requirements are not af-fordable for many low-end applications. However, in such cases, simulating merely the Interaural Time Difference (ITD) and Interaural Intensity Difference (IID) proper direc-tional sensation can be achieved. With other words, the direction of a sound source can be made perceptible accurately without any real spectral transformation, using only time delaying (ITD) and amplification (IID). However, at this method, the spatial sensation is absent, the listener feels as if the source was in his head – though in the given direction – therefore it is called lateralization [7]. A difficulty is that the IID is highly frequency dependent; the ITD – as expected – is substantially independent of the frequency. Summarized, lateralization can be an alternative method of 2D spatializa-tion to indicate a direction through the auditory display espe-cially for narrow band signals and to achieve low computa-tional requirements.

The designed system

As we introduced, our basic aim is to produce an instrument for the blind, which is able to indicate the environmental obstacles by such a stereo sound effect as if it had originated from the obstacles themselves. Our purpose is to detect only the nearest (most dangerous) object in front of the user and to show its horizontal position through the auditory display. To accomplish these tasks we have decided to utilize the tech-niques described in the previous sections (2-D echolocation and spatialization) in a digital signal processor based portable instrument. We have selected an inexpensive fixed point DSP

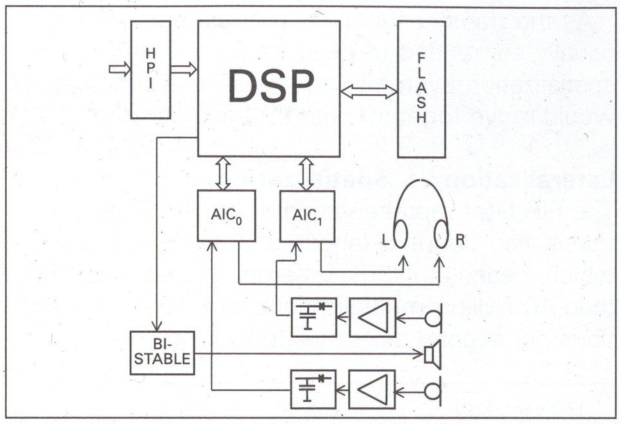

Figure 4 The simplified block-diagram of the hardware

[7] that has low power consumption and sufficient computa-tional capacity, so it is particularly suitable for the given task. In the following the structure and operation of the designed system will be briefly described.

The Structure of the System

The simplified block diagram of the hardware can be seen in the next page in Fig. 4. The main characteristics of the ap-plied TMS 320VC5402 type DSP are: 16-bit fixed point, low voltage (V_{Core} =1.8 V, V_{I-O} =3.3 V), 100 MIPS max., 16 k word on-chip RAM.

The Host Port Interface (HPI) of the DSP is used to download the program from a host PC to the DSP, but the processor can boot from the flash RAM too, once the program is stored. This is a very flexible configuration, since the modification of the software through the HPI is quick and simple while the flash RAM provides portable operation.

Two parallel Analog Interface Circuits (AICs) type TLV320AIC10 are used with dual purposes. They digitize the envelope-demodulated echo signal of the left and right re-ceiver and provide the digital-to-analog conversion of the binaural sound for the earphones. The parameters of the codecs (22 kHz sampling rate and 16 bit resolution) are suit-able for both tasks [10].

As Fig. 4 illustrates, the signals of the receivers are amplified and then envelope-demodulated. Since the signal envelope contains all available information from the environment (ex-cept the Doppler-shift, which is neither significant nor useful for us), the digitization of the original modulated carrier is not necessary.

The ultrasound transmitter is driven by the DSP's timer out-put through a flip-flop producing a frequency of 40 kHz. Lower frequencies would be better considering attenuation, but it would also disturb sensitive animals (like guide-dogs), which is not allowable in our case.

Some important auxiliary circuits used for impedance matching, etc. are not shown in the figure.

The Operation of the System

The simplified flow chart of the software can be seen in Fig. 5.

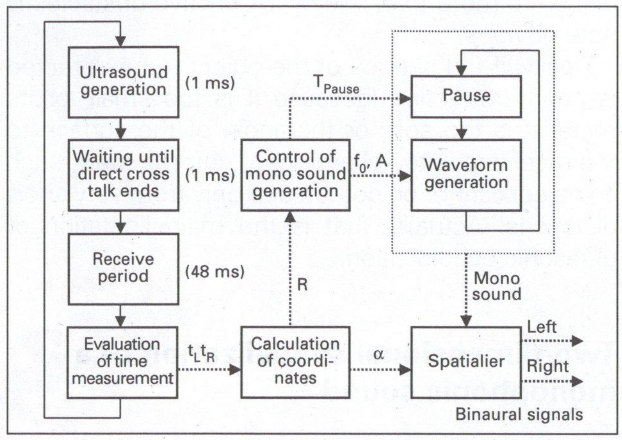

Figure 5 The simplified flow-chart of the software

The time length of the echolocation is 50 ms. This is in corre-spondence with the attenuation of the echo signal, which is satisfactorily high after this period not to disturb a new meas-urement. Actually the timings of the localization procedure have been experimentally determined [1].

During the receive operation the samples of the two envelope signals are stored in the DSP RAM (requiring 2 kword mem-ory). Then, at the evaluation, the time delays are determined by a time (distance) varying threshold method and using also linear interpolation between samples.

The calculation of coordinates is based on (4) and (5); the DSP implementation of (4) is trivial, the directional angle (?) is determined by mapping to a time delays' difference table placed in the DSP memory.

Parallel with the echolocation, a mono signaling sound gen-erator loop is running. As symbolized in Fig. 5, a waveform followed by a short pause is generated continuously. The selected waveform itself is planned to be a digitized signal of a musical instrument but could be any other signal pleasant for the human hearing. The base frequency (f_0) and amplitude (A) of the generated waveform as well as the length of the pause (T_{Pause}) are adjusted on the basis of the distance (R). The closer is the obstacle the higher is the pitch and the vol-ume and the shorter the pause. The nonlinear mapping of the distance to the previous parameters is done also with the help of tables stored in the DSP memory.

The final step is the 2-D spatialization of the mono signaling sound in accordance with the measured direction (α). The left and right HRIR functions are placed into the RAM, and a pointer indexed by α selects the left and right FIR filters' parameters among them. The original HRIR set publicized in [8] results in 128 tap FIR filters for synthesis at 44.1 kHz, but as we use 22 kHz, they are down-sampled, halving the filter length to 64. Thus we lose the frequency components above 10 kHz, which degrades the efficiency of the spatialization, but reduces also the computational and memory require-ments. Since the worst-case direction error is not less than 5^0 (5), and the maximum detectable angle is not more than 750, it is adequate to store 16 HRIR pair (with 50 resolution be-tween -75^0 and $+75^0$) requiring 2 kword RAM. The reduction of the computational load is even more convincing: for the real-time two-channel convolution, only a total of 5.7 million arithmetical instructions per second are needed. Because this number is only a small fraction of the capabilities of the DSP, power savings can be achieved by decreasing the rate of the master clock and/or using power down modes in the remain-ing time.

The parameters of the spatialization are refreshed 20 times per second, so even rapid changes of the environment can be followed as long as the localization is reliable.

The Portable Device

The portable device consists of a central unit attached to the ultrasound sensors and to the earphones through wires. The central unit contains all electronic parts currently on two separate boards connected to each other. This is for flexible experimentation, as one board is the digital part including the analog interfaces, and the other one is the analog part (ampli-fiers and demodulators). We plan to integrate the electronic parts into a Walkman like instrument. The ultrasound trans-ducers and earphones are mounted on a baseball-hat so the wearing of the navigation aid should not be uncomfortable or conspicuous. Since the sensors as well as the earphones are fixed to the head, no head tracking is required.

Experimental results

The previously described device is currently under manufac-turing; therefore we could make measurements only with an earlier, experimental version. The experimental system is based on a Starter Kit [11], completed with additional circuits on several boards. In fact, the experimental and the designed new system are nearly identical both structurally and func-tionally. The main differences of the experimental version to the designed one are: no flash RAM, an earlier DSP type and AIC [11], (more power consumption, less RAM and computational capacity...).

In the followings we briefly introduce some measurement results obtained with the experimental device.

Digitization of the Echo Signals

At this experiment the device was connected to a PC, and after a localization measurement the left and right digitized envelope signals were transferred to the host PC from the DSP memory. Then the data were displayed with the help of MATLAB functions. Fig. 6 shows the received signals when the sensors were directed to a wall in a distance of about 1.3 m in a normal (not empty) room.

The direct impulse and later, at about 7.5 ms, the first re-flected impulse followed by other reflections and reverbera-tions are visible on both figures. The pictures show only the first 30 ms of the signals to better emphasize the details.

Localization Tests

At this sequence of experiments the accuracy of the two-dimensional object-localization was investigated. The ex-perimental device was connected to the host PC and the left and right time delays (t_L, t_R) – measured with the simple threshold method – were transferred to the computer. The device was directed to a wall in various distances and direc-tions. At a given position several localization measurements were performed to provide data for statistical analysis. The results are presented in Table 1.

We also experienced that the maximum detectable distance is about 6 m, and the maximum detectable direction angle is between 40^0 and 50^0. Also it was agreed that a more sophisti-cated echo detection method would be essential for more precise and reliable direction detection.

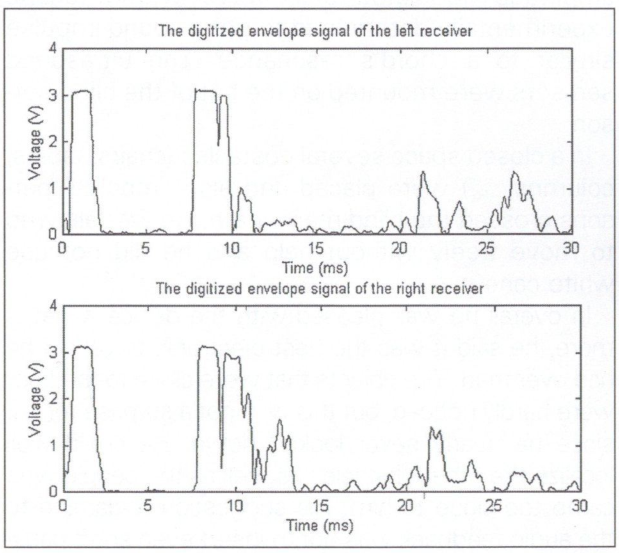

Figure 6 He digitized envelope signals of a localization measurement from the DSP memory

(N is the number of measurements made at the given configu-ration. R' and a' are the measured coordinates, their mean and standard deviation were calculated using the Matlab mean(.) and std(., 1) functions.) The standard deviations of the mea-

Actual coordinates		Statistics of measurement results			
R [m]	α [°]	N	Mean(R') [m]	Mean(α') [°]	Std(α') [°]
0.97	22.6	176	0.98	30.1	0.6
1.26	15	198	1.27	16.4	3.9
1.42	10	202	1.43	10.7	2.0
1.60	5	222	1.61	8.5	2.0
1.61	0	187	1.63	0.3	1.2
1.64	-5	189	1.65	-3.7	3.8
1.45	-10	163	1.47	-8.8	3.3
1.21	-15	173	1.23	-16.0	3.3
0.86	-22.6	251	0.88	-19.6	0.0
1.87	0	197	1.87	-0.2	3.0
3.26	0	169	3.24	6.2	4.8

Table 1 Localization measurement results

sured distances are not presented in the table because they were less than 1 cm at each localization sequence.

Subjective Test

To investigate the usability the navigation aid – and to verify our ideas – a blind man was asked to test the experimental device.

During the subjective test the DSP was connected to the PC only for program download. Again, the simple threshold method was used for time measurements. In order to reduce the complexity of the software, the lateralization method was applied in the instrument instead of spatialization. The wave-form of the signaling sound was an amplitude-modulated sine wave with a shape experimentally determined to get a sound impulse similar to a chord's resonance. The ultrasound sen-sors were mounted on the hat of the blind person.

In a closed space several obstacles (chairs, tables, columns, ...) were placed and also 'impolite' persons crossed the blind man's path. He was allowed to move freely without help and he did not use white cane.

In overall he was pleased with the device, what is more, he said it was the best electronic travel aid he had ever tried. The objects that were close to the floor were hardly noticed, but this was not a surprise for us, since he nearly never looked down. He could well localize the other obstacles as well as the people who came too close to him. His suggestion re-garding to the audio feedback was not to insert even short pauses into the signaling sound.

Conclusions

A navigation aid for the blind designed at our laboratory was introduced in this paper. The system detects the nearest ob-stacle in front of the user, using 2-dimensional echolocation, and indicates its horizontal location by real spatial sound. In the future it can be supplemented with a third ultrasound receiver to be able for 3D localization and feedback. The designed instrument is not completed yet, but an experimental version proved that its principles are suitable and usable in practice. However, the system cannot solve the blinds' ultimate problem of the environment perception. It has limits due to the characteristics of the ultrasound reflections such that many object can barely be detected, which have very small or soft surfaces. The directional characteristics of the transduc-ers also bound the range of the localization. Despite these difficulties, we do hope that the designed instrument will efficiently aid the blind at travel and every day hard life be-cause it is based on the natural hearing abilities of the hu-mans.

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Public application of speaker verification algorithm using both dynamic and static parameters

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Introduction

Automatic systems recognizing a speaker's voice are divided into two groups: automatic identification systems and automatic verification systems. Verification systems enable or disable access to an object by a password. Automatic verification systems are used in military, forensic and commercial applications. This paper describes one of the possibilities where to implement a speaker verification system.

Where can we implement a speaker verification system?

Secure access voice identification can be used whether or not in combination with fingerprints, facial information, identity card or signature. If we leave out military and some other special security systems, there are a lot of commercial possibilities where to implement voice security systems. Let us take, for example, safes, quick access to doors, car protection against thefts, the protection of electric systems like TV, video, and last, but not least the protection of computers.

If we take computers, there can be various ways how to protect our data, or prevent direct access to computer. These are:

- Data protection by coding files, and their decoding controlled by a voice identification system
- Protecting the computer from fast access by screensaver that uses speaker verification system
- Protecting the whole computer directly at the booting stage

This paper outlines the screensaver version.

The screensaver version of speaker verification systems

The screensaver described here is an application running under Windows OS. Roughly speaking, this application is a speaker verification system implemented in an ordinary screensaver program with an "scr" extension and copied into the system directory. The application, which is activated by the OS, runs, till the first interruption is sent from the mouse or the keyboard. After the interruption the computer verifies the identity of the user via his voice password. We distinguish two very important modes of screensaver applications: the setting mode and the active mode. These modes are described below:

The setting mode of the screensaver

The setting mode serves to create templates and set thresholds for the verifications. As we can see in Fig. 1., our verification system will check both name and the password of the user. Therefore the first thing we should to is to set the system by creating a template recording.

These templates will serve to compare the speaker's voice the moment he tries to access the computer.

After creating several templates it is recommended to test them. There are two buttons to test the

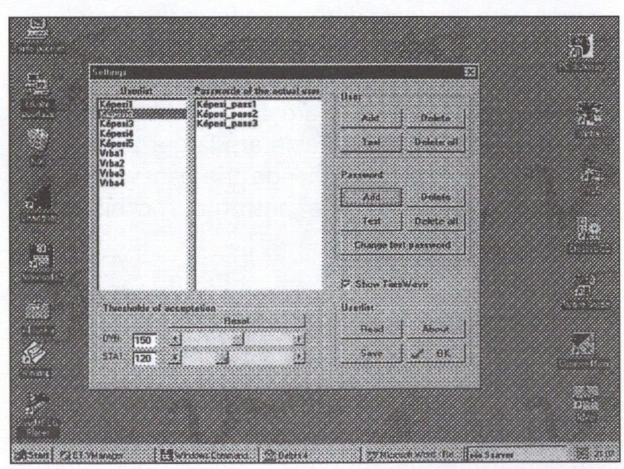

Figure 1 The "setting window" of the screensver

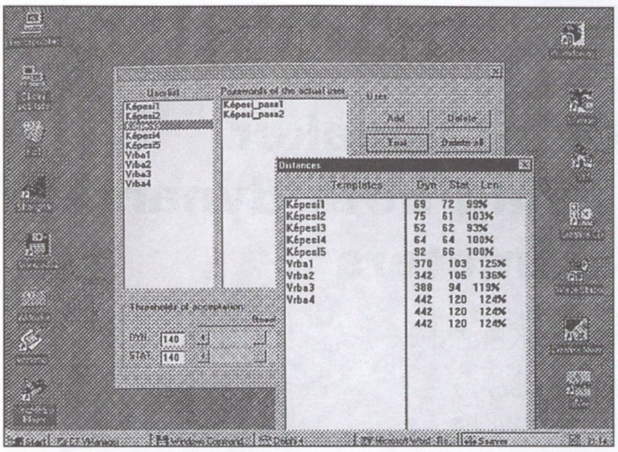

Figure 2 The table of distances appears when we test our templates

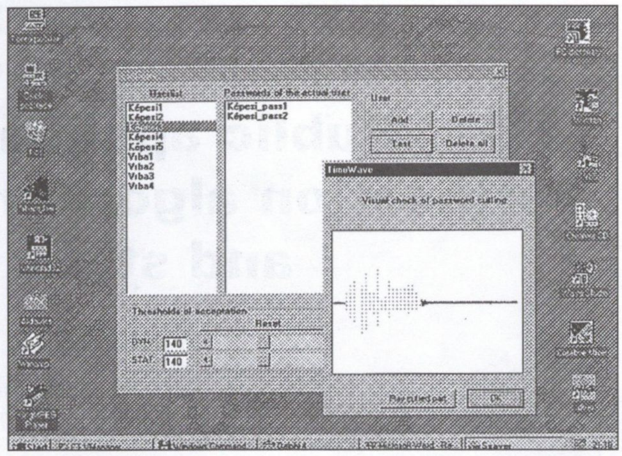

Figure 3 The "TimeWave window" of the screensaver serves for the visual checking of password cutting

references created. First we can check our username. The question is how close to the templates the repetition of our name is. Everything depends on this similarity, which is indicated by the distances between the actual (i.e. test) and the referential templates. In the setting mode these distances are always displayed in a table, see Fig.2. The displayed values are: dynamic distance, static distance and the length of the actual recording compared with the lengths of all the references. This table is used to set the thresholds of acceptance. These thresholds will described below.

Choosing passwords

As we know, there are some speech sounds which are very hard to distinguish from the noise in sampled digital form (f, s, h, t, p...). We should choose passwords, which can be cut out correctly by the computer in cases when a noise appears in the background, For this purpose a window has been added which shows the cutting of the chosen password. The designation of the first and the last segments in the recording. Fig.3. shows the "Timewave" window when a background noise is present. The user's aim should be to find a password, which he can pronounce with the least difference from the templates. In this case, the table with the distances (Fig.2.) is very useful because it gives an opportunity to check the similarity. When we are testing our templates and the distances in the table are larger than any of the set thresholds or to change the passwords. The smaller the distance, the better our choice of a password is.

The active mode of the screensaver

The screensaver could be activated either manually or by the OS after a set period has elapsed. When creating a screensaver, the first thing we should do is to deactivate the Ctrl-Alt-Del, Alt-Tab, and Ctrl-Esc system-key combinations.

After an interruption has come from either the mouse or from the keyboard, the computer first asks from the username. If ht user has been recognized, the computer reads the user's password-list (see Fig.1.), and after that it asks the user to say his password. This password will only be compared with the user's password templates.

As we have mentioned before, there are two thresholds in the system which determine whether the actual username or password will be accepted or rejected. There are two thresholds because there are two algorithms testing simultaneously the similarity of the templates

The verification is done in two ways at the same time: checking the parameters of the speaker's vocal tract and testing the correctness of the password. The first method uses static symptoms, which represent the vocal fold parameters of the speaker. In the second case, where the dynamic parameters are used for recognition, a matrix of dynamic parameters is stored for all references. Since these matrices are not of the same width, the Dynamic Time Warping algorithm has been implemented for lenght normalization.

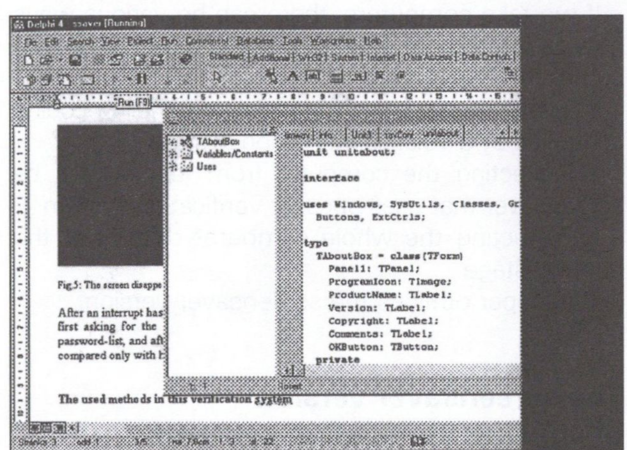

Figure 4 The screen disappearing and appearing in the active (saving) mode

Parametric representation of speech in the verification system

Digitizing the signal:

After the computer has asked for the username of the password, a recording process is started, and the computer will create a temporary file, containing the most recent pronunciation. The signal from the mic is passed through a lowpass filter with a cutoff frequency about 5kHz and digitized at a sampling rate of 11kHz with 16-bit resolution PC board. The signal to noise ratio, which strongly depends on the mike, will always be better than 25dB. In this work no emphasis is applied.

Designation of the first and last segment

Fig.5. shows how prominent the energy difference is between the clear segment and a segment with speech signal. The Figure demonstrates an idal case, where the recording is without any noise. In the case that noise appears in the background of recording, it can happen, that the energy of non-speech-segments before or after the active password is relatively high. To avoid this situation the algorithm was complemented with an energy check of 3 successive active segments as active or not active is done by comparing the energy of three frequency bands. Fig. 3. Shows the password cutting where a background noise is present.

By continuous segmentation we search for the first three active segments. The first of these three segments will be denoted as the first segment of the password, and the last active segment from the entire temporary recording will be denoted as the last segment.

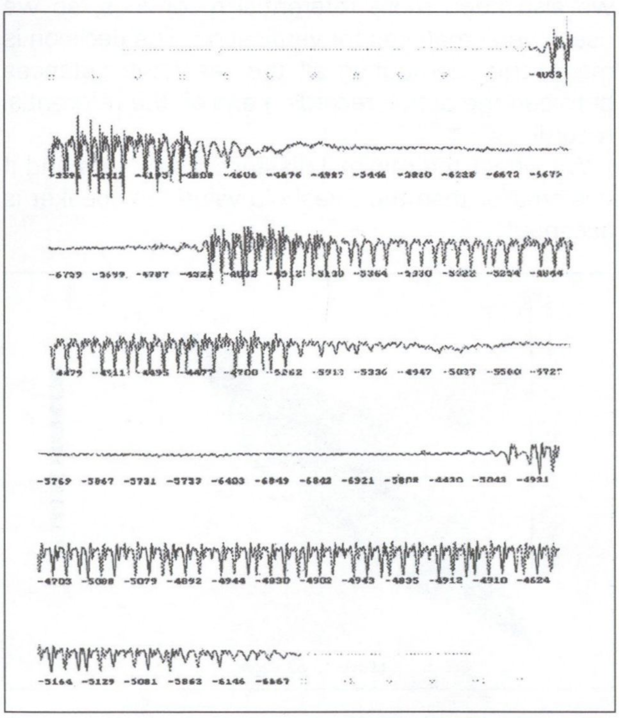

Figure 5 Automatic designation of the first and last segment in case of clear signal

Parameter representation

Dynamic parameters

The length of the cut segments, which are considered to be active, is 23.17 ms. This length corresponds to 256 samples. This verification system uses 50% overlap of analyzed segments. In the case of dynamic parameters, a vector of 30 delta-filtered cepstral coefficients was stored for each segment. In the literature these coefficients are referred to as Orthogonal Partial Cepstral Coefficients OPCC, and are computed by the following formula

$$\Delta C_i(t) = \frac{\sum_{k=-K}^{K} k \cdot C_i(t+k)}{\sum_{k=-K}^{K} k^2}, \qquad (1)$$

where C_i is the i-th Linear Cepstral Coefficient, and ΔC_i is the i-th Orthogonal Partial Cepstral Coefficient. The number of segments used for one computation is 2K+1. Currently filtered are sections of 180 ms, which corresponds to 9 analyzed segments with 25% overlap. In this case K=4. Generally, this filtering is applied to time trajectories of STFT.

Dynamic parameters of referential and test recordings are compared in the DTW-based procedure described in the following.

Static parameters

The second method implemented in this system is based on the static vocal fold parameters. This means that only one vector containing the mean values of chosen parameter representation describes the whole password. Both the LFCC and the MFCC coefficients were tested for our purpose. The results were nearly the same so we use the LFCC vector in the following.

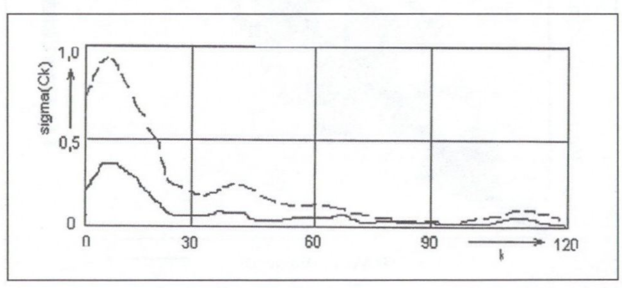

Figure 6 The mean scatter of distances between referential and test LFCCs. Full line: referential and test records of the same speaker, dotted line: referential and test records of different speakers.

To obtain static parameters, we only use voice segments so that the static behavior of the speaker's vocal tract is represented by a vector of mean values of the Linear Cepstral Coefficients we apply the weighted distance

$$D1=\sum_{m=1}^{L} w_m(x_{im}-y_{jm})(x_{im}-y_{jm}) \tag{2}$$

where $w_m=1/V(m)$ The weighting function was derived from Fig.6.

$$V(0)=V(1)=0$$
$$V(k+2)=\exp(-k/7)+0.2\,,\quad k=0...N \tag{3}$$

Description of the dynamic time warpin algorithm

This is a simplified form of the original algorithm of Furui, where every segment is represented by a vector of above mentioned coefficients. The whole DTW algorithm is divided into these parts: computing the local and cumulated distances, finding the right time-transform function, and computing of the final distance.

Computing the "Region"

Let us have the same word with two pronunciations of different lengths, i.e. with a different number of segments as shown in Fig.7. Axis X always represents the longer word, and axis Y represents the word with fewer segments. To compute the "Region" means that we will compute all the local distances between the i-segment of the longer word and the j-th segment of the shorter one.

By definition these distances are computed only for the active area. This area is specified as shown in [5]. The distance between two vectors of OPCC coefficient is:

$$D2=\sum_{m=1}^{L}(x_{im}-y_{jm})(x_{im}-y_{jm}) \tag{4}$$

Computing the Cumulated distances

This is done step by step along axis X. This means that we look for the respective equivalent segment cut from the shorter record for each segment of the longer record with the least distance. The segments along axis X are applied only once and none of them is left out. We did not use any weighting function. The only local limiting DTW function was used according to Fig.8.

The correct transformation function

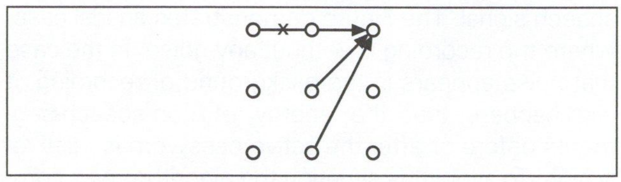

Figure 8 The local limiting function of DTW used in our system

Is sought step by step by means of the minimal cumulated distance.

The resultant distance is given by the last value of (D[I,J]) divided by the number of analyzed segments in the longer utterance. If we only verify one person, we also need more referential recordings, so we use several matrices for verification. The decision is made after computing all the resultant distances between the actual recording and all the referential recordings.

We select the minimal distance from them and if it is smaller than the threshold value the speaker is accepted.

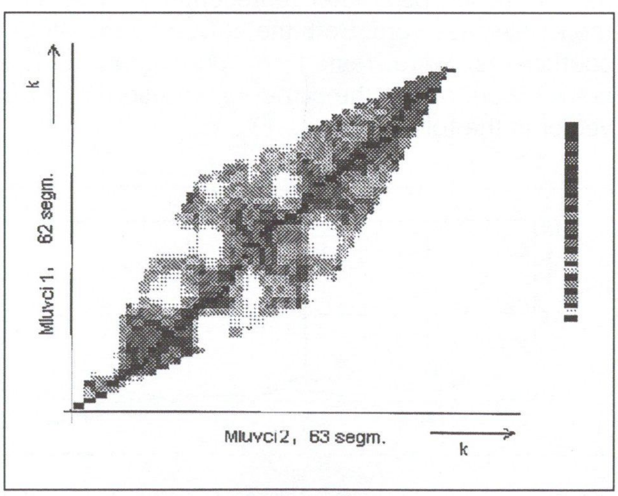

Figure 7 Local distances. Referential and test recordings of the same speaker. The vertical strips represent increasing distance intervals.

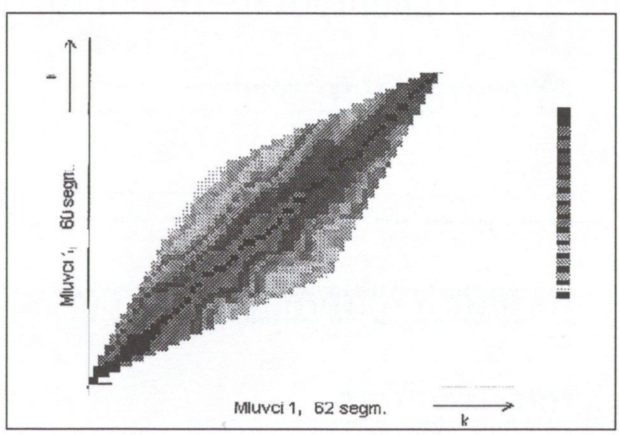

Figure 9 Cumulated distances. Referential and test recordings of the same speaker. The resultant distance is 58. The vertical strips represent increasing distance intervals.

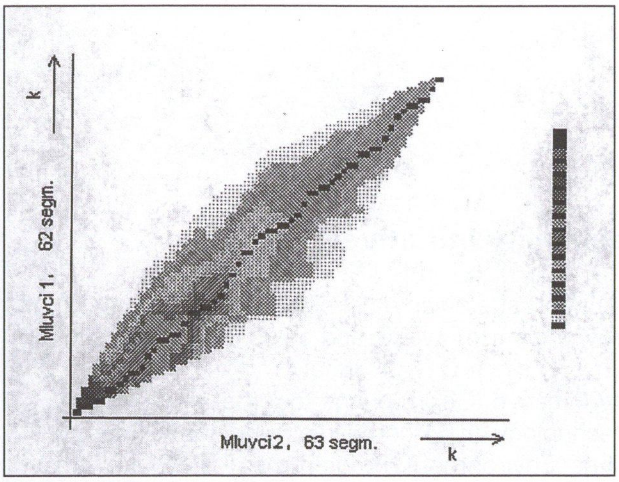

Figure 10 Cumulated distances. Referential and test recordings belong to different speakers. The resultant distance is 182. The vertical strips represent increasing distance intervals.

Conclusion

The screensaver described in this paper represent only one of the possibilities where speaker verification system could be implemented. This form of application is very simple and fast htanks to easy implementation in computers, and it doesn't require any HW expansion. The only thing we need is a microphone and a speaker system.

The verification system uses both dynamic and static speech parameters. The comparison of dynamic parameters is done by the DTW algorithm and the speech signal is represented for each segment by a vector of OPCC parameters. We obtain these delta coefficients by delta filtering of cepstral time trajectories.

Beside these dynamic parameters there are also static derived parameters, which describe the vocal tract of the speaker better than the dynamic ones do. The inadequacy of the system described here is the application of a one-level DTW method. This result in limiting the password to a one-word password. In the case that we are pronouncing our name, we should pronounce it as one word in order to preserve continuity. In spite of this insufficiency this application of the verification system could be very efficient and reliable.

Acknowledgement

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News

Launch of the Global Environment Sustainability Initiative

An initiative to improve the global environment and support sustainable development by promoting business practices and technologies that saves energy, minimizes waste and helps bridge the "digital divide" was launched on the occasion of World Environment Day (5. June)

The new alliance, called the Global e-Sustainability Initiative (GeSI), brings together some of the world's biggest information and communications technology (ICT) companies and their industry associations and is supported by the United Nations Environment Programme (UNEP) an the International Telecommunication Union (ITU).

A key contribution of ICT to environmental protection is in the transport sector. Videoconferencing, tele-banking, tele-learning and teleshopping, for example, can eliminate the need for travel so reducing the traffic (less congestion and pollution) and the emission of greenhouse gases - the major cause of global warning.

The participating companies in GeSI have agreed on an exciting range of activities ranging from environmental management in their internal operations, to exploring options for remote and disadvantaged communities in developing countries to get online.

Over the next two years, the GeSI will support research on the role that information and communications technology can play in advancing sustainable development - climate change, waste reduction and the digital divide are among the main issues that will be addressed first. Participating companies are also looking into how best to "outreach" their knowledge and experience to enable businesses around the world to take new opportunities and expand markets while displaying corporate social and environmental responsibility at the same time. All GeSI members are striving to improve their own internal environmental performance.

Evolution of network management technologies

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Internet Service Providers (ISPs) introduced numerous new services and telecommunication infrastructures in the recent past, continuously. This process will go on in the future and we will look in the face of inhomogenous, meshed and distributed network infrastructures. Not only the evolution of topology and the structure of networks operators but also the new services are challenges for network management systems. In this paper, an overview will be given about the traditional methods and systems from these points of view and some requirements and outlooks of the recent research aspirations will be discussed. The focus of this paper is a review about the research ambitions of the new network management architectures. **This work has been supported in part by the European Union in the context of ACTS, project ELISA (http://www-st.inf.tu-dresden.de/elisa/).*

Introduction

Today's telecommunication, IT networks and network management are changing rapidly in terms of size, technology and the number of organisations involved. Current network management and control software is oriented toward servicing single administration domains. The need of QoS drived and distributed network architectures is a big challenge at the present. Specifically, to provide an extensible approach to the dynamic integration, management, and runtime assessment of various network protocols in live network operation is necessary. In this paper, the management aspects of this challenge will be highlighted.

Some aspects of the traditional management systems

The ISO/OSI approach of management systems are no more sufficient for recent services in the field of internet and/or IT networks. The OSI approach is basically a static and centralised layered model with separated management functions and software modules, running on mainframes and/or management workstations. The classical abstraction model of the network management approach (Fig. 1.) isn't fully appropriate for modern management architectures. First of all, the telecommunication and IT provider structure isn't separated and centralised any more. Second, in an inhomogenous network, the requirements of new services (QoS classes, protocol's, security requirements, etc.) aren't manageable with these centralised architectures any more. Fore example, the migration of the networks depend on application requirements, QoS requirements, security requirements, provider structure, routing technics, etc., the security management depends on provider structure, applications, etc., the billing-accounting management depends on architecture, provider structure (and agreements), QoS classes in the network, etc. As a consequence of the above mentioned mutual relations is, that instead of a centralised management we need a distributed architecture for service providers.

The ITU-T, ETSI, ISO and IETF approaches follow the layered models, such as TMN, and layered protocol stacks such as CMIP or SNMP, and the realisations of

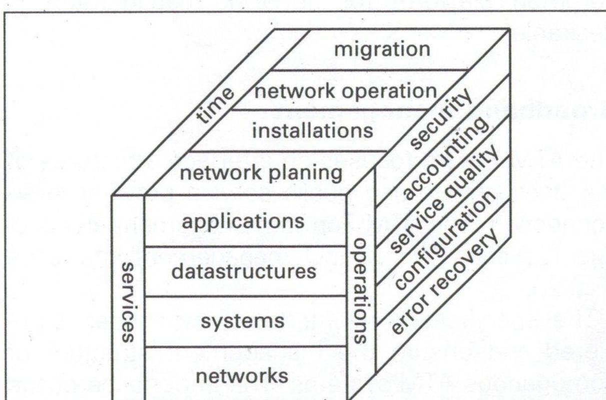

Figure 1 The classical management system structure

these architectures are application independent, autonomous systems. The Telecommunications Management Networks (TMN) approach is an interconnection of the carrier's Operation Systems (OS) and the packet or circuit switched data networks using one of the TMN standard interfaces (X, F or Q). ITU-T propose for this connection the ISO/OSI connection protocol CMIP and for the data structure the Common Management Information Service Elements (CMISE) structure. The next section presents some management aspect of high speed wide area networks' management.

State-of-the-art study of wide area networks management

In the following joint IP/ATM infrastructure will be discussed a from QoS requirements' point of view.

Management of IP traffic over ATM networks

The IP over ATM technics spreading out and the management of this network focusing on traffic management, QoS management on ATM cell level and IP level (packet and cell loss statistics, error recovery, etc.) moreover on congession management, configurations management and admission control. The present management systems don't support the classification of services, more specifically, the application-dependent QoS architectures. In order to realise such a system, first of all, we need a scalable network such as the Resource reSerVation Protocol (RSVP) supporting a routing scheme and management architectures, which handle these reservation technics. The ATM itself supports service classes, but the mapping of the requirements of applications on ATM classes isn't possible in the mentioned protocol stacks. The use of SNMP as a management platform is possible over an IP/ATM network architecture. The coexistence of the internet and the ATM (and others high speed networks) technology is predictable and a common platform for network management is desirable.

Broadband management

The ATM Forum focuses on interface structures of the user, private and public service provider interconnection. The ATM Forum management architecture identifies five distinct management interfaces (Fig. 2.).

The specification of interfaces have been completed and ensure the management structure of homogenous ATM systems. The importance of the ATM Forum standard is significant between public and private network operators, but not popular at user parts because native ATM services seldom work at users.

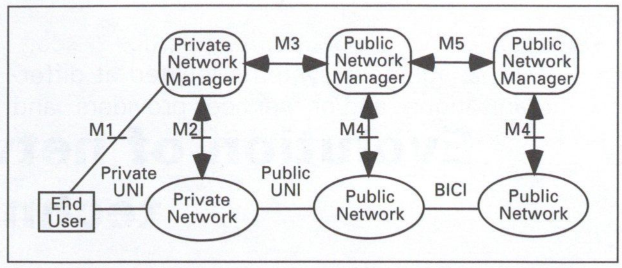

Figure 2 The ATM Forum management architecture

The ATM Forum standardised MIB (TM 4.0) structure include traffic management class information (the atmVpcTable and atmVccTable MIB tables) provides per connection traffic management. The extension to the SNMP MIB defined in [11] will allow for the whole ATM Adaptation Layer (AAL1÷5) structure and the Service Type structure to carry information regarding the configuration of ATM. Some broadband research projects are focused on the integration of new network elements and functions into standardised network management architectures. One of these is the TERRACE [15] project, which presents a TMN based model and extends the model with broadband specific behaviours (i.e. traffic control, admission control, congestion control).

In the next section we will dealing with distributed management architectures and research aspirations on this field.

Distributed network management approaches

A Gateway based approach

In this section, an extended gateway based cooperative TMN approach will be described. To fulfil the requirements of a cooperative and distributed management architecture we can extend the conventional TMN architecture with gateways, which transports the management information over a heterogeneous (partly not ITU-T like) network.

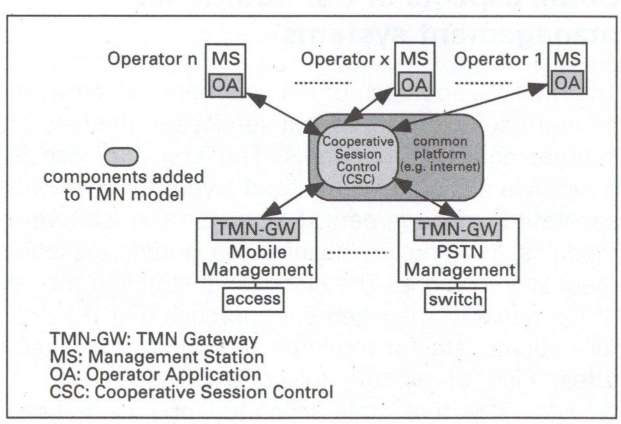

Figure 3 Cooperative TMN model

Fig. 3. shall give an overview of the cooperative TMN architecture. The cooperating parties are conventional management systems located at different organisations and/or service providers and human network operators. The functionalities of the system components allow for operators to operate over a distributed resources as described below.

The TMN gateway (TMN-GW) receives commands, guidance information and data for action to be performed by cooperating parties. The TMN-GW include the conversation of commands, messages, and handle the standard management functions (authorisation, fault reports, alarm reports, etc.). The operator applications (OA) are responsible for interaction between human operators and the cooperative session control (CSC) system. From realisation point of view a Java based environment may be a good choice for this application platform.

The CSC controls the corresponding multi-party sessions based on information which is represented by multi-party task descriptions. The task descriptions are represented by object oriented hypermedia documents and using for example the internet technology. It is possible to realise this representation with RFC recommendations e.g. IP, HTML, HTTP, etc.

Open Distributed Processing

The Open Distributed Processing (ODP) approach basically is an abstract model for management systems with five view points. The concept of ODP isn't to realise a new formal description, but to use such models and specification technics, which are standardised and used at present. These viewpoints (namely: enterprise, information, computation, engineering, technology viewpoints) describe the networks and support the model creation and simulation. The ODP isn't a vertical layered system (such as the OSI model) any more but also a distributed model with mutual influences between the viewpoints.

Common Object Request Broker Architecture

The Common Object Request Broker Architecture (CORBA) is an object oriented approach with Object Request Broker (ORB) technics. The ORB manages the control transfer and data transfer to the object implementation and back to the client. The clients in the architecture have access to an object and can perform operations on it. An object implementation provides the semantics of the object, usually by defining data for the object instance an code for the object's method. Fig. 4.a. provides a simplified view of the CORBA architectures.

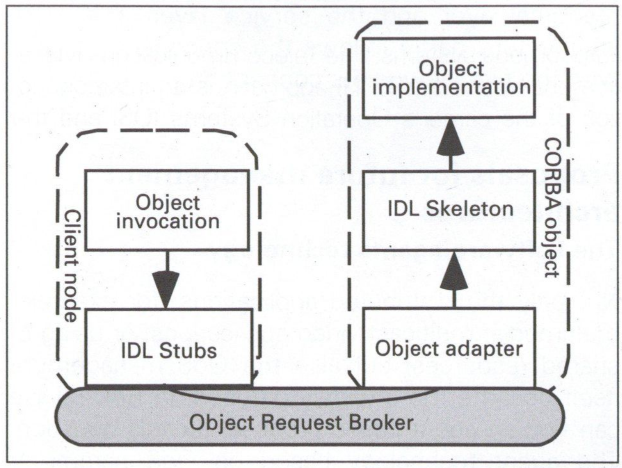

Figure 4a CORBA system architecture

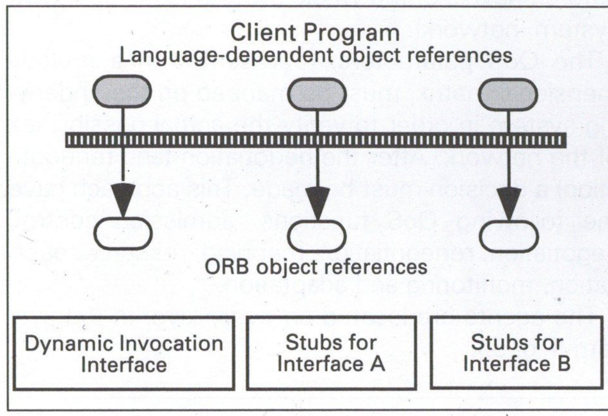

Figure 4b The structure of a Typical Client

The CORBA architecture is a distributed Client-Server architecture with an implementation of independent data structures. Fig. 4. b. presents a client structure with different interfaces. The CORBA ORB architecture is designed to allow interoperation with a wide range of object systems. A foreign Object system can interoperate with the CORBA architecture via the ORB gateway.

An application based on CORBA can be seen as a collection of independent software components or CORBA objects. From applications' point of view the CORBA architecture is a well-defined system and with synchronisation procedures it is allowed to build up a cooperative, distributed network management architecture.

TINA-C

Telecommunication Information Networking Architecture Consortium (TINA-C) presents an object-oriented approach with tree (sub-)architecture elements: Service, Network, Computing. The Computing element is an adaptation for telecommunication requirements, based of OMG's CORBA conception. TINA broke the telecommunication system into three layers: the element layer, the resource layer and the service layer. The TINA architecture partly follows the conception of ODP with the five viewpoints conception.

Proposals for future management architectures

The software agents technology

Not only the distributed applications (for example: multimedia, multicast video applications, or using of shared resources) but also the QoS management itself become more widely diffused. In Ref. [2] we can find an agent based proposal for this question. The agent technology based on QoS parameter negations between the agents, which are located on different levels of the whole system (user, operating system, network).

The QoS parameters, represented in a multidimensional matrix, must be mapped on the underlying system in order to verify the actual possibilities of the network. After the negotiation (and renegotiation) a decision must be made. This approach have the following QoS functions: admission control, negotiation, renegotiation, mapping, resource reservation, monitoring and adaptation.

The agents are located on every layer in the system (Fig. 5.).

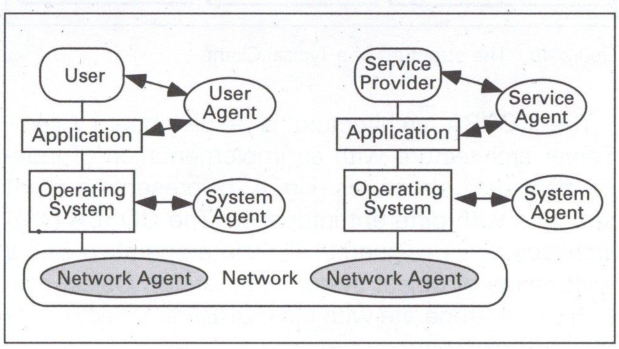

Figure 5 The system architecture of the agent technolog

In this architecture, the network is managed and controlled by a set of Network Agents that can be activated on demand. Each component involved has to contribute with certain guaranties on its own resources, when providing end-to-end QoS. Each Agent can collect information about the nearest agents to control their own resources and can influence (or influenced by) the behaviour of other agents. The software realisation of Agents are possible with Java applets which are downloadable on demand from servers.

The Agent technology is useful for enhanced QoS in the internet world. Recently, RSVP (and diffserv architecture) have been proposed to provide guaranteed quality of services as well. In praxis, network resources, such as routers, do not yet fully support RSVP. In a tunnel architecture [1] it is possible to define an overlay network over the non-RSVP router architecture to ensure the requested QoS requirements of applications. The tunnel architecture is dynamically configurable in order to handle the network failures and the renegatiotion between the changed application and/or user demand. The SNMP management protocol is an appropriate choice for the organisation of the configuration of the overlay RSVP capable network.

Active networks

Active networking is motivated by providing networks protocols that are more flexible and extensible. For forthcoming multimedia applications (multicast video, teleconferencing, collaborative applications, etc.) an active networks architecture will be proposed, which means in praxis as a heterogeneous group communication. We describe below a RSVP signalised QoS filter approach for these applications in an internet (AMnet) environment [7]. From the management system point of view the system working autonomous i.e. the filters loaded automatically on AMnet node if in the QoS negotiation phase it is requested. The impact of a such system is restricted on configuration, accounting and service quality management's functions. The RSVP protocol can easily extended to configure and control QoS filters. In this case, the user selected a desired QoS class and the network trying to satisfy this request. In the QoS negotiation phase the RSVP protocol (with a coupled RSVP daemon), which is extended with a QF daemon to exchange QF objects, build up the QoS filter multicast tree. Fig. 6. shall give an overview about an AMnet node architecture.

The AMnet approach based on active networks nodes provide specific services to individual users.

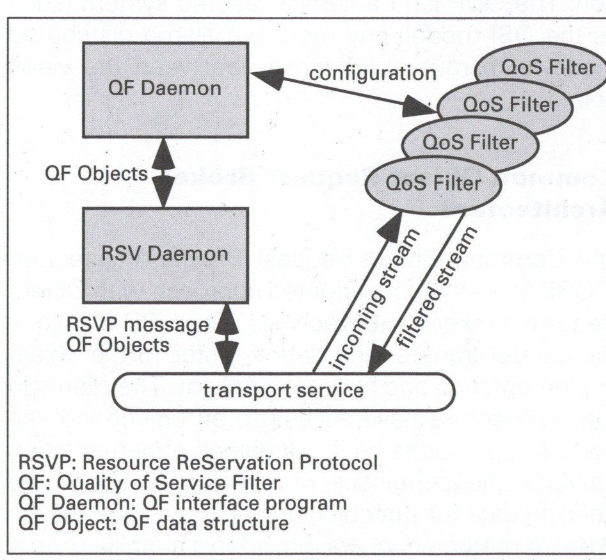

Figure 6 AMnet node architecture with QoS filters

The management architectures of active networks are under study. The main challenges in such a networks in key-words are the following: the management elements changing dynamically, the Management Informations Base (MIB) structure must be dynamically adjusted, the management of active components must be automated in runtime moreover standardised API is required for nodes to mapping the functionalities.

Open programmable networks

The problem is that current process of changing network protocols is lengthy and difficult. A programmable network is distinguised from any other networking environment by the fact that it can be programmed from minimal set of APIs. In the open programmable networks architectures views the network as a distributed programming system, and provides a programming language-like model for expressing new active protocols ('A' protocol family) in terms of operation at nodes [10]. The architectures of the system allows new (not yet standardised) protocols to be dynamically deployed at routers (and end systems), without the need for coordination and interaction between co-existing protocols. The interconnection of nodes may be on IP level or some other network protocol level which ensure the transport of the 'A' protocols. Different applications are able to introduce a set of 'A' protocols into the networks. These architectures support the software agents technology in all respect, but the technical requirements of nodes are high considering the present node (router) constructions. The node must be handle the traditional packets (such as IPv4 or others) and on-the-fly handle the new 'A' capsules. This approach allow for users (applications) to deploy protocols (rapidly, automatically and dynamically) those nodes they are needed and it isn't necessary consensus about the kinds or definition of the protocols. To pursue this goals we need an Applications Programmable Interface (i.e. networks API) from network management point of view to manage the mentioned architectures. The appropriate API candidate is under study [9], [14]. These APIs allow service providers to manipulate the states of the network using middleware toolkits. The whole open programmable networks architectures are in research state and the specification of APIs are in process.

Conclusion

The most significant trends in network architecture design are being driven by the emerging needs for global mobility, virtual networking, and active networks technology. Critical to the deployment and the management of these future networks is the need to provide consistency and control over dynamic change, and to limit the impact that such changes have on performance and stability. The gateway based architecture (described in section 3.1) is a pragmatic evolution step towards the TMN model.

An ODP based management realisation isn't yet exist. In this paper, we have discussed how a system management framework can be drived from current internet and active networks technology. It is also described services and their requirement in the context of new network management frameworks. To pursuing an unified paradigm for managing change in conventional and active networks i.e. in a heterogeneous environment is highly desired.

List of Acronyms

ATM	–	Asynchronous Transfer Mode
API	–	Application Programming Interface
CMIP	–	Common Management Information Protocol
CMISE	–	Common Management Information Service Environment
CORBA	–	Common Object Request Broker Architecture
CSC	–	Cooperative Session Control
ETSI	–	European Telecommunication Standards Institute
HTML	–	HyperText Markup Language
HTTP	–	HyperText Transfer Protocol
IP	–	Internet Protocol
IDL	–	Interface Definition Language
IPv4	–	Internet Protocol version 4
IETF	–	Internet Engineering Task Force
ISO/OSI	–	International Standards Organisation/Open Systems Interconnections
ISP	–	Internet Service Provider
IT	–	Information Technology
ITU-T	–	International Telecommunication Union-Telecom
MIB	–	Management Information Base
ODP	–	Open Distributed Processing
OMG	–	Object Management Group
ORB	–	Object Request Broker
RSVP	–	Resource reSerVation Protocol
RFC	–	Request for Comments
SNMP	–	Simple Network Management Protocol
TINA-C	–	Telecommunication Information Networking Architecture Consortium
TMN	–	Telecommunications Management Networks
QoS	–	Quality of Service

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Introduction to the world of mobile ad hoc routing protocols

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Mobile ad hoc networks are self-organising, quick-to-install networks made up from mobile tools, without any backbone line or a well-defined network topology. Due to the difficulties caused by mobility and the unknown topology, attempts to create a system that could be generally applied under any conditions failed so far. Consequently a lot of specific constructions have been developed to cope with problems arising in various areas. Notwithstanding their extremely different features the created ad hoc networks can be grouped into three major classes in terms of the applied protocols: reactive, proactive and hybrid ones – the last was developed through the combination of the first two.

The basic principle of reactive, or in other words, on demand protocols is as follows: minimum topology information is stored by each mobile node, and if the nodes want to communicate with each other then the required route information are determined before the data exchange. The main advantage of this principle comes from the fact that the nodes have to reserve a relatively small memory space for route entries which is essential for mobile sets due to the scarce resources. On the other hand searching for a route prior to communication results in a considerable delay in the transmission of data, thus the network becomes practically unsuitable for real time (for example multimedia) applications.

The basic concept of proactive protocols is contrary to the previous principle: in this arrangement the nodes always have the most up-to-date topology information for the whole network. Consequently the delay before data forwarding is reduced to the minimum, but the resource requirement is considerable.

Further advantages and drawbacks could be mentioned for both protocol types, but it is already clear that the two basic principles are in total contrast. The hybrid protocols that can be applied in a much wider range than purely reactive or proactive ones were developed with a view to bridge over the differences. Networks using hybrid protocols will presumably become more and more popular in the future, this is why the present article also lays great emphasis on their presentation.

Proactive protocols

The basic concept of proactive protocols is that the nodes maintain up-to-date, consistent routing data about all other nodes of the network. The nodes try to realise this with routing tables and regular updating messages. In the next part the specific features of three such protocols are analysed.

Destination Sequenced Distance Vector (DSDV) protocol

This protocol is based on an improved Bellman-Ford route searching algorithm capable of eliminating circular references. Every node stores the route to other nodes in the routing table. A table entry includes the following fields: destination identifier (address), the next node on the route to the destination, the distance and a unique serial (sequence) number. The serial number is derived from the destination. The protocol provides for the consistency of tables by means of periodic and forced (triggered) messages. In the periodic messages the protocol sends all "routing" information available for the given node, using, if needed, several protocol data units. With the forced message only the information changed since the last periodic message are sent by the node, this way the signalling traffic in the network can be significantly reduced. This message can be stored in one protocol data unit, and comprises the address and the distance of the destination as well as a serial number. The nodes use the route with the highest serial number. If a node receives two forced messages with the same serial number then the one where the route length is shorter is used for the updating of the table, that is the protocol is optimising for the shorter route. Before sending out the updating messages the nodes wait a certain time (stabilisation period) so as to be able to eliminate the undesired effect of messages arriving one after the other, comprising shorter and shorter routes. Otherwise the node would send a new updating message to the adjacent ones on receipt of each fresh message. By waiting for the expiration of the stabilisation period such cases can be managed, since the shortest route may be selected from the updating messages received during this time, and the selected one can be forwarded.

Cluster-Gateway Switching Routing (CGSR) protocol

In case of the application of CGSR protocol the nodes are logically organised into a hierarchy: the network includes nodes of various level and the clusters thereof. In terms of level or type the nodes are classified in three categories: gateway (connected to two clusters as well), clusterhead and simple node.

Through the application of clusterheads code separation as well as the efficient allocation of channels and bandwidth within the cluster can be realised. The clusterhead is selected according to a distributed algorithm. Frequent clusterhead changes reduce data transmission time therefore the LLC (Least Clusterhead Change) algorithm is applied by the protocol so as to minimise clusterhead changes. With this algorithm clusterhead selection is initiated only if two clusterheads are brought into direct connection with each other or a node gets out of the coverage area of all clusterheads.

The CGSR is based on DSDV (Destination Sequenced Distance Vector) protocol but the DSDV was modified so as to make the most of advantages offered by the hierarchical structure: routes lead only through clusterheads and gateways in the system. If a node wants to send a message then the message is addressed to the clusterhead, and the clusterhead forwards the message to the next gateway. The advantage of this arrangement is that only the routes leading to the clusterhead need to be maintained within the groups.

In this case the modified DSDV uses two tables: the traffic routing table and the so-called cluster member table. In the traffic routing table the next node where the message is to be sent to, the distance and a serial number can be found for every destination (target) group, while the cluster member table includes the destination node group assignments.

A typical route in the network is as follows:

$$C_1G_1C_2G_2 \ldots C_iG_iC_i+1,$$

where C_i is a clusterhead and G_i is a gateway

The gateway – gateway routing is not allowed because, due to the code separation, significant extra costs would perhaps be required for the communication of the two gateways (code harmonisation), moreover the clusterhead has always a higher priority when sending messages than the other nodes.

The drawback of the protocol is that apart from the route maintenance updating messages of DSDV further updating messages are also required to maintain the group, considering that cluster member tables as well have to be updated in the network.

Wireless Routing Protocol (WRP)

The purpose of the protocol is to maintain traffic routing information between all the nodes. Four tables are used by each node: distance table, traffic routing table, connection-costs table and Message Repeat List (MRL) table. The purpose of the protocol is to maintain routing information between all the nodes.

Message Repeat List (MRL) table

Each MRL entry contains the serial number of the updating message, the message repeat number, the "acknowledgement required" flag bit vector with one entry for every neighbour as well as a list showing the kinds of data sent in the updating message. The MRL table stores which updating has to be repeated and which neighbour has to acknowledge the updating message. An updating message includes several updates where the units inform their neighbours about the change of links. An updating data has the following fields: the address and the distance of the destination and the node preceding the destination.

If the link between two nodes is eliminated, the affected nodes send updating messages to their neighbours. The neighbours modify the distance table and search for a new route. If they succeed in finding a new route, this information is sent back to the nodes concerned.

The nodes "learn" about the existence of their neighbours from the acknowledgement and other messages. If a node fails to send data to a neighbour within a certain time then the neighbour assumes that the link has been released. This is why every node sends so-called "hello" messages to its neighbours in the absence of other messages.

The advantage of WRP protocol is the method applied for avoiding circular messages/references (ensuring acyclic architecture). The nodes communicate the distance of every destination node and the address of the preceding node in the network. The route searching algorithm evades the "count-to-infinity" problem, enforcing the consistency check of predecessor (previous) information. This way the convergence is faster if a link is released (eliminated).

Reactive protocols

The basic principle is as follows: the route from the source to the destination is identified only when requested by the source. In such cases the source calls for some routing algorithm which examines the possible permutations of routes. If the route selection was successful, the chosen one is maintained while requested by the source or while the destination can be reached via this route.

Ad hoc On-Demand Distance Vector (AODV) routing protocol

The AODV was developed through the upgrade of DSDV protocol. The advantages of AODV over DSDV are that fewer messages have to be broadcast and the nodes need not store information relating to several routes between the source and the destination. No route information are stored at all by nodes where no route goes through. This is essential from the aspect of memory management.

When a node (the source) wants to establish connection with another node (the destination) it examines whether there is a valid route between them. If no valid route is available the routing process is launched by the source. First a Route Request (RREQ) packet is sent to the adjacent nodes. The neighbours forward the received packets to their respective adjacent nodes and so on. This process is continued until a RREQ packet reaches the destination or arrives at an intermediate node which has a valid route up to the destination (fresh enough route).

If the above algorithm is applied it may happen that the identified route contains circles and/or not the latest route information are stored by the nodes. In order to avoid these problems a sequencing method is used by the AODV protocol. Each node has its own sequence number and a Broadcast Identifier (ID). The source increments the broadcast identifier upon every route request. The broadcast identifier and the identifier (IP address) of the node identify unambiguously a given route request. The source puts the broadcast identifier and the sequence number – considered to be the best – of the destination node in the RREQ packet. A RREQ packet received by an intermediate node is forwarded to the adjacent nodes only in the case where the destination sequence number stored in the intermediate node is lower than the one included in the packet.

During the forwarding of RREQ packets each intermediate node records the address of the neighbour which the first RREQ packet has been received from. When a RREQ packet reaches the destination or an intermediate node with a valid route up to the destination (fresh enough route), the node receiving the packet sends a Route Reply (RREP) message to the neighbour from which the RREQ packet arrived. The adjacent node forwards the RREP packet to the recorded neighbour, located on the route, and so on. During the forwarding of RREP message towards the source each node records the sender of the RREP message. When the source receives the RREP packet, all the nodes included in the route will know which neighbour the packets arriving from the source are to be sent to, thus the route may be considered as set up.

The nodes register a timer also for each route entry. If the route is not used within a predefined time then the entry becomes invalid. Route entries may become invalid also if one of the nodes on the route is dislocated. In such cases the source side neighbour of the dislocated node sends a link failure message towards the source. Every node having received this message cancels its route entry. If this message reaches the source, then the source has to issue a new route request provided that the link in question is needed.

Considering the protocol structure AODV can be applied only for symmetric links.

Dynamic Source Routing (DSR) protocol

If this protocol is used, the nodes are equipped with route cache memory units as well. Entries relating to routes applied by them are stored in these cache memory units. The content of the cache memory unit is updated when the node makes a new route entry.

If a node wants to send a packet to another node, first it examines whether there is a valid route entry in the cache memory. If there is a valid entry, then this entry is used for forwarding the packet. In the absence of such entry a route request packet is sent to the neighbours. This packet includes the IP address of the source, the IP address of the addressee and a unique identifier. Upon receipt of this packet the given node examines whether there is a valid route entry to the destination in its own cache memory unit. Should the result be positive a route reply message is generated.

The reply message is always generated if the node receiving the packet is the destination. The reply message contains the full route information (route record) comprised in the route request packet. If the node fails to find a valid route entry to the destination then puts its own IP address into the packet and forwards the packet to the neighbours. The broadcasting of messages is limited by the protocol in a way that the node processes only those messages that have not been encountered so far or where its own IP address is not included.

When created, the reply message has to be forwarded to the source by the generating node. If the given node has a valid route entry to the source in its cache memory, then this entry is used for sending the reply packet. In the absence of valid entry two cases are possible. Either symmetric links are used by the system, consequently the packet can be returned to the sender via the reverse route, or the links are not symmetric, therefore the node must launch a route request.

Cache memory units are maintained through the use of error packets and acknowledgements. Upon a node's unsuccessful attempt to establish connection with another node an error packet is generated. As a result of this error packet the node cancels its entry belonging to the other node in the cache memory unit. The entry in the cache memory becomes invalid also if no acknowledgement is received to the sent out packets.

Associativity-Based Routing (ABR) protocol

This protocol provides for the elimination of circular messages (references) and deadlocks in the system and prevents the appearance of duplicated packets. The operation is based on the degree of associativity stability. Each node periodically sends a sign of life to the adjacent nodes. As a result of this sign of life the nodes increment the value of the stored variable assigned to the sending node. The value of the variable shows the extent of associativity stability. If a node remains at the same place for a long time this value will be high reflecting reliable links. The variables stored in the nodes are reduced to zero if the node to which the variable is assigned becomes located outside the communication distance.

The purpose of ABR protocol is to provide routes with as long life as possible. When a node wants to request a route to another node, a Broadcast Query (BQ) message is sent to all the other nodes. If the other nodes detect the BQ packet they attach to it their own address, the QoS information and the stored degree of associativity stability relating to the adjacent nodes, then forward the packet. The destination node selects the most stable route from the data comprised in the received packets. If there are several routes with equal stability then the route containing the minimum number of hops is chosen by the destination node. When this is done the destination sends back a REPLY message to the source over the selected route. Every node having received the REPLY message makes the entry belonging to the selected route active. Since always only one route is selected, the possibility of repeating a packet is excluded.

Should a source node be dislocated, an RN[1] message is sent by it to the nodes on the route requesting them to delete their route entries, then a new BQ-REPLY cycle is initiated by the source. If the destination is dislocated then the nearest node on the route sends an LQ(H) (Localized Query) message towards the destination where H indicates the number of hops. Upon receipt of this message the destination sends back a reply over the best route, otherwise a time-out occurs at the sending node and the right to attempt is transferred to the source side neighbour of the sending node. The transfer of the right to attempt can be initiated by sending an RN[0] message. If this method fails to bring success up to the half of the route then a new BQ process is launched by the source.

If a route is no more needed it can be deleted by sending an RD (Route Delete) packet.

Signal Stability Routing (SSR) protocol

The Signal Stability Routing (SSR) protocol includes two interworking protocols: the Dynamic Routing Protocol (DRP) and the Static Routing Protocol (SRP). The DRP maintains the Signal Stability Tables (SST) and the Routing Tables (RT). The signal strength of the given node's neighbours is stored in the Signal Stability Table. The table is updated periodically based on the signs of life sent by the adjacent nodes. The DRP classifies the links established with adjacent nodes as strong or weak connections. All transmissions are implemented via the DRP. The DRP, after having updated the table entries, transfers the packets to the SRP. When the given node is the destination the SRP transfers the packet to the stack, otherwise forwards it to the node recorded in the Routing Table. If there is no such entry in the Routing Table, a route search is launched.

Only not processed route search packets sent via a strong channel are forwarded by the nodes. The destination node accepts only the first route search packet because presumably this was sent through the strongest route. Following the reception of the first packet the destination sends a reply message via the reverse route. The nodes receiving the reply update their Routing Table.

If the source does not receive an answer within a given time, then the value of PREF bit in the header of the search packet is changed indicating that weak channels can also be accepted and sends again the packet.

Should the route be broken somewhere the intermediate nodes inform the source about the failure and, as a result, a new route search is initiated by the source. If the link is broken at the source then the source notifies the other nodes of the fault.

Hybrid protocols

These protocols cannot be ranked unambiguously in the above-mentioned two classes. The hybrid protocols try to combine the advantages of both types, therefore they usually have several levels and apply "sub-protocols" of different types at the various levels.

Core Extraction Distributed Ad hoc Routing Algorithm (CEDAR)

The ad hoc network is a dynamic network created between a group of mobile units, where the channel is a shared and "expensive" resource. The applications, however, need links of suitable quality. Therefore the developers set the following aims when creating CEDAR:

- Use a distributed route calculation algorithm.
- The least possible nodes should take part in the calculation and maintenance of routes.
- Each node should maintain only those routes that lead to its destination nodes.
- Old, unused routes shall be identified and eliminated.
- If the topology is stabilised the network has to converge towards the optimum.
- A reserve (stand-by) route shall be provided, if possible, for cases when the primary route is damaged and is being recalculated.
- The algorithm shall be robust, even if it involves the deterioration of optimum conditions. (Figure 1.)

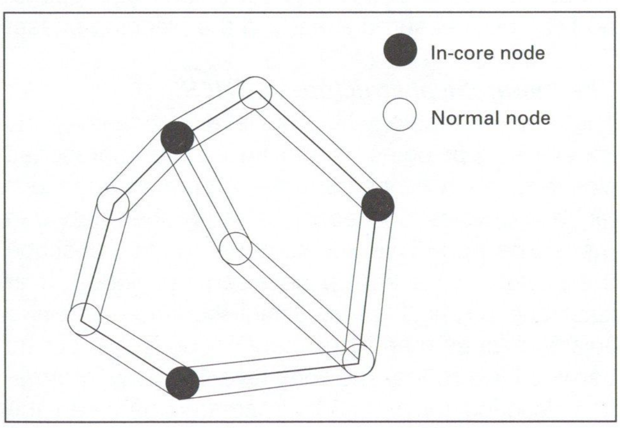

Figure 1 Topology of CEDAR network

The operation of the protocol is as follows: as the first step the core of the network is created using a distributed selection algorithm. The core of the network is a subset of nodes with which the network can be covered in a way that the distance between any unit and some in-core unit is not longer than two hops (Figure 1.). The out-of-core nodes do not take part in the route calculation, instead request a route from the nearest node when wish to communicate with another node. This way the least possible units participate in the route calculation, and updating messages have to be forwarded only between a few units. Only locally available data are used by the in-core nodes for route calculation. The updating of these data is also unique, implemented with so-called increasing and decreasing waves. With the help of these waves the data of stable links are spread over the whole network whereas the data of unstable links are forwarded only within a limited range. Consequently the probability that even a far node calculates the route on the basis of still existing links is high.

The in-core nodes communicate with each other by means of a special multiple-sending message forwarding, the so-called core-broadcast. If this method is used, a node does not forward the packet to all neighbours, only to certain nodes – in close co-operation with the Medium Access Control (MAC) layer. Therefore it is assumed that the individual identifier of all received or heard core-broadcast messages is stored temporarily by every node. This way the majority of packet retransmissions can be eliminated.

In the network maximum two out-of-core units can be located between two in-core nodes. These "adjacent" in-core nodes maintain a virtual connection made up of several real links, thus the superfluous packets can be filtered out at the intermediate units. It is essential to note that a node may request a route only from one in-core node, but can communicate with several in-core nodes if included in the virtual connection.

The protocol spreads the status information of links using so-called increasing and decreasing waves. In each link one node is responsible for bandwidth monitoring and providing information, and in case of a two-way link this is true for both nodes. If the bandwidth is changed, the controllers (the nearest in-core nodes) are informed one after the other, requesting them to launch increasing or decreasing waves. The controllers spread these waves within a part of the core, using core-broadcast. Each in-core node has two message queues: one contains the increasing wave messages, this is the increasing queue, while the other – the decreasing queue – accommodates the decreasing wave messages.

Elements of the increasing queue are forwarded periodically by the node, whereas a message placed in the decreasing queue is sent out immediately, therefore the decreasing waves spread much more quickly than the increasing ones. The decreasing waves suppress (blank) the increasing waves, this way a dynamically changing connection will be known only in the local environment, whereas the data of a stable link may spread far away as well. On the route the waves may be transformed into each other. When the Time To Live (TTL) level of a message becomes zero, further nodes receive a decreasing wave where the bandwidth of the link is 0, and the TTL level is infinite (thus the link entry is deleted), but this wave is not transmitted either through the whole network, because if a node does not have an entry for the link included in the message, throws away the decreasing wave message instead of forwarding. Consequently all "far" nodes delete out-of-date data relating to the given link.

With a view to avoid a large "overhead" waves are generated by the nodes only in the case where the change exceeds the threshold level. Logarithmic comparison seems to be reasonable in this case so as to be able to compare the change with the actually used bandwidth of the given link.

When requesting route calculation the source node must specify a Quality of Service parameter as well, concretely the required bandwidth. After that the controller of the source tries to find a route of adequate bandwidth to the destination. First the controller of the destination is searched for by making use of core-broadcast. The newly established core-route between the controller of the source and that of the destination can be already used to seek for a route meeting the given criteria, since along this route the in-core nodes know all the links and the bandwidth thereof. By making use of local information and topology the controller of the source attempts to find a route between the source node and a node of such a controller located the farthest on the core-route which ensures the required bandwidth. After that the remote controller seeks for a suitable route towards the destination. Finally the appropriate route is set up, and from that time on the packets are transmitted on the found new route instead of the core one. Therefore the core-route is used by the in-core nodes only until the appropriate route is found.

CEDAR is a robust and adaptive protocol, reacting quickly and efficiently to the changes of the network. It is most likely that a stable route with adequate bandwidth is set up, and no significant additional traffic is generated even in the case of very dynamic networks either (waves).

Zone-based Hierarchical Link State (ZHLS) routing protocol

The ZHLS is an ad hoc routing protocol based on the Global Positioning System (GPS). When using this protocol the network is divided into distinct (non-overlay) zones. Each node knows its own GPS co-ordinates (consequently the identifier of its own zone), the topology within its own zone and the links between the zones. The routing is performed on two levels: within the zone and between zones. No special nodes (for example clusterheads) are required by the protocol in spite of its hierarchical nature, consequently no bottleneck occurs in the network and no problem is caused by the dropping out of nodes either. Only the zone identifier and node identifier of the destination node are required for the routing, therefore the route can be modified in an adaptive way between the source and the destination depending on the changes of network topology. Simulation results published in connection with the protocol show that fewer messages have to be exchanged in the course of route search and the updating of topology information if ZHLS is used than in case of the application of on demand routing algorithms (for example: LSR – Link State Routing protocol) based on flooding. The distinct (non-overlay) zones make possible the reuse of frequencies.

Zone map

In ZHLS the network is divided into zones. Each node identifies its own geographic position with the help of some positioning system (for example: GPS), then – in the knowledge of the geographic position – the zone where the node can be found is also identified.

The nodes have a zone map for determining the zone identifier; the zone map is created by the designers of the system during planning. The size of zones depends on the mobility of nodes, on the density of the network, on the strength of communication and on the propagation characteristic. The division can be based on geographic and radio wave propagation considerations. The geographic division is simpler, whereas the division in terms of radio aspects is more complicated due to the frequency reuse demands. The radio based division can be applied in cases where the conditions of propagation can be measured already in the planning phase.

The hierarchical structure of ZHLS

The network topology has two levels: node-level and zone-level topologies. If two nodes are approached closer to each other than the communication distance, a physical connection is established between them. The node level topology information describe the physical links set up between the nodes. If at least one physical link is established between two adjacent zones then virtual connection is also set up between the zones. The zone level topology information describe the virtual links created between the zones.

Two kinds of LSP (Link State Packet) messages are used by the nodes to communicate topology information: node LSP and zone LSP. The node LSP is transmitted within a given zone, whereas the zone LSP is forwarded over the whole network.

Structure of routing tables

Within zone
Each node sends out a link request packet. Those adjacent nodes that receive such a packet give a reply by sending back a message with a format <zone identifier, node identifier>. When all the answers have arrived at the initial node, the node generates a node LSP message. The node LSP packet includes the identifiers of adjacent nodes within the zone as well as the zone identifiers of neighbours outside the zone. The node LSP message is transmitted within the zone. The nodes create their routing tables on the basis of node LSP packets, by making use of some algorithm searching for the shortest route.

As a result of movements physical links can be cancelled and new ones may be created, therefore the above procedure must be carried out periodically. It might happen that a node is roaming into another zone, therefore timers are assigned to the table entries and those where a time-out occurs are deleted.

Between zones
Those nodes that are physically linked with node(s) located in some adjacent zone are called gateway nodes.

Since each node LSP contains the zone identifiers of the connected zones, every node will know which adjacent zones are in virtual connection with its own zone. Therefore nodes located in the same zone will create and forward over the network the same zone LSP. The zone LSPs will be forwarded towards the other zones by the gateway nodes. Finally the zone LSPs will be known in the whole network, thus the current zone level topology will be available for the nodes. The zone level routing tables are created in this case as well with the help of some algorithm searching for the shortest route. The above procedure also is executed periodically.

The gateway nodes forward only those LSPs that have not been forwarded so far, so the network traffic generated by LSP is reduced. No timers are applied for zone LSPs; the routing tables are changed only when a virtual link is broken or a new connection is established.

Positioning and routing
Let's assume that node "a" wants to send data to node "z". In the first step "a" examines the routing table belonging to its own zone. Should "z" be included in the table, its data are immediately forwarded by "a". Otherwise "z" is found in another zone, so "a" issues a positioning request by sending out the messages <"a", the own zone identifier of "a", "z", X>, where X indicates one of the other zones. The nodes receiving the messages forward the packets, according to their own zone level tables, towards the X zones. If some gateway node finds "z" in its own node level table then a reply message is generated with the following format: <"z", zone identifier of "z", "a", zone identifier of "a" >. If "a" receives this packet then the zone identifier and node identifier of "z" will be available for it, consequently "a" will be able to send the data. All nodes that are not in the zone of "z" will use their zone level table for the transmission of packets, whereas the node located in the zone of "z" applies the node level table when wishing to send a packet.

Figure 2 illustrates the distribution of route lengths created by ZHLS protocol, comparing it with the distribution of lengths for routes set up by a purely reactive protocol (LSR).

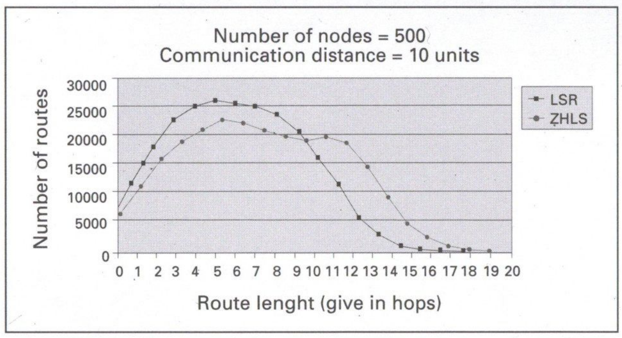

Figure 2 Distribution of route lenghts for all nodepairs included in the network

Summary

We would like to emphasise that, in addition to the protocols described here, several other solutions have been developed, but none of them is capable of providing an efficient solution in itself to all the problems arising in an ad hoc environment.

The Quality of Service and the security are not fully elaborated areas yet for these protocols. Although some algorithms implement partially QoS, but the level offered by Internet protocols has not been reached so far.

On the other hand it is worth following with attention the area of mobile ad hoc networks, because dynamic development can be experienced, and major progress can be expected in the near future.

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Simplified design of electromagnetic coils suitable for programming

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engineer

This paper sums up a topic of the book written by the author entitled "Solenoids and electromagnetic control elements". In this book a "coil constant" is introduced through which the design of electromagnets becomes considerably easier and also the programming of appropriate calculations are made easier.

Design principles

The literature dealing with coil design cites generally a technique which is based on the calculation of the average diameter of the bobbin and then the determination of the number of turns, the length of wire and the resistance of the coil follows. The real problem here is the determination of the diameter of wire. In this case the calculation uses a root type equation which is not really suitable for computer assisted programming. The process of calculation is widely known in the literature so we do not present it here. However, in the following sections our new method will be illustrated via practical examples.

First of all we want to note that the diameter and other parameters of the copper wire can be specified in both of the usual unit system. In this paper we will use the metric system based on SI as introduced in 1980. However, we have to take into account the Anglo-American system being still in use since in the tables we specify also AWG (American Wire Gauge) parameters. Originally the "Brown and Sharp" system of AWG specified the thickness of iron plates. Here the Gage number identifies the plate thickness and recently also the wire gauge. For example the diameter of an AWG No. 1. wire is 0.2893 inch, i.e. 7.3 mm. The diameter of a No. 30 wire is 0.001 inch, i.e. 0.0254 mm. It should be noted that a higher AWG number means smaller diameter.

Should any unit be used, our technique based on coil demand can be well implemented since it takes into account the number of turns per unit area. This paper focuses on the geometry of coils and makes the assumption that electrical and thermal parameters (supply voltage, current, temperature) are known, including coil resistance.

The method described here can be used for design calculations and also for the verification of calculations and their programming.

Initial calculation

The first step is the determination of the cross-section of window (Aw) and the coiling volume (Wv) for circular bobbin. Design parameters are as follows:

Permitted outer diameter of the coil: ODw = 30 mm

Length of coil: L = 27 mm

Diameter of coil body: IDw = 15 mm

Using these data the cross-section of windows can be calculated:

$Aw = [(ODw - IDw)/2] \times L$ [mm^2]

The coil volume is the product of the efficient sides of the bobbin and the length of coil:

$Wv = (ODw^2 - IDw^2) \times 0.785\ L$ [mm^3] and $0.785 = \pi/4$.

In case the outer diameter of the coil is not specified, this parameter can be calculated as follows:

$ODw = (2\ Aw/L) + IDw$ [mm].

With the specified geometrical parameters:

Window cross-section:

$$Aw = [(30 - 15)/2] \times 27 = 202.5\ \text{mm}^2.$$

Coil volume:

$$Wv = (302 - 152) \times 0.785 \times 27 = 14{,}306\ \text{mm}^2.$$

Applications

Example 1: Determination of number of turns and resistance

Let the diameter of the naked wire be 0.1 mm (see metric wires, table 1, naked wire diameter 0.1 mm and single insulated wire diameter 0.113 mm). If the wire is spooled one turn above the other, then theoretically 7831 turns can be placed in an area of one cm^2. However, taking into account the filling coefficient of 84%, the number of turns will be in practice 0.84 x 7831 = 6578 turns as it can be seen in Table 1, column turns/cm^2.

The full length of wire in the coil is the product of he number of turns and the unit side, i.e. 6578 x 10 mm = 65 780 mm = 65.78 m.

According to the table, resistance of the wire is 219 ohm/100 m, hence the full resistance of the coil is 0.6578 x 219 = 144 ohm.

Remarks:

1. In theory the number of turns was determined by the fact that the square of the diameter of the insulated wire specifies the area of a square touching the wire: ($0.1132\ mm^2 = 0.012769\ mm^2$) so in an area of 1 cm^2 = 100 mm^2 round 100/0.012769 = 7831 turns can be placed.

2. The resistance of the wire can be calculated from the reference value for 100 m (column 3 in table) as follows:

R = $(219/100)/d^2$ = 2.19/0.01 = 219 ohm/100 m, i.e. for 0.6578 m R = 144 ohm.

In all other cases table 1-3 specifies the resistance of 100 m wire for diameters 0.04 ... 1.5 mm.

Example 2: Resistance to be specified for given number of turns and bobbin

Let us suppose that the number of turns in the coil is 3000 and this should be placed in a volume of 14 306 mm^3 according to previous calculations.

The greatest wire diameter has to be found which fits the given window cross section.

Here again we use the table of metric wires where the unit of area is cm^2 and the unit of volume is cm^3. Accordingly, the window required for 3000 turns of wire (Aw) is 202.5 mm^2, i.e. 2.025 cm^2 and the number of turns per window area is 3000/0.025 = 1.481 turn/cm^2.

Generally speaking:

Number of turns /cm^2 taken from table = given number of turns/given window cross section (in cm^2).

Now we look for the appropriate wire diameter in the table. This is found at 0.23 mm. The associated number of turns per unit area for single isolated wire is 1428.

Hence the new window cross-section:

Aw = 3000/1428 cm^2 = 2.1 cm^2

The new window height:

Hw = Aw/L = 2.1/2.7 cm = 0.777 cm = 7.77 mm.

The outer diameter of the wiring:

ODw = 2x7.77 + 15 mm = 30.84 mm, i.e. we use somewhat greater outer side diameter than 30 mm.

According to the modified bobbin diameter the volume of the coil changes as well:

Wv = ($ODw^2 - IDw^2$) x 0.785 x L = (30.542 – 152) x 0.785 x 27 = 14 999 mm^2 = 15 cm^2.

Since the resistance per volume unit of the wire is 5.92 ohm/cm^3 in the table, resistance of the coil will be:

R = 5.92 x 15 = 88.8 ohm.

Example 3: Wiring parameters to be specified for given resistance value

The bobbin to be used is the same as in the previous example. The resistance required should be 2500 ohm.

Parameters of the bobbin:

Window area:

Aw = 202.8 mm^2, i.e. 2.025 cm^2

Volume of wiring:

Wv = 14 380 mm^3, cca. 15 cm^3.

Since in this volume we want to place a coil with 2500 ohm resistance, the resulting unit resistance is 2500/15 = 166.6 ohm.

In the table we find that the nearest value is associated to 0.09 mm wire diameter which is 220 ohm/cm^3, so the volume of the coil will be smaller:

Wv = 2500 / 220 = 11.36 cm^3 < 14.94 cm^3. As we can see, this is less than 15 cm^3, i.e. the window area is not completely filled.

For the calculation of the number of turns first we shall calculate the bobbin which can be taken from the volume parameters:.

Wiring volume:

Wv = (ODw2 – IDw2) x k

hence

ODw = $(Wv/k\ IDw2)^{1/2}$ where k is a newly introduced constant. Its value:

0.785L = 0.785 x 27 mm = 21.2 mm.

In our case the outer diameter of the wiring is:

ODw = $(11\ 360/21.2 + 15^2)^{1/2}$ = 27.58 mm.

The next step is the specification of wiring area, Aw: Aw = (27.6 – 15)/2 x 27 = 170 mm^2 = 1.7 cm^2.

According to the table a wire of 0.09 mm diameter provides 8156 turns per cm^2. In our case:

N = 8156 x 1.7 = 13 862 turns.

Finally the complete parameter set:

Wire diameter: 0.09 mm, single isolated wire

Number of turns:	13 862
Resistance:	2500 ohm
Outer diameter of coil:	27.6 mm
Wiring area:	1.7 cm^2
Volume of wiring:	11.36 cm^3

Calculation with the use of a wiring constant

This method of calculation is used when both the number of turns and the coil resistance is given. The method provides the appropriate wire diameter. Using previous methods this calculation would take a longer time since we should try all possible wire diameter – from the smallest to the greatest. However, with the use of a coil constant to be specified below, the calculation is much easier. The value of the coil constant is the ratio of the number of turns to resistance, depending also on the geometrical dimensions and wire size. The last column in the metric wire table indicates values for 1 cm^2 wiring area and 1 cm^3 volume. As seen in the table, this value depends on wire diameter.

Use of the table: the required number of turns is converted into 1 cm^2, the resistance into 1 cm^3, their ratio is taken and this value is looked for in the last column (K). Generally this value is between two adjacent values, so we get two wire diameter values. Any of these two values can be used at the discretion of the designer.

Example 4.
In example 3 w could place 13 862 turns of wire diameter 0.09 mm in a corrected wiring area of 1.7 cm^2. The resistance of the wiring was 2500 ohm. Now we calculate the parameters of this wiring with the use of the coil constant.

Initial parameters:	
Wiring area:	2 cm^2
Wiring volume:	15 cm^3
Number of turns:	13 862
Resistance:	2500 ohm

The next step is the calculation of the coil constant:

Number of turns:	13 862/2 = 6 931 per cm^2
Resistance:	2500/15 = 166 ohm per cm^3
Coil constant:	6931/166 = 41.7

When compared to the metric wire table we find that the value of 41.7 falls between 37 and 46, associated to wire diameters 0.09 mm and 1.0 mm respectively. Now we can follow on steps described in Example 1.

Without calculations we can turn also to a graphical procedure using the attached diagram.

Use of the wiring diagram

The use of the diagram speeds up the process of identification of a wire diameter for a given number of turns and volume. In case of programming, this also facilitates the testing procedure.

The given volume can be selected between 0.5 and 10.0 cm^3. Wire diameters are indicated on the inner side of the left vertical axis in mm. The outer side indicates the resistance for the volume of 1 cm^3.

In most cases the required wire diameter falls between two values since the diagram is based on resistance values. In such cases it is practical to use the smaller diameter.

Example: Use of the diagram
If we want to coil 1000 turns then in case of 1 cm^3 volume the wire of 0.28 mm diameter shall be used which gives a resistance of 2.72 ohm.

In case of 3 cm^3 volume the wire diameter falls between 0.5 and 0.55. Choosing the smaller 0.5 mm wire the resulting resistance will be 0.29 ohm/cm^3, thus the resistance of the coil is 3 x 0.29 = 0.87 ohm.

Choosing the volume of 0.5 cm^3 will produce 0.19, i.e. 0.2 mm wire diameter. For the 0.19 mm wire diameter 12.0 ohm/cm^3, hence the resistance will be 12/2 = 6 ohm.

The use of the diagram provides an overview on the problem and it is useful in the initial calculations. The previously described methods allow for more accurate calculations.

The table can be used inversely as well. For a given resistance value we can find numbers of turns on the horizontal axis for different volumes.

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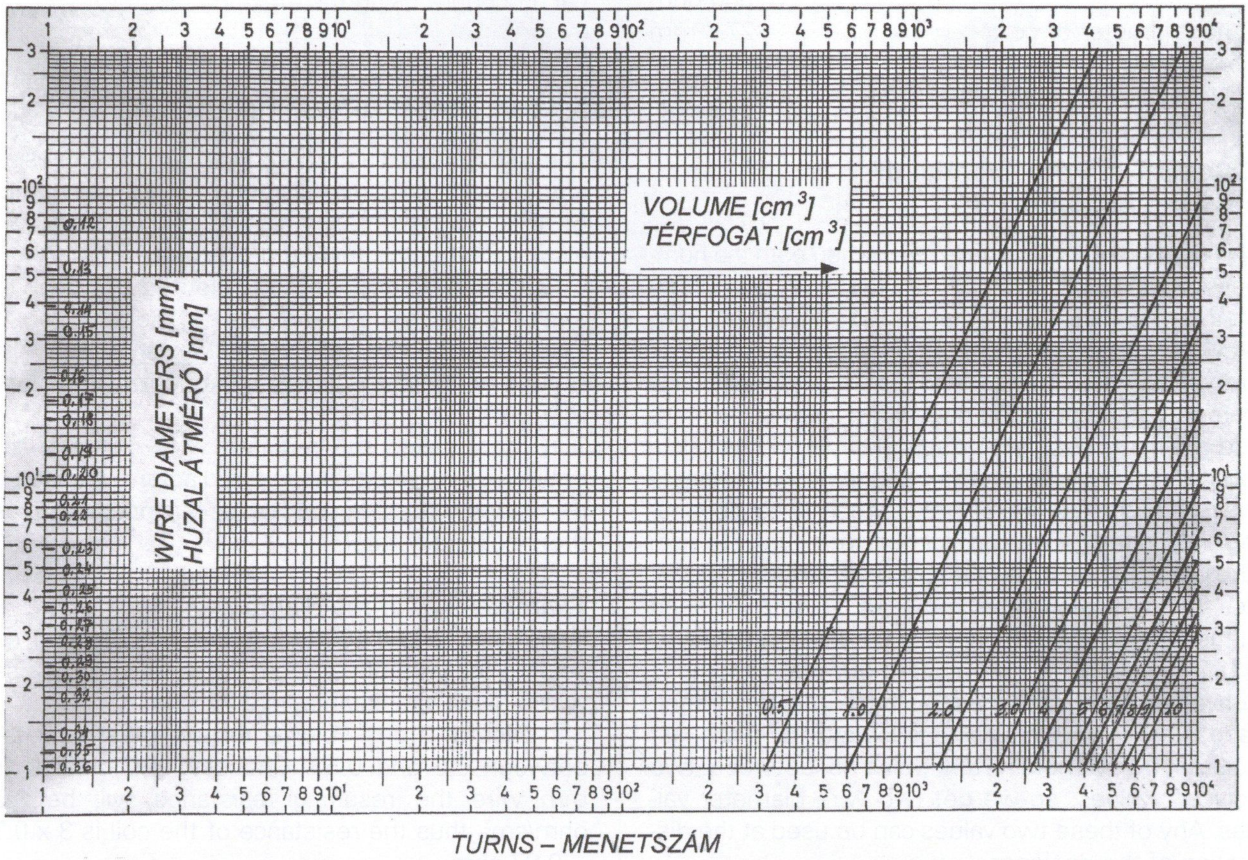
VOLUME [cm^3]
TÉRFOGAT [cm^3]
WIRE DIAMETERS [mm]
HUZAL ÁTMÉRŐ [mm]
TURNS – MENETSZÁM

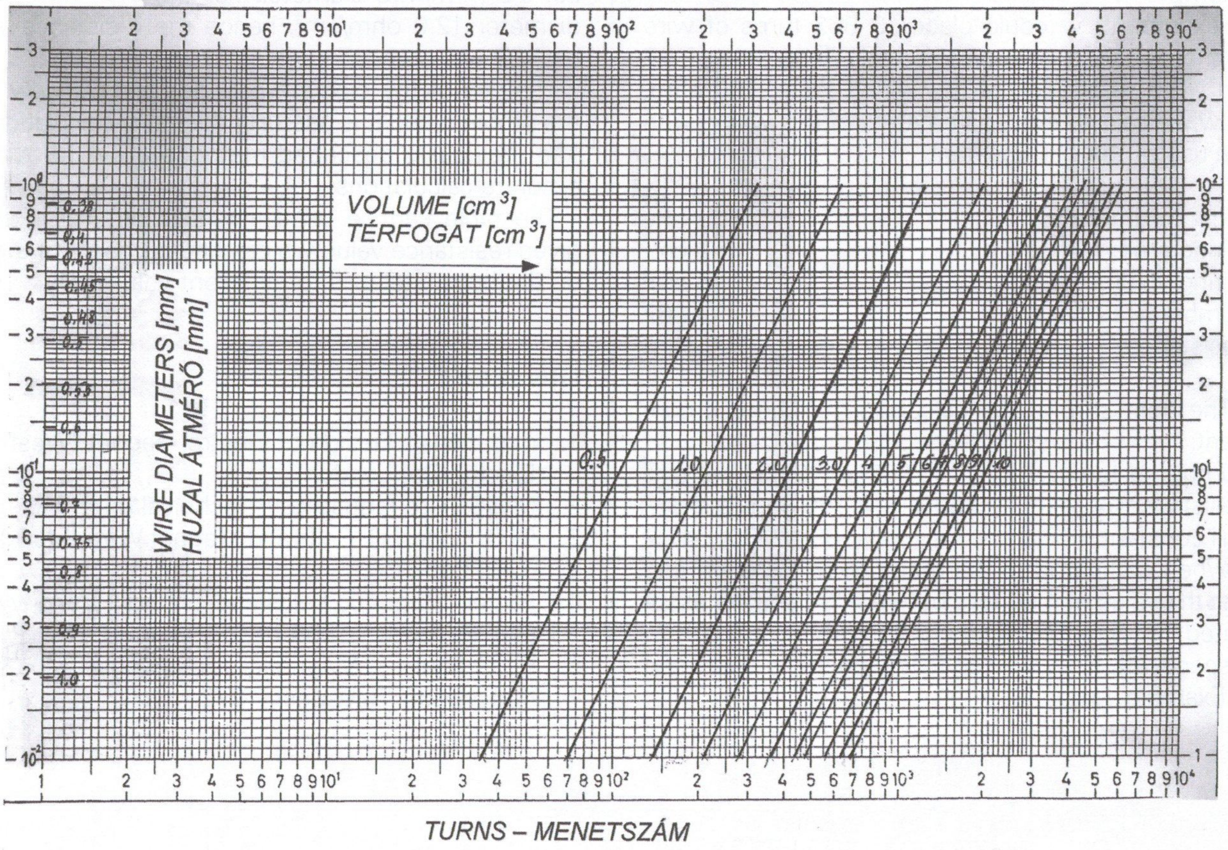
VOLUME [cm^3]
TÉRFOGAT [cm^3]
WIRE DIAMETERS [mm]
HUZAL ÁTMÉRŐ [mm]
TURNS – MENETSZÁM

CHART 1. - METRIC COIL WINDING

AWG	Bare copper dia. [mm]	Bare copper ohm / 100 m	Bare copper cross-section	Single build copper diameter [mm] min.	Single build copper diameter [mm] nom.	Single build copper diameter [mm] max.	Single build copper kg / 100 m	Single build copper turns / cm²	Single build copper ohm / cm³	Double build copper diameter [mm] min.	Double build copper diameter [mm] nom.	Double build copper diameter [mm] max.	Double build copper kg / 100 m	Double build copper turns / cm²	Double build copper ohm / cm³	K cm / ohm
45	0,04	1 372,025	0,00126	0,044	0,047	0,051	0,001176	36 215	4968,8	0,047	0,051	0,055	0,001214	30 757	4220	7,288
44	0,05	878,096	0,00196	0,053	0,057	0,061	0,001817	24 622	2162	0,050	0,058	0,066	0,001828	23 781	2088	11,39
42	0,06	609,789	0,00283	0,064	0,069	0,074	0,002625	17 433	1063	0,068	0,073	0,078	0,002681	15 575	949,7	16,4
41	0,07	448,008	0,00385	0,075	0,080	0,085	0,003565	12 968	580,98	0,081	0,086	0,091	0,003662	11 222	502,8	22,32
40	0,08	343,006	0,00503	0,086	0,091	0,096	0,004648	10 022	343,76	0,093	0,099	0,105	0,004797	8 468	290,5	29,15
39	0,09	271,017	0,00636	0,095	0,101	0,107	0,005855	8 136	220,5	0,102	0,109	0,116	0,006020	6 985	189,3	36,9
38	0,10	219,524	0,00785	0,107	0,113	0,119	0,007246	6 656	146,12	0,115	0,123	0,129	0,007478	5 518	121,1	45,55
37	0,11	181,425	0,00950	0,117	0,123	0,129	0,008736	5 618	101,92	0,127	0,134	0,141	0,009014	4 733	85,87	55,12
	0,12	152,447	0,01131	0,127	0,133	0,139	0,010366	4 805	73,251	0,137	0,145	0,153	0,010693	4 042	61,62	65,6
36	0,13	129,896	0,01327	0,139	0,146	0,153	0,012220	3 987	51,789	0,149	0,158	0,167	0,012578	3 404	44,22	76,98
35	0,14	112,002	0,01539	0,149	0,156	0,163	0,014135	3 492	39,111	0,159	0,168	0,177	0,014516	3 011	33,72	89,28
	0,15	97,566	0,01767	0,159	0,163	0,173	0,016092	3 192	31,143	0,169	0,178	0,187	0,016594	2 682	26,17	102,5
34	0,16	85,752	0,02011	0,171	0,179	0,185	0,018487	2 652	22,741	0,181	0,189	0,199	0,018848	2 379	20,4	116,6
	0,17	75,960	0,02270	0,184	0,191	0,198	0,020900	2 467	18,739	0,194	0,203	0,212	0,021364	2 188	16,62	131,6
33	0,18	67,754	0,02545	0,194	0,201	0,208	0,023382	2 391	16,2	0,204	0,214	0,224	0,023912	1 965	13,31	147,6
	0,19	60,810	0,02835	0,204	0,213	0,218	0,026087	1 983	12,059	0,214	0,224	0,234	0,026559	1 793	10,9	164,4
32	0,20	54,881	0,03142	0,214	0,221	0,228	0,028765	1 842	10,109	0,226	0,236	0,246	0,029438	1 615	8,863	182,2
	0,21	49,779	0,03464	0,223	0,231	0,239	0,031666	1 686	8,3927	0,235	0,247	0,257	0,032417	1 428	7,108	200,9
	0,22	45,356	0,03801	0,233	0,237	0,249	0,034519	1 602	7,2661	0,245	0,257	0,269	0,035488	1 362	6,178	220,5
31	0,23	41,498	0,04155	0,243	0,251	0,259	0,037886	1 428	5,9259	0,255	0,267	0,279	0,038700	1 203	4,992	241
	0,24	38,112	0,04524	0,253	0,261	0,269	0,041205	1 321	5,0346	0,265	0,277	0,289	0,042050	1 172	4,467	262,4
	0,25	35,124	0,04909	0,264	0,274	0,284	0,044824	1 198	4,2078	0,277	0,289	0,303	0,045653	1 077	3,783	284,7
30	0,26	32,474	0,05309	0,274	0,284	0,294	0,048428	1 115	3,6208	0,287	0,300	0,313	0,049346	1 000	3,247	307,9
	0,27	30,113	0,05726	0,284	0,294	0,304	0,052172	1 041	3,1348	0,297	0,310	0,323	0,053121	936	2,819	332,1
	0,28	28,001	0,06158	0,294	0,304	0,314	0,056055	973	2,7245	0,307	0,320	0,333	0,057035	878	2,458	357,1
29	0,29	26,103	0,06605	0,304	0,314	0,324	0,060077	912	2,3806	0,317	0,333	0,343	0,061284	856	2,234	383,1
	0,30	24,392	0,07069	0,317	0,327	0,337	0,064431	888	2,166	0,330	0,343	0,356	0,065483	807	1,968	410
28	0,32	21,438	0,08042	0,337	0,347	0,357	0,073185	789	1,6915	0,352	0,365	0,378	0,074443	713	1,529	466,5
	0,34	18,990	0,09079	0,357	0,369	0,381	0,082642	697	1,3236	0,373	0,387	0,401	0,083978	643	1,221	526,6
	0,35	17,920	0,09621	0,368	0,380	0,392	0,087586	657	1,1774	0,384	0,398	0,412	0,088960	608	1,09	558
27	0,36	16,939	0,10179	0,378	0,390	0,402	0,092596	624	1,057	0,394	0,408	0,422	0,094007	570	0,965	590,4
	0,38	15,202	0,11341	0,398	0,411	0,424	0,103117	565	0,8589	0,416	0,431	0,446	0,104770	511	0,777	657,8
26	0,40	13,720	0,12566	0,421	0,432	0,447	0,114203	509	0,6984	0,436	0,451	0,466	0,115850	467	0,641	728,8
	0,42	12,445	0,13854	0,441	0,455	0,467	0,126034	458	0,57	0,459	0,489	0,489	0,129185	397	0,494	803,6
25	0,45	10,841	0,15904	0,470	0,484	0,498	0,144348	405	0,439	0,488	0,504	0,520	0,146288	373	0,404	922,5
	0,48	9,528	0,18096	0,502	0,516	0,530	0,164209	356	0,3392	0,520	0,536	0,552	0,166274	330	0,314	1050
24	0,50	8,781	0,19635	0,522	0,536	0,550	0,178020	330	0,2898	0,540	0,556	0,572	0,180164	307	0,27	1139
23	0,55	7,257	0,23758	0,572	0,586	0,600	0,214989	276	0,2003	0,590	0,607	0,624	0,217448	257	0,187	1378
	0,60	6,098	0,28274	0,619	0,636	0,653	0,255444	234	0,1427	0,640	0,659	0,678	0,258369	218	0,133	1640
22	0,65	5,196	0,33183	0,669	0,686	0,703	0,299387	201	0,1044	0,690	0,709	0,728	0,302537	188	0,098	1925
21	0,70	4,480	0,38485	0,722	0,740	0,758	0,347397	173	0,0775	0,742	0,762	0,782	0,350641	163	0,073	2232
	0,75	3,903	0,44179	0,772	0,790	0,808	0,398354	152	0,0593	0,772	0,790	0,808	0,398354	152	0,059	2562
20	0,80	3,430	0,50265	0,821	0,841	0,862	0,452963	134	0,046	0,821	0,841	0,862	0,452963	134	0,046	2915
	0,85	3,038	0,56745	0,871	0,891	0,912	0,510904	119	0,0362	0,871	0,891	0,912	0,510904	119	0,036	3291
19	0,90	2,710	0,63617	0,921	0,942	0,964	0,572516	107	0,029	0,921	0,942	0,964	0,572516	107	0,029	3690
	0,95	2,432	0,70882	0,971	0,992	1,014	0,637441	97	0,0235	0,971	0,992	1,014	0,637441	97	0,023	4111
18	1,00	2,195	0,78540	1,022	1,045	1,069	0,706468	87	0,0191	1,045	1,057	1,069	0,708944	85	0,019	4555
	1,05	1,991	0,86590	1,072	1,095	1,119	0,778397	86	0,0171	1,095	1,119	1,144	0,783613	73	0,014	5022
	1,10	1,814	0,95033	1,122	1,145	1,169	0,853813	79	0,0143	1,146	1,170	1,194	0,859495	69	0,013	5512
17	1,15	1,660	1,03869	1,172	1,195	1,219	0,932716	72	0,0119	1,195	1,220	1,246	0,938643	64	0,011	6024
	1,20	1,524	1,13097	1,225	1,250	1,275	1,016331	66	0,01	1,248	1,274	1,300	1,022278	59	0,009	6560
	1,25	1,405	1,22718	1,275	1,301	1,325	1,102513	61	0,0085	1,298	1,309	1,350	1,104562	55	0,008	7118
16	1,30	1,299	1,32732	1,325	1,351	1,375	1,191936	56	0,0073	1,348	1,374	1,400	1,198089	50	0,007	7698
	1,35	1,205	1,43139	1,375	1,401	1,425	1,284847	52	0,0063	1,398	1,414	1,430	1,288439	48	0,006	8302
	1,40	1,120	1,53938	1,427	1,453	1,479	1,381815	49	0,0054	1,450	1,477	1,540	1,388718	44	0,005	8928
15	1,45	1,044	1,65130	1,477	1,503	1,529	1,481719	45	0,0047	1,477	1,514	1,554	1,484977	41	0,004	9578
	1,50	0,976	1,76715	1,527	1,548	1,570	1,583589	42	0,0041	1,550	1,577	1,604	1,592486	38	0,004	10249

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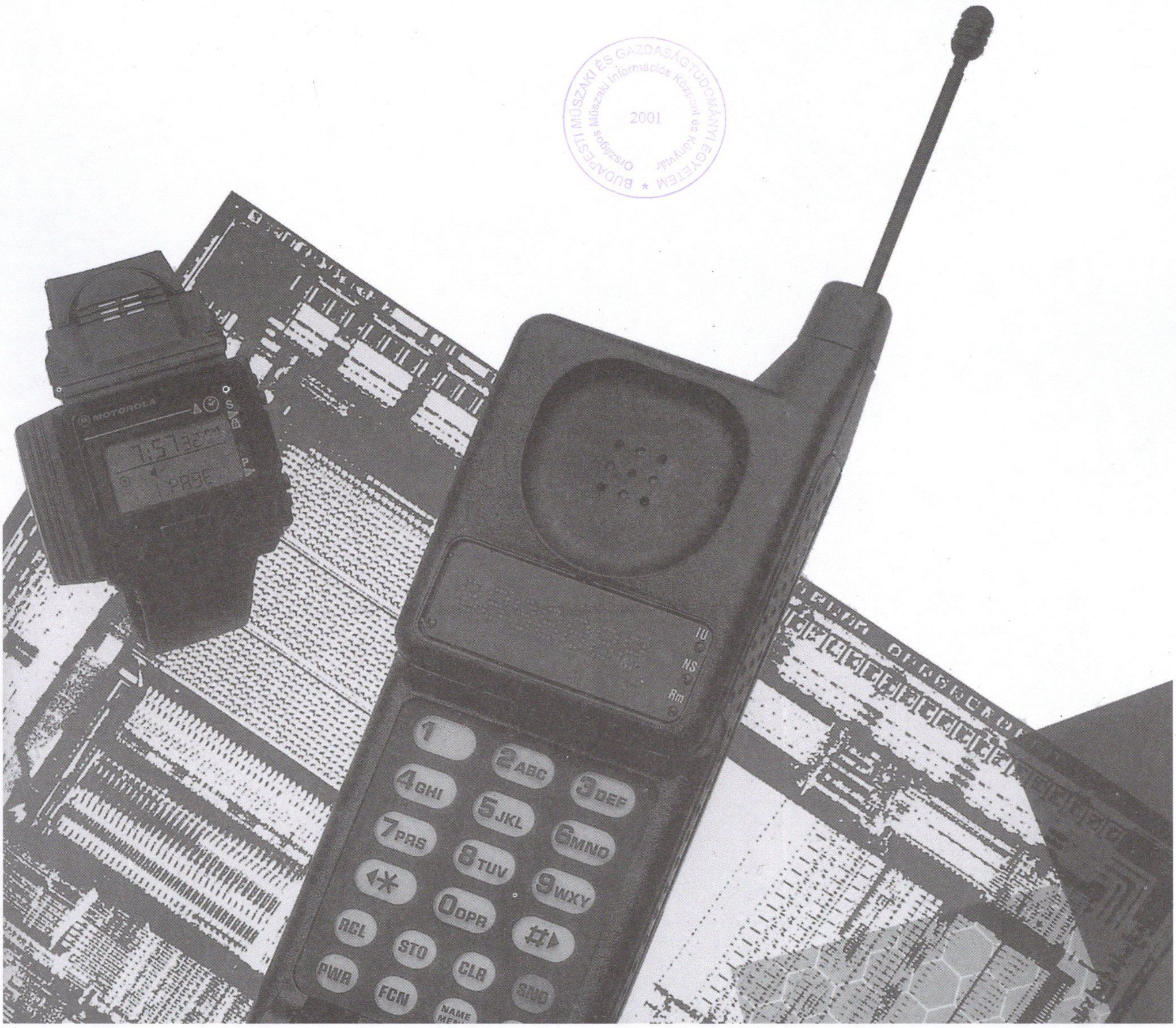
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