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Voice Biometrics over the Internet in the Framework of COST Action 275

Laurent Besacier,[1] Aladdin M. Ariyaeeinia,[2] John S. Mason,[3] Jean-François Bonastre,[4] Pedro Mayorga,[1] Corinne Fredouille,[4] Sylvain Meignier,[4] Johann Siau,[2] Nicholas W. D. Evans,[5] Roland Auckenthaler,[5] and Robert Stapert[6]

[1] *CLIPS/IMAG, 38041 Grenoble Cedex 9, France*
Emails: laurent.besacier@imag.fr; pedro.mayorga-ortiz@imag.fr

[2] *Department of Electronic, Communication and Electrical engineering, University of Hertfordshire, Hatfield, AL10 9AB, UK*
Emails: a.m.ariyaeeinia@herts.ac.uk; j.siau@herts.ac.uk

[3] *Department of Electrical and Electronic Engineering, University of Wales Swansea, Swansea SA2 8PP, UK*
Email: j.s.d.mason@swansea.ac.uk

[4] *LIA, University of Avignon, 84911 Avignon Cedex 9, France*
Emails: jean.francois.bonastre@lia.univ-avignon.fr; corinne.fredouille@lia.univ-avignon.fr; sylvain.meignier@lia.univ-avignon.fr

[5] *School of Engineering, University of Wales Swansea, Swansea SA2 8PP, UK*
Emails: n.w.d.evans@swan.ac.uk; eeaucken@swansea.ac.uk

[6] *Aculab, Milton Keynes, MK1 1PT, UK*
Email: robert.stapert@aculab.com

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The emerging field of biometric authentication over the Internet requires both robust person authentication and secure computer network protocols. This paper presents investigations of vocal biometric person authentication over the Internet, both at the protocol and authentication robustness levels. As part of this study, an appropriate client-server architecture for biometrics on the Internet is proposed and implemented. It is shown that the transmission of raw biometric data in this application is likely to result in unacceptably long delays in the process. On the other hand, by using data models (or features), the transmission time can be reduced to an acceptable level. The use of encryption/decryption for enhancing the data security in the proposed client-server link and its effects on the transmission time are also examined. Furthermore, the scope of the investigations includes an analysis of the effects of packet loss and speech coding on speaker verification performance. It is experimentally demonstrated that whilst the adverse effects of packet loss can be negligible, the encoding of speech, particularly at a low bit rate, can reduce the verification accuracy considerably. The paper details the experimental investigations conducted and presents an analysis of the results.

Keywords and phrases: voice biometrics, speaker verification, packet loss, compression, Internet.

1. INTRODUCTION

The ever-increasing use of the Internet-enabled devices is resulting in normal activities in day-to-day life, such as banking and shopping, being conducted without face-to-face or personal contacts. A natural consequence of this is the obsolescence of certain conventional means of identification. Examples of these are photo ID cards and passports. On the other hand, the conventional authentication means such as personal identification numbers and passwords, which are equally applicable to local and remote identity verification, can be easily compromised or forgotten. In view of the above, it appears that biometrics is the only means that can satisfy the requirements for remote identity verification in terms of both appropriateness and reliability. This is because firstly, biometric data can be easily captured, stored, processed, and described electronically. Secondly, it uses an intrinsic aspect of a human being for identity verification. Consequently, it is not so susceptible to fraud as passwords or personal identification numbers.

The deployment of biometrics on the Internet, however, is a multidisciplinary task. It involves person authentication techniques based on signal processing, statistical modelling, and mathematical fusion methods, as well as data communications, computer networks, communication protocols, and online data security.

The necessity for the latter discipline is due to the fact that an online robust biometric authentication strategy would be of little or no value if, for instance, hackers could break into the personal identification server to control the verification of their pretended identities, or could access personal identification data transmitted over the network.

The original aim of the Internet was to provide a means of sharing information, thus security was not of major concern. As the Internet has evolved, many security implications and bandwidth issues have arisen. There are many potential threats to any system that relies on the Internet as a communication medium. The potential benefits of biometric identity verification over the Internet have highlighted issues of security and network performance that need to be tackled more effectively [1].

In general, network performance varies widely with the geographical location of the clients, server type, and network resources. There is variation in the response time from session to session even if the connection is made to the same server. This is because in each session, data packets may travel through a different route [2]. There is a difference in the performance of the dial-up Internet service, integrated subscriber digital network (ISDN), asymmetric digital subscriber line (ADSL), cable modem, and leased line as they all have a different bandwidth and response time. This will undoubtedly affect the performance of biometric verification systems in terms of speed, reliability, and the quality of service.

Over IP networks, both speech and image-based biometrics are viable alternative approaches to verification. Focusing on speech biometrics, some predictions for the year 2005 show that 10% of voice traffic will be over IP. This means that speaker verification technology will have to face new problems. The most common architecture seems to be client-server-based where a distant speaker verification server is remotely accessed by the client for authentication. In this scenario, the speech signal is transmitted from the client terminal to a remote speaker verification server. Coding of the speech signal is then generally necessary to reduce transmission delays and to respect bandwidth constraints. Many problems can appear with this kind of architecture, particularly when the transmission is made via the Internet:

(i) firstly, transcoding (the process of coding and decoding) modifies the spectral characteristics of the speech signal, and thereby can adversely affect the speaker verification performance;
(ii) secondly, transmission errors can occur on the transmission line: thus, data packets can be lost (e.g., with UDP transport protocols which do not implement any error recovery);
(iii) thirdly, the time response of the system is increased by coding, transmission, and possible error recovery processes. This delay (termed "jitter" as used in the domain of computer networks) can be potentially very disturbing. For example, in some applications (e.g., man-machine dialogue), speaker verification is only one subsystem amongst a number of other subsystems. In such cases, the effective operation of the whole system depends heavily on the response time of the individual subsystems;
(iv) finally, speech packets (or other personal information) transmitted over IP could be intercepted and captured by impostors, and subsequently used, for instance, for fraudulent access authorisation.

To our knowledge, this paper is the first to present an overview of issues and problems in the above area. These include architecture and protocol considerations (Section 2), speaker verification robustness to speech coding and packet loss over IP networks (Section 3), and wireless mobile devices (Section 4). This work is currently conducted in the framework of COST Action 275 (http://www.fub.it/cost275/).

2. ARCHITECTURE AND PROTOCOL CONSIDERATIONS IN BIOMETRICS OVER THE INTERNET

This part details an analysis carried out to determine the right balance in the transmission method for the purpose of implementing applications involving biometric verification. These tests were conducted in different geographical locations within the UK. However, most of the local area network (LAN) tests were carried out in the premises of the University of Hertfordshire.

2.1. Biometrics applied

The raw biometric data can have different sizes depending on its type. For instance, voice or face biometric datasets are considerably larger than that of fingerprint. In any case, the data contains the identity of an individual and should be treated with utmost care. Therefore, it is necessary to have an appropriate architecture and method of transmission in order to provide a high level of protection against uncertainties.

2.1.1. Client-server architecture

An effective client-server structure for biometrics on the Internet has recently been proposed by some authors of this paper [3]. This realisation (Figure 1) consists of 3 distinct components, each performing a specific task. The client part consists of users (clients) requesting appropriate services from the server. A main role of the server is to respond to these requests. However, from time to time, it itself becomes a client to the central database and requests services from it.

The modular nature of the proposed structure is also necessary for performing software updating effectively. For example, the client module dynamically obtains information relevant to its process, and the updates to its software are provided by the server. As a result, it is ensured that the client software will always be up-to-date, and modifications or improvements can be gradually rolled in.

In order to maintain data integrity, the transmission channel needs to be secured and encrypted. This will ensure

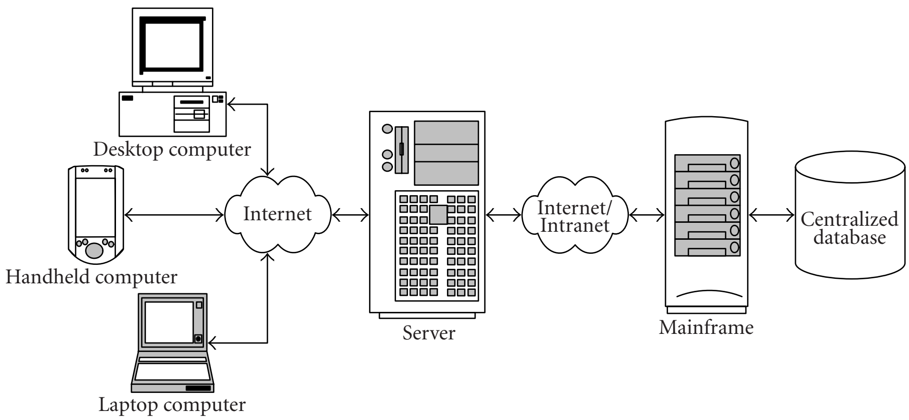


FIGURE 1: Client-server architecture.

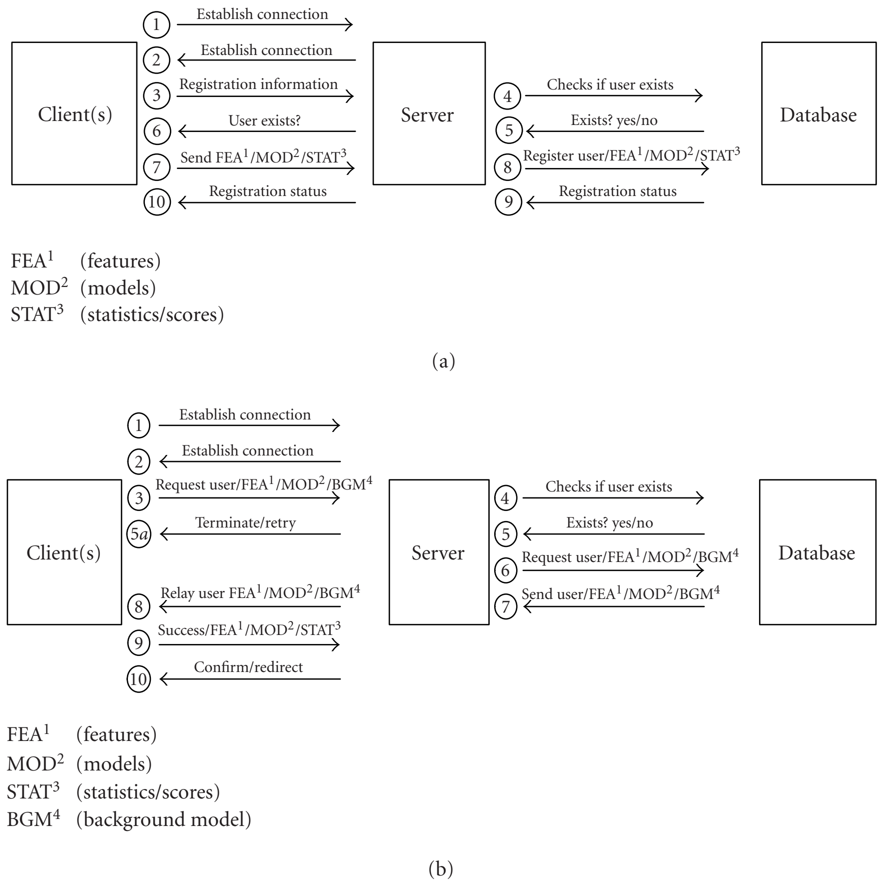


FIGURE 2: Proposed client-server architecture. (a) Enrolment process. (b) Verification process.

that data sent from the client to the server and vice versa will be of no use to others even if they breach the system.

Figure 2 illustrates the operation of the proposed system in terms of its enrollment and verification processes. It should be noted that although the system is ideally suited to speaker verification, it could also be adapted to suit other types of biometrics. The operation can be described as follows.

The database acts as the central storage area for all biometric data and also as a server to the main server. Each server has its unique identifier that allows its connection to the database. All communications between the server and database are secured and encrypted. Distributed/different servers from different geographical locations can therefore connect to the central database through a fast network link.

During the enrollment process, the client initially establishes a connection with the server. This is known as the handshaking process in which the client and server establish the identity of both machines for that particular session. The encryption key (Section 2.1.3) is also exchanged at this time. The registration information is then sent to the server. Once a confirmation is obtained from the server that the user does not exist in the system, the client is prompted to send the biometric features, models, and statistics over to the server to be enrolled. These are encrypted before transmission. The server then forwards this information to the database and thus enrolling the user to the system.

When a user returns to verify his/her identity, the client machine establishes a connection with the server, whereby during the handshaking process, a different key will be allocated to secure the connection for the session. The client then requests the server to provide data files associated with the user. The server then requests the relevant information from the central database and relays the data back to the client. The client machine uses this information to perform a verification test. If the test result is positive, the statistics regarding the success of the verification is sent back to the server to be stored into the central database.

Depending on the level of security required, the function of the client machine, and the location of the client machine, some operations can be adapted to optimise the performance-to-security ratio appropriately. For example, when a home PC is used, the data files can be stored on the local computer for later use. This will result in reducing the amount of data transfer necessary between the client and the server. However, when the client uses a station which is not registered as his/her own, then the data files provided by the server will need to be removed from the client station after each process is completed in order to improve the security measures.

An advantage of the above architecture is that it will allow, and accommodate, future expandability and upgradeability beyond that achievable with a conventional software-based system architecture. Additionally, unlike some newly developed online recognition systems (http://www.biometrika.it), the proposed architecture eliminates the need for the installation of software on local terminals. This enhances the usability of the online recognition system considerably as it allows access from any station and any location.

Moreover, the proposed architecture requires only minimal data to be transmitted between client-server-database, as opposed to the transmission of the full raw biometric data. The emergence of load-balancing and distributed systems technology provides the possibility of having servers distributed at different remote locations. This in turn further reduces the time-lag in client-server communications.

2.1.2. *Data format*

As in most client-server architectures, a set of instructions is needed to enable communications between the client software and the server software. The instructions for the system follow a format similar to that shown in Figure 3. The start tag contains one of control, data, or key tags as appropriate for the correct operation of the system.

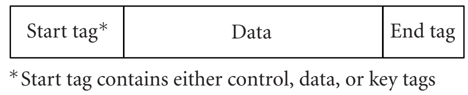


FIGURE 3: Data format tags.

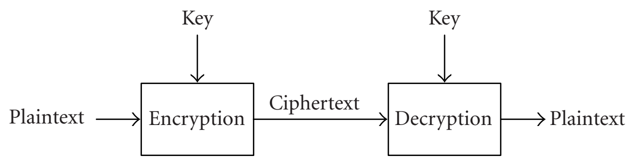


FIGURE 4: Encryption/decryption process.

It is worth noting that the biometric information transferred should be in the form of characteristic features rather than raw data. This will reduce the size of the data to be transferred. Moreover, with this approach, the load on the server can be reduced by performing parts of the processing on the client machine.

2.1.3. *Data security*

The transmission of data over the network requires some form of security measure. Sensitive data such as biometrics needs to be encrypted to prevent others from misusing it. Therefore, the link between the client and server has to be secure throughout the entire process to prevent access or attacks from a hostile source.

To secure the link between the client and the server effectively, the data transmitted between them needs to be in encrypted form. Encryption is a process of disguising/ciphering a message which hides its contents by representing it in a different form. For the purpose of decryption, the exact key used for the encryption process will be needed to restore the original message. Without knowing the key, it will be practically impossible to access the message contents. This process is summarized in Figure 4.

A well-known algorithm for encrypting and decrypting messages is Blowfish [4]. This algorithm is in the public domain and is considered for the purpose of this study. A main advantage of Blowfish is that it is significantly faster than data encryption standard (DES) [5]. A description of Blowfish is presented in the following section.

2.1.4. *Blowfish*

Blowfish is a 64-bit block cipher, and the algorithm consists of two parts. These are a key-expansion part and a data-encryption part. Key expansion converts a key of at most 448 bits into several subkey arrays in a total of 4168 bytes. The data is then encrypted via a 16-round Feistel network, where each round consists of a key-dependent permutation and a key- and data-dependent substitution. All operations are XORs and additions on 32-bit words. The only

Table 1: Dependence of the transmission time(s) on the file size and connection type.

File size (bytes)	Connection					
	Dial-up 56 k	Cable/DSL 512 k	Cable/DSL 1 M	LAN 10 M	LAN 100 M	LAN 1 G
87 k	12.43	1.36	0.68	0.07	0.01	0.6×10^{-3}
130 k	18.57	2.03	1.02	0.10	0.01	1.0×10^{-3}
173 k	24.71	2.70	1.35	0.14	0.01	1.4×10^{-3}
216 k	30.86	3.38	1.69	0.17	0.02	1.7×10^{-3}
259 k	37.00	4.05	2.02	0.20	0.02	2.0×10^{-3}
302 k	43.14	4.72	2.36	0.24	0.02	2.4×10^{-3}
345 k	49.29	5.39	2.70	0.27	0.03	2.7×10^{-3}
388 k	55.43	6.06	3.03	0.30	0.03	3.0×10^{-3}
431 k	61.57	6.73	3.37	0.34	0.03	3.4×10^{-3}
517 k	73.86	8.08	4.04	0.40	0.04	4.0×10^{-3}
603 k	86.14	9.42	4.71	0.47	0.05	4.7×10^{-3}
690 k	98.57	10.78	5.39	0.54	0.05	5.4×10^{-3}
776 k	110.86	12.13	6.06	0.61	0.06	6.1×10^{-3}
862 k	123.14	13.47	6.73	0.67	0.07	6.7×10^{-3}
1024 k	146.29	16.00	8.00	0.80	0.08	8.0×10^{-3}

additional operations are four indexed array data lookups per round.

Blowfish uses a large number of subkeys for encryption or decryption and these keys must be precomputed before any of the above processes can be carried out. The generation of the subkeys involves two arrays consisting of eighteen 32-bit P-arrays subkeys $P_1 \cdots P_{18}$ and four 32-bit S-boxes with 256 entries each.

The calculation of the subkeys is detailed in Schneier's paper [4]. In general, generating the subkeys is a computationally expensive process and requires a total of 521 iterations. However, these keys can then be stored and reused.

2.2. Experimental analysis

The most common connection to the Internet is normally via a dial-up service which ideally offers a maximum transmission speed of 56 kbps. However, cable/ADSL services are becoming more and more available. In an ideal situation, these offer services with transmission speeds of up to 1 Mbps downstream (receiving data) and 512 kbps upstream (sending data). However, the most common transmission speeds of these for receiving and sending data are 512 kbps and 256 kbps, respectively. It should also be noted that these transmission rates might vary considerably during a given connection.

2.2.1. Theoretical transmission rates

The basic approach to calculate the time taken to transmit a file from one location to another via the Internet is based on the following equation:

$$T_s = \frac{Fsz \times 8}{Cnx}, \tag{1}$$

where T_s is the time taken in seconds, Fsz is the file size in bytes, and Cnx is the connection speed in bps.

The above equation assumes an ideal situation where the connection to the Internet and to the destination servers is achieved at the maximum throughput. This, however, is not the actual case on a day-to-day basis.

A comparison of the calculated theoretical transmission time for different file sizes and different connection types is presented in Table 1.

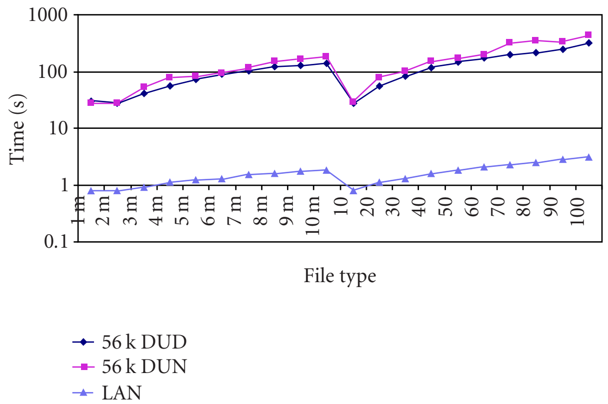
As observed in this table, even in an ideal situation, the use of a dial-up connection involves relatively a long transmission time.

2.2.2. Experimental transmission rates

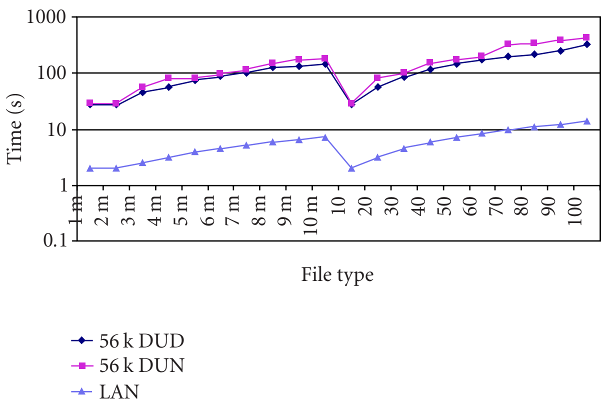
Experiments were conducted at different times using two types of common Internet connections with the file size varying from 4 kb to 900 kb. The files used were signals generated from white noise. These audio files were of 1 to 10 seconds in length. The two types of connection used were a 56 k dial-up connection service and a LAN. The results of this experimental study are given in Figure 5. As it is observed, the transmission time in practice is significantly longer than that suggested theoretically.

The results in Figure 5 clearly indicate that verification over the Internet is unfavourably influenced by the performance of the network. To minimize this, it seems advantageous to compress data before its transmission.

The next set of experiments was based on the transmission of audio models rather than raw data. The previous set of white noise files (Section 2.2.2) was preprocessed and the features were extracted using LPCC-12. These were used to generate audio models based on a VQ with a codebook size of 64. The results of this study are presented in Table 2. As observed, due to the use of VQ, considerable reduction in the file size is achieved. This in turn has resulted in significant reduction in transmission time.



(a)



(b)

FIGURE 5: Experimental transmission rates (DUD: dial-up daytime; DUN: dial-up nighttime). (a) Transmission times without encryption. (b) Transmission times with encryption.

As part of this study, a second set of experiments was conducted based on the encryption of VQ files using the Blowfish algorithm. The results of this investigation are also shown in Table 2. It is seen that there is a slight increase in the overall transmission time in this case. This is due to the initial processing time needed to prepare the data prior to transmission and the time taken to decrypt the data at the receiver. The resultant increase in the overall transmission time is negligible and often not noticeable.

These experimental results indicate the difficulties introduced by the transmission of raw data over the Internet, especially when the file sizes are too large. The results presented were based on the use of audio signal files. It should be noted that image-based biometric data files are of considerably larger sizes. The transmission of such raw files over the Internet may sometimes result in unacceptably long delays in the verification process.

2.3. Comments

A client-server architecture for biometric verification over the Internet has been proposed and described in detail. Based on an analysis of the characteristics of the proposed architecture, its advantages have been discussed, and it has been shown that it provides a practical and systematic approach to the implementation of biometric verification on the Internet. Using a set of experimental investigations, it has been shown that, in practice, it may not be feasible to transmit raw biometric data over the Internet as this can cause unacceptably long delays in the process. It has been demonstrated that the transmission of data models (or features) instead of raw material will significantly reduce the transmission time. Another possibility is to compress biometric data before its transmission. Such compression, however, may unfavourably influence the robustness of biometric techniques (see the next part). Finally, it has been argued that the client-server link should be made secure by encrypting the data before its transmission. It has been shown that the increase in the overall transmission time due to this process is relatively small.

TABLE 2: Transmission time for 4 KB audio models (DUD: dial-up daytime; DUN: dial-up nighttime).

LPCC12 VQ64	Transmission time(s)	
	Without encryption	With encryption
56 k DUD	1.9	2.3
56 k DUN	2.6	2.7
LAN	0.1	0.2

3. SPEAKER VERIFICATION EXPERIMENTS OVER IP NETWORKS

In Section 2, it has been notably shown that transmitting raw biometric data over the Internet may lead to unacceptably long delays. However, recently, considerable progress has been achieved in transmitting voice over the Internet for communication purposes. Thus, this section proposes a methodology for evaluating the speaker verification performance over IP network. The idea is to duplicate an existing and well-known database used for speaker verification (XM2VTS) by passing its speech signals through different coders and different network conditions representative of what can occur over the Internet. Some partners of COST 275 are also evaluating the influence of image and video compression on face recognition performance, again using XM2VTS as it is a multimodal database. Section 3.1 is dedicated to the database description and to the degradation methodology adopted, whereas Second 3.2 presents the speaker verification system and some results obtained with this IP-degraded version of XM2VTS.

3.1. Database used and degradation methodology

3.1.1. XM2VTS database

In acquiring the XM2VTS database (http://www.ee.surrey.ac.uk/Research/VSSP/xm2vtsdb/), 295 volunteers from the University of Surrey visited a recording studio four times at approximately one-month intervals. On each visit, (session) two recordings (shots) were made. The first shot consisted of speech while the second consisted of rotating head movements. Digital video equipment was used to capture the entire database. At the third session, a high-precision 3D model of the subjects head was also built using an active stereo system provided by the Turing Institute. We have chosen this database since many partners of COST Action 275 already use it. The work described in this paper was made on its speech part, where the subjects were asked to read three sentences twice. The three sentences remained the same throughout all four recording sessions and a total of 7080 speech files were made available on 4 CD-ROMs. The audio, which had originally been stored in mono, 16 bit, 32 kHz PCM wave files, was down-sampled to 8 kHz. This is the input sampling frequency required in the speech codecs considered in this study.

3.1.2. Codec used

H323 is a standard for transmitting voice and video. A famous H323 videoconferencing software is for example NetMeeting™. H323 is commonly used to transmit video and voice over IP networks. The audio codecs used in this standard are G711, G722, G723.1, G728, and G729. We propose to use in our experiments the codec which has the lowest bit rate: G723.1 (6.4 and 5.3 kbps), and the one with the highest bit rate: G711 (64 kbps: 8 kHz, 8 bits). Influence of these codecs on speech recognition was evaluated in a former study we made [6], it is thus very exciting to know what will be the results on the speaker verification task.

3.1.3. Packet loss

Simulation with the Gilbert model

There are two main transport protocols used on IP networks. These are UDP and TCP. While UDP protocol does not allow any recovery of transmission errors, TCP includes some error recovery processes. However, the transmission of speech via TCP connections is not very realistic. This is due to the requirement for real-time (or near real-time) operations in most speech-related applications [7]. As a result, the choice is limited to the use of UDP which involves packet loss problems. The process of audio packet loss can be simply characterised using a Gilbert model [8, 9] consisting of two states (Figure 6). One of the states (state 1) represents a packet loss and the other state (state 0) represents the case where packets are correctly transmitted. The transition probabilities in this statistical mode, as shown in Figure 6, are represented by p and q. In other words, p is the probability of going from state 0 to state 1 and q is the probability of going from state 1 to state 0.

Different values of p and q define different packet loss conditions that can occur on the Internet. The probability that n consecutive packets are lost is given by $p(1-q)^{n-1}$. If $(1-q) > p$, then the probability of losing a packet in state 1 (after having already lost a packet) is greater than the probability of losing a packet in state 0 (after having successfully received a packet) [9]. This is generally the case in data transmission on the Internet where packet losses occur

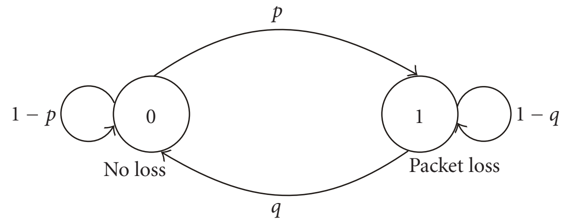


Figure 6: Gilbert model.

as bursts. Note that $p + q$ is not necessarily equal to 1. When p and q parameters are fixed, the mean number of consecutive packets lost can be easily calculated as p/q^2. Of course, the larger this mean is, the more severe the degradation is. Different values of p and q representing different network conditions considered in this study are presented in Table 3 [8, 9].

Real-conditions packet loss

In order to investigate the effects of real network conditions as well, it was decided to play and record the whole speech part of XM2VTS through the network. This was carried out by playing the speech dataset into a computer which was set up for videoconferencing. For this purpose, a transatlantic connection was established between France and Mexico using videoconferencing software. The microphone on the French site was then replaced with the audio output of a computer playing the speech material in XM2VTS. Due to numerous network breakdowns, the transmission of material had to be conducted using several different connections established on different days and at different times. This, of course, provided variations in network conditions that occur in the case of real applications. Table 3 presents a summary of the different coders and simulated network conditions that were considered.

(i) Two degraded versions of XM2VTS were obtained by applying G711 and G723.1 codecs alone without any packet loss.

(ii) Six degraded versions of XM2VTS were obtained using simulated packet loss conditions: 2 conditions (average/bad) ×3 speech qualities (clean/G711/G723.1). The simulated average and bad network conditions considered in this study corresponded to 9% and 30% speech packet loss rates, respectively. Each packet contained 30 milliseconds of speech which was consistent with the duration proposed in Real Time Protocol (RTP) (used under H323).

(iii) One degraded version of XM2VTS based on real network conditions. The transmission was spread from 12/9/02 to 1/10/02 and the mean packet loss rate was 15%. The detailed packet loss conditions for each part of the database are described in Figure 7. Each bar corresponds to a different transmission day and thus to a different transmission condition. We see that in the worst cases, real packet loss rate is around 30%; this

Table 3: Summary of the simulated IP degradation plan (3 codecs $*$ 3 network conditions give 9 different degradations).

Codecs	None (128 kbps)	G711 (64 kbps)	G723.1 (5.3 kbps)
Network condition	No packet loss	Average $p = 0.1$; $q = 0.7$	Bad $p = 0.25$; $q = 0.4$

figure corresponds approximately to the mean packet loss rate measured after simulated IP degradation with $p = 0.25$ and $q = 0.4$ (called bad condition in Table 3). On the other hand, in the best cases, real packet loss rate is around 10% and even less; this corresponds approximately to our simulated "average" condition ($p = 0.1$; $q = 0.7$ in Table 3) for which mean packet loss rate is around 9%.

3.2. *Speaker verification experiments with the ELISA system*

The ELISA consortium groups several public laboratories working on speaker recognition. One of the main objectives of the consortium is to emphasize assessment of performance. Particularly, the consortium has developed a common speaker verification system which has been used for participating at various NIST speaker verification evaluations campaigns [10, 11].

ELISA system is a complete framework designed for speaker verification. It is a Gaussian mixture model (GMM) based system [12] including audio parameterisation as well as score normalization techniques for speaker verification.

This system was presented at NIST from 1998 to 2002 and showed the state-of-the-art performance. ELISA is now collaborating with COST Action 275 concerning performance assessment of multimodal person authentication systems over the Internet. ELISA evaluated the speaker verification performance using the COST 275 dedicated database detailed in Section 3.1.

3.2.1. *Speaker verification protocol on XM2VTS*

For the purpose of this investigation, the Lausanne protocol (configuration 2) is adopted. This has already been defined for the XM2VTS database. There are 199 clients in the XM2VTS database. The training of the client models is carried out using full session 1 and full session 2 of the client part of XM2VTS. Test accesses of 398 clients are obtained using full session 4 ($\times 2$ shots) of the client part. Using the impostor part of the database (70 impostors $\times$ 4 sessions $\times$ 2 shots $\times$ 199 clients $=$ 111440 impostor accesses) 111440 impostor accesses are obtained. The 25 evaluation impostors of XM2VTS are used to develop a world model. The text-independent speaker verification experiments are conducted in matched conditions (same training/test conditions).

3.2.2. *ELISA system on XM2VTS*

The ELISA system on XM2VTS is based on the LIA system presented to NIST 2002 speaker recognition evaluation. The speaker verification system uses 32 parameters: 16 linear frequency cepstral coefficients (LFCC) + 16 DeltaLFCC. Silence frame removal is applied before centring (CMS) and reducing vectors.

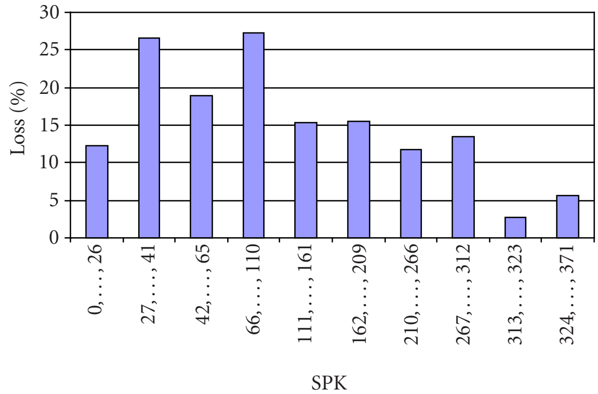


Figure 7: Packet loss measurements for real transmission over IP (different groups of speakers SPK represent different connections).

For the world model, 128-Gaussian component GMM was trained using Switchboard II phase II data (8 kHz landline telephone) and then adapted (MAP [13], mean only) on XM2VTS data (25 evaluation impostors set). The client models are 128-Gaussian component GMM developed by adapting (MAP, mean only) the previous world model.

Decision logic is based on using the conventional log likelihood ratio (LLR). No LLR normalisation such as Znorm [14], Tnorm [15], or Dnorm [16] is applied before the decision process.

3.2.3. *Results*

The speaker verification performance with the simulated degraded versions of XM2VTS is presented in Table 4. We can see that whatever the packet loss level is (no packet loss, average condition, or bad condition), the equal error rate (EER) remains very low for clean speech (no codec) or slightly compressed speech (G711). Based on these results, it can be concluded that, even at a high rate, packet loss alone is not a significant problem for text-independent speaker verification. Comparing these results with those for speech recognition [17], it can be said that the speaker verification performance is far less sensitive to packet loss. On the other hand, the last column of Table 4 shows that the speaker verification performance is adversely affected when the speech material is encoded at low bit rates (e.g., using G723.1). In that case, packet loss increases the degradation. These results are in agreement with those in Section 4 of this paper, describing the performance of speaker verification over wireless mobile devices.

TABLE 4: Results (EER%) of the experiments using degraded XM2VTS.

Network condition	Codecs		
	Clean (128 kbps)	G711 (64 kbps)	G723.1 (5.3 kbps)
No packet loss	0.25%	0.25%	2.68%
Average Network condition $p = 0.1; q = 0.7$	0.25%	0.25%	6.28%
Bad Network condition $p = 0.25; q = 0.4$	0.50%	0.75%	9%

4. SPEAKER VERIFICATION EXPERIMENTS OVER WIRELESS MOBILE DEVICES

Most wireless mobile networks are susceptible to packet loss to some degree. Whilst there exist many strategies to combat packet loss, such as retransmission or packet recovery [17, 18, 19], online identity verification applications may still operate effectively from semi real-time voice streams. This is possible because there is no intrinsic requirement on latency in the case of retransmission. In this part, speaker verification accuracy is assessed against the level of packet loss in wireless mobile devices.

The packet loss scenario is contrasted with degradation coming from additive noise. The degrading effect of ambient noise on automatic speech and speaker recognitions is widely acknowledged and known to be large even for relatively low noise levels. Thus a comparison is made between the two forms of degradation by using otherwise identical experimental conditions.

The remainder of this part is organised as follows. Section 4.1 addresses packet loss in typical wireless and IP networks and its effects on speaker verification. Section 4.2 addresses additive noise and speech enhancement.

Experimental work on the 2000-speaker SpeechDat Welsh [20] database is presented in Section 4.3 with results of experiments using both simulated packet loss and speech enhancement after contamination by additive real car noise.

4.1. *Packet loss in mobile networks*

Some degree of packet loss is inherent in mobile networks. Lost packets might be caused by variable transmission conditions, or the hand-over between neighbouring cells as a wireless mobile device roams about the network.

Approaches dealing with packet loss recovery are generally controlled by the routing protocol adopted in the network architecture. For automatic speech recognition applications where time-sequence information is more critical, packet loss might have a significant impact on performance.

Lost packets might then be retransmitted or some form of compensation employed [17, 18, 19]. In contrast, as seen in Section 3, for speaker verification, a limited degree of packet loss might not have a too detrimental effect, particularly in text-independent mode. This form of speaker verification is generally less dependent on time-sequence information, and there is some evidence in a related study of computational efficiency [21] that speaker verification systems might be relatively insensitive to packet loss. One potential anomaly in this hypothesis, equally applicable to both speech and speaker recognitions, is the effect of lost packets on dynamic features which are computed from their static counterparts over some small window, typically in the order of 100 milliseconds or more. Unless appropriately compensated, packet loss of static features would lead to corrupt dynamic features and performance degradation. This difficulty is circumvented here by assuming that the transmitted features are in fact specific to speech and speaker recognitions rather than conventional codec parameters (as defined in the ETSI AURORA standard [22]). As a consequence, packet loss encompasses both static and dynamic features. Preliminary experiments using a Gilbert model (Section 3.1.3) showed very little sensitivity to the patterns of packet loss, so a balanced loss ($p = 0.25$ and $q = 0.5$) is simulated here with the emphasis placed on the total loss as a percentage of the original.

Experiments are performed with a conventional implementation of a GMM [23] as used by most of today's text-independent speaker verification systems.

4.2. *Additive noise*

The second degradation considered here typifies the conditions under which wireless mobile devices are commonly used, namely, with a meaningful level of background noise.

The consequences of such additive noise are

(i) direct contamination of the speech signal,
(ii) induced changes in the speaking style of the persons subjected to the noise, known as the Lombard reflex [24].

In these experiments, noise is added to the speech recordings thereby minimising any Lombard effects. The noise is added at a moderate level of 15 dB SNR. Subsequently, for completeness, a simple speech enhancement process is applied to the degraded signal.

The form of enhancement considered here has the option of returning the speech to the time domain. Such an approach might lead to suboptimal compensation in terms of recognition performance but nonetheless offers benefits in terms of integration into existing systems and communications networks.

Perhaps the first notable work in this field is that of Boll [25] and Berouti et al. [26] both in 1979. Speech enhancement for human-to-human conversation was performed by an approach still known today as spectral subtraction.

Subsequently, Lockwood and Boudy [27] applied spectral subtraction extensively to automatic speech recognition.

There are many approaches and applications of spectral subtraction. Of particular interest here is an implementation of spectral subtraction termed quantile-based noise estimation (QBNE), proposed by Stahl et al. [28]. QBNE is an extension of the histogram approach presented by Hirsch and Ehrlicher [29]. The main advantage of these approaches is that an explicit speech, nonspeech detector is not required. Noise estimates are continually updated during both non-speech and speech periods from frequency-dependent, temporal statistics of the degraded speech signal. An efficient implementation of QBNE, important in the context of mobile systems, is described in [30].

4.3. Experimental results

4.3.1. Database

The experimental work here was performed on the Speech-Dat Welsh database [20]. The data consists of 2000 speakers recorded over a fixed telephony network. One thousand of the 2000 speakers were used to create a world model and the other 1000 speakers used for speaker model training and testing. Training was performed on approximately 30 seconds of phonetically rich sentences per speaker with a total of about 8 hours for the world model. Two separate text-independent tests used either a 4-digit string, or a single digit, per speaker per test, giving 1000 tests per experiment. Features are standard MFCC-14 static concatenated with 14 dynamic coefficients.

4.3.2. Packet loss and additive noise degradations

To simulate packet loss, approximately 50% of speech features are discarded from the test set, iteratively. No attempt is made to recover these lost vectors although the minimum number of feature vectors per test is capped to two.

Some results are presented in Figures 8 and 9. The detection error trade-off (DET) curves show the system to be highly resilient with minimal increases in error rates until over 75% of the feature vectors are lost, the first three profiles being very close together. This is true for both plots: (Figure 8), the longer, 4-digit string test utterances and (Figure 9) the shorter, single-digit test utterance. Interestingly, in both cases, the profiles diverge toward the left. Considering the 4-digit case (left plot), this indicates that for operating points accepting high false acceptances in return for lower false rejections, the system is particularly robust against packet loss: just 2% false rejections with 50% false acceptances at the extreme case of 98% data loss.

Evidence is presented again in Figure 10 where the EERs are plotted against percentage vector loss and it is clear that the performance begins to degrade only after over 75% of the vectors are lost. This is very much in line with the findings of Section 3 and of McLaughlin et al. [21] who report that a factor of 20 losses can be tolerated before meaningful speaker verification degradation occurs. This finding supports the idea that, in the context of *text-independent* speaker recognition where time sequence information is less critical, there is a large redundancy in typical speech frame rates.

To simulate speaker verification in adverse conditions, the test data is artificially contaminated with car noise at a moderate level of approximately 15 dB SNR.

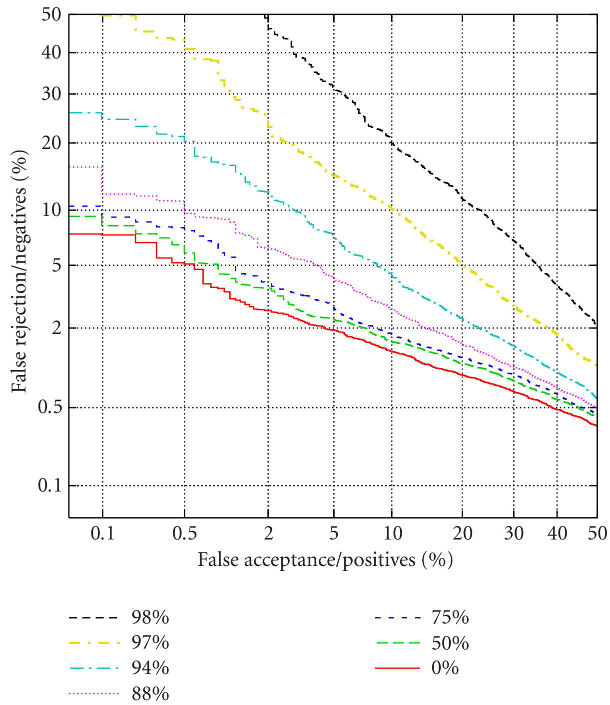


Figure 8: Speaker verification performance for varying degrees of feature vector loss, from 0 up to 98% (with a minimum of 2 feature vectors maintained in all tests) for 4-digit string tests.

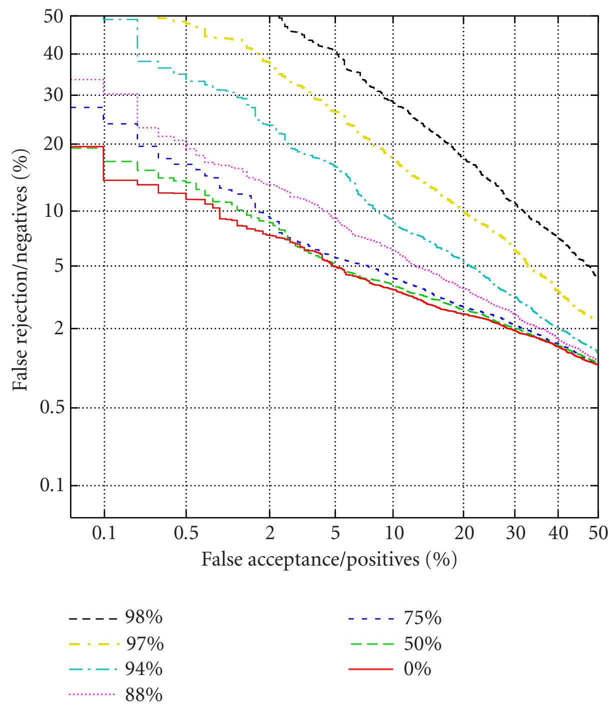


Figure 9: Speaker verification performance for varying degrees of feature vector loss, from 0 up to 98% (with a minimum of 2 feature vectors maintained in all tests) for single-digit tests.

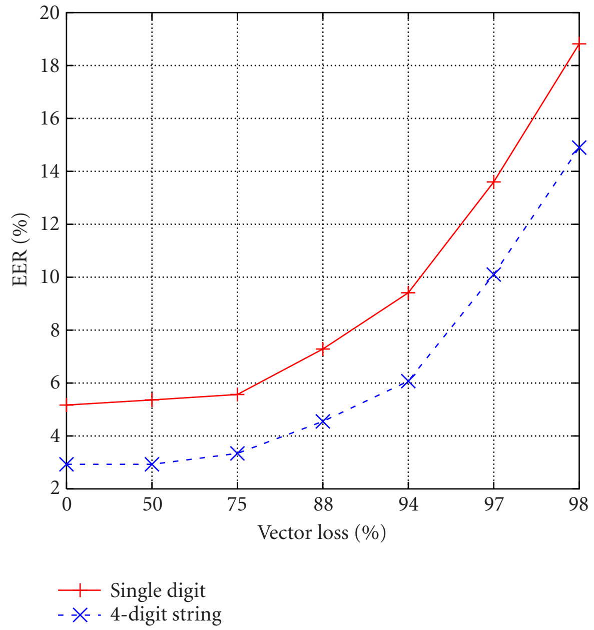


FIGURE 10: EER against feature vector loss (%) for test utterances of 4-digit string (lower profile) and single-digit utterance (upper profile). In all cases, minimum test length is maintained at two vectors.

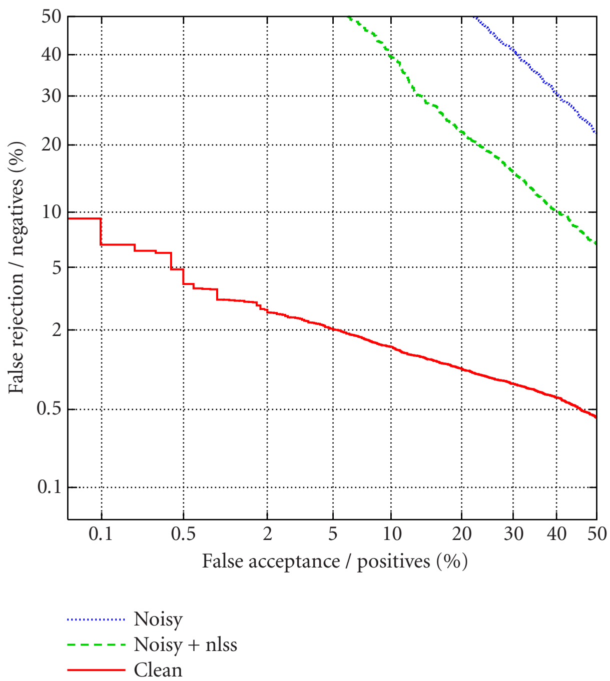


FIGURE 11: Speaker verification performance for the 4-digit string test set with top profile: 15 dB SNR added noise; middle profile: 15 dB SNR added noise plus speech enhancement; and bottom profile: original baseline.

Figure 11 illustrates the effects. The three profiles are for the original telephony test data (bottom profile), the contaminated test data (top profile), and the contaminated data after processing with the speech enhancement approach outlined above (middle profile).

Clearly, the levels of performance degradations are marked, even after compensation. This serves to illustrate how relatively small the degradation from packet loss might prove to be in relation to additive noise.

5. CONCLUSION

This paper has focused on the emerging need of vocal biometric user authentication over the Internet. More precisely, it has presented the constraints tied with the use of the Internet transmission channel, at the protocol level and at the speech signal level.

At the protocol level, the proposed results have shown that a client-server architecture for vocal biometric user authentication over the Internet involves the transmission of data models or features instead of raw biometric materials. A data encryption process for the client-server link has also been recommended.

At the signal level, the experiments have shown that the packet loss is not a main problem for text-independent vocal person authentication. This is in contrast with previous speech recognition experiments where packet loss was found to reduce the accuracy significantly. Moreover, a large degradation of the performance is observed where a low bit rate coder is used. In this case, packet loss increases the degradation.

Experiments using artificially noised wireless audio records have confirmed that environmental noise remains a main drawback for vocal biometric authentication over the Internet.

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Laurent Besacier received his Ph.D. degree in computer science in April 1998 on "A parallel model for automatic speaker recognition" from the University of Avignon, France. Then he spent one and a half year at IMT (Switzerland) as an Associate Researcher working on M2VTS European project (multimodal person authentication). Since September 1999, he has been an Associate Professor in the University of Joseph Fourier (Grenoble). He carries out research on automatic speech and speaker recognition within the GEOD team at CLIPS Lab. He published about 30 papers on various aspects of speech recognition and speaker recognition. He is in the board of AFCP, the French Speaking Speech Communication Association. He is the supervisor of over 5 Ph.D. students in the area of speaker and speech recognition. His research interests lie in automatic speech and speaker recognition: indexation and tracking of audio documents.

Aladdin M. Ariyaeeinia received his B.Eng degree in telecommunication engineering, M.S. in digital signal processing, and Ph.D. degree in artificial intelligence in 1976, 1982, and 1986, respectively. He was awarded Chartered Engineer status in 1988. In 1986, he joined the University of East Anglia as a Senior Research Fellow. Over the last fourteen years, Ariyaeeinia has been working in the Faculty of Engineering and Information Sciences, the University of Hertfordshire. During this period, he has been conducting research on various aspects of speech processing in close collaboration with industry. He is now a Reader in signal processing, and responsible for leading the Multimedia and Internet Technologies Group. Ariyaeeinia's current research interests include speaker and language recognition, speaker-based audio-visual data indexation, biometrics-based recognition over the Internet, and speech enhancement. He has over 40 publications, and has served on various scientific committees.

John S. Mason is a Senior Lecturer at the Department of Electrical and Electronic Engineering. He received his M.S. and Ph.D. degrees from the University of Surrey in 1971 and 1974, respectively, joining the University of Wales Swansea as a Lecturer in May 1973. In 1979, he took up a one-year appointment as a Senior Research Engineer at Hewlett Packard Ltd in South Queensferry, and in 1994 he was invited to work on an international project in the Australian National University, Canberra, as a Visiting Research Fellow. From the time of his Ph.D. studies through today, his research interest has focused on digital signal processing. Of a particular note is the work done on finding solutions to complex Chebyshev approximations, widely acknowledged as the first to solve this long-standing problem. More recently, his research has revolved around speech and speaker recognition and multimedia signal processing. In these areas, he has served on the technical committees of a number of international research meetings.

Jean-François Bonastre has been an Associate Professor at the LIA, the University of Avignon computer laboratory since 1994. He studied computer science in the University of Marseille and obtained a DEA (Master) in artificial intelligence in 1990. He obtained his Ph.D. degree in 1994, from the University of Avignon, and his HDR (Ph.D. supervision diploma) in 2000, both in computer science, both on speech science, more precisely, on speaker recognition. J.-F. Bonastre is the current President of the AFCP, the French Speaking Speech Communication Association (a Regional Branch of ISCA). He was the Chairman of the RLA2C workshop (1998) and a member of the Program Committee of Speaker Odyssey Workshops (2001 and 2004). J.-F. Bonastre has been an Invited Professor at Panasonic Speech Technology Lab. (PSTL), Calif, USA, in 2002.

Pedro Mayorga received his diploma in physics from Universidad Autonoma de Baja California (UABC), Ensenada, Baja California, in 1992. He received the M.S. degree in digital systems from Instituto Politecnico Nacional de Mexico in 1998 with a thesis on fingerprints recognition using neural networks. He is currently a Ph.D. student at CLIPS laboratory, Institut National Polytechnique de Grenoble, France. His Ph.D. research involves the application of digital signal processing and speech recognition to the vocal recognition servers. From 1988 to 1992, he worked in the computer laboratory, Facultad de Ciencias, UABC, Ensenada, Baja California. From 1993 to 1994, he was an Associate Research and Associate Professor at the Instituto de Ingenieria, UABC, Mexicali, Baja California, at the Department of Electrical and Electronic. He was a teacher in microprocessors and digital control systems courses at the same university department. From 1993 to 1994, he worked as a part-time Associate Professor at the Instituto Tecnologico de Mexicali (ITM). Since 1994, he has been a whole-time Associate Professor at the Department of Electrical-Electronic, ITM. From 1994 to 2000, he was teaching in microprocessors, microcontrollers, digital control, signal processing, and signal and systems courses at the ITM Engineering School.

Corinne Fredouille obtained her Ph.D. degree in 2000 in the field of automatic speaker recognition. She has joined the computer science laboratory LIA, University of Avignon, and more precisely the speech processing team, as an Assistant Professor in 2003. Currently, she is an active member of the European ELISA Consortium, of AFCP, the French Speaking Speech Communication Association, and of ISCA/SIG SPLC (Speaker and Language Characterization Special Interest Group).

Johann Siau received his B.Eng degree in electrical and electronic engineering in 1997. In 2001, he joined the University of Hertfordshire as a full-time academic staff. Over the past few years, Johann has been working with the Faculty of Engineering and Information Sciences and during this period, he has been conducting research on various aspects of speaker verification technologies, including verification over the Internet. He is currently managing a significant part of the multimedia network at the department and is a member of the Multimedia and Internet Technologies Group. Johann's current research interests include speaker recognition, biometrics-based recognition over the Internet, and network security and vulnerabilities.

Nicholas W. D. Evans received the M.Eng degree in electronics and computing science from the University of Wales Swansea in 1999. He then joined the Speech and Image Research Group to pursue his Ph.D. degree sponsored by the Engineering and Physical Sciences Research Council. In 2002, he became a Lecturer in communications at the School of Engineering. Nick's research interests include time-frequency analysis for noise estimation, speech enhancement, noise compensation, noise robust automatic speech recognition, and biometric speaker verification. Nick is a member of ISCA and IEE.

Roland Auckenthaler worked with Ensigma Ltd. as a Teaching Company Associate from 1998 to 2000 and received his Ph.D. degree from the University of Wales Swansea in 2002 in the area of speaker verification. He now works with Ubiquity Software Corporation in the area of Internet telephony and does part-time research within the University of Wales Swansea. Roland is also a holder of a patent in the area of speaker verification.

Robert Stapert moved from the Netherlands to the UK in 1996. There, in his capacity as a Ph.D. student, he spent four years at Swansea University's Speech and Image Processing laboratory. His theme was enhancing speaker verification by means of time sequence information. He completed his Ph.D. in 2000. Since then, he has been employed at Aculab in Milton Keynes, UK, as a member of their digital signal processing team, working as a Software Engineer. He is responsible for the design and development of Aculab's speaker verification product. Further, he is working on projects related to text to speech, speech recognition, as well as various nonspeech related projects.

EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

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Many important applications of multimedia revolve around the detection of humans and the interpretation of human behavior, for example, surveillance and intrusion detection, automatic analysis of sports videos, broadcasts, movies, ambient assisted living applications, video conferencing applications, and so forth. Success in this task requires the integration of various data modalities including video, audio, and associated text, and a host of methods from the field of machine learning. Additionally, the computational efficiency of the resulting algorithms is critical since the amount of data to be processed in videos is typically large and real-time systems are required for practical implementations.

Recently, there have been several special issues on the human detection and human-activity analysis in video. The emphasis has been on the use of video data only. This special issue is concerned with contributions that rely on the use of multimedia information, that is, audio, video, and, if available, the associated text information.

Papers on the following and related topics are solicited:

- Video characterization, classification, and semantic annotation using both audio and video, and text (if available).
- Video indexing and retrieval using multimedia information.
- Segmentation of broadcast and sport videos based on audio and video.
- Detection of speaker turns and speaker clustering in broadcast video.
- Separation of speech and music/jingles in broadcast videos by taking advantage of multimedia information.
- Video conferencing applications taking advantage of both audio and video.
- Human mood detection, and classification of interactivity in duplexed multimedia signals as in conversations.
- Human computer interaction, ubiquitous computing using multimedia.
- Intelligent audio-video surveillance and other security-related applications.

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Special Issue on

Signal Processing for Location Estimation and Tracking in Wireless Environments

Call for Papers

In recent years, the proliferation of mobile computing devices and wireless technologies has fostered a growing interest in location-aware systems and services. The availability of location information on objects and human beings is critical in many military and civilian applications such as emergency call services, tracking of valuable assets, monitoring individuals with special needs in assisted living facilities, location-assisted gaming (e.g., Geocaching), etc.

Existing positioning systems can be categorized based on whether they are intended for indoor or outdoor applications. Within both of these application areas, there are two major categories of position estimation techniques, as discussed below.

- *Geometric techniques*–Position is estimated by exploiting time of arrival (TOA), time difference of arrival (TDOA), angle of arrival (AOA) or other information derived from the relationship between the geometry of an array of receivers and the modeled propagation characteristics of the transmitted signal.
- *Mapping approaches*–Position is estimated based on comparison of local measurements to a "map" of expected distribution of the measured values. For example, in a wireless LAN application, received signal strength (RSS) might be observed either at the location of the client or at a remote reference point. Mapping approaches are also known as location fingerprinting.

Although geometric approaches have the potential to achieve higher precision than mapping approaches, they generally require direct-path signal reception or accurate environmental information at the receiver and often perform poorly in complex multipath environments. On the other hand, estimation accuracy of mapping approaches is limited by both the accuracy of the reference map and the accuracy of observed measurements. Furthermore, frequent and extensive site-survey measurements are often needed to accommodate the time varying nature of wireless channels, structural changes in the environment, and upgrades of wireless infrastructure.

In addition to snapshots of AOA, TOA, TDOA or RSS measurements, motion models or prior knowledge of structural constraints can often be used to enhance location estimation accuracy for mobile objects by "tracking" location estimates over time. Trackers that integrate such information into the computation of location estimates are generally implemented using techniques such as Kalman filters, particle filters, Markov chain Monte Carlo methods, etc.

The purpose of the proposed special issue is to present a comprehensive picture of both the current state of the art and emerging technologies in signal processing for location estimation and tracking in wireless environments. Papers are solicited on all related aspects from the point of view of both theory and practice. Submitted articles must be previously unpublished and not concurrently submitted for publication on other journals.

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

- Received signal strength (RSS), angle-of-arrival (AOA), and time-based location estimation
- Ultrawideband (UWB) location estimation
- Bayesian location estimation and tracking
- Pattern recognition and learning theory approaches to location estimation
- Applications of expectation-maximization (EM) and Markov chain Monte Carlo (MCMC) techniques
- Applications of electromagnetic propagation modeling to location estimation
- Mitigation of errors due to non-line-of-sight propagation
- System design and configuration
- Performance evaluation, performance bounds, and statistical analysis

- Computational complexity and distributed computation
- Distributed location estimation
- Synchronization issues
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GUEST EDITORS:

Richard J. Barton, Department of Electrical and Computer Engineering, University of Houston, Houston, TX 77204, USA; rbarton@uh.edu

Rong Zheng, Department of Computer Science, University of Houston, Houston, TX 77204, USA; rzheng@cs.uh.edu

Sinan Gezici, Mitsubishi Electric Research Labs, Cambridge, MA 02139, USA; gezici@ieee.org

Venugopal V. Veeravalli, Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801, USA; vvv@uiuc.edu

Special Issue on Track Before Detect Algorithms

Call for Papers

Seamless detection and tracking schemes are able to integrate unthresholded (or below target detection threshold) multiple sensor responses over time to detect and track targets in low signal-to-noise ratio (SNR) and high clutter scenarios. These schemes, also called "track-before-detect (TBD)" algorithms are especially suitable for tracking weak targets that would only very rarely cross a standard detection threshold as applied at the sensor level.

Thresholding sensor responses result in a loss of information. Keeping this information allows some TBD approaches to deal with the classical data association problem effectively in high clutter and low SNR situations. For example, in detection scenarios with simultaneous activation/illumination from different signal sources this feature allows the application of triangulation techniques, where in the case of contact tracking approaches essential information about weak targets would often be lost because these targets did not produce signals that cross the normal detection threshold. Extending this example to a multi-sensor network scenario, a TBD algorithm that can use unthresholded (or below threshold) data has the potential to show improved performance compared to an algorithm that relies on thresholded data. In low SNR situations, this can substantially increase performance particularly in the case of a dense multi-target scenario.

Naturally, TBD algorithms consume high computational processing power: An efficient realization and coding of the TBD scheme is mandatory.

Another issue that arises when using the TBD scheme is the quality of the sensor model: Practical experience with thresholded data shows that a coarser modelling of the likelihood function might be sufficient and often leads to robust algorithms. How much have these sensor models to be improved in order to allow the TBD algorithms to exploit the information provided with the unthresholded data?

TBD algorithms that are well known to the tracking community are the likelihood ratio detection and tracking (LRDT), maximum likelihood probabilistic data association (MLPDA), maximum likelihood probabilistic multihypothesis tracking (MLPMHT), Houghtransform based methods and dynamic programming techniques; also related are the probability hypothesis density (PHD), the histogram probabilistic multi- hypothesis tracking (H-PMHT) algorithms, and, of course, various particle filter approaches. Some of these algorithms are capable of tracking extended targets and performing signal estimation in multi-sensor measurements.

The aim of this special issue is to focus on recent developments in this expanding research area. The special issue will focus on one hand on the development and comparison of algorithmic approaches, and on the other hand on their currently ever-widening range of applications such as in active or passive surveillance scenarios (e.g. for object tracking and classification with image and video based sensors, or scenarios involving chemical, electromagnetic and acoustic sensors). Special interest lies in multi-sensor data fusion and/or multi-target tracking applications.

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GUEST EDITORS:

Yvo Boers, Surface Radar, Thales Nederland B.V. Haaksbergerstraat 49, 7554 PA Hengelo, The Netherlands; yvo.boers@nl.thalesgroup.com

Frank Ehlers, NATO Undersea Research Centre, Viale S. Bartolomeo 400, 19138 La Spezia, Italy; frankehlers@ieee.org

Wolfgang Koch, FGAN-FKIE, Neuenahrer Strasse 20, D53343 Wachtberg, Germany; w.koch@fgan.de

Tod Luginbuhl, Naval Undersea Warfare Center, 1176 Howell Street, Newport, RI 02841-1708, USA; t.e.luginbuhl@ieee.org

Lawrence D. Stone, Metron Inc., 11911 Freedom Drive, Suite 800, Reston, VA 20190, USA; stone@metsci.com

Roy L. Streit, Metron Inc., 11911 Freedom Drive, Suite 800, Reston, VA 20190, USA; streit@metsci.com

Special Issue on

Advanced Signal Processing and Pattern Recognition Methods for Biometrics

Call for Papers

Biometric identification has established itself as a very important research area primarily due to the pronounced need for more reliable and secure authentication architectures in several civilian and commercial applications. The recent integration of biometrics in large-scale authentication systems such as border control operations has further underscored the importance of conducting systematic research in biometrics. Despite the tremendous progress made over the past few years, biometric systems still have to reckon with a number of problems, which illustrate the importance of developing new biometric processing algorithms as well as the consideration of novel data acquisition techniques. Undoubtedly, the simultaneous use of several biometrics would improve the accuracy of an identification system. For example the use of palmprints can boost the performance of hand geometry systems. Therefore, the development of biometric fusion schemes is an important area of study. Topics related to the correlation between biometric traits, diversity measures for comparing multiple algorithms, incorporation of multiple quality measures, and so forth need to be studied in more detail in the context of multibiometrics systems. Issues related to the individuality of traits and the scalability of biometric systems also require further research. The possibility of using biometric information to generate cryptographic keys is also an emerging area of study. Thus, there is a definite need for advanced signal processing, computer vision, and pattern recognition techniques to bring the current biometric systems to maturity and allow for their large-scale deployment.

This special issue aims to focus on emerging biometric technologies and comprehensively cover their system, processing, and application aspects. Submitted articles must not have been previously published and must not be currently submitted for publication elsewhere. Topics of interest include, but are not limited to, the following:

- Fusion of biometrics
- Analysis of facial/iris/palm/fingerprint/hand images
- Unobtrusive capturing and extraction of biometric information from images/video
- Biometric identification systems based on face/iris/palm/fingerprint/voice/gait/signature
- Emerging biometrics: ear, teeth, ground reaction force, ECG, retina, skin, DNA
- Biometric systems based on 3D information
- User-specific parameterization
- Biometric individuality
- Biometric cryptosystems
- Quality measure of biometrics data
- Sensor interoperability
- Performance evaluation and statistical analysis

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GUEST EDITORS:

Nikolaos V. Boulgouris, Department of Electronic Engineering, Division of Engineering, King's College London, London WC2R 2LS, UK; nikolaos.boulgouris@kcl.ac.uk

Juwei Lu, EPSON Edge, EPSON Canada Ltd., Toronto, Ontario M1W 3Z5, Canada; juwei@ieee.org

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Arun Ross, Lane Department of Computer Science & Electrical Engineering, West Virginia University, Morgantown WV, 26506, USA; arun.ross@mail.wvu.edu

Special Issue on
Signal Processing for Data Converters

Call for Papers

Data converters (ADCs and DACs) ultimately limit the performance of today's communication systems. New concepts for high-speed, high-resolution, and power-aware converters are therefore required, which also lead to an increased demand for high-speed and high-resolution sampling systems in the measurement industry. Present converter technologies operate on their limits, since the downscaling of IC technologies to deep submicron technologies makes their design increasingly difficult. Fortunately, downscaling of IC technologies allows for using additional chip area for digital signal processing algorithms with hardly any additional costs. Therefore, one can use more elaborate signal processing algorithms to improve the conversion quality, to realize new converter architectures and technologies, or to relax the requirements on the analog design. Pipelined ADCs constitute just one example of converter technology where signal processing algorithms are already extensively used. However, time-interleaved converters and their generalizations, including hybrid filter bank-based converters and parallel sigma-delta-based converters, are the next candidates for digitally enhanced converter technologies, where advanced signal processing is essential. Accurate models constitute one foundation of digital corrected data converters. Generating and verifying such models is a complex and time-consuming process that demands high-performance instrumentation in conjunction with sophisticated software defined measurements.

The aim of this special issue is to bring forward recent developments on signal processing methods for data converters. It includes design, analysis, and implementation of enhancement algorithms as well as signal processing aspects of new converter topologies and sampling strategies. Further, it includes design, analysis, and implementation of software defined measurements for characterization and modeling of data converters.

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

- Analysis, design, and implementation of digital algorithms for data converters
- Analysis and modeling of novel converter topologies and their signal processing aspects
- Digital calibration of data converters
- Error identification and correction in time-interleaved ADCs and their generalizations
- Signal processing for application-specific data converters (communication systems, measurement systems, etc.)
- New sampling strategies
- Sampling theory for data converters
- Signal processing algorithms for data converter testing
- Influence of technology scaling on data converters and their design
- Behavioral models for converter characterization
- Instrumentation and software defined measurements for converter characterization

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GUEST EDITORS:

Christian Vogel, Signal Processing and Speech Communication Laboratory, Graz University of Technology, 8010 Graz, Austria; vogel@tugraz.at

Håkan Johansson, Department of Electrical Engineering, Linköping University, 581 83 Linköping, Sweden; hakanj@isy.liu.se

Boris Murmann, Department of Electrical Engineering, Stanford University, Stanford, CA 94305-4070, USA; murmann@stanford.edu

EURASIP JOURNAL ON ADVANCES IN SIGNAL PROCESSING

Special Issue on Distributed Space-Time Systems

Call for Papers

Diversity is a powerful technique to mitigate channel fading and to improve robustness to cochannel interference in a wireless network. Space-time wireless systems traditionally use multiple colocated antennas at the transmitter and receiver along with appropriate signal design (also known as space-time coding) to realize spatial diversity in the link. Typically this diversity can augment any frequency and time diversity available to the receiver. Multiple antennas also offer the ability to use spatial multiplexing to dramatically increase the data rate.

A recent development in this area aims at dispensing with the need for colocated antennas. Popularly known as the cooperative diversity technique, this uses the antennas at multiple user terminals in a network in the form of a virtual antenna array to realize spatial diversity in a distributed fashion. Such techniques create new challenges in the design of wireless systems.

The purpose of this call for papers is to address some of these challenges such as new protocols for cooperative diversity, cross-layer design, cooperative multiplexing, space-time coding for distributed antennas, cooperative channel estimation and equalization, selecting the right users for participating in a cooperative network, modulation specific issues like OFDMA and CDMA, and distributed space-time processing for sensor networks.

Papers on the following and related topics are solicited for this special issue:

- New protocols for cooperative diversity systems
- Cross-layer protocol design
- Signal design for distributed space-time systems
- Cooperative channel estimation and equalization
- Cooperative MIMO systems
- Distributed space-time processing for sensor networks
- Power allocation in distributed space-time systems
- Fast algorithms and efficient architectures for virtual MIMO receivers
- Energy efficient relay network architectures

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GUEST EDITORS:

Ranjan K. Mallik, Department of Electrical Engineering, Indian Institute of Technology - Delhi, Hauz Khas, New Delhi 110016, India; rkmallik@ee.iitd.ernet.in

Arogyaswami J. Paulraj, Department of Electrical Engineering, Stanford University, 232 Packard, 350 Serra Mall, Stanford, CA 94305, USA; apaulraj@stanford.edu

Mrityunjoy Chakraborty, Department of Electronics and Electrical Communication Engineering, Indian Institute of Technology - Kharagpur, Kharagpur 721302, West Bengal, India; mrityun@ece.iitkgp.ernet.in

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George K. Karagiannidis, Department of Electrical and Computer Engineering, Aristotle University of Thessaloniki, 54 124 Thessaloniki, Greece; geokarag@auth.gr

Special Issue on

Cooperative Localization in Wireless Ad Hoc and Sensor Networks

Call for Papers

One of the major requirements for most applications based on wireless ad hoc and sensor networks is accurate node localization. In fact, sensed data without position information is often less useful.

Due to several factors (e.g., cost, size, power), only a small fraction of nodes obtain the position information of the anchor nodes. In this case, a node has to estimate its position without a direct interaction with anchor nodes and a cooperation between nodes is needed in a multihop fashion. In some applications, none of the nodes are aware of their absolute position (anchor-free) and only relative coordinates are estimated instead.

Most works reported in the literature have studied cooperative localization with the emphasis on algorithms. However, very few works give emphasis on the localization as estimation or on the investigation of fundamental performance limits as well as on experimental activities. In particular, the fundamental performance limits of multihop and anchor-free positioning in the presence of unreliable measurements are not yet well established. The knowledge of such limits can also help in the design and comparison of new low-complexity and distributed localization algorithms. Thus, measurement campaigns in the context of cooperative localization to validate the algorithms as well as to derive statistical models are very valuable.

The goal of this special issue is to bring together contributions from signal processing, communications and related communities, with particular focus on signal processing, new algorithm design methodologies, and fundamental limitations of cooperative localization systems. Papers on the following and related topics are solicited:

- anchor-based and anchor-free distributed and cooperative localization algorithms that can cope with unreliable range measurements
- derivation of fundamental limits in multihop and anchor-free localization scenarios
- new localization algorithms design methodologies based, for example, on statistical inference and factor graphs
- low-complexity and energy-efficient distributed localization algorithms
- distributed ranging and time synchronization techniques
- measurement campaigns and statistical channel modeling
- algorithm convergence issues
- UWB systems
- localization through multiple-antenna systems
- experimental results

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GUEST EDITORS:

Davide Dardari, WiLAB, Department of Electronics, Computer Science and Systems (DEIS), University of Bologna at Cesena, Italy; ddardari@ieee.org

Chia-Chin Chong, DoCoMo Communications Laboratories USA, Inc., Palo Alto, CA 94304-1201, USA; cchong@docomolabs-usa.com

Damien B. Jourdan, Athena Technologies, Inc., Warrenton, VA 20187, USA; jourdan@alum.mit.edu

Lorenzo Mucchi, Department of Electronics and Telecommunications, University of Florence, 50121 Florence, Italy; mucchi@lenst.det.unifi.it

Special Issue on Information Theoretic Methods for Bioinformatics

Call for Papers

Information theoretic methods for modeling are at the center of the current efforts to interpret bioinformatics data. The high pace at which new technologies are developed for collecting genomic and proteomic data requires a sustained effort to provide powerful methods for modeling the data acquired. Recent advances in universal modeling and minimum description length techniques have been shown to be well suited for modeling and analyzing such data. This special issue calls for contributions to modeling of data arising in bioinformatics and systems biology by information theoretic means. Submissions should address theoretical developments, computational aspects, or specific applications. Suitable topics for this special issue include but are not limited to:

- Normalized maximum-likelihood (NML) universal models
- Minimum description length (MDL) techniques
- Microarray data modeling
- Denoising of genomic data
- Pattern recognition
- Data compression-based modeling

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

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GUEST EDITORS:

Jorma Rissanen, Computer Learning Research Center, University of London, Royal Holloway, TW20 0EX, UK; jorma.rissanen@mdl-research.org

Peter Grünwald, Centrum voor Wiskunde en Informatica (CWI), National Research Institute for Mathematics and Computer Science, P.O. Box 94079, 1090 GB Amsterdam, The Netherlands; pdg@cwi.nl

Jukka Heikkonen, Laboratory of Computational Engineering, Helsinki University of Technology, P.O. Box 9203, 02015 HUT, Finland; jukka.heikkonen@tkk.fi

Petri Myllymäki, Department of Computer Science, University of Helsinki, P.O. Box 68 (Gustaf Hällströmin katu 2b), 00014, Finland; petri.myllymaki@cs.helsinki.fi

Teemu Roos, Complex Systems Computation Group, Helsinki Institute for Information Technology, University of Helsinki, P.O.Box 68, 00014, Finland; teemu.roos@hiit.fi

Juho Rousu, Department of Computer Science, University of Helsinki, P.O. Box 68 (Gustaf Hällströmin katu 2b), 00014, Finland; juho.rousu@cs.helsinki.fi

EURASIP JOURNAL ON WIRELESS COMMUNICATIONS AND NETWORKING

Special Issue on Multimedia over Wireless Networks

Call for Papers

Scope

In recent years there has been a tremendous increase in demand for multimedia delivered over wireless networks. The design and capabilities of the mobile devices and the services being offered reflect the increase in multimedia usage in the wireless setting. Applications that are in the process of becoming essential to users include video telephony, gaming, or TV broadcasting. This trend creates great opportunities for identifying new wireless multimedia applications, and for developing advanced systems and algorithms to support these applications. Given the nature of the channel and of the mobile devices, issues such as scalability, error resiliency, and energy efficiency are of great importance in applications involving multimedia transmission over wireless networks.

The papers in this issue will focus on state-of-the-art research on all aspects of wireless multimedia communications. Papers showing significant contributions are solicited on topics including but are not limited to:

- Error resilience and error concealment algorithms
- Rate control for wireless multimedia coding
- Scalable coding and transmission
- Joint source-channel coding
- Joint optimization of power consumption and rate-distortion performance
- Wireless multimedia traffic modeling
- Wireless multimedia streaming
- Wireless multimedia coding
- QoS for wireless multimedia applications
- Distributed multimedia coding

Authors should follow the EURASIP Journal on Wireless Communications and Networking manuscript format described at http://www.hindawi.com/journals/wcn/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP Journal on Wireless Communications and Networking's Manuscript Tracking System at http://www.hindawi.com/mts/, according to the following timetable:

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GUEST EDITORS:

Dan Lelescu, R&D lab, Imaging Division, Micron Technology, Inc., 2125 O'Nel Drive, San Jose, CA 9513, USA; danlelescu@yahoo.com

Peter Schelkens, Deptartment of Electronics and Information Processing (ETRO), Vrije Universiteit Brussel (VUB), Pleinlaan 2, 1050 Brussel, Belgium; Peter.Schelkens@vub.ac.be

Kameswara Rao Namuduri, Deptartment of Electrical and Computer Engineering, Wichita State University, 1845 Fairmount, Wichita, 67260 KS, USA; kamesh.namuduri@wichita.edu

EURASIP JOURNAL ON IMAGE AND VIDEO PROCESSING

Special Issue on Image and Video Processing for Disability

Call for Papers

New technologies represent a great opportunity for the improvement of life and independent living of the disabled and elder people. Over the last decade, active researches have produced novel algorithms for blind, deaf, mute people or for people with severe motor disabilities. These researches are strongly related with the development of new dedicated systems for human-computer interactions.

Whatever the kind of handicap, image processing can provide a significant help for disability compensation to avoid the gap increasing between disabled and nondisabled people with respect to the new technologies.

Researches for new systems for disabled people are multidisciplinary research from engineering sciences (computer science, HCI, automatic, electronics, etc.) and human sciences (psychology, cognition, etc.). Here we are focusing on researches involving image and video processing for disability. However, multimodal signals-based systems can be considered.

The goal of this special issue is to provide original contributions in the field of image and video processing for disability.

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

- Eye-gaze analysis and interpretation
- Head motion analysis
- Human behavior modeling
- HCI for disabled people
- Hand-gesture analysis and interpretation
- Sign language recognition
- Modality replacement
- Multimodal systems for disabled
- Facial expressions interpretation

In each case, works should be related to an application dedicated to disabled or elder people's help.

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GUEST EDITORS:

Dimitrios Tzovaras, Informatics and Telematics Institiut, Centre for Research and Technology Hellas, 1st Km Thermi-Panorama Road, Thermi_Thessaloniki, GR 57001, Greece; tzovaras@iti.gr

Patrice Dalle, Equipe de Traitement et Compréhension d'Image (TCI), Institute de Recherche en Informatique de Toulouse (IRIT),Université Paul Sabatier, 118 Route de Narbonne, 31062 Toulouse cedex 4, France; dalle@irit.fr

Thierry Pun, Centre Universitaire d'Informatique (CUI), Université de Genèva, 24 Rue Général Dufour, 1211 Geneva 4, Switzerland; thierry.pun@cui.unige.ch

Alice Caplier, Laboratoire des Image et des Signaux (LSI), Institute National Polytechnique de Grenoble (INPG), 46 avenue Félix Viallet, 38031 Grenoble cedex, France; caplier@lis.inpg.fr

Special Issue on Anthropocentric Video Analysis: Tools and Applications

Call for Papers

Humans are a basic entity in most videos. Lately, there has been increased interest in devising automated video analysis algorithms that aim to extract, efficiently describe, and organize information regarding the state or state transition of individuals (identity, emotional state, activity, position and pose, etc), interactions between individuals (dialogue, gestures, engagement into collaborative or competitive activities like sports), physical characteristics of humans (anthropometric characteristics, 3D head/body models), and so forth. Such information can be utilized in a multitude of important applications that include, but are not limited to:

- Human computer interaction, ubiquitous computing
- Video characterization, classification, and semantic annotation
- Video indexing and retrieval
- Temporal video segmentation (shot and scene boundary detection) and summarization
- Intelligent video surveillance, access control, and other security related applications

High quality and original contributions on the following (nonexhaustive) list of topics related to anthropocentric video analysis and its applications are solicited:

- Detection and tracking of humans or human body parts
- Action recognition and human behavior analysis
- Emotional state recognition
- Anthropocentric video characterization, semantic annotation, indexing, retrieval, temporal segmentation and summarization
- Efficient description schemes for anthropocentric video information
- Dialogue detection, LiP activity detection, visual speech recognition
- Hand gesture recognition
- 3D modeling of humans
- Person verification and recognition

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GUEST EDITORS:

Nikos Nikolaidis, Department of Informatics, Aristotle University of Thessaloniki, Greece; nikolaid@aiia.csd.auth.gr

Maja Pantic, Department of Computing, Imperial College London, UK; m.pantic@imperial.ac.uk

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IEEE ICME 2007 Call for Papers
2007 International Conference on Multimedia & Expo (ICME)

July 2-5, 2007
Beijing International Convention Center, Beijing, China

Sponsored by: Circuits and Systems Society, Communications Society, Computer Society, and Signal Processing Society.

IEEE International Conference on Multimedia & Expo is a major annual international conference with the objective of bringing together researchers, developers, and practitioners from academia and industry working in all areas of multimedia. ICME serves as a forum for the dissemination of state-of-the-art research, development, and implementations of multimedia systems, technologies and applications. ICME is co-sponsored by four IEEE societies including the Circuits and Systems Society, the Communications Society, the Computer Society, and the Signal Processing Society. The conference will feature world-class plenary speakers, exhibits, special sessions, tutorials, and paper presentations.

Prospective authors are invited to submit a four-page paper in double-column format including authors' names, affiliations, and a short abstract. Only electronic submissions will be accepted. Topics include but are not limited to:

- Audio, image, video processing
- Virtual reality and 3-D imaging
- Signal processing for media integration
- Multimedia communications and networking
- Multimedia security and content protection
- Multimedia human-machine interface and interaction
- Multimedia databases
- Multimedia computing systems and appliances
- Hardware and software for multimedia systems
- Multimedia standards and related issues
- Multimedia applications
- Multimedia and social media on the Internet

A number of awards will be presented to the Best Papers and Best Student Papers at the conference. Participation for special sessions and tutorial proposals are encouraged.

SCHEDULE

- Special Session Proposals Due: **December 1, 2006**
- Tutorial Proposals Due: **December 1, 2006**
- Regular Paper Submissions Due: **January 5, 2007**
- Notification of Acceptance: **March 19, 2007**
- Camera-Ready Papers Due: **April 16, 2007**

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First Call For Papers

Creating exact 3D moving images as ghost-like replicas of 3D objects has been an ultimate goal in video science. Capturing 3D scenery, processing the captured data for transmission, and displaying the result for 3D viewing are the main functional components. These components encompass a wide range of disciplines: imaging and computer graphics, signal processing, telecommunications, electronics, optics and physics are needed.

The objective of the **3DTV-Conference** is to bring together researchers and developers from academia and industry with diverse experience and activity in distinct, yet complementary, areas so that full scale 3D video capabilities are seemlessly integrated.

Topics of Interest

3D Capture and Processing
- 3D time-varying scene capture technology
- Multi-camera recording
- 3D photography algorithms
- Synchronization and calibration of camera arrays
- 3D view registration
- Multi-view geometry and calibration
- Holographic camera techniques
- 3D motion analysis and tracking
- Surface modeling for 3-D scenes
- Multi-view image and 3D data processing

3D Transmission
- Systems, architecture and transmission aspects of 3D
- 3D streaming
- Error-related issues and handling of 3d video
- Hologram compression
- Multi-view video coding
- 3D mesh compression
- Multiple description coding for 3D
- Signal processing for diffraction and holographic 3DTV

3D Visualization
- 3D mesh representation
- Texture and point representation
- Object-based representation and segmentation
- Volume representation
- 3D motion animation
- Dense stereo and 3D reconstruction
- Stereoscopic display techniques
- Holographic display technology
- Reduced parallax systems and integral imaging
- Underlying optics and VLSI technology
- Projection and display technology for 3D videos
- Human factors

3D Applications
- 3D imaging in virtual heritage and virtual archaeology
- 3D Teleimmersion and remote collaboration
- Augmented reality and virtual environments
- 3D television, cinema, games and entertainment
- Medical and biomedical applications
- 3D Content-based retrieval and recognition
- 3D Watermarking

Paper Submission

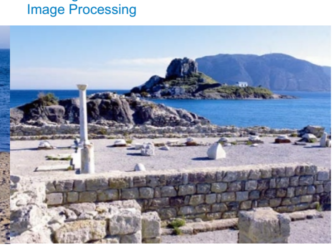
Prospective contributors are invited to submit full papers electronically using the on-line submission interface, following the instructions available at http://www.3dtv-con.org. Papers should be in Adobe PDF format, written in English, with no more than four pages including figures, using a font size of 11. Conference proceedings will be published online by the IEEE.

Important Dates

1 December 2006	Special sessions and tutorials proposals
15 December 2006	Regular Paper submission
9 February 2007	Notification of acceptance
2 March 2007	Submission of camera-ready papers

3DTV NoE — ITI-CERTH — Information Society Technologies — Institute of Electrical and Electronics Engineers — European Association for Signal and Image Processing — MPEG-IF





The International ITG / IEEE Workshop on Smart Antennas WSA 2007 February 26-27, 2007 Vienna

Call for Papers

The International ITG / IEEE Workshop on Smart Antennas WSA 2007 provides a forum for presentation of the most recent research on smart antennas. The objective is to continue, accelerate, and broaden the momentum already gained with a series of ITG Workshops held since 1996: Munich and Zurich'96, Vienna and Kaiserslautern'97, Karlsruhe' 98, Stuttgart'99, Ilmenau'01, Munich'04, Duisburg'05, and Ulm'06. This call for papers intends to solicit contribu-tions on latest research of this key technology for wireless communication systems.

Workshop topics include, but are not limited to:

- Antennas for beamforming and diversity
- Channel measurements
- Spatial channel modeling
- Beamforming
- Diversity concepts
- Space-time processing
- Space-time codes
- MIMO Systems
- Multicarrier MIMO
- Multiuser MIMO
- Cooperative and sensor networks
- Crosslayer optimisation
- Radio resource management
- Cellular systems
- Link, system and network level simulations
- Hard- and software implementation issues

There will be oral as well as poster presentations.
The workshop will be jointly organized by the Institute of Communications and Radio Frequency at Vienna University of Technology and the ftw. Telecommunications Research Center Vienna in cooperation with the VDE, ÖVE, and the IEEE on February 26-27, 2007 in Vienna, Austria

Organizers and Workshop Chairs

Markus Rupp,
E-Mail: mrupp@nt.tuwien.ac.at
Christoph Mecklenbräuker,
E-Mail: cfm@ftw.at
Information about the workshop can soon be found at: http://www.ftw.at/

Technical program committee

Jørgen Bach-Andersen	David Gesbert	Michael Meurer	Werner Teich
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Alex Gershman	Tadashi Matsumoto	Giorgio Taricco	

5th International Symposium on

Image and Signal Processing and Analysis ISPA 2007

September 27-29, 2007, Istanbul, Turkey

http://www.isispa.org

Call for Papers

The 5th International Symposium on Image and Signal Processing and Analysis (ISPA 2007) will take place in Istanbul, Turkey, from September 27-29, 2007. The scientific program of the symposium consists of invited lectures, regular papers, and posters. The aim of the symposium is to foster interaction of researchers and exchange of new ideas. Prospective authors are invited to submit their manuscripts reporting original work, as well as proposals for special sessions.

Co-Operations and Co-Sponsorships

- European Association for Signal Processing (EURASIP)
- IEEE Region 8*

Symposium Topics

A. Image and Video Processing
B. Image and Video Analysis
C. Image Formation and Reproduction
D. Signal Processing
E. Signal Analysis
F. Applications

For a detailed list of subtopics please visit ISPA 2007 web site.

Important Dates

Submission of full paper: February 15, 2007
Notification of acceptance/rejection: April 15, 2007
Submission of camera-ready papers and registration: May 15, 2007

Symposium Venue

Located in the center of the Old World, Istanbul is one of the world's great cities famous for its historical monuments and scenic beauties. It is the only city in the world which spreads over two continents: it lies at a point where Asia and Europe are separated by a narrow strait - the Bosphorus. Istanbul has been the cradle for many civilizations for over 2500 years and has a very rich history. It has been the capital of three great empires, the Roman, Byzantine and Ottoman empires, and for more than 1,600 years over 120 emperors and sultans ruled the world from here. Istanbul is the heart of Turkey with respect to entertainment, culture, education, shopping, imports and exports, tourism and the arts. The symposium will be organized in the congress center of the Bogazici University.

Paper Submission Procedure

Papers including title, author list and affiliations, figures, results, and references should not exceed six A4 pages. Detailed instructions for electronic submission are available on the ISPA web site. All papers will be subject to a peer-review process with at least two reviewers. All accepted papers will be published in the symposium proceedings in book form and on CD-ROM, which will be available through IEEE Publications Center and in IEEExplore digital library.

Call for Special Session Proposals

Prospective organizers of special sessions are invited to send proposals to Special Session Co-Chairs, according to instructions provided on the ISPA web site. The aim of a special session is to provide an overview of the state-of-the-art and current research directions in specific fields of image and signal processing and analysis.

Best Student Paper Award

Best Student Paper Award in the amount of 300 EUR will be given to a student author. The student's name must appear first on the paper and the paper must be presented at the symposium to be eligible for the award.

* request pending

Organizing Institutions
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Sabanci University, Turkey
University of Zagreb, Croatia

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Aytul Ercil
aytulercil@sabanciuniv.edu
Sabanci University, Turkey

Sven Loncaric
sven.loncaric@fer.hr
University of Zagreb, Croatia

Program Co-Chairs
Maria Petrou
maria.petrou@imperial.ac.uk
Imperial College London, UK

Tapio Saramaki
tapio.saramaki@tut.fi
Tampere Univ. of Tech., Finland

Special Sessions Co-Chairs
Bulent Sankur
bulent.sankur@boun.edu.tr
Bogazici University, Turkey

Robert Bregovic
bregovic@cs.tut.fi
Tampere Univ. of Tech., Finland

Publicity Chair
Enis Cetin
cetin@bilkent.edu.tr
Bilkent University, Turkey

Local Arrangements Chair
Aysın Ertuzun
email address
ertuz@boun.edu.tr
Bogazici University

Publications Chair
Marko Subasic
marko.subasic@fer.hr
University of Zagreb, Croatia

Registration Chair
Gulbin Akgun
gulbin@sabanciuniv.edu
Sabancı University, Turkey

Information Tech. Chair
Vjekoslav Levacic
vjekoslav.levacic@fer.hr
University of Zagreb, Croatia

Symposium Secretariat
Gulbin Akgun
Sabanci University, Faculty of Engineering and Natural Sciences
Tel: 216 483 9543
Fax: 216 483 9550
secretariat@isispa.org

ISSPA 2007

International Symposium on Signal Processing and its Applications

in conjunction **with the International Conference on Information Sciences, Signal Processing and their Applications**

12 – 15 February 2007, Millennium Hotel, Sharjah, U.A.E.

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University of Sharjah, UAE

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University of Sharjah, UAE

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University of Sharjah, UAE
H. A. Al-Hammady
Etisalat University College, UAE

Web and IT
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DSTO, Australia
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Tokyo Institute of Technology, Japan
M. Gabbouj, Europe
Tampere University of Technology, Finland
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ENIT, Tunisia
Y. Zhang, America
Villanova University, USA

Call For Participation

ISSPA 2007 marks the 20th anniversary of launching the first ISSPA in 1987 in Brisbane, Australia. Since its inception, ISSPA has provided, through a series of 8 symposia, a high quality forum for engineers and scientists engaged in research and development of Signal and Image Processing theory and applications. Effective 2007, ISSPA will extend its scope to add the new track of information sciences. Hence, the intention that the previous full name of ISSPA is replaced after 2007 by the following new full name:
International Conference on Information Sciences, Signal Processing and their Applications. ISSPA is an IEEE indexed conference.

ISSPA 2007 is organized by the University of Sharjah, College of Engineering, Etisalat University College and the American University of Sharjah.

The regular technical program will run for three days along with an exhibition of signal processing and information sciences products. In addition, tutorial sessions will be held on the first day of the symposium. Presentations will be given in the following topics:

1. Filter Design Theory and Methods
2. Multirate Filtering & Wavelets
3. Adaptive Signal Processing
4. Time-Frequency/Time-Scale Analysis
5. Statistical Signal & Array Processing
6. Radar & Sonar Processing
7. Speech Processing & Recognition
8. Fractals and Chaos Signal Processing
9. Signal Processing in Communications
10. Signal processing in Networking
11. Multimedia Signal Processing
12. Nonlinear signal processing
13. Biomedical Signal and Image Processing
14. Image and Video Processing
15. Image Segmentation and Scene Analysis
16. VLSI for Signal and Image Processing
17. Cryptology, Steganography, and Digital Watermarking
18. Image indexing & retrieval
19. Soft Computing & Pattern Recognition
20. Natural Language Processing
21. Signal Processing for Bioinformatics
22. Signal Processing for Geoinformatics
23. Biometric Systems and Security
24. Machine Vision
25. Data visualization
26. Data mining
27. Sensor Networks and Sensor Fusion
28. Signal Processing and Information Sciences Education
29. Others
30. Special Sessions

Prospective authors were invited to submit ***full length*** *(four pages) papers* ***via the conference website*** *for presentation in any of the areas listed above (showing area in submission). Submission of proposals for student session, tutorials and sessions on special topics were sent to the conference secretary. All articles submitted to ISSPA 2007 are* ***peer-reviewed*** *using a* ***blind review process*** *by at least two independent reviewers.*

For more details see

www.isspa2007.org/

Important Deadlines:

Full Paper Submission:
October 14, 2006
Tutorials/Special Sessions Submission:
October 14, 2006
Notification of Acceptance:
December 3, 2006
Final Accepted Paper Submission:
December 19, 2006

Conference Secretary

A-K. Hamid
University of Sharjah, UAE
Tel : +971 6 5050932
Fax :+971 6 5050872
E-mail: akhamid@sharjah.ac.ae