experiments, with clean and then coded speech (see Fig. 3). By using the five-dimensional pruned MBCM, ASV is able to catch 91.3% of impostors for clean speech and 88.2% for coded speech, on average.

Figs. 2 and 3 reveal a continuous mean increase in ASV accuracy with MBCM dimension and thus with code length. It is hoped that ASV will be further improved by issuing customers with longer personal codes (i.e. ten digits). Long term variation of the speech signal should also be catered for. In this respect, an updated enrolment signature could be computed from test and previous enrollment signatures each time a customer accesses a service. A more complete investigation needs to consider a customer base composed of female as well as male speakers. Whether a single MBCM may be trained to cater for both genders simultaneously warrants investigation. Another approach would rely on two MBCMs, one for each gender and on an automatic selection of the appropriate MBCM according to customer gender. The latter is possible with the use of accurate gender gates [8].

In summary, augmenting an SR security system with ASV is proposed. This second level of security is aimed at protecting access to commercial services from impostors who would know the names and associated codes of legitimate customers and thus be able to defeat the SR system. When an identity is claimed, through the utterance of a code, the ASV system produces a signature which is then compared to a known signature for that identity. The distinguishing feature of such a system is its implementation, which is independent of customer base size. Consequently, its commercial exploitation should be much more straightforward than those of previously proposed ASV systems.

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Vector quantisation with MAP estimator using reliability information from turbo-codes

Keang-Po Ho

Minimum mean-squared distortion can be achieved by a maximum *a posteriori* probability (MAP) estimator utilising the reliability information available from the turbo-code channel decoder. Using the optimum MAP estimator in vector quantisation, numerical results show that MAP detection provides a significant improvement over the conventional mapping-based hard-decision source decoder.

When turbo-codes are decoded, reliability information is provided as the likelihood of the transmitted binary data [1]. Taking advantage of the reliability information (or, in general, any decoder with soft output), the performance of vector quantisation transmitted through a noisy channel can be greatly improved. Combined with the turbo-code structure, a maximum *a posteriori* probability (MAP) estimator can be used to provide a minimum mean-squared distortion estimation of the original transmitted analogue vector. Instead of decoding the binary data, the log-likelihood ratio (LLR) from turbo-codes is used directly in the MAP estimator. The main contribution of this Letter is to derive the optimum estimator-based decoder when LLR from turbo-codes is available.

Fig. 1 shows a schematic diagram of a communication system to transmit vector quantised analogue signals. In the system, the encoder inputs an analogue continuous sample $\mathbf{x}$, quantises it using a quantiser Q, and then maps the results to a symbol selected from binary codewords $\{b_1, ..., b_N\}$ with length $K = \log_2 N$. The binary codewords are concatenated, encoded by the turbo-code encoder, and sent through the noisy channel. The system in Fig. 1*a* uses conventional mapping-based hard-decision decoder. The turbo-code channel decoder outputs the binary codeword, which is subsequently mapped to the reconstruction levels of the vector quantiser. Using the LLR of the turbo-decoder, the system in Fig. 1*b* uses a MAP estimator to provide an estimation of the original transmitted analogue signal to minimise the mean-squared distortion.

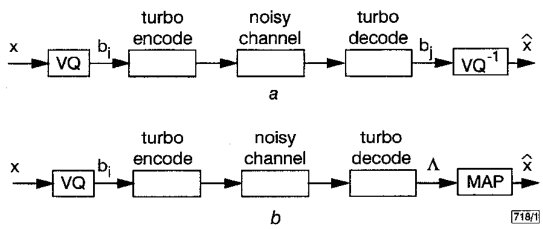

Fig. 1 *Schematic diagram of communication systems for vector quantiser using turbo-codes*

a With conventional hard-decisoin mapping-based decoder
b With MAP estimator-based soft-decision decoder

The turbo-decoder provides the LLR Λ of each bit, according to [1]

$$\Lambda = \ln\left[\frac{\Pr(u_k = 1|\text{channel output})}{\Pr(u_k = -1|\text{channel output})}\right] \qquad (1)$$

where u_k is the transmitted bit. Conventionally, the decoder of Fig. 1*a* makes a hard decision on the LLR to estimate the corresponding bit which is the sign of the LLR, i.e. 1 if LLR > 0 and –1 otherwise. The LLR can be used to calculate the bit *a posteriori* probability according to

$$P(u_k = \pm 1|\Lambda) = \frac{\exp(\pm\Lambda)}{1+\exp(\pm\Lambda)} \qquad (2)$$

The LLR *a posteriori* probability is $P(\Lambda|u_k) = P(u_k|\Lambda)p(\Lambda)/P(u_k)$ from the Bayes rule. Usually, $P(u_k = \pm 1) = 1/2$ (in the later numerical example, $P(u_k = 1) = 0.4952$) and $p(\Lambda)$ are the same for both $u_k = \pm 1$. Ignoring all common factors, we may write $P(\Lambda|u_k) = P(u_k|\Lambda)$. The LLR *a posteriori* probability within a binary codeword can be used to calculate a set of LLR *a posteriori* probabilities $\{P_1, ..., P_N\}$, where $P_i = P(\vec{\Lambda}|b_i)$ is the LLR a posteriori probability given the binary codewords b_i, and $\vec{\Lambda} = (\Lambda_1, ..., \Lambda_K)$ is the LLR within a binary codeword. Using these *a posteriori* probabilities, the minimum squared error estimator for the original input signal is the MAP estimator [2, 3]:

$$\hat{x}(\vec{\Lambda}) = \sum_{i=1}^{N} E\{\mathbf{x}|b_i\}p(b_i|\vec{\Lambda}) = \sum_{i=1}^{N} \mathbf{c}_i p(b_i|\vec{\Lambda}) \qquad (3)$$

where $\mathbf{c}_i = E\{\mathbf{x}|b_i\}$, $i = 1, ..., N$ is the codeword or centroid of each vector quantisation partition. From the Bayes rule, $p(b_i|\vec{\Lambda}) = p(\vec{\Lambda}|b_i)p_{b_i}/p(\vec{\Lambda})$, the soft decoder is

$$\hat{x}(\vec{\Lambda}) = \frac{\sum_{i=1}^{N} P_i p_{b_i} \mathbf{c}_i}{\sum_{i=1}^{N} P_i p_{b_i}} \qquad (4)$$

where p_{b_i} is the *a priori* probability of the binary codewords b_i. In Bakus and Khandani [4], the output vector is estimated by $\hat{x}(\overline{\Lambda}) = \Sigma_{i=1}^{N} P_i\mathbf{c}_i$ for a scalar quantiser with the assumption that all binary codewords have the same *a priori* probability. In general, the system distortion using this approximation [4] is very close to the optimum result. However, in some special cases with huge differences in *a priori* probabilities, the approximation is much worse than the optimum decoder (eqn. 4). Furthermore, this Letter shows the result for the more powerful vector quantiser instead of for scalar quantisers [4].

The proposed LLR-based MAP decoder is tested for vector quantisation. The source is a first-order Gauss-Markov source with a correlation coefficient of 0.9. The vector quantiser is designed using the LBG algorithm [5] for a dimension of eight with a rate of 1 bit/sample. The 16-state rate-1/2 turbo-codes are the same as that in [1] with a block length of 256×256 and generator of $G_1 = 37$ and $G_2 = 21$. To limit the computation complexity, the number of iterations is limited to six. The error probability as a function of the channel signal-to-noise ratio (SNR) in our simulation is the same at that in [1].

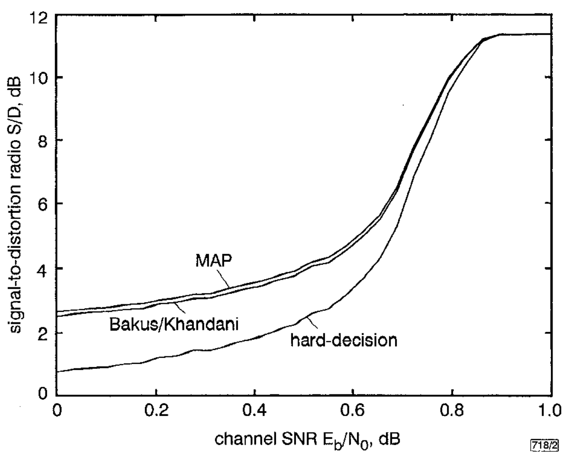

Fig. 2 *Signal-to-distortion ratio against channel signal-to-noise ratio E_b/N_0 for eight-dimensional vector quantiser decoded by hard-decision mapping, MAP estimator, and Bakus/Khandani decoder* [4]

Fig. 2 shows the signal-to-distortion ratio (S/D) against the channel SNR per bit, E_b/N_0. The performances provided by the MAP estimator, the Bakus/Khandani decoder [4], and the hard-decision mapping-based decoder are shown for comparison. The MAP estimator for minimum mean-squared distortion provides the smallest distortion and the largest S/D. At a low channel SNR, the improvement over the conventional hard-decision mapping based decoder is ~2dB. The MAP estimator performs ~0.10–0.16dB better than the Bakus/Khandani decoder [4]. While the improvement of the MAP estimator over the Bakus/Khandani decoder [4] is small, it is confirmed that the Bakus/Khandani decoder [4] is a sub-optimal decoder.

In conclusion, the optimum MAP estimator based on the reliability information provided by a turbo-code channel decoder is derived to minimise the mean-squared distortion. For an eight-dimensional vector quantiser at a rate of 1 bit/sample, the MAP estimator can achieve an ~2dB performance improvement over the conventional mapping-based decoder, and performs better than another, previously published soft-decision decoder [4].

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Virtual circuit connection method for RSVP multicasting supporting heterogeneous receivers on the ATM network

Dongwook Lee and Kiseon Kim

The authors suggest a novel virtual circuit connection method based on the reverse traversing technique to minimise the waste of network bandwidth resources, when the internet protocol multicast is interoperated using the resource reservation protocol over an asynchronous transfer mode network. Simulation results show that, as the number of receivers increases, the bandwidth requirements on all links of the network of the proposed scheme become more advantageous than those of other conventional methods.

Resource reservation protocol over asynchronous transfer mode: In recent years, the use of internet applications such as audio, video, and video conferencing as well as text-based services has grown enormously [1]. To support applications requiring real-time performance guarantees, the internet engineering task force (IETF) has developed the resource reservation protocol (RSVP). The RSVP allows users to communicate with the networks for their quality of service (QoS) requirements, as shown in Fig. 1 [2], where an example of RSVP message flows is illustrated. There are one sender S1, two end-receivers R1 and R2 and three routers in the network. When a receiver originates a reservation request, it can also request a confirmation message to indicate that its request was installed in the network. A successful reservation request propagates upstream along the multicast tree in which one path can be R1, SW2, SW1 and S1 until it reaches a point where an existing reservation is equal to or greater than that being requested. At point SW1, the arriving request is merged with the reservation in place and need not be forwarded. At each merge point, only the largest request and any accompanying confirmation-request objects are forwarded upstream.

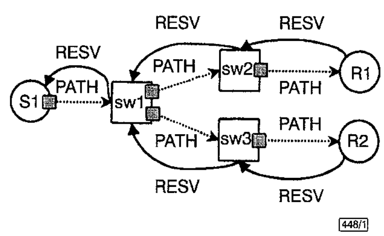

Fig. 1 *RSVP message flows*

——▸ reservation message
·········▸ path message
▣ reserved resources

ATM (asynchronous transfer mode) is realised as a platform to provide a high-speed data link layer service with a certain QoS and a unique user-to-network interface. ATM Forum UNI 3.x and 4.0 support the permanent virtual circuit (PVC) and switched virtual circuit (SVC). They also provide the point-to-point virtual circuit (VC) and the point-to-multipoint VC connection with various real-time services. By using these VCs, it is possible to interoperate the RSVP over ATM. The scheme of mapping the RSVP to ATM VCs is explained in detail in [3], where the conventional methods use either a point-to-multipoint VC or point-to-point VCs for each receiver to establish a VC for RSVP. The point-to-multipoint VC of ATM is a single VC and all receivers receive data with same QoS. Therefore, this is suitable when all receivers request data with the same QoS. When the internet protocol (IP)