removed speech features for recognition. It is worth noting that the bias estimation process of the proposed method is non-iterative, so it is computationally efficient.

Experimental results: The effectiveness of the proposed orthogonal transform-based SBR (OTSBR) method was examined by simulations using a multi-speaker continuous Mandarin speech recognition task. The database was generated by ten speakers including eight males and two females. It contained, in total, 3050 utterances including 2572 training utterances and 478 testing utterances. Each utterance comprised several syllables and was uttered in such a way that every syllable was clearly pronounced. All speech signals were digitally recorded into a PC with a SoundBlaster card through a microphone and sampled at 16kHz. An adverse testing speech database was constructed artificially by passing each utterance of the clean-speech testing set through a filter which simulated a telephone channel. A set of 32 simulated filters generated from a large telephone-speech database was used in this study. All speech signals were divided into 20ms frames with 10ms frame shifts for feature extraction. A set of 25 features, including 12 MFCCs, 12 delta MFCCs, and a delta log-energy was extracted for each frame. A sub-syllable-based hidden Markov model (HMM) recogniser was constructed from the clean-speech training set by the maximum likelihood training algorithm. It consists of 100 three-state right-final-dependent initial models, 39 five-state context-independent final models, and a single-state non-speech model. The baseline SBR method used three separate codebooks for the three feature sets containing 12 MFCCs, 12 delta MFCCs, and a delta log-energy, respectively [1]. For the proposed OTSBR method, the orthogonal transform coefficients of these 25 features were calculated for all utterances in the training set and used to create 25 codebooks.

Table 1: Performance of baseline SBR method

Codeword number	Bias deviation	Syllable accuracy	Relative bias estimation time
		%	
128	132.4	63.0(57.2)	0.5
256	114.6	64.6(58.7)	1.0
512	140.2	62.4(56.6)	2.0
1024	122.8	64.1(58.3)	4.0

Table 2: Performance of proposed OTSBR method

Window length/ window shift	Bias deviation	Syllable accuracy	Relative bias estimation time
		%	
4/1	46.4	70.1	0.21
4/3	46.3	70.3	0.08
6/1	46.0	70.0	0.21
6/3	45.8	69.9	0.08
8/1	46.7	70.1	0.21
8/3	46.0	70.4	0.08

Table 1 shows the performance of the baseline SBR method. The average bias deviation of the bias estimation is defined by

$$D_{bias} = \frac{1}{N_{utt}} \sum_{u=1}^{N_{utt}} \left(\sum_{k=1}^{24} \left(bias_{est}(u,k) - bias_{des}(u,k) \right)^2 \right) \quad (10)$$

where N_{utt} is the total number of utterances in the testing set, $bias_{est}(u, k)$ and $bias_{des}(u, k)$ are, respectively, the estimated and desired biases of utterance u and feature element k. Here $bias_{des}(u, k)$ was obtained by taking the average of the differences between the kth features of bias-corrupted speech and of clean speech of all frames in utterance u. It is noted that the calculation of D_{bias} only involves 12 MFCCs and 12 delta MFCCs. In Table 1, the numbers within the parentheses and outside the parentheses for syllable accuracy are the results of the first and tenth iterations, respectively. As to the bias deviation and the relative bias estimation time, only the results of the tenth iteration are shown in Table 1. The performance of the OTSBR method is shown in Table 2. Here, the bias estimation times are normalised to that of the baseline SBR method with 256 codewords. It can be seen from Table 2 that all cases using a different window length and window shift have comparable recognition performances. They are all much better than those achieved by using the baseline SBR method. They also all have smaller bias estimation times.

Conclusions: We have proposed a new SBR method using orthogonal transforms to improve the accuracy of bias estimation for adverse Mandarin speech recognition. Experimental results have confirmed that the proposed method outperformed the conventional SBR method significantly both in terms of the recognition performance and the computation speed.

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Threshold-type call admission control in wireless/mobile multimedia networks using prioritised adaptive framework

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Limitations to the bandwidth of wireless links has motivated the development of adaptive multimedia services where the bandwidth of a call can be dynamically adjusted. A threshold-type call admission control algorithm is proposed for quality of service provisioning; a nonlinear programming model is formulated for determining the optimal threshold values.

Introduction: Limitations to the bandwidth of wireless links has motivated the development of adaptive multimedia services which can operate over a wide range of available bandwidths [1]. That is, it is possible to overcome the link overload condition by reducing the bandwidth of individual calls. For example, handoff blocking due to bandwidth limitations can be avoided. A bandwidth adaptation algorithm (BAA) that reduces/expands the bandwidth of individual calls is invoked in the event of a new call arrival, a call completion, or an incoming/outgoing handoff call.

Under this adaptive framework, the quality of service (QoS) parameters are expressed in terms of the call blocking probability and the call degradation probability. The call degradation probability is the probability that a call will be allocated less than its maximum bandwidth at a given time. Call admission control (CAC) is required to satisfy the above QoS parameters. Recently, prioritisation (or 'differentiation') in the Internet has become of extreme importance. We believe that this concept will be reflected in wireless/mobile networks in the near future. Thus, we take prioritisation into consideration in our adaptive multimedia framework.

Model description: Suppose that there are K classes of adaptive multimedia services. Their priority levels are ordered in a decreasing manner; i.e. calls belonging to class i have higher priority than calls belonging to class j if $i < j$. That is, the lower is the priority of a class, the more preferable it becomes to adapt the bandwidth of calls of that class. The bandwidth of a class i call takes its discrete value from the set B_i. We adopt the BAA proposed in [2] where the bandwidth of calls with lower priority is always preferably reduced/increased. (See [2] for details).

New class i (i = 1, 2, ..., K) call arrivals into a cell are Poissonian with mean rate λ_i. Handoff call arrivals of class i are also assumed to be Poissonian with mean rate $1/h_i$. The call holding time (CHT) of a class i call is assumed to follow an exponential distribution with mean $1/\mu_i$. The cell residence time (CRT), i.e. the amount of time during which a mobile terminal stays in a cell before handoff, is assumed to follow an exponential distribution with mean $1/h$. We assume that the CRT is independent of class. The total bandwidth capacity in each cell is the same and is denoted by C.

Call admission control: In a threshold-type CAC algorithm, a newly arriving call of class i is blocked if the current number of ongoing class i calls is equal to or greater than t_i. An incoming handoff call of class i is accepted regardless of the number of ongoing class i calls; if the available bandwidth is insufficient, bandwidth adaptation is performed to accommodate the call. The threshold CAC algorithm is chosen because the steady state probability is easily tractable via the product form [3]. The steady probability of state $\mathbf{x} = (x_1, x_2, ..., x_K)$, where x_i denotes the number of ongoing class i calls in the cell, is determined by choosing a threshold value for each class i, namely t_i. Ultimately, the t_i should be determined for each class for which the revenue is maximised. Here, for each on-going class i call, revenue is assumed to be accrued at rate r_i regardless of the currently allocated bandwidth.

By means of the product form solution, $\mathbf{x}$ is given by

$$\pi(\mathbf{x}) = \frac{1}{G}\prod_{i=1}^{K} p_i(x_i) \tag{1}$$

$$G = \sum_{\forall \mathbf{x}}\prod_{i=1}^{K} p_i(x_i) \tag{2}$$

$$p_i(x_i) = I(x_i \le t_i)\left(\frac{\lambda_i + h_i}{\mu_i + h}\right)^{x_i}/x_i! + I(x_i > t_i)\left(\frac{\lambda_i + h_i}{\mu_i + h}\right)^{t_i}\left(\frac{h_i}{\mu_i + h}\right)^{x_i - t_i}/x_i! \tag{3}$$

where G is the normalisation constant, and $I(\cdot)$ is an indicator function which is 1 if its argument is true and 0 otherwise. Then, our CAC algorithm can be formulated as a nonlinear programming (NLP) problem as shown below. Finally, the optimal values of decision variables t_1, t_2, ..., t_K are determined by solving the NLP problem:

Maximise

$$\sum_{\forall \mathbf{x}}(\mathbf{r}\cdot\mathbf{x})\pi(\mathbf{x})$$

subject to

$$\sum_{\forall \mathbf{x}, x_i > 0}\pi(\mathbf{x})\frac{d_i(\mathbf{x})}{x_i} \le P_{D_i}$$

and

$$\sum_{\forall x_i \ge t_i}\pi(\mathbf{x}) \le P_{B_i}$$

where $d_i(\mathbf{x})$ is the number of class i calls for which the bandwidth is allocated to be less than its maximum bandwidth in $\mathbf{x}$, which is determined by the BAA. Let P_{Di} and P_{Bi} denote the required upper bounds of the call degradation probability and the call blocking probability of a class i call, respectively. In addition, $\mathbf{r} = (r_1, r_2, ..., r_K)$ is the revenue vector. Note that the call blocking probability applies only to new calls. An incoming handoff call is almost always accepted; if the available bandwidth is insufficient, the BAA is initiated to make room for the handoff call. (We implicitly assume that the case in which a handoff call should be blocked will not happen.)

To solve the above NLP problem, we adopt the 'separable programming' technique, which investigates every possible case of threshold value for all classes. As a result, all the nonlinear terms in the above NLP problem are changed into constants, thereby transforming the NLP problem into a linear programming (LP) problem.

Let $\mathbf{t} = (t_1, t_2, ..., t_K)$ be a vector of threshold values. By introducing Boolean variables $A_{(0,0,...,0)}$, $A_{(1,0,...,0)}$, ..., $A_{(max(t_1),max(t_2),...,max(t_K))}$ instead of $\mathbf{t}$, the nonlinear expressions (eqns. 1 – 3) are changed into the linear expressions below. Here, $max(t_i)$ means the maximum possible integer value of threshold of class i. Note that only one of the above Boolean variables is 1, while the others are 0; for example, in three classes, $A_{2,3,5} = 1$ means that $\mathbf{t}$ = (2, 3, 5) is the threshold vector for the optimal solution.

$$\sum_{\forall \mathbf{t}} A_{\mathbf{t}} = 1 \tag{4}$$

$$\pi(\mathbf{x}) = \sum_{\forall \mathbf{t}} A_{\mathbf{t}}\left[\frac{1}{G(\mathbf{t})}\prod_{i=1}^{K} p_i(x_i, \mathbf{t})\right] \tag{5}$$

$$G(\mathbf{t}) = \sum_{\forall \mathbf{x}}\prod_{i=1}^{K} p_i(x_i, \mathbf{t}) \tag{6}$$

$$p_i(x_i, \mathbf{t}) = I(x_i \le t_i)\left(\frac{\lambda_i + h_i}{\mu_i + h}\right)^{x_i}/x_i! + I(x_i > t_i)\left(\frac{\lambda_i + h_i}{\mu_i + h}\right)^{t_i}\left(\frac{h_i}{\mu_i + h}\right)^{x_i - t_i}/x_i! \tag{7}$$

Finally, the threshold values for the maximum revenue can be found by solving the above LP problem.

Numerical results: We simulated the environment where $K = 3$ and C = 3Mbit/s. Table 1 shows the bandwidth values (in Kbit/s) as in [1] and the revenue rates. The upper bounds of the call blocking probabilities were set to 0.5 for all classes. The upper bounds of the call degradation probability of class 0,1, 2 were set to 0.1, 0.2, and 0.3, respectively. Note that class 0 had the highest priority. Here the new call arrival rate of each class was 0.3, while the mean of handoff call arrival rate was set to be half of the mean of the new call arrival rate. In addition, the mean values of the CHT and the CRT were assumed to be 500 and 100 s, respectively.

Table 1: Bandwidth values and revenue rates

Service	B_i	r_i
Class 0	{64, 80, 96,112, 128}	1
Class 1	{128, 160, 192, 224, 256}	3
Class 2	{256, 288, 320, 352, 384}	8

Table 2: Revenues of threshold sets

(t_1, t_2, t_3)	Revenue
(7, 5, 14)	43.929670
(7, 6, 13)	43.509179
(7, 6, 14)	44.633431
(7, 7, 13)	43.949723
(8, 5, 13)	43.349376
(8, 5, 14)	44.473667
(8, 6, 13)	44.053130
(8, 6, 14)*	45.177361
(8, 7, 13)	44.493615

Table 2 shows the revenues of t_i tuples which satisfy the upper bounds of both the call blocking probability and the call degradation probability around the optimal solution. Note that the tuple (8, 6, 14) marked by * is the solution of our LP model, which verifies that we have found the optimal threshold values that maximise revenue.

Conclusion: Adaptive multimedia services will become pervasive in future wireless/mobile networks, especially considering the highly fluctuating link bandwidth availability. In an adaptive multimedia framework, the bandwidth of a call dynamically takes a value from a set of discrete values, depending on situations. Prioritisation among multiple classes of service is also taken into account such that the bandwidth values of lower priority calls are preferably adapted in overload conditions.

A threshold-type call admission control algorithm has been proposed. A nonlinear programming (NLP) model has been formulated to determine threshold values for obtaining the optimal solution. The NLP model is then transformed into a linear programming model via separable programming. Numerical results verify that the result of the proposed method achieves maximum revenue while satisfying the quality of service requirements.

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Simplified ultrasonic regular-sampled PWM technique

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The well-established regular-sampled pulse width modulation technique is simplified to produce a strategy with reduced computational complexity, but without leading to a significant deterioration in performance. The technique is particularly suitable for applications requiring ultrasonic carrier frequencies to reduce harmonic distortion and for those closed-loop feedback drive systems that are limited by sampling rates and control algorithm execution times.

Introduction: Advances in fast-switching power devices offer the possibility of increasing the range of switching frequencies into the ultrasonic range [1, 2]. These developments have resulted in reduced harmonic distortion due to the high switching frequencies and have led to the possibility of using simpler pulsewidth modulation (PWM) strategies. The regular-sampled pulse width modulation (RS PWM) technique is well documented and is widely used for the control of both single and three phase inverters [3, 4]. The technique is defined by simple algebraic equations and may be realised using both low-cost hardware and software implementations.

The proposed new modulation technique extends the principle of using a single sample to produce more than one switching pulse. Simulation and experimental results show that excellent performance is achievable using the resulting simplified RS PWM strategy.

Three-level RS PWM: Fig. 1 shows the production of three-level PWM, where switching takes place between three voltage levels (–1, 0 and 1, say) [5]. As shown in Fig. 1, the first sample of the modulating function is taken at time T_k and is used to modulate the leading edge of the PWM pulse. The second sample, taken at time T_{k+1} is used to modulate the trailing edge. The leading and trailing edge times, δ_k and δ_{k+1} respectively, are calculated from eqns. 1 and 2.

$$\delta_k = \frac{T}{2}[1 - M \sin \omega T_k] \tag{1}$$

and

$$\delta_{k+1} = \frac{T}{2} M \sin \omega T_{k+1} \tag{2}$$

where T is the sample period, T_k is the kth sample and M is the modulation index.

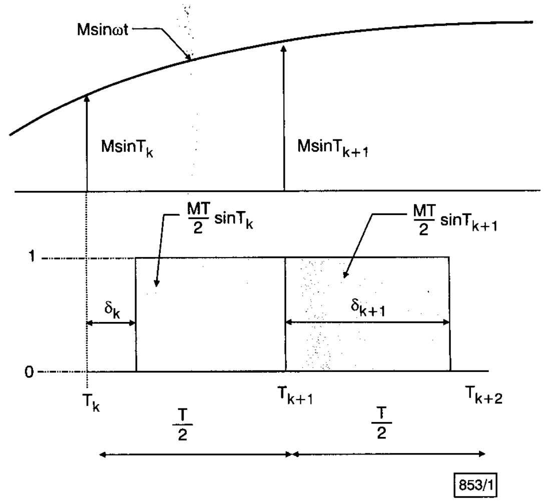

Fig. 1 *Production of one pulse for three-level asymmetric regular-sampled PWM*

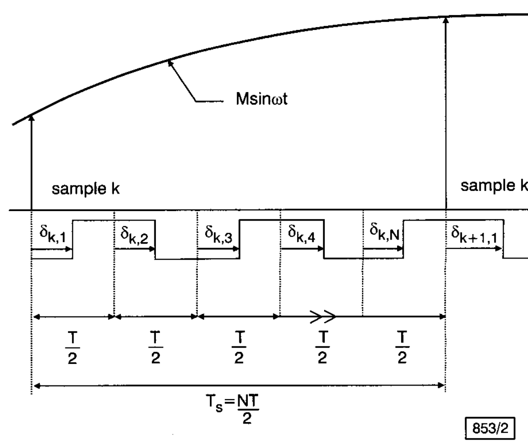

Fig. 2 *Definition of switching angles and pulses for simplified RS PWM technique*

Simplified RS PWM: Fig. 2 shows how PWM is produced by regularly sampling the modulating wave every N half carrier periods. Sample k is taken at time T_k and, in contrast to the conventional RS PWM technique where the next sample is taken after (at most) $N = 2$ half carrier periods, sample $k+1$ is taken only after N half carrier periods, where $N > 2$, have elapsed. Sample k is then used to calculate switching pulse time $\delta_{k,j}$ and is further used to define all the pulse times ($\delta_{k,1}$ to $\delta_{k,N}$) over the N half sample periods, such that $\delta_{k,1} = \delta_{k,3} = \delta_{k,5}$ etc. and $\delta_{k,2} = \delta_{k,4}$ etc. Consequently, the number of calculations required to produce the PWM is reduced by a factor of $1/N$ when compared to the standard RS PWM method.

Modification of eqns. 1 and 2 results in the general equation (eqn. 3) for the pulse angle

$$\delta_{k,j} = \frac{T}{2}\left[\frac{1}{2} + (-1)^k\left\{\frac{1}{2} - (-1)^j M \sin \omega T_k\right\}\right] \tag{3}$$

and eqn. 4 for the switching angle $\alpha_{k,j}$ for three-level RS PWM