baseline. This indicates that the EPA is a promising approach for robust speaker recognition under the mismatch handset condition.

Table 1: Speaker recognition rates (%) on ten different handsets in HTIMIT database achieved by GMM/CMS, GMM/CMS + pitch and EPA + GMM/CMS + pitch methods in different dimensional eigen-prosody space, respectively

	Dimension	senh	cb1	cb2	cb3	cb4	el1	el2	el3	el4	pt1	Average
GMM/CMS	—	77.7	76.0	73.7	26.3	39.6	72.8	64.2	65.3	70.2	61.6	62.7
GMM/CMS + pitch	—	78.6	77.2	76.0	28.3	42.5	74.9	65.9	69.4	72.5	65.3	65.1
EPA + GMM/CMS + pitch	50	95.1	82.7	82.7	35.8	49.7	82.4	74.0	72.3	76.9	69.1	72.0
	100	95.4	84.1	82.1	41.0	53.2	81.5	76.0	76.6	76.9	72.0	73.9
	200	98.3	84.7	85.3	39.9	52.9	83.8	77.2	76.9	76.3	72.3	74.7

Conclusions: We have presented a novel eigen-prosody analysis approach to capture the long-term prosodic information for robust speaker recognition under a mismatch environment. Unlike the N-gram and DHMM-based methods, which require a lot of speech data to reach a reasonable performance, the EPA requires only a few training/test utterances. Experimental results have shown that it is a promising method to alleviate the problem of handset mismatch. Furthermore, EPA can be applied against channel mismatch.

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Integrated rounding method for real number bit distribution over DMT systems

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Most bit loading algorithms proposed in the literature originally distribute a non-integer number of bits to the subcarriers of a discrete multitone (DMT) system and then, employing an iterative algorithm they round these assigned numbers to integers. Introduced is the concept of 'the quality factors' of the subcarriers and a new rounding algorithm is proposed based on them. Simulations comparing this proposal and the rounding proposed in the literature reveal that the proposed scheme results in power savings up to 11.1%, without affecting negatively the bit error rate.

Introduction: In [1] the analytical equation that maximises the overall data rate R_T for a predetermined bit error rate (BER) and power budget P_T is extracted, whereas in [2] the analytical equation that minimises the BER for a given P_T and R_T is given. Then in both cases the real number bit distribution R_i of the ith subcarrier is rounded to the integer R_i' that is nearest to R_i. Finally, if the predetermined criteria P_T and/or R_T are not fulfilled, an iterative procedure follows that adds or subtracts bits from the subcarriers according to the difference $\Delta R_i = R_i - R_i'$. This last step results in an integer number bit distribution R_{qi} that has slightly diverted the system from the optimal (but practically unattainable) real number bit distribution, leaving consequently, room for improvement. In this Letter we propose a new rounding method for a real number bit distribution over the subchannels of a discrete multitone (DMT) system, introducing the concept of 'the quality factors'.

Existing rounding and proposed solution: Assume a DMT system consisting of N subcarriers. Every subchannel i has its own gain α_i and additive Gaussian noise with variance n_i. Assume also that the analytical equations in [1] or [2] distribute R_i bits per subcarrier. A possible rounding of R_i to $\lceil R_i \rceil$ introduces a power increase $\Delta P_{u,i}$ and a rate increase $\Delta R_{u,i}$, whereas rounding R_i to $\lfloor R_i \rfloor$ introduces a power decrease $\Delta P_{d,i}$ and a rate reduction $\Delta R_{d,i}$, where $\lceil x \rceil$ is the smallest integer that is greater than x and $\lfloor x \rfloor$ is the greatest integer that is smaller than x. The existing rounding method considers only the ΔR_i parameter and not ΔP_i although variations of R_i do not lead to proportional variations of P_i (according to (7) the relationship between them is of logarithmic nature). In our proposal, the criterion for rounding is a 'quality factor' assigned to every subcarrier i. Let $Q_{u,i}$ be that factor defined as:

$$Q_{u,i} = \frac{\Delta P_{u,i}}{\Delta R_{u,i}} \quad (1)$$

$Q_{u,i}$ expresses the extra power consumption per bit increase if R_i is rounded to $\lceil R_i \rceil$. Those subcarriers that have the smallest $Q_{u,i}$ quality factors add the lowest possible power per bit increase and are candidates to be rounded to $\lceil R_i \rceil$, whereas those with the highest $Q_{u,i}$ should be rounded to $\lfloor R_i \rfloor$. The quality factors $Q_{u,i}$ are calculated using (2) (see 'Appendix'):

$$Q_{u,i} = \frac{[Q^{-1}(\text{BER}/4)]^2 (2^{\lfloor R_i \rfloor + 1} - 2^{R_i}) n_i}{3\alpha_i (\lfloor R_i \rfloor + 1 - R_i)}, \quad i = 1, \ldots, N \quad (2)$$

Out of the N subcarriers we have to locate M subchannels with the lowest $Q_{u,i}$. Those will add power and bits to the total power budget and the total number of bits, respectively. The rest of the N–M subcarriers will subtract bits and power and the constraints P_T and/or R_T for a given BER and a given channel will be fulfilled. Thus, $Q_{u,i}$ are sorted in ascending order, forming the vector **y**. The boundary M between the two groups is estimated using the bisection method. The bisection method stops when M satisfies the inequalities:

$$\sum_{k=1}^{M} \left| \Delta P_{u,i}^{(k)} \right| < \sum_{k=M+1}^{N} \left| \Delta P_{d,i}^{(k)} \right| \quad (3)$$

$$\sum_{k=1}^{M+1} \left| \Delta P_{u,i}^{(k)} \right| > \sum_{k=M+2}^{N} \left| \Delta P_{d,i}^{(k)} \right| \quad (4)$$

where index k in the summation is the ranking position of subcarrier i in the sorted vector **y**. If (3) and (4) are satisfied, then the total power transmission is as close as possible but less than P_T. The algorithm's steps are:

1. Calculate $Q_{u,i}$ using (2) and form the sorted vector **y**.
2. Let $d = 1$, $u = N$ and $M = \lfloor (u+d)/2 \rfloor$.
3. Calculate the two parts of the inequalities (3) and (4).
4. If only (3) is satisfied then $d = M$, $M = \lfloor (u+d)/2 \rfloor$. Go to 3.
Else if only (4) is satisfied then $u = M$, $M = \lfloor (u+d)/2 \rfloor$. Go to 3.
Else STOP.
5. $R_{qi} = R_i + |\Delta R_{u,i}^{(k)}|$ for $k = 1, \ldots, M$ and $R_{qi} = R_i - |\Delta R_{d,i}^{(k)}|$ for $k = M+1, \ldots, N$.

Obviously when the initial conditions include a target bit rate that must be fulfilled, then (3) and (4) are turned into (5) and (6) and the bisection method stops when the determined M satisfies both of them.

$$\sum_{k=1}^{M} \left| \Delta R_{u,i}^{(k)} \right| > \sum_{k=M+1}^{N} \left| \Delta R_{d,i}^{(k)} \right| \quad (5)$$

$$\sum_{k=1}^{M-1} \left| \Delta R_{u,i}^{(k)} \right| < \sum_{k=M}^{N} \left| \Delta R_{d,i}^{(k)} \right| \quad (6)$$

When both (5) and (6) are satisfied it means that the data rate is greater than R_T but as close as possible to it.

Simulation results: We have compared the algorithms in [1, 2] in terms of transmitting power per bit, using the existing rounding method and the proposed one. In this study we have used 256 subcarriers. The maximum bit rate per carrier was fixed to 6 bits. Fig. 1 presents the simulations for 200 up to 550 bits per DMT symbol. According to the ANSI T1.413 these data rates result in 800 Kbit/s up to 2.2 Mbit/s for Asymmetric Digital Subscriber Line (ADSL). Significant performance improvement is achieved when our system is loaded with more than 350 bits/symbol. For the maximum load, our proposal results in power savings up to 11.1%, whereas the power savings on average are 1.6–4.3%. In both cases the BER was constantly below 10^{-4}. Sorting the quality factors introduces $N \log_2 N$ iterations whereas around 10 iterations are needed on average for the bisection method convergence. Hence, the introduced complexity is $O(N \log_2 N + 10)$. According to [3] the existing method's complexity is $O(10N + 2N)$ on average. For $N = 256$ subcarriers the proposed rounding demands 2058 iterations whereas the existing demands 3072.

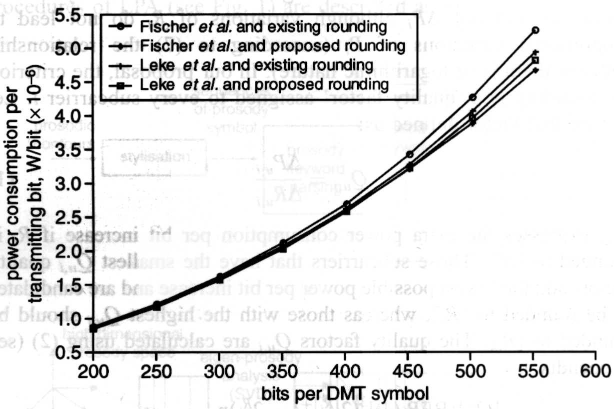

Fig. 1 *Transmitting power per bit for different bit loads. Bit loading algorithms [1, 2] and existing against proposed rounding*

Conclusion: In this Letter we propose a new rounding algorithm for real number bit distribution. Our proposal introduces less computational complexity compared with the existing methods. Moreover, it improves the overall performance of a DMT system in terms of total transmitting power especially for heavily loaded systems.

Appendix: A good upper bound for the bit error rate of a QAM system is given by

$$\mathrm{BER} = 4Q\left(\sqrt{\frac{3P_i\alpha_i}{(2^{R_i}-1)n_i}}\right)$$

where P_i is the average transmitting power per QAM symbol for subcarrier i [4]. Solving for P_i for a predetermined BER we have

$$P_i = [Q^{-1}(\mathrm{BER}/4)]^2(2^{R_i}-1)n_i/(3a_i) \tag{7}$$

Rounding to the smaller integer that is greater than R_i means that subcarrier i will carry $\lfloor R_i \rfloor + 1$ bits. So, from the definition (1) of the quality factor $Q_{u,i}$ we have:

$$Q_{u,i} = \frac{[Q^{-1}(\mathrm{BER}/4)]^2(2^{\lfloor R_i \rfloor+1} - 2^{R_i})n_i}{3\alpha_i(\lfloor R_i \rfloor + 1 - R_i)} \tag{8}$$

which is (2).

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