is a direct consequence of achieving required performance with lower CNR and CIR compared to other noncoherent techniques, additional filtering of ACI due to inherent PLL selectivity, possible relaxation of filtering requirements for ACI in the radio front-end, and easy reconfiguration of PLL parameters such as damping factor and bandwidth to accommodate different operating conditions. Furthermore, PLL-based receivers are suitable for integration, leading to a single-chip radio receiver for DECT [6].

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Provisioning QoS features for input-queued ATM switches

S. Li and N. Ansari

A new scheduling algorithm is proposed to improve on existing algorithms designed for input-queued ATM switches. By assigning a session weight according to its queue length normalised by its rate and using maximum weight matching to obtain a match, the proposed algorithm can avoid starvation of slow sessions, thus providing good delay properties as well as fair services, and at the same time reducing traffic burstiness.

Introduction: The input-queued switching architecture is becoming an attractive alternative for designing very high speed switches owing to its scalability. At any given time slot, an input-queued switch can transmit a cell from each input, and likewise can transmit a cell to each output. Input contention occurs when multiple cells stored in a given input are directed to different outputs, and similarly, output contention occurs when multiple cells from different inputs are directed to the same output. Input and output contentions are the major problems limiting the throughput of an input-queued switch.

Maximising the throughput of an input-queued switch is equivalent to finding maximum matching in a bipartite graph [1]. Maximum size and maximum weight matching based algorithms have been proposed to achieve 100% throughput for uniform and non-uniform traffic [2 – 4]. However, only aiming to maximise throughput could have adverse effects on traffic shape and quality of service (QoS) features such as delay and fairness. In this Letter, we propose a new maximum weight matching based algorithm, referred to as longest normalised queue first (LNQF),to achieve 100% throughput and provide QoS features at the same time.

Switch and traffic models: Consider an $N \times N$ input-queued switch. To provide QoS features, the switch resources, i.e. the bandwidth and storage, should be allocated on a per-session basis. Therefore, a per-session queueing architecture is adopted in which a separate FIFO queue is maintained for each session in every input. Denote $Q_{i,j}$ as the virtual output queue (VOQ) which consists of all sessions arriving at input i and directed to output j. Since sessions in the same VOQ have the same destination, each VOQ can be treated as an entity in resolving the contentions. Denote $I_{i,j,k}$ as the kth session in $Q_{i,j}$, and $\lambda_{i,j,k}$ as its arrival rate. The arrival rate to $Q_{i,j}$ can thus be expressed as $\lambda_{i,j} = \Sigma_k \lambda_{i,j,k}$. The aggregate arrival process to input i is said to be uniform if $\lambda_{i,m} = \lambda_{i,n}$, $\forall m \neq n$, $1 \leq m$, $n \leq N$. Otherwise, the process is said to be non-uniform. The traffic pattern is admissible if and only if $\Sigma_{j=1}^{N} \lambda_{i,j} \leq 1$ and $\Sigma_{i=1}^{N} \lambda_{i,j} \leq 1$. The traffic in a real network is highly correlated from cell to cell, and cells tend to arrive at the switch in bursts. Therefore, the on-off bursty traffic model is adopted, in which the burstiness is defined as the ratio of the peak cell rate to the average cell rate.

LNQF algorithm: Denote $l_{i,j,k}(n)$ as the length of the FIFO queue corresponding to session $I_{i,j,k}$ in $Q_{i,j}$ and $l_{i,j}(n) = \Sigma_k l_{i,j,k}(n)$ as the length of $Q_{i,j}$ at time slot n. In LNQF, the weight of a VOQ is set to be the normalised queue length, which is the total queue length of the VOQ divided by its rate, i.e. $w_{i,j}(n) = l_{i,j}(n)/\lambda_{i,j}(n)$. Let $\underline{W}_i(n) = (w_{i,1}(n), w_{i,2}(n), ..., w_{i,N}(n))^t$ be the weight vector associated with input i, and $S = [S_{i,j}(n)]$ be the service matrix which indicates the match between inputs and outputs. $S_{i,j}(n)$ is set to 1 if input i is scheduled to transmit a cell to output j, and 0 otherwise. Let $\underline{S}_i(n) = (S_{i,1}(n), S_{i,2}(n), ..., S_{i,N}(n))^t$ be the service vector associated with input i. The LNQF scheduler performs the following in each time slot n:

(i) Each input computes the normalised queue length of each VOQ stored in it, sets it as the weight of that VOQ, and sends the weight vector $\underline{W}_i(n)$ to the scheduler.

(ii) The scheduler finds a match that achieves the maximum aggregate weight under the constraint of unique pairing, i.e.

$$\arg\max_{S} \left[\sum_{i,j} S_{i,j}(n) w_{i,j}(n) \right]$$

such that $\Sigma_i S_{i,j} = \Sigma_j S_{i,j}(n) = 1$, sends the service vector $\underline{S}_i(n)$ to the corresponding input, and uses the service matrix $[S_{i,j}(n)]$ to configure the fabric.

(iii) Each input computes the normalised queue length of each session in the matched VOQ indicated by $\underline{S}_i(n)$, and selects the session with the longest normalised queue length for transmission.

A switch is stable if and only if the expected queue length in the switch does not increase without bound. It can be proved, similarly to the proof in [3], that the switch using LNQF is stable for all admissible traffic patterns.

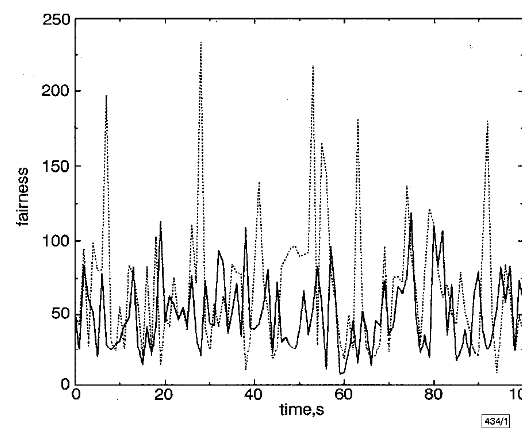

Fig. 1 *Fairness comparison showing LNQF against LQF*

——— LNQF
··········· LQF

Performance: A 4×4 input-queued switch was considered for simulations in which bursty traffic was generated based on the on-off traffic model. The average burst length was chosen to be 20 cells and the burstiness was 2. The traffic was non-uniform, i.e. the arrival rates of the VOQs in the same input were different, and were 0.5, 1, 2 and 5Mbit/s. Two sessions in each VOQ, a fast session with a rate four times that of a slow session, were generated. A traffic load of 0.95 was assumed, and each simulation lasted 100s.

The fairness of LNQF and LQF is shown in Fig. 1, where the fairness is defined as

$$F_S = \max_{\forall i,j,j\neq i} \left| \frac{W_i(t_1,t_2)}{r_i} - \frac{W_j(t_1,t_2)}{r_j} \right|$$

where $W_i(t_i, t_2)$ is the number of cells delivered for session i during the time interval $(t_1, t_2]$, and r_i is the rate of session i. The smaller the amount of fairness, the fairer the server is. As shown in the Figure, the average fairness of LNQF is smaller than that of LQF, and so is the variation of the fairness.

Table 1: Statistics of simulation results

Schedulers	LNQF	LQF	OCF
Average delay (μs) of VOQ1 session 1 λ = 0.1Mbit/s	10.8	123.5	17.4
Average delay (μs) of VOQ1 session 2 λ = 0.4Mbit/s	16.2	54.5	17.0
Average delay (μs) of VOQ2 session 1 λ = 0.2Mbit/s	13.0	75.8	15.2
Average delay (μs) of VOQ2 session 2 λ = 0.8Mbit/s	15.3	25.1	15.7
Average delay (μs) of VOQ3 session 1 λ = 0.4Mbit/s	13.3	38.7	15.2
Average delay (μs) of VOQ3 session 2 λ = 1.6Mbit/s	15.2	12.2	15.6
Average delay (μs) of VOQ4 session 1 λ = 1Mbit/s	15.1	17.6	15.7
Average delay (μs) of VOQ4 session 2 λ = 4Mbit/s	16.4	5.2	16.1
Average queue length (cell)	128.1	129.3	132.4
Fairness	50.3	68.1	50.5
Average transmission time (time slot) per burst:	135.6	138.1	92.4

Table 1 summarises the performance comparison among LNQF, LQF and OCF:

(i) All three algorithms have similar average queue lengths, indicating that switches using these algorithms yield similar throughput.

(ii) LQF tends to starve the slow session, i.e. the average delay of a slow session is much higher than that of a faster session, while LNQF provides comparable delay for each session.

(iii) The average time for OCF to complete transmitting a burst is much shorter than that for LNQF, implying that LNQF is a better 'traffic shaper' in reducing burstiness.

Conclusions: We have proposed a new algorithm, LNQF, which improves on existing algorithms in terms of delay, fairness, and burstiness, and demonstrated through simulations that LNQF is a starvation-free algorithm which can provide better fairness than LQF, and better effectiveness in smoothing traffic than OCF.

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Stereophonic speech recognition in noise using compensated hidden Markov models

D.M. Brookes and M.H. Leung

A novel procedure is presented for noise compensation in hidden Markov model speech recognisers. The procedure uses two microphone signals and, unlike previous approaches, does not require the noise spectrum to be stationary even in the short term. Results are presented showing that the performance of the compensated system equals or exceeds that obtained using matched training.

Introduction: Most speech recognition systems are designed for use with only a single microphone and their performance generally degrades substantially when background noise is present. It is possible in such systems to compensate for the noise either by modifying the input signal or by adjusting the speech models that are used in recognition [1 – 3]. In either case, it is normally necessary to estimate the parameters of a noise model during non-speech intervals and to assume that they remain unchanged during subsequent speech. The noise model may assume a stationary spectrum or may represent the noise as a dynamic process but, even in the latter case, the model parameters are assumed to be constant in the short term.

In many environments, such as offices and vehicles, it is feasible to use a stereo pair of microphones for speech recognition. In this work, we show that, when two microphone signals are available, it is possible to estimate the noise with much weaker stationarity assumptions and hence to achieve improved speech recognition performance even in the presence of noise sources whose spectrum varies unpredictably.

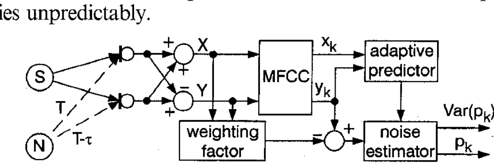

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Fig. 1 *Block diagram of noise estimation procedure*

Noise estimation: A block diagram of the noise estimation procedure is shown in Fig. 1, in which S and N denote the speaker and a typical noise source, respectively. If the speaker is not equidistant from the two microphones, it is necessary to estimate the speaker's direction and to compensate the microphone signals for any differential delay [4]. The time delays from the noise source to the microphones are, respectively, T and T–τ, as shown.

The two microphone signals are combined to form a sum signal with spectrum $X(j\omega) = 2S(j\omega) + P(j\omega)$ and a difference signal with spectrum $Y(j\omega)$. If the noise arises from a number of sources N_i that are independent of S, then the ratio of the noise powers, $|P|^2$ and $|Y|^2$, is given by

$$\frac{|P(j\omega)|^2}{|Y(j\omega)|^2} = \frac{\left|\sum_i N_i(j\omega)\cos(\frac{1}{2}\omega\tau_i)\right|^2}{\left|\sum_i N_i(j\omega)\sin(\frac{1}{2}\omega\tau_i)\right|^2} \quad (1)$$

where τ_i, the relative time delay of the two microphone signals for N_i, depends on the direction of the noise source. We assume that τ_i are stationary over intervals of a few seconds but that N_i may be time varying.

The expression in eqn. 1 can be simplified if, at each particular frequency, the noise comprises multiple echos originating from a single dominant sound source, $N(j\omega)$ In this case $N(j\omega)$ will be a common factor in both numerator and denominator and the ratio $|P/Y|$ will be stationary even if $N(j\omega)$ is time varying. We estimate the interfering noise component $|P|$ by measuring $|Y|$ and assuming that the ratio $|P/Y|$ is stationary over short periods.

For each input frame the spectra X and Y are transformed into the mel-cepstral domain [5] to obtain cepstral coefficients x_k and y_k for $k = 0$ to K. When speech is present, we cannot observe the noise component P directly so we estimate its cepstral coefficients from y_k using

$$p_k \simeq a_k y_k + b_k \quad (2)$$