Performance Analysis of Dynamic Packet Scheduling within a cdma2000 Broadcast Network

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***Abstract*— cdma2000 1xEV-DO, one of the prominent third generation mobile communication systems, provides Broadcast and Multicast Services (BCMCS) to meet increasing demand for multimedia data services. We propose a dynamic packet scheduling algorithm that works with a retransmission scheme for the BCMCS scheduler. Integrated with EDF (Earliest Deadline First) real-time scheduling, the proposed algorithm not only adapts to dynamic contexts efficiently but also satisfies the real-time requirements of broadcast streams. Furthermore, by supporting handoff in this dynamic scheduling environment, we can reduce the discontinuity of service when a mobile node moves into a new service area. Extensive simulations have quantitatively verified the efficiency of our approach even when mobile nodes move around among many service areas.**

I. Introduction

The third-generation (3G) cellular technologies now being deployed worldwide offer a host of services that can be used to support communications among members of specific groups and to disseminate information. Among them, broadcast and multicast services (BCMCS) are currently being standardized by various mobile wireless standard bodies such as 3GPP and 3GPP2. The 3GPP2 group recently baselined the specifications for a cdma2000 high-rate broadcast packet-data air interface [1] [2] [3] [4]. The goal of these specifications is a system that can deliver multimedia broadcast and multicast traffic with minimum resource usage by both the radio access and core networks. In addition, users expect minimum latency when joining or leaving the network, and multimedia video streams must be delivered continuously over radio channels as mobile users move around.

In general, a wireless radio channel has a much higher error-rate than that of wired link. The unreliable and error-prone nature of the radio channel is the major challenge in the servicing of video streams over cdma2000 1xEV-DO broadcast networks. Wireless radio channels are affected by time-varying fading and interference conditions, which may lead to corrupted packets. These errors tend to occur in clusters or bursts, with relatively long error-free intervals between them. Error control is mandatory when data are sent over wireless access networks to an end-user. Forward error correction (FEC) is the control method most commonly used when sending a broadcast video stream over a lossy network such as a wireless radio channel. In BCMCS, the MAC protocol uses Reed-Solomon (RS) coding as the method of forward error correction. As a result, if the channel condition is good, many slots will be overbooked due to the parity information required by Reed-Solomon. Additionally, using Reed-Solomon, the capacity for error recovery declines when the condition of a channel deteriorates, because FEC error recovery schemes become ineffective as the length of the error bursts increases. By using more intelligent automatic retransmission request (ARQ) techniques to protect the transmitted video stream, we can save many slots when the condition of a channel is good, and reduce the necessary playback quality degradation when the channel condition is bad. Additionally, application-aware retransmission of corrupted packets exploiting the characteristics of MPEG-4 FGS (Fine Granular Scalability) coded video streams, becomes possible.

In the existing packet scheduling technique used by BCMCS, each broadcast video stream is assigned its own time slot for delivery to the mobile node, and these assignments are determined statically in advance. But this static scheduler, which is currently incorporated in the BCMCS specification, cannot adapt to an environment in which content streams change dynamically [5]. We therefore propose a dynamic scheduler based on the EDF (Earliest Deadline First) algorithm to replace the existing cdma2000 1xEV-DO BCMCS scheduler.

We also propose a method for efficient handoff when a mobile node moves into a new service area, and analyze its performance with respect to packet error rate. In the current BCMCS, soft combining is allowed on the forward link during high-speed broadcast service transmission, because multiple sectors may transmit the same data. However, using a dynamic packet scheduler, the packets are individually scheduled at each base station, and so soft combining is impossible. As an alternative method of preventing degradation of playback quality during the handoff of a node, retransmission of lost packets, identified by the sequence numbers in the packet headers, can be requested.

The remainder of this paper is organized as follows: in Section II, we review work related to our own research and, in Section III, we analyze the performance of the proposed handoff technique. The experimental environment and our results are described in Sections IV and V. Section VI concludes the paper.

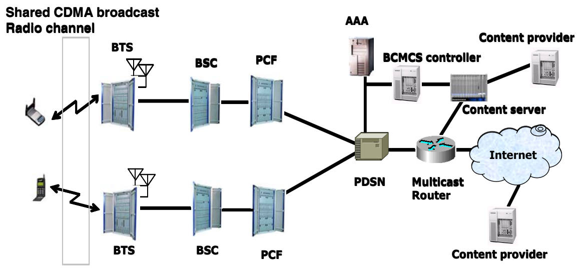


Fig. 1. The cdma2000 BCMCS network architecture.

II. RELATED WORK

Kang et al. [5] proposed a dynamic packet scheduler based on EDF for cdma2000 1xEV-DO BCMCS, with the service architecture depicted in Figure 1. Multimedia video streams have timing constraints which means that a video frame is useful only if it arrives at the buffer of the MPEG video decoder before a specified deadline, and so an EDF-based packet scheduling algorithm is appropriate. The dynamic scheduler works with a retransmission scheme for priority-based error recovery. The proposed scheme not only adapts to dynamic contexts efficiently but also satisfies the real-time requirements of broadcast streams while making it unnecessary to add additional parity information to the original content stream: the slots thus saved can be used more flexibly, for instance to retransmit or to unicast packets. Using this architecture, admission can easily be controlled using a simple utilization bound test when a retransmission request occurs or a new broadcast flow starts.

In the current BCMCS scheme, when a mobile node requests a BCMCS content stream, packets are transmitted from the PDSN (Packet Data Serving Node) to the PCF (Packet Control Function), which uses a timestamp to make sure that each packet is received simultaneously at all base stations. However, using a dynamic scheduler, all packets are scheduled by their deadlines and thus the scheduling order of each packet can be different in each service area. When a mobile node enters a new service area, it is possible that the packets it requires have already been transmitted by the base station in that area. This can lead to appreciable degradation in playback quality, especially if the lost packets are base-layer packets in a video stream encoded using MPEG-4 FGS (Fine Granular Scalability) [6] [7].

In this study, we exploit the FGS characteristics of the MPEG-4 Part 2 standard. This scalable video coding scheme features a layered bit-stream structure in which different layers have different levels of importance. While accuracy in the base layer is essential for decoding video streams, the enhancement layers are more tolerant of channel errors. Thus, it seems reasonable that the more important layers should be provided with more protection against errors than the less important layers. The Reed-Solomon code that is currently used for error recovery in BCMCS does not exploit this possibility, and treats all packets equally. We guarantee delivery of the base-layer packets by giving them more opportunities for retransmission, thus avoiding abrupt drops in service quality.

In the framework proposed by Kang et al. [8], new video streams can be admitted even when no available slots remain, although the playback quality is somewhat degraded as a result of dynamic adjustment of the scalability of the video streams that are already being serviced. To adjust the scalability of each video stream, three policies – fair degradation, victim selection and proportionally fair degradation – are proposed. This adaptive framework has the following key features:

- **Graceful quality degradation**: unlike non-scalable video, scalable video can adapt its representation to bandwidth variations. When the network has to drop packets, it does so with awareness of the video that the packets represent. As a result, perceived quality is gracefully degraded even under severe channel conditions.
- **Efficiency**: when there is excess bandwidth, it is used efficiently in a way that maximizes the perceived quality or the revenue.
- **Fairness**: the resource can be shared to enhance the average playback quality.
- **Guaranteed delivery of the base layer**: the base layer stream is guaranteed by giving it more opportunities for retransmission. This avoids abrupt drops in service quality.

III. THE PROPOSED SCHEME

The previous work mentioned in Section II did not consider handoff events triggered by mobile nodes. The utilization of slots fluctuates as a mobile node enters or leaves the current service area, which may influence the number of retransmission requests. The utilization of slots affects the error recovery capacity of the retransmission scheme: the chance of retransmission falls as slot utilization increases, meaning that fewer corrupted packets will be recovered; and the opposite is also true.

In broadcast multimedia applications, a packet generated at the content server needs to be received before a specified deadline, and the EDF algorithm endeavors to achieve this. When a mobile node fails to receive packets as a result of handoff, or just because of bad air conditions, it sends retransmission request messages through the reverse unicast channel with the sequence numbers of the lost packets, which are put into queues for retransmission. There is one queue for the base layer and another for the enhancement layer. The EDF scheduler first selects the broadcast packet which has the earliest deadline; but if there are no broadcast packets to service, it selects one from the retransmission queue containing base-layer packets, on the basis of its deadline; finally, if there are no base-layer packets to be serviced, the scheduler services an enhancement-layer packet.

IV. THE EXPERIMENTAL ENVIRONMENT

Our simulator consists of a video frame generation module, an admission control module, a handoff request generation module, a scheduler, and an error generation module, as shown in Figure 2. The handoff request module generates handoff

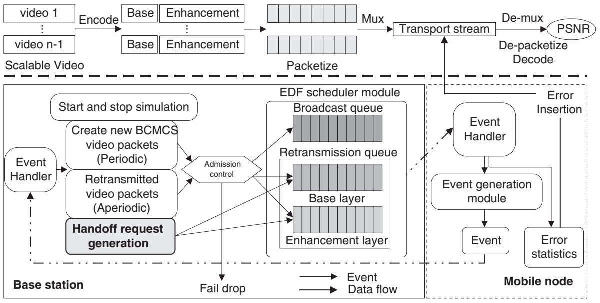


Fig. 2. Structure of the simulator.

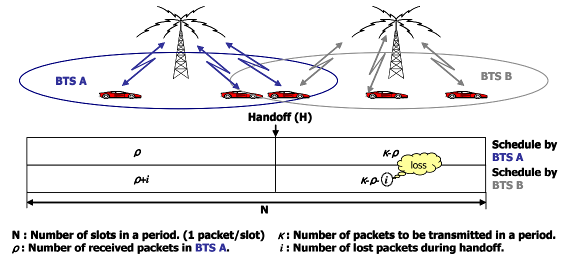


Fig. 3. Packet loss during handoff.

events for the mobile nodes. These arrive in a Poisson distribution with mean inter-arrival time $1/\lambda$. Departures of mobile nodes from the service area also follow a Poisson distribution with a mean interval of $1/\lambda$. Packets deemed to have been lost wait in a queue to be scheduled. Whether a video packet is received successfully at the mobile node is determined by an error generation module, using the simple threshold model suggested by Zorzi [9], which can simulate the error sequences generated by data transmission on a correlated Rayleigh fading channel: these errors occur in clusters or bursts with relatively long error-free intervals between them. By choosing different values for the physical-layer (MAC-layer) bit error-rate and $f_d NT$ (the Doppler frequency normalized to the data-rate with block size N), we can model different degrees of correlation in the fading process. The value of $f_d NT$ determines the correlation properties, which are related to the mobile speed for a given carrier frequency. When $f_d NT$ is small, the fading process has a strong correlation, which means long bursts of errors (slow fading). Conversely, the occurrence of errors has a weak correlation for large values of $f_d NT$ (fast fading). In our experiments, we set the values of $f_d NT$ to 0.01, 0.02 and 0.03. We used the Foreman test bench video sequences streamed at 30 frames per second, with a total of 10,000 frames. Results are presented for average physical-layer packet error rates ($PER_{physical}$) of 1%, 3%, 5% and 10%. Each video stream is handled with our reference MPEG-4 FGS codec, which is derived from the framework of the European ACTS Project Mobile Multimedia Systems (MoMuSys) [10], but modified for our experiment. It consists of a 100 kbps base-layer bit-rate and a 120 kbps enhancement-layer bit-rate with 120 levels (1 kbps step). All experiments were performed with the value of mean handoff inter-arrival time ($1/\lambda$), 60 seconds (one minute). We compared the proposed scheme with the original FEC-based static scheme employed in BCMCS. The (16, 12, 4) Reed-Solomon code, capable of recovering up to 4 octet erasures in each Reed-Solomon code word, was used, and the number of MAC packets per ECB row was 16 in our experiments. Reed-Solomon is used as an erasure code not as an error correcting code. To evaluate the two error recovery schemes, errors (generated as mentioned above) are injected into the original transport stream of target video sequence as shown in Figure 2 and the peak signal-to-noise ratio (PSNR) of the resulting video stream is calculated as an estimate of the quality of a reconstructed image compared with an original image.

V. PERFORMANCE EVALUATION

1) Analysis of handoff in a dynamic packet scheduling environment: In a dynamic packet scheduling environment, a mobile node which is moving toward a neighboring cell may lose packets even when there is no handoff delay, because all packets are scheduled in each service area independently, as shown in Figure 3. When a mobile node enters a new service area, and handoff occurs at time H, packets that have not been received from the previous cell (BTS A) may already been transmitted in the new cell (BTS B). The expected number of packets lost (E_{lost}) during handoff can be calculated as follows:

$$E_{lost} = \sum_{H=0}^{N} \sum_{\rho=Max(0,\kappa-N+H)}^{Min(\kappa,H)} \sum_{i=Max(0,\kappa-N+H-\rho)}^{Min(\kappa-\rho,H-\rho)} \left[\frac{1}{N} \times \frac{{}_H C_\rho \times {}_{N-H} C_{\kappa-\rho}}{{}_N C_\kappa} \times \frac{{}_H C_{\rho+i} \times {}_{N-H} C_{\kappa-\rho-i}}{{}_N C_\kappa} \times i \right]. \quad (1)$$

As mentioned in Section III, the error recovery capacity of the proposed scheme is determined by the current slot utilization. Assuming that handoff requests from the neighboring service area follow a Poisson distribution with arrival and departure rate λ, the maximum slot utilization in the current service area (U_{cur}) is as follows:

$$Max(U_{cur}) = U_P + \sum_{\psi=1}^{node\ count} \left(\frac{e_\psi}{p_\psi}\right) \sum_{\kappa=1}^{Retry_{Max}} (\varepsilon_\psi)^\kappa + U_{handoff} . \quad (2)$$

where e_ψ represents the number of required slots in a period p_ψ for the video stream being received by mobile node ψ, and ε_ψ is the average steady-state MAC-layer packet error rate ($PER_{physical}$) of node ψ. A mobile node is able to make up to $Retry_{Max}$ retransmission requests. U_P is the utilization of n periodic video frames which have flow IDs from 1 to n, when the number of slots required by the forward video stream i is e_i, and the period of the video stream i is p_i slots. U_P and e_i are given as follows:

$$U_P = \sum_{i=1}^{n} \frac{e_i}{p_i} \text{ , } e_i = \left\lfloor \frac{b_i}{\mu_i} \right\rfloor \text{ .} \tag{3}$$

where the bit-rate of video stream i is b_i kbps and the number of bits that can be forwarded in one slot is μ_i.

P_{drop} is the probability that retransmission packet τ is not admitted into the wait queue and is therefore not scheduled again, and this probability can be defined as

$$P_{drop} = \begin{cases} 0 & \text{if } \tau \text{ is admitted by utilization bound test.} \\ 1 & \text{if } \tau \text{ fails to admit to the system.} \end{cases}$$

Thus the expected value of $P_{drop}(E_{drop})$ is calculated as:

$$E_{drop} = \left(\frac{|1-U_p| - (1-U_p)}{2(1-U_p)} \right) \left(1 - \frac{1}{U_{cur}} \right) \text{ .} \tag{4}$$

In our proposed scheme, the maximum packet error rate for packet τ after the $Retry_{Max}$th retransmission ($Max(PER)$) is expressed as (5):

$$Max(PER) = \varepsilon \left\{ E_{drop} + (1 - E_{drop})(\varepsilon) \right\}^{(Retry_{Max})} \text{ .} \tag{5}$$

where ε is the average steady-state packet error rate of all mobile nodes. $U_{handoff}$ is the average utilization factor achieved by packets during handoff. As the mean inter-arrival time is $1/\lambda$, $U_{handoff}$ is calculated as follows:

$$U_{handoff} = \frac{\lambda E_{lost}}{600}, \text{ (600 slots/sec) .} \tag{6}$$

2) Simulation results: Figure 4 shows the average number of packets lost during handoff as the handoff position H changes. The two curves represent the results of 1,000 and 10,000 trials; the video is streamed at 220kbps; and the period is 1 second (N=600). With more trials, the graph stabilizes to a smooth curve. The expected number of lost packets during handoff is 2.107 and 2.111 for 1,000 and 10,000 trials respectively, which is nearly in agreement with the theoretical value of 2.094, derived from Equation (1).

Figure 5 shows the changing utilization of slots as the number of mobile nodes and $PER_{physical}$ increases. The slot utilization increases as the number of mobile nodes increases because of the growing number of retransmission requests. Also, more packets request retransmission as the channel condition deteriorates. When $PER_{physical}$ is 1% or 3%, the utilization falls below unity even when the number of mobile nodes reaches 60, which means that slots are still available after corrupted packets have been retransmitted up to $Retry_{Max}$ times ($Retry_{Max} = 2$ in our experiment). However, when $PER_{physical}$ are 5%, 7% or 10%, the utilization arrives at unity as the number of mobile nodes exceeds 50, 40 and 30 respectively. After these saturation points, $PER_{recovered}$ (PER in upper layer) increases abruptly because there are no slots left for additional retransmission, as shown in Figures 6, 7 and 8, which demonstrate how $PER_{recovered}$ changes as the number of mobile nodes increases for value of $PER_{physical}$ is 3%, 5% and 10%.

The error recovery capacity of Reed-Solomon is not affected by the node count or the number of handoff requests. However, the performance of our scheme is strongly related to the node count, the channel condition, and the number of handoff requests, as described above, because these factors all influence slot utilization. As the number of mobile nodes increases and the channel condition deteriorates, $PER_{recovered}$ increases with our scheme, but it still shows an improved playback quality (PSNR), because it protects the base layer by retrying transmission repeatedly.

Figures 9 and 10 show the average PSNR for different numbers of mobile nodes, with values of $PER_{physical}$ between 1% and 10%, and f_dNT set to 0.02. Our scheme shows better playback quality even when the proposed handoff scheme is

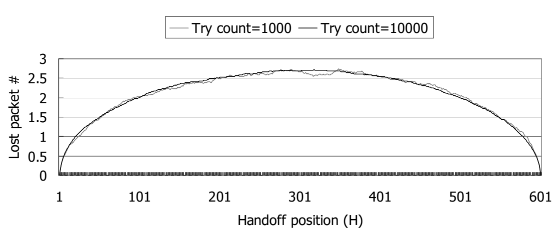


Fig. 4. Number of lost packets during handoff.

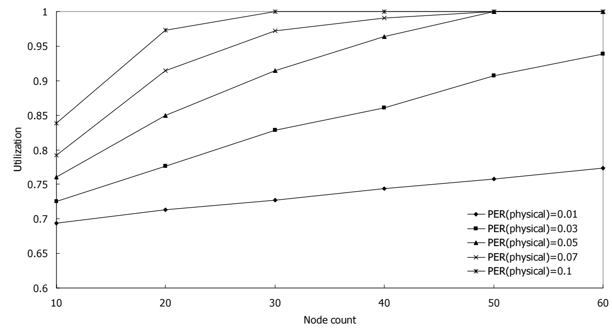


Fig. 5. Utilization of slots.

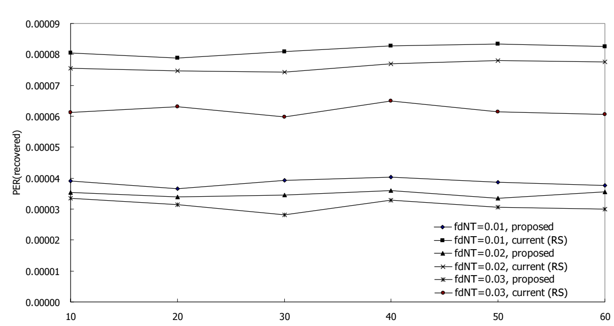


Fig. 6. $PER_{recovered}$ when $PER_{physical}$ is 3%.

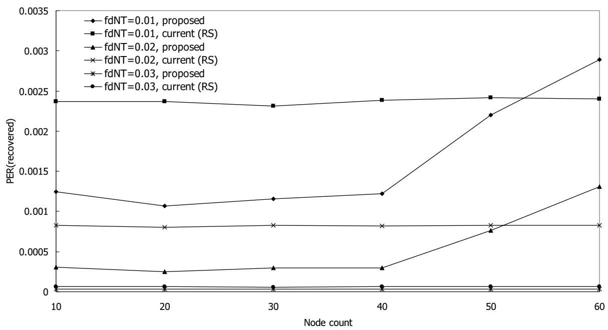


Fig. 7. $PER_{recovered}$ when $PER_{physical}$ is 5%.

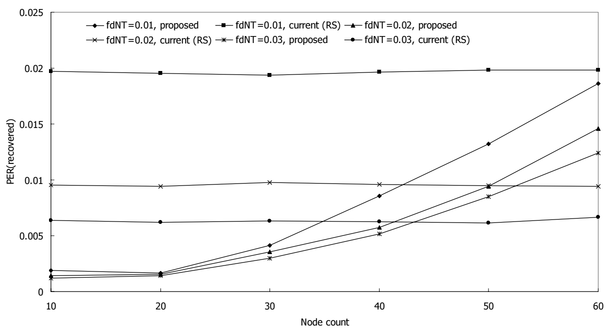
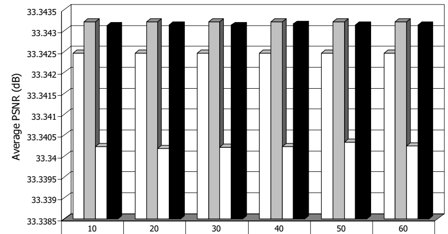


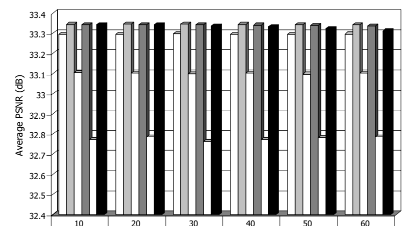
Fig. 8. $PER_{recovered}$ when $PER_{physical}$ is 10%.



	10	20	30	40	50	60
1%, current (RS)	33.342465	33.342465	33.342465	33.342465	33.342465	33.342465
1%, proposed	33.343212	33.343212	33.343212	33.343212	33.343212	33.343212
3%, current (RS)	33.340228	33.340183	33.34021	33.340225	33.340333	33.340245
3%, proposed	33.343121	33.343134	33.343129	33.343157	33.343138	33.343132

Node count

Fig. 9. PSNR when $PER_{physical}$ is 1% and 3% ($f_dNT = 0.02$).



	10	20	30	40	50	60
5%, current (RS)	33.293851	33.293078	33.296773	33.294855	33.294866	33.295132
5%, proposed	33.342701	33.342774	33.342798	33.342621	33.341404	33.340313
7%, current (RS)	33.105127	33.10108	33.098492	33.100246	33.095984	33.10108
7%, proposed	33.341526	33.341541	33.34085	33.33936	33.337337	33.334901
10%, current (RS)	32.773121	32.784654	32.761429	32.772879	32.782023	32.78368
10%, proposed	33.340257	33.339642	33.335627	33.330665	33.322714	33.311071

Node count

Fig. 10. PSNR when $PER_{physical}$ is 5%, 7% and 10% ($f_dNT = 0.02$).

also supported. As the handoff request intervals get shorter (the effect is not depicted in this paper) and the number of mobile nodes increases, $PER_{recovered}$ for our scheme increases, and eventually exceeds that of the current scheme; but the average PSNR remains higher, and so the relative advantage of our scheme grows as $PER_{physical}$ increases. For example, the $PER_{recovered}$ of our scheme exceeds that of the current scheme when $PER_{physical}$ is 5% and the number of mobile nodes is greater than 50. However, Average playback quality is about 0.05dB higher than the current scheme. In all cases, with either scheme, the value of f_dNT affects the error recovery capacity. When the value of f_dNT is small (slow fading), the errors are more bursty, which degrades error recovery capacity, as shown in Figures 6-8.

VI. Conclusion

We have analyzed the performance of a dynamic packet scheduling scheme with retransmission, in an environment in which mobile nodes move between service areas. During handoff of a mobile node to a new service area, packets can be lost because packet scheduling is performed within each service area independently. We derived the expected number of lost packets during handoff and validated it by simulation. As the number of handoff requests increase, slot utilization increases due to the retransmission of lost packets and this result in the degradation of error recovery capacity after the slot utilization exceeds unity; but our scheme still achieves better playback quality. By protecting base-layer data packets by means of preferential retransmission, we can avoid abrupt drops in service quality. The resulting PSNR values are higher than those achieved by the current Reed-Solomon based error recovery scheme, even though the packet recovery capacity reduces as the number of mobile nodes increases.

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