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An embedded packet train and adaptive FEC scheme for effective video adaptation over wireless broadband networks

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Abstract: With the rapid growth of wireless broadband technologies, such as WLAN and WiMAX, quality streaming video contents are available through portable devices anytime, anywhere. The layered multicast system using scalable video codecs has been proposed as an efficient architecture for video dissemination taking account of user and link diversities. However, in the wired/wireless combined best-effort based heterogeneous IP networks which provide more fluctuation in available bandwidth and end-to-end delay, the performance of streaming systems has been greatly degraded due to frequent packet loss, resulting from either wired congestion or wireless fading/shadowing. In this paper, we present a real-time embedded packet train probing scheme for estimating end-to-end available bandwidth so as to accomplish effective congestion and error control. This is facilitated by effective classification of packet loss sources, delay trend detection algorithm and flexible transmission rate of packets. Under the proper wireless channel modelling and estimation, our layered structure can allow appropriate subscription of video layers and adaptively insert necessary amount of forward error correction (FEC) packets so as to achieve QoS optimized system for scalable video multicasting.

Key words: Adaptive FEC, Available bandwidth estimation, Layered streaming, Congestion control algorithm
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INTRODUCTION

With the emergence of wireless technology, various multimedia services, e.g., audio and/or video, are reaching us today through portable devices anytime, anywhere and even more increasingly accessible in the near future. High quality video streaming over wireless IP is one of the most attractive applications by ongoing deployment of wireless local area network (WLAN) hotspots and even powerful WiMAX mobile coverage. However, the present Internet is not providing adequate quality of service (QoS) for users and not ready to become the universal network satisfying all our multimedia communication needs.

Scalable multimedia disseminated through receiver driven layered multicast (Liu and Hwang, 2003) is believed to be a promising technique solving challenges of network resource limitation and receiver diversities simultaneously. Specifically, users requesting the same contents can share the data stream across networks and receive the service in the best quality affordable depending on its link and device capabilities. To provide scalable video quality, H.264/AVC scalable extension using motion-compensated temporal filtering (MCTF) (Schäfer *et al*., 2005) together with spatio/temporal/SNR scalability has shown coding efficiency comparable to that of single-layer codecs.

Nevertheless, the quality of video delivery is largely affected by error rate and bandwidth variation. In today's heterogeneous Internet, various wireless access technologies such as WLAN, cellular network, and bluetooth, may co-exist with wired backbone introducing different link layer control mechanisms and high bit error rates. The consequence of the wired/wireless combined best-effort based IP networks thus provides more fluctuations in available bandwidth and end-to-end delay, causing more media decoding degradations.

To overcome these preblems, there are two approaches main to achieving end-to-end QoS support: network centric and end-system centric (Zhang *et al*., 2005). A network centric approach provides prioritized services to different traffic classes in networking equipments between senders and receivers. Representative proposals are DiffServe networks on top of conventional IP as well as 802.11e and WiMAX for wireless environments. An end-system centric method instead depends solely on senders and receivers and performs congestion and error control to maintain most media quality. For example, Tan and Zakhor (2001) proposed a layered FEC scheme for layered video multicast using equation based rate control. In (Hsiao *et al*., 2005a), the amount of transmitted FEC is further decided by modelled wireless channel estimation. A new end-system driven solution featuring embedded probing is proposed in this paper for more flexible layer construction and subscription while being reliable in diverse channel conditions. Through effective integration of packet loss classification, effective probing, congestion control via layered structure, and packet level FEC, our proposed system is specially designed for wired/wireless layered multicast applications considering scalable extension of H.264/AVC. Simulation results show optimal video quality can be sustained throughout changing network condition and different users based on our flexible structure. The optimality comes from the best trade-off between number of video layers subscription and number of additional FEC packets insertion to simultaneously satisfy the estimated available bandwidth and the estimated wireless channel error condition.

The paper is organized as follows: Section 2 addresses some existing research works related to our proposed system. We then introduce our proposed architecture in Section 3. Section 4 discusses the simulations and results, followed by the conclusion in Section 5.

EXISTING RELEVANT RESEARCH

Packet loss is one of the most damaging factors in multimedia networking. It can be due to wired network congestion or wireless fading/shadowing. These two losses should be handled differently to achieve effective QoS performance. We will start with some discussions on congestion control, and then error control.

Congestion control and delay trend model

For end-to-end QoS, congestion control is one of the important tasks required at end-systems. Basically, end-systems adjust data rates according to observed network conditions, i.e., packet loss and delay statistics. TCP transport protocol is a typical example. However, the long delay created by retransmission used in TCP is not practical in multimedia applications, such as real-time streaming. Unlike TCP, many UDP control schemes widely used in multimedia communication try not to grab more channel shares than a TCP session under the same environment. TFRC (Floyd *et al*., 2000) is a well known equation-based protocol which manages to get smooth and accurate round trip time (RTT), though its RTT-biased result is arguably representing the actual data rate.

Available bandwidth estimation tools also play an important role for layered congestion control. Generally, receivers perform bandwidth detection before subscribing to a proper layer according to its estimated available bandwidth. PLM (Legout and Biersack, 2000) utilized packet pair for layered video streaming over fair scheduler networks. BIC (Liu *et al*., 2005) modified delay trend detection from Pathload (Jain and Dovrolis, 2002) for a receiver-driven layered multicast protocol, and proposed an effective probing scheme with minimum additional network traffic for each receiver to subscribe to appropriate multicast group.

The delay trend model is proposed in BIC under first-in-first-out and fluid assumptions. In this method, if a stream is transmitted with data rate lower than or equal to that of the end-to-end available bandwidth, the relative one-way delay time of the received data packet stream at the receiver end will not show increasing trend. Although the fluctuating Internet traffic makes the delay trend detection of observed events more challenging, the BIC proposed a full-search scheme which can tolerate the traffic diversity.

Error control and wireless channel

Due to the bursty nature of multi-channel wireless fading, packets can be lost at a much higher rate than in wired channel. To provide error control, the packet level FEC erasure code (such as Reed-Solomon code) (Rizzo, 1997) has been proposed for error correction. Through adequate selection of (n,k) parameters based on desired protection level and affordable redundancy, the media stream can be decoded without error, as long as any subset of k or more packets in an n-packet block are successfully received. To efficiently decide the adequate error protection level, wireless channel error condition should be adaptively estimated. Gilbert/Elliot's two-state Markov model is commonly used to simulate the bit error rate resulting in the packet loss. This method can be approximated as a two-state Markov chain with parameters p and q representing transition probabilities from (packet) loss state to received state and from received state to loss state respectively. Thus, the steady state average packet loss rate p_{L} is:

$$p_{\mathrm{L}} = \frac{q}{p+q}. \tag{1}$$

The FEC decoding error rate, e_{FEC}, defined as the error rate of protected data, for a specific pair of p, q and (n, k) can be calculated using this model (Zhang *et al*., 2004) as:

$$e_{\mathrm{FEC},(n,k)} = \sum_{m=n-k+1}^{n} P(m,n), \tag{2}$$

$$P(m,n) = \sum_{s=1}^{n-m+1} p_{\mathrm{L}} G(s) R(m, n-s+1), \tag{3}$$

with

$$G(s) = \begin{cases} 1, & \text{for } s=1, \\ p(1-q)^{s-2}, & \text{for } s>1, \end{cases} \tag{4}$$

$$R(x,y) = \begin{cases} G(y), & \text{for } x=1, \\ \sum_{s=1}^{y-x+1} g(s) R(x-1, y-s), & \text{for } 2 \le x \le y, \end{cases} \tag{5}$$

$$g(s) = \begin{cases} 1-p, & \text{for } s=1, \\ p(1-q)^{s-2}, & \text{for } s>1, \end{cases} \tag{6}$$

where $p(m,n)$ is the probability of m lost packets out of n consecutive packets. Conclusively, the right amount of FEC packets can thus be decided assuming p and q estimated at receivers.

Along with error control, some actions need to be taken to maintain the performance of congestion control protocol when it comes to wireless. Congestion control can only deal with congestion loss while error control is for wireless loss. Incorporating packet loss classification (PLC) into the control loop is one of the promising ways (Hsiao *et al*., 2005b) to improve the efficiency and is used in our paper to discriminate two loss sources.

LAYERED FEC STRUCTURE WITH EMBEDDED PROBING

Fig.1 illustrates our proposed wired/wireless multicast system, where media quality degradation resulting from wireless loss is protected by adequate FEC erasure codes with embedded probing performed first to assure enough available bandwidth for redundancy and fair share with other sessions. The fundamental principle of this proposed system is the decoupling of several important modules (scalable video layer creation, PLC, bandwidth probing, adaptive FEC insertion) and conduct an effective integrated trade-off analysis to reach optimal number of video layers and FEC protection levels under all the resource constraints.

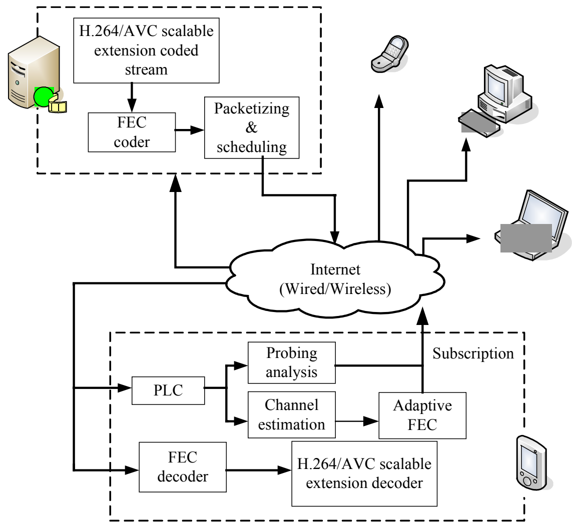

Fig.1 System diagram

Layered structure of scalable video

Our scalable video is created by the MCTF scalable extension of H.264/AVC, which is an emerging compression technology with coding efficiency comparable to that of the original H.264/AVC standard. With MCTF using lifting framework, temporal decomposition can be achieved nicely for SNR, temporal and spatial scalabilities. Aggregation and zero-padding of network abstraction units may need to compose packets in identical size when encoding FEC.

Video data and error protection codes are formatted in layers, where each layer is assigned to a multicast group (as shown in Fig.2) with rate $r_{i,j}$, i and j are indexes of video and FEC layers respectively. In this setup, layers with j=0 contain only video streams, otherwise j indicates level of protection for a specific i. There are totally V enhancement video layers and F protection levels for every video layer.

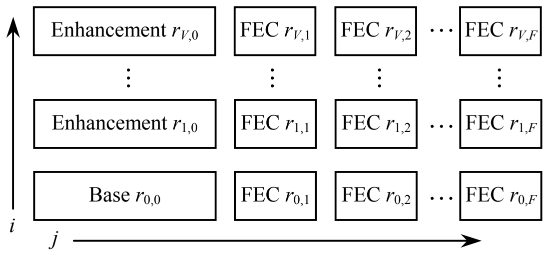

Fig.2 Rates of (*i,j*) in layers

To encode layered FEC, we decide feasible block parameters ($n_{i,j}$,k_i) for video layer i through network analysis under the structure illustrated in Fig.3. During the process, target decoding error rates, $e_{\mathrm{FEC},i}$, as well as k_i are set first, then the estimated p, q value sets representing wireless channel conditions are introduced into Eq.(2) for evaluating minimum $n_{i,j}$ satisfying:

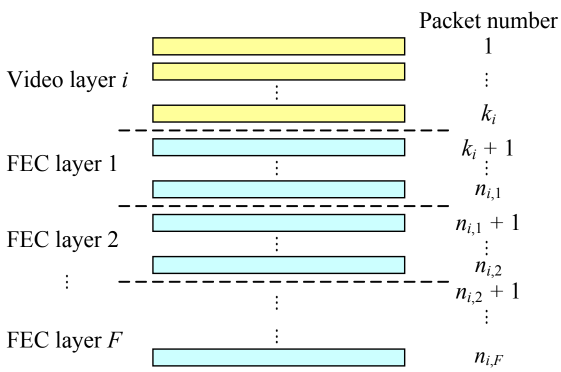

Fig.3 Layered FEC structure

$$e_{\mathrm{FEC},(n_{i,j},k_i)} \le e_{\mathrm{FEC},i}. \tag{7}$$

In other words, by modelling the network conditions using F number of p, q sets, we can derive the corresponding $n_{i,j}$, j=1, …, F in ascending order to generate F protection layers. In case we want to judge how many FEC layers to subscribe to after the whole scenario has been constructed, we can again choose a data layer i, plug in newly estimated p, q, find minimum n_{new} for Eq.(7) through the same process, and then pick j with $n_{i,j}$ closest to but larger than n_{new}.

Regarding inter-layer dependency and protection levels for every layer, the overall data rate is:

$$R_{v,f} = \sum_{i=0}^{v} \sum_{j=0}^{f} r_{i,j}, \tag{8}$$

where v is the number of enhancement layers subscribed and f, being the same for every i, is the FEC layers requested. Unequal protection to different video layers is supported by different $e_{\mathrm{FEC},i}$, not by f. For instance, under channel condition (p,q)=(0.8,0.2), we preset $e_{\mathrm{FEC},0}$=0.01, $e_{\mathrm{FEC},5}$=0.01, and k_0=k_5=8 resulting in $n_{0,j}$=17 and $n_{5,j}$=15. It also implies that a video layer requires all lower ones to be available for decoding (due to the cumulative layer structure of the adopted scalable codec), while not every sub-layer FEC is needed. Therefore, receivers acquire distinct video quality and amount of protection by subscribing to proper groups.

Embedded probing

To increase the data rate, either for more video data or for loss recovery, estimation of available bandwidth must be done in advance to prevent congestion. In contrast to BIC which requires specific synchronized and sequential scheduling for probing, we embed probing streams in regular ones through effective scheduling of packet transmission and take advantage of the fact that streaming systems usually have decoding buffer to tolerate some amount of delays. Moreover, the scheduling is receiver independent with no need to change all the time.

As shown in Fig.4 of our embedded probing, the stream is periodically separated into probing and regular intervals alternatingly with period T. The length of probing interval, t_{p}, is further divided into certain uniform probing regions according to the number of possible layers to go to, e.g., r_{p1} to r_{p4} in the figure. In each region, previously generated packets are delayed in transmission for the purpose of creating temporarily higher sending rate within a region.

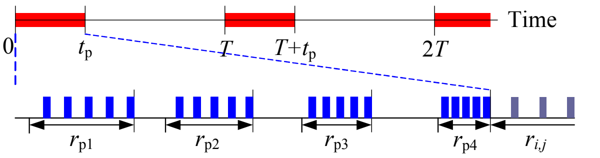

Fig.4 Embedded probing

At the receiver side, the interval and region a packet belongs to can be distinguished from its RTP timestamp; then fullsearch delay trend detection (Liu *et al.*, 2005) is applied to packets in time slots at objective rates. Since 50 packets is the shortest effective length of probing streams in general cases (Liu, 2004), the duration of each probing region is set to be the time to send 50 packets in regular intervals of base layer $(i,j)=(0, 0)$ and remains the same for every layer.

Other important facts are that the probing rate we are looking for is an aggregation rate $R_{v,f}$, not $r_{i,j}$. The aggregation rates of all possible target subscriptions should be included in a probing interval. More details will be addressed in the next section.

Quality optimization

Users at the receiver end always expect better quality. Since packet loss in multimedia streaming, which means loss of data, causes more serious quality degradation than encoding quality loss, our end-system design always fits recovery needs before trying to get more video data.

Given that the layered streams are ready at the server, what layers to subscribe to is based on three types of information: channel estimation (p,q), probing results (available bandwidth), and observed packet loss rates. Relying on PLC, we can continuously monitor $\hat{p}$, $\hat{q}$ (update every T seconds for enough samples) and congestion packet loss (update every second). Table 1 addresses all possible cases and reactions in our proposed architecture where five moves are allowed in the system. "–" means do not care because the rate is getting lower or low congestion loss is a must for a positive probing. Every time period T, receivers come out with a (p, q) pair, it is mapped to demand FEC, j, for current video layer i conditioning on Eq.(7). If the new j is larger than current j, we categorize it as "worse" in wireless condition. "Better" and "same" are defined analogously. For each condition, we increase FEC, decrease FEC, and change video quality accordingly taking into account the embedded probing results. For example, if the wireless channel is worse and probing for more FEC in the same video layer is failed, we go to path II with more FEC but less quality; if channel does not change and probing for higher rate at the same protection level is positive, it is time to subscribe more video contents. Fig.5 shows a partial view of Fig.2 with five single-step moving directions notated.

Table 1 Possible system reactions

Channel estimation	Probing result	Congestion loss	Action	Action path
Worse	Positive	–	Increase FEC	I
	Negative	"High"		II
Better	–	–	Decrease FEC	III
Same	Positive	–	Change quality	IV
	Negative	High		V

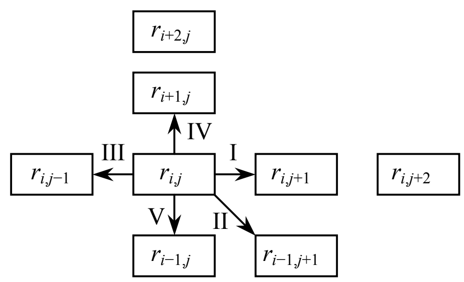

Fig.5 Optimization paths

Therefore, only the paths involving higher rates, I and IV, need probing. If we allow two step jumps, i to i+2 and j to j+2, totally four probing rates are required in every probing interval. In layers $i\neq0$ and $j\neq0$, set every r_{p1} to r_{p4} at $r_{i,j+1}$, $r_{i,j+2}$, $r_{i+1,j}$, and $r_{i+2,j}$ respectively. If i=0, r_{p3} and r_{p4} are $r_{0,j}+r_{1,j}$, and $r_{0,j}+r_{1,j}+r_{2,j}$. If j=0, r_{p1} and r_{p2} are $r_{i,0}+r_{i,1}$, and $r_{i,0}+r_{i,1}+r_{i,2}$. The aggregation rates of probing regions should match target rates. Depending on channel quality, end users select regions 1 and 2 or 3 and 4 for delay trend detection and make decisions to stay or take one of five moves accordingly. The only feedback information needed is the resulting change of layer subscription.

SIMULATIONS

Two experiments were carried to evaluate our proposed architecture in the NS2 network simulator (http://www.isi.edu/nsnam/ns/). Using the topology of Fig.6, we first simulated the embedded probing mechanism under various conditions and then examined the overall layered multicast system in practical scenarios.

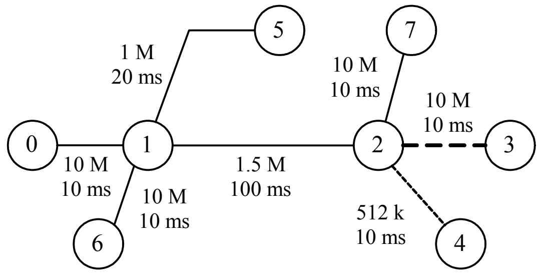

Fig.6 The simulation topology

Embedded probing performance

Conventionally, the target probing rate, which is higher than the current data rate, is attained by sending either dummy packets or, in layered multicast, upper layer packets to increase the overall data rate. The proposed embedded probing, making no change to the overall (long term) data rate to every receiver, may relax some congestion caused by the probing traffic, especially at the lower-than-current rate just before formatted target rate condition.

We first compared the performance in terms of one probe (a train of 50 packets) accuracy between embedded and extra-packet probing under fullsearch delay trend detection. In real-time applications probing tools are utilized for a quick decision for later compensation control schemes. Fig.6 shows all testing streams, Nodes 0 to 3, were 400 kbps probing at 500 kbps and performed a probing with threshold of 0.6 in each test. The cross traffic, Nodes 6 to 7, was configured at a constant bit rate (CBR) during one test along with random dithering. Differences between probing rate and available bandwidth were ±10, 25, 50, 100 kpbs using 100 tests for each rate difference. No wireless loss was introduced.

Fig.7 shows an example of probing and decision making of an embedded stream. The fluctuating curve is the actual available bandwidth calculated at the bottleneck queue. In this +50 case, average available bandwidth was 550 kbps (compared to 500 kbps probing rate). The probing began at the time of symbol "□" and ended in less than 0.5 s with a positive result indicated as "+", which was correct.

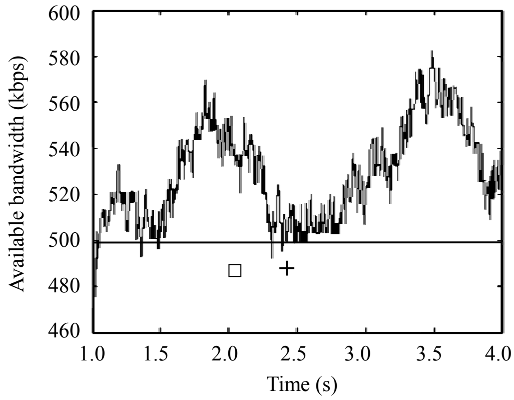

Fig.7 A probing example

One-probe error rates are illustrated in Fig.8. Error rates decreased linearly when difference between available bandwidth and probing rate became larger. Also, the two methods performed very similarly. The embedded one was slightly better at ±10 kbps and only 1% worse than extra-packet method after 50 kbps. In very close cases, such as 10 kpbs, high error rates could be expected due to the fluctuating nature of the Internet traffic although average available bandwidth was steady.

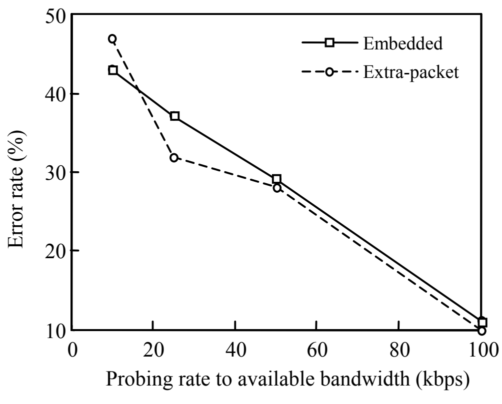

Fig.8 One-probe error rates illustration

System performance

To evaluate the performance of the proposed system, we specified all parameters and created three scenarios. They were: (1) with CBR (same as in Section 4.1) cross traffic, but no wireless loss, (2) with CBR cross traffic, and fixed wireless model (p,q), and (3) with significant cross traffic and wireless channel variation.

We set 5 high quality video layers at cumulative rates 300 k, 500 k, 750 k, 1 M, and 1.2 M bps. Regarding FEC, Table 2 illustrates all details from (p,q) sets to resulting $n_{i,j}$, where V=4 and F=6. All layers were not necessarily to be of the same rate. In addition, two more thresholds for observed congestion, P_c, and wireless, P_w, loss rates were both 1%. The system will immediately go either lower layer or get more protection considering the last probing outcome, if observing loss rates are higher than thresholds. For probing, T was 10 s and 0.8 s for each region.

Table 2 The (p,q) sets and the resulting $n_{i,j}$

j	1	2	3	4	5	6
P_L	3%	5%	10%	15%	20%	30%
p	0.97	0.95	0.90	0.85	0.80	0.70
q	0.03	0.05	0.10	0.15	0.20	0.30
$e_{FEC,i}$	0.005	0.005	0.005	0.005	0.005	0.005
k_i	8	8	8	8	8	8
$n_{i,j}$	10	11	12	14	15	19
$n_{i,j}/k_i$	1.250	1.375	1.500	1.750	1.875	2.375

In the first testing environment, cross traffic of 500 kbps was running from Node 6 to Node 7 (Fig.6). Therefore we expected that the available bandwidth at receiver Nodes 3, 4, 5 were around 1 M, 512 k, and 1 M. Fig.9 shows results at 750 k, 500 k, and 1 M respectively. Nodes 3 and 4 were on layers with rates closest to but no more than available bandwidth. Node 5 was right on the available bandwidth due to its more dedicated link. Also, we can see slight jitters of receiving rates as the result of embedded packet probing which are four regions per period.

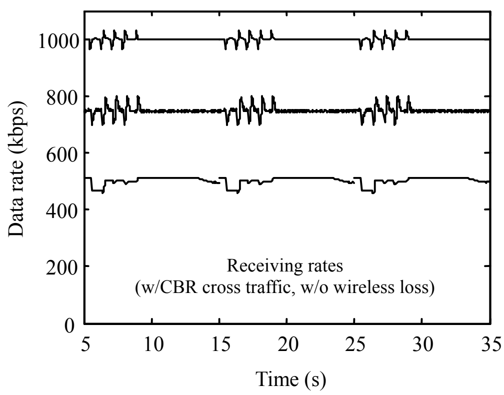

Fig.9 Receiving rates under cross traffic and wireless loss

The simulation setups for the second scenario are shown in Table 3. Still under 500 k cross traffic, we modelled the wireless links 2-3 and 2-4 with (p,q)=(0.85,0.09), different from the constructing values shown in Table 2, and ran the model for 50 s. The decoding error rates e_{FEC} were both significantly lower than packet loss rates. In the last row, $n_{i,j}$ subscribed, 14, for Node 3 was actually larger than the desired value, 13, derived from estimated $(\hat{p},\hat{q})$, due to the lack of $n_{i,j}$=13.

Table 3 Results of wireless model (p,q)=(0.85,0.09)

Parameter	Node 3	Node 4
(i,j)	(1, 4)	(0, 3)
$(\hat{p},\hat{q})$	(0.840, 0.089)	(0.838, 0.086)
P_w	0.068	0.094
P_c	0.002	0
e_{FEC}	0.0006	0.0012
Average rate	822.7	404.2
$n_{i,j}$ (n need)	14 (13)	12 (12)

A full simulation of a series of subscription layers changes at Node 3 during critical network condition is shown in Fig.10. Besides the receiving data rate (solid line), available bandwidth (dotted line) started from 1 M, dropped to 600 k, then went up to 750 k. Average wireless loss (dashed line) increased from 3% to 10%. The 100 ms stream [ignored startup phase (Liu *et al*., 2004)] starting from (i,j)=(3,1) reacted as follows:

(1) Dropped to (i,j)=(2,1) at the 28th second due to congestion loss.

(2) Dropped to (i,j)=(1,1) at the 35th second again due to congestion loss.

(3) Since $(\hat{p},\hat{q})$ had not been updated, automatically subscribed to one more FEC layer for (i,j)=(1,2) due to wireless loss and previous probing.

(4) Subscribed to layer (1,4) based on the new probing and $(\hat{p},\hat{q})$.

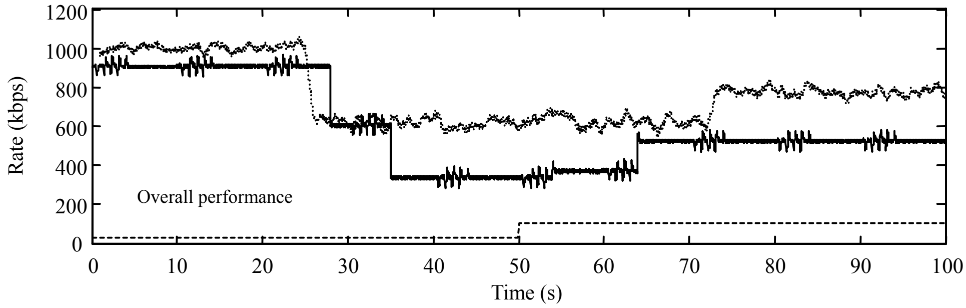

Fig.10 System performance

(5) Stayed on the current layer because layer (2,4) needed 875 kbps which was larger than the 750 kbps available, while the overall decoding error rate was kept low at 0.0037.

CONCLUSION

In this paper, we combined an effective embedded probing scheme, flexible layered FEC structure, and efficient optimization steps to propose a layered multicast system for streaming video over wireless networks. The probing is comparable to traditional style; the system can rapidly adjust to the wired/wireless channel condition so as to protect the content. Tests in real Internet and fine grain layers can be expected in the future.

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