Prioritization-based Layered Multicast for Fixed/Mobile Networks with Fast Convergence and Inter-Session Fairness

Kenichi Kashibuchi, Tarik Taleb, Abbas Jamalipour, Yoshiaki Nemoto, and Nei Kato

Abstract— **Multimedia streaming services are becoming popular in both wired and wireless networks. Layered multicast is a widely accepted approach for streaming multimedia data to a large number of users. Existing layered multicast approaches do not interact well with network dynamics. Indeed, upon a change in network conditions, they require a long time till they can appropriately adjust their data transmission rate. Additionally, they do not achieve fairness when users from different sessions share the bandwidth of a bottleneck link.**

In this paper, we propose a scheme that allows newly-arriving users to promptly converge their data transmission rates to the most optimal rate that best suit the current conditions of the network without degrading the system fairness. The proposed scheme is based on the fact that layered multicast uses priority-based packet dropping policies. In the proposed scheme, two newly-defined packet messages are considered: "low priority join" and "normal join" messages. To join a session, a user first subscribes to all corresponding layers by issuing "low priority join" messages. It then computes packet drops experienced on each layer. If packets of a given layer experience a drop rate higher than a predetermined threshold, the user leaves that layer and all higher layers. The user then "officially" joins the remaining layers by transmitting "normal join" messages. This operation helps users to subscrive to only layers whose aggregate bandwidth fits the current network conditions.

The performance of the proposed scheme is evaluated through computer simulations and is compared against the Receiver-driven Layered Multicast (RLM) scheme. The results show that the proposed scheme achieves appropriate bandwidth utilization from the start of the session. The results demonstrate also that the proposed scheme is effective in managing handover in mobile networks and achieves better Quality of Service (QoS) in heterogeneous mobile environments.

Index Terms— **layered multicast, priority dropping, QoS, hand-off management**

I. INTRODUCTION

Along with the recent advances and on-going improvements in broadband Internet access technologies, a plethora of wide-band multimedia services has appeared. Simultaneous streaming of such services to a potentially large number of users is a challenging task for current unicast-based Internet

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technologies. Multicast is seen as an attractive solution for large-scale streaming of these multimedia services. Whilst multicast is optional in existing Internet Protocol version 4 (IPv4) and is not widely spread, in next generation Internet Protocol, Internet Protocol version 6 (IPv6), all hosts and routers are required to support multicast. Therefore, it seems that realtime streaming will use multicast along with the spread of IPv6.

On the other hand, current wireless accesses, such as Wireless LAN (WLAN), Worldwide Interoperability for Microwave Access (WiMAX), and $3^{rd}-4^{th}$ generation cellular systems ($3-4G$), enable broadband communications in mobile networks. Along with further development of such access technologies, mobile users (laptop computers with wireless interfaces for WLAN or mobile phones) may desire to receive multimedia services while they are on move.

However, the progress of such various Internet access technologies means that users may have to access different networks with different available bandwidths. For instance, a user may access the Internet at nearly $100\,\text{Mbps}$ (*e.g.* via a Fiber To The Home (FTTH) line) and another user may access the Internet at a few Mbps (*e.g.* via a cellular network). In general IP multicast, since the server streams data at only a single rate, multimedia data are transmitted with same contents and same quality (bit rate) for all users. Accordingly, in order to deliver data to as many users as possible, streaming rates should take into account the available bandwidths of users. Indeed, users with enough available bandwidth should receive data at high rates and users with lower available bandwidth should receive data at low quality (low bit rate).

In order to deal with such network diversity, the cumulative layered multicast approach has been proposed [1] [2]. In layered multicast, streaming servers encode the multimedia data in a base layer and several other enhancement layers. They transmit each layer on a different multicast channel. In case of receiver-driven congestion control, receivers assess whether to add or drop layers (*i.e.* to join or leave a multicast group) according to the network conditions. Receivers first join the base layer. They then join the upper enhancement layers in sequence if the network conditions allow. In layered encoding, the more layers users receive, the higher quality they can get. However, since the enhancement layers depend on lower layers, the user cannot decode data without lower layers. If packets of some lower layers are dropped in the

network, packets from higher layer become useless. As a result, the effective packet loss rate becomes high. Therefore, since lower layers provide more important data and should be given higher priority, routers dealing with layered multicast approaches should be equipped with a priority-based packet dropping policy [3][4]. Among receiver-driven layered multi-cast approaches, Receiver-driven Layered Multicast (RLM) is a notable example [2]. A major drawback of RLM consists in the following. In the beginning of a session, users join the base layer and then join the higher enhancement layers one-by-one. Therefore, RLM requires a long time until an appropriate multicast group is found. Additionally, RLM does not have inter-session fairness.

Moreover, when providing multimedia streaming services to mobile users, users' mobility should be considered. Indeed, when a mobile node performs handoff between two base stations (access points), if the two cells have different available bandwidths, the user cannot receive video data at a rate suitable for the new cell immediately after the handoff. This bandwidth disparity can be due to the difference in traffic load in both wireless cells or use of different wireless access techniques with different link speeds, such as WLAN, 3G, 4G, or WiMAX. If the user moves from a higher bandwidth network (*e.g.* WLAN) to a lower bandwidth network (*e.g.* 3G) without changing the total number of layers (*i.e.* bit rate), the new network may be congested and a number of packets may be dropped. This will eventually affect the streaming quality. On the other hand, if the mobile node moves from a lower bandwidth network to a higher bandwidth network, no adjustment to the streaming rate will degrade the service quality in an environment where higher quality can be obtained. This results in a waste of the network resources.

To solve these issues, this paper proposes a scheme that quickly converges to the most appropriate rate while guaranteeing fairness among all sessions. The mechanism of the proposed scheme is extended further to mobile networks by taking into account the mobility of users.

The remainder of this paper is structured as follows. Section II surveys recent research work on layered multicast techniques for multimedia streaming. Section III introduces the proposed scheme. Following this, Section IV describes the simulation setup and discusses the simulation results. Finally, the paper concludes in Section V.

II. RELATED WORK

To tackle issues related to network diversity, two types of multicast streaming can be envisioned: replicated streaming and layered streaming. In the former [5], which is also called as simulcast approach, a video server transmits mul-tiple streams carrying the same data with different qualities (different bit rates). Each stream is multicast on a different multicast channel. Each user joins the multicast group that best satisfies its needs. In the latter, the server divides its data into layers: a base layer and several other enhancement layers. It transmits each layer to a different multicast group. The base layer can be independently decoded and provides the basic-level quality. The enhancement layers can be decoded together with the base layer. Users join the base layer group

and join as many enhancement layer groups as the network condition permits. In general, the layered multicast approach is more advantageous than the simulcast approach in bandwidth utilization, especially in highly heterogeneous networks. A detailed description of the two approaches can be founded at [6].

Transcoding is another technique for multicast streaming [7]. In this approach, video streams are filtered or transcoded to the optimal rate at routers, application gateways, or wireless base stations in the network. A trivial drawback of this approach is that it causes significant load to these network elements.

Among layered multicast approaches, RLM and packet Pair receiver-driven cumulative Layered Multicast (PLM) [8] are notable examples. In RLM, a server transmits data packets of each layer using different multicast group addresses and users join as many layers as their available bandwidths allow. If a user has sufficient bandwidth, he/she joins the next enhance-ment layer. In contrast, if he/she detects network congestions, he/she quits receiving the highest enhancement layer. This procedure is called "join-experiments". At the beginning of a session, RLM assesses its available bandwidth in a slow start-like fashion. That is, users converge to the optimal rate by subscribing to next enhancement layers incrementally until congestion occurs (*i.e.* packet losses). Therefore, it needs a long convergence time. Furthermore, joining higher layers may yield packet losses and may degrade the streaming quality. In addition, since old users do not change their groups, newly-arriving users may not find sufficient bandwidth. This ultimately yields unfairness among users.

To solve these issues, PLM has been proposed. PLM estimates the available bandwidth based on a packet pair mechanism without join-experiments. All packets on all layers are transmitted in pairs. At the beginning of a session, users simply join the base layer. Users then estimate their available bandwidth C seconds ($C = 1$ in [8]) after receiving the first packet pair and check whether they should add or drop layers. Therefore, PLM can converge to the optimal rate faster than RLM, without inducing losses. However, PLM requires that all routers in the network should implement a fair scheduler. Moreover, as the bandwidth estimation is based on an accurate estimation of propagation delays, PLM may fail in the bandwidth estimation in case the network load becomes heavy.

On the other hand, in mobile networks, users freely perform handoff upon changing their point of attachment to the net-work. To guarantee smooth handoff, several approaches have been proposed. In [9] multiple paths are established between the server and a mobile node during handoff. Admittedly this scheme provides smooth handoff for streaming media. Never-theless it uses multiple paths during the time a mobile node exists in the cell overlapping area. As in the case of a handoff between 3G and WLAN, if the distance of the cell overlapping area is long, it is unacceptable to use multiple paths for a long time as it causes redundant transmissions of important data. Although it does not consider the layered multicast approach, the research work outlined in [10] realizes a seamless handoff for streaming video by using two simultaneous connections

through two separate WLANs. The handoff occurrence time is computed based on the delay difference. This approach is however effective for only horizontal handoffs and not vertical ones.

III. PROPOSED SCHEME

This section describes in detail the proposed scheme. It first presents the preconditions that are required for the implementation of our scheme.

A. Preconditions

The proposed scheme is based on the cumulative layered streaming approach, similarly to RLM and PLM. As in any layered streaming mechanism, all routers are assumed to be multicast-capable and to support some priority-based dropping disciplines.

Input multimedia data are encoded into a set of n cumulative layers $(L_1, ..., L_n)$. All subsets $\{L_1, ..., L_i\}_{i \leq n}$ provide the same content. The provided quality increases as index i increases. Since the lowest layer contains the most important data, packets from the lowest layer are assigned higher priority. Indeed, the base layer L_1 has the highest priority (the lowest drop probability) among all layers. Higher layers have lower priority as index i increases. Let B_i and P_i denote the bit rate and the priority of layer L_i, respectively. P_i depends on the aggregate bandwidth of the $(i-1)$ cumulative layers B_{i-1}^c:

$$B_{i-1}^c = \begin{cases} \sum_{k=1}^{i-1} B_k & \text{for } i \in \{2, ..., n\} & \text{(1a)} \\ 0 & \text{for } i = 1 & \text{(1b)} \end{cases}$$

We assign these priorities at the streaming servers using the class field in the IPv6 packet header. Similarly in spirit to the Assured Forwarding (AF) class of the DiffServ CodePoint (DSCP) [11], we consider the setup of twelve priority levels. An example of this priority setting is given in Table I. In this example the values of P_i are calculated as follows:

$$P_i = \begin{cases} 12 - \max\left(0, \lfloor \log_2 \frac{B_{i-1}^c}{10^4} \rfloor\right) & \text{for } i \in \{2, ..., n\} & \text{(2a)} \\ 12 & \text{for } i = 1 & \text{(2b)} \end{cases}$$

In the proposed scheme, two signaling packets are defined: "low priority join" and "normal join". The former is transmitted by users to local multicast routers in order to join a

TABLE I
AN EXAMPLE OF PRIORITY SETTING.

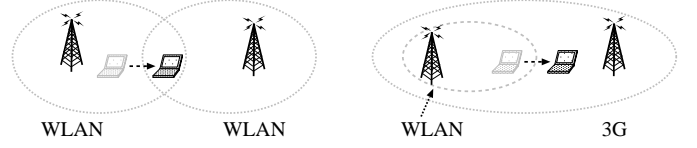| Priority P_i	DSCP	Cumulative Rate B_{i-1}^c
12	AF41	<20 kbps
11	AF42	<40 kbps
10	AF43	<80 kbps
9	AF31	<160 kbps
8	AF32	<320 kbps
7	AF33	<640 kbps
6	AF21	<1.25 Mbps
5	AF22	<2.5 Mbps
4	AF23	<5 Mbps
3	AF11	<10 Mbps
2	AF12	<20 Mbps
1	AF13	≥20 Mbps

Fig. 1.   Two different scenarios of handoff.

session or conduct join-experiments. Upon receiving a low priority join message, multicast routers subsequently decrease the priority of the forwarded streaming packets by one and forward them. In other words, the priority of layer L_i (P_i) is decremented to ($P_i - 1$), not P_{i-1}. These streams with lower priority do not affect the network because their packets are discarded first when a downstream router is congested. A normal join message is transmitted by users to local multicast routers in order to receive packets with normal priority P_i. After receiving a normal join message, multicast routers forward packets without changing their priority.

All multicast routers maintain an internal table with information on source address, multicast group address, outgoing interface identifier, and priority level. In the table, the priority field can be set to either "N" or "L". L indicates that the priority of packets is decreased by one by this multicast router, while N indicates that packets are forwarded without changing their priority.

Additionally, when we apply our scheme to mobile users, it is assumed that wireless cells overlap with each other as shown in Fig. 1. To access two networks in parallel, a mobile node needs to be simply equipped with two wireless interfaces. This assumption is based on the work presented in [12], in which a single physical WLAN interface is used to simultaneously access multiple WLANs. Moreover, along with a further integration of wireless technologies, it will become normal for a mobile node to have an interface that can simultaneously access different types of wireless networks.

B. Description of the proposed scheme

As previously mentioned, a server S encodes multimedia data into a set of n cumulative layers $(L_1, ..., L_n)$. Layer L_i is multicast to group G_i with priority P_i. Lower layers are transmitted with higher priorities (i.e. $P_i > P_{i+1}$).

At the beginning of a session, users transmit "low priority join" messages $\{(S, G_i), i = 1, ..., n\}$ to local multicast routers. Upon receiving a low priority join message (S, G_i), multicast routers first check whether the stream corresponding to L_i is flowing or not (i.e. whether the entry $(S, G_i, *^1, N)$ exists in the internal table). If a multicast router does not have the entry $(S, G_i, *, N)$, it adds the entry (S, G_i, IF, N) to its table and forwards the low priority join message (S, G_i) to its upper multicast router. Here, IF indicates the interface identifier that received the low priority join message. On the contrary, if the multicast router has the entry $(S, G_i, *, N)$, it adds the entry (S, G_i, IF, L) to the table and subsequently decreases the priority of the forwarded packets from P_i to ($P_i - 1$). It then transmits packets to the requesting users

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[1]Asterisk($*$) denotes unspecified interface.

or lower multicast routers. In this manner, multicast routers forward the low priority join message (S, G_i) to upper routers until the expected stream is found.

After receiving the first packet of group G_i, which is transmitted with the lower priority $(P_i - 1)$, users calculate the packet loss rate ρ_i experienced by packets of layer L_i for an interval of time T_D (detection timer). This packet loss rate can be simply calculated from the sequence number available in the packet header of Real-time Transport Protocol (RTP) [13]. If ρ_k, the packet loss rate of L_k, is smaller than a predefined threshold θ, users transmit a normal join message (S, G_k) to the local multicast router. The setting of θ indicates the system tolerance level in terms of packet drops and depends on Forward Error Correction (FEC) redundancy. If ρ_k is above θ, the users stop receiving packets of layer L_k and higher layers by leaving multicast groups $\{G_i, i = k, ..., n\}$.

Upon receiving a normal join message (S, G_k), a multicast router verifies whether it has the entry (S, G_k, IF, L) or not. In case it does not have the entry (S, G_k, IF, L), it forwards the normal join message (S, G_k) to the upper multicast router. On the contrary, when it has the entry (S, G_k, IF, L), it modifies the entry from (S, G_k, IF, L) to (S, G_k, IF, N) to indicate that it is forwarding packets with normal priority P_k.

After a given period of time T_J (join timer), users send low priority join message (S, G_k) to perform the join-experiment operation with lower priority. After receiving the first packet of layer L_k, users calculate the packet loss rate ρ_k on the layer L_k for an interval of time T_D, as in the beginning of the session. When ρ_k is smaller than the threshold θ, users transmit a normal join message (S, G_k) to the local multicast router and join L_k with normal priority. In case ρ_k exceeds θ, users interpret this join-experiment as failure and leave layer L_k. They then multiplicatively increase the join timer T_J as follows:

$$T_J \leftarrow \alpha \cdot T_J \tag{3}$$

where α is a constant. In the proposed scheme, T_D and T_J are set to random values uniformly chosen from $[0.5 \cdot T_D, 1.5 \cdot T_D]$ and $[0.5 \cdot T_J, 1.5 \cdot T_J]$, respectively. This setting aims to avoid simultaneous requests from different users.

C. Explanation of the proposed scheme via an example

In this section, we describe the procedures of the proposed scheme via an example using Figs. 2 and 3. The latter corresponds to the time interval $[t_0, t_1]$ in Fig. 2. In this example, server S encodes multimedia data into a set of four cumulative layers $(L_1, L_2, L_3,$ and $L_4)$. Each layer L_i is transmitted to multicast group G_i with priority P_i $(i = 1, 2, 3, 4)$.

At time $t = 0$, receiver R_1 joins a session from server S and receives data from three layers (i.e. L_1, L_2, and L_3) (Fig. 3(a)). After a while, a new user R_2 joins the same session by issuing low priority join messages $\{(S, G_i), i = 1, 2, 3, 4\}$ to the local multicast router MR_3. Since MR_3 does not have any entry $(S, G_i, *, N)$, it adds the entries $\{(S, G_i, IF_5, N), i = 1, 2, 3, 4\}$ and forwards the low priority join messages $\{(S, G_i), i = 1, 2, 3, 4\}$ to its upper multicast router MR_2. MR_2 then adds the entries $\{(S, G_m, IF_4, L), m = 1, 2, 3\}$ and (S, G_4, IF_4, N), and forwards the low priority
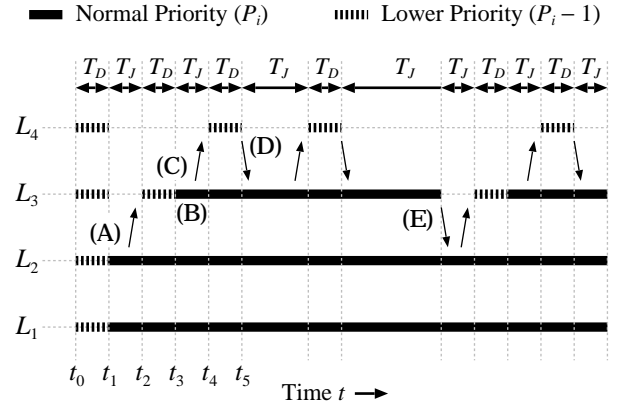

Fig. 2. An example of R_2's layer subscription.

join message (S, G_4) to its upper multicast router MR_1. In response, MR_1 adds the entry (S, G_4, IF_1, L) to its internal table (Fig. 3(b)). It then forwards packets of layer L_4 to receiver R_2 via MR_2 and MR_3 with lower priority. In its turn, MR_2 transmits data of layers L_1, L_2, and L_3 to R_2 via MR_3 also with lower priority (Fig. 3(c)).

We assume that R_2 receives the first packet of group G_i at time $t = t_0$. At $t = t_1 = (t_0 + T_D)$, R_2 calculates the packet loss rate ρ_i on layer L_i. In this example, ρ_3 is assumed to be above the threshold θ. Therefore, the user transmits a normal join messages $\{(S, G_k), k = 1, 2\}$ to MR_3 and leaves L_3 and L_4. Since MR_3 does not have the entries (S, G_k, IF_5, L), it forwards the normal join messages (S, G_k) to MR_2 (Fig. 3(d)). In response, MR_2 resets the entries (S, G_k, IF_4, L) to (S, G_k, IF_4, N). Afterwards, MR_2 subsequently forwards the packets to MR_3 without changing the priority (i.e. with P_k) (Fig. 3(e)).

At $t = t_2 = (t_1 + T_J)$, R_2 transmits a low priority join message (S, G_3) to perform a join-experiment with lower priority and receives data of L_3 with lower priority $(P_3 - 1)$ (Fig. 2(A)). At $t = t_3$, if the calculated packet loss rate ρ_3 is not larger than θ, R_2 transmits a normal join message (S, G_3) and joins L_3 with normal priority (Fig. 2(B)).

In a similar way, at $t = t_4$, R_2 transmits a low priority join message (S, G_4) and receives L_4 with lower priority $(P_4 - 1)$ (Fig. 2(C)). At $t = t_5$, if ρ_4 exceeds θ, R_2 stops receiving layer L_4 and T_J is multiplicatively increased (Fig. 2(D)). After time T_J, R_2 performs again a join-experiment with lower priority. If R_2 infers a possible congestion occurrence from the loss rate of L_3, R_2 leaves L_3 (Fig. 2(E)).

D. Handoff management in the proposed scheme

In the remainder of this section, we explain how the proposed scheme can be applied in mobile networks. This application is similar in spirit to the idea presented in [14] [15]. Fig. 4 illustrates the idea. A mobile node (MN) instantly measures radio strength or link quality. Prior to handoff, MN receives data via base station BS_1 using wireless interface IF_1. When MN moves into the new cell of base station BS_2, a new network address is given to the MN's wireless interface IF_2. When radio strength or link quality through IF_1 goes down

(a) Initial state when R_1 is receiving the session.

(b) R_2 joins the session with lower priority.

(c) R_2 receives stream with lower priority.

(d) R_2 transmits normal join.
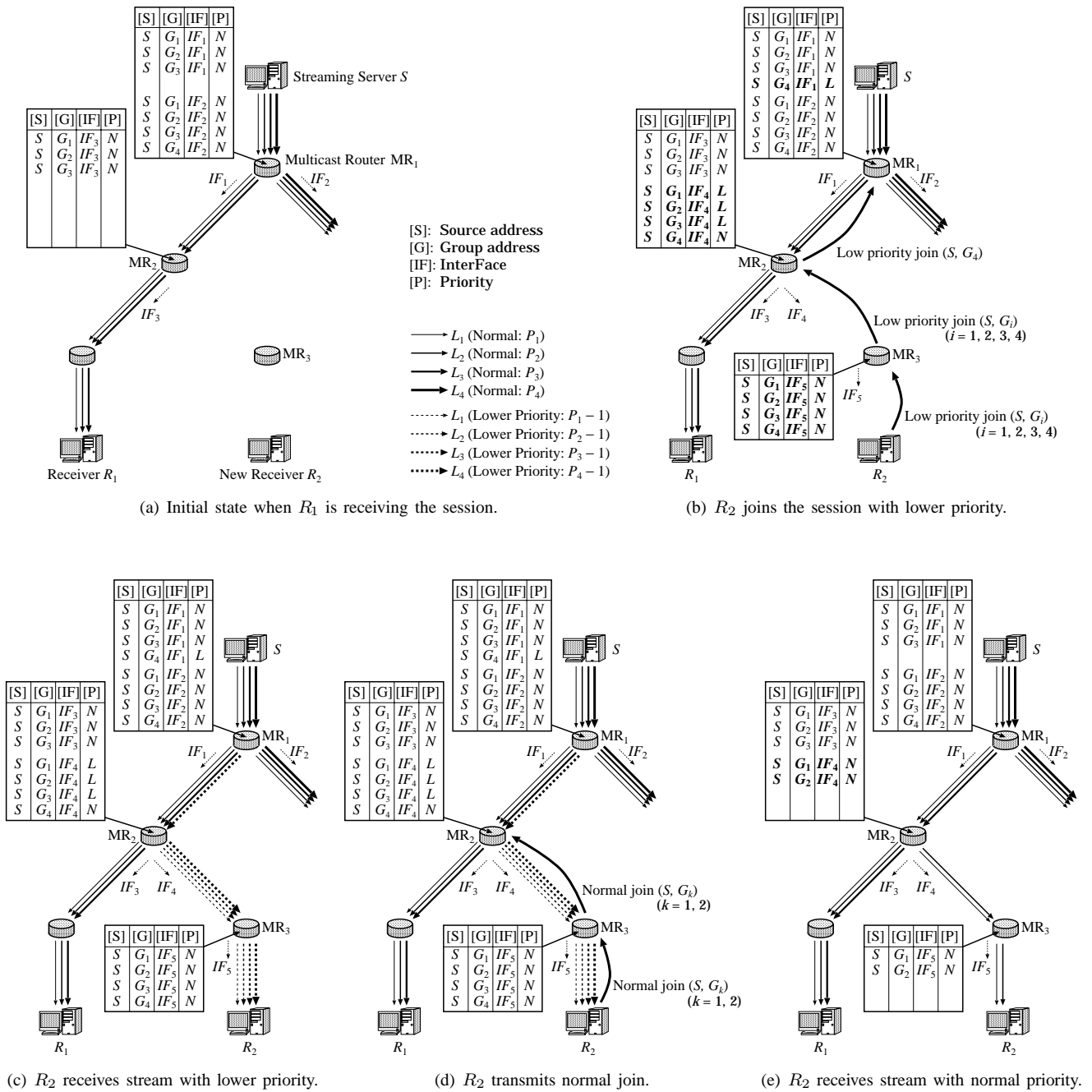
(e) R_2 receives stream with normal priority.

Fig. 3.   Approach of the proposed scheme.

below a predefined threshold, MN transmits low priority join messages $\{(S, G_i), i = 1, ..., n\}$ to the local multicast router through IF_2, as shown in Fig. 4. MN then receives all layers L_i with lower priority $(P_i - 1)$ through IF_2. After receiving the first packet of L_i, MN calculates ρ_i, the packet loss rate of L_i, for a time T_D. If ρ_k exceeds a predefined threshold θ, MN leaves multicast groups corresponding to layer L_k and higher layers. Besides, it transmits normal join messages $\{(S, G_i), i = 1, ..., k - 1\}$ to the local multicast router, in order to join the lower layer with normal priority. After receiving the stream with normal priority, MN leaves the multicast groups through IF_1 and shifts to IF_2. Henceforth, it conducts join-experiment with lower priority through IF_2 every T_J period of time.
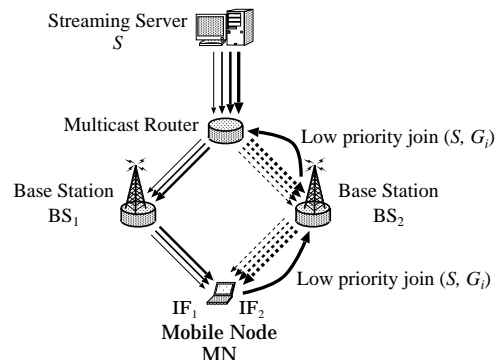

Fig. 4.   Application of the proposed scheme to mobile networks.

IV. PERFORMANCE EVALUATION

In order to verify the effectiveness of our scheme, we implement the algorithm in the Network Simulator (ns-2.28) [16]. We carry out several simulations and compare it against RLM. This section gives a detailed description of the simulation environment and discusses the simulation results.

Similarly to [2], layered sources are modeled as a set of Constant Bit Rate (CBR) streams with fixed packet sizes (500 Bytes). Packets are generated at times defined as follows:

$$T_k = \begin{cases} 0 & \text{for } k = 0 \quad \text{(4a)} \\ T_{k-1} + \Delta + N_k & \text{for } k > 0 \quad \text{(4b)} \end{cases}$$

where Δ is a fixed interval deliberately chosen to meet the target bit rate. N_k is uniformly distributed on $[-\Delta/2, \Delta/2]$. The traffic sources have twelve layers streamed at bit rates B_i [kbps]:

$$B_i = \begin{cases} 32 & \text{for } i = 1 \quad \text{(5a)} \\ 32 \cdot 2^{i-2} & \text{for } i \in \{2, ..., 12\} \quad \text{(5b)} \end{cases}$$

Cumulative bit rates range thus from 32 kbps till 64 Mbps. They are exponentially spaced to simulate highly heterogeneous environments.

In our scheme, all routers need to support some priority disciplines as previously mentioned. In the conducted simulations, we use Weighted Random Early Detection (WRED) [17] for queuing. In WRED, while the average queue size is between a minimum threshold $TH_{\min}$ and a maximum threshold $TH_{\max}$, the arriving packets are either dropped or queued, depending on the packet drop probability as shown in Fig. 5. If the average queue size is larger than $TH_{\max}$, arriving packets are automatically dropped. Mark Probability Denominator (*MPD*) indicates the fraction of packets discarded when the average queue depth is $TH_{\max}$. WRED selectively discards lower priority traffic when a router begins to get congested. It provides also differentiated performance characteristics for different classes of service. We set the parameters of WRED as listed in Table II. Here, it should be emphasized that all routers in the network do not have to set the same parameter. On the other hand, RLM uses drop tail queue [2]. The maximum size of queues is set to 20 packets.

In the proposed scheme, we set both the initial value of T_J (T_J^{init}) and T_D to one second. The multiplicative coefficient α,
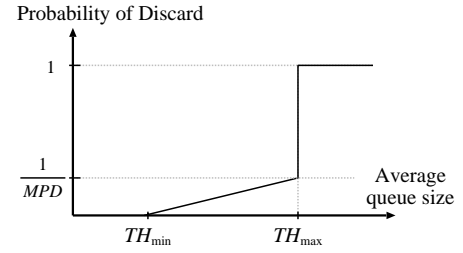

Fig. 5. Packet drop probability of WRED.

used in the computation of join timer (T_J) (equation (3)), is set to two. RLM parameters are set to the same values as in [2]. The used multicast routing protocol is Distance Vector Multicast Routing Protocol (DVMRP). Both wired and wireless links are error-free throughout this paper.

Since the users cannot decode the higher layer without receiving all the lower layers in layered multicast, we use goodput to evaluate the performance of both schemes. Here, goodput is defined as throughput of the layers below the highest layer that users can sequentially achieve with less than $\theta = 10$ [%] of packet loss rate. This metric indicates the number of bytes received and actually decoded by users.

A. Wired network

To better understand the behavior of the proposed scheme, we first conduct simulations using a simple network topology made of only wired links as shown in Fig. 6. We consider a scenario where receivers R_1 and R_2 join a session provided by server S_1 at time $t = 50$ [sec] and $t = 200$ [sec], respectively. Fig. 7 plots the goodput transition of both users. In this scenario, R_1 and R_2 can receive data at a maximum bandwidth of 100 Mbps and 10 Mbps, respectively. Fig. 7 indicates that both receivers succeeded in receiving data at appropriate rates, and that is the case for both schemes. However, it is observed that compared to the proposed scheme, RLM requires a significant time (over 100 sec) till it can make full utilization of the available network resources. The reason behind this performance consists in the fact that unlike RLM, our scheme allows users to estimate the available bandwidth by joining all layers with lower priority. In RLM, users need to conduct several join-experiments till they find out the available bandwidth. During this query for the available bandwidth,

TABLE II

WRED SETTING (UNIT: PACKETS)

Priority	DSCP	WRED parameters [packets]		
		$TH_{\min}$	$TH_{\max}$	*MPD*
12	AF41	11	15	20
11	AF42	10	14	16
10	AF43	9	13	13
9	AF31	8	12	10
8	AF32	7	11	9
7	AF33	6	10	8
6	AF21	5	9	7
5	AF22	4	8	6
4	AF23	3	7	5
3	AF11	2	6	4
2	AF12	1	5	3
1	AF13	0	4	2
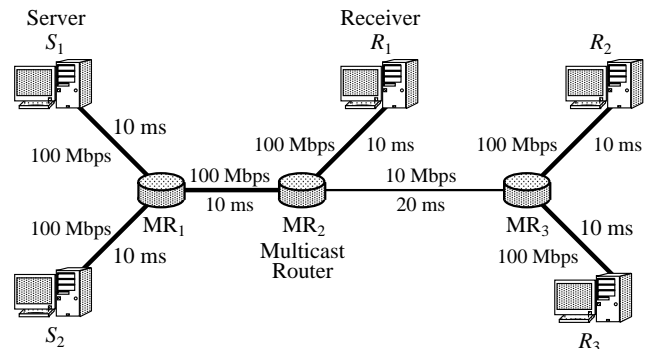
Fig. 6. A wired network topology.

(a) Proposed scheme

(b) RLM
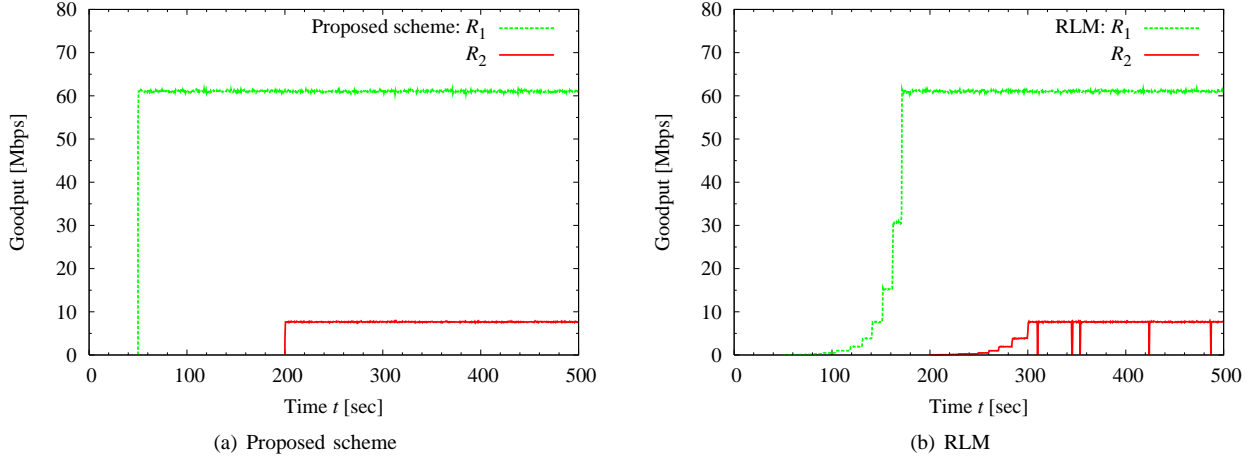
Fig. 7. R_1 and R_2 join the same session at $t = 50$ [sec] and $t = 200$ [sec], respectively.
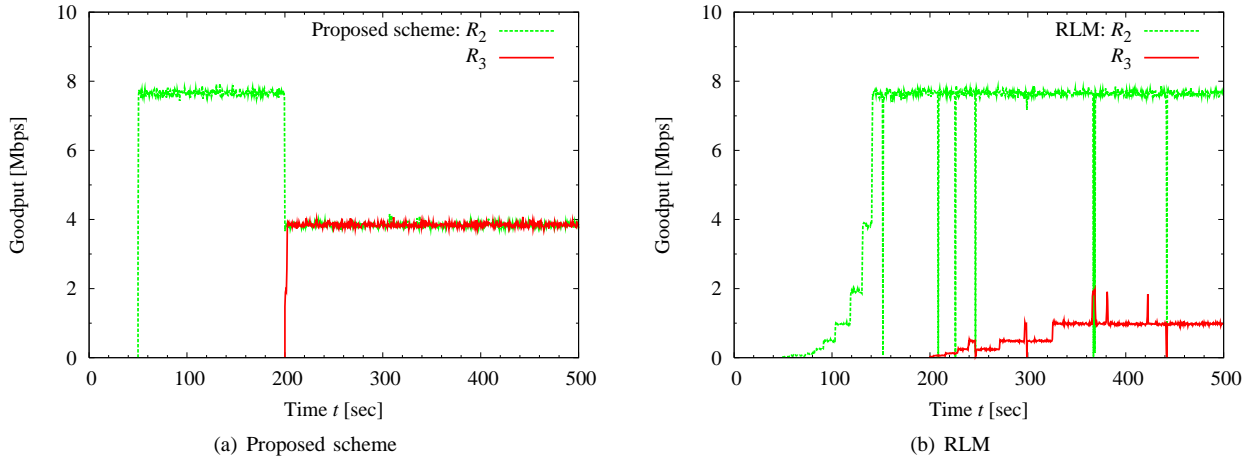

(a) Proposed scheme

(b) RLM

Fig. 8. R_2 and R_3 join two different sessions at $t = 50$ [sec] and $t = 200$ [sec], respectively.

significant packet drops occur. On the other hand, in the proposed scheme, users conduct join-experiments at only one time with lower priority. Accordingly, users can receive data at appropriate bit rates from the beginning of the session.

To investigate the performance of our scheme when two different sessions share the same bottleneck link, we consider a scenario where R_2 joins a session from server S_1 at time $t = 50$ [sec] and R_3 joins another session from another server S_2 at $t = 200$ [sec]. The results are plotted in Fig. 8. The figure demonstrates that RLM results in strong unfairness between the two users. Indeed, it shows that R_2 conquers most of the available bandwidth when RLM is in use. On the other hand, in case of the proposed scheme, the figure indicates that both users share the network resources in a fair manner. This performance is attributable to the priority dropping mechanism of the proposed scheme. Observe also that both users experience some oscillations in their goodput transition in case of RLM whereas their goodputs remain stable in case of the proposed scheme. The reason behind this performance underlies beneath the fact that users perform join-experiments with lower priority in the proposed scheme.
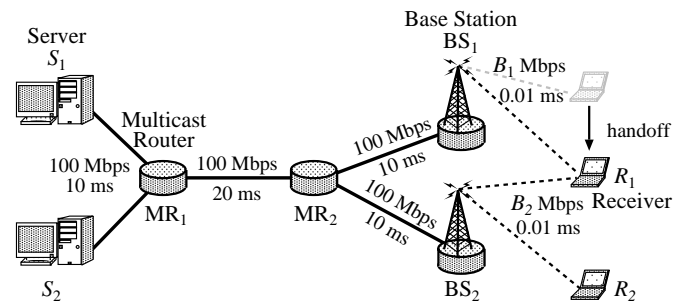

Fig. 9. Hybrid wired/wireless network topology.

B. Hybrid wired/wireless network

In the remainder of this section, we investigate the performance of our scheme in mobile environments. At the beginning of the simulation, a mobile node R_1 resides in the cell of BS_1. It then moves into the cell overlapping area of BS_1 and BS_2, and performs handoff as shown in Fig. 9. In this simulation, two scenarios are envisioned: the mobile node moves from a lower bandwidth cell to a higher bandwidth cell or vice versa. Figs. 10 and 11 graph the goodput
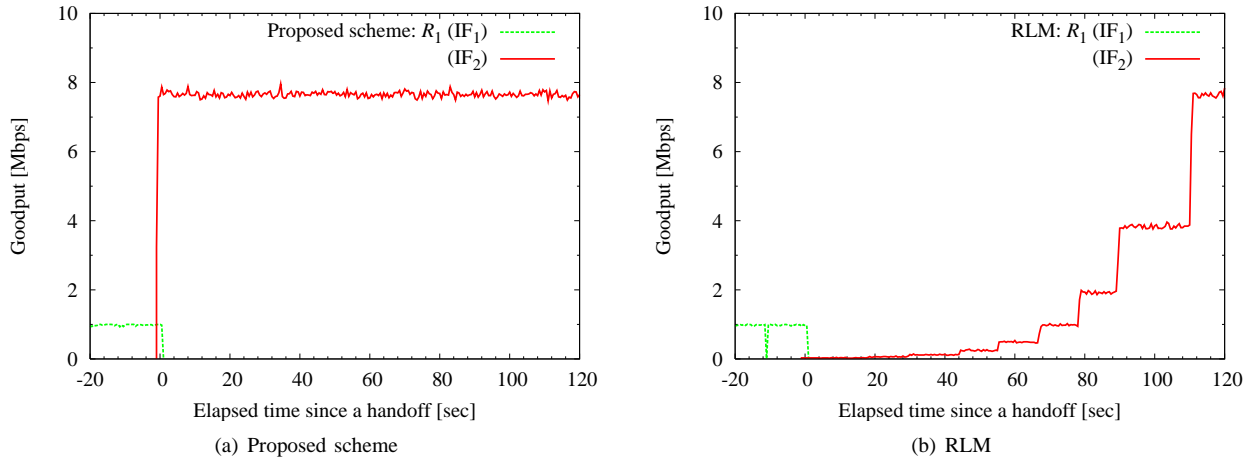
Fig. 10. R_1 performs handoff from 1 Mbps to 11 Mbps when only a single session exists.
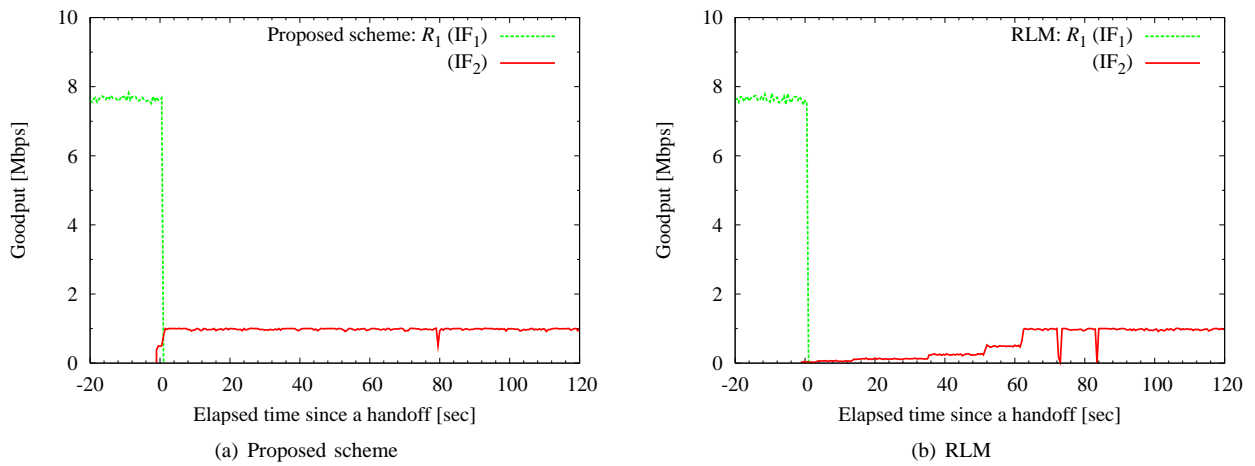

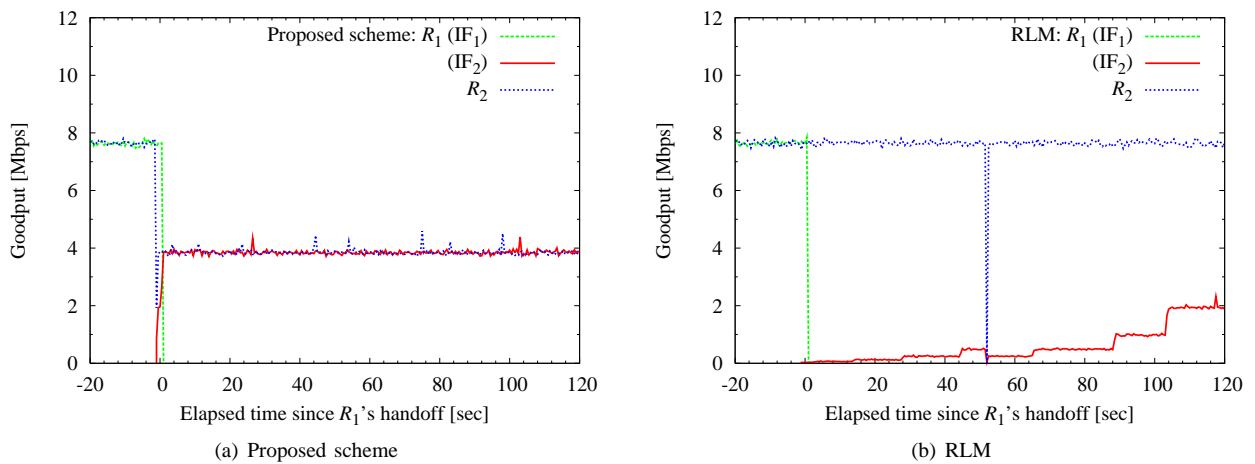Fig. 11. R_1 performs handoff from 11 Mbps to 1 Mbps when only a single session exists.

Fig. 12. R_1 performs handoff from BS_1 to BS_2 and shares 11 Mbps with R_2 after handoff.

transition when receiver R_1 performs handoff from 1 Mbps network to 11 Mbps network (*i.e.* $B_1 = 1$ [Mbps], $B_2 = 11$ [Mbps]), and from 11 Mbps to 1 Mbps, respectively. These graphs demonstrate that the proposed scheme enables R_1 to receive the stream at rates suitable to the available bandwidth immediately after handoff. On the other hand, RLM needs longer convergence time as it adds the upper layers one-by-one.

Next, we evaluate the performance of the proposed scheme when receiver R_1 performs handoff and shares the available bandwidth of the new network with another user joining a different session. At the beginning of the simulation, receivers R_1 and R_2 reside in cells of BS_1 and BS_2, respectively. R_1 then moves into the cell overlapping area, performs handoff, and starts sharing the cell resources with R_2. Bandwidth of the two cells is set to 11 Mbps. Fig. 12 shows the goodput transition of both receivers. It is clear that the proposed scheme succeeds in fairly dividing the bandwidth of the new network between the two receivers immediately after R_1's handoff. However, in case of RLM, R_2 remains conquering most of the network resources. This unfairness issue can be explained in the same way as when users share the bandwidth of the same bottleneck link.

V. CONCLUSION

In this paper, we proposed a layered multicast streaming scheme. The proposed scheme exploits the fact that layered multicast uses priority-based packet dropping policies. In our method, a user joins all layers with lower priority at the beginning of a session. He/she then calculates the packet loss rate on each layer. If the packet loss rate on a certain layer exceeds a predefined threshold, the user leaves that layer and higher layers. The user then receives packets of lower layers with normal priority. By so doing, users can converge fast to the optimal bit rate (most suitable rate to the network conditions) at the beginning of a session. Additionally, as they conduct join-experiments with lower priority at regular time intervals, users are always aware of the bandwidth availability of the underlying network without dropping packets of lower layer.

The performance of the proposed scheme was investigated through several simulations. Performance evaluation relied on computer simulation. The obtained results revealed that our scheme enables users to fairly share the bandwidth with other users. It achieves appropriate streaming rate from the beginning of the session. The efficiency of the proposed scheme in mobile networks is also confirmed. Indeed, our scheme guarantees smooth handoff and enables mobile nodes to receive streams at rates appropriate to network conditions immediately after handoff occurrence. The good fairness of the proposed scheme in mobile environments is also verified.

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