

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

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**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 subscribe 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

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<sup>rd</sup>–4<sup>th</sup> 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 100Mbps (*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

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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 ( $L_1, \dots, L_n$ ). All subsets  $\{L_1, \dots, L_i\}_{i \leq n}$  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  $B_{i-1}^c$ :

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

$$(1b)$$

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:

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

$$(2b)$$

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	$\geq 20$ Mbps

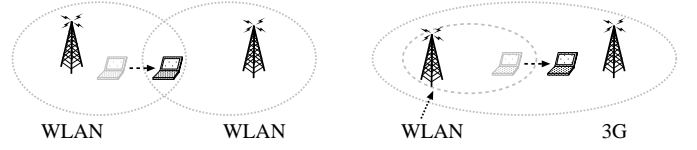


Fig. 1. Two different scenarios of handoff.

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 ( $L_1, \dots, L_n$ ). 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  $\{(S, G_i), i = 1, \dots, n\}$  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  $(S, G_i, *, N)$  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

<sup>1</sup>Asterisk(\*) denotes unspecified interface.



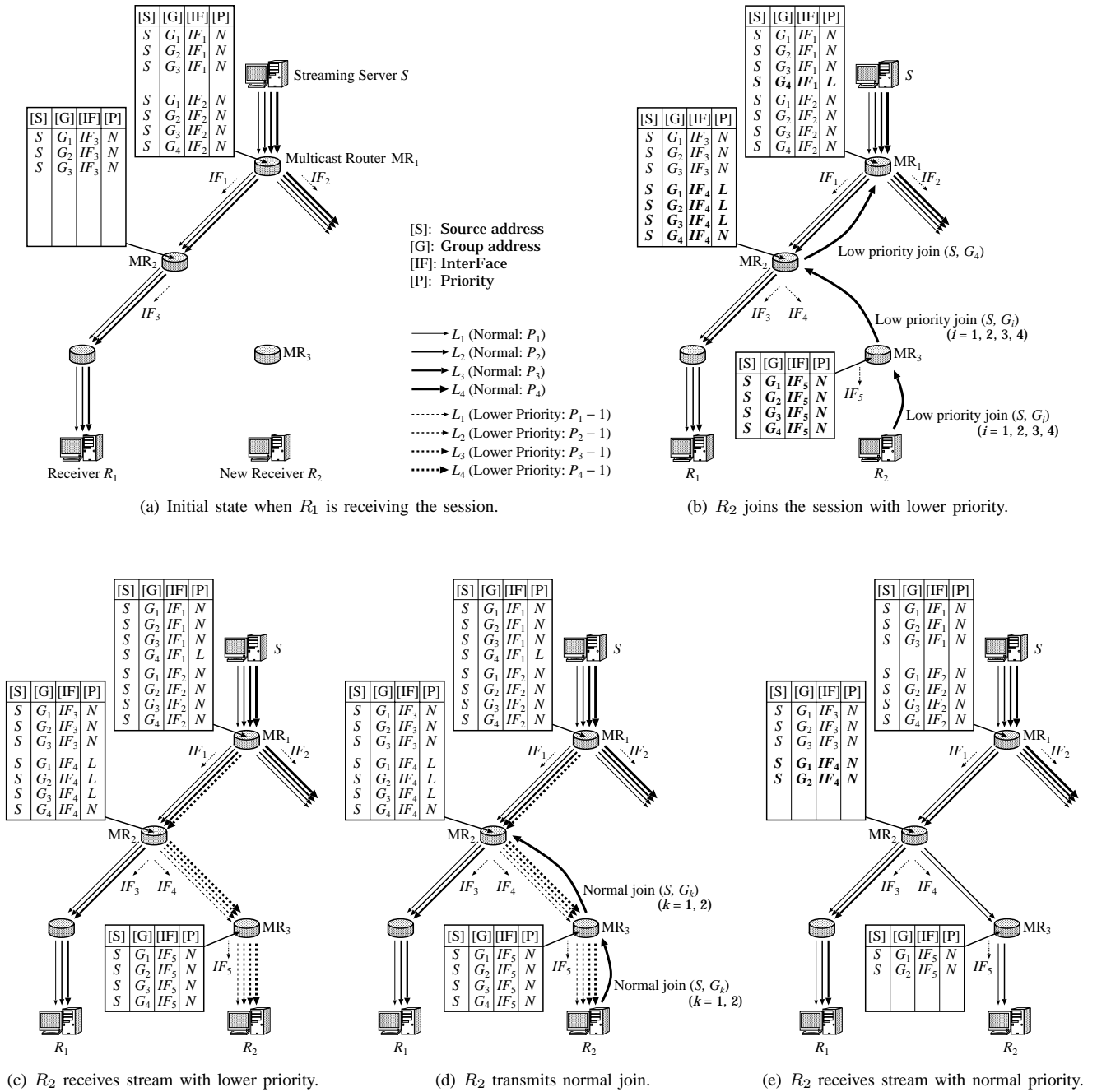


Fig. 3. Approach of the proposed scheme.

below a predefined threshold, MN transmits low priority join messages  $\{(S, G_i), i = 1, \dots, n\}$  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  $\rho_i$ , the packet loss rate of  $L_i$ , for a time  $T_D$ . If  $\rho_k$  exceeds a predefined threshold  $\theta$ , MN leaves multicast groups corresponding to layer  $L_k$  and higher layers. Besides, it transmits normal join messages  $\{(S, G_i), i = 1, \dots, k - 1\}$  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.

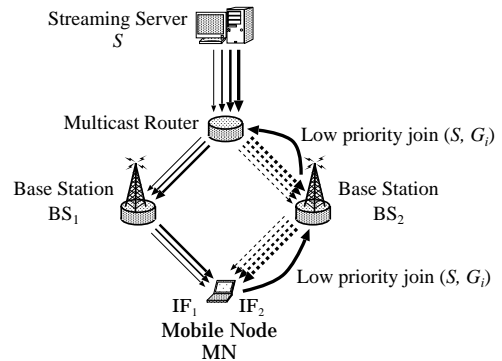


Fig. 4. Application of the proposed scheme to mobile networks.



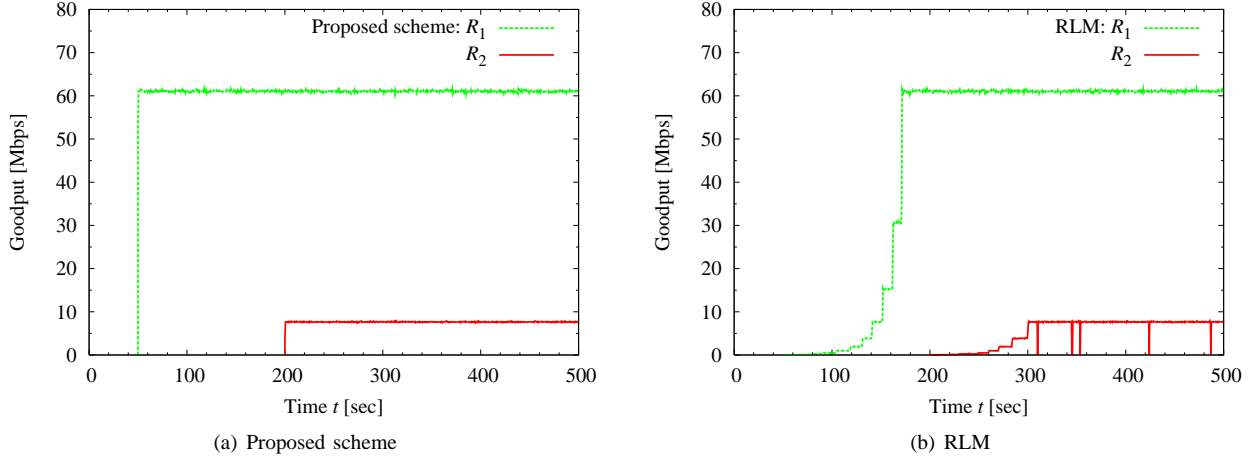


Fig. 7.  $R_1$  and  $R_2$  join the same session at  $t = 50$  [sec] and  $t = 200$  [sec], respectively.

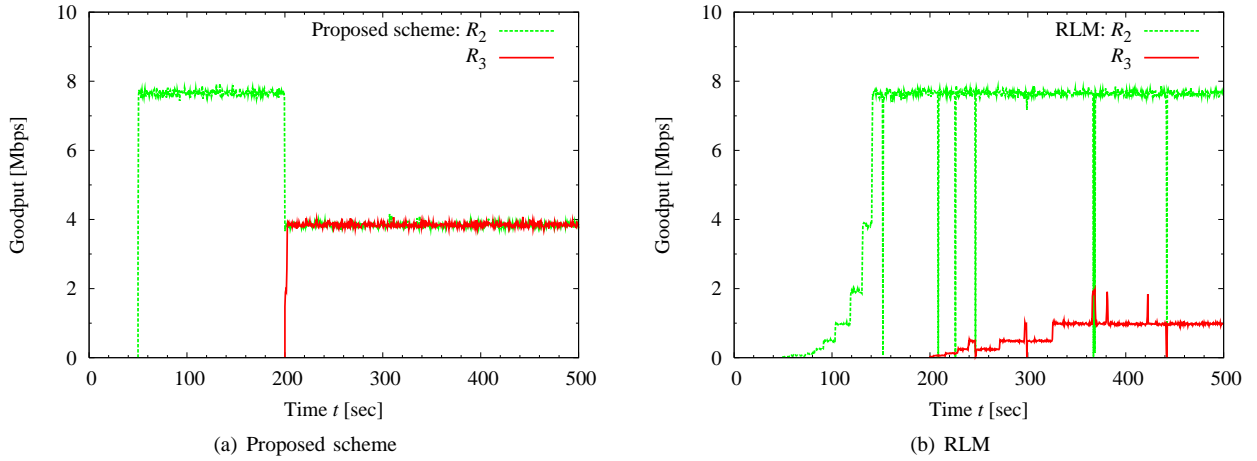


Fig. 8.  $R_2$  and  $R_3$  join two different sessions at  $t = 50$  [sec] and  $t = 200$  [sec], respectively.

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.

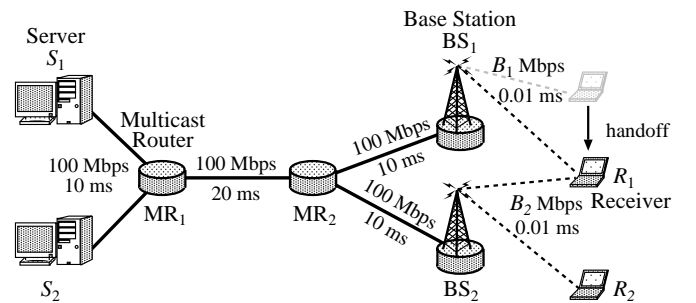


Fig. 9. Hybrid wired/wireless network topology.

### 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





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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