Bandwidth Aggregation-Aware Dynamic QoS Negotiation for Real-Time Video Streaming in Next-Generation Wireless Networks

Juan Carlos Fernandez, *Student Member, IEEE*, Tarik Taleb, *Member, IEEE*, Mohsen Guizani, *Fellow, IEEE*, and Nei Kato, *Senior Member, IEEE*

***Abstract*—In next generation wireless networks, Internet service providers (ISPs) are expected to offer services through several wireless technologies (e.g., WLAN, 3G, WiFi, and WiMAX). Thus, mobile computers equipped with multiple interfaces will be able to maintain simultaneous connections with different networks and increase their data communication rates by aggregating the bandwidth available at these networks. To guarantee quality-of-service (QoS) for these applications, this paper proposes a dynamic QoS negotiation scheme that allows users to dynamically negotiate the service levels required for their traffic and to reach them through one or more wireless interfaces. Such bandwidth aggregation (BAG) scheme implies transmission of data belonging to a single application via multiple paths with different characteristics, which may result in an out-of-order delivery of data packets to the receiver and introduce additional delays for packets reordering.**

The proposed QoS negotiation system aims to ensure the continuity of QoS perceived by mobile users while they are on the move between different access points, and also, a fair use of the network resources. The performance of the proposed dynamic QoS negotiation system is investigated and compared against other schemes. The obtained results demonstrate the outstanding performance of the proposed scheme as it enhances the scalability of the system and minimizes the reordering delay and the associated packet loss rate.

***Index Terms*—Bandwidth aggregation, mobile video streaming, packet scheduling, QoS negotiation, wireless networks.**

I. Introduction

ALONG with the ever-growing demand for real-time multimedia services (e.g., video streaming, video conferencing, online interactive games, and IPTV) that require high-quality quality-of-service (QoS) support, such as guaranteed bandwidth, delay, jitter and error rate, ISPs are required to extend their ranges of services to allow users to utilize these multimedia applications with certain level of QoS. Currently the service level agreement (SLA) is made by the user via contract with the ISP. The SLA is static for the contract period and is applied equally to the overall traffic between the end-user and the network, regardless of the different service levels required by different applications [1].

To ensure an efficient provision of real-time video applications in wireless networks, mobile users should be able to dynamically negotiate their QoS requirements, represented by the service level specifications (SLSs), with the access network. This negotiation should be performed per session. The network operator must guarantee the negotiated SLS during the entire course of the session, which is a challenging task because of the mobility of users. In addition, transmission of high quality video requires high bandwidth that is difficult to guarantee because of the resource constraints in current wireless networks. However, mobile users equipped with multiple wireless interfaces, in combination with ISPs providing services through different wireless technologies, ought to make simultaneous use of these interfaces to connect to the network and aggregate the available resources via these interfaces. Thus, users can enhance the perceived quality of their applications.

Two scenarios can be envisioned for a mobile terminal depending on its role as a sender or receiver. The latter case is the most common one where the sender can be a fixed multimedia server. In the proposed system, a mobile terminal negotiates its service level with the network upon performing handoff to a new access point. A set of mechanisms are applied to define a service level for the mobile terminal [2]. If the terminal acts as a sender, it then adjusts its transmission rate to the agreed bandwidth. In case the terminal operates as a receiver (which is the case that we consider in this paper), it notifies the corresponding source of the agreed service level and the sender accordingly adjusts its streaming rate [3]. A proxy in the middle of the network, between the wireless and wired parts of the network, runs the scheduling operation to ensure that packets are transmitted in order to the terminal via the different wireless interfaces.

The contribution of this paper is fourfold. Firstly, we propose a dynamic QoS negotiation system for real-time video applications that enables mobile users to negotiate their desired service levels and to reach them by using their available interfaces. The proposed system supports initial negotiation, renegotiation in a small time scale, and mobility of users. Secondly, we demonstrate the ability of the proposed mechanism in supporting QoS continuity while users are on the move. Thirdly, we demonstrate the need for controlling the bandwidth aggregation mechanisms

Manuscript received September 30, 2008; revised April 07, 2009. Current version published September 16, 2009. The associate editor coordinating the review of this manuscript and approving it for publication was Prof. Aggelos K. Katsaggelos.

J. C. Fernandez and N. Kato are with the Graduate School of Information Sciences (GSIS), Tohoku University, Sendai 980-8579, Japan (e-mail: carlos@it.ecei.tohoku.ac.jp; kato@it.ecei.tohoku.ac.jp).

T. Taleb is with NEC Europe Ltd., Heidelberg, Germany (e-mail: tarik.taleb@nw.neclab.eu).

M. Guizani is with the Department of Information Science, Kuwait University, College for Women, Safat 13060, Kuwait (e-mail: mguizani@ieee.org).

Color versions of one or more of the figures in this paper are available online at http://ieeexplore.ieee.org.

Digital Object Identifier 10.1109/TMM.2009.2026086

to guarantee an efficient and fair use of the network resources. Finally, we propose a packet scheduling strategy to cope with the packet reordering issue in multi-path video transmission, which minimizes the reordering delay at end-terminals and its associated packet loss rate.

The remainder of this paper is organized in the following fashion. Section II reviews the existing work related to service level negotiation, uses of multiple interfaces in wireless networks, and multi-path scheduling strategies. Section III describes the proposed scheme. Section IV evaluates the performances of the proposed schemes. Finally, concluding remarks are presented in Section V.

II. Related Work

Several protocols for service level negotiation have been proposed, such as common open policy service for service level specification (COPS-SLS) [4], resource negotiation and pricing protocol (RNAP) [5], and service negotiation protocol (SrNP) [6]. Furthermore, two protocols have been proposed to support QoS negotiation in wireless networks by considering users' mobility, namely, QoS generic signaling layer protocol (QoS GSLP) [7] and dynamic service negotiation protocol (DSNP) [8]. QoS GSLP uses mobility and traffic pattern prediction to prefigure the next point of attachment of a mobile user and delivers the SLS to that access point, reducing thereby the handoff negotiation delay. This method highly increases the complexity of the system and makes it shortly scalable. DSNP informs all neighboring base stations (BSs) of the current BS of the SLS of a user. Each time the user negotiates for a new SLS, the QoS global server (QGS) delivers the new SLS to the current BS and its neighbors. In this fashion, all potential points of attachment after the users' handoff already have information on the users' SLS. This mechanism presents scalability problems in terms of signaling overhead and data storage. In the mechanism presented in [9], when a mobile station (MS) performs handoff, the new BS consults the previous BS for the SLS of the MS. This approach is highly scalable, as the MSs' SLSs are delivered just to the new BSs. However, the handoff negotiation delay is incremented by the roundtrip time between the two adjacent BSs. Another mechanism to inform BSs of users' SLSs is mentioned in [10]. In this mechanism, after a user negotiates its service level with the network, the QGS delivers an SLS token to the user. The token contains all details associated with the service level and traffic specifications. The token is encrypted and can be decrypted by only BSs in a given domain. When a user performs handoff within the domain, it sends the SLS token to the new BS, which decrypts it and performs traffic conditioning. In this way, the handoff negotiation delay is minimized as no communication is required between the MS and the QGS or between BSs. However, this method gives rise to some security concerns as malicious users can obtain the SLS token from genuine users and steal their service levels. A detailed survey on the above mentioned protocols can be found in [11].

Mobility of users in wireless networks introduces some issues to the network administrator as users need to keep their connections active when they move from one access point to another. The handoff process can be in homogeneous networks (horizontal handoff) [12], [13] , or between heterogeneous networks (vertical handoff) [14], [15].

The use of multiple interfaces in wireless networks was firstly addressed by stream control transmission protocol (SCTP) [16], which uses multiple interfaces to ensure high reliability. Some variations of the original SCTP have been presented in [17]–[19], which are able to distinguish among losses due to congestion and radio channel failures to better select the path for data transmission. The work presented in [20] provides mechanism to monitor the one-way delay variation throughout the available paths. A variation of SCTP was developed to provide bandwidth aggregation, particularly for the provision of real-time applications to wireless mobile users. Load-sharing SCTP (LS-SCTP) [21] introduces a new functionality to SCTP by involving all available paths in data communication and aggregating their bandwidth to improve the performances of real-time applications. Multimedia multiplexing transport protocol (MMTP) [22] is a link-layer aware protocol designed for transmitting multimedia data over mobile systems. It makes simultaneous use of every available communication channel.

Bandwidth aggregation involves multiple paths in the data transmission and raises the need for an adequate distribution of data load over several paths. Moreover, packets (belonging to the same application) transmitted through different paths may experience different latencies, resulting in out-of-order delivery to the final destination [23]. Packets arriving out-of-order need to be stored in a buffer until they can be delivered to the application in a proper order. If they arrive later than their playback time, they are discarded. To cope with packet reordering in multi-path environments, several scheduling strategies have been proposed, most of which are based on round robin mechanisms, such as weighted fair queuing (WFQ), weighted round robin (WRR), weighted interleaved round robin (WIRR), and surplus round robin (SRR) algorithms. The earliest delivery path first (EDPF) scheme [24] is currently one of the most promising scheduling algorithms. It is based on the estimation of the delivery times of packets through each available path. It schedules packets via the path with the earliest delivery time. Another interesting work is presented in [25] that faces the problem of joint path selection to optimize the streaming of stored video sequences on multipath networks. In the following section, we describe our proposed scheme in detail.

III. Bandwidth Aggregation-Aware Dynamic QoS Negotiation System

The proposed dynamic QoS negotiation system allows users to define and request their desired service levels, which can be accepted or rejected by the service negotiation entity. In case of rejection, the QGS proposes a different service level to MS. MS accepts or rejects such an offer. Moreover, at any time MS can upgrade or downgrade a previously negotiated service level. On the other hand, the QGS may require degrading the service level when resources become scarce [2]. Disregarding the situation, a new SLS is established when both MS and the QGS receive positive responses from each other. Some important aspects of the QoS negotiation system are as follows.

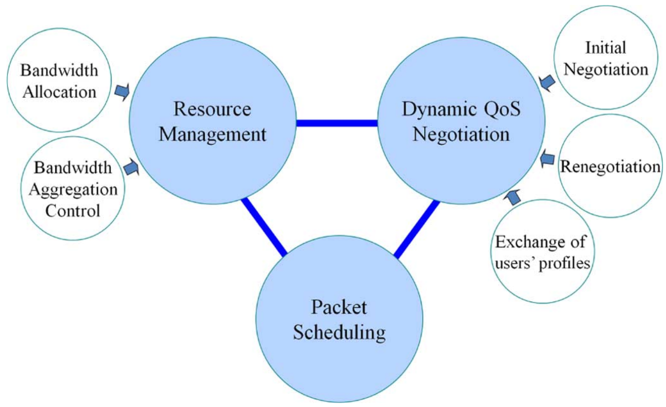


Fig. 1. Major elements of the bandwidth aggregation-aware QoS negotiation system.

1) Even when an MS is able to negotiate SLSs through all its interfaces, each SLS is associated to one specific interface. Thus, MS should perform an initial negotiation through each interface it attempts to use to connect to the network.

2) An MS handoff refers to the event where MS changes its point of attachment in one specific wireless network technology. It is unlikely that an MS performs handoff through more than one interface at the same time.

3) MSs are able to perform handoffs between BSs of the same wireless technologies only.

Fig. 1 shows the major elements of the bandwidth aggregation-aware QoS negotiation system. The *resource management* module consists of two components: *1) bandwidth allocation*, a mechanism that divides the time-slot among users based on their agreed service level and *2) bandwidth aggregation control*, a mechanism that ensures the fairness and scalability of the system. This is achieved by controlling the total amount of bandwidth, assigned to each user through all the available interfaces, not to exceed the bandwidth indicated in the user's SLA.

The *dynamic QoS negotiation* module includes the following components: *1) initial QoS negotiation*, a procedure that enables users to obtain the service levels demanded by their applications, *2) renegotiation*, applied when a user desires to renegotiate an already established SLS during the communication course, and *3) exchange of users' profiles*, a set of fast and scalable mechanisms used by the system to securely exchange information on users' profiles among access points. The *packet scheduling* module aims to minimize the reordering packet delay at the receiver as well as the associated packet loss rate. Packet reordering and packet drops occur due to the fact that each path may have different capacity and different propagation delay.

A. Envisioned Architecture

The QoS negotiation takes place at the IP layer and is based on differentiated services (DiffServ). The components of the envisioned architecture for bandwidth aggregation-aware dynamic QoS negotiation are schematically shown in Fig. 2. The figure depicts one of the multiple domains administered by different ISPs and offering services through different wireless technologies. The domain consists of a QGS, an authentication, authorization and accounting (AAA) server, a network proxy (NP), a number of BSs, and a population of MSs.

QGS is the entity for service level negotiation; it decides the admissibility of service requirements based on the service level that the user is allowed to receive as well as the current available resources in the network [2]. The AAA server is used to confirm that a user, who is requesting a specific service level, is permitted to obtain it.

BS is the entity where SLSs are applied. It uses DiffServ to enforce different service levels to users. BSs constantly inform the QGS about their local resource availability and receive SLSs of users for traffic conditioning. BSs control access on the wireless channel by dividing the bandwidth into transmission slots. The key to providing QoS to mobile users consists in the assignment and guarantee of a time-slot for data transmission for each user during the encourse of its session.

NP is the key entity to schedule data packets via multiple paths; it should be informed of the time-slot assigned to a user by each BS through which the user negotiated SLS. NP is responsible for maintaining global information about the aggregated bandwidth of users, such as negotiated wireless paths for data transmission and time-slots assigned through each path.

B. Resource Management

1) Bandwidth Allocation: As mentioned earlier, to guarantee the negotiated service level to users, BSs implement a time-slot division approach, which allows each MS to use the total bandwidth of the channel during the time-slots exclusively assigned to the MS. Thus, BSs avoid collisions in the wireless channel and provide users with strict QoS, rather than relative QoS where data packets belonging to users with a similar service level compete among them to get access to the link. Each MS is allocated a specific period of time to use the wireless link. The time-slot assigned to an MS depends on the amount of bandwidth specified in its SLS. Thus, the time-slot size varies from one MS to another. It is calculated as follows:

$$\delta_i = \frac{BW_{SLS_i}}{BW_T} * \Delta \tag{1}$$

where BW_{SLS_i}, BW_T, and Δ denote the bandwidth specified in the SLS of the MS_i, the total bandwidth of the wireless link from the BS, and the time-slot interval, respectively. The time-slot interval is the continuously repeated time period in which all MSs will be served.

Let us define slot synchronization delay as the time a packet has to wait at the BS queue since its arrival time until the beginning of the time-slot of its corresponding MS. Burst delay is in turn defined as the time a packet has to wait at the BS queue due to the bursty nature of real-time traffic. These packets have to wait for later time-slots to be transmitted. In addition, if the next packet to be transmitted is too large to be processed during the remainder of the time-slot, the packet will wait for the time-slot of its corresponding MS in the next round. This also means that the current time-slot will have an unused remainder at the end.

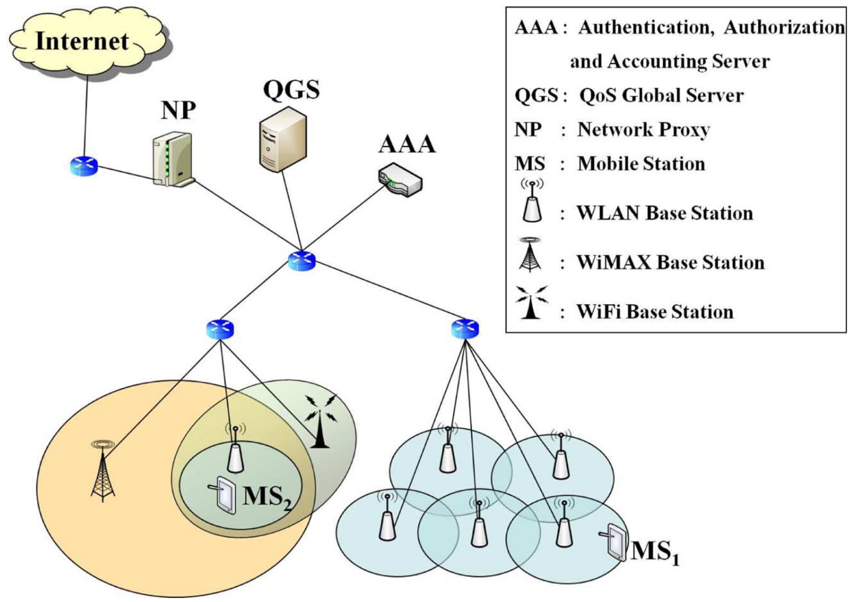


Fig. 2. Envisioned architecture for dynamic QoS negotiation in next-generation wireless networks.

Based on the above discussion, an appropriate size of the time-slot interval should be carefully determined to decrease the synchronization delay and the burst delay, while keeping the network utilization high. In [26], it was demonstrated that the unused bandwidth at the end of time-slots does not depend on the time-slot size. It only depends on the distribution of packet sizes. The same work also demonstrated that

$$U = \frac{\Delta - E(R)}{\Delta}; E(R) \leq \Delta \quad (2)$$

where U and $E(R)$ denote the network utilization and the expected value of the unused portion of the time-slot of an MS, respectively. Equation (2) indicates that increasing the time-slot interval length yields better network utilization. On the other hand, decreasing the time-slot interval size lowers the packet delay. Thus, the size of the time-slot interval should be appropriately selected to keep a balance between the network utilization and the communication delay. In our previous work [27], we empirically demonstrated that $(\Delta = 0.1\ \mathrm{s})$ achieves a low delay as well as high utilization of the bandwidth.

2) Bandwidth Aggregation Control: Mobile stations are able to connect to the network via multiple interfaces simultaneously. This introduces a new issue related to SLA management. When an MS negotiates the service level for its traffic, the QGS confirms from the AAA server that the MS is allowed to receive the requested service level. The AAA server verifies if the requested service level exceeds the agreed SLA of the MS or not. Since, in bandwidth aggregation scenarios, MSs are able to negotiate SLSs through several interfaces, such a verification method is not suitable, as the MS can negotiate SLSs through all its interfaces, and the network separately verifies each SLS:

$$BW_i \leq BW_{SLA}; \quad 1 \leq i \leq n \quad (3)$$

where BW_i, BW_{SLA}, and n denote the bandwidth negotiated through the i interface, the bandwidth specified in the SLA of the MS, and the number of interfaces of the MS, respectively. Thus, an MS may obtain up to n times the bandwidth indicated in its SLA

$$\sum_{i=1}^{n} BW_i \leq n \times BW_{SLA}. \quad (4)$$

On the other hand, some other MSs may get their SLS requests rejected due to the unfair service level assignments.

To guarantee an efficient and fair use of the network resources among all competing MSs, in a bandwidth aggregation system, the network operator should consider using some or all available interfaces to ensure the service quality in case a single SLS (provided by a single interface) does not meet the pre-agreed SLA. In the same manner, if the aggregate SLSs provided by multiple interfaces exceed the pre-agreed SLA, the network operator should hinder the user from using some of the interfaces to ensure a fair utilization of network resources among all active MSs. Thus, the network should ensure that the total bandwidth assigned to an MS, via its available interfaces, does not exceed that of the agreed SLA, as shown in the following:

$$\sum_{i=1}^{n} BW_i \leq BW_{SLA}. \quad (5)$$

In the envisioned architecture, the AAA server performs the bandwidth aggregation control mechanism. Indeed, the AAA server keeps track of the SLSs negotiated by each MS.

C. Dynamic QoS Negotiation

1) Initial QoS Negotiation: Upon connecting to the network, an MS negotiates with the QGS regarding its service level. Firstly the MS requests predefined services available in the network. When the MS obtains the requested information, it sends a service negotiation request; the request is received by the BS and forwarded to the QGS. The QGS consults with the AAA server to verify whether the MS is authorized to receive the requested service. In case of acceptance, the QGS sends the

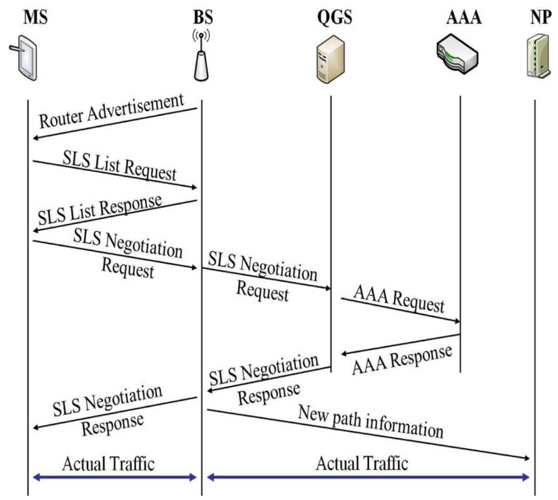


Fig. 3. Signaling for initial QoS negotiation.

new SLS to the corresponding BS in order to perform traffic conditioning. The QGS also notifies the successful service level negotiation to the MS via the BS. The BS assigns a time-slot to the MS for data transmission and delivers the quadruple $\{$ MS, IF-id, BS, time-slot $\}$ to the NP to add the new path to the available paths for data transmission belonging to the MS. The (IF-id) is the corresponding MS' wireless interface identification. Right after that, the MS starts using the service. This procedure is conceptually depicted in Fig. 3. If the MS is not authorized to acquire the requested service or there are not enough resources to satisfy it, the request is rejected and a negative negotiation response is sent to the MS, which includes the reasons for turning down the request and the available resources for which the MS can currently negotiate.

Fig. 4 shows the general procedure that users follow to negotiate the required bandwidth for their applications. The users attempt to get the whole required bandwidth through any of the available interfaces, starting from the interface with the strongest signal and following a descending order of the signal strength. Recall that every time a user's negotiation request is rejected by QGS, QGS informs the available bandwidth to the user. In case of the user could not obtain the requested bandwidth through any interface, it evaluates whether the sum of the available bandwidth of each interface satisfies the required bandwidth. In affirmative case, the user negotiates the available bandwidth through each interface until reaching the requested bandwidth. Otherwise, the user should consider downgrading the requested bandwidth or waiting for better network conditions.

Negotiation Process' Algorithm

```
n: Number of wireless interfaces of the MS
BW_req: Required bandwidth for the MS
For i=1 to n {BW_available(i) = 0}
Total_available_BW = 0

For i = 1 to n {
    Negotiate BW_req through interface i
    If successful { Exit the negotiation process}
    Else {
                BW_available(i) = BW indicated in the negotiation response
                        message as available BW.

                Total_available_BW = Total_available_BW + BW_available(i)
    }
}

If Total_available_BW ≥ BW_req {
    Negotiate the available BW through each interface until reaching BW_req
    Exit the negotiation process
}
Else {
    If it is possible to reduce BW_req to Total_available_BW {
                        Negotiate the available BW through each interface
     Exit the negotiation process
    }
        Else {wait for the network condition to be suitable for the application's
        requirements}
}
```

Fig. 4. Pseudo-code of the negotiation process.

2) QoS Renegotiation: Service level renegotiation is required when an MS is currently receiving services from the network and one of the following three cases occurs: *1)* the service requirement of the MS changes, *2)* the resources in the network become scarce and the QGS requires the MSs to degrade their existing SLSs, and *3)* the MS performs handoff and the available resources in the new subnet are not enough to guarantee the current SLS. For the two first cases, the renegotiation is similar to the initial QoS negotiation procedure apart from the fact that the MS keeps receiving services during the renegotiation period. If the QGS rejects the new service level requested by the MS, its current service level is retained. On the other hand, in the third case, the service is stopped until a new SLS is successfully negotiated. After each successful QoS renegotiation, NP is informed of the quadruple $\{$ MS, IF-id, BS, time-slot $\}$ to update the information of the MS's wireless interface. In case of renegotiation due to handoff, NP simply redirects the traffic from the previous BS to the new BS.

3) Exchange of Users' Profiles: QoS and mobility functionalities are not independent. Coupling between the two functionalities occurs because QoS is tied to a specific path and paths change as a result of handoff. Without this coupling, when handoff occurs, the state becomes not associated with the new path. Thus there is no QoS on the new path and the unused QoS state on the old path affects the use of the network resources. Thus, QoS and mobility management should be coupled to ensure the continuity of service level perceived by mobile users while they perform handoffs between different access points. The work in [28] presents an interesting classification of mobile applications based on their mobility management requirements and also investigates the handoff performance of the existing mobility management protocols for these applications.

One of the most relevant issues in wireless networks is to track the location of an MS and inform the appropriate BS of the SLS of the MS. There are several ways to inform the new BS of MS's SLS.

1) *Mobility pattern prediction:* The QGS prefigures the next BS to which the MS will perform handoff and delivers the SLS to that BS, thus reducing the handoff negotiation delay [29]. However, this method increments highly the complexity of the system and accordingly impact its scalability.

2) *Broadcasting SLSs:* The QGS delivers the SLS of each MS to every BS in the domain. This method is simple and minimizes the handoff negotiation delay, as all the BSs know in advance the SLS of every MS. Thus upon handoff, the new BS is able to immediately perform traffic conditioning. However, this mechanism presents scalability problems in terms of signaling overheads and data stored at the BSs.

3) *DSNP:* It is an enhancement to broadcasting SLSs method that was introduced by Chen *et al.* [8]. The QGS delivers the SLS of the MS to the current BS and its neighboring BSs. Thus, all possible new BSs already know the MS's SLS. This method decreases the signaling overheads as well as the amount of SLSs stored at the BSs. However, both signaling and data stored still remain high, as several BSs receive and store in their tables the SLS of the MS every time it performs handoff.

4) *SLS delivery on demand:* The QGS delivers the MS's SLS to the new BS in response to an SLS solicitation message sent by the MS and forwarded by the new BS. This method reduces the signaling overhead and the data stored at the BSs, as the SLS of the MS is delivered to the appropriate BS only on demand. However, such a reduction on the signaling overhead and data stored comes at the price of handoff negotiation delay that is increased by the round trip delay between the BS and the QGS.

5) *BS-collaboration approach:* An enhancement to SLS delivery on demand method was introduced in our previous work [9]. When an MS performs handoff, the new BS consults the previous BS for the SLS of the MS. By this way, this method reduces the handoff negotiation delay as the round trip delay among two adjacent BSs should be shorter than that among BSs and QGS. This approach is highly scalable as the MS's SLS is delivered only to the new BS.

6) *Encrypted SLS:* As in [10] after a user negotiates its service level with the network, the QGS delivers an SLS token to the user. The token contains all details associated with the service level and traffic specifications. The token is encrypted and cannot be deciphered by MSs. However, any BS into the same domain can decrypt it by using a network specific secret key. When a user performs handoff within the domain, it simply sends the SLS token to the new BS, which decrypts it and performs traffic conditioning. Accordingly, the handoff negotiation delays as well as the signaling overheads are minimized. However, this method presents some security concerns. Indeed, malicious users can obtain the SLS token from genuine users and steal their service levels.

In the proposed QoS negotiation system, to inform the new BS of the current service level of the MS, we developed an enhanced version of the encrypted SLS approach called extended encrypted SLS (EESLS), which tackles its security issues. We implement public-key cryptography at the BSs, also known as asymmetric cryptography, in which the key used to encrypt a message differs from the key used to decrypt it. Thus, each BS has a pair of cryptographic keys: a public key and a private key. The private key is kept secret while the public key is announced to the MSs by using the router advertisement message. To keep the complexity at the BSs low, the MSs encrypt the messages related to authentication only at the beginning of the service negotiation process and upon handoff. Additionally, the generation of the public and private keys pair for each BS is delegated to the QGS. Moreover, the QGS uses symmetric cryptography with all the BSs within the domain, which uses a single secret key for both encryption and decryption.

The initial QoS negotiation is shown in Fig. 3. The BSs include their public keys into the router advertisement messages. The MS generates and includes a password into the SLS negotiation request message and encrypts this message with the public key of the BS. The BS decrypts the message and forwards it to the QGS. Upon acceptance, the QGS encrypts the new SLS of the MS along with the password of the MS, termed as token, and sends it to the BS into the SLS negotiation response message. The BS forwards the message to the MS and decrypts the SLS to perform traffic conditioning. Thus, the MS gets its own token. That token works only when it is sent by the QGS to a BS. Therefore, even if a malicious user steals the token, it cannot do anything with the same, because the MS should include some security information when it attempts to get services from other BSs.

As an MS moves and attempts to change its point of attachment to the network within the same domain (intra-domain handoff), it receives the router advertisement message from the new BS, and uses the public key of that new BS to encrypt a handoff negotiation message, which contains the token, the MS' password, and the current time, as shown in the following:

$$HNM = Encrypted_{BS}\{Token, password, time\} \quad (6)$$

where *HNM* denotes the handoff negotiation message that is encrypted with the public key of the BS

$$Token = Encrypted_{QGS}\{SLS, password\} \quad (7)$$

where the *Token* is encrypted with the key of the QGS, and can be decrypted by any BS in the domain.

Then the MS sends the HNM message to the new BS, which decrypts the message with its private key, decrypts the token with the network's global key, and compares the MS's password into the token with the password in the message. If they match, the new BS verifies if the message is recent enough by verifying the time into the message. If the time is recent enough, the new BS performs traffic conditioning by applying the SLS of the MS. The MS receives the handoff negotiation response message and starts receiving the service.

In this fashion, we ensure that a malicious user cannot steal the service level of an MS just by intercepting its token. On the other hand, if a malicious user gets the handoff negotiation message of an MS, it cannot be used at the same BS because only one MS can receive services with a single SLS. The time in the handoff negotiation message is used to ensure that a malicious user cannot use the SLS of a legitimate user when it performs handoff to another BS. In such a case, the time in the message is not recent. The malicious user cannot use the handoff negotiation message of the legitimate MS in a different BS because the pairs of keys are different for the BSs; therefore, the new BS is unable to decrypt the message.

After that, the new BS informs the QGS that it is currently providing services to the MS in order to update the available resources of the new BS in the QGS database. Moreover, when the previous BS detects that the MS is no longer active in the coverage area, it erases the SLS of the MS from its database, releases the associate resources, and informs the QGS of its new state of available resources. These operations ensure that BSs store information on SLSs of only users that they are currently serving. In case the new BS is unable to guarantee the SLS, it forwards the handoff negotiation messages to the QGS. Then the QGS sends a negative handoff negotiation response to the MS, informing the MS of available service levels that the new BS can offer.

When an MS moves out to a new domain (inter-domain handoff), the MS negotiates a new SLS with the QGS of the new domain, because getting the SLS of the MS from the QGS in the previous domain may be more costly than negotiating a new one.

The main goal of contemporary researches in mobile networks is to provide seamless handoff, which is not always possible; in some cases the available resources in the new BS may not be enough to guarantee the SLS of the MS. In such a case, the QGS asks the MS to downgrade its SLS. Such downgrade of the service level affects the quality perceived by the user. Therefore, the user notifies the corresponding source of the new service level and the sender accordingly adjusts its streaming rate.

D. Packet Scheduling

The successful transmission of data belonging to a single application via multiple paths depends on the appropriate scheduling strategy [23], [30] . Most of the previously proposed scheduling algorithms are based on round robin scheduling algorithms. Recently a new technique was introduced, named earliest delivery path first (EDPF) [24] that focuses on scheduling by estimating the delivery time of the next packet through each path. By using this estimation, EDPF schedules the packets via the path with the earliest delivery time.

As previously mentioned, our QoS negotiation system implements a time-slotted approach for bandwidth allocation at the BSs. Thus, each MS is allocated a specific period of time to access the wireless channel. At any given time-slot, only one MS is allowed to transmit/receive data through a particular BS. The time-slot size varies from an MS to another, because the length of the time-slot depends on the amount of bandwidth negotiated by MSs.

After each successful QoS negotiation or renegotiation, the BS assigns a specific time-slot to the MS and informs the NP of the specific beginning and ending times of the time-slot for the MS. Using these two parameters, NP can make an accurate estimation of the delivery time of the next packet for the MS through each available path.

Since no suitable scheduling algorithm exists for the time-slot approach implemented in our QoS negotiation system, we developed an enhanced version of EDPF called time-slotted earliest delivery path first (TS-EDPF). TS-EDPF uses the time-slot assigned to the MS through each available path for an accurate computation of the delivery time of the next packet.

To estimate the delivery time of a packet via a specific path, TS-EDPF computes the time at which the packet arrives at the BS by computing the time at which the transmission can begin at the BS on the path. Then, it adjusts this time so it is within the time-slot assigned to the MS. By adding the transmission delay, we obtain the delivery time of the packet which should be within the time-slot of the MS.

The time at which the transmission can begin at the BS is denoted as

$$S_i{}^l = MAX(a_i + D_l, A_l) \tag{8}$$

where a_i and D_l denote the time at which packet i arrives at the NP and the delay from the NP to the BS along path l, respectively. A_l denotes the time instant when path l will be available for the next transmission.

To adjust $S_i{}^l$ to be within the time-slot assigned to the MS, let $[X_l, Y_l]$ be the time-slot period for the MS through path l and X_l' be the starting time of the subsequent time-slot. Furthermore, let $\Gamma\left(S_i^l, l\right)$ be the function that returns the next valid time at which the transmission can commence at the BS on path l based on the time-slot $[X_l, Y_l]$:

$$\Gamma(S_i{}^l, l) = \begin{cases} S_i{}^l, & \text{if } S_i{}^l \in [X_l, Y_l] \\ X_l', & \text{otherwise.} \end{cases} \tag{9}$$

To compute the transmission delay for packet i via link l, denoted by $T_i{}^l$, let L_i be the size of packet i and let B_l denote the bandwidth of the wireless link on path l. It should be reminded that in a time-slot division system, each MS uses the total bandwidth of the link during a short period of time:

$$T_i{}^l = \frac{L_i}{B_l}. \tag{10}$$

Then, the algorithm computes the time at which the transmission of packet i can be completed at the BS on path l, denoted by E_i^l:

$$E_i{}^l = \Gamma(MAX[a_i + D_l, A_l], l) + T_i{}^l. \tag{11}$$

Finally, it should be ensured that the transmission of packet i is completed within the time-slot assigned to the MS. Let $\Theta\left(E_i^l, l\right)$ be the function that returns the next valid time at which the transmission of packet i can be completed at the BS on path l based on the time-slot $[X_l, Y_l]$:

$$\Theta(E_i{}^l, l) = \begin{cases} E_i{}^l, & \text{if } E_i{}^l \in [X_l, Y_l] \\ X_l' + T_i{}^l, & \text{otherwise.} \end{cases} \tag{12}$$

The delivery time of packet i, through path l, can be then computed as follows:

$$d_i{}^l = \Theta\left(\Gamma(MAX(a_i + D_l, A_l), l) + T_i{}^l, l\right). \tag{13}$$

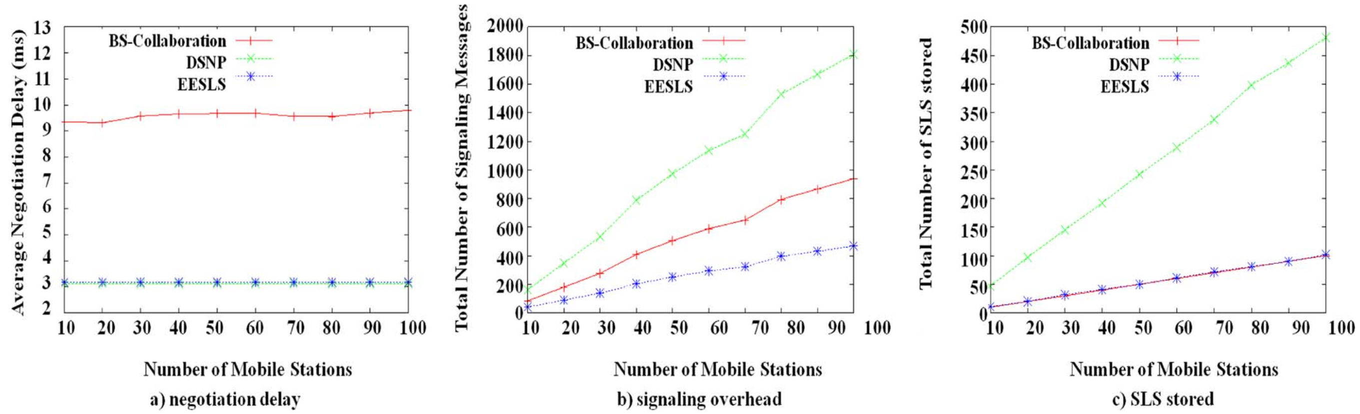


Fig. 5. Overall performance of dynamic QoS negotiation module for varying number of mobile stations.

TS-EDPF estimates the delivery time of a packet through each available path and then schedules the packet via the path with the earliest delivery time.

IV. Performance Evaluation

This section presents and discusses the performance of the proposed bandwidth aggregation-aware dynamic QoS negotiation mechanism in three different parts. Firstly, we verify the applicability of the proposed mechanism to exchange users' profiles. Secondly, we demonstrate the necessity for the BAG control mechanism. Finally, we showcase the merits of the proposed TS-EDPF scheduling algorithm.

A. Exchange of Users' Profiles

We set up a simulation environment using the network simulator (NS-2) [31] to evaluate the applicability of the proposed mobility management mechanism. As mobility management deals with MSs performing handoffs between BSs of the same wireless technology, for this evaluation, we consider that users are employing only one wireless interface (the same wireless technology for all the users). The mobility of MSs follows the reference point group mobility (RPGM) model. To provide a wide area for the users to move around, we consider the coverage area of five BSs where MS_1 is located as shown in Fig. 2. The number of MSs roaming over the coverage area varies from 10 to 100. The simulation starts when all MSs have already initiated their service levels. The major issue in providing QoS in wireless networks consists in the mobility of users (where seamless and lossless handoffs need to be guaranteed). Therefore, the focus of this evaluation is on the service level negotiation upon intra-domain handoff, as this is the most frequent handoff performed by MSs.

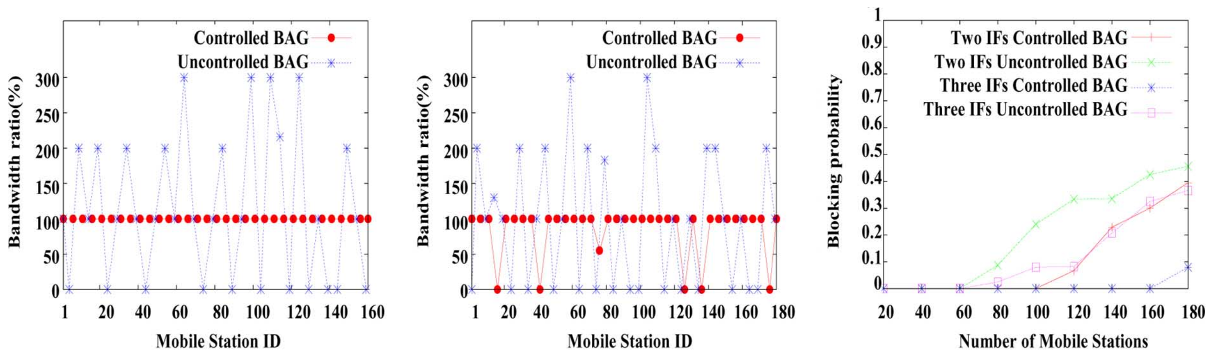
Fig. 5(a) shows that both the proposed extended encrypted SLS (EESLS) and DSNP exhibit close handoff negotiation delays associated to the round trip delays from the MSs to the BSs. The slight difference among them is attributable to the decryption time of the handoff negotiation message in case of the EESLS method. On the other hand, the BS-collaboration method shows the highest handoff negotiation delay because of the fact that the new BS gets the SLS from the previous BS, which requires communication between the two BSs. This increases the overall negotiation delay, whereas in DSNP the new BS already has the SLS or receives the SLS from the MS in case of the EESLS scheme. Fig. 5(b) shows that the proposed EESLS scheme has the lowest signaling overhead as the MSs deliver their own SLS to only the next point of attachment. Fig. 5(c) demonstrates that both the BS-collaboration approach and the EESLS method require the storage of a lower number of SLSs at the table of BSs than that of DSNP. The small difference between the BS-collaboration method and the EESLS method is because when an MS performs handoff in the BS-collaboration method, the previous BS is asked to deliver the SLS of that MS, and right after that, the previous BS erases that SLS from its table. On the other hand, in the EESLS method, the previous BS erases the SLS of an MS when it realizes that the MS is not active in its coverage area.

We proposed an enhanced version of encrypted SLS mechanism that addresses its security limitations and makes it robust enough to prevent malicious users from stealing the service levels of legitimate users. EESLS highly increases the scalability of the system by minimizing the signaling overhead and the required data storage. This reduces the size of the tables of BSs and also the time required to search into these tables. EESLS also achieves low handoff negotiation delay, which is essential to provide seamless handoff and to ensure the continuity of the service.

B. Bandwidth Aggregation Control Mechanism

To demonstrate the benefits of using the BAG control mechanism, several simulations were conducted. In these simulations, we evaluate the system operating at its critical state (i.e., large number of users) that correspond to 160 MSs, and also when the system's capacity is saturated (e.g., 180 MSs). All MSs are equipped with three wireless interfaces (IFs), which are assumed to correspond to the wireless technologies supported by the same ISP. Additionally, we set the locations of users to the overlapped coverage area of these three wireless technologies as shown in Fig. 2 (i.e., the area where MS_2 is located). Thus, each MS is able to negotiate and receive services through one, two, or three BSs. The bandwidth level specified in the SLA of each MS varies from 300 Kbps to 2 Mbps.

In this performance evaluation, two SLS negotiation approaches are considered. In the first approach, named uncontrolled BAG, the AAA assumes any single SLS request, which



a) Individual ratio of bandwidth used to that of SLA (160 MSs)

b) Individual ratio of bandwidth used to that of SLA (180 MSs)

c) Blocking probability for different numbers of MSs

Fig. 6. Overall performance of the bandwidth aggregation control scheme.

does not exceed the bandwidth specified in the SLA of the MS, to be valid. In the second approach, called controlled BAG, the AAA keeps track of the total bandwidth currently used by the MS and ensures that the total bandwidth assigned to an MS, via its available interfaces, does not exceed that of the agreed SLA.

Fig. 6(a) and (b) shows the ratio of the individual bandwidth actually used by each MS to that of its agreed SLA. The figures consider two populations of mobile users, 160 and 180 MSs, respectively. The figures demonstrate that when BAG is not controlled, some mobile stations acquire up to three times their agreed bandwidth depriving others from having accesses to the bandwidth to which they are subscribed. This intuitively results in an unfair service, a fact that is illustrated in the variation of the bandwidth ratio from 0% to 300%. When the BAG control mechanism is in use, each MS receives a bandwidth in the range of 0% to 100% of its SLA. In case of 160 MSs, shown in Fig. 6(a), all MSs are provided with bandwidths equal to that of their SLAs. This demonstrates that the BAG control mechanism efficiently distributes the available bandwidth of the three BSs. In the absence of such a BAG control mechanism, the system ends up by allocating 300% of SLAs to few MSs, 200% of SLAs to other MSs, and 0% to many MSs. This obviously puts both the scalability and fairness of the system in question. When the network is visited by a high number of mobile nodes [180 MSs in Fig. 6(b)] and the network resources become scarce, the BAG control mechanism rejects the requests from some MSs, but its performance remains comparatively much better than that of the uncontrolled BAG approach.

The blocking probabilities for different numbers of mobile stations are shown in Fig. 6(c). Based on the number of wireless technologies in use, two scenarios are considered. Firstly, we consider the use of two and three interfaces. The goal behind this experiment is to investigate the impact of the number of deployed interfaces on the system scalability. The results indicate that in case of the BAG control mechanism, the system starts blocking requests when the number of mobile stations exceeds 100 and 160 when two and three IFs are used, respectively. In the absence of such a BAG control mechanism, the blocking probability gets non-null values earlier, in the presence of few MSs (i.e., 60 MSs when three IFs are used). Based on the above results, it can be concluded that in the absence of a BAG control mechanism, MSs are allocated bandwidths exceeding those of their SLAs. This renders the ISP unable to control its own resources, ultimately resulting in an unfair service and a high blocking probability.

C. Packet Scheduling

In order to verify the effectiveness of the proposed multi-path scheduling algorithm TS-EDPF, we conducted several simulations. As comparison terms, we used the three most suitable algorithms for the proposed QoS negotiation system, namely weighted round robin (WRR), weighted interleaved round robin (WIRR), and earliest delivery path first (EDPF) scheduling algorithms. WRR and WIRR are able to use the knowledge of the negotiated bandwidth through each available path for an accurate and effortless distribution of packets among them. On the other hand, the EDPF scheduling algorithm additionally makes use of the delays between the network proxy and the BSs to estimate the delivery times of packets via each available path. Unless otherwise specified, the time-slot interval is set to 0.1 s; i.e., an MS accesses a BS once every 100 ms.

Three video applications are used in this simulation: CBR_1, VBR_1, and VBR_2. CBR_1 is a constant bitrate application with a data rate of 2.6 Mbps. VBR_1 and VBR_2 are variable bitrate video traces collected from [32]. VBR_1 corresponds to the MPEG-4 trace of the movie Jurassic Park-I, generated at high quality with peak rate equal to 2.6 Mbps and mean rate of 790 kbps that represents 30.38% of the peak rate. VBR_2 corresponds to the MPEG-4 trace of a soccer game also generated at a high quality with peak rate of 3.2 mbps and mean rate of 1.140 mbps that represents 35.63% of the peak rate. The duration of the three video applications is 3600 s. We consider one MS equipped with three wireless interfaces of different technologies supported by the same service provider. The MS executes one by one the three video applications in different sessions. For each session, the MS negotiates an aggregate bandwidth of 100% of the video application bitrate for CBR or peak rate for VBR traffic, respectively. The maximum transmission delay for packets of the simulated applications is set to 300 ms.

Table I summarizes the results for the three video applications. The buffer size indicates the largest number of packets that were queued in the buffer awaiting playback. The bandwidth ratio indicates the effective use of the aggregated bandwidth. The disorder delivery radio indicates the proportion of packets that arrived in an out-of-order manner. The results in the

TABLE I
EVALUATION OF THE SCHEDULING ALGORITHMS

	CBR_1				VBR_1 (Jurassic park I)				VBR_2 (Soccer game)			
Algorithm	buffer size (pkts)	Bw ratio (%)	disorder delivery ratio (%)	Packet loss ratio (%)	buffer size (pkts)	Bw ratio (%)	disorder delivery ratio (%)	Packet loss ratio (%)	buffer size (pkts)	Bw ratio (%)	disorder delivery ratio (%)	Packet loss ratio (%)
TS-EDPF	1	97.16	1.3	0.002	3	30.80	3.3	0.001	12	35.84	3.1	0.002
EDPF	19	97.13	61.8	2.84	16	30.69	50.95	4.15	18	35.79	54.89	4.72
WRR	17	97.14	52.6	4.53	17	30.68	37.33	7.14	21	35.81	43.71	7.70
WIRR	21	97.13	61.8	2.79	16	30.69	53.51	5.05	22	35.78	56.89	4.97

table demonstrate that the proposed TS-EDPF scheme outperforms the three other schemes in terms of the overall quantifying parameters. Indeed, TS-EDPF shows the best performances for CBR traffic with a disorder delivery ratio of 1.3%, maximum buffer size of only one packet, and the packet loss of 0.002%. For VBR_1, the disorder delivery ratio was 3.3%, the maximum buffer size was three packets, and the packet loss was found to be 0.001%. As for VBR_2, the disorder delivery ratio was 3.1%, the maximum buffer size was 12 packets, and the packet loss was 0.002%. This good performance of TS-EDPF is attributed to the adoption of time-slot based policy that was not considered in the other three schemes. It should be noted that a value of 0.002% as packet loss rate means that the scheduling algorithm is accurately delivering data packets to the MS. The results indicate that all schemes use efficiently the aggregated bandwidth for the CBR application, achieving fairly high throughputs. On the other hand, for VBR_1 and VBR_2, the bandwidth utilization rates are around 30% and 35%, respectively. This means that around 70% and 65% of the negotiated bandwidth for VBR1 and VBR2 remain unused during the applications' running times (i.e., 3600 s).

In general, the results in Table I demonstrate the effectiveness of the proposed TS-EDPF scheduling algorithm as it achieved by far the smallest buffer sizes, smallest disorder delivery ratios, and the lowest packet loss rates for the three considered video applications. The results also demonstrate that the aggregated bandwidth was used up to 97.16%. That is because the video packets cannot be fragmented. Thus if the time needed to transmit the next packet is larger than the remainder time of the time-slot, the packet will be transmitted during the time-slot in the next round. This implies that there may be an unused time at the end of each time-slot. The proposed TS-EDPF aims to minimize that amount of unused bandwidth by scheduling later-arriving smaller packets to be transmitted during that remaining time. Thus, TS-EDPF schedules a few packets to arrive in out-of-order at the receiver to maximize the bandwidth utilization. These packets arriving out of order do not affect the performance of the scheduling process, as they arrive earlier than when they are scheduled in a strict order. They only affect the buffer size, as they have to wait at the MS' buffer until the preceding packets arrive.

D. Efficient Bandwidth Utilization

As mentioned earlier, for VBR applications large amounts of the negotiated bandwidth remain unused. This is because the peak rate of the VBR traffic is usually reached only once; during the remainder of the transmission, the data rate is much lower than the peak rate. For an efficient bandwidth utilization, we implemented a priority queue scheme at BSs. Thus, when a time-slot of an MS starts, the queue associated to this MS becomes the priority queue, which will be exclusively served during the time-slot. When the priority queue becomes empty, BS serves the best-effort traffic queue.

TABLE II
EVALUATION OF TS-EDPF WITH PRIORITY QUEUE FOR VBR_2

Algorithm	buffer size (pkts)	Bw ratio (%)	disorder delivery ratio (%)	Packet loss ratio (%)
TS-EDPF	12	35.84	3.1	0.002
TS-EDPF & Priority Queue	21	98.71	5.8	0.009

Table II demonstrates the effectiveness of the incorporated priority queue scheme as it increases the aggregated bandwidth utilization ratio from 35.84% for VBR_2 to 98.71. However, the priority queue scheme slightly affects the transmission of VBR_2, by increasing the disorder delivery ratio from 3.1% to 5.87%, the buffer size from 12 packets to 21 packets, and the packet loss rate from 0.002% to 0.009%. Thus, the bandwidth utilization during VBR traffic transmission is highly increased using the priority queue at the BSs, at the cost of a slight increment in the packet loss of the VBR application. The priority queue scheme also mitigates the unused bandwidth at the end of the time-slots (as mentioned in Section III-B) by serving best-effort traffic during these times.

V. CONCLUSION

In next-generation wireless networks, mobile computers will be equipped with several wireless interfaces that enable users to connect to different networks at the same time. In this paper, we proposed a new scheme that allows users to dynamically negotiate QoS profiles with different networks. The proposed scheme supports initial negotiation, renegotiation, bandwidth aggregation, and mobility. A new method to inform the QoS profile of a user to BS towards which the user is moving was presented, and its applicability was demonstrated through computer simulations. We showed that the proposed scheme achieves the shortest negotiation delays and reduces overhead in terms of both signaling messages and state information storage. The bandwidth aggregation mechanism mitigates the resource constraints in wireless networks. It helps users to negotiate their desired service levels and reach them by using one or more interfaces. Simulation results showed the need for a bandwidth aggregation control mechanism to maintain a scalable and fair use of the network resources. Finally, an enhanced version of the EDPF scheduling algorithm was proposed to adapt it to the bandwidth allocation scheme implemented in our QoS negotiation system. We demonstrated via simulations that the proposed TS-EDPF scheduling algorithm largely mitigates the packet reordering issue and the packet loss rate.

REFERENCES

[1] C. Ward, M. J. Buco, R. Chang, and L. Luan, "A generic SLA semantic model for the execution management of E-business outsourcing contracts," in *Proc. 3rd Int. Conf. E-Commerce (EC-Web 2002)*, Berlin, Germany, Sep. 2002.

[2] T. Taleb and A. Nafaa, "A fair and dynamic auction-based resource allocation scheme for wireless mobile networks," in *Proc. IEEE ICC'08*, Beijing, China, May 2008.

[3] T. Taleb, K. Kashibuchi, A. Leonardi, S. Palazzo, K. Hashimoto, N. Kato, and Y. Nemoto, "A cross-layer approach for an efficient delivery of TCP/RTP-based multimedia applications in heterogeneous wireless networks," *IEEE Trans. Veh. Technol.*, vol. 57, no. 6, pp. 3801–3814, Nov. 2008.

[4] T. Nguyen, N. Boukhatem, Y. Doudane, and G. Pujolle, "COPS-SLS: A service-level negotiation protocol for the Internet," *IEEE Commun. Mag.*, vol. 40, no. 5, pp. 158–165, May 2002.

[5] X. Wang and H. Schulzrinne, "An integrated resource negotiation, pricing, and QoS adaptation framework for multimedia applications," *IEEE J. Select. Areas Commun.*, vol. 18, no. 12, pp. 2514–2529, Dec. 2000.

[6] SrNP: Service Negotiation Protocol, Tequila Consortium, 2001. [Online]. Available: http://www.ist-tequila.org/deliverables.

[7] J. A. Colas *et al.*, "Connecting ambient networks—Architecture and protocol design (release 1)," *Ambient Networks Consortium, Deliverable D 3.2*, Mar. 2005.

[8] J. C. Chen, A. McAuley, V. Sarangan, S. Baba, and Y. Ohba, "Dynamic service negotiation protocol (DSNP) and wireless DiffServ," in *Proc. ICC 2002*, New York, Apr. 2002.

[9] J. C. Fernandez, T. Taleb, N. Ansari, K. Hashimoto, N. Kato, and Y. Nemoto, "Dynamic QoS negotiation for next generation wireless communications systems," in *Proc. WCNC 2007*, Hong Kong, China, Mar. 2007.

[10] M. Barry, A. T. Campbell, and A. Veres, "Distributed control algorithms for service differentiation in wireless packet networks," in *Proc. IEEE INFOCOM 2001*, Anchorage, AK, Apr. 2001.

[11] V. Sarangan and J. C. Chen, "Comparative study of protocols for dynamic service negotiation in the next-generation Internet," *IEEE Commun. Mag.*, vol. 44, no. 3, pp. 151–159, Mar. 2006.

[12] I. F. Akyildiz, J. Xie, and S. Mohanty, "A survey on mobility management in next generation all-IP based wireless systems," *IEEE Wireless Commun.*, vol. 11, no. 4, pp. 16–28, Aug. 2004.

[13] N. Banerjee, W. Wu, S. Das, S. Dawkins, and J. Pathak, "Mobility support in wireless Internet," *IEEE Wireless Commun.*, vol. 10, no. 5, pp. 54–61, Oct. 2003.

[14] R. Ramjee, K. Varadhan, L. Salgarelli, S. R. Thuel, S. Y. Wand, and T. L. Porta, "Hawaii: A domain-based approach for supporting mobility in wide-area wireless networks," *IEEE/ACM Trans. Netw.*, vol. 10, no. 3, pp. 396–410, Jun. 2002.

[15] M. Liu, Z. Li, X. Guo, and E. Dutkiewicz, "Performance analysis and optimization of handoff algorithms in heterogeneous wireless networks," *IEEE Trans. Mobile Comput.*, vol. 7, no. 7, pp. 846–857, Jul. 2008.

[16] R. Stewart *et al.*, "Stream control transmission protocol," in *RFC 2960*, Oct. 2000.

[17] S. Fu and M. Atiquzzaman, "SCTP: State of the art in research, products, and technical challenges," *IEEE Commun. Mag.*, vol. 42, no. 4, pp. 64–76, Apr. 2004.

[18] L. M. F. Yu and V. C. M. Leung, "A new method to support UMTS/WLAN vertical handover using SCTP," *IEEE Wireless Commun.*, vol. 11, no. 4, pp. 44–51, Aug. 2004.

[19] R. Fracchia, C. Casetti, C. Chiasserini, and M. Meo, "WiSE: Best-path selection in wireless multihoming environments," *IEEE Trans. Mobile Comput.*, vol. 6, no. 10, pp. 1130–1141, Oct. 2007.

[20] M. Jain and C. Dovrolis, "End-to-end available bandwidth: Measurement methodology, dynamics, and relation with TCP throughput," *IEEE/ACM Trans. Netw.*, vol. 11, no. 4, pp. 537–549, Aug. 2003.

[21] A. Abdelal, T. Saadawi, and M. Lee, "LS-SCTP: A bandwidth aggregation technique for stream control transmission protocol," *Comput. Commun.*, vol. 27, no. 10, pp. 1012–1024, Jun. 2004.

[22] L. Magalhaes and R. Kravets, "MMTP—Multimedia multiplexing transport protocol," in *Proc. 1st ACM Workshop Data Communications in Latin America and the Caribbean (SIGCOMM-LA 2001)*, San Jose, Costa Rica, Apr. 2001.

[23] T. Taleb, T. Nakamura, and K. Hashimoto, "Multi-source streaming in next generation mobile communication systems," in *Proc. IEEE ICC'08*, Beijing, China, May 2008.

[24] K. Chebrolu and R. Rao, "Bandwidth aggregation for real-time applications in heterogeneous wireless networks," *IEEE Trans. Mobile Comput.*, vol. 5, no. 4, pp. 388–403, Apr. 2006.

[25] D. Jurca and P. Frossard, "Media flow rate allocation in multipath networks," *IEEE Trans. Multimedia*, vol. 9, no. 6, pp. 1227–1240, Oct. 2007.

[26] G. Kramer, B. Mukherjee, and G. Pesavento, "Ethernet PON (ePON): Design and analysis of an optical access network," *Photonic Netw. Commun.*, vol. 3, no. 3, pp. 307–319, Jul. 2001.

[27] J. C. Fernandez and N. Kato, "Multi-path video streaming in wireless networks using time-slot based scheduling," *Int. J. Bus. Data Commun. Netw. (IJBDCN)*, to be published.

[28] S. Mohanty and I. F. Akyildiz, "Performance analysis of handoff techniques based on mobile IP, TCP-migrate, and SIP," *IEEE Trans. Mobile Comput.*, vol. 6, no. 7, pp. 731–747, Jul. 2007.

[29] I. F. Akyildiz and W. Wang, "A predictive user mobility profile for wireless multimedia networks," *IEEE/ACM Trans. Netw.*, vol. 12, pp. 1021–1035, Dec. 2004.

[30] T. Taleb, D. Mashimo, K. Hashimoto, N. Kato, and Y. Nemoto, "On how to mitigate the packet reordering issue in the explicit load balancing scheme," in *Proc. IEEE Global Information Infrastructure Symp. (GIIS)*, Marrakech, Morocco, Jul. 2007.

[31] CB/LBNL/VINT: Network Simulator ns (Version 2). [Online]. Available: http://www.isi.edu/nsnam/ns/.

[32] MPEG-4 and H.263 Video Traces for Network Performance Evaluation, 2006. [Online]. Available: http://www.tkn.tu-berlin.de/research/trace/trace.html.

Juan Carlos Fernandez (S'07) received the B.E. degree in computer science from the Universidad Central de Venezuela (UCV), Caracas, in 2000 and the M.S. degree from the Graduate School of Information Sciences (GSIS) at Tohoku University, Sendai, Japan, in 2008. Currently, he is working toward the Ph.D. degree at GSIS, Tohoku University.

His research interests are in the areas of wireless networking, quality of service, real-time multimedia applications, and video streaming.

Tarik Taleb (M'05) received the B.E. degree (with distinction) in information engineering and the M.Sc. and Ph.D. degrees in information sciences from the Graduate School of Information Sciences (GSIS), Tohoku University, Sendai, Japan, in 2001, 2003, and 2005, respectively.

He is currently working as a Senior Researcher at NEC Europe Ltd, Heidelberg, Germany. Prior to his current position and until March 2009, he worked as an Assistant Professor at the GSIS, Tohoku University. From October 2005 until March 2006, he was working as a Research Fellow with the Intelligent Cosmos Research Institute, Sendai, Japan. His research interests lie in the field of 3GPP system architectural enhancements, mobile multimedia streaming, wireless networking, inter-vehicular communications, congestion control protocols, handoff and mobility management, satellite and space communications, and network security.

Dr. Taleb is on the editorial board of the IEEE TRANSACTIONS ON VEHICULAR TECHNOLOGY, the IEEE WIRELESS COMMUNICATIONS, IEEE COMMUNICATIONS SURVEYS & TUTORIALS, and other Wiley journals. He also serves as Vice Chair of the Satellite and Space Communications Technical Committee of the IEEE Communication Society (ComSoc) (2006–present). He has been on the technical program committee of different IEEE conferences, including Globecom, ICC, and WCNC, and chaired some of their symposia. He is the recipient of the 2009 IEEE ComSoc Asia-Pacific Young Researcher award (June 2009), the 2008 TELECOM System Technology Award from the Telecommunications Advancement Foundation (March 2008), the 2007 Funai Foundation Science Promotion Award (April 2007), the 2006 IEEE Computer Society Japan Chapter Young Author Award (December 2006), the Niwa Yasujirou Memorial Award (February 2005), and the Young Researcher's Encouragement Award from the Japan chapter of the IEEE Vehicular Technology Society (VTS) (October 2003).

Mohsen Guizani (S'83–M'90–SM'98–F'09) received the B.S. (with distinction) and M.S. degrees in electrical engineering and the M.S. and Ph.D. degrees in computer engineering in 1984, 1986, 1987, and 1990, respectively, from Syracuse University, Syracuse, NY.

He is currently a Professor and the Chair of the Information Science Department at Kuwait University, Safat, Kuwait. He is also an Adjunct Professor at Western Michigan University (WMU), Kalamazoo. He was the Chair of the Computer Science Department at WMU from 2003–2008. He was the Chair of the Computer Science Department at the University of West Florida, Pensacola, from 1999–2003. His research interests include computer networks, wireless communications and mobile computing, and optical networking. He is the author of six books and more than 200 publications in refereed journals and conferences.

Dr. Guizani currently serves on the editorial boards of six technical journals and is the Founder and EIC of *Wireless Communications and Mobile Computing* Journal published by John Wiley (http://www.interscience.wiley.com/jpages/1530-8669/) and the *Journal of Computer Systems, Networks and Communications* (http://www.hindawi.com/journals/) published by Hindawi, Inc. He is also the Founder and General Chair of the IEEE International Conference of Wireless Networks, Communications, and Mobile Computing (IWCMC). He guest edited a number of special issues in IEEE journals and magazines. He also served as member, Chair, and General Chair of a number of conferences. He received both the Best Teaching Award and the Excellence in Research Award from the University of Missouri-Columbia in 1999 (a college wide competition). He won the best Research Award from KFUPM in 1995 (a university wide competition). He was selected as the Best Teaching Assistant for two consecutive years at Syracuse University, 1988 and 1989. He is the past Chair of TAOS and current Chair of WTC IEEE ComSocTechnical Committees. He is a member of the IEEE Communication Society, the IEEE Computer Society, ASEE, and a senior member of ACM.

Nei Kato (M'03–A'04–SM'05) received the M.S. and Ph.D. degrees in information engineering from Tohoku University, Sendai, Japan, in 1988 and 1991, respectively.

He joined the Computer Center of Tohoku University in 1991 and has been a full Professor in the Graduate School of Information Sciences since 2003. He has been engaged in research on computer networking, wireless mobile communications, image processing, and neural networks. He has published more than 160 papers in journals and peer-reviewed conference proceedings.

Dr. Kato currently serves as Vice Chair of Technical Committee of Satellite Communications, IEICE, is a technical editor of IEEE WIRELESS COMMUNICATIONS MAGAZINE (2006), an editor of the IEEE TRANSACTIONS ON WIRELESS COMMUNICATIONS (2008), and a co-guest-editor for IEEE WIRELESS COMMUNICATIONS MAGAZINE SI on Wireless Communications for E-healthcare. He has served as a symposium co-chair for GLOBECOM 07 and ChinaCom 08, and TPC member for a large number of IEEE international conferences including ICC, GLOBECOM, WCNC, and HPSR. His awards include Minoru Ishida Foundation Research Encouragement Prize (2003), Distinguished Contributions to Satellite Communications Award from the IEEE Communications Society, Satellite and Space Communications Technical Committee (2005), the FUNAI information Science Award (2007), the TELCOM System Technology Award from Foundation for Electrical Communications Diffusion (2008), and the IEICE Network System Research Award (2009). Besides his academic activities, he also serves on the expert committee of Telecommunications Council, Ministry of Internal Affairs and Communications, and chairperson of ITU-R SG4, Japan. He is a member of the Institute of Electronics, Information and Communication Engineers (IEICE).