A Bandwidth Aggregation-aware QoS Negotiation Mechanism for Next-Generation Wireless Networks

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***Abstract*— The transmission of high quality video requires high bandwidth. Ensuring constantly high bandwidth in wireless environments is a challenging task given constraints in the current wireless network resources. Current mobile computers are equipped with multiples wireless interfaces that can be used to improve the video quality by aggregating the bandwidth of these interfaces. Such Bandwidth Aggregation (BAG) approach involves multiple paths in communication and gives rise to a number of issues related to the management of the Service Level Agreement (SLA) and packet reordering.**

To guarantee an efficient and fair management of SLA, this paper presents a bandwidth aggregation-aware QoS negotiation mechanism that enables users to dynamically negotiate their desired service levels and to reach them through the use of bandwidth aggregation. This operation is performed while ensuring a fair use of the network resources among all competing users. To cope with packet reordering, a new scheduling strategy is presented. The performance evaluation of the proposed bandwidth aggregation-aware QoS negotiation scheme and the proposed scheduling algorithm are conducted via simulations and the results are discussed.

I. Introduction

Along with the diversity and the on-going advances in wireless network technologies (e.g., IEEE 802.xx, GPRS, UMTS and Bluetooh), mobile computers are equipped with multiple wireless interfaces (IFs). When the coverage areas of different wireless technologies overlap, terminals are able to maintain simultaneous connections through the corresponding interfaces. End-terminals can increase their communication throughput by transmitting/receiving data via multiple paths. However, such multi-path delivery of data gives rise to a number of issues. Indeed, as each path may have different capacity and different propagation delay, data packets may be received in an out-of-order manner at the receiver side. Another issue is pertained to the management of the Service Level Agreement (SLA) of users.

Effectively, to ensure an efficient provision of real-time video applications in wireless networks, mobile users should be able to dynamically negotiate their Service Level Specifications (SLS) with the access network during the entire course of the connection [1]. It should be reminded that SLS indicates the quality level (in terms of bandwidth, delay, and packet loss) that a network operator should guarantee for a subscriber. The network operator should ensure that users are given their requested SLSs by checking their Service Level Agreements (SLA), contracted with the Internet Service Provider (ISP) [2].

When Bandwidth Aggregation (BAG) is possible via different interfaces, the network operator should consider the use of 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 the use of some interfaces to ensure a fair utilization of network resources among all active subscribers. In this context, the contribution of this paper is three fold. Firstly, we demonstrate the imperative need for controlling BAG mechanisms to guarantee a fair use of the network resources. Secondly, we introduce a BAG control mechanism to enhance the working of our recently proposed Dynamic SLS Negotiation mechanism [3]. Finally, as a packet scheduling strategy, we propose an enhanced version of the EPDF scheme [4] to cope with the packet reordering issue in multi-path video transmission.

The remainder of this paper is organized in the following fashion. Section II highlights some research work pertained to BAG. Section III describes the proposed BAG control mechanism. It also presents the experimental results that justify the need for BAG control in dynamic QoS negotiation schemes. Section IV presents the proposed enhancements to the EDPF scheme to minimize the delay due to packet reordering in multi-path wireless environments. Section V shows the performance evaluation of the enhanced scheduling mechanism. Finally, concluding remarks are given in Section VI.

II. Related Work

With a further and full integration of wireless networks, it will become common for a mobile node to have interfaces of different wireless technologies. A special hardware that can simultaneously access all types of wireless technologies shall be then available. To allow a mobile node to simultaneously register multiple Care-of-Addresses (CoAs), Mobile IP (MIP) simultaneous binding option [5] [6] is used. On the other hand, to keep senders always informed of these CoA registrations directly from the mobile nodes, the route optimization option [7] is used.

The use of multiple interfaces in wireless devices has been studied for different purposes. For instance, Stream Control Transmission Protocol (SCTP) [8] uses multiple interfaces to ensure high reliability. Multiple interfaces are also used for bandwidth aggregation, particularly for the provision of bandwidth-intensive real-time applications to wireless mobile users. Load-Sharing SCTP (LS-SCTP) [9] introduces a new functionality to SCTP by involving all the active transmission paths in data communication and aggregating their bandwidths to share the data load between two end-points. The bandwidth of mobile users with multiple interfaces is aggregated at the

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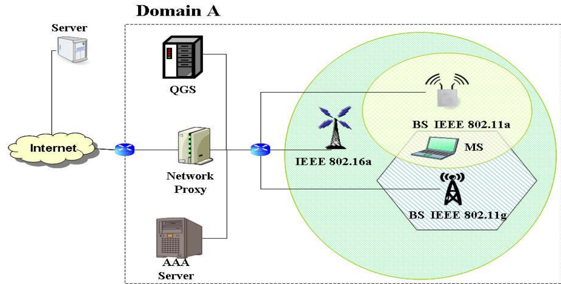

Fig. 1. The envisioned architecture for BAG-aware QoS negotiation.

transport layer in pTCP (parallel TCP) [10]. Multimedia Multiplexing Transport Protocol (MMTP) [11] is a link-layer aware protocol designed for transferring multimedia data on mobile systems. It makes simultaneous use of every communication channel available to send data.

In a bandwidth aggregation scenario, packets of the same flow are transmitted over multiple interfaces. While this operation has many advantages, it makes packets of the same application experience different latencies, resulting in out-of-order delivery to the final destination and delay jitter. For connectionless-oriented protocols such as User Datagram Protocol (UDP), addition of buffering capabilities to end terminals can ensure coherent reception and recover the original timing relationships between the transmitted data. However, when the used interfaces exhibit significantly different channel conditions, a significant jitter can be experienced and the use of a small buffer will not be efficient enough. For applications based on connection-oriented protocols (e.g., TCP), such disorder in packet reception results in the transmission of unnecessary duplicate acknowledgments (DupAcks). Indeed, current implementations of TCP work on the assumption that out-of-order packets indicate network congestion. TCP senders mistakenly halve their congestion windows when packets are reordered.

To cope with packet reordering in multi-path environments, various scheduling strategies have been proposed. The round robin scheduling mechanism is the first scheduling scheme in literature and forms the basis for many scheduling mechanisms. It is suitable for environments with paths homogeneous in terms of bandwidth and delay. For heterogeneous paths, Weighted Fair Queuing (WFQ), Weighted Round Robin (WRR), Weighted Interleaved Round Robin (WIRR) and Surplus Round Robin (SRR) are notable scheduling schemes.

The concept of QoS negotiation in wireless networks has simplified the scheduling operation as the access network guarantees certain amount of bandwidth to mobile users during their connection. Thus knowledge on the bandwidth of each path is available: no monitoring of any kind is required. The Earliest Delivery Path First (EDPF) scheme [4] exploits such characteristic and focuses its operations on finding out the best path for the delivery of each packet. A brief overview of the EDPF scheme, along with its advantages and pitfalls, will be given in Section IV. In the following section we describe our proposed BAG-aware QoS negotiation architecture.

III. BAG-aware QoS Negotiation

This section describes the proposed BAG-aware QoS negotiation mechanism. Before delving into details of the proposed scheme, first is a description of the key components of the envisioned architecture and the major operations behind our recently proposed dynamic QoS negotiation scheme [3].

A. Architecture Description

The network is divided into a number of domains administrated by different ISPs. Each domain consists of a QoS Global Server (QGS), an AAA (Authentication, Authorization, and Accounting) server, a number of Base Stations (BSs), and a population of mobile users, termed henceforth as Mobile Stations (MS). The key components of the architecture are schematically depicted in Fig. 1.

QGS, introduced also in the Dynamic Service Negotiation Protocol (DSNP) [12], basically functions as a Policy Decision Point (PDP) defined in the Policy Framework presented in [13]. It performs service level negotiation and is responsible for maintaining global information about the available resources in the whole domain. Based on this information, it admits or rejects a service level request. BSs are responsible for applying different service levels to MSs and for controlling the traffic flow of all MSs in their coverage areas. BSs inform QGS of their local resource availability and receive SLS of mobile users for traffic conditioning.

In our recently proposed QoS negotiation mechanism [3], when a user logs into the network for the first time, it requests QGS for predefined services available in the network. As soon as the MS obtains the requested information, it starts the negotiation procedure with QGS. Upon receiving a service request from MS, QGS consults its corresponding AAA server to determine if the requested service is legitimate. In case of acceptance, QGS delivers the new SLS to the appropriate BS in order to condition the traffic for the requesting MS. It also sends a positive service negotiation response to the MS. After that, the MS starts using the service. If the MS is not authorized to acquire the requested service or there are not enough resources to satisfy the requested service, a negative service negotiation response is dispatched to the MS. The response message includes the reasons for rejecting the request and the available resources that the MS can currently renegotiate for. A detailed description of the QoS negotiation procedure can be found in [3] [14].

B. Bandwidth Aggregation Issues and Adequate Solutions

In the context of SLA management, BAG may introduce new issues to operators of wireless network systems. Indeed, since i) the possibility of having the coverage areas of two or more BSs overlapped and ii) the use of multiple IFs have not been considered in the design of current service level negotiation mechanisms, a MS may receive or send data at bandwidths higher than what it deserves, in other words higher than what it subscribed for. This will result in an unfair service as it will damage the service quality perceived by other unfortunate subscribers. Another issue of BAG consists in the delivery of data via multiple paths, which leads to packet

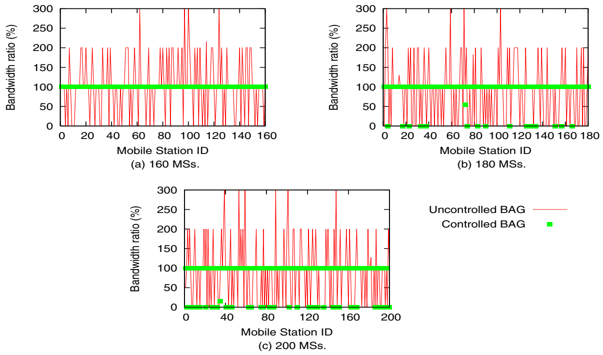

Fig. 2. Ratio of actually used individual bandwidth to that of the agreed SLA for different populations of mobile users.

reordering. In case of video streaming, packet reordering may result in extra delay in playback at the receiver side.

The two above mentioned issues can be resolved by the addition of an effective bandwidth aggregation control mechanism to the SLS negotiation mechanism and the development of an efficient scheduling strategy, respectively. The former should ensure that each MS does not receive or send data at a bandwidth higher than what is indicated in its SLA agreement with the ISP. The latter will be discussed in the next section.

To illustrate the benefits behind the use of a BAG control mechanism, we conduct some simulations using the Network Simulator (NS2). In the conducted simulations, unless otherwise specified all mobile stations are equipped with three interfaces. The three interfaces are assumed to correspond to different wireless technologies supported by the same ISP in a single domain as shown in Fig. 1. The number of MSs is varied from 20 to 200. The bandwidth level indicated in the SLA of each MS varies from 300Kbps to 2Mbps.

Two SLS negotiation approaches are studied. In the first approach, mobile users negotiate their SLSs with the network through their interfaces. The network verifies only if the requested bandwidth in single SLSs requests do not exceed the contracted one in SLA. This approach is henceforth referred to as Uncontrolled BAG method. In the second approach (dubbed as Controlled BAG), the network ensures that the total bandwidth assigned to a mobile station, via its available interfaces, does not exceed that of the agreed SLA.

Fig. 2 shows the ratio of the individual bandwidth actually used by each mobile station to that of its agreed SLA. The figure considers the case of three populations of mobile users. The figure shows that when BAG is not controlled, some mobile stations get up to three times their agreed bandwidth depriving others from having access to the bandwidth they subscribed for. 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, all mobile nodes are provided with bandwidths equal to that of their SLA. This demonstrates that the BAG control mechanism makes efficient use of the aggregate bandwidth of the three simulated bases stations. In the absence of such BAG control mechanism, the

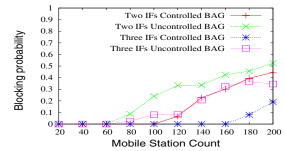

Fig. 3. Blocking probability for a different number of mobile stations.

system ends up by allocating 300% of SLA to few MSs, 200% of SLA to others 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 and 200 MSs) and the network resources become scarce, the BAG control mechanism rejects requests of some mobile nodes but its performance remains comparatively much more outstanding than that of the uncontrolled BAG approach.

Fig. 3 plots the blocking probability of MSs for different numbers of mobile stations. Based on the number of wireless technologies in use, two scenarios are considered: Use of two interfaces and use of three interfaces. The goal behind this experiment is to investigate the impact of the number of deployed interfaces on the system scalability. The figure shows 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 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 that of their SLA. This renders the ISP unable to control its own resources. This ultimately results in an unfair service and high blocking probability.

C. BAG Control Performing Entity

Having demonstrated the need for a BAG control mechanism to ensure an efficient QoS negotiation system, our next step is to determine the best entity that should be performing such BAG control operation. Limiting our studies to the case where BAG is allowed among only BSs belonging to the same domain, the BAG control mechanism can be carried out at the QGS of the domain in which a MS is currently residing. If the visited domain is different than the home domain, the QGS of the visited domain will request the QGS of the home domain for the SLA of the MS. This operation is performed only once, precisely upon the entrance of the MS to the visited domain, and shall cause no signaling overhead to the system.

IV. Scheduling Strategy

As mentioned previously, BAG results in an additional delay at the receiver side due to the packet reordering issue. Consequently, some of the packets of the real-time video applications experience delays higher than their timers and ultimately get discarded. To cope with this issue, as mentioned earlier, there are several scheduling strategies such as Weighted Round Robin (WRR), Weighted Interleaved Round Robin (WIRR), Surplus Round Robin (SRR) and the most recently developed Earliest Delivery Path First algorithm (EDPF) [4]. EDPF is the most notable scheduling algorithm to deliver packets in order

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through multiple paths, minimizing the packet reordering delay at the receiver. It bases its scheduling on a prior knowledge of the available bandwidth at each interface. The key idea behind EDPF algorithm lies on the estimation of the delivery time of the next packet through each path. Using this estimation, EDPF transmits the packets via the path with the earliest delivery time. The delivery time is estimated as follows:

$$d_i{}^l = MAX(a_i + D_l,\, A_l) + \frac{L_i}{B_l} \tag{1}$$

where d_i^l, a_i, and D_l denote the delivery time of packet i through path l, the time at which packet i arrives at the proxy, and the delay from the proxy to the BS along the path l, respectively. A_l, L_i, and B_l denote the time instants when path l will be available for next transmission, the size of packet i, and the bandwidth of path l. Thus, the first component computes the time at which the transmission can begin at the BS on path l, and the second component computes the packet transmission time from such a BS.

In the envisioned QoS negotiation system, we consider a time-slotted approach for bandwidth allocation at the BSs. In other words, each MS is allocated a specific period of time to use the wireless channel. At any given time, only one MS is allowed to transmit/receive data through a particular BS. The size of the time-slot allocated to a given MS through a particular BS corresponds to the bandwidth agreed for the MS divided by the total bandwidth of the wireless link. As a result, the time-slot size varies from an MS to another. The BS has knowledge on the specific beginning and ending times of the time-slot for each MS attached to it. Using these two parameters, the network proxy can make an accurate estimate of the delivery time of the next packet for each MS through each available path. Given the fact that the EDPF scheme is a generic protocol in its nature and does not clearly describe how the delivery time of the next packet is computed, as improvements to the original design of EDPF we suggest the use of these two parameters for an accurate computation of the delivery time of the next packet. The enhanced version of EDPF is dubbed Time-Slotted Earliest Delivery Path First (TS-EDPF) and the suggested modifications are described hereunder.

The delay on a path l from the proxy to the BS denoted by D_l in Eq. (1) should include the total sum of queuing delay, transmission time and link propagation delay to the next entity for all entities along the path from the Network Proxy (including itself) to the BS. In this way, we can estimate the packet delivery time more accurately.

We should now ensure that the time, at which the transmission can begin at a BS, is within the slot time assigned to a MS by the BS. Let $[X_l,\, Y_l]$ be the time-slot period for the MS through path l, and S_i^l be the time at which transmission of packet i can begin at the BS on path l. Furthermore, let $Start(S_i^l, l)$ 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 $[X_l,\, Y_l]$.

$$S_i{}^l = MAX(a_i + D_l,\, A_l) \tag{2}$$

$$Start(S_i{}^l, l) = \begin{cases} S_i{}^l & if\ S_i{}^l \in [X_l,\, Y_l] \\ X_l & otherwise \end{cases} \tag{3}$$

where X_l is the starting time of the subsequent time-slot.

Similarly, we should ensure that the transmission of any packet i at the BS is completed within the time interval $[X_l,\, Y_l]$. Let E_i^l be the time at which transmission of packet i can finish at the BS on path l. Let $Finish(E_i^l, l)$ denote the function that returns the next valid time at which the transmission of the packet i can finish at the BS on path l based on the time slot $[X_l,\, Y_l]$. Both E_i^l and $Finish(E_i^l, l)$ can be expressed as follow:

$$E_i{}^l = Start(\,MAX[a_i + D_l,\, A_l],\, l) + \frac{L_i}{B_l} \tag{4}$$

$$Finish(E_i{}^l, l) = \begin{cases} E_i{}^l & if\ E_i{}^l \in [X_l,\, Y_l] \\ X_l + \frac{L_i}{B_l} & otherwise \end{cases} \tag{5}$$

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

$$d_i{}^l = Finish(\,Start[\,MAX(a_i + D_l,\, A_l),\, l] + \frac{L_i}{B_l},\, l) \tag{6}$$

This algorithm schedules the packet i on path p where

$$p = \{\, l : d_i{}^l \,<\, d_i{}^m,\, l \,\leq\, m \,\leq\, N\} \tag{7}$$

Here N is the number of interfaces. p is the path through which packet i is delivered at the earliest. Then the value of A_p is updated to p, the time when the BS on link p is available for starting the transmission of the next packet.

V. Experimental Evaluation

Having described our TS-EDPF scheduling algorithm, we now evaluate its performance and compare it against the performance of other scheduling algorithms such as WRR, WIRR, and the original EDPF.

In our QoS architecture, after each successful bandwidth negotiation between a MS and the network, the network proxy is informed of i) the amount of bandwidth negotiated for the MS, ii) BS from which the MS will receive the service, and iii) the time-slot assigned to the MS. We perform several simulations using Network Simulator (NS2). We consider one MS equipped with three interfaces that correspond to different wireless technologies supported by the same ISP in a single domain as shown in Fig. 1. The MS has an aggregate bandwidth of 640 Kbps. It initiates a video streaming application at this data rate from a video server.

Two scenarios are studied for evaluating the proposed scheduling method, namely Scenario 1 and Scenario 2. The time slots in Scenarios 1 and 2 are set to 1s and 0.1s, respectively.

Table I shows the results for both scenarios. The buffer size reflects the largest number of packets that were queued in the buffer awaiting playback. The reordering delay indicates the delay in playback due to packet reordering. The bandwidth ratio indicates how much bandwidth the end-terminal could indeed use out of the negotiated and agreed upon bandwidth. The results of the table demonstrate that the proposed TS-EDPF scheme outperforms the three other

Table I
Comparison among scheduling algorithms.

	Scenario 1: Δ = 1s			Scenario 2: Δ = 0.1s		
Algorithm	buffer size (pkts)	reordering delay (ms)	Bw ratio (%)	buffer size (pkts)	reordering delay (ms)	Bw ratio (%)
WRR	38	627	99.22	7	104	99.78
WIRR	39	640	99.19	5	79	99.81
EDPF	33	625	99.19	5	75	99.78
TS-EDPF	1	19	99.87	1	19	99.88

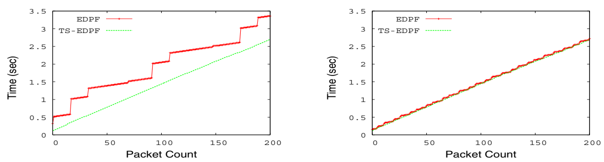

Fig. 4. Playback time of packets

schemes in terms of the three quantifying parameters. Indeed, the proposed scheme ensures high utilization of the network resources while minimizing the number of packets received out of order and thus reducing the associated reordering delay. This good performance is attributable to the time-slot based policy enforcement strategy adopted by the TS-EDPF and lacking in the other three schemes. The results of the table indicate also that all schemes achieve fairly high throughput. However, we notice the impact of the length of the time slot on the bandwidth ratio obtained in case of WRR, WIRR, and EDPF, whereas the throughput achieved in case of TS-EDPF remains stable and not sensitive to the length of the slot-time.

Fig. 4 shows the actual playback time of the first two hundred packets delivered by the two algorithms; EDPF and TS-EDPF. In case of small values of the time slot length (e.g., Δ = 0.1s), both schemes exhibit the same behavior. However, when the time slot is set to larger values, the TS-EDPF scheme outperforms the EDPF scheme. Indeed, the playback of packets remains steady when TS-EDPF is in use, regardless of the time slot length. To illustrate the idea with more clarity, we plot the reordering delay in playback experienced by packets in Fig. 5. For the sake of better plot, we present the case of (Δ = 0.1s).[1] From Fig. 5, we notice that among the first two hundred packets only two packets arrived in out-of-order in case of TS-EDPF. However, in case of the original EDPF, a quite number of packets arrived out of order and this resulted in a relatively higher reordering delay compared to TS-EDPF.

VI. Concluding Remarks

In next generation wireless networks, mobile users will be equipped with several wireless interfaces that will enable them to get connected to different wireless networks. In the context of SLS management, this will create some issues. Indeed, some users may achieve throughputs higher than what they deserve while others get their SLS requests blocked. This will lead to an unfair service. To cope with such an issue, this paper argued the addition of a bandwidth aggregation

[1]In case of Δ = 1s, the packet reordering delay experienced in EDPF is significantly larger than that in case of TS-EDPF (as it can be inferred from Fig. 4-a).

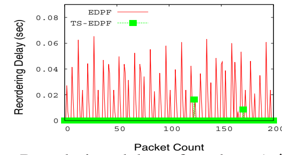

Fig. 5. Reordering delay of packets (Δ = 0.1s).

control mechanism to current QoS negotiation systems. To cope with the packet reordering issue, introduced by the use of multiple heterogeneous channels for communication, a fixed slot-time based policy enforcement strategy is added to the Earliest Delivery Path First scheduling mechanism. Extensive simulations were conducted and encouraging results were obtained. Indeed, the results demonstrated that in the presence of the proposed bandwidth aggregation control mechanism, the system tends to be more scalable and fair. The simulations also illustrated that employment of a time-slotted approach for bandwidth allocation largely mitigates the packet reordering issue and the associated playback delay. Finally, the actual implementation of the proposed QoS negotiation architecture along with the proposed mechanisms forms the focus of our future research work.

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