

Improved rtPS Scheduling with QoE metrics in Joint WiMAX / Satellite Networks

Anastasia Lygizou, Spyros Xergias and Nikos Passas

National and Kapodistrian University of Athens

Dept. of Informatics and Telecommunications

Panepistimiopolis, Ilissia

Athens 15784

University of Athens, Greece

Abstract—This paper improves a previously proposed scheduling algorithm that is responsible to share the allocated capacity to the uplink traffic of an integrated satellite and WiMAX network. The target of this improvement is to schedule traffic of real time connections based on Quality of Experience (QoE) metrics. After a bibliographic search on QoE metrics, the FC-MDI (Frame Classification-Media Delivery Index) metric is improved in order to be used in two proposed alternative algorithms for the scheduling of real time connections. Simulation results show that the proposed algorithms, especially the one with rate adaptation, considerably improve the QoE and the mean delay of real-time connections.

Keywords— DVB-RCS, WiMAX, QoE, rtPS, scheduling.

I. INTRODUCTION

IEEE 802.16 [1] is a standard that aims at filling the gap between local and wide area networks, by introducing an advanced system for metropolitan environments. In this system, also known as WiMAX, both point-to-multipoint (cellular) and mesh mode configurations can be supported, while node mobility is also covered by amendment 802.16e [2]. One of the main advantages of the standard is the large degree of flexibility it provides by supporting a wide range of traffic classes with different characteristics and quality of service (QoS) requirements. This is attained through a large set of parameters that allow users to describe in detail their traffic profiles and service needs. On the other hand, Digital Video Broadcasting – Return Channel Satellite (DVB-RCS) [3] is an open standard for bi-directional transmission of digital data over the satellite network. It employs satellite transmission using combinations of C, Ku and Ka bands. DVB-RCS is a fully mature open, satellite communication standard with highly efficient bandwidth management, making it a cost-efficient alternative in many cases. It mainly describes the uplink direction of a satellite network, providing advanced QoS capabilities for requesting and acquiring capacity for demanding services.

The advantage of combining the two technologies is that a satellite network can be used for interconnecting WiMAX islands with the Internet and avoiding layout of expensive backbone infrastructures. This can provide reliable solution, especially in rural areas or locations affected by

environmental factors, e.g. islands, mountains, etc. However, a satellite network experiences large round trip delays that can deteriorate quality especially for real-time applications. In [4], we have investigated how the two networks can co-operate, especially in terms of QoS, in order to reduce end-to-end delays and packet losses due to expiration. In this work, we extend [4] towards improving a part of the proposed mechanism which shares capacity to real-time connections of the WiMAX network based on the use of QoE metrics. Quality of Experience (QoE) is the overall performance of a system from the users' perspective.

The paper is organized as follows. Section II presents the main characteristics of WiMAX and DVB-RCS, focusing on their QoS capabilities. Section III presents a brief state-of-the-art on QoE metrics, their use in management tools and our proposal for using in our mechanism. Section IV describes the basic architecture of our mechanism, our previous proposal, as well as the proposed improvements in the sharing of capacity to real-time traffic based on the above QoE metric. Section V contains the description of the simulation model used for evaluation purposes together with the obtained results. Finally, section VI contains our conclusions.

II. QoS PROVISION IN WiMAX AND DVB-RCS

A. QoS provision in WiMAX

The system architecture consists of Base Stations (BSs), each one responsible for a specific area cell, and stationary Subscriber Stations (SSs). The communication path between SSs and BS has two directions: uplink (from SSs to BS) and downlink (from BS to SSs), multiplexed either with Time Division Duplex (TDD) or Frequency Division Duplex (FDD). Transmission parameters, including the modulation and coding schemes, may be adjusted individually for each SS on a frame-by-frame basis. A TDD frame is divided into a downlink subframe, and an uplink subframe. The TDD framing is adaptive in that the bandwidth allocated to the downlink versus the uplink direction may vary.

The downlink subframe begins with information necessary for frame synchronization and control. This is followed by the frame control section, containing the DL-MAP and UL-MAP fields, that state the physical slots (PSs) at which bursts begin

in both directions. Through the UL-MAP, the BS determines the transmission opportunities of its subordinates SSs, based on the bandwidth requests of each SS. Bandwidth requests are transmitted through special purpose information elements referred as BW-Requests. Each SS having decoded the corresponding control information contained in the UL-MAP, knows exactly during which PSs of the uplink subframe it is allowed to transmit and what kind of transmission it can make.

WiMAX can support multiple communication services (data, voice, video, etc.) with different QoS requirements organized into different connections. Each connection is associated with a single service flow and specifies a set of traffic and QoS parameters that quantify its traffic behavior and QoS expectations. This set includes the minimum reserved traffic rate (in bits/sec), the maximum sustained traffic rate (in bits/sec), the maximum latency (in ms), the tolerated jitter (maximum delay variation in ms), the traffic priority (values 0-7, with 7 the highest), etc.

The standard defines five different services. The first one is the Unsolicited Grant Service (UGS) that supports real-time data streams consisting of fixed-size data packets transmitted at periodic intervals, such as Voice over IP without silence suppression. The second one is Real-time Polling Service (rtPS) that supports data streams consisting of variable-sized data packets that are transmitted at fixed intervals, such as MPEG video. These applications have specific bandwidth requirements, as well as a maximum acceptable latency. Late packets that miss the deadline are considered useless. The third one is extended rtPS (ertPS), that better supports real-time service flows that generate variable size data packets on a periodic basis, e.g., VoIP with silence suppression. The fourth one is Non-real-time Polling Service (nrtPS). This service is for non-real-time connections that require better than best effort service, e.g., bandwidth intensive file transfer. These applications are time-insensitive but require a minimum bandwidth allocation. Finally, the Best Effort service (BE) is for best effort traffic with no QoS guarantee. The applications of this kind of service share the remaining bandwidth after allocation to the rest of the services is completed. BE uses only contention mode.

The traffic scheduler located at the BS decides on the allocation of the physical slots in each time frame. Uplink scheduling is performed by the BS with the aim of providing each SS with enough bandwidth for uplink transmissions or opportunities for extra transmission requests.

B. QoS provision in DVB-RCS

According to the DVB-RCS standard [3], a satellite network consists of: a geostationary (GEO) satellite, Return Channel Satellite Terminals (RCSTs) and the Network Control Centre (NCC) responsible to control transmissions to/from the RCSTs.

The satellite access scheme is based on Multi-Frequency Time Division Multiple Access (MF-TDMA). MF-TDMA allows a group of RCSTs to communicate with the NCC using a set of carrier frequencies, each of which is divided into time-slots. The timeslots of the return link are organized and

numbered so that the network is able to allocate them to individual RCSTs. These timeslots are organized in frames, which are then organized in superframes.

The NCC allocates to each active RCST a series of TRAFFIC bursts (bursts that are used for carrying useful data from the RCST to the Gateway(s)/RCST), each defined by a frequency, a bandwidth, a start time and a duration, in order to avoid collisions between the terminals.

Whenever required, each RCST issues a capacity request and the NCC allocates the capacity through the Terminal Burst Time Plan (TBTP) at the beginning of every superframe. The standard defines five capacity request categories. Continuous Rate Assignment (CRA) is a capacity provided in full for each and every superframe, without the need for requests. Such capacity shall be negotiated directly between the RCST and the NCC. Rate Based Dynamic Capacity (RBDC) is rate capacity which is requested dynamically by the RCST. Each request overrides all previous RBDC requests from the same RCST, up to a maximum rate limit negotiated directly between the RCST and the NCC. To prevent a terminal anomaly resulting in a hanging capacity assignment, the last RBDC request received from a given terminal shall automatically expire after a time-out period equal to 2 superframes. Volume Based Dynamic Capacity (VBDC) which is requested dynamically by the RCST (in slots per frame), in a cumulative way. (i.e. each request is added to the total volume request of the same RCST). The cumulative total per RCST shall be reduced by the amount of this capacity category assigned in each superframe. Absolute Volume Based Dynamic Capacity (AVBDC) which is requested dynamically by the RCST (in slots per frame), in an absolute way. Finally, Free Capacity Assignment (FCA), which is capacity assigned in this category is intended as bonus capacity which can be used to reduce delays on any traffic which can tolerate delay jitter.

III. STATE-OF-THE-ART OF QUALITY OF EXPERIENCE

A. Categorization of QoE metrics

QoE reflects the overall performance of a system from the users' perspective. QoE is related to but differs from QoS, which embodies the notion that hardware and software characteristics can be measured, improved and perhaps guaranteed. In contrast, QoE expresses user satisfaction both subjectively and objectively, which are the two main categories of QoE metrics [5].

Subjective quality metrics represent the most accurate method for obtaining quality ratings. In subjective experiments, a number of "subjects" (typically 15-30) are asked to watch a set of video clips and rate their quality. The average rating over all viewers for a given clip is also known as the Mean Opinion Score (MOS), ranging from 1 (poor) to 5 (excellent), which is the main subjective metric. There exist standard methods for conducting subjective video quality evaluations, such as the ITU-R BT.500-11 [6]. Some variants included in the standard are: Double Stimulus Impairment Scale (DSIS), Double Stimulus Continuous Quality Scale

(DSCQS), Single Stimulus (SS), Single Stimulus Continuous Quality Evaluation (SSCQE), Stimulus Comparison Adjectival Categorical Judgement (SCACJ), Simultaneous Double Stimulus for Continuous Evaluation (SDSCE) and Absolute Category Rating (ACR). The differences between them are minimal and mainly depend on the particular application considered.

Objective quality metrics are algorithms and formulas that measure, in a certain way, the quality of a stream. The most commonly used objective measures for video are: Mean Squared Error (MSE) and Peak signal to noise ratio (PSNR) (which is a logarithmic representation of MSE), ITS' Video Quality Metric (VQM) [7], EPFL's Moving Picture Quality Metric (MPQM) [8], Color Moving Picture Quality Metric (CMPQM) [4], Normalization Video Fidelity Metric (NVFM) [8], and Structural Similarity Index Metric (SSIM) [9]. With a few exceptions, objective metrics propose different ways of comparing the received sample with the original one, typically by computing a sort of distance between both signals.

Finally, a hybrid approach between subjective and objective evaluation has been proposed in [10]. It is a technique that allows approximation of the value obtained from a subjective test but automatically. In more detail, Pseudo-Subjective Quality Assessment (PSQA) metric starts by selecting the factors that may have an impact on the quality, such as: codec, bandwidth, loss, delay, and jitter. Then these factors are used to generate several distorted video samples. These samples are subjectively evaluated by a panel of observers. The results of the observations are then used to train a Random neural network (RNN) in order to capture the relation between the factors that cause the distortion (objective approach) and the perceived quality by real-human (subjective approach).

B. The use of QoE metrics in management tools

There is a large number of papers that use QoE metrics to measure video quality but very few that use QoE metrics for QoE management. [11] is the only work in UMTS that investigates the possibility of using QoE as metric for scheduling decision. In order to get QoE feedback in real time, the PSQA technique is used. Loss rate (LR) of video packet and mean loss burst size (MLBS) are considered as the quality-affecting parameters for the training of RNN. MLBS parameter is the average length of a sequence of consecutive lost packets in a period of time and captures the way losses are distributed in the flow as this affects dramatically the perceptual quality of the video. After the training of RNN, MOS is estimated in real time, so that the scheduler can get MOS scores for making scheduling decision. Two algorithms are proposed, the QoE-CI and the QoE-PF. The first one has the objective of maximizing system throughput while taking into account the quality of experience of video-streaming users. The second one has the goal to maximize fairness between users while keeping QoE of video users acceptable. The main conclusion of this paper is that QoE-PF is fairer than QoE-CI because it takes into account the average throughput of each station, while QoE-CI gives higher quality

score than QoE-PF since it privileges station with better signal condition. [12] and [13] propose a novel rate-adaptation mechanism based on quality of experience, using PSQA tool for obtaining mean opinion score in real-time. The parameters used in PSQA are the loss rate of the I frames, loss rate of the P frames, loss rate of the B frames, and the MLBS of the I frames. The idea of the proposed scheme is to use QoE feedback from mobile stations to provision the current condition of the network and then adapt the rate accordingly. In [14], a novel packet scheduling algorithm for multi-hop wireless networks that jointly optimizes the delivery of multiple video, audio, and data flows according to the QoE metrics is developed. A previously proposed model to determine user satisfaction is used, where quality is given in terms of the PSNR, while MOS is produced through a non-linear curve mapping PSNR to MOS. The proposed scheduler locates sets of packet combinations across all active flows of all users that pass the node that would satisfy a given buffer reduction. For each of these combinations, an estimation of the user satisfaction expressed in MOS decrease for each flow is calculated. The scheduler then drops the packets whose combination results in the smallest decrease in QoE satisfaction based on a proposed cost function. [15] proposes the same scheduler for multi-service scheduling with an emphasis on video content, but also for audio communication and file transfer. There is also difference in the model that determines user satisfaction in video content, which is based on a previously proposed video distortion model. The computed distortion is mapped to PSNR, and then mapped to a MOS score using a non-linear relation.

C. QoE metrics in our mechanism

The target of this paper is to improve a previously proposed mechanism, in order to make the scheduling of rtPS connections based on the use of QoE metrics. QoE metrics are usually used for the assessment of the transmission of video on different network conditions, and rarely used in scheduling solutions, while they have never been used till now for scheduling in satellite networks. Subjective metrics are the most accurate for QoE measurements, as they are evaluated by real-human. Their main shortcoming is that they are time-consuming and high-cost in man power. Thus, they cannot be easily repeated several times nor used in real-time (being a part of an automatic process). As we need the proposed improvement to be part of an automatic procedure, subjective and hybrid QoE metrics are excluded in our case. From the already proposed solutions in other kind of networks, the solutions proposed in [11-13] have the drawback of using the PSQA metric for scheduling and QoE management. On the other hand, the solution proposed in [14-15] is considered complex, as it calculates the QoE produced by every possible packet dropping. Our proposal aims to be simpler in order to be used in satellite networks, which have the drawback of delays. For all these reasons, the FC-MDI metric was chosen to be used in the existing mechanism [18], as it is an objective metric that gives a different weight to the loss of different categories of voice and video frames. The FC-MDI (Media

Delivery Index based on Frame Classification)metric is an extension of the MDI (Media Delivery Index) metric [16], an objective metric that contains two numbers separated by colon: the delay factor (DF) and the media loss rate (MLR). DF is time value indicating how many milliseconds the buffer must be able to contain to eliminate jitter, while MLR is computed difference between number of media packets received during an interval and number of media packets expected during an interval, everything scaling in the value of one second. Because the MLR is a rate, some important information is lost, such as whether the IP packets lost are consecutive or not. It does not consider the quality degradation that suffered some propagated loss from previous temporally related frames, so [17] proposes FC-MDI which takes frame classification into account to improve the performance of the MDI measurement. It distinguishes the packet loss based on the frame classification, and gives the different frame a different weight. In all types of frames, I-frame plays the most important role, as the rest frame of the whole group of picture (GOP) cannot decode normally if the I-frame is lost. Compared with B-frame, P-frame relies less on its previous I-frames and P-frames. FC-MLR (Media Loss Rate based on Frame Classification) improves the definition of the MLR and takes frame classification into account as follows:

$$\frac{I_{P_{Loss}} \alpha + P_{P_{Loss}} \beta + B_{P_{Loss}} \gamma}{I_{P_{Loss}} + P_{P_{Loss}} + B_{P_{Loss}}}$$

where α, β, γ are weights with ($3 \geq \alpha > \beta > \gamma \geq 0, \alpha + \beta + \gamma = 3$) and $I_{P_{Loss}}, P_{P_{Loss}}$, and $B_{P_{Loss}}$ are respectively the number of lost I, P and B frames. The results of experiments demonstrate that when two videos of different qualities have a same number of total dropped-packet, the traditional MDI measurement cannot tell the difference between them, as MDI does not take into account the quality degradation that suffers some propagation loss from previous temporally related frames, while FC-MDI possesses a distinguishing feature.

The FC-MDI takes frame classification into account by giving different weights to the number of I-frames lost, P-frames lost and B-frames lost. However, it does not take into account if the frames lost from a specific category are consecutive or not, which makes a difference. In this paper, the LA-MDI is proposed (which is an improvement of FC-MDI), in order to give a greater importance to the consecutive lost frames of a specific category. In the LA-MDI, the definition of the DF is the same with its definition in the simple MDI, where the LA-MLR improves the definition of the FC-MLR in order to take into account the consecutive lost frames as follows:

$$\frac{ngI \alpha + ngP \beta + ngB \gamma}{ngI + ngP + ngB}$$

where α, β, γ are weights with ($3 \geq \alpha > \beta > \gamma \geq 0, \alpha + \beta + \gamma = 3$), $I_{P_{Loss}}, P_{P_{Loss}}$, and $B_{P_{Loss}}$ are respectively the number of lost I, P, B frames, and ngI, ngP , and ngB are respectively the number of group of lost I, P, B frames. In order to explain better the role of the number of group of lost frames of a specific category, suppose that in a sequence of frames of a specific category, the frames sent at position 2, 5, 7, 9, 10, ... , are lost,

then the next loss vector 1011010100... is get. In this loss vector the number of group of lost frames is four. In the opposite, suppose the frames sent at position 2, 3, 4, 5, 6, ... , are lost, then the next loss vector 1000001111... is get, and the number of group of lost frames is one. Both cases have the same FC-MLR, while the value of the LA-MLR in the first case is lower than in the second one. The greater the number of group of lost frames, the more dispersed the lost frames are, and so the QoE is better. Generally, as the FC-MLR and the LA-MLR grows, the QoE becomes worse as the number of lost frames increases.

IV. PROPOSED SCHEDULING SOLUTION

A. General operation

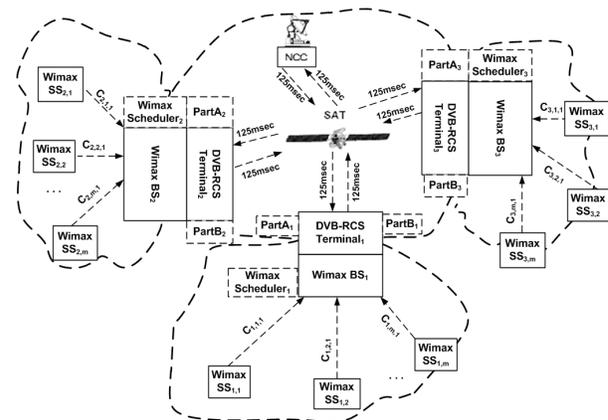


Fig. 1 Network architecture with three BS/RCST

In [4], an interconnection of a satellite and a WiMAX network is proposed, assuming that one or more of the RCSTs are also WiMAX BSs serving a number of SSs as shown in Fig. 1. This integrated scheduling provision mechanism consists of three main parts:

PartA is an entity at the RCST/BS that makes the capacity requests following a prediction-based approach, PartB is an entity at the NCC that allocates resources and creates the TBTP, while PartC is an entity at the RCST/BS that shares the given capacity among its WiMAX subscribers. PartB accepts the capacity requests made from all PartAs, processes them and creates the TBTP in order to allocate the capacity of a superframe among the different RCSTs. The architectural elements of RCST are shown in Fig. 2. PartC, located at the RCST/BS, contains the scheduling algorithm that is responsible to share the allocated capacity, to the uplink traffic arriving from the WiMAX network. In more detail, PartC classifies uplink traffic arriving from the SSs into five queues: UGS_queue, rtPS_queue, ertPS_queue, nrtPS_queue, BE_queue based on each packet's QoS service type. It then interprets TBTP (knows exactly which slots has been assigned to it) and selects which packets will be transmitted. This selection is made based on a priority scheme: it first selects packets from the UGS_queue, then from the rtPS_queue, then from the ertPS_queue, then from the nrtPS_queue and finally

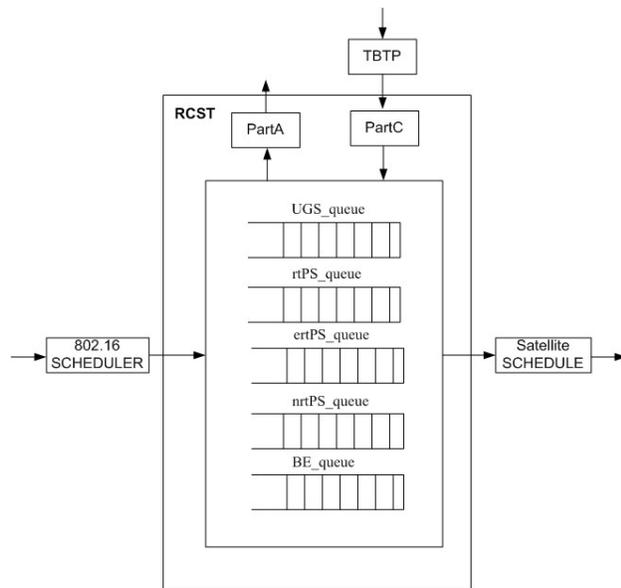


Fig. 2 Architectural elements of RCST

from BE_queue. Finally, it is also responsible to discard packets that are expired based on the deadlines set for their transmission to the satellite network and keep statistics on the packets transmitted and discarded.

In [4] the *RTFS* (Real Time FIFO Scheduler) algorithm treats the transmission of packets of video connections with the logic of a FIFO queue. The packets of all *rtPS* connections are inserted in one queue based on the order of their arrival. During the superframe, the PartC transmits, whenever it has available capacity based on the TBTP, the packets from this queue. A packet is dropped, if it has been expired due to delay. The performance of the mechanism was demonstrated in [4].

In [18] the *FC_MDI* metric is used in order to improve the scheduler of *rtPS* connections. The proposed *FC_MDI_S* algorithm makes the following procedures in the beginning of every superframe: It firsts drops the packets that are expired due to delay factor. Then, it computes the *FC-MLR* value of every connections based on the lost I, P, B frames in the previous superframe. Finally, it sorts the video connections based on the computed *FC-MLR* value of the connections under two versions. The first version is named *FC_MDI_SG* and has a greedy logic. In order to preserve the connections that have good quality, the connections are sorted based on *FC-MLR* value in ascending way, from the best quality to the worst. This will lead to the maintenance of the quality of some connections and the starvation of some other connections. The second version is named *FC_MDI_SF* and has a fair logic. In order to be fair and maintain all connections (even in worse quality), the connections are sorted in the opposite way than the previous version from the worst quality to the best.

During the superframe, the PartC transmits whenever it has available capacity based on the TBTP. The *FC_MDI_SG* version transmits all the packets of one category, giving priority to I frames, then to P frames and last to B frames, and

then moves on to packets of the same category of another connection. On the contrary, the *FC_MDI_SF* version transmits one packet of one category from all connections, and then another packet of the same category from all connections, until exhausting all the packets of this category. After the transmission of all the packets of the previous category, it moves on to the next category giving priority to I frames, then to P frames and last to B frames. Fig.3 presents an example of transmission of packets under the previously proposed versions. In more detail, Fig.3a presents the frames of three connections as they have arrived in the buffer of PartC, Fig.3b presents the transmission of frames under *FC_MDI_SG*

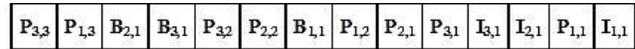


Fig.3a Packets of three connections to the buffer of PartC

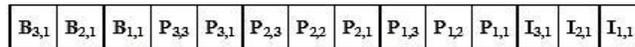


Fig.3b Transmission of packets under *FC_MDI_SG* algorithm

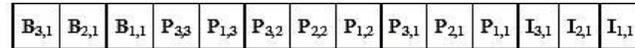


Fig.3c Transmission of packets under *FC_MDI_SF* algorithm

version and Fig.3c presents the transmission of frames under *FC_MDI_SF* version. The pointer *i* in the *Frame_Category_{i,j}* of Fig.3 shows the connection, while the pointer *j* shows the order of the frame of the specific frame category of *i* connection.

The greedy and fair version of the proposed *FC_MDI_S* algorithm have the same performance concerning the goodput and mean delay, as the logic of the versions for sharing capacity is the same. They differentiate in the way they deal with the different connection ids, where the connections have differentiated goodput under the greedy version and equal goodput under the fair version. Finally, simulation results show that the *FC_MDI_S* algorithm improves the QoE performance relatively to the *FIFO_S* algorithm, and it substantially improves the mean delay of the connections.

B. Improvement of *rtPS* scheduling in PartC

The target of this paper is to further improve the scheduling algorithm of PartC, in order to make the scheduling of *rtPS* connections based on the use of the proposed *LA-MDI* metric.

In the beginning of every superframe, the proposed algorithm, referred to as *LAQoE*, drops the packets that are expired due to delay factor. Then, it sorts the video connections based on their mean *LA-MLR*. The mean *LA-MLR* of a connection in superframe *t* is defined as $\frac{\sum_{k=t-T+1}^t LA_MLR_k}{T}$, where *T* is a small number of superframes (time window), in order to reflect the quality of the connection in the recent past. Two alternatives are studied for sorting the connections according to the mean *LA-MLR*. The first alternative is named *LAQoEG* and has a greedy logic. In order

to preserve the connections that have good quality, the connections are sorted based on mean LA-MLR in ascending way, from the best quality to the worst. This maylead to the maintenance of the quality of some connections and the starvation of some others. The second alternative is named LAQoEF and has a fair logic. In order to be fair and maintain all connections (even in worse quality), the connections are sorted in the opposite way than the previous algorithm based on the mean LA-MLR of the connections from the worst

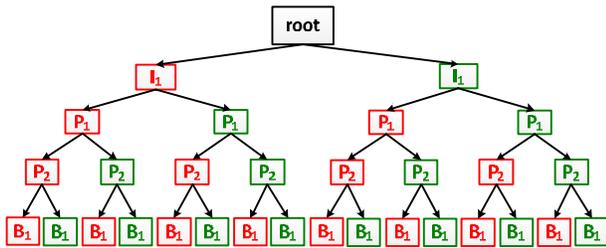


Fig. 4 Example of a QoE tree

quality to the best.

In the beginning of every superframe, the PartC has accepted the TBTP generated from the NCC, so it has the knowledge of the available capacity for transmission. For every connection with the order of the previous sorting, the PartC creates a binary tree named QoE Tree (QoET) based on the available capacity for this connection.

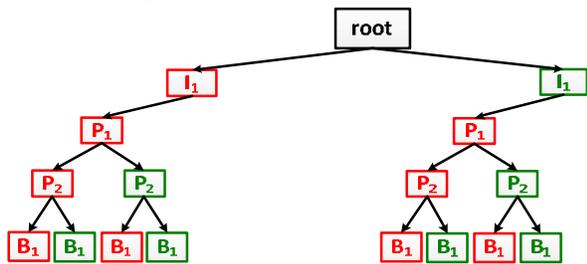


Fig. 5 Example of a QoE tree where frame P1 is expired

PartC knows from PartA the sequence of packets that have arrived during the previous superframe. For every rtPS connection, PartC constructs a QoET that represents the possible combinations of packet transmission in this superframe. If, for example, PartC wants to transmit the sequence of I1P1P2B1 packets, then the QoET that is constructed is shown in Fig. 4.

Every path of the tree represents a combination of packet transmission, where a red node shows that a packet is not transmitted and a green node that a packet is transmitted. Knowing the TBTP, PartC can compute if a packet will expire due to delay before its time for transmission. If the packet expires, then naturally it is not transmitted. Suppose this happens for packet P1, then the constructed QoET is presented in Fig. 5. In addition, the construction of a path stops, if its capacity comes to the available capacity that this connection has for transmission. Suppose that this happens if packets I1P2 are transmitted, then the constructed QoET tree is presented in Fig. 6.

The leaves of the tree also contain the information of the LA-MLR metric for the specific path, which is easy to

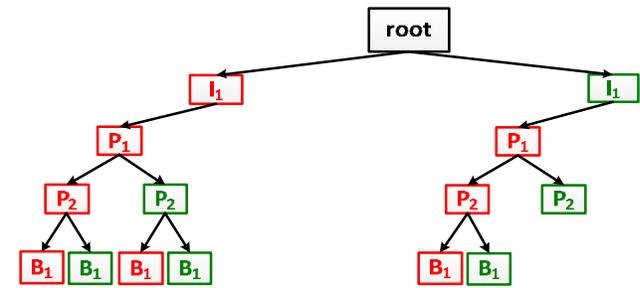


Fig. 6 Example of a QoET tree where packet P1 is expired and the available capacity of the connection is overcome with the transmission of I1 and P2 frames

compute as we know the sequence of lost frames from every different category, as well as the total amount of bytes to be transmitted.

The LAQoE selects the path (sequence of packets) from the QoET of this connection with the best LA-MLR value. The available capacity for the next connection is reduced by the size of transmitted bytes of the selected path.

During the superframe, the PartC transmits, whenever it has available capacity based on the TBTP, the packets from the path selected of a connection based on the order of the sorted connections. If the packets of the selected paths of all connections are transmitted and PartC has still available capacity, then it transmits packets that have arrived in this superframe, using the logic of the FC_MDI_S algorithm. The transmission of these packets as well as the dropping of the packets is measured to the computing of the LA-MLR of the connections to the next superframe.

The LAQoE algorithm with rate adaptation

The LAQoERA algorithm is an improvement of the LAQoE algorithm that makes rate adaptation. PartC has the possibility of transmitting video in three rates: high, medium and low. Low quality is corresponded to rate 1, medium quality to rate 2 and high quality to rate 3. The greater the LA-MLR metric becomes, the worse it is. In the LAQoERA algorithm, the corresponding path of the QoET is not able to be transmitted upon a LA-MLR threshold. Instead, the connection transmits to a lower quality. If it is already in the lowest quality, then the connection transmits the best path that it is able to.

The LAQoERA algorithm differentiates the sorting of video connections, the creation of the QoET and the selection of the transmitting path so as to take into account the rate of video connections. The sorting of the video connections is based on the mean LA-MLR of the connections and the mean rate (mR) of the connections. The mR of a connection in superframe t is defined as $\frac{\sum_{i=t-T}^{t-1} Rate_i}{T}$, where T is a small number of superframes (time window), in order to reflect the rate of the connection in the recent past. The connections are sorted according to the mean LA-MLR and mR under two versions. The first version is named LAQoERAG and has a greedy logic.

In order to preserve the connections that have good quality, the connections are sorted based on mR in descending way, from the best rate to the worst, and then based on mean LA-MLR in ascending way, from the best quality to the worst. The second version is named LAQoERAF and has a fair logic. In order to be fair and maintain all connections (even in worse quality), the connections are sorted in the opposite way than the previous algorithm based on mR in ascending way, from the worst rate to the best, and then based on mean LA-MLR in descending way, from the worst quality to the best.

The difference in the creation of the QoET from the previous algorithm is that there are flags in every path showing if this path is able to be transmitted in rate 3, rate 2 and rate 1. The flag of one rate becomes false only when the capacity of a path overcomes the available capacity of this connection. If the flags of the three rates are false, then the path stops. In addition there are flags in the whole tree showing the existence of a path in rate 3, rate 2 or rate 1.

Finally, the LAQoERA algorithm selects the path (sequence of packets) with the best LA-MLR value in the best rate that this connection has the ability to transmit. This is shown from the flags of the QoET. If the flag of the whole tree in rate 3 is true, then the path with the best LA-MLR metric will be selected (from these paths that have the respective flag in rate 3 set to true). If the LA-MLR metric of the selected path is over a threshold (thresrate3), then PartC prefers to transmit in lower grade but in better quality. The same procedure is repeated for rate 2. If the path selected in rate 2 has the LA-MLR metric over a threshold (thresrate2), then PartC will select the path with the best LA-MLR metric in rate 1.

The available capacity for the next connection is reduced by the size of transmitted bytes of the path selected in the respective rate.

V. SIMULATIONS

In order to measure the performance of the proposed algorithms, we accommodated the simulation program presented in [4]. The program is constructed in C++ and simulates the full operation of WiMAX network, as well as the DVB-RCS for the return link of a satellite network. We use the simulation scenario presented in [4] with three DVB-RCS terminals each one interconnecting a WiMAX network, all with the same number of subscribers (Fig. 1). In the previous simulation scenario, every SS had multiple types of traffic, including video, compressed and uncompressed voice, ftp and http. In order to present the difference of the proposed mechanisms regarding the QoE of the video connections, in the present simulation scenario every SS has only one video connection. The same video trace is used for every SS, in order to present the difference between the greedy and fair versions. The source of this video trace is the "Alladin" film from "http://trace.eas.asu.edu/TRACE/ltvt.html" in high quality ("Verbose_Alladin.dat" file). Especially, for the LAQoERAG and LAQoERAF algorithms, we also use the same video trace in medium ("Verbose_Alladin_10.dat" file) and low quality ("Verbose_Alladin_10_14_18.dat" file).

The time frame length in WiMAX is set to 1msec, the packet size to 54 bytes and the modulation to 64-QAM for all SSs, leading to a transmission speed of 120Mbps (as indicated in the standard). The latency used in the WiMAX network for rtPS connections is 50msec.

The maximum transmission rate in the return link of the satellite network is 6Mbit/s, while the duration of the frame is set to 50msec and the superframe to 500msec, equal to the round trip delay. During the logon phase, each RCST terminal sets the CRA_level equal to zero (in order to present the difference between the quantity of the requested slots), the

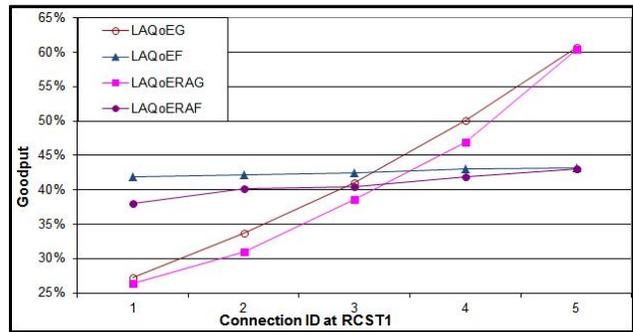


Fig.7 Goodput per connection id for five connections per SS

RBDC_max to 700kbps, the RBDC_timeout equal to 2 and the VBDC_max equal to 11 slots per frame. The latency used in the satellite network is 300msec. The values for thresrate3 and thresrate2 are set equal to 2.5, while the values of weights are $\alpha=1.75$, $\beta=1$, $\gamma=0.25$. Finally, the value of time window T used in mean LA-MLR and mean Rate in LAQoE algorithm are set equal to 3.

The greedy and fair versions of the proposed algorithms have the same performance concerning the goodput, mean delay and loss rate, as the logic of the algorithms for sharing capacity is the same. They differentiate in the way they deal with the different connection ids. Fig. 7 presents the goodput per connection id for five connections per SS. This figure shows the different performance of the greedy and the fair

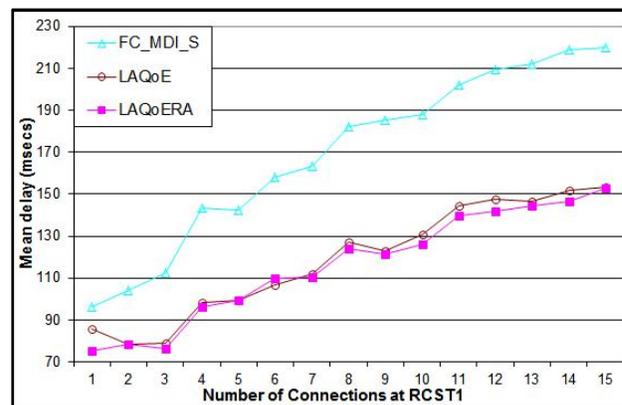


Fig. 8 Mean delay per proposed algorithm

algorithms, where the connections have differentiated goodput under the greedy versions LAQoEG and LAQoERAG.

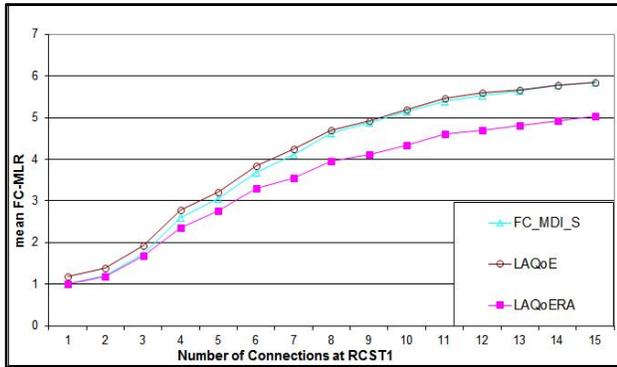


Fig. 9 Mean FC-MLR per proposed algorithm

On the other hand, the fair versions *LAQoEF* and *LAQoERAF* have equal goodput, which is less than the goodput of the best connection id and better than the goodput of the worst connection id of the greedy versions. It is due to the operator of the system to choose between them.

Fig.8 presents that the *LAQoE*, and *LAQoERA* algorithms reduce the mean delay of the video connections. This is due to

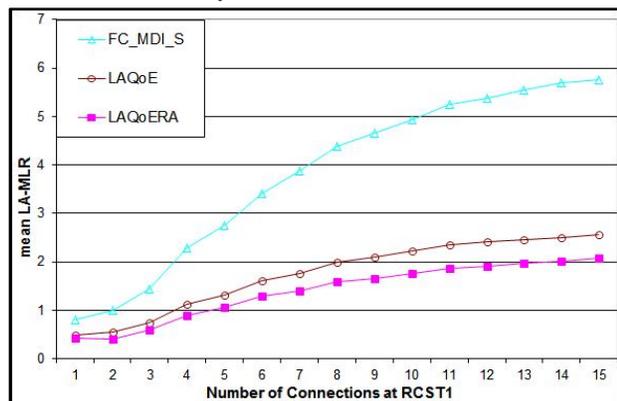


Fig. 10 Mean LA-MLR per proposed algorithm

the philosophy of the algorithms that take into account the TBTP to the construction of QoET and the selection of packets for transmission with the best QoE metric. This is a substantially improvement, as we prefer video connections to have reduced delay.

The two lastly proposed algorithms use the LA-MLR metric for their QoE evaluation. Fig. 9 shows that the *FC_MDI_S* and the *LAQoE* algorithms have the same mean FC-MLR value, while Fig. 10 shows that the *LAQoE* algorithm improves the LA-MLR value regarding to the *FC_MDI_S* one, as it takes account the number of group of lost frames of different categories. This is a proof of the differentiation and improvement of the LA-MLR metric. Fig. 10 presents the mean LA-MLR value for all connections of a SS. This figure shows that the two lastly proposed algorithms substantially improve the QoE performance of the video connections. Especially, the *LAQoERA* algorithm has the best QoE performance. This is due to the rate adaptation of this algorithm, which loses the least of the transmitted information.

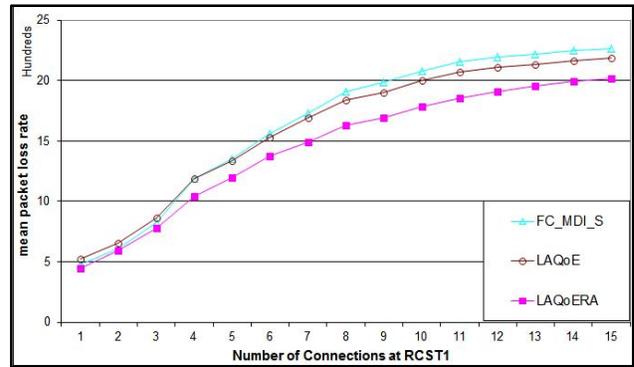


Fig. 11 Loss rate per proposed algorithm

It may transmit in lower quality but it transmits more information. This is better presented in Fig. 11, which presents the percentage of lost bandwidth. Fig. 12 presents the mean rate of the connections of a SS, which shows how the mean rate of the connections in *LAQoERA* algorithms is reduced as the number of connections of a SS is increased.

From the presented results, we conclude that the *LAQoE* algorithm further reduces the mean delay of the connections, and improves the QoE performance of the video connections

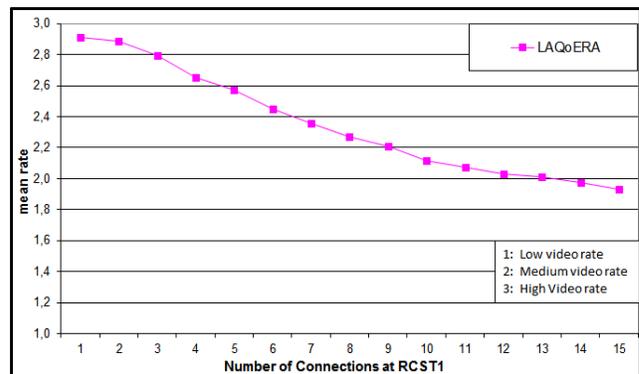


Fig. 12 Mean rate of the connections of a SS

relatively to the *FC_MDI_S* algorithm. This is due to the philosophy of this algorithm which serves the sequence of packets with the best QoE metric. Finally, the *LAQoERA* algorithm has the best mean delay and QoE performance for video connections, as it loses less of the transmitted information due to the rate adaptation that it makes.

VI. CONCLUSION

In this paper, we improve a previously proposed scheduling algorithm, named *FC_MDI_S*. This algorithm is responsible to share the allocated capacity to the uplink traffic arriving from the WiMAX network in an integrated satellite/WiMAX network. *FC_MDI_S* algorithm makes the scheduling using the *FC_MDI* QoE metric. This is considered novel, as QoE metrics are mainly used for the assessment of video quality and not for scheduling. Especially in satellite networks, QoE metrics have never been used in management

tools. We propose an improvement of the FC_MDI metric named LA_MDI. We propose and evaluate two alternative algorithms based on this new metric named *LAQoE* and *LAQoERA*. The second algorithm is an improvement of the first one that also makes rate adaptation. Simulation results show that the proposed algorithms, and especially the second one, considerably improve the QoE of video connections and reduce their mean delay.

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