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A study of multicast congestion control for UMTS

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SUMMARY

In this paper, we study the applicability of multicast congestion control over universal mobile telecommunications system (UMTS) networks. We analyze two well-known multicast congestion control schemes for fixed networks, namely TCP-friendly multicast congestion control and pragmatic general multicast congestion control. We investigate their behavior when they are employed in UMTS networks and we analyze the problems arose when these mechanisms are applied over the wireless links of the UMTS terrestrial radio-access network. Additionally, we propose necessary improvements to these legacy schemes and explain the necessity of these modifications. The proposed schemes are implemented in the ns-2 network simulator and are evaluated under various network conditions and topologies. Finally, we measure the performance of the proposed modified schemes and compare them with the corresponding legacy mechanisms. Copyright © 2009 John Wiley & Sons, Ltd.

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1. INTRODUCTION

Multicast is an efficient method for data transmission to multiple destinations using fewer network resources. Its advantage is that the sender's data are transmitted only once over the links that are shared along the paths to a targeted set of destinations. 3rd Generation Partnership Project (3GPP) recognized the need for the support of real-time multicast multimedia transmission in cellular networks and initiated the standardization of multimedia broadcast/multicast service (MBMS) framework of universal mobile telecommunications system (UMTS) [1, 2].

Congestion control is a policy that adapts the source transmission rate according to the network congestion. In IP multicast, User Datagram Protocol (UDP) is used for the transport layer. This protocol does not implement any congestion control. Instead, the Transmission Control Protocol

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(TCP) adapts its transmission rate according to network congestion. The coexistence of multicast traffic and TCP traffic may lead to unfair use of network resources. In order to prevent this situation, the deployment of multicast congestion control is indispensable. This kind of congestion control is well known as TCP-friendliness [3].

The adoption of a multicast congestion control in cellular networks poses an additional set of challenges. All the algorithms for congestion control treat the packet loss as a manifestation of network congestion. This assumption does not always apply to networks with radio links, in which packet loss is often induced by reasons other than network congestion like noise or radio link error. In these cases, the network reaction should not be a drastic reduction of the sender's transmission rate [4]. Another limitation is that the mobile terminals' computing power cannot afford complicated statistics and traffic measurements, which in turn means that such operations should not be executed on the mobile equipment.

In this paper, we investigate the applicability of two well-known multicast congestion control schemes over UMTS networks: the TCP-friendly multicast congestion control (TFMCC) and the pragmatic general multicast congestion control (PGMCC). Both schemes belong to the class of single-rate congestion control schemes. Such schemes are simple enough, so as to meet a prime objective for UMTS multicast services, which is scalability to applications with thousands of receivers.

We show that the degradation of the radio channels in the UMTS terrestrial radio-access network (UTRAN) causes malfunctions in the legacy TFMCC and PGMCC schemes. The innovation of our work stems from the fact that the original schemes are partly modified and extended in order to support the particularities of the UTRAN. Our proposals introduce minor modifications in the UMTS architecture. Furthermore, complicated operations like statistics and traffic measurements are avoided to be performed on mobile equipment. Last but not the least, we measure the performance of the modified TFMCC and PGMCC schemes in a comparative way.

The paper is structured as follows. Section 2 briefly describes the related work to the examined scientific domain. Section 3 focuses on the description of the TFMCC and PGMCC mechanisms. Furthermore, the problems of the applicability of the above schemes in UMTS are also described in Section 3. Section 4 is dedicated to presentation of the proposed modifications to the PGMCC and TFMCC mechanisms and Section 5 presents the simulation results. Finally, some concluding remarks and planned next steps are briefly presented in Section 6.

2. RELATED WORK

Multicast congestion control problem in fixed networks is still a domain of active research and a lot of solutions have been proposed until now. A technical problem of major importance in multicast congestion control is scalability. When the source receives a negative feedback of congestion notification from the network, it adapts its transmission rate. In order to avoid a feedback implosion, it is proposed that the receiver of the worst congestion level is selected as the representative [5, 6]. The representative transmits feedback information for congestion control and the number of feedbacks is limited. Another advantage of the use of a single receiver is that the excessive restriction of transmission rate is avoided when the sender receives multiple negative feedbacks which, however, originate from different receivers.

In contrast to the multicast congestion control problem in fixed networks, very few solutions have been proposed for the variation of this problem in cellular networks. The most strongly

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related work is presented in [7]. This paper presents a TCP-friendly mechanism for the multicast congestion control over UMTS. This mechanism is based on the TFMCC algorithm. The proposed scheme introduces improvements in the TFMCC functionality in order to solve the current limiting receiver (CLR) selection problem [6].

In [8] the wireless-caused representative selection fluctuation problem is presented. This situation causes frequent change in the representative of wireless multicast congestion control. The sender adjusts its transmission rate to the tentative worst receiver, which brings severe performance degradation to wireless multicast. The authors propose two possible solutions and through performance evaluation in various situations, it is concluded that the end-to-end approach is more sensitive for its inferring error. On the other hand, the active service approach leads to significant performance improvement. This work applies to the extended class of wireless access networks and is not well aligned with 3GPP specifications for the UMTS cellular networks.

3. BACKGROUND AND PROBLEM FORMULATION

3.1. Multicast congestion control

The legacy multicast congestion control schemes that we study are the TFMCC and the PGMCC. The main advantages of these schemes are the simplicity and the scalability [9]. In the following paragraphs we briefly present the above mechanisms and analyze their operation.

3.1.1. TFMCC. TFMCC is a well-known equation-based multicast congestion control mechanism. In order to compete fairly with TCP, TFMCC receivers individually measure the prevalent network conditions and calculate a rate (that is TCP-friendly) on the path from the sender to themselves. The rate is determined using an equation for TCP throughput, which roughly describes TCP's sending rate as a function of the loss event rate, round-trip time (RTT), and packet size. The sending rate of the multicast transmission is adapted to the receiver experiencing the worst network conditions. TFMCC mechanism works as follows:

- Each receiver measures the loss event rate and its RTT to the sender.
- Each receiver then uses this information, together with an equation for TCP throughput, to derive a TCP-friendly sending rate.
- Through a distributed feedback suppression mechanism, only a subset of the receivers is allowed to give feedback to prevent a feedback implosion at the sender.
- Receivers whose feedback is not suppressed report the calculated transmission rate back to the sender in so-called receiver reports. The receiver reports serve two purposes: inform the sender about the appropriate transmit rate, and allow the receivers to measure their RTT.
- The sender selects the receiver that reports the lowest rate as CLR. Whenever feedback with an even lower rate reaches the sender, the corresponding receiver becomes CLR and the sending rate is reduced to match that receiver's calculated rate. The sending rate increases when the CLR reports a calculated rate higher than the current sending rate.

Any realistic equation giving TCP throughput as a function of loss event rate and RTT would be suitable for use in TFMCC. However, it should be noted that the TCP throughput equation used should always reflect TCP's retransmit timeout behavior, as this dominates TCP throughput at higher loss rates. The throughput equation currently recommended for TFMCC is a slightly simplified version of the throughput equation for Reno TCP from [10]:

$$T_{\mathrm{TCP}} = \frac{8s}{t_{\mathrm{RTT}}(\sqrt{2p/3} + (12\sqrt{3p/8})p(1+32p^2))}$$

In the above equation, the expected throughput T_{TCP} is the transmit rate in bits/second.

The expected throughput T_{TCP} of a TCP flow is calculated as a function of the steady-state loss event rate p, the round-trip time t_{RTT}, and the packet size s. Each TFMCC receiver measures its own loss event rate and estimates its RTT to the sender. It then uses this equation to calculate T_{TCP}, which is an estimate of the throughput a TCP flow would achieve on the network path to the corresponding receiver under the same network conditions. For full details of TFMCC, we refer the reader to [6].

3.1.2. PGMCC. PGMCC is a single-rate multicast congestion control scheme, which is TCP-friendly, achieves fast response to variations in network conditions and is suitable for both non-reliable and reliable data transfers. In general, the receiver reports are a fundamental component of PGMCC. They are sent back to the sender as NAK or ACK fields and based on them the sender regulates the transmission rate. The NAKs consist of the following fields:

- the identity of the receiver (`recv_id`),
- the loss rate measured locally (`recv_loss`),
- the highest data packet sequence number received (`recv_lastseq`).

Additionally, the receiver with the worst throughput is elected as acker being in charge of sending positive ACKs to the sender. The identity of the acker is carried as a field in each data packet. The ACKs contain the same loss report as NAKs and two additional fields:

- the sequence number of the data packet, which triggered the ACK (`ack_seq`),
- a 32-bit bitmap indicating the receive status of the 32 last packets (`bitmask`).

The loss rate is estimated in each receiver. The estimated loss rate (`recv_loss`) is sent back to the multicast sender with NAKs and ACKs. In order to measure its loss rate, each receiver interprets the packet arrival rate as a discrete signal (1 for lost packets, 0 otherwise). The signal is then passed through a discrete-time linear filter, whose response and computational costs are chosen accordingly. The sender regulates the transmission rate based on the worst throughput received.

The main goals of PGMCC are TCP-friendliness and scalability. The latter is achieved by decentralizing functionality as much as possible. The adoption of NAK suppression and the election of a single node to be in charge of sending ACKs are solutions that only require a constant amount of processing cost on nodes, independently of the group size.

Another advantage is the fast response that is achieved by introducing positive ACKs from the acker instead of NAKs proposed by other schemes. The adoption of positive ACKs permits a more timely distribution of information. For full details of PGMCC, we refer the reader to [5].

3.2. Overview of UMTS multicast

3.2.1. UMTS architecture. From the physical point of view, UMTS network architecture is organized in two domains: the user equipment (UE) and the public land mobile network (PLMN). UE

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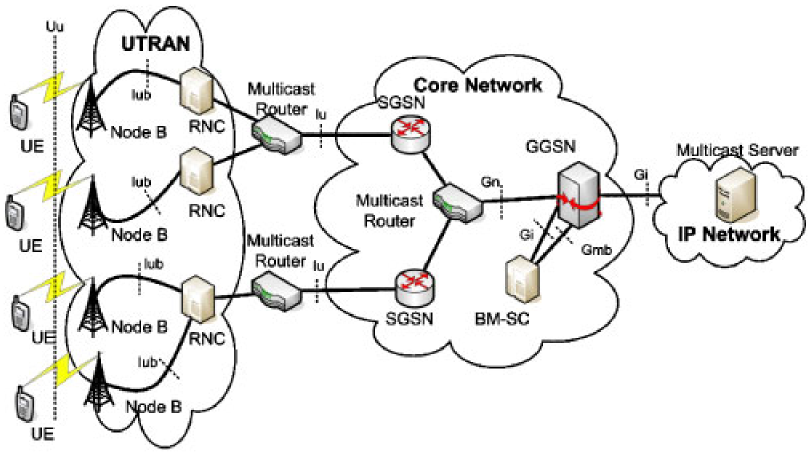

Figure 1. UMTS architecture.

constitutes the mobile user terminal. The PLMN is further divided into two land-based infrastructures: the UTRAN and the core network (CN) (Figure 1). The UTRAN handles all radio-related functionalities. The CN is responsible for maintaining subscriber data and for switching voice and data connections.

The UTRAN consists of two kinds of nodes: the radio network controller (RNC) and the Node B. The Node B constitutes the base station and provides radio coverage to one or more cells. It is connected to the UE via the Uu interface and to the RNC via the Iub interface. The Uu is a radio interface based on the wideband code division multiple access technology.

The CN consists of two kinds of general packet radio service support nodes (GSNs), namely gateway GSN (GGSN) and serving GSN (SGSN) (Figure 1). The SGSN is the centerpiece of the packet switched domain. It provides routing functionality, it manages a group of RNCs and it interacts with the home location register, which is a database permanently storing subscribers' data. The SGSN is connected to GGSN via the Gn interface and to RNCs via the Iu interface. GGSN provides the interconnection between the UMTS network and the external packet data networks like the Internet [11, 12].

3.2.2. MBMS service. The MBMS is defined by 3GPP as an IP datacast type of service, which can be offered via existing UMTS cellular networks. As the term MBMS implies, there are two types of MBMS service mode: the broadcast and the multicast. In broadcast mode, data are delivered to a specified area without knowledge of the receivers in the area. Since the multicast mode is more general than the broadcast one, we shall focus on the operation of the MBMS multicast mode.

The basic MBMS architecture is almost the same as the existing UMTS architecture in the PS-domain. The most significant modification is the addition of a new node called broadcast multicast–service center (BM–SC). This node performs the multicast group management and provides interfaces for the interaction with the content providers. The BM–SC also authorizes and charges the content provider. In order to reduce the implementation costs, the MBMS has been designed so as to introduce minor changes to the existing radio and CN architectures. For simplicity

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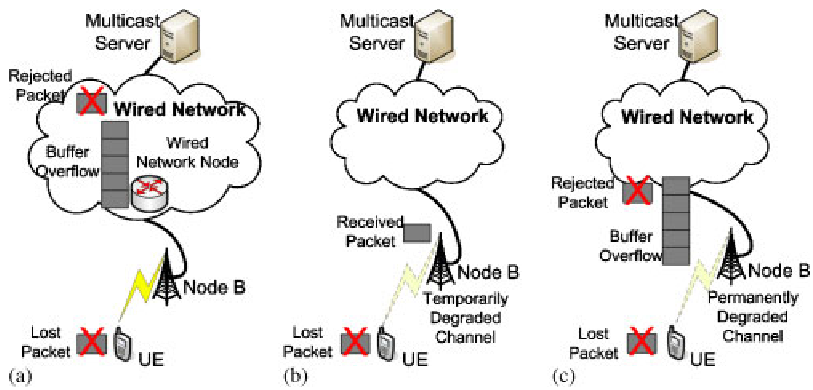

Figure 2. Packet loss induced by: (a) network congestion; (b) temporary radio channel degradation; and (c) permanent radio channel degradation.

reasons, in our analysis, we shall consider that the functionality of the BM–SC is incorporated in the GGSN (Figure 1).

3.3. Radio channel degradation

The multicast congestion control schemes for fixed data networks treat packet losses as a manifestation of network congestion. These legacy schemes translate the packet loss as packet rejection that is caused by buffer overflow in a network node and the network's reaction to the packet losses is a drastic reduction of the sender's transmission rate. The goal is the optimization of the network resources utilization in order to avoid buffer overflows and packet retransmissions.

The above described scenario may also occur in a UMTS networks. As shown in Figure 2(a), a packet rejection caused by overflow in a wired network buffer occurs. The UE translates this packet rejection as a packet loss, and the action that should be taken by the multicast server in order to resolve this situation is the reduction of its transmission rate.

However, the applicability of the legacy congestion control schemes for the multicast congestion control over cellular communication systems, like UMTS, faces a major problem. In cellular networks there are some cases where the packet loss may not mean network congestion. In other words, the quality of a radio channel may be degraded due to reasons other than network congestion. These reasons may be noise or signal fading. During a degradation period, the bit error rate of the radio link may become temporarily very high but, normally, after that period the radio link is expected to recover. In this situation the packet loss does not depend on the arrival rate of the packets, but on the duration of the radio channel degradation. After the end of the fading period, the packet losses will be resolved without the need for reduction of the transmission rate. If the packet loss is caused by radio channel degradation, the reduction of the transmission rate will not affect the packet loss. Figure 2(b) depicts the case when a packet loss occurs due to temporary radio channel degradation.

The radio channel degradation may affect the performance of the legacy schemes. If a UE suffers from fading, then the packet loss rate for this UE will temporarily increase. This increment of the packet loss rate may cause the selection of this UE as the representative. The next step is the reduction of the sender's transmission rate according to the acceptable sending rate calculated at the examined UE. The problem is that this reduction is unnecessary because it is radio and not network caused. When the UE recovers from the degraded radio quality phase, the target throughput of the representative will increase. If a lot of UEs participate in the multicast group, there is a high probability that, at a given time, a UE suffers from fading. Soon, another UE suffering from channel degradation will be selected as representative and will cause reduction of the transmission rate. Eventually, the radio channel degradation will cause a significant and steady degradation of the service performance. As a consequence, the network's reaction to the packet losses due to radio channel degradation should not be a drastic reduction of the sender's transmission rate.

Another possible scenario is the case that, under certain conditions, a permanent degradation of the radio channel affects a specific UE. When a permanent degradation occurs on the radio link, the buffer of the Node B will possibly overflow and some packets will be rejected. Subsequently, Node B will request the retransmission of the rejected packets from its corresponding RNC. Obviously, the steady incapability of the UE to receive multicast packets affects the performance of the whole multicast service and the UE suffering from permanent radio channel degradation should be candidate as the current representative. Figure 2(c) visualizes this case.

4. ADAPTATION TO UMTS

In order to adapt the TFMCC and PGMCC mechanisms to the previously described situations, we introduced some necessary improvements to these schemes. The proposed mechanisms follow a design very similar to that of the legacy congestion control schemes. New functionality has been added to some nodes of the UTRAN to deal successfully with the temporary and the permanent radio channel degradation situations.

4.1. New packet field

In the proposed mechanisms, the nodes located at the border between wireless and wired network (i.e. the Node Bs) have an additional functionality. This is to provide the receivers (i.e. the UEs) with information about their measured packet loss rate. Each UE is informed by its serving Node B for the packet loss that the Node B measures. The information is piggybacked in the data packets of a multicast session. In order to achieve a minimal cost, a new packet field has been added in the data packet of the proposed schemes. This packet field may be an 1-bit information and, thus, minor modifications are needed to be introduced in the headers of the data packets. From now on, we shall refer to this packet field as packet loss indication.

On the other hand, the formats of the receiver reports containing the acceptable sending rate are preserved in their legacy forms. Nevertheless, the way that the acceptable sending rate is calculated in the receivers has changed. In fact, the way that the acceptable sending rate field is calculated, depends on the type of the radio channel degradation. The radio channel degradation may be temporary or permanent. Although the effects of temporary radio channel degradation should be filtered out by the scheme, the permanent radio channel degradation impacts should not be ignored.

4.2. Temporary radio channel degradation

In this subsection we present the modifications introduced in the legacy TFMCC and PGMCC schemes in order to deal successfully with the temporary radio channel degradation. By the term temporary radio channel degradation we mean any situation that causes increment of the packet losses over the radio channel for a short time period. The main goal of the modified schemes is to filter out the packet losses caused by this situation. As it has already been explained, the reason is that this kind of packet loss does not mean network congestion and, consequently, no reduction of the sender's transmission rate is needed.

In the legacy schemes, the loss rate is estimated in each receiver. This loss rate is further used for the calculation of an acceptable transmission rate. The calculated acceptable loss rate is sent back to the multicast sender with the receiver reports and then the sender selects the current representative and the actual sending rate. The intention of the modified schemes is to ignore the packet losses caused by temporary radio channel degradation. Consequently, the packet losses caused by temporary radio channel degradation are excluded from the input of the acceptable transmission rate calculation.

The new packet field introduced in the data packets permits each UE to identify the reason of a packet loss. The UE compares the packet loss received from Node B with its measured packet loss. In general, the following cases are distinguished:

- When both the Node B and the UE encounter a packet loss, this packet loss is considered to be caused due to network congestion (Figure 2(a)). Consequently, this kind of packet loss is taken into consideration during the calculation of the acceptable sending rate in the UEs.
- On the other hand, when the two values differ, the UE concludes that the reason for the difference is losses at the radio link caused by temporary radio channel degradation (Figure 2(b)). This kind of packet loss is not related to the network congestion and, consequently, the reduction of the transmission rate will not affect this packet loss. In this case, the packet loss is not accounted during the calculation of the acceptable sending rate.

The packet loss indication introduced in the data packet is set by the Node B and, as we have already mentioned, may be an 1-bit information. This means that 0 indicates a correctly received packet in Node B, whereas 1 indicates a previously lost packet. Each UE reads this information when receives a data packet. The packet loss flag sequence from the data packets is recorded in each UE and is used for the estimation of the packet loss and the calculation of the acceptable sending rate.

4.3. Permanent radio channel degradation

At this point we shall present the adaptation of the legacy schemes to the situation where permanent radio channel degradation occurs. If a permanent degradation of the radio channel affects a specific UE, then after a short time period, the buffer of the Node B will overflow and some packets will be rejected (Figure 2(c)). Obviously, this steady incapability of the UE to receive multicast packets affects the performance of the whole multicast service and the UE suffering from permanent radio channel degradation should be selected as the current representative.

In the proposed mechanisms, in case that a packet rejection has occurred in Node B due to buffer overflow, then when writing the packet loss indication, Node B considers this packet as unreceived (lost) packet. Thus, Node B sets the packet loss indication to 1 in the header of the data packets that have previously been rejected due to overflow of its buffer. Consequently, the receiving UEs consider these packet losses as general packet losses that are happened due to network congestion. Thus, these packets are accounted by the UE during the estimation of the loss rate and, subsequently, during the calculation of the acceptable sending rate.

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To sum up, in our proposed schemes the Node B sets the packet loss indication to 1 in case that:

- The packet was previously lost.
- The packet was rejected from the Node B's buffer due to permanent radio channel degradation.

It is obvious that the above described improvement makes our proposed mechanisms suitable for permanent degradation of the radio channel. It is assured that this situation is not hidden from the UEs. In fact, the UEs gets always informed of the packet rejections caused by buffer overflow in their serving Node B.

5. EXPERIMENTAL RESULTS

The modified TFMCC and PGMCC schemes are evaluated through simulations toward two directions. The first direction is to examine whether the proposed mechanisms preserve the benefits of the corresponding legacy schemes in comparison with their initial presentations in [5, 6]. The second direction is to evaluate the behavior of the mechanisms under conditions when temporary or permanent radio channel degradation occurs. In all cases, the results are presented in a comparative way.

In order to simulate the multicast transmission over UMTS, we use the enhanced UMTS radio-access network extensions extension of ns-2 [13] along with the implementation of the MBMS multicast packet forwarding mechanism described in [14]. Furthermore, the code used to implement and evaluate the TFMCC and the PGMCC in [5, 6], respectively, is incorporated in the simulator. Finally, the legacy TFMCC and PGMCC schemes are modified and extended in order to support the UMTS environment characteristics. The simulation parameters are presented in Table I.

In the following paragraphs we present in detail the performance evaluation of the proposed modified schemes and their comparison them with the corresponding legacy mechanisms.

5.1. *Intra/inter-protocol fairness*

The first aspect that we examine is the intra-protocol fairness of the proposed schemes. More specifically, we examine the fairness of the proposed schemes toward other competing flows of the same protocol when they share common wired or wireless links. To this direction, we consider a UMTS network with a single-bottleneck topology. The bottleneck is applied over a link connecting an SGSN with an RNC node (Iu interface) (Figure 1). We monitor the throughput over a wireless link connecting the UEx with Node By. UEx belongs to two multicast groups and is receiving two

Table I. Simulation parameters.

Number of SGSNs	3
Number of RNCs	20
Number of Node Bs/cells	80
Number of multicast servers/groups for TFMCC traffic	5
Number of multicast servers/groups for PGMCC traffic	5
Number of servers for TCP traffic	50
Number of UEs	500
Number of UEs in each multicast group	Varies from 10 to 100
Number of multicast groups that a single UE joins	Varies from 0 to 2

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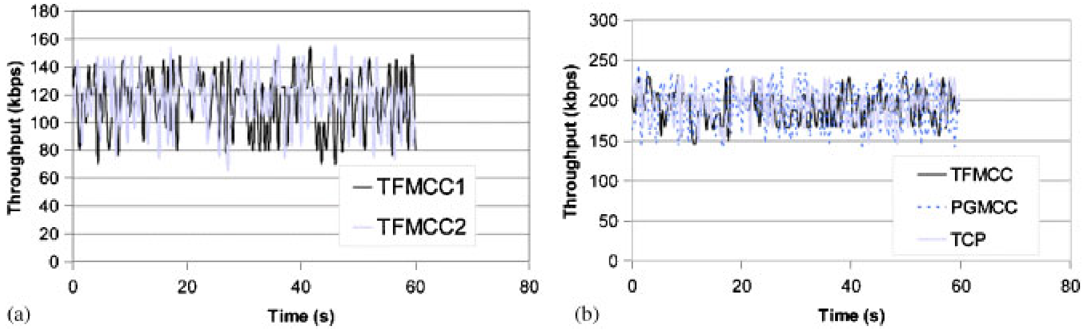

Figure 3. Throughputs in a single-bottleneck UMTS network: (a) two competing TFMCC instances and (b) TFMCC versus PGMCC versus TCP flow.

instances of multicast traffic from two different external multicast servers. Figure 3(a) illustrates the throughput of each competing multicast flow when the modified TFMCC is used as congestion control mechanism. The average throughput of TFMCC1 flow is about 98.6% of the average TFMCC2 throughput. In the case of the modified PGMCC congestion control mechanism, the average flow almost matched each other. Similar results are obtained for many other scenarios. For example, when no congestion exists or when congestion exists over an Gn interface (connects GGSN with SGSN nodes). The intra-protocol fairness for both the proposed schemes is therefore confirmed under all the congestion scenarios.

Furthermore, we examine all the aspects of inter-protocol fairness of the proposed schemes. The inter-protocol fairness of the proposed schemes is the fairness toward the competing TCP flows along with the fairness which the proposed schemes show to each other when they use common links. In the examined simulation topology, UEx is receiving multicast traffic from two multicast servers. The first multicast server uses the modified TFMCC as a congestion control mechanism, whereas the other one uses the modified PGMCC. At the same time, UEx is receiving TCP traffic from an external node. Figure 3(b) shows the throughput of the TFMCC flow against the PGMCC and TCP flows. The average throughput of all flows closely matches each other. The average throughput of TFMCC is 89% and that of PGMCC is 94% of the average throughput of TCP. Moreover, TFMCC achieves a smoother rate than TCP and PGMCC.

Similar results are obtained for many other scenarios. In all network topologies and congestion levels, the available throughput of the bottleneck link is evenly shared among the competing TCP and modified TFMCC and/or PGMCC schemes and TFMCC is achieving the smoother rate. Consequently, the TCP-friendliness and inter-protocol fairness of each one of the proposed schemes against the other are confirmed.

5.2. Responsiveness to changes

An important concern in the design of congestion control schemes is their responsiveness to changes in the network conditions. This behavior is investigated using the single-bottleneck topology. During the simulation, we change the applied loss rate of the bottleneck link. The simulation lasts 150 s. During this time interval three different loss rates are applied over the Iu interface. A TFMCC flow along with a PGMCC and a TCP flow are monitored, while sharing the bottleneck link. The results of the simulation for the three competing flows are presented in Figure 4.

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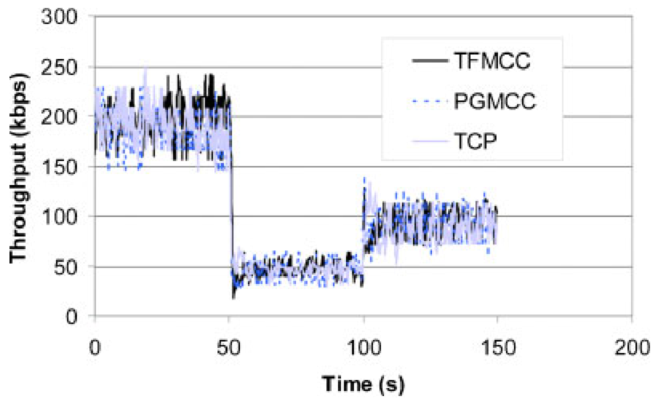

Figure 4. Responsiveness to changes in the loss rate.

As shown in Figure 4, the proposed schemes' throughputs closely match the TCP throughput at all three loss levels. Moreover, the adaptation of the sending rate is fast enough for both the proposed schemes. The results show that the PGMCC scheme has immediate reaction to the changes in the loss rate. Actually, the simulator logs show that the UEs need 1200–2000 ms after the change of the loss rate in order to adapt to the new loss rate. These figures of response time are close enough to the corresponding time of TCP (about 1000–1500 ms). On the other hand, TFMCC has slower reaction since its adaptation time is about 1500–2500 ms. However, as in the experiment described in the previous subsection, the TFMCC preserves the smoothest rate of all schemes.

A similar simulation setting is used in order to investigate the responsiveness to changes in the RTT. The results are similar to those above and, consequently, the excellent reactivity of the modified TFMCC and (especially) of the modified PGMCC to changes in congestion level of the UMTS network is confirmed. The above results also prove that during the application of the proposed schemes over the UMTS, one of the main advantages of the legacy schemes, which is the fast responsiveness to changes, is not affected.

5.3. Temporary radio channel degradation

The next aspect of our experiments is the evaluation of the proposed schemes when packet losses occur due to temporary radio channel degradation. We consider a UMTS network with a loss model presented in [15] and we simulate a radio channel degradation period by applying an error rate over the packets transmitted via the radio links. The packet loss rate is selected to be relatively small and is assumed to be stable in a period of 20 s. During each 5 s period, the applied packet loss rate is selected randomly from a range from 0% up to 5%. The examined schemes are the TCP protocol along with both the proposed and the legacy congestion control schemes. The experiment is executed five times, once for each of examined schemes, with exactly the same parameters. During each instance of the experiment we measure the throughput of each one of examined schemes.

Figure 5 depicts the throughput ratio of the modified schemes (they are referred as 'MOD_TFMCC' and 'MOD_PGMCC') against the corresponding legacy schemes (they are

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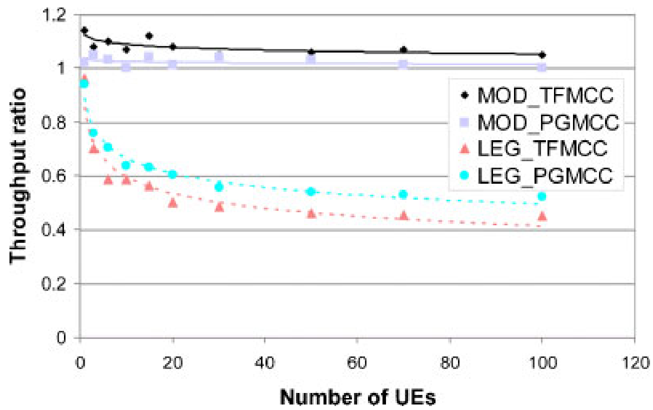

Figure 5. Throughput of proposed versus legacy schemes during temporary radio channel degradation.

Table II. Quantitative improvement between the proposed and the legacy schemes during temporary radio channel degradation.

	Number of UEs									
Scheme	1	3	6	10	15	20	30	50	70	100
MOD_TFMCC	1.14	1.08	1.1	1.07	1.12	1.08	1.05	1.06	1.07	1.05
MOD_PGMCC	1.02	1.05	1.03	1	1.04	1.01	1.04	1.03	1.01	1
LEG_TFMCC	0.96	0.7	0.585	0.587	0.562	0.502	0.484	0.462	0.456	0.452
LEG_PGMCC	0.94	0.76	0.705	0.637	0.63	0.602	0.557	0.538	0.528	0.522

referred as 'LEG_TFMCC' and 'LEG_PGMCC'). The throughput of each scheme is normalized by the TCP throughput (thus the TCP throughput ratio is omitted since it always coincides with 1). The horizontal axis shows the number of the UEs belonging in the examined multicast group. The vertical axis shows the average throughput that is normalized by the corresponding TCP one.

The packet losses can be identified correctly at the UEs and are ignored during the calculation of the acceptable sending rate. The modified PGMCC scheme throughput closely approaches the mean throughput of the TCP protocol, whereas the TFMCC scheme has a slightly bigger (about 8%) throughput than the TCP. On the other hand, significant multicast service degradation takes place when the legacy schemes are used for the congestion control. The service degradation is getting bigger when the number of the UEs participating in the multicast group increases. Table II and Figure 5 confirm the efficient operation of the proposed schemes when radio channel degradation occurs.

5.4. *Permanent radio channel degradation*

The last concern of our experiments is the evaluation of the proposed schemes when packet losses occur due to permanent radio channel degradation. In the examined UMTS network, the radio link connecting the UEx with the Node By is permanently degraded.

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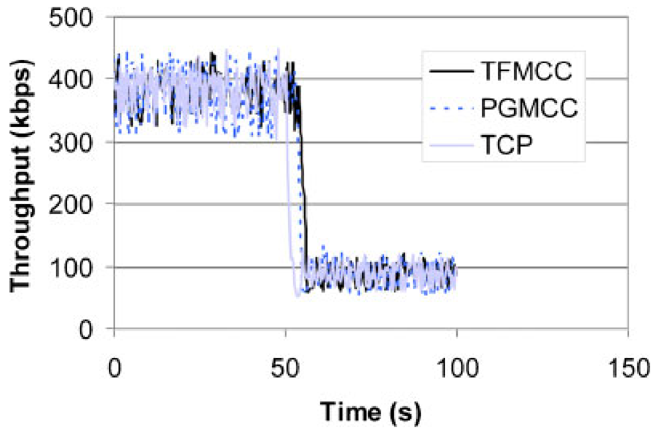

Figure 6. Throughput of proposed schemes versus TCP during permanent radio channel degradation.

We simulate this wireless channel degradation period by applying a loss rate of 20% over the packets transmitted via the corrupted wireless link. In the beginning of the simulation, no radio channel degradation occurs. After 50 s of simulation, we apply the error rate over the wireless channel connecting the UEx with Node By. Then, we monitor the changes in the throughput of the corrupted wireless link for 100 s. The experiment is executed three times, one for each modified scheme and one for the TCP traffic, with exactly the same parameters. The results of this experiment are presented in Figure 6.

The simulation results prove that the proposed schemes react efficiently to the permanent wireless channel degradation. In the beginning of the simulation, where no congestion exists, the throughput matches the available bandwidth of the wireless link. After 50 s of simulation, the 20% packet loss rate is applied. The modified schemes, unlike TCP, do not immediately react to this degradation because they consider it as temporary degradation. Soon after, the buffer of the Node B overflows and the modified congestion control schemes are able to distinguish the nature of the degradation. It takes about 5 s for PGMCC and 6 for TFMCC to fully adapt to the new network conditions. This time interval may differ according to the bit-rate of the transmission and the size of the buffer in Node B. The corresponding time interval for TCP is less than 2 s.

The results confirm the behavior of the proposed schemes when permanent radio channel degradation occurs. The packet losses in the Node B are considered as network congestion and are not ignored during the calculation of the acceptable sending rate in the UE. This kind of packet losses causes reduction of the transmission rate. In case multiple radio channels are degraded, the simulation results (not listed here) are similar. The representative is a UE among the ones being connected with permanently degraded radio links.

6. CONCLUSIONS AND FUTURE WORK

In this paper we presented two congestion control mechanisms for multicast transmissions over UMTS. The proposed schemes are based on the well-known TFMCC and PGMCC mechanisms.

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Both the legacy schemes belong to the category of TCP-friendly, single-rate multicast congestion control mechanisms and they have designed mainly for fixed networks. In order to be used in cellular networks, a number of modifications are required to both mechanisms.

In particular, a new 1-bit packet field is added in the data packet of both schemes in order to confront the permanent and temporary radio channel degradation phenomena. The impacts are minor and restricted only in two nodes of the UMTS network, namely the Node Bs and the UEs. The additional functionality of the UE has as main target to examine the cause of a potential packet loss. The proposed schemes respect the limited computing power of the UEs and no demanding operations are introduced in those network nodes.

In order to evaluate the proposed schemes, we performed simulation experiments. The main conclusion is that both schemes preserve the benefits of the legacy mechanisms over the UMTS cellular network. This means that the schemes are fair toward other competing instances of same scheme over various network topologies and congestion levels. They also show TCP-friendliness toward competing TCP flows. The modified TFMCC scheme preserves a very smooth rate in comparison with the TCP and the modified PGMCC. The results of the experiments also demonstrate the efficient response of the mechanisms to changes in the congestion level for both loss rate and RTT. Furthermore, it has to be mentioned that an additional benefit of the legacy schemes that is also preserved is scalability.

The other aspect of our experiments was the performance monitoring of the schemes when radio channel degradation occurs. We confirmed that when temporary radio channel degradation occurs at the radio link, the packet losses can be identified correctly at the UEs and are ignored at the calculation of the acceptable sending rate. On the other hand, the legacy schemes react to the above situation with reduction of the sending rate and the multicast service is degraded permanently.

Finally, we examined the behavior of our proposed schemes against permanent radio channel degradation. In this case, the packet losses due to buffer overflow in the Node B are considered as network congestion and are not ignored during the calculation of the acceptable sending rate in the UE. Eventually, this kind of packet losses causes reduction of the transmission rate.

The step that follows this work may be the evaluation of multicast congestion control schemes belonging in other classes (like multi-rate or layered schemes). It may be examined whether these schemes can be applied over UMTS multicast and possible modifications may be proposed. These schemes may also be evaluated through comparison with the schemes presented in this paper.

Furthermore, another step may be the formulation of a multicast group control mechanism dedicated to UMTS networks. In some cases, a permanently degraded channel connecting a specific UE may cause a large reduction of the transmission rate and eventually generic multicast service degradation. It may be specified under which circumstances radio channel degradation would cause expulsion of a UE from a multicast group.

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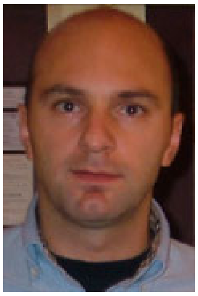

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