Congestion Control for Multicast Transmission over UMTS

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ABSTRACT

In this chapter, we present a novel mechanism for the multicast congestion control over Universal Mobile Telecommunications System (UMTS) networks. The proposed mechanism is based on the well known TCP-Friendly Multicast Congestion Control (TFMCC) scheme. TFMCC is an equation-based multicast congestion control mechanism intended to scale to groups of several thousand receivers. It is based on the concept of Current Limiting Receiver (CLR). The CLR is the receiver that the sender believes currently has the lowest expected throughput of the multicast group. The key challenge in the design of the new scheme lies in extending TFMCC in order to support the UMTS Terrestrial Radio-Access Network (UTRAN). Our major contribution is that the TFMCC mechanism was improved in order to cope with the packet losses caused by either the temporary or the permanent degradation of the wireless channel. One of the major advantages of the proposed scheme is that it introduces minor modifications in the UMTS nodes with respect to the computing power of the mobile terminals. Finally, we implement our approach in the ns-2 network simulator and we evaluate it under various conditions. The simulation results are presented along with their analysis.

1. INTRODUCTION

Third Generation (3G) mobile cellular networks promise the provision of advanced services along with high data rates. In the meantime, the requirement for real time multimedia data transmission which addresses to user groups is increasing. Services like videoconference or distance learning are demanding features which load the network nodes and consume a large portion of the throughput provided by the network [1]. Universal Mobile Telecommunications System (UMTS) constitutes the most prevalent standard of the 3G cellular networks. Despite the high capacity that UMTS networks provide, the expected demand will certainly overcome the available resources. This is the reason why multicast transmission is one of the major goals for UMTS and 3G networks in general.

Multicast is an efficient method for data transmission to multiple destinations. Its advantage is that sender's data are transmitted only once over links which are shared along the paths to a targeted set of destinations. Data duplication is restricted only in nodes where the paths diverge to different subnetworks [2]. Multicast routing has been adopted by the Internet for more than ten years. Internet Protocol (IP) multicast is a bandwidth-conserving technology that reduces traffic by simultaneously delivering a single stream of data over a multicast tree. On the other hand, the wireless communication medium has itself a broadcast nature which is suitable for the adoption of multicast routing over cellular networks. The 3G Partnership Project (3GPP) is a global body dedicated to developing 3G specifications. In the beginning of the current decade, 3GPP recognized the need for the support of multicast routing in UMTS networks. As a result, the standardization of Multimedia Broadcast/Multicast Service (MBMS) framework started in 2002 [3].

Congestion control is a policy that regulates the source transmission rate according to the network congestion. In IP multicast, the User Datagram Protocol (UDP) is used for the transport layer. This protocol does not implement any congestion control. Instead, the Transmission Control Protocol (TCP) regulates its transmission rate according to network congestion. This means that 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 [4].

The adoption of a multicast congestion control in cellular networks poses an additional set of challenges which are related to the existence of wireless links and mobile terminals. In the first place, all the algorithms for congestion control treat the packet loss as a manifestation of network congestion. This assumption may not apply to networks with wireless links, in which packet loss is often induced by noise, wireless link error or reasons other than network congestion. As a consequence, the network reaction should not be a drastic reduction of the sender's transmission rate [5]. Secondly, due to the fact that the physical radio resources (frequencies and code sequences) are limited, Radio Resource Management (RRM) is a key process. It administers with high flexibility and efficiency the scarce radio resources while at the same time keeps service constraints. RRM performs itself congestion control over the radio links and its strategy should be considered during the design of the congestion control mechanism [6]. Last but not least, the mobile terminals' computing power cannot afford complicated statistics and traffic measurements. Consequently, such operations should be avoided to be held on mobile equipment.

In this chapter, we present a novel mechanism for the multicast congestion control over UMTS networks. The proposed mechanism is based on the well known TCP-Friendly Multicast

Congestion Control (TFMCC) scheme. TFMCC is an equation-based multicast congestion control mechanism that extends the TCP-Friendly Rate Control (TFRC) [7] protocol from the unicast to the multicast domain in the Internet. It belongs to the class of single-rate congestion control schemes. Such schemes inevitably do not offer multiple transmission rates as layered schemes do. However, they are much simpler so as to meet a prime objective for UMTS multicast services, that is scalability to applications with thousands of receivers [4].

In our proposed mechanism, the TFMCC scheme is partly modified and extended in order to support the particularities of the UMTS Terrestrial Radio-Access Network (UTRAN). The major problem of the applicability of TFMCC over UMTS is the Current Limiting Receiver (CLR) problem. The CLR problem is caused when the wireless channel quality is temporarily degraded. Minor modifications in the UMTS architecture are required by our proposed scheme. New functionalities are introduced in two nodes of the UMTS network in order to deal with the CLR problem. These impacts concern the User Equipment (UE) and the Node B. The additional functionalities allow each UE to identify the reason of a packet loss. The UE can conclude whether a packet loss has been caused by wireless channel degradation or by network congestion. Additionally, another aspect of the proposed mechanism is the handling of the permanent degradation of the wireless link.

This chapter is structured as follows: subchapter 2 provides an overview of the work related to the scientific domain. In subchapter 3, we briefly present concepts like the TFMCC algorithm, the UMTS networks and the MBMS service. Moreover, we describe the problem of the applicability of congestion control over the wireless access networks. Subchapter 4 is dedicated to the proposed congestion control mechanism. Subchapter 5 describes the simulation experiments. Finally, some concluding remarks and planned next steps are stated in subchapters 6 and 7, respectively.

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. We use two distinct properties to classify the existing approaches [8]:

- The rates delivered to the receivers in a session. Existing approaches generally fall into three categories: single-rate [4], multi-rate [9] and layered [10].
- The place where adaptation is performed. It is either at the end systems (end-to-end service) [4], [10] and [11], or at the intermediate network nodes (active service) [12].

A technical problem of major importance in multicast congestion control is scalability. When the source receives a negative feedback of congestion notification inside the network, it regulates its transmission rate. In order to avoid a feedback implosion, the majority of the researchers, like the authors of [13] and [14], suggest that the receiver of the worst congestion level should be selected as the representative. In this approach, only 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, no specific solutions and algorithms have been proposed for the variation of this problem in cellular networks.

Despite radio network congestion being a widely recognized and identified problem, few relevant studies have been published. The most strongly related publication is [15]. However, this publication refers to the extended class of wireless access networks (including WLANs) and it is not well aligned with 3GPP specifications for the UMTS cellular networks.

In [15], the authors investigate the wireless-caused representative selection fluctuation problem in wireless multicast congestion control. This problem is caused by the frequent degradation of the wireless channel and the subsequent bursty packet loss. This situation causes frequent change of the representative. The sender adjusts its transmission rate to the tentative worst receiver, which brings severe performance degradation to wireless multicast. In this paper, two possible solutions are proposed, an end-to-end approach and an active approach. Finally, through performance evaluation in various situations, it is concluded that the end-to-end approach is sensitive for its inferring error. On the other hand, the active service leads to significant performance improvement.

3. OVERVIEW OF THE DOMAIN

In this subchapter we describe in brief some basic concepts of the examined scientific domain. The TFMCC algorithm and the UMTS system along with its MBMS service are presented. Finally, the CLR selection problem is analyzed.

3.1 TFMCC Mechanism

TFMCC is a well known equation-based multicast congestion control mechanism that extends the TFRC protocol [7] from the unicast to the multicast domain. It constitutes a congestion control scheme which not only aims to reduce packet loss and improve bandwidth utilization but also is fair towards competing TCP flows, i.e. is TCP-friendly. TFMCC belongs to the class of single rate congestion control schemes and applies at the end systems (end-to-end service). Such schemes inevitably do not offer multiple transmission rates as layered schemes do. However, they are much simpler so as to offer scalability to applications with thousands of receivers [4].

TFMCC uses a control equation derived from a model of TCP's long term throughput to directly control the sender's transmission rate. The loss event rate and the Round-Trip Time (RTT) are the parameters that define this target throughput. Each receiver calculates its target throughput and considers it as the acceptable sending rate from the sender to itself.

TFMCC uses a feedback scheme which allows the receiver calculating the slowest transmission rate to always reach the sender. This scheme is based on the concept of the Current Limiting Receiver (CLR). The CLR is the receiver that the sender believes currently has the lowest expected throughput of the multicast group. Moreover, the TFMCC design ensures that the sender gets feedback from the receivers experiencing the worst network conditions without being overwhelmed by feedback (feedback implosion is suppressed).

For full details of TFMCC, we refer the reader to [4].

3.2 UMTS Architecture

From the physical point of view, the UMTS network architecture is organized in two domains. This basic split considers the User Equipment UE and the Public Land Mobile Network (PLMN). The UE is used by the subscribers to access the UMTS services while the PLMN is a network established by an operator to provide mobile telecommunications services to the public. The PLMN is further divided into two land-based infrastructures: the UMTS Terrestrial Radio-Access Network (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 first is the Radio Network Controller (RNC) and the second is the Node B. The Node B constitutes the base station and provides radio coverage to one or more cells (Figure 1). The Node B 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 (WCDMA) technology. A single RNC with all the Node Bs connected to it, is called Radio Network Subsystem (RNS).

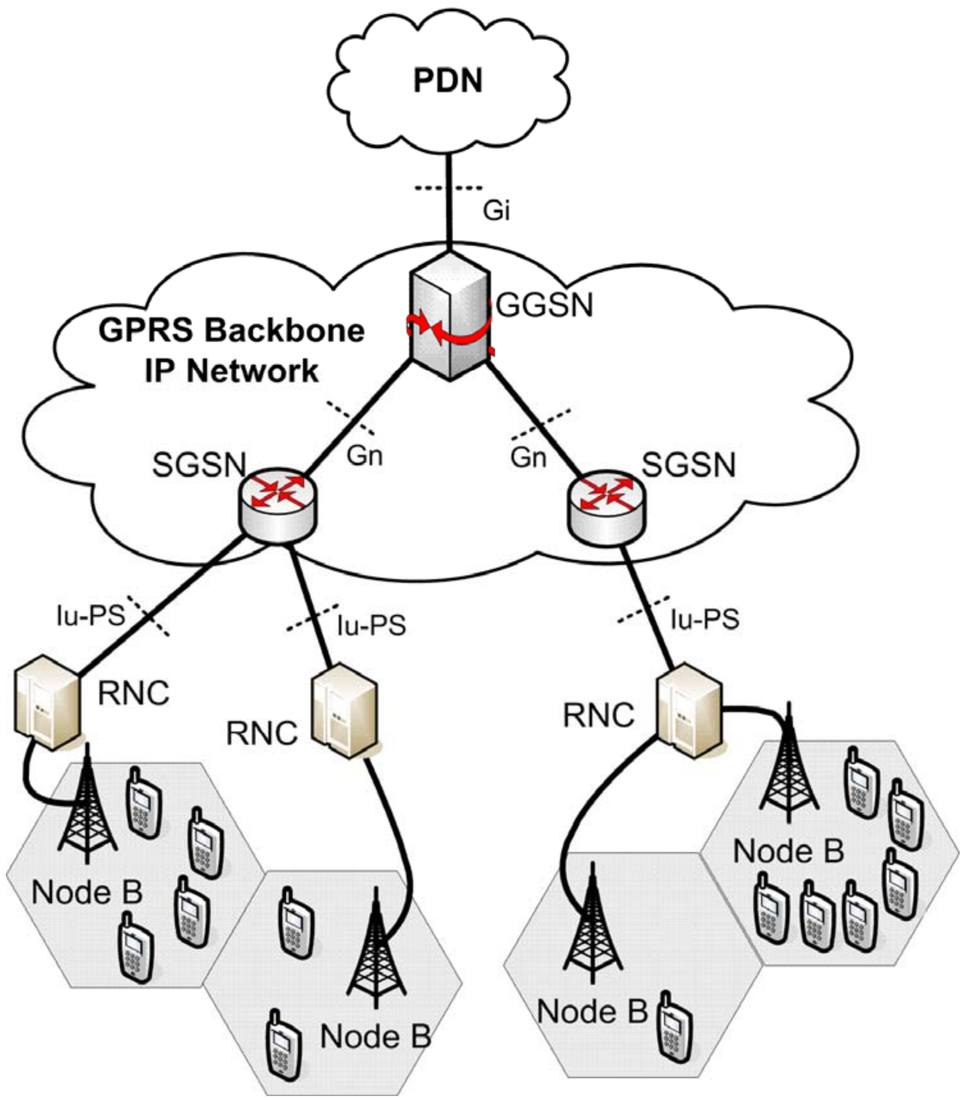

Figure 1. UMTS architecture.

The CN is logically divided into the Circuit-Switched (CS) domain and the Packet-Switched (PS) domain. All of the voice related traffic is handled by the CS-domain while the PS domain handles the packet transfer. The entities of the CS portion of the CN will not be described because the purpose of this paper is to focus on multicast. The PS-domain is more relevant and therefore, in the remainder of this paper, more attention will be devoted to the PS-functionality. The PS-domain of the CN consists of two kinds of General Packet Radio Service (GPRS) Support Nodes (GSNs), namely Gateway GSN (GGSN) and Serving GSN (SGSN) (Figure 1). The SGSN is the centerpiece of the PS-domain. It provides routing functionality, it manages a group of RNSs and it interacts with the Home Location Register (HLR) 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 (PDNs) like the Internet [16].

Before a UE can exchange data with an external PDN, it must first establish a virtual connection with that PDN. Once the UE is known to the network, the packet transfer is based on the Packet Data Protocol (PDP). An instance of a PDP type is called PDP context. When a PDP context needs to be established, a PDP context activation procedure takes place. If this procedure is successful, it leads to the creation of two GPRS Tunneling Protocol (GTP) sessions dedicated to the subscriber: one between the GGSN and the SGSN over the Gn interface and another between the SGSN and the RNC over the Iu interface. The IP packets destined for an application using a particular PDP context are routed using the assigned GTP tunnels to the appropriate RNC. The RNC recovers the GTP-tunneled packet and transmits it to the UE [1].

Data transmission in the UTRAN is based on transport channels. The transport channels define the characteristics of the data transfer according to the service requirements. Despite the fact that there are several types of transport channels specified for UMTS, we will focus on the two most important types: the Dedicated Channel (DCH) and the Forward Access Channel (FACH). The DCH carries information exchanged between a specific UE and the upper network levels. It exists both in the downlink and the uplink direction. Instead, the FACH exists only in the downlink direction. FACH is a common channel and, consequently, a single FACH can carry information for more than one UE in a cell. The existence of multiple types of transport channels in combination with the capability of switching between the different types, allows higher flexibility and a more efficient use of the scarce radio resources while at the same time keeping service constraints [6].

3.3 MBMS Service

As we mentioned above, the 3GPP is currently standardizing the MBMS service [17] and [18]. Actually, the MBMS is defined as an IP Datacast (IPDC) type of service, which can be offered via existing GSM and UMTS cellular networks. As the term Multimedia Broadcast/Multicast Service implies, two types of service mode exist in MBMS service: the broadcast and the multicast. In the broadcast mode, data are delivered to all the receivers roaming in a specific area. On the other hand, in the multicast mode the receivers have to declare their interest for the data reception. The service then decides whether the user may receive data or not. During the rest of our analysis we will focus on the multicast mode. The multicast mode is the most complicated and also covers all the aspects of the broadcast mode.

The basic MBMS architecture is almost the same as the existing UMTS architecture in the PS-domain. Figure 2 illustrates the basic MBMS architecture for UMTS. The most significant modification of the UMTS architecture is the addition of a new node called Broadcast Multicast–Service Center (BM-SC).

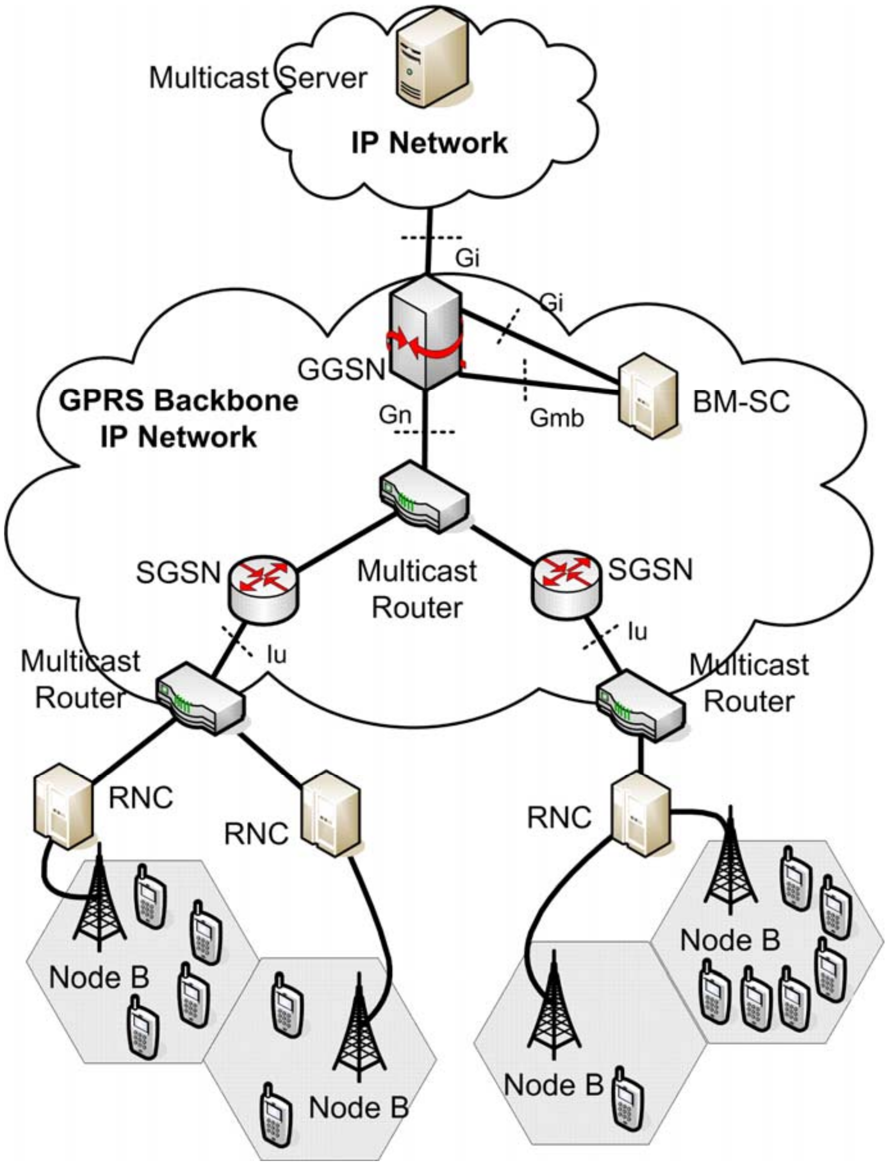

Figure 2. MBMS architecture for UMTS using IP multicast.

The BM-SC is a data source unique to MBMS. In this node the MBMS data are scheduled and interfaces are provided for the interaction with the content provider. The BM-SC may authorize and charge the content provider. At this point, it must be clarified that the data source may not originate from an external PDN, but may also originate from within the UMTS network.

In order to reduce the implementation costs, the MBMS has been designed to introduce only minor changes to existing radio and core network architectures. For simplicity reasons, in our analysis, we will consider the functionality of the BM-SC incorporated in the GGSN.

The reception of an MBMS multicast service is enabled by certain procedures. These are: Subscription, Service Announcement, Joining, Session Start, MBMS Notification, Data Transfer, Session Stop and Leaving, [17] and [18]. Figure 3 presents the sequence of the MBMS multicast service phases.

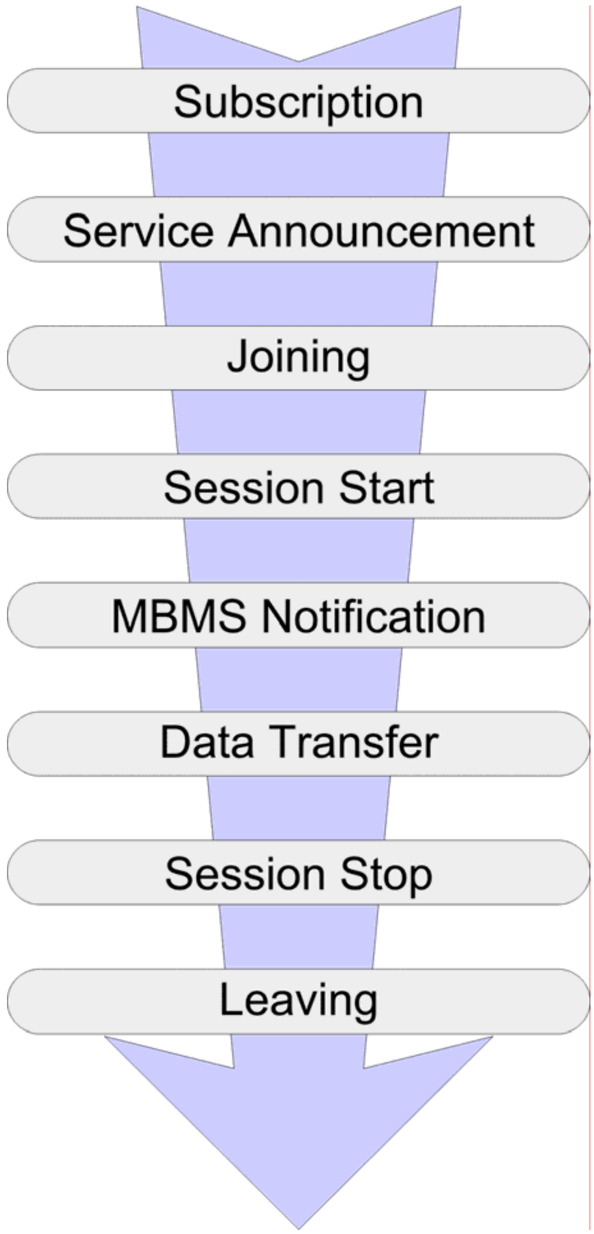

Figure 3. Phases of MBMS multicast service provision.

3.4 CLR Selection Problem

In wireless communication systems like UMTS, the packet loss may not mean network congestion. The quality of a wireless link may be degraded due to signal fading. During a fading period, the bit error rate of the wireless link may become very high but, normally, after that period the wireless link is expected to recover.

The traditional congestion control mechanisms translate the packet loss as buffer overflow in the network nodes, i.e. as network congestion. Consequently, the action taken in order to resolve this situation is the reduction of the sender's transmission rate. Nevertheless, if the packet loss is caused by fading, the reduction of the transmission rate will not affect the packet loss. This is due to the fact that the packet loss does not depend on the arrival rate of the packets but on the wireless channel degradation. Finally, the packet loss will be resolved after the end of the fading period, without a transmission rate regulation.

Obviously, the wireless channel degradation may affect the performance of the TFMCC mechanism. If we suppose that a UE suffers from fading, then the packet loss probability for this UE will temporarily increase. This increment of packet loss may cause the selection of this UE as the CLR. The next step is that the transmission rate of the multicast server will be reduced according to the packet loss of the examined UE. The problem is that this reduction is unnecessary because it is wireless-caused. After the recovering from bad wireless quality phase, the target throughput of the CLR will be improved. If a lot of UEs participate in the multicast group, there is a high probability that another UE suffers from fading. Soon, another UE suffering from channel degradation will be selected as CLR and will regulate the transmission rate. Eventually, the wireless-channel degradation will cause a significant and steady degradation of the performance of the TFMCC mechanism and of the multicast service. During this analysis we shall refer to this problem as the CLR selection problem.

4. THE PROPOSED MECHANISM

As we have already mentioned, the proposed mechanism follows a design very similar to that of the TFMCC scheme. Nevertheless, new functionality has been added to the existing mechanism in order to deal with the CLR selection problem.

The basic principles that govern the proposed mechanism are the following:

1. Each UE measures its packet loss rate using the packet loss history scheme of TFMCC.
2. Each Node B measures its packet loss rate. This information is written to the heading of the data packets and is then read by the UEs. This is a new functionality which combats the CLR selection problem in UMTS networks. This functionality does not exist in the TFMCC scheme and is explained below.
3. Each UE measures or estimates the RTT to the multicast server. This is achieved through an approach inherited from TFMCC. In more detail, timestamped feedback is sent to the multicast server. The server then echoes the timestamp and the corresponding UE_id in the header of a data packet. This approach causes minor traffic overhead in the network.
4. Each UE uses a control equation to calculate an acceptable sending rate from the sender back to it. The input parameters for the control equation are the loss rate and the RTT measured by the UE.
5. The feedback scheme of TFMCC is adopted. This scheme has devised a way in that the feedback from the receiver calculating the slowest transmission rate always reaches the sender. In addition, the feedback is filtered using randomized timers in order to avoid a feedback implosion.

In the proposed mechanism, the nodes located at the border between wireless and wired network (i.e. the Node Bs) have an additional responsibility. This responsibility is to provide the receivers (i.e. the UEs) with information about their measured packet loss. This means that each UE is informed by its serving Node B, of the packet loss that the Node B measures. This information is piggybacked in the data packets of a multicast session.

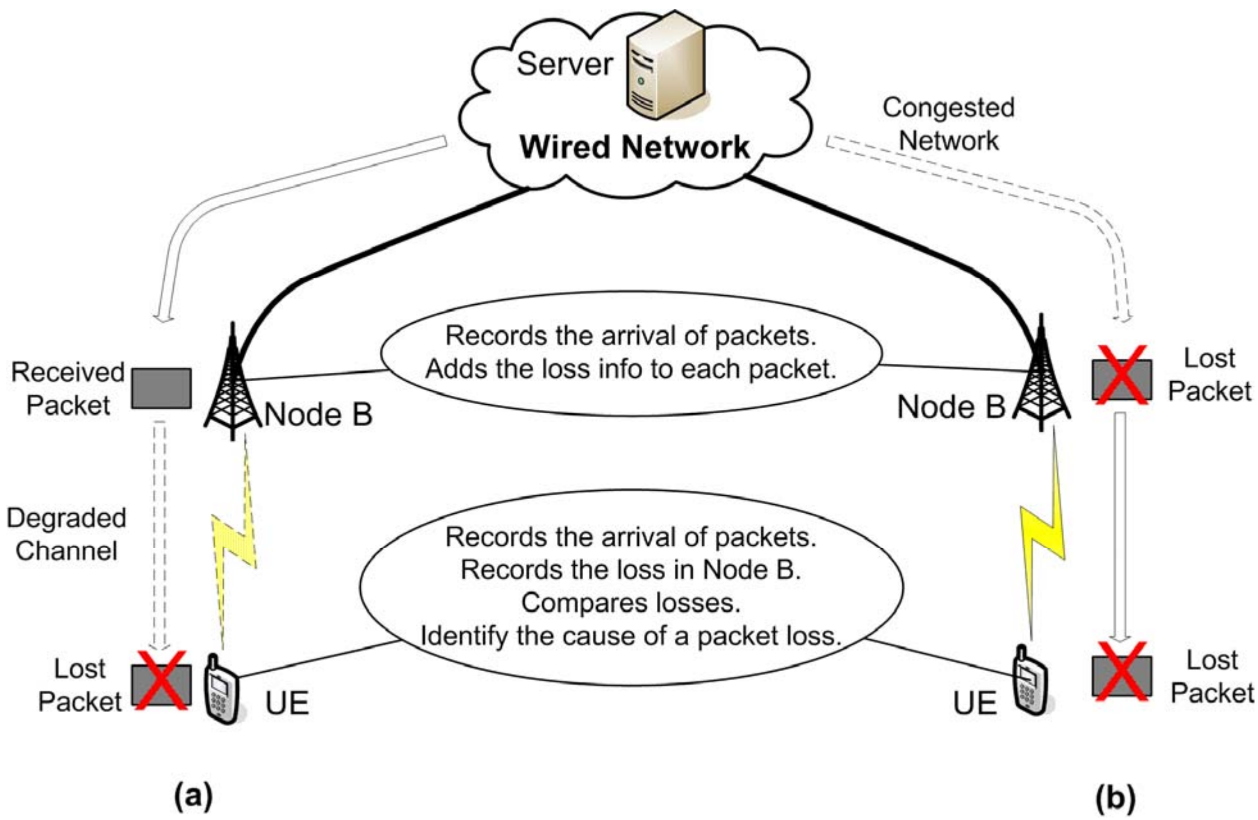

Figure 4. Packet losses at the UE: (a) Packet loss due to wireless channel degradation.

(b) Packet loss due to network congestion.

This additional functionality of the Node Bs, 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 the two values differ, the UE can conclude that the reason for the difference is losses at the wireless link caused by wireless channel degradation. 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 at the CLR selection. Figure 4(a) visualizes this functionality of the UE.

- On the other hand, when both the Node B and the UE encounter a packet loss, this packet loss is considered to be caused due to network congestion. Consequently, this kind of packet loss is taken into consideration during the CLR selection. This scenario is depicted in Figure 4(b).

At this point we will present another aspect of the proposed mechanism. Consider the case that, under certain conditions, a permanent degradation of the wireless channel affects a specific UE. When a permanent degradation occurs on the wireless link, the buffer of the Node B will overflow and some packets will be rejected. Normally, this UE should be a CLR-candidate. In the proposed mechanism, during permanent channel degradation, the Node B counts the rejected packets as general packet losses which happened due to network congestion. These packets are taken into account by the UE during the CLR selection. This scenario is illustrated in Figure 5.

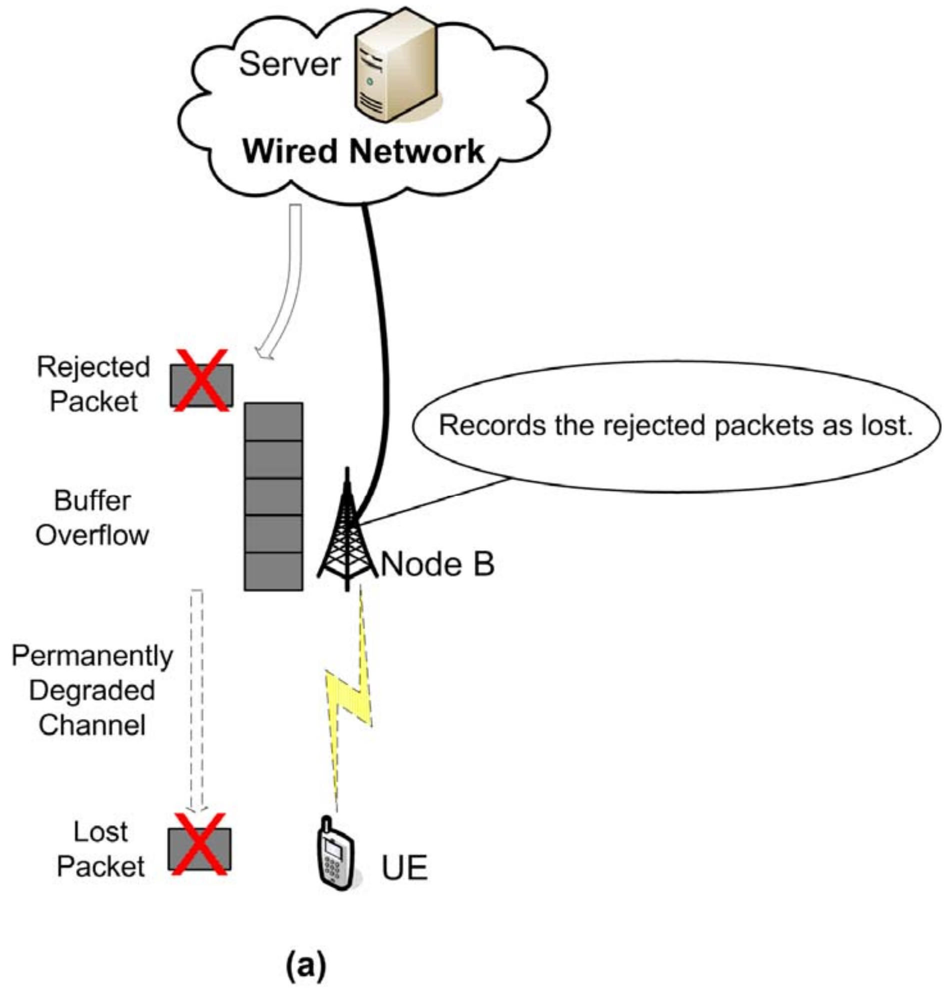

Figure 5. Permanent wireless channel degradation causes buffer overflow in the Node B.

As we mentioned above, this permanent degradation is not hidden from the UE. In fact, the UE is informed of the packet losses caused by buffer overflow. This functionality makes our proposed mechanism suitable not only with the CLR selection problem, but also with the permanent degradation of the wireless channel.

5. EXPERIMENTS

The above described mechanism was implemented and subjected to extensive evaluation through simulation. The evaluation was conducted towards two directions. The first was to examine that the proposed mechanism preserves the benefits of TFMCC as they are presented in [4]. The other was to evaluate the behavior of the mechanism against the CLR selection problem.

5.1 Simulation Environment

For the verification of the proposed mechanism the ns-2 network simulator [19] along with its EURANE extension were used. The Enhanced UMTS Radio Access Network Extensions (EURANE) for ns-2 [20] comprises of extensions for the support of UMTS network functionality. The three UTRAN nodes which ns-2 does not support, namely RNC, Node B and UE, are implemented in EURANE. Moreover, EURANE supports the functionality of the following transport channels: FACH, RACH, DCH and HS-DSCH.

Given the fact that the ns-2 simulator does not support the multicast transmission in UMTS, we implemented the multicast packet forwarding mechanism described [21]. In order to use this multicast scheme, we had to introduce the Routing Lists in each node of the UMTS network except for the UEs.

The next step was the installation of the TFMCC scheme. The codes used to implement and evaluate the TFMCC by the authors of [4] are provided at [22]. We adopted this TFMCC implementation in order to evaluate our proposed variation of TFMCC.

Finally, the generic TFMCC was modified and extended in order to support the UMTS environment. In more detail, the implementation of the TFMCC was enhanced in order to support the functionality of the Node B and the UE as described in subchapter 4. The Node B implementation was modified in order to provide the UEs with information about its measured packet loss. This means that each UE is informed by its serving Node B, of the packet loss that the Node B measures. This information is piggybacked in the data packets of a multicast session. One bit in the header of the data packet is enough for the provision of this information. On the other hand, the UE implementation was modified in order to read this bit and to take the decision whether a packet loss should be accounted at the calculation of its acceptable sending rate.

5.2 Fairness

The first aspect that we examined was the TCP-friendliness of TFMCC. In more detail, we considered the fairness of TFMCC towards the competing TCP flows when they share wired or wireless links. We tested the TFMCC fairness in various conditions. Below, we present the TFMCC behavior in a non-congested and in a congested UMTS network.

In the first place, fairness towards competing TCP flows was analyzed using a non-congested UMTS network (Figure 6). We monitored the throughput over a wireless link connected the UEx with Node By. We supposed that UEx belongs to a multicast group and receives TFMCC traffic. At the same time, this UE receives TCP traffic from an external node.

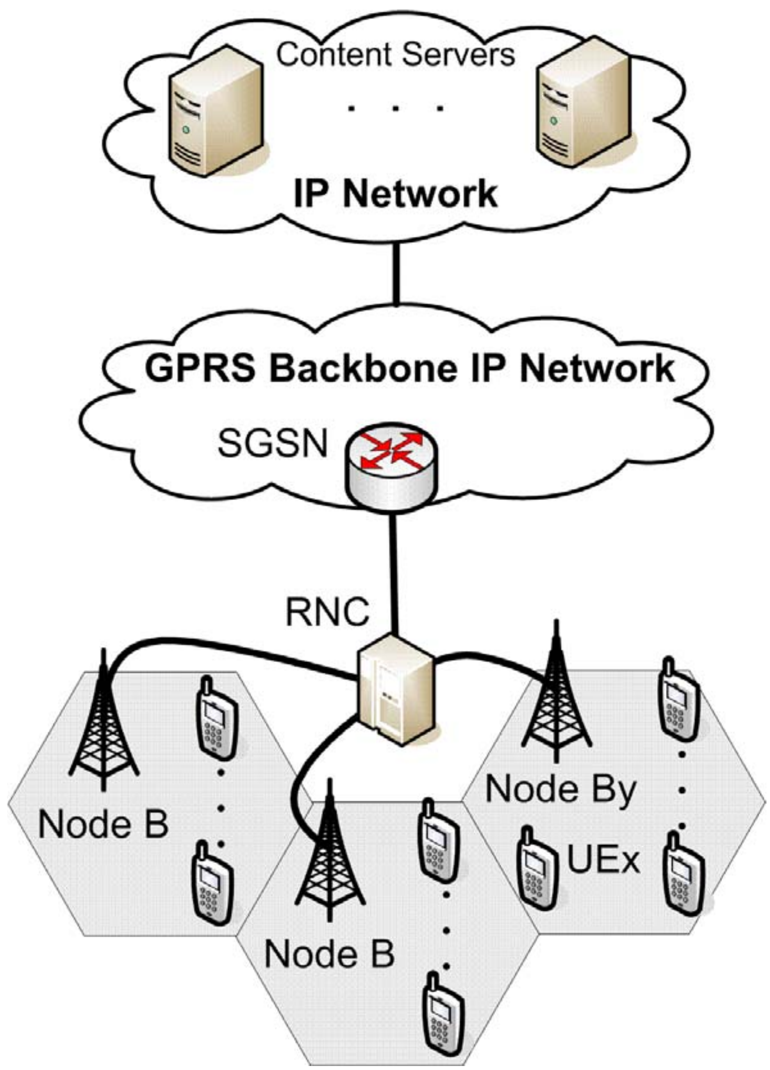

Figure 6. Non-congested UMTS topology.

Figure 7 illustrates the throughput of TFMCC flow against that of the TCP flow.

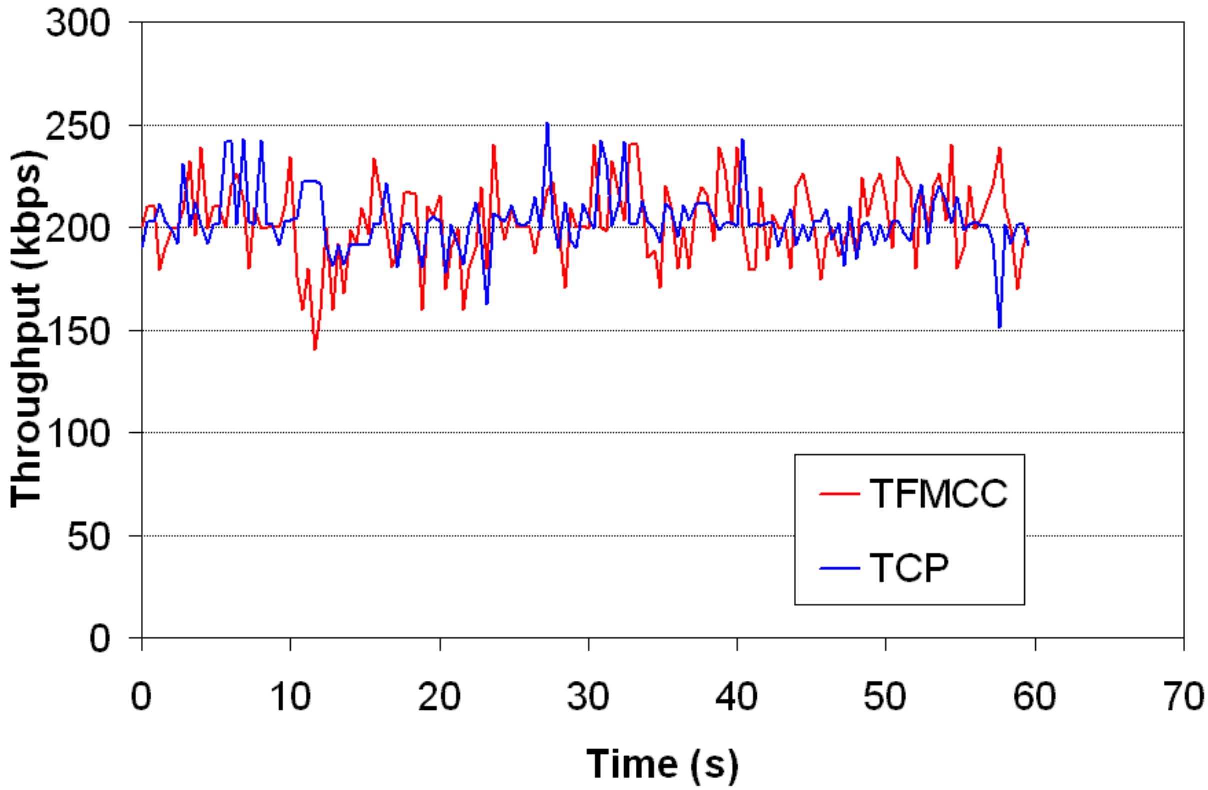

Figure 7. TFMCC flow vs. TCP flow in a non-congested UMTS network.

Due to our initial assumption that no congestion exists, the capacity of the wireless link poses a threshold of 384kbps for the throughput of the flows towards the examined UE. As it was expected, the available bandwidth is fairly shared between the flows. Figure 7 confirms that the average throughput of TFMCC flow closely matches the average TCP throughput. Actually, both the average throughputs match the half of the available capacity of the UTRAN wireless link.

The next step of our experiment was to examine the fairness of the mechanism in a congested UMTS network. We considered the single-bottleneck topology depicted in Figure 8. The bottleneck was applied over a link which connects an SGSN with an RNC node (Iu interface).

As shown in Figure 8, a number of sending content servers are connected to a number of receiving UEs through a common bottleneck. In more detail, 15 servers send TCP traffic to as many UEs, whereas 5 multicast servers send TFMCC traffic to as many multicast groups.

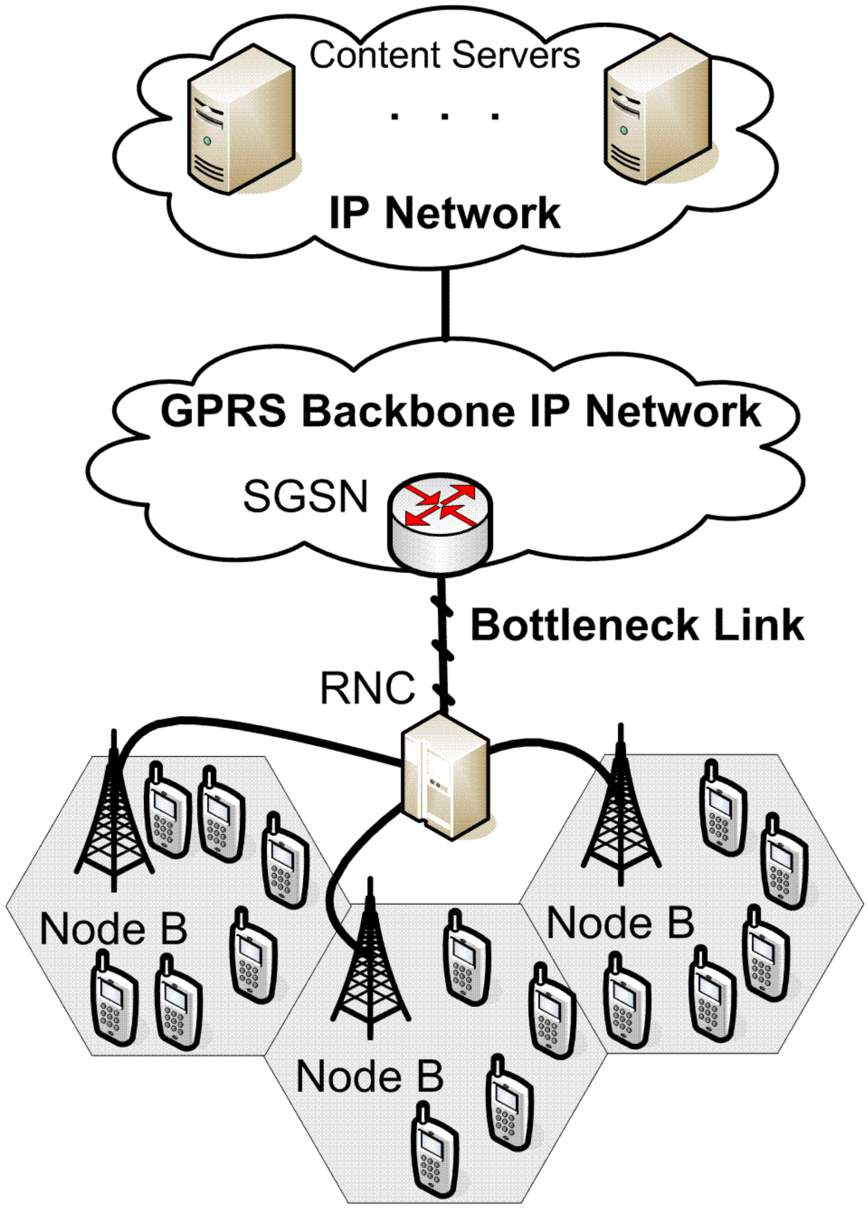

Figure 8. Single-bottleneck topology.

Figure 9 shows the throughput of a TFMCC flow against two sample TCP flows (out of 15). The average throughput of TFMCC closely matches the average TCP throughput. Moreover, TFMCC achieves a smoother rate than the TCP.

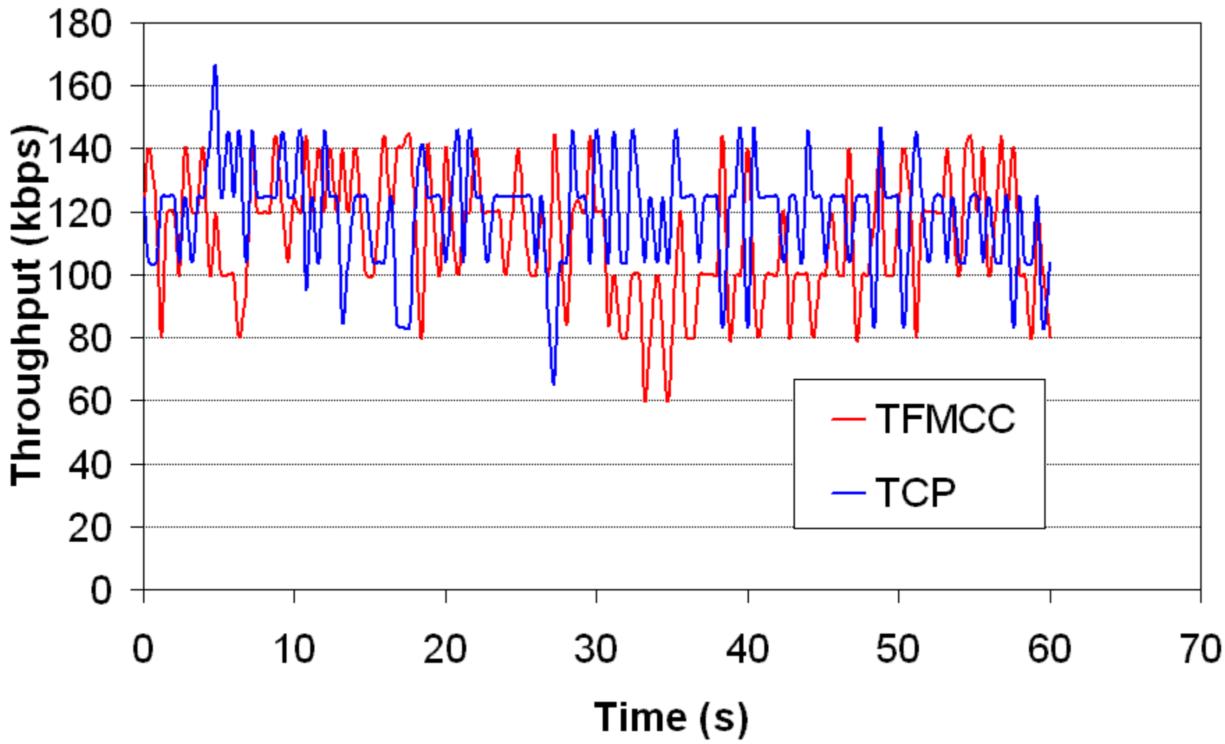

Figure 9. TFMCC flow vs. TCP flow in a single-bottleneck UMTS network.

Similar results can be obtained for many other scenarios. For example, if we suppose that congestion exists over a Gn interface (connects GGSN with SGSN nodes). In this case, the available throughput of the bottleneck link is evenly shared among the competing TFMCC and TCP flows. The TCP-friendliness of the proposed scheme was therefore confirmed under all the congestion scenarios.

5.3 Responsiveness to Changes

An important concern in the design of congestion control protocols is their responsiveness to changes in the network conditions. This behavior was investigated using the single bottleneck topology of Figure 8.

During the simulation we changed the applied loss rate of the bottleneck link. The simulation lasted 150 seconds. During this time interval three different loss rates were applied on the Iu interface. The TFMCC flow was monitored along with two TCP flows sharing the bottleneck link. The results of the simulation for the three competing flows are presented in Figure 10.

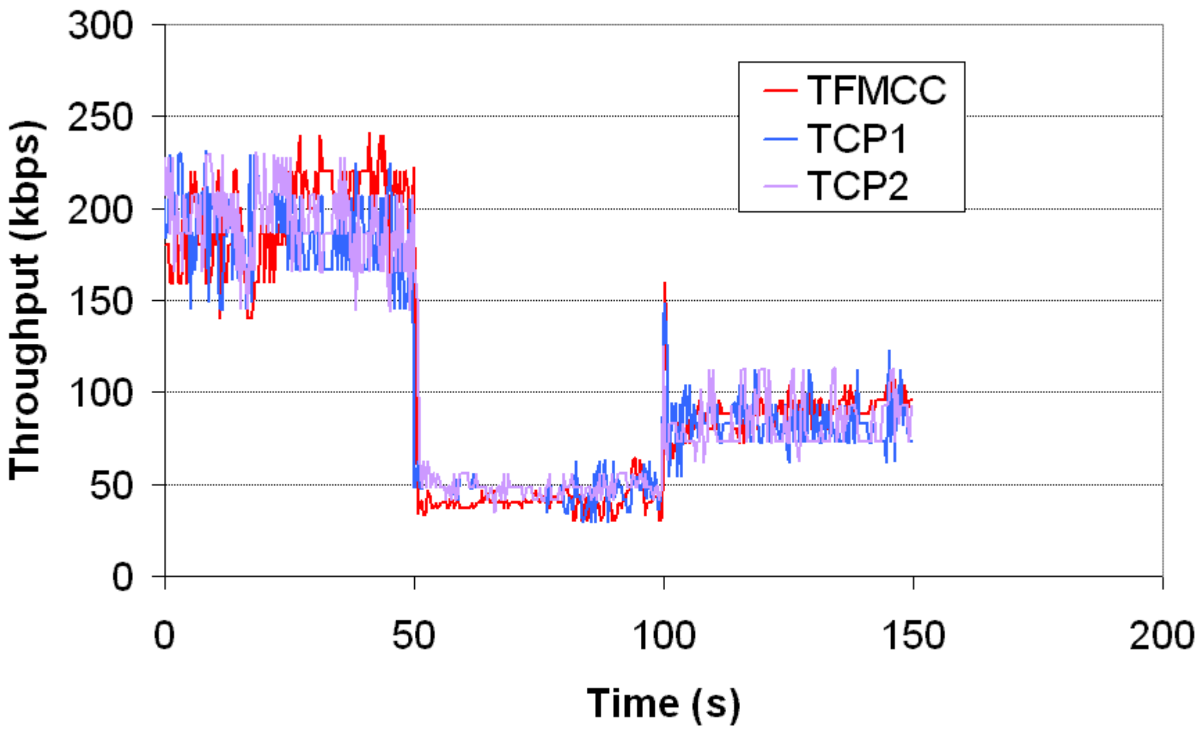

Figure 10. Responsiveness to changes in the loss rate.

As shown in Figure 10, TFMCC matches closely the TCP throughput at all three loss levels. Moreover, the adaptation of the sending rate is fast enough. Actually, the simulator logs show that the UEs need 1500-2000ms 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).

A similar simulation setting was used in order to investigate the responsiveness to changes in the RTT. The results are similar to those above. The above experiment confirms the excellent reactivity of the TFMCC to changes in congestion level of the UMTS network. Moreover, it confirms that during the application of TFMCC over the UMTS the properties and the benefits of this scheme are not affected.

5.4 Reaction to Wireless Channel Degradation

The next concern of our experiments was the evaluation of the proposed scheme when wireless-caused packet losses occur. We simulated a UMTS network and assumed a degradation of the wireless channels. In more detail, we simulated the wireless channel degradation by applying an error rate over the packets transmitted via the wireless links. We examined the proposed scheme for different number of UEs belonging in the multicast group.

In Figure 11, our proposed scheme is referred as modified TFMCC (mod_TFMCC). On the other hand, the typical TFMCC algorithm presented in [4] is referred as TFMCC. The horizontal axis shows the number of the UEs belonging in the examined multicast group. Both mechanisms were examined for up to 100 UEs participating in the multicast group. The vertical axis shows the average throughput which is normalized to the corresponding TCP one. The results when 5% wireless-caused packet loss is applied are presented in Figure 11.

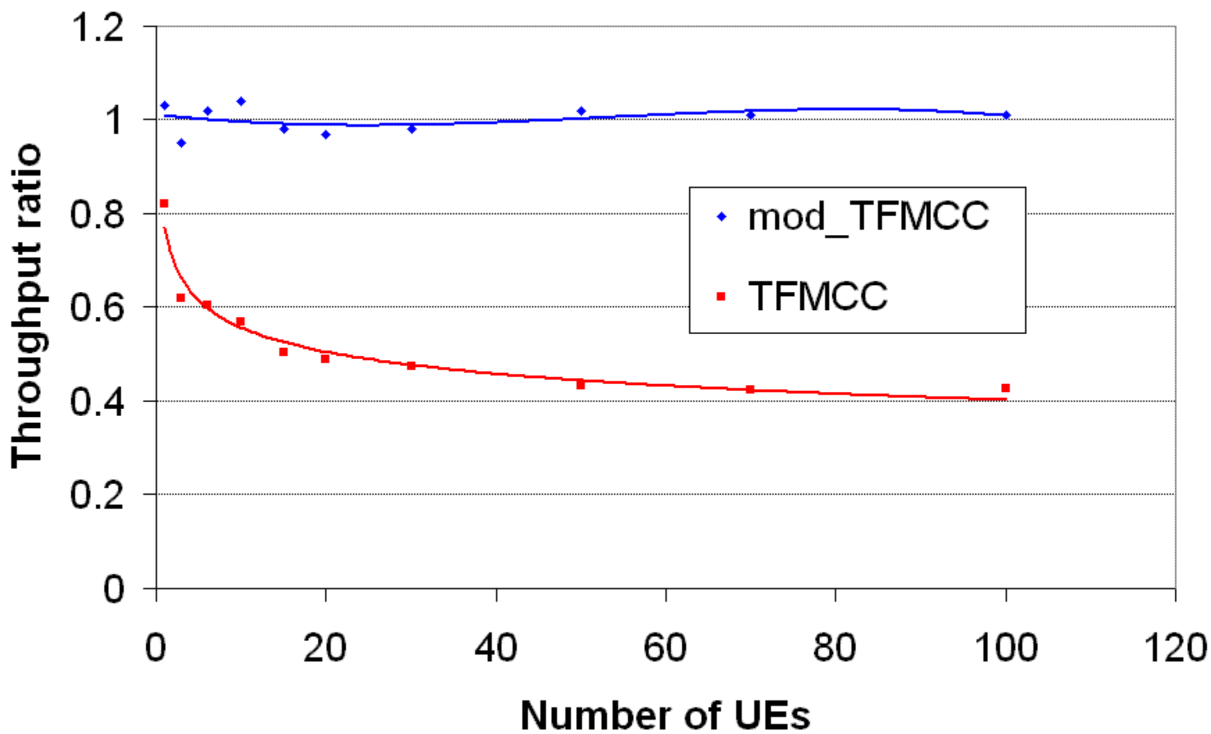

Figure 11. Throughput of our proposed scheme vs. TFMCC when wireless-caused packet losses occur.

The results depicted in Figure 11 confirm the excellent behavior of our proposed scheme when wireless channel degradation occurs. The wireless-caused packet losses can be identified correctly at the UEs and be ignored at the calculation of the acceptable sending rate. This means that the CLR selection problem can be overcome and significant improvement is added on the TFMCC application over the UMTS.

5.5 Permanent Wireless Channel Degradation

The last concern of our experiments was the evaluation of the proposed scheme when wireless-caused packet losses occur in a permanent manner. We examined the behavior of the modified TFMCC when a wireless channel is permanently degraded so as to lead to buffer overflow and packet rejections in the corresponding Node B. Figure 12 depicts the simulation setting. In the

UMTS network of this setting we assumed a permanent degradation of the wireless link that connects the UEx with the Node By.

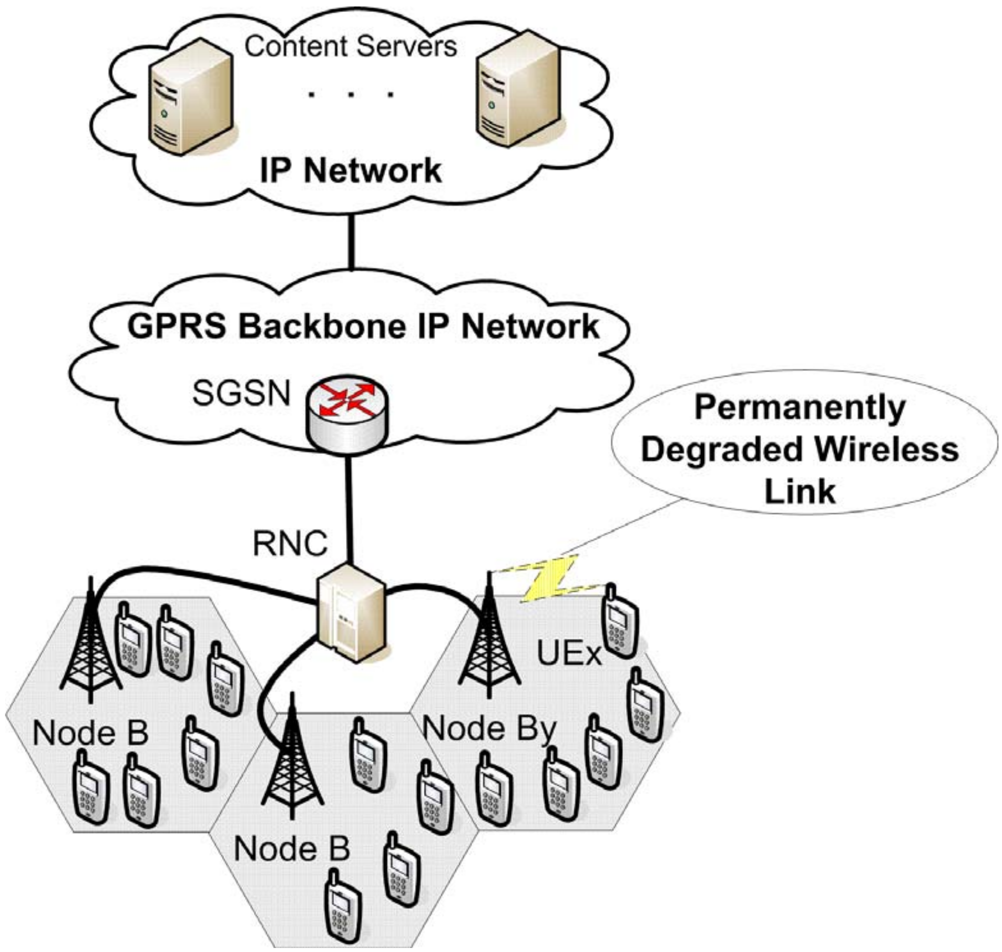

Figure 12. Permanent wireless-channel degradation topology.

In more detail, we simulated the wireless channel degradation by applying an error rate of 50% over the packets transmitted via the corrupted wireless link. In the beginning of the simulation, no wireless channel degradation occurred. After 50 seconds of simulation, we applied the error rate over the wireless channel connecting the UEx with Node By. We monitored the changes over the throughput of the corrupted wireless link for 100 seconds. The results of our experiment are presented in Figure 13.

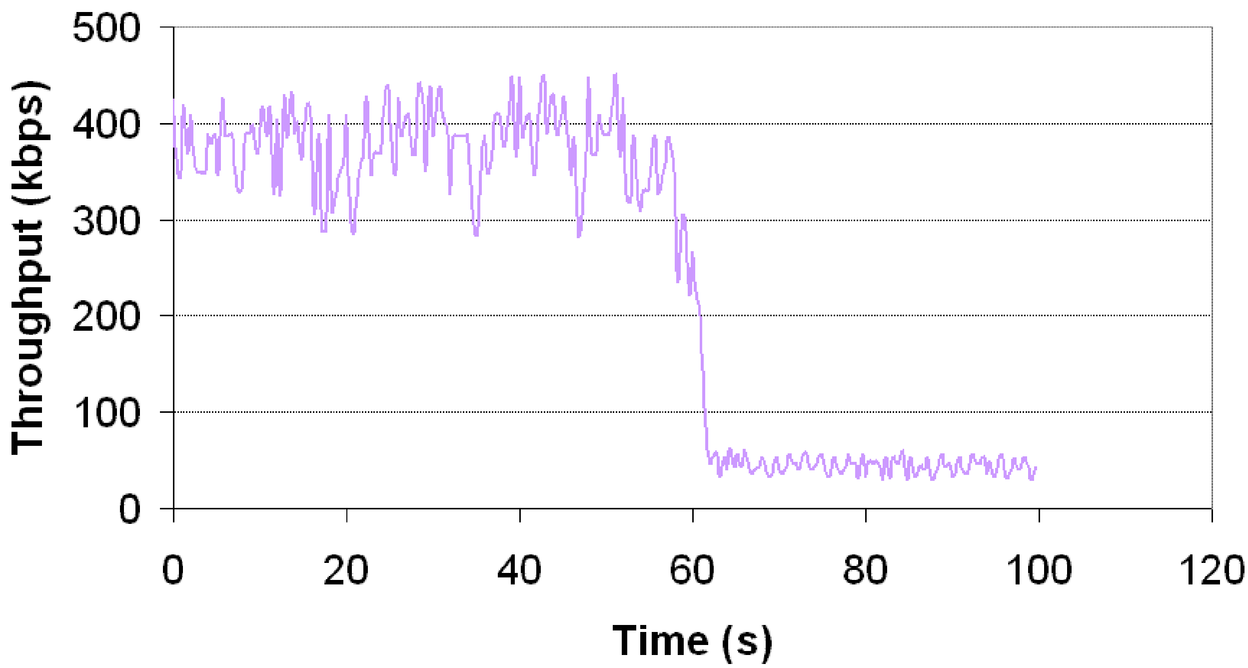

Figure 13. Throughput of our proposed scheme vs. TFMCC when wireless-caused packet losses occur.

The simulation results prove that our proposed scheme reacts to the permanent wireless-channel degradation. In the beginning of the simulation, no congestion exists and the throughput matches with the available bandwidth of the wireless link. After 50 seconds of simulation, the 50% packet error rate is applied. Obviously, the network does not immediately react to this degradation because it considers it as wireless-caused degradation.

During the period between 50 and 60 seconds of simulation, the buffer of the Node B overflowed and the network was able to distinguish the nature of the degradation. It took about 10 seconds to adapt to the new network conditions, but this time interval may differ according to the bit-rate of the transmission and the size of the buffer in Node B.

The results confirm the behavior of our proposed scheme when permanent wireless 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. Eventually, this kind of packet losses causes reduction of the transmission rate.

In case that multiple wireless channels are degraded, the simulation results (not listed here) are similar. The CLR is a UE among the ones being connected with degraded wireless links and calculating the minimal acceptable sending rate.

6. CONCLUSIONS

We have described a congestion control scheme for the multicast transmission over UMTS. The proposed scheme is based on the well known TFMCC mechanism. The TFMCC is a TCP-friendly, single-rate multicast congestion control mechanism intended to scale to groups of several thousand receivers. Nevertheless, the legacy TFMCC algorithm has been designed for fixed networks and had to be modified before being applied to wireless networks.

In wireless communication systems like UMTS, the CLR selection problem appears. This is because in this kind of networks the packet loss may not mean network congestion but may be caused by wireless link degradation. The proposed scheme does not translate the wireless-caused packet loss as buffer overflow in the network. Consequently, no reduction of the sender's transmission rate is performed in order to resolve this situation.

We have evaluated the proposed scheme through simulation experiments. We concluded that it preserves the benefits of TFMCC algorithm over the UMTS cellular network. The mechanism is fair towards competing TCP flows over congested and non-congested links. Moreover, the results of the experiments demonstrate the very good reactivity to changes in the congestion level for both loss rate and RTT.

Additionally, simulation experiments were performed in order to examine the proposed scheme against the CLR selection problem. The results confirm the excellent behavior of our proposed scheme when wireless channel degradation occurs. The wireless-caused packet losses can be identified correctly at the UEs and are ignored at the calculation of the acceptable sending rate. This means that the CLR selection problem can be overcome and significant improvement is added on the TFMCC application over the UMTS.

Finally, we examined the behavior of our proposed scheme against permanent wireless 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.

Minor modification in the UMTS architecture is needed. Actually, the impacts concern only two nodes of the UMTS network; the Node Bs and the UEs. The additional functionality in UE is to examine whether a packet loss is wireless caused or not. The reading of a single bit in each multicast packet header is sufficient for this purpose. On the other hand, each Node B has to set this bit info over the packet header if it has encountered a packet-loss. The proposed scheme therefore respects the limited computing power of the UEs and no demanding operation is introduced in those network nodes.

7. FUTURE WORK

The step that follows this work is the evaluation of different congestion control schemes for UMTS networks. Other TCP- friendly multicast congestion control schemes like pgmcc [23] will be investigated and modified in order to meet the UMTS requirements. Additionally, the applicability over UMTS of different multicast congestion control approaches will be examined.

Multicast architectures like multi-rate [9], layered [10], end-to-end services [4], [10] and [11] and active services [12] will be evaluated for their applicability. These emerging schemes will be also evaluated through comparison. This comparison will examine several aspects like the efficiency and the cost of implementation of each scheme.

Furthermore, we will try to formulate a multicast group control mechanism dedicated for the UMTS networks. In some cases, permanent wireless channel degradation may cause a large reduction to the transmission rate and eventually multicast service degradation. It will be specified under which circumstances wireless channel degradation will cause rejection of a corrupted UE from a multicast group.

Furthermore, we will try to take advantage of the broadcasting nature of the wireless channels. This broadcasting nature is a promising platform for enhancing the legacy multicast schemes and implementing efficient wireless multicast schemes.

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