Impact of Acknowledgments on Application Performance in 4G LTE Networks

by

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Abstract

4G LTE is a new cellular phone network standard to provide both the capacity and Quality of Service (QoS) needed to support multimedia applications. Recent research in LTE has explored modifications to the current QoS setup, creating MAC layer schedulers and modifying the current QoS architecture. However, what has not been fully explored are the effects of LTE retransmission choices and capabilities on QoS. This thesis examines the impact of using acknowledgments to recover lost data over the wireless interface on VoIP, FTP and MPEG video applications. Issues explored include interaction between application performance, network transport protocols, LTE acknowledgment mode, and wireless conditions.

Simulations show that LTE retransmissions improve FTP throughput by 0.1 to 0.8 Mb/s. With delay sensitive applications, like VoIP and video, the benefits of retransmissions are dependent on the loss rate. When the wireless loss rate is less than 20%, VoIP has similar performance with and without LTE retransmissions. At higher loss rates the use of LTE retransmissions adds degrading the VoIP quality by 71%. With UDP video, the choice of retransmissions or not makes little change when the wireless loss rates are less than 10%. With higher wireless loss rates, the frame arrival delay increases by up to 539% with LTE retransmissions, but the frame rate of the video decreases by up to 34% without those retransmissions.

LTE providers should configure their networks to use retransmission policies appropriate for the type of application traffic. This thesis shows that VoIP, FTP and video require different configurations in the LTE network layers.

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1 Introduction

In recent years, mobile technology has increased both in capability and penetration in everyday use. Cisco reports that over half a billion mobile devices and connections were added in 2013 and 77% of this growth came from smartphones [16]. Global mobile data traffic grew 81% and mobile video traffic took up 53% of all mobile data traffic in 2013. Cisco also found that 4G connections generated 14.5 times more traffic on average than non 4G connections. While the total number of 4G connections only account for 2.9% of all mobile connections they account for 30% of mobile traffic. Cisco predicts that by 2018 4G traffic will account for half of all mobile traffic. With the growing usage of 4G networks research into LTE and other 4G protocols is important for businesses and users alike.

A common problem for all networks is managing loss and minimizing the delay of data. For wireless networks this is especially problematic as the wireless link is constantly shifting in signal quality based on local interference sources like buildings or other radio wave emitting objects and mobility of the communicating nodes. All wireless networks need to define how to handle diverse radio environments while maintaining a good quality for users.

As the noise in a signal seen by a device increases, LTE decreases the amount of bits sent to improve the chances of the data arriving without error. When data is lost LTE provides retransmissions at two different network layers to reduce loss. LTE also has multiple parameters that adjust how retransmissions are managed in the network. However, the exact settings used by a network are not known to its users or application developers. Further, the network provider may not know what settings they should use.

The choice of how retransmissions are managed has a large impact on the end user applications. Some programs require all data sent to be received without error like emails or file transfers. However other application like VoIP or live video streaming need minimal network delay and can accept some loss. With the multiple ways LTE can configure retransmissions it is possible that some network configuration may be beneficial to some types of applications

while harmful to others.

In previous research into LTE retransmissions authors have focused on one of two retransmission techniques at the Media Access Control (MAC) and RLC layers in LTE but not both. Kawser et al. [29] looked at how the maximum number of MAC layer retransmissions do not always improve the total amount of data lost. Makidis [30] simulated RLC retransmissions with Web and FTP traffic. While this work shows some impacts of using retransmissions it does not look at any of the adjustable parameters with the RLC layer. Other research into LTE retransmissions like Asheralieva et al. [9] looked at VoIP simulations in LTE with and without MAC layer retransmissions. This work did not look at the impact of the RLC layer retransmissions.

This work examines how configurations in the RLC layer for retransmission of data lost in the wireless last hop impact applications like VoIP and video that are delay sensitive and FTP, which is throughput sensitive. Preferred settings are found through simulations of an LTE network with varied loss rates on the wireless link. These simulations are supported by an open source To complete this work new code is added to the LTE simulator to support the use of negative acknowledgments (NACKs) in acknowledged mode (AM), which is part of the LTE specification but was not implemented in the simulator.

A built in capability of the simulator sets when the wireless noise is great enough to lose data. A Gilbert Elliot model is used to define at what time, how long and what radio frequencies experience enough noise to induce loss. First, experiments use VoIP, FTP and video applications at 10% wireless loss (a goal according to the LTE specification). These tests adjust two parameters in LTE, the timers t-Reordering and t-StatusProhibit. The results indicate what settings are preferred for the applications. Next these parameters are fixed based on the earlier findings and the amount of wireless loss is adjusted. From these tests we find if the applications perform better when LTE performs extra retransmissions to recover lost data or not.

Simulations show that LTE retransmissions improve FTP throughput by 0.1 to 0.8 Mb/s.

With delay sensitive applications, like VoIP and video, the benefits of retransmissions are dependent on the loss rate. When the wireless loss rate is less than 20%, VoIP has similar performance with and without LTE retransmissions. At higher loss rates the use of LTE retransmissions adds degrading the VoIP quality by 71%. With UDP video, the choice of retransmissions or not makes little change when the wireless loss rates are less than 10%. With higher wireless loss rates, the frame arrival delay increases by up to 539% with LTE retransmissions, but the frame rate of the video decreases by up to 34% without those retransmissions.

In Section 2 background needed for this work is covered. This includes an overview of LTE and how retransmission work, the simulation tool this work uses, and the types of applications tested. In Section 3 other research into LTE, retransmissions in LTE and types of applications people use on mobile devices. Section 4 covers the specific questions this work looks at and how they are tested. The results of these simulations are in Section 5. This work is then concluded in Section 6 and a brief description of future work is provided in Section 7.

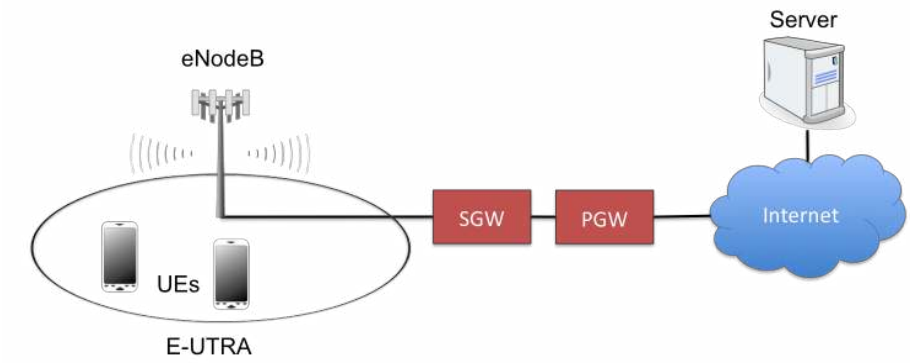


Figure 1: Evolved Packet Core

2 Background

2.1 Overview of 4G LTE

Long Term Evolution (LTE) is one of the current 4G cellular phone technologies deployed in multiple countries around the world. The core network for LTE is known as System Architecture Evolution (SAE). This network is split into two primary areas, evolved packet core (EPC) that makes up the connection from gateways on Internet protocol (IP) networks to the enhanced node b (eNodeB) access points and evolved universal mobile telecommunications system terrestrial radio access (E-UTRA) that makes up the wireless last link to user devices known as user equipment (UE) [6]. Unlike previous phone network technologies like global system for mobile communications (GSM), general packet radio service (GPRS) and niversal Mobile Telecommunications System (UMTS), LTE is entirely packet switched. GSM was a circuit switched network while GPRS and UMTS supported both circuit and packet switching [21]. The LTE network is depicted in **Figure 1**. The components of the network are:

- Serving Gateway (SGW) - A node that determines which eNodeB to route and send packets to.
- Packet Delivery Network Gateway (PGW) - Connectivity to external networks like the

Internet.

- enhanced node b (eNodeB) - Wireless access point for the radio interface E-UTRA.
- user equipment (UE) - End devices that connect to the LTE network such as phones, tablets and computers with LTE connect cards.

2.1.1 Network Layers

In this section we give a brief overview of the layers that make up the wireless link in LTE. From bottom to top these are the physical, MAC, RLC and Packet Data Convergence Protocol (PDCP) layers that handle both user data and LTE control messages. Above these layers LTE has additional control layers called the Radio Resource Control (RRC) and Non Access Stratum (NAS). The following sections pay particular attention to topics needed for understanding retransmissions in LTE and size of transmission opportunities.

2.1.1.1 Physical Layer

The physical layer handles transmitting data over radio waves between the eNodeB and UE. LTE can transmit data in physical resource blocks (PRBs) that take up 0.5 ms slots in time and 15kHz of radio bandwidth. The more contiguous radio frequencies a LTE network has the more data can be sent in parallel. The smallest physical resource LTE can send is a transport block that takes up 1 ms transmission time interval (TTI) (2 slots) and some number of PRBs depending on what the network has available and what the scheduler allows. In LTE the maximum number of PRBs in a transport block is 100.

Channel Quality As for other wireless standards, LTE will monitor the quality of the radio signal and adjust the encoding rate for better performance. A UE will regularly check the receive signal quality and report a CQIs value (among others values) to the eNodeB. The CQI represents the data encoding scheme and code rate to use so that the block level error rate (BLER) of received data is at most 10% [17]. BLER is a major metric in determining

Table 1: CQI Table [4].

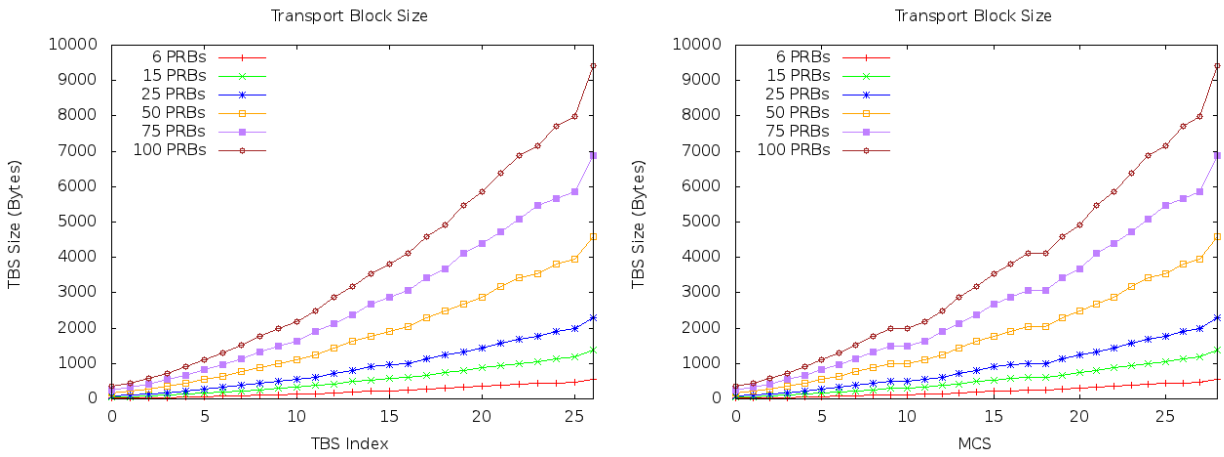
CQI index	Modulation	Code Rate x 1024	Efficiency
0	out of range		
1	QPSK	78	0.1523
2	QPSK	120	0.2344
3	QPSK	193	0.3770
4	QPSK	308	0.6016
5	QPSK	449	0.8770
6	QPSK	602	1.1758
7	16QAM	378	1.4766
8	16QAM	490	1.9141
9	16QAM	616	2.4063
10	64QAM	466	2.7305
11	64QAM	567	3.3223
12	64QAM	666	3.9023
13	64QAM	772	4.5234
14	64QAM	873	5.1152
15	64QAM	948	5.5547

how to send data in LTE. BLER is specifically defined as the ratio of the number of erroneous blocks received to the total number of blocks sent where an erroneous block is a transport block whose cyclic redundancy check is wrong [7]. The modulation determines how many bits can be encoded per orthogonal frequency-division multiplexing (OFDM) symbol (the method of encoding bits in the radio spectrum) and the coding rate is the ratio of the number of bits transmitted compared to the number of bits that could be transmitted. **Table 1** shows CQI values and their meaning.

The eNodeB takes the CQI value and determines a MCSs value between 0 and 31 (though 29 - 31 are reserved for future use). The MCS value selected will not necessarily map to the same modulation encoding that the UE requested with its CQI. The MCS values and their modulation (listed in 2, 4 and 6 bits) are shown in **Table 2** along with a TBS index. The TBS index is combined with the number of PRBs to determine how large the transport block sent is. Possible transport block sizes are given in table 7.1.7.2.1-1 of [4]. For example, a good signal to noise ratios (SINRs) will have a high MCS value of 28, which maps to a transport block index of 26. If there are 100 PRBs available then the transport block will be

Table 2: MCS Modulation and TBS Index [4].

MCS index	Modulation Order	TBS index
0	2	0
1	2	1
2	2	2
3	2	3
4	2	4
5	2	5
6	2	6
7	2	7
8	2	8
9	2	9
10	4	9
11	4	10
12	4	11
13	4	12
14	4	13
15	4	14
16	4	15
17	6	15
18	6	16
19	6	17
20	6	18
21	6	19
22	6	20
23	6	21
24	6	22
25	6	23
26	6	24
27	6	25
28	6	26
29	2	reserved
30	4	reserved
31	6	reserved

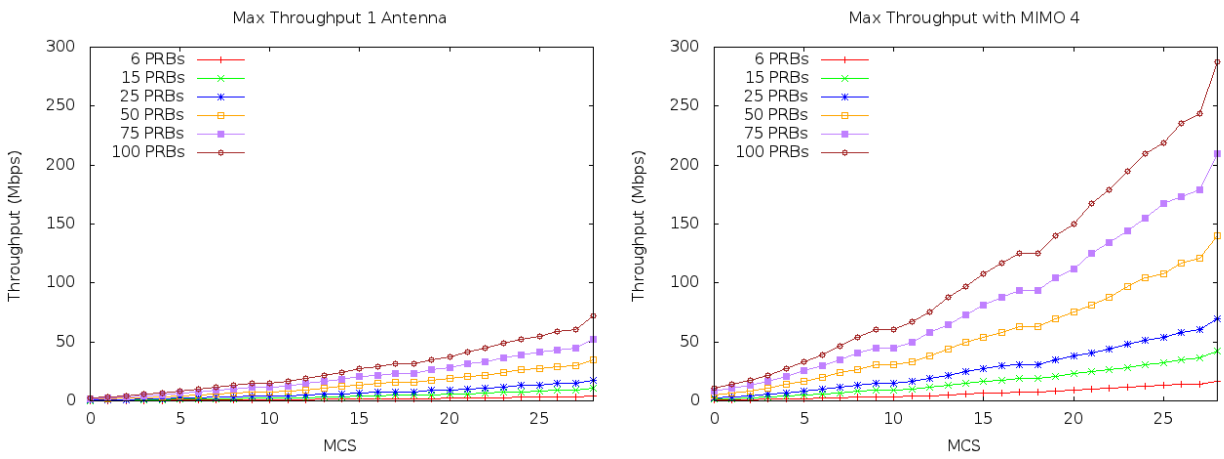


(a) TBS Index. (b) MCS value.

Figure 2: Maximum transport block size.

9,422 bytes. Conversely a bad SINR could produce an MCS value and transport block index of 0. If the same 100 PRBs are available then the transport block will be only 349 bytes. **Figure 2** shows the mapping of the size of the transport blocks in bytes based on the TBS index and MCS value. Here we show the results when 6, 15, 25, 50, 75 and 100 resource blocks are used. These are the maximum number of resource blocks available under six different bandwidth ranges that LTE supports [8]. The actual size of the transport block the user receives would also depend on how many of the available resource blocks the scheduler uses.

Maximum Downlink Throughput The maximum throughput a user could see in an LTE network is based on many factors. Assume that a user has ideal SINR conditions. Also assume that a user is the only person in a particular network cell so the scheduler on the eNodeB can give that one user's data all of the network resources. This would lead to transport blocks of size 75,376 bits. If we further assume that the user will not be sending any traffic back to the eNodeB then all transmission opportunities could have data for the user. With transport blocks sent every millisecond the user could get 71.9Mb/s of data. LTE also supports the use of transmitting and receiving on multiple antennas called



(a) Max throughput without MIMO.

(b) Max throughput with MIMO.

Figure 3: Maximum possible throughput in LTE

multiple input multiple output (MIMO). LTE defines that 1x1, 2x2, 3x2 and 4x2 MIMO setups are possible [42]. If a user had a phone that supported 4x2 MIMO (4 receiving and 2 transmitting antennas) then the maximum throughput the user could see would be 287.5Mb/s. The maximum possible throughput is shown in **Figure 3** for the six sets of maximum LTE bandwidths.

2.1.1.2 MAC Layer

The Media Access Control (MAC) layer maps logical and physical communication channels between the physical and RLC layers. The MAC layer provides scheduling and performs HARQ functions to recover damaged transport blocks [3].

Service Data Units (SDUs) and Protocol Data Units (PDUs) While this is being presented here the concept of service data unit (SDU) and protocol data unit (PDU) exist in the MAC, RLC and PDCP layers. A SDU is the packet of data that a layer gets from or gives to a higher layer. So the RLC layer would give the MAC layer a packet of data and this packet would then be considered a MAC SDU. A PDU is the packet of data that a layer gives to or gets from a lower layer. So the MAC layer would take its SDU, perform its

processing and then create a MAC layer PDU that it then provides to the physical layer. The MAC, RLC and PDCP layers will take SDUs from higher layers, perform some processing to create PDUs to give to lower layers and take PDUs from lower layers, perform some processing to create SDUs to provide to higher layers. The MAC layer packs only one SDU per PDU.

Hybrid Automatic Repeat Request (HARQ) The MAC layer's HARQ process attempts to correct for any data lost during transmission. For each transport block received the HARQ mechanism checks to see if the data was received successfully. This is done using forward error correction (FEC) bits and parity bits that come with the transport block [3]. If no errors are detected then an acknowledgment (ACK) is returned to the sender and the data is passed up to the higher network layers. If an error is detected then a NACK is returned and the transport block is kept. On future retransmissions multiple copies of the damaged data can be combined to create the original data [3]. More information on how these retransmissions work is in Section 2.1.3.

2.1.1.3 Radio Link Control (RLC) Layer

The Radio Link Control (RLC) Layer provides segmentation and reassembly between the original data frames and those encoded in the transport blocks [2]. How the RLC layer deals with segmentation and reassembly depends on the mode of transmission in use. The RLC layer has several transmission modes, transparent mode (TM) for control data, unacknowledged mode (UM) that just handles out-of-order reassembly and acknowledged mode (AM) that provides acknowledgments for transport blocks that could not be repaired or retrieved through the HARQ mechanisms at the MAC layer. While these three modes are different they all share some features. All modes support a variable size of RLC SDU that must be a multiple of 8 bits. The data PDUs made in the RLC layer come from RLC SDU handed down from the PDCP layer. Also all control PDUs are created in the RLC layer itself. All

three layers can only transmit PDUs when the lower MAC layer informs of a transmission opportunity.

transparent mode (TM) TM is used for control messages only and not user application data. There is no segmentation or concatenation of RLC SDUs . Unlike in UM and AM there are no RLC headers added to the PDU. Any PDU received by this layer are delivered directly to the PDCP layer as there has been no segmentation or reassembly to take into account.

unacknowledged mode (UM) UM is depicted graphically in **Figure 4**. When transmitting, the RLC layer will perform segmentation/concatenation of RLC SDUs to accommodate the size of the transport block, which is reported by the MAC layer. The receiving side of UM will reorder the PDUs as necessary. After a PDU arrives out of order a timer is used to wait just in-case the MAC layer HARQ process can recover the missing data. The RLC SDUs are created from the PDUs and delivered to the higher PDCP layer. Any SDUs that cannot be recreated due to loss from lower layers are discarded. The receiving side maintains a receiving window to support reordering of RLC PDUs delivered out of order.

UM Reordering Window The receiving side of an RLC UM entity keeps a set of state variables that define the reordering window. The reordering window tracks what PDUs are missing and may be recovered by the MAC layer HARQ retransmissions. This window is described by the following state variables.

- VR(UR) - The sequence number of the lowest PDU considered for reordering.
- VR(UH) - The sequence number following the highest received PDU.
- VR(UX) - The sequence number following the sPDU that triggered the t-Reordering timer. This timer is triggered when PDUs arrive out of order.

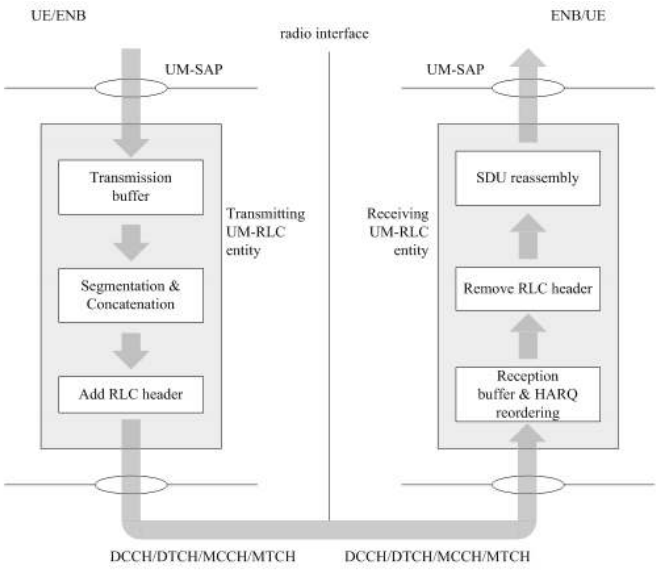


Figure 4: Unacknowledged Mode [2]

acknowledged mode (AM) AM is depicted graphically in **Figure 5**. Like UM this mode also performs segmentation and reassembly procedures. Unlike UM this mode can request retransmission of lost PDUs. To support these services the RLC layer uses transmitting/receiving windows and buffers along with the use of ACKs and NACKs to inform senders of lost PDUs that need to be resent.

AM Sliding Windows Data transmissions with RLC AM use a sliding windows. Each PDU the RLC layer creates in AM has a RLC header that includes a sequence number (SN). This SN is used with the sliding window so only PDUs with a SNs inside the transmitting window are sent and only PDUs with a SNs inside the sliding window are accepted (all others are dropped).

The sender's transmission window (as seen in **Figure 6**) can be described by four variables [2] [30]:

- VT(A) - The sequence number following the last in-sequence acknowledged PDU (lower bound of window).
- VT(S) - The sequence number of the next block of PDU to be transmitted for the first

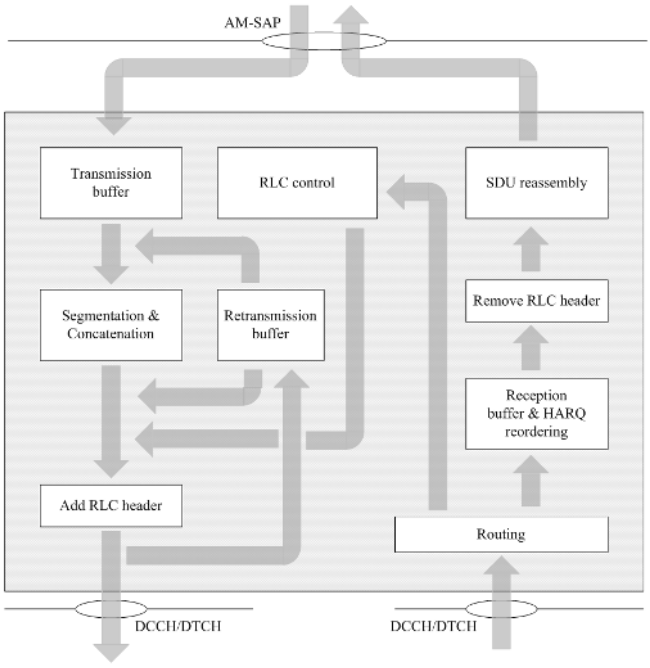


Figure 5: Acknowledged Mode [2]

time. It is updated once the data is transmitted.

- VT(MS) - The sequence number of the first PDU that can be rejected by the receiver (upper bound of window).
- VT(WS) - The size of the transmission window.

The receiver's reception window (as seen in **Figure 7**) can be described by five variables (note this figure displays three of the five) [2] [30]:

- VR(R) - The sequence number following the last in sequence PDU received (lower bound of window).
- VR(H) - The sequence number following the highest accepted PDU that has been received so far.
- VR(MR) - The sequence number for the first PDU that is rejected (upper bound of window).

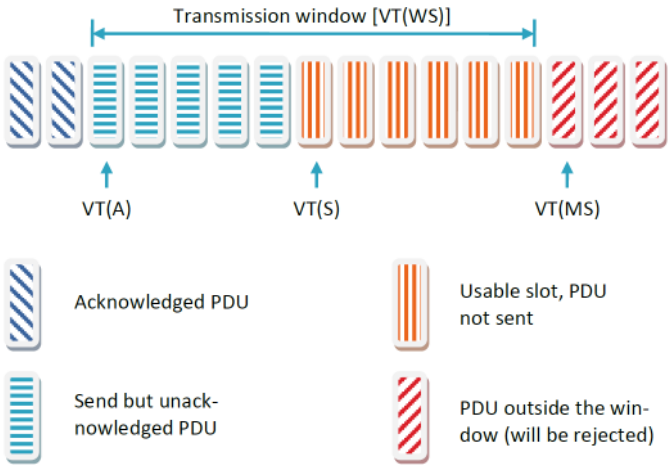


Figure 6: AM Transmission Window [30]

- VR(X) - The sequence number following the sequence number for PDU that triggered the t-Reordering timer. This timer is triggered when PDUs arrive out of order.
- VR(MS) - The sequence number that can be reported as the ACK sequence number in a STATUS PDU.

More information on how retransmissions work in the RLC layer is in Section 2.1.3.

Service Data Units (SDUs) and Protocol Data Units (PDUs) In the RLC layer PDUs are either data PDUs or control PDUs. Data PDUs can be transmitted over UM or AM while control PDUs are sent over TM [2]. When using TM one SDU is used to create one PDU with no header. In UM and AM the PDU could contain one or more full or partial SDUs along with headers. There are multiple possible sizes and formats for thePDU headers depending on the transmission mode and the use of data or control PDUs. One example of how RLC layer segmentation works with SDUs and PDUs can be seen in **Figure 8**. Here the red dotted lines indicate that an SDU was segmented to fit into the available PDU.

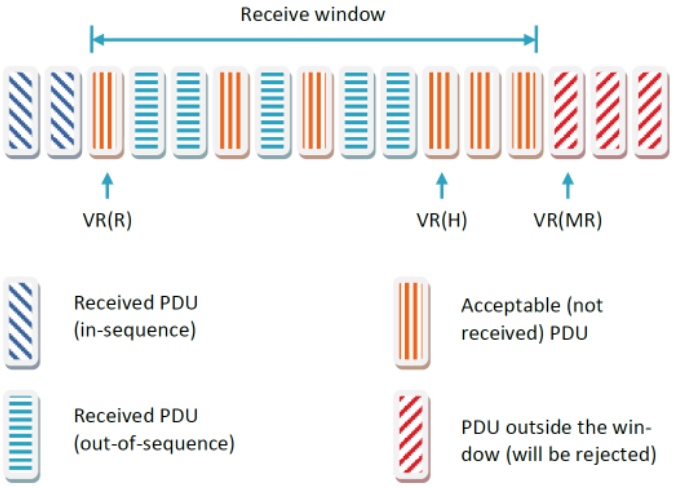


Figure 7: AM Reception Window [30]

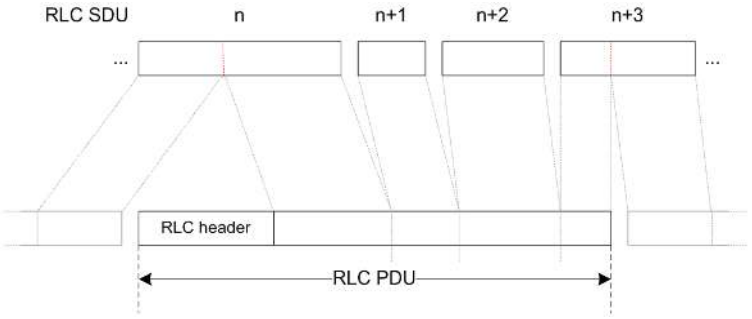


Figure 8: Example RLC SDU Segmentation [1]

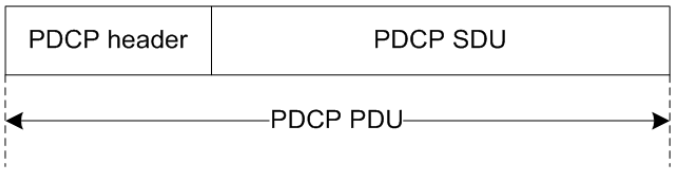


Figure 9: PDCP PDU [1]

2.1.1.4 Packet Data Convergence Protocol (PDCP) Layer

The Packet Data Convergence Protocol (PDCP) layer provides capabilities such as encryption/decryption, robust header compression for VoIP, integrity protection and SN to provide duplicate removal [1]. The structure of the PDCP PDU is fairly simple. One PDCP SDU makes up the payload and the header is either one or two octets long. The structure of this PDU can be seen in **Figure 9**. When carrying user data the PDCP SDU will contain one IP packet.

2.1.1.5 Radio Resource Control (RRC) Layer

The Radio Resource Control (RRC) layer is primarily concerned with the management of the connections in E-UTRA [1]. This is the highest LTE related layer on a UE. The eNodeB also has a RRC layer along with the NAS layer.

2.1.1.6 Non Access Stratum (NAS) Layer

The NAS layer exists only on the eNodeB and not on the UEs. Similar to the RRC layer it handles control functions though at a higher level. The services and functions of the NAS layer include quality of service (QoS) resource control, handling the mobility of idle devices, paging origination and the configuration and control of security [1].

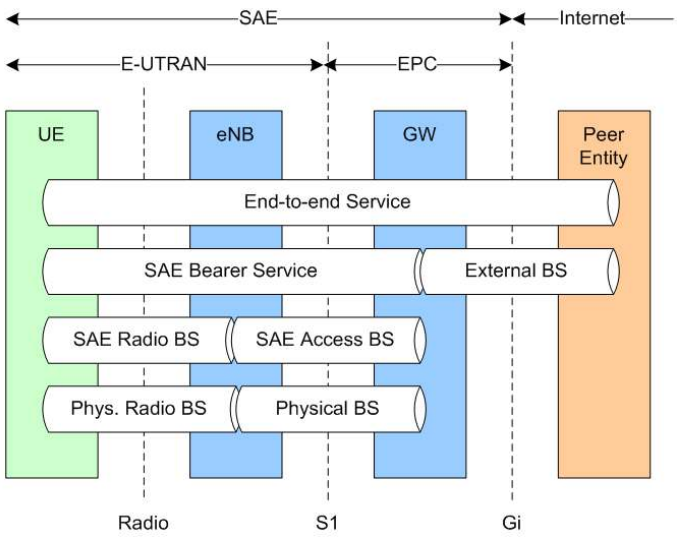


Figure 10: SAE Bearer Service Architecture [1]

2.1.2 Quality of Service

One of the main goals for 4th generation cellular network technologies is to provide QoS support to handle new demand for multimedia applications. The LTE specification defines some goals and capabilities to provide QoS and leaves some components up to implementers.

In LTE there are bearers, which uniquely identify packet flows that receive the same QoS treatment [20]. The structure of these bearers across the LTE network can be seen in **Figure 10**. There is a high level end-to-end bearer that logically connects the UE to whatever entity they are interacting with. In reality this bearer is made up of multiple components that need to deal with the specific protocols and physical layer capabilities that make up the SAE. For instance the bearer that connects the UE to the eNodeB is called a radio bearer as it deals with the physical radio interface. In LTE QoS features can be applied from the UE connected to the radio interface to the PGW. After this the data is being transmitted over the open Internet and the LTE network can no longer manage QoS.

Traffic gets mapped to the bearers based on a packet filter. These filters are called traffic flow templates (TFTs). The TFT filters packets based on protocol, IP address range, port numbers and uplink/downlink direction. All packet flows that are mapped to a particular

bearer receive the same packet-forwarding treatment like scheduling, queue management and other QoS techniques [20]. There is one bearer per QoS class (defined in the next section) and IP address of the UE [20]. While a UE will have multiple bearers they can be classified as guaranteed bit rate (GBR) and non-guaranteed bit rate (GBR). As their names suggest GBR bearers needs to support a minimal bit rate for all of its packet flows while non-GBR bearers make no such promise [20].

Bearers are also classified based on how they are created. A bearer can be either a default or dedicated bearer [20]. The default bearer is created when the UE first connects to the LTE network. This default bearer is also a non-GBR bearer as it must exist no matter what the network conditions are. Any other bearer that gets created to fill a particular QoS requirement is considered a dedicated bearer. The TFT of the default bearer allows all traffic while each dedicated bearer gets a specific TFT to separate its traffic from other flows.

2.1.3 LTE Retransmissions and Applications

For a developer the high bandwidth and QoS considerations in LTE are beneficial for many applications. However while the LTE specification provides many options there is little the application developer can do. The only features that the developer can potentially control that impact LTE are the IP address, port numbers and protocols to use when accessing content on the Internet. These features can trigger specific radio bearers as discussed in Section 2.1.2. From the application developers perspective however LTE is a black box. There is no information on what potential radio bearers are setup or how to access them. While a phone call is likely to get a high quality of service but a VoIP application a developer builds cannot indicate what their application needs. There is no information to the developer on if they will receive AM or UM operations in the RLC layer.

While there is no way for an application developer to know what they will be getting it would be helpful to know what settings in LTE cause what results on the end application. This work focuses on settings in the MAC and RLC layers as they deal with retransmissions

on the wireless link. For applications we look at examples with delay constraints such as live voice and video.

2.1.3.1 MAC

As described earlier the MAC layer will attempt to fix lost or damaged data using HARQ. HARQ is a stop and wait protocol so no new data can be sent until the transmission is acknowledged. This wait time would add large delays to LTE so the MAC layer compensates by maintaining multiple HARQ processes in parallel [17]. When frequency division duplexing (FDD) is used (uplink and downlink are separated by radio frequencies) their are eight HARQ processes for downlink and uplink traffic. When time division duplexing (TDD) is used (uplink and downlink are separated by transmission at specific TTIs) the number of processes for downlink and uplink is dependent on the TDD configuration.

The timing on retransmissions are dependent on if synchronous or asynchronous HARQ is used. With asynchronous the ID of the specific HARQ process is sent with the ACK or NACK. This way a receiver can change when they send this control message. With synchronous HARQ the receiver sends the ACK/NACK in a specific subframe, which indicates the HARQ processes it is associated with. In LTE the downlink uses asynchronous HARQ while the uplink uses synchronous [17].

For downlink transmissions HARQ will send up to 3 retransmissions [8]. If any of these packets arrive without error or if any of them run through soft combination (combining bad PDUs to create the original version) produces a valid packet then the data can be sent to the RLC layer. If after 3 retransmissions the data cannot be reproduced through soft combining then it is considered lost. Since multiple processes are sending data and possibly waiting retransmissions the MAC layer can receive good packets before an earlier bad packet is corrected. It is up to the higher RLC layer to re-order the data.

Since each HARQ process must wait for an ACK or NACK there is a delay before the next message can be sent. The delay comes from two times the propagation delay (sending

the message from one node to another and then the ACK or NACK) and the time to process the message at the receiver. Radio waves propagate at the speed of light and since LTE has to support cells as large as 100km from the tower [17] this translates to a propagation delay of less than a millisecond.

While the downlink uses asynchronous HARQ and can request retransmissions at an indeterminate time, there is still a schedule that leads to minimal delay. In FDD there are eight HARQ processes that transmit one after the next. This means that after a process transmits it will not get another opportunity until after the other 7 HARQ processes have sent their data. An example of this is given by [17] and is depicted in **Figure 11**. Please note that the transmission and reception windows shown in **Figure 11** for the UE are offset. This is meant to indicate how the UE needs to account for its distance from the eNodeB and adjust its clock to remain synchronous. This difference is called a timing advance and is managed with the help of the eNodeB.

The eight processes send a message every subframe (1 ms interval). Say that a HARQ process at the eNodeB sends its transport block at time n. After a propagation delay the message arrives at the UE at its time n. The UE then has until time n+4 to determine if an ACK or NACK is required. At n+4 the UE can send its control message, which arrives at the eNodeB at its time n+4. At this point a total of 5 HARQ processes could have sent data leaving another 3 processes to send their data before the original process can either send new data or resend old at time n+8. This gives us a round trip time of 8 ms for the HARQ operation when using 8 HARQ processes [17]. If the eNodeB needs to resend the data the maximum of 3 times then the UE will not get the last retransmission until time n+24. After processing (3 ms) the final transmission the UE would likely not be able to provide data to the RLC layer until about 28 ms. Now asynchronous HARQ could be used to change the order of retransmissions. If this happens however one HARQ process could get a shorter round trip time (RTT) for its retransmissions at the expense of another process getting an increased RTT. This example shows how all HARQ processes can be fair to one another.

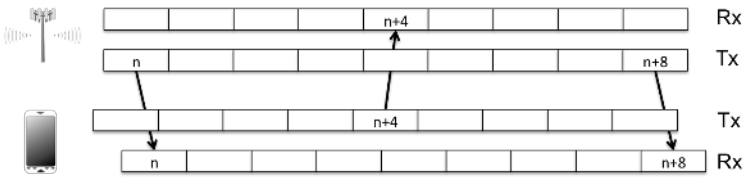


Figure 11: HARQ retransmissions for downlink traffic. This diagram was modified from an example in [17].

For TDD the total delay between sending data and getting the response is not fixed. There are multiple TDD frame setups that place downlink and uplink transmission opportunities at different times.

2.1.3.2 RLC

The RLC layer supports transmissions of user data using either the unacknowledged mode (UM) or acknowledged mode (AM). While AM enforces retransmissions both UM and AM support reordering of data to account for the MAC layer out of order PDU reception. In both modes a timer called t-Reordering is used. This timer is started when PDUs arrive at the RLC layer out of order. If the MAC layer can deliver the missing PDUs before t-Reordering expires then the timer is stopped and the received PDUs can be converted into RLC SDUs and delivered to the PDCP layer. If however t-Reordering expires then any missing PDUs are considered lost.

The value of t-Reordering has an impact on LTE performance. If t-Reordering is too long and the RLC layer will delay in delivering data it has or delay in indicating data needs to be retransmitted. If t-Reordering is too short then the RLC layer may give up on data or identify data for retransmissions that the MAC layer may still deliver.

If RLC AM is used then lost data will be requested for retransmission. AM requests retransmissions by sending a STATUS control message that includes an ACK sequence number and a sequence number for every missing PDU as a NACK. Only sequence numbers that are less than the ACK sequence number can be NACKed. The state variable VR(MS) holds the SN that is sent as an ACK so only missing PDUs from VR(R) (lower bound of receive

window) to VR(MS) can be NACKed. The variable VR(H) is the SN after the largest SN received. VR(H) can be greater than VR(MS).

As new PDUs arrive in sequence VR(R), VR(MS) and VR(MR) are all updated normally. Once a PDU arrives out of order VR(X) is set to the SN of the PDU that just arrived and the timer t-Reordering is started. VR(H) is also updated as this is the new highest seen SN. While t-Reordering is running new PDUs that arrive cause different status variable updates. As long as the PDU with SN VR(X) does not arrive then VR(R), VR(MR) and VR(MS) will not update, though VR(H) could. If the missing PDU arrives before the timer expires then VR(R), VR(MR) and VR(MS) are all updated and the timer is stopped. If however t-Reordering expires then VR(MS) is updated to the first SN for the PDU after VR(H) for which not all bytes have been received. This allows for the missing PDU to now b included as a NACK in a STATUS message. Also when the timer expires if VR(H) is greater than VR(MS) then other PDUs are missing. If so then t-Reordering are started again. At this point during the second run of t-Reordering the original missing PDU may arrive. If so then VR(R) and VR(MR) will be able to update again as it was this original missing PDU that stopped the lower bound of the receiving window.

For every PDU in AM there is a retransmission count. When a PDU is considered for retransmission its count is incremented. This makes it up to the transmitter when a PDU should be abandoned. In AM there is a constant called maxRetxThreshold that identifies the maximum number of retransmissions for a PDU. However the LTE specification does not say that a PDU is dropped once its maximum retransmission count is reached. Instead a message is sent up to the control RRC layer. In the RRC layer the notification of a PDU reaching its maximum retransmission threshold is considered a radio link failure [5]. When the UE has detected a radio link failure its RRC layer attempts to reconnect to the RRC layer at the eNodeB. If a connection is reestablished then the UE releases its RLC and PDCP entities and existing MAC configuration. This will also cause any existing data received at the RLC layer but not yet delivered to the higher layers to be lost. This includes all RLC

layer data for any other existing connections as well.

The transmitting side of the RLC connection can request a STATUS message through a poll request. The poll request comes in the form of one bit in the header of a data PDU that is sent. The transmitter keeps track of two constants to determine when a poll is requested. These are pollPDU and pollByte. These constants indicate how many PDUs or bytes to send before requesting a poll for status. A poll is requested whenever one of these constants is exceeded.

The STATUS message from the receiving RLC AM entity contains the ACKs and NACKs for the transmitter. The receiving side makes a STATUS message based on a few actions. First if the timer t-Reordering expires then the receiver will send a STATUS message. Also the transmitter can request that a STATUS message be sent in the RLC header of transmitted data. The transmitter keeps track of two state variables, pollPDU and pollByte. These are maximum the number of PDUs and maximum number of bytes sent between the transmitter asking for STATUS messages. Once one of these two constants are reached the transmitter asks for a new STATUS message. At the same time the transmitter will start the timer t-PollRetransmit. If the STATUS message does not arrive before t-PollRetransmit expires then the transmitter assumes its poll for STATUS was lost and it sends another. Not only will the transmitter resend its poll request when the timer expires but it also checks the status of its buffers. If there is no new data or retransmitted data to send but there is data pending ACKs then the LTE specification says that the transmitter can either put the last sent data (VT(S) - 1) into the retransmission buffer, or put all unacknowledged PDUs into the retransmission buffer.

Another AM timer on the receiver side is is t-StatusProhibit. As long as this timer is running the RLC layer cannot send any status messages with ACKs or NACKs. This setting is important because it can delay the RLC layer asking for retransmission and therefore delay delivering PDUs that have already been received.

2.2 ns-3

ns-3[1] is an open source discrete-event network simulator. ns-3 comes with modules for simulating network components, protocols and physical transmission methods. The goal of ns-3 is to improve the core architecture, software integration and educational components of ns-2[2]. Funding for ns-3 initially came from the National Science Foundation[3] and the Planéte[4] group at INRIA Sophia Antipolis[5]. Additional educational and government agencies have provided founding for ns-3 as well. While ns-2 is still widely used in research the number of ns-3 users has grown. Between 2009 and 2012 ns-3 was downloaded 50,792 times. As of the 3.16 release of ns-3 there were 113 listed authors and 25 maintainers of the project[6].

One of the modules built into the normal ns-3 distribution is an LTE/EPC Network Simulator[11]. This module, called LENA, is developed by the Centre Tecnológic de Telecomunicacions de Catalunya[7]. Development of the LTE module for ns-3 is ongoing.

The current version of the LTE module allows the creation of simulation networks involving Internet host connected to cellular phone towers through the EPC that then communicate over wireless with UEs. In the LTE module users can specify radio bearers for communication, scheduling algorithms, use of RLC transmission modes, location of towers and other settings.

Since the LTE module is integrated into ns-3 it can also take advantage of other existing ns-3 modules. This includes use of normal Internet hosts for UEs to communicate with, adding buildings to cause obstructions to the propagation of wireless signals, mobility models to factor movement of UEs, wireless propagation loss models and other features.

[1] https://www.nsnam.org/
[2] http://nsnam.isi.edu/nsnam/index.php
[3] http://www.nsf.gov/
[4] http://www-sop.inria.fr/planete/index.html
[5] http://www.inria.fr/centre/sophia/
[6] https://www.nsnam.org/overview/statistics/
[7] http://networks.cttc.es/mobile-networks/software-tools/lena/

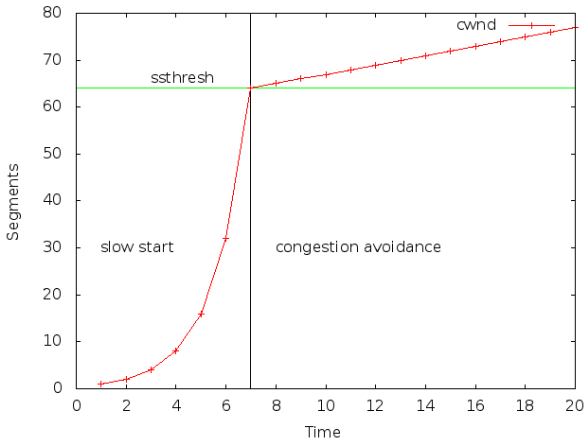


Figure 12: Simple example of congestion window growth in TCP.

2.2.1 Transmission Control Protocol (TCP)

transmission control protocol (TCP)[18] provides a reliable, ordered delivery of bytes between computers connected on the Internet. TCP senders add sequence numbers to packets and TCP receivers acknowledge packet receptions. Unacknowledged TCP packets are retransmitted based on information about successfully received sequence numbers. The various TCP congestion control algorithms proposed over the last 25 years roughly aim to maximize throughput while avoiding congestion at Internet routers.

While many TCP congestion control algorithms have been proposed, they all have several common features that are defined in RFC 2581[19]. The TCP sender keeps a dynamic congestion control window (cwnd) that limits the number of outstanding transmitted packets before an acknowledgment (ACK) needs to arrive from the receiver. **Figure 12** shows an example of TCP cwnd growth. The TCP connection begins in *slow start* mode with cwnd = 1 and doubles the size of cwnd every packetround trip time (RTT) (i.e., the time from when a packet is sent until an ACK is received back at the source). TCP stays in slow start mode until cwnd reaches the slow start threshold (ssthresh), indicated by the green line in **Figure 12**. Upon reaching ssthresh, TCP switches to the congestion avoidance phase, where cwnd is slowed to linear growth to avoid network congestion. The exact management of the cwnd growth depends on the specific congestion control algorithm used.

As of version 3.19, ns-3 natively supports the congestion control algorithms are Reno, NewReno, Tahoe, Westwood and Westwood+. There are also software modules for ns-3 that can provide access to the operating systems TCP stack but this is only supported on some systems and does not provide the fine grained logging of the native ns-3 code.

2.2.2 User Datagram Protocol (UDP)

user datagram protocol (UDP)[34] provides delivery of bytes between computers connected on the Internet with a minimal protocol overhead. This protocol only defines the source port, destination port, length of the data and a checksum along with the message. There is no reliable delivery of data with UDP. If a packet is lost then it cannot be recovered. UDP is meant for applications that do not require any reliability in a message reaching its destination. The low overhead of UDP makes it ideal for real time applications where the delay of waiting for the retransmission of data is unacceptable.

2.3 Applications

2.3.1 VoIP

Voice over IP (VoIP) and conversational video allow users to communicate with each other over the Internet in real time. Providing a good QoS for these types of applications is measured by the quality of the communication the users have. Does the user experience lost parts of the conversation? Does it take too long to receive data that the users are left waiting for responses? Is the audio and video clear and understandable? This section briefly introduces some basic components to VoIP and conversational video systems and discusses what can be considered a good QoS.

While there are many protocols and technologies for manipulating audio and video data they all follow a few fundamental steps. When a source has audio or video data to send it first encodes it into binary data for transmission. The type of encoding used could employ

techniques to recover lost data or minimize the size. The sender then packetizes the data and sends it. The receiver normally keeps a buffer to hold incoming data. The buffer is used to provide a steady playback as the arrival of data is subject to network delays. The data is then decoded and presented to the user. In general the sources for delay are encoding/decoding, packetization, network latency and buffering [28] [10].

The choice of encoder impacts how much data is generated and the rate of transmission needed for the conversation. While there are many types of encoders one common example is G.711 [36]. This coded creates 160 bytes of application data. This is encapsulated in a real time protocol (RTP) packet with a 12 byte header to provide some timing information on how to play out the data. This is placed in a UDP packet (8 byte header) and then in an IP frame (20 byte header). This codec transmit data every 20 ms creating a data rate of 64Kb/s.

For users on either end of a connection to maintain a conversation either through VoIP or video they need to receive data quickly with a minimum of loss. Unfortunately packet loss and transmission delays are at odds with one another. TCP provides reliable delivery through retransmissions, which adds delay. UDP adds minimal overhead to data, which improves delay but lost data cannot be recovered. Holding live conversations are more sensitive to delay than packet loss [15]. The International Telecommunication Union (ITU) recommends that one way delay should be less than 400 ms at most while a limit of 150 ms would allow most applications to run without any users realizing the delay[39].

Both the delay experienced in receiving VoIP packets and packet loss impact the quality of the conversation. One way to measure how delay and packet loss both impact perceived user quality there is the E-model [14]. The E-model has an R-Factor *(R)* that indicates the overall quality of the conversions based on the impact of delay *(i(d))* and loss *(i(l))* and is shown in **Equation 1**.

$$R = 94 - i(d) - i(l) \tag{1}$$

Table 3: Listening Quality Scale with MOS [37]

Speech Quality	MOS
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

The impact from delay is calculated from **Equation 2**. Here the one way delay *(d)* is used in two different equations depending on if the delay is smaller or larger than 177.3 ms.

$$i(d) = \begin{cases} 0.024 \times d & d \leq 177.3 \\ 0.024 \times d + 0.11 \times (d - 177.3) & \text{d } > 177.3 \end{cases} \tag{2}$$

The impact from packet loss, *(i(l))* is calculated using **Equation 3**.

$$i(l) = 30 \times log(1 + 15 \times l) \tag{3}$$

The result of **Equation 1** produces a number from 0 to 94. 94 is the best quality while 0 is the worst. This value can then be used to calculate a G-Model mean opinion score (MOS) [12]. The MOS was originally created as a metric for getting a users opinion of voice call quality. The ITU outlines how to map the R value to the MOS in [38]. R values less than 0 make a MOS of 1. Values between 0 and 100 are set using **Equation 4**. Any R value greater than 100 results in a MOS of 5. **Equation 1** can only create an R value as high as 94 but this is because this particular equation for the E-Model R factor is slightly different than others. The original MOS ranges from 1 - 5 as seen in **Table 3**. **Equation 4** will only produce a maximum MOS of 4.5.

$$E(R) = 1.0 + 0.035 \times R + 0.000007 \times R \times (R - 60) \times (100 - R) \tag{4}$$

2.3.2 FTP

file transfer protocol (FTP) is a protocol for transferring a file from one computer to another over a network. FTP creates a connection from one host to another and sends the contents of a file using TCP to ensure all of the entire file arrives at the receiver without error. While FTP is not concerned with delay like a VoIP or a video application is it is concerned with the overall throughput available to it. TCP will transfer data as fast as possible on the network so the current capacity of a network determines how much TCP can send and therefore how long it will take to transfer the entire file. FTP is a simple example for a TCP application to use in comparison with the VoIP and video applications. The TCP throughput is used to measure application performance.

2.3.3 Video

A video is just a series of pictures presented at a particular frame rate (there is also the audio component but the largest part of the data is the image). When the pictures are presented at a particular frame rate a viewer perceives motion by the changes in the picture. Video compression techniques take advantage of the fact that not all of the information in every picture is required. While there are multiple video encoding formats the examples used in this work use MPEG [22]. MPEG relies on temporal and spatial redundancy reduction. The idea is that across time and space (part of the image) there is content that is duplicated across frames that could be removed. In MPEG there are three types of frames, I, P and B. The I frame has the least amount of compression and provides a basic image. A P frame references a previously seen I frame and denotes what content in the image has changed for the future. A B frame also indicates portions of an image that change but it refers to previous and upcoming frames. MPEG then encodes a video into a series of I, P and B frames to reduce the total amount of data needed for the video. While the P and B frames denote changes there are still multiple I frames with a larger copy of the image. This is so video playback can provide random access [22]. A similar idea of removing redundant parts

of images that either cannot be detected or can be recreated by the player is a part of many video compression schemes. The frame rate at the receiver is used to measure application performance.

3 Related Work

As cellular networks and mobile devices have become more common in society, research into these areas has been ongoing. This section covers research in mobile applications and aspects of LTE. First, research into mobile applications shows that communication focus is prominent for users. Research into LTE is then considered. This includes measurements of real application data in LTE networks, some aspects of wireless retransmissions and a specific look at VoIP in LTE.

3.1 Applications in Cellular Networks

Xu et al. [41] examines the types of applications used by phones across the US by a tier-1 cellular network provider. The data set for this study was collected from all links within the tier-1 network from the radio access (cell phone towers) to the Internet. This data was collected over one week in August 2010. The authors found that 20% of mobile applications dealt with content local to the users such as news and radio stations for the area around the user's location. They also found that certain applications were likely to be used together (one after another) like multiple news applications. Many applications had identifiable usage patterns. News applications are used more in the morning for instance. Other applications like games are more likely when the user is traveling, e.g., while waiting in an airport. One important finding related to this work is that the top data consumer application was a personalized Internet radio, which was responsible for more than 3 TB of data over the week. This shows that streaming audio is a popular application for mobile devices.

Böhmer et al. [13] studied the applications users ran on their mobile phones and how they used them. The authors created an Android application that collected data on when applications were installed, opened, closed, updated and installed. The application sent this data back to the authors to use for their study. To encourage users to participate the application would also make recommendations on other applications the user may be

interested in. The authors collected data for 4,125 users from August 2010 to January 2011. They found that, on average, users spend 59.23 minutes per day on their mobile devices. The authors found that mobile application were also most likely used for communication through phone calls, emails, texts and other communication applications. This research shows that communication type applications that are delay sensitive are used regularly.

3.2 LTE Measurements

Huang et al. [25] examined data collected from an LTE network in a US city in October 2012 amounting to 2.9 TB of data. The authors characterized the network protocols, network characteristics and types of applications used in the dataset. They found that TCP made up 95.3% of traffic flows and 97.2% of bytes. The majority of the remaining flows and bytes used UDP. Downlink data sent from the tower to the phone took up the majority of the data in the network. Of the top 5% of flows (by payload size) 74.4% were related to video or audio applications. One result of this research is that the RTT from the LTE network to the UE was shorter than the RTT of the LTE network to external Web servers. The authors reported the normalized RTT for connection going from the UE to a monitoring node inside the LTE network and then connections from the monitoring node to the external server handling the application request. For about 40% of all data collected, the RTT of the connection from the UE to the monitoring node was larger than from the monitoring node to the external server. For the rest of the data collected the RTT from the UE to the monitor was smaller than from the monitor to the external server. The authors also found that 38.1% of TCP flows did not have any TCP layer retransmissions. While Huang et al [25] collected a large volume of real data, they did not have access to data from the LTE network layers. We do not know what RLC settings the networks used or how many PRBs were available.

Jiang et a.l [27] examined TCP performance in 3G and 4G networks including LTE. Their measurements show the existence of large buffers in cellular networks leading to buffer bloat problems. The authors found that at least for the Android operating system (which

is open source) the maximum window size for TCP was a static setting based on the type of network used. The reason for this is to limit the number of bytes in flight to reduce the increased RTT due to having large buffers. The idea is that the more data gets pushed through the network the more data has to sit in buffers meaning the time to get the data increases. This issue was found in both a AT&T HSPA+ 3G network and a Verizon LTE 4G network. The LTE network had higher throughput and lower RTTs compared to the 3G network but both have problems due to the static maximum TCP window size. The authors proposed an algorithm that would dynamically adjust the maximum window size to balance throughput and RTT concerns. This work shows how throughput and latency issues are still a concern in existing LTE networks. While this work looked at making adjustments to TCP, there is no information on the configuration of the underlying LTE network that handled the traffic.

3.3 HARQ

Kawser et al. [29] looked at saving radio resources by limiting the maximum number of HARQ retransmissions. Their reasoning was that for nodes experiencing poor radio conditions having three retransmissions would not provide a successful retransmission or soft combining to recover the lost data. The authors used an LTE link layer simulator to examine the difference in the BLER of the wireless transmission with bad SINR. Their results show only a small performance improvement for the full three retransmissions. The authors instead suggest that for poor radio conditions only one or two retransmissions should be allowed in HARQ. This work only looks at performance related to the MAC layer HARQ process. It does not include any retransmissions from the RLC layer, nor does it show how these retransmissions and delays impact high level applications.

Ikuno et al. [26] examined how HARQ layer retransmissions create a SINR shift in the BLER curve. In LTE, a BLER of 10% or less is the target used to set the modulation and encoding scheme. The received SINR at the UE is used to create BLER curves showing

what the expected BLER is for different SINRs. As the SINR worsens the BLER increases. Changing the modulation and encoding scheme moves the BLER curve, for example, going from 64 to 16 quadrature amplitude modulation (QAM) would move the the curve so the same SINR would have a lower BLER. The authors developed a model that takes into account the number of HARQ retransmissions and the current modulation scheme to predict the SINR gain in decibels. When compared to simulations run using a Matlab based downlink physical layer simulator [32], this work showed an error rate in predicting the SINR improvement of less than 1% for QPSK and 16 QAM but a 11% error rate for 64 QAM. This work does not look at any of the possible applications of this research but with the ability to better predict the improvements from HARQ retransmissions, a system could make better decisions on how to transfer data

3.4 RLC

Makidis [30] implemented and evaluated RLC AM for his Master's thesis. The author implemented their RLC AM in ns-2 and tested with TCP based application like Web browsing with hypertex transfer protocol (HTTP) and FTP traffic. Simulations showed that using RLC AM works well with applications that experience contention. Comparing RLC AM with two other selective repeat protocols showed that the adaptive selective repeat is preferred when maximizing TCP throughput. The throughput for Web browsing traffic in RLC AM was significantly lower than both the selective repeat variants. RLC AM performed better when dealing with large file transfers with FTP.

While Makdis implemented the RLC AM protocol in ns-2, he did not implement the other LTE layers. This is problematic as the loss rate applied to these simulations do not take the MAC layer HARQ retransmissions into account. The author did not implement the RLC UM or run any tests with this option. This is possibly because the author was trying to show the impact of retransmissions, but the MAC layer would have provided retransmissions for comparison. Since the author stated that the RLC parameters he used

were shown experimentally to have good results, there was no adjusting of these parameters to show what changes would occur to the end results.

3.5 VoIP in LTE

Asheralieva et al. [9] simulated running VoIP applications over LTE. These authors focused on two packet scheduling mechanisms and if HARQ was used or not. The authors' found that HARQ can improve QoS for VoIP services. These authors simulation took into account scheduling along with the physical and MAC layers. They did not include the RLC layer that would handle reordering of packets during packet loss. They also did not consider at if the RLC layer was enforcing retransmission of lost packets.

Masum et al. [31] examined the end-to-end delay of VoIP applications in LTE networks. The authors used the OPNET simulator and created example networks to examine a baseline VoIP network, a congested VoIP network and a VoIP congested network with an FTP traffic mix. In these scenarios the authors modified speed of UEs, packet loss and the available bandwidth (changing the number of available PRBs). They found that when there is no movement the end-to-end delay is slightly higher for networks congested with only VoIP traffic, however the authors do not examine why their simulations produced these results. In the other scenarios the end-to-end delay was better when nodes were mobile. In congested VoIP networks the speed of the mobile UEs had little impact on packet loss. For the network with mixed FTP traffic, stationary nodes saw little packet loss while mobile nodes saw more.

While the work of Masum et al shows the delay perceived in VoIP over LTE, they did not adjust any of the settings in the RLC layer. They also did not report any information about performance in the MAC or RLC layers. The highest average packet loss across their tests was less than 4%.

4 Approach

4.1 Problem Statement

The LTE specification allows for retransmissions in both the MAC and RLC layer. The MAC layer retransmissions recover most lost packets with low latency while the RLC layer adds retransmissions to keep wireless error rates below 1% to improve TCP performance [17]. The use of QoS classes and radio bearers allow for multiple RLC entities and potentially multiple configurations for the use of AM/UM and other configurations. However, network providers do not publish information on what their QoS setups are and data flows can only be classified by their protocol, IP addresses, ports used and the direction of traffic (uplink/downlink). As it stands the application developer does not know if the application is connected through the RLC AM for higher reliability or through UM for lower latency. There is also no information for the developer on what settings are being used inside the RLC layer for timeouts or status messages.

As an example assume a developer knows their application will need to send sets of messages in bursts. It may be best if this string of messages is not interrupted by RLC AM STATUS messages. If the LTE network posted its QoS classes and the RLC layer settings the developer could potentially setup the protocol and port numbers their applications use to ensure the traffic gets sent to the QoS class with the best RLC settings for their application.

4.1.0.1 Objectives

This work examines how adjustments to the configurable parameters in the RLC layer impacts application level performance for applications. Specifically, this work looks at applications with different QoS concerns and characteristics. VoIP applications need to keep both a low delay and low packet loss rate while sending a small amount of data at a relatively low rate. FTP applications are mostly concerned with throughput so files can be moved

as quickly as possible and without any loss. Video applications send large amounts of data that require a low amount of packet loss. However the format of the video adds additional concerns. A video for a live conversation like a video conference will be concerned with the delay on arriving video frames and will allow some packet loss. A video of a past recorded event like a movie on the other hand can use a playout buffer to compensate for more delay while expecting few if any packets to be lost. The following outlines how we will test these applications within LTE.

4.1.0.2 VoIP with varied RLC settings

Test a VoIP application with different settings for t-Reordering and t-StatusProhibit timers.

Methods Configure ns-3 simulations to use a constant bit rate application that uses UDP. The application is designed to send traffic at a constant rate of 64 Kb/s to align with the G.711 encoding standard [36]. Simulation runs use AM and UM with different settings for the t-Reordering and t-StatusProhibit timer with some wireless loss. The delay of arriving UDP packets and percent of packet lost is used to calculate the MOS to determine the settings for these timers that produce the best results.

4.1.0.3 VoIP with AM vs UM with varied loss

Test a VoIP application with constant settings in RLC AM and UM with different amounts of loss.

Methods Using the same application setup for testing the RLC layer settings this test compares using the preferred settings found previously with a varied amount of wireless loss to find under what conditions AM or UM perform better. Once again the MOS is used to determine the overall quality of the conversation.

4.1.0.4 FTP with varied RLC settings

Test the performance of transferring a file from a server to a UE using varied RLC settings.

Methods Configure ns-3 simulations to use an FTP -like application where TCP New Reno is used to send as much data as possible from a server to the UE. The settings for the t-Reordering and t-StatusProhibit are varied with AM and UM to find settings that produce the highest throughput for the application.

4.1.0.5 FTP with AM vs UM with varied loss

Test the performance of transferring a file from a server to a UE using both AM and UM with different amounts of loss.

Methods Using the preferred settings for t-Reordering and t-StatusProhibit run the FTP simulation for both AM and UM with a varied amount of wireless loss to find under what condition AM or UM perform better based on the application throughput.

4.1.0.6 MPEG video with varied RLC settings

Test the performance of MPEG video transferring from a server to the UE using varied RLC settings.

Methods Configure ns-3 simulations to use an MPEG UDP application. The application is designed to send UDP packets based on a trace file from a real MPEG-4 encoding of a video. These simulations adjust the settings for t-Reordering and t-StatusProhibit timers long with using AM and UM to find reasonable settings for these timers.

4.1.0.7 MPEG video with AM vs UM

Test the performance of MPEG videos when using AM and UM and varying loss.

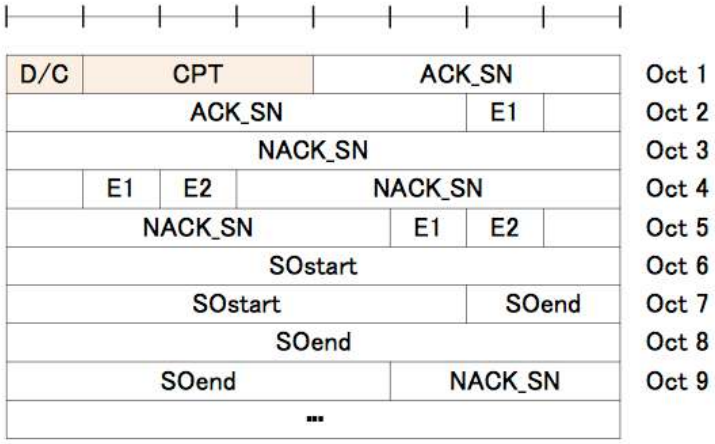


Figure 13: Status PDU for RLC AM [2]

Methods The MPEG video simulation is run using the preferred settings found for t-Reordering and t-StatusProhibit found earlier. This test tries the simulation using AM and UM with a varied amount of wireless loss to find under what conditions AM or UM may be preferred.

4.2 Simulator Additions

As of version 3.16 of ns-3 the LTE module did not support using NACKs in RLC AM. An existing discussion on this exact issue was found in the ns-3 user forum[8]. We joined the discussion and over the course of several months added NACK support to the simulator.

The initial contribution was a simple change to the creation of the RLC status PDU so it could serialize and de-serialize NACK SNs. The RLC status PDU can be seen in **Figure 13**. ACK SN represents the highest SN that can be ACKed followed by SNs for each of the NACKed PDUs. Originally the LTE simulator did not support the use of the NACK SN. Instead the simulator always sent back one NACK in the status message with a value of 1024. Our contribution to the code allowed a set of SNs to be added to the status PDU that would be listed as NACKed.

The code for adding the NACK sequence numbers was simple. When the RLC AM code

[8]https://groups.google.com/forum/#!topic/ns-3-users/CEfmMX3IRBw

constructed a STATUS PDU it would iterate over its receive window to find what PDUs were missing and add them to a list for the STATUS PDU. Adding these numbers to the PDU involved creating some serialization code to embed the sequence numbers and corresponding de-serialization code to extract the numbers later. In **Figure 13**, the ACK_SN field holds the highest sequence number that can be ACKed at the time of the STATUS. This value is stored in the RLC AM state variable VR(MS). After ACK_SN there is the E1 bit. This is the extension bit 1 that and it indicates if a NACK_SN, E1 and E2 bit follow. A value of 0 indicates no NACK_SN while a value of 1 indicates that at least one NACK_SN follows. Our code sets the first E1 bit to 1 and adds the first sequence number for the first missing PDU. If the list of sequence numbers to NACK still had elements, the next E1 bit would be set to 1 and the next NACK_SN was listed until all NACKs were added to the STATUS. The E2 or extension bit 2 field indicates if the previous NACK_SN was a partial NACK. For a partial NACK the SOstart and SOend field indicate what part of the received PDU was in error. However, since the rest of the RLC AM code in the simulator did not support sending partial SDUs we did not add this to the implementation. As a result we always set the E2 field to 0 indicating that this was a NACK for a full PDU.

The initial code given to the LTE module developers had some errors. After the developers reported back problems, we iterated over the code to fix the issues in adding and removing the NACK sequence numbers to and from the STATUS PDUs. The LTE module developers also added checks to ensure the STATUS PDU would not add more NACK sequence numbers to the PDU than could be sent given a specific transport block size.

After the code to add a list of NACK sequence numbers was working, the next set of additions to the simulator was based on determining which sequence numbers needed to be ACKed or NACKed for a STATUS PDU. As mentioned in Section 2.1.1.3 the receiving side of the RLC AM layer has five state variables. VR(R), VR(MR), VR(X), VR(MS) and VR(H). These variables hold the sequence numbers for the lower bound of the window, upper bound of the window, sequence number after the PDU triggering reordering, ACK sequence

number and the largest sequence number seen so far, respectively. The upper bound of the receive window (VR(MR)) is not important in determining what sequence numbers are used in the STATUS PDU so we do not consider this.

VR(R) is the lower bound of the receiving window and VR(H) is the highest sequence number seen so far. For our first pass at finding the NACK sequence numbers we added a loop to examine the receive buffer between VR(R) inclusive and VR(H) exclusive. If the buffer did not have a PDU for the given sequence number in this range, it was marked as a NACK. However, this was not correct. In our original implementation we did not know that VR(MS) was the highest possible ACK value. This was reported back by the LTE module developers. Our code changed so that the reception buffer was checked from VR(R) inclusive to VR(MS) exclusive. However, the issue on how VR(MS) and what should be NACKed became an issue of discussion with the LTE module developers.

Initially the developers thought that the only sequence numbers that could be NACKed were those higher than the sequence number ACKed. The reason for this was a misunderstanding in the RLC specification [2]. The issue was over how the variable VR(MS) was incremented. Specifically section 5.1.3.2.3 of [2] states.

"When a RLC data PDU with SN = x is placed in the reception buffer, the receiving side of an AM RLC entity shall: if all byte segments of the AMD PDU with SN = VR(MS) are received: update VR(MS) to the SN of the first AMD PDU with SN >current VR(MS) for which not all byte segments have been received;" [2]

This passage seems to indicate that VR(MS) grows up to the first missing PDU and therefore only sequence numbers larger than the ACK could be used for NACKs. The reason for this disconnect is due in part to how the specification document is written. Section 5.1.3.2.3 does give the impression that VR(MS) cannot allow sequence numbers smaller than ACK to be used for a NACK. However section 5.1.3.2.4 of [2] indicates that when the timer t-reordering expires VR(MS) is incremented to the SN of the first PDU with a sequence number >= VR(X) for which not all byte segments have been received. This is a consistent

issue that we had with the developers. A section of the specification would indicate a variable behaved a particular way and then another section sometimes pages away would add a caveat. This made it very easy for all of us to become confused. After this discussion the developers updated the code to account for incrementing VR(MS) on t-Reordering expiring.

The impact of updating VR(MS) on the expiration of t-Reordering is to allow for the lower MAC layer to recover transport blocks before considering a PDU lost. This way the RLC layer will not request retransmissions until after the MAC layer has a chance to recover the data. Hence, the setting of the t-Reordering timer can have an impact on the performance of LTE by either waiting too long and adding delays or by forcing unneeded retransmissions.

For the transmitting RLC AM entity we added code so that the NACKed sequence numbers were used to determine which PDU needed to be moved to the retransmission buffer and how to update the transmitting side state variables VT(A), VT(MS) and VT(S). As explained in Section 2.1.1.3 these are the lower bound of the transmitting window, upper bound of the transmitting window and the sequence number of the next new PDU to transmit. I had some initial confusion as how the lower bound of the transmission window (VT(A)) was set but conversations with the module developers cleared this up. In the end I added code so that upon receiving a STATUS PDU, the transmitter would update VT(A) to be either equal to the ACK sequence number or the first NACK sequence number, which ever came first.

During testing of using NACKs described in Section 5 the developers and I realized there was a problem where the transmission window could stall. The issue was that the transmitter only knows what to transmit and retransmit based on the STATUS PDUs received from the receiver. During testing a scenario came up where the transmitter sent data along with a request for a STATUS PDU. However this PDU was lost. As the transmitter still had data to send it continued to make and send PDUs. As the new set of data arrived at the receiver the missing PDU is noticed and the timer t-Reordering is set. The variable VR(MS) is not incremented past the missing PDU until after t-Reordering expires so as more data comes in

the STATUS messages that go back just ACK the last received PDU before the one that is missing. The missing PDU is not NACKed until after t-Reordering expires. This all works as expected, however the transmitter finished sending new data before t-Reordering expired. Once t-Reordering did expire VR(MS) is updated and the receiving side can make a STATUS PDU with the corresponding NACK. However since the transmitter is done sending data, it does not send any requests for STATUS messages. Also since the transmitter did not get a NACK it does not move any data to the retransmission buffer. As a result the sender has nothing new to send and nothing to resend, just data that has not been ACKed. The receiver meanwhile is never asked for a STATUS and none are sent. This caused the transmission to stall. After examining this problem and reviewing the specification I found a note that when t-Reordering expires the receiver should send a STATUS message even if none were requested. This had been missed because of how it was written in as a note to a section in the specification. After reviewing this with the developers, they modified the code to send this STATUS message and all of the remaining tests passed.

With these changes the RLC AM in the LTE module is still not complete. As mentioned the STATUS PDU does not support indicating only parts of PDUs as being NACKed. Additionally the RLC layer in general can pack multiple SDUs into one PDU but does not currently support splitting an SDU to fit the remaining room in a PDU.

4.3 Simulator Settings

4.3.1 VoIP

To simulate a VoIP application in ns-3 we used a method similar to the one described by Ramroop in [35]. In this work Ramroop looked at both VoIP and MPEG video examples in ns-3. For VoIP the author chose the G.711 codec. This codec has an application output of approximately 64Kb/s. Each data packet has a payload of 160 bytes, which gets encapsulated by the RTP protocol (12 bytes), then UDP (8 bytes) and finally IP (20 bytes). Since ns-3

does not have a module for RTP the author added the 12 bytes of RTP header to the data packet for a size of 172 bytes. These are then encapsulated by UDP and IP as supported in ns-3. To simulate a VoIP conversation, Ramroop used an on/off ns-3 application that allows control either directly or through a random value (with a specified mean) when to turn sending on and off. For simplicity we decided to make our VoIP example a simple constant bit rate. This is not like a real VoIP conversation but since we are interested on the delay of packet arrival and loss, it is an acceptable trade off.

In ns-3 we created a constant bit rate (CBR) sending UDP application. We set the payload size in bytes to be 172 bytes (160 bytes of data and 12 of RTP header) and set an interval on sending packets of 0.02 seconds to generate 50 packets per second.

4.3.2 FTP

To simulate FTP, ns-3 has a bulk send application that transfers data as fast as possible using TCP. The application sets the maximum number of bytes to transfer, allowing the simulation to send files of varied sizes. The TCP stack in ns-3 supports only a few of the early TCP congestion control algorithms. These are Reno, NewReno, Tahoe, Westwood and Westwood+. ns-3 does support accessing the TCP stack of the operating system through extensions like the network simulation cradle (NSC) and direct code execution (DCE). While NSC and DCE allows the use of modern TCP implementations, these components are not part of the default ns-3 and do not work on all operating systems. Attempts to use NSC with the LTE module caused the TCP simulations to end before sending all data. The reason for this error is unknown and there was not enough time to thoroughly examine the cause. Similarly, there was not enough time to examine how DCE could be used. As a result these simulations use the ns-3 default of NewReno. This means that the TCP implementation will take longer to fill the congestion window, opposed to newer algorithms.

Table 4: MPEG frame trace.

Frame No.	FrameType	Time[ms]	Length[byte]
1	I	0	7559
2	P	120	3442
3	B	40	1865
4	B	80	2021
5	P	240	2282
6	B	160	1532
7	B	200	1853
8	P	360	2216
9	B	280	1515
10	B	320	1664

4.3.3 Video

For video we use a sample application that comes with ns-3. This applications takes as its input a trace file that defines an MPEG 4 video. The trace file lists four columns for a row number, frame type, time to send the packet and size of the packet in bytes as seen in **Table 4**. Example trace files for this application are available from the Telecommunications Network Group at the Technische Universität Berlin[9].

The video trace file used in these simulations is for a soccer match and has an average bit rate of about 1.05 Mb/s. The trace file application in ns-3 reads the trace file and sends UDP packets for the video frames at the specified time intervals. The application does not support any notion of playback buffers or how the video will look to the receiver. Instead we examine packet loss and the difference in delay of the packets as they arrive at the UE. Since this application uses UDP, we set the maximum packet size to 1,460 bytes, leaving more than enough room for the IP header. Any video frames larger than this are broken into multiple UDP packets.

[9]http://www-tkn.ee.tu-berlin.de/research/trace/trace.html

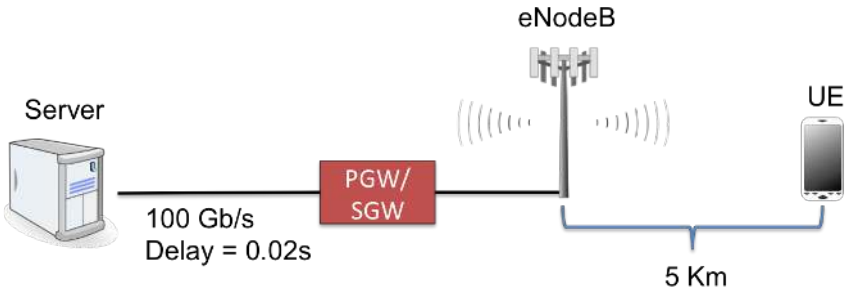


Figure 14: Network used during simulations.

4.3.4 Simulated Network

In the ns-3 simulations, we use the network configuration shown in **Figure 14**. Our UE connects to a server on the Internet to retrieve either the VoIP or video examples. The Internet is connected to the LTE network through the PGW. The PGW then connects to a SGW before connecting to an eNodeB. In the LTE module the PGW and SGW are considered to be one node that implements the operations of both devices. In the simulator we set our Internet link to have a bandwidth of 100 Gb/s, a delay of 20 ms and an maximum transmission unit (MTU) of 1500 bytes.

The connection point from the PGW to the eNodeB does not have settings on bandwidth or delay. The eNodeB has a height of 30 meters. The UE is placed 5 kilometers from the eNodeB and is positioned 1 meter from the ground. We chose the positioning of the UE to get a specific SINR for the received signal as discussed in **Section 4.3.5**.

4.3.5 Constant Settings

ns-3 and the LTE module provide many settings. To limit the scope of this work some of these settings are left constant throughout the tests. This section details these settings.

4.3.5.1 Mobility

For our simulations the UE nodes remain stationary throughout the simulation. Mobile wireless nodes often see variable network conditions due to multipath fading, moving between cells and entering bad signal locations. In this work we are not looking at situations where a wireless node moves between cells, we are instead using fading trace files to apply wireless loss.

4.3.5.2 Distance

The distance from the eNodeB and UE is the same for all of the simulations. In the ns-3 simulator the distance between the nodes impacts the attenuation on the wireless signal causing nodes further away to experience a worse SINR. Unless otherwise noted the distance between the eNodeB and the UE in these simulations is 5Km. The reason for this was to create a baseline SINR so the UE would report a CQI of 8. This CQI was chosen based on real measurements described in Appendix B.

4.3.5.3 Available PRBs

As mentioned in Section 2.1.1 an LTE network can support 6 different configurations in bandwidth ranges providing different number of available PRBs. For all of these simulations we set the network to have the the maximum frequency range of 20MHz and 100PRBs. This is the default for the simulator.

4.3.5.4 Basic Radio Environment

ns-3 and the LTE module provide multiple tools for creating the radio environment. This impacts the SINR the nodes experience. These settings include the power of the transmitters, background noise, multipath fading models buildings, building materials, interference sources and fading traces to create a realistic radio environment. In this work, we focus on how LTE handles loss but we do not look into specific origins of the loss. We use the fading trace file

to apply loss as this allows the fine granularity of selecting specific radio frequency ranges and time frames to apply a poor SINR. We do not examine having interference from any other wireless transmitters or from having obstructing buildings.

For propagation loss ns-3 has a few options. These simulations use the FriisPropagationLossModel since the only factor that impacts SINR is the distance between the eNodeB and the UE. In these simulations the FriisPropagationLossModel sets the base line SINR that the fading trace file manipulates.

The settings on transmission powers and noise remain constant throughout the simulations. For the baseline radio environment that the fading trace file manipulates, we created a setup where the SINR experienced by the UE causes it to send a CQI of 8 to the eNodeB. The reason for this was that 8 was the average CQI requested from a phone that we took measurement on for a week as described in Appendix B. Both the eNodeB and UE can have their transmission powers and noise figures set. The transmission power as its name suggests sets the amount of power used to transmit. The noise figure is an additional SINR setting caused by imperfections in the antenna for the device. For this work we keep the noise figure on both devices set to 0. To set the transmission power we first tried the default settings (30 dBm and 10 dBm for the eNodeB and eNodeB respectively). We then adjusted the position of the nodes and the transmission power so that both nodes could communicate with one another with the UE requesting a CQI of 8. Using the default settings for the transmission power, the UE only requested a CQI of 8 when it was so far away that its own transmissions could not be heard by the eNodeB. We therefore lowered the transmission power on the eNodeB and moved the UE closer (to 5Km) so that the UE would request a CQI of 8 and the eNodeB would be able to hear the transmissions from the UE. The final settings on the transmission powers was 20dBm for the eNodeB and 10dBm for the UE.

The LTE module also allows for setting the E-UTRA absolute radio frequency channel number (EARFCN) for both the downlink and the uplink [8]. The EARFCN identify the carrier frequency used in the uplink and downlink. These were left on their default settings of

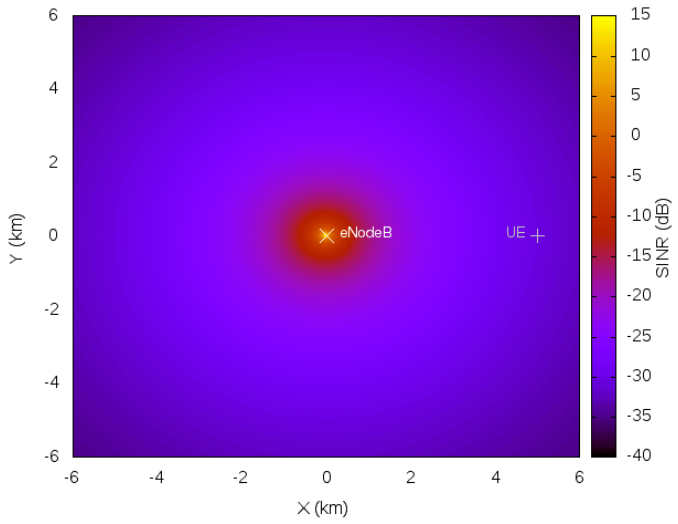


Figure 15: Default simulation radio map.

100 for downlink and 18,100 for the uplink. These settings create the basic radio environment seen in **Figure 15**. This figure shows the simulated area with the position of the eNodeB and the UE. The color in the figure shows the SINR at all locations in the simulation. Around the eNodeB the SINR is high since the tower is transmitting. The further away from the tower the lower the SINR.

4.3.5.5 Adaptive Modulation and Coding (AMC)

The LTE module provides two different adaptive modulation and coding (AMC) implementations that determine how SINR maps to CQI and MCS values. These simulations use the default MiErrorModel for determining the AMC. This model uses the received SINR at the UE to determine the BLER the UE would likely receive. The UE uses this information to pick a CQI value to reduce or eliminate all wireless loss.

4.3.5.6 Error Models

The LTE module allows for enabling or disabling the physical layer errors for data and control messages separately. Since this work is interested in retransmissions where the LTE

control messages work as intended, we set the data error model enabled and the control error model disabled.

4.3.5.7 Buffers

Both AM and UM use buffers to store data for transmission. However, since AM is supposed to provide reliable data transmission the ns-3 LTE module implementers set its buffers without a limit so there should be no dropped data due to buffer overflow. UM on the other hand does have a setting to set the maximum size of the buffer. Since we are interested in losses caused by the physical layer and not an overflowing buffer we set the UM buffer, limit to be 9,999,999 bytes to avoid any dropped data.

4.3.5.8 MAC Scheduler

The LTE module allows setting different MAC schedulers to determine how data is sent and received from all of the UEs. Since these simulations only have one UE, we use the default proportional fair MAC scheduler.

4.3.5.9 RLC MaxRetxThreshold

The RLC AM variable maxRetxThreshold determines how many RLC layer retransmissions are allowed. When a RLC PDU surpasses this retransmission limit, a radio link failure message is sent to the RRC layer as described in Section 2.1.3. The ns-3 LTE model does not support sending the radio link failure message to the RRC layer. As a result making any adjustments to this setting do not have an impact. Since the radio link failure causes the UE to reset its connection, this work assumes that if a RLC PDU passes its maximum retransmission threshold then the communication will have failed. The default in the ns-3 simulator for maxRetxThreshold is 5.

4.3.5.10 RLC PollBytes and PollPDU

The RLC AM variables PollBytes and PollPDU determine how many PDUs or bytes to send before requesting a STATUS message from the receiver. For this work we decided to focus on the timers t-Reordering and t-StatusProhibit so we leave these variables to their default in the simulator of 1 PDU and 50 bytes.

4.3.5.11 RLC t-PollRetransmit

The RLC AM timer t-PollRetransmit is used by the transmitter to determine how long after requesting a STATUS message it should wait before sending the poll request again. In this work we decided to focus on the timers t-Reordering and t-StatusProhibit. This variable is left with its default setting from the simulator of 100 ms.

4.3.6 Variable Settings

4.3.6.1 Wireless Loss

To adjust the wireless loss experienced by the UE we used a fading trace file. The fading trace file specifies a SINR value to apply to a specific PRB at a specific TTI. As a result we can set when and where data can be lost. Normally the fading trace file is generated through a Matlab script with a set of parameters to create a realistic radio environment[10]. For these simulations we are interested in examining how loss and retransmissions impact the higher level application. We could try generating realistic fading trace files but this means we lose control over the specific loss instances. Instead we created our own fading traces to control exactly when and where loss occurs.

For the UE to lose any data the SINR must be low enough that the signal is disrupted by the noise. LTE is designed to adjust the modulation and coding to keep the BLER to less than 10% [17]. This means that as the SINR gets lower the UE requests lower CQI values.

[10]https://www.nsnam.org/docs/models/html/lte-user.html

The eNodeB can then respond by choosing a different modulation and coding scheme that sends less data but with a better chance of it arriving undamaged. While the goal is for a BLER of 10% the ns-3 LTE module does not automatically add a random error to the transmission. The simulator uses the SINR combined with the modulation and coding to determine if a particular transport block would be lost. Since the UE requests a lower CQI when the SINR worsens, the simulator does not see any transport blocks lost until the SINR is worse than the eNodeB can compensate. This means that as long as the MCS that the eNodeB transmits at is above 0 there will be no errors.

When the MCS is 0 there might still be no errors. However once the MCS is at 0 the eNodeB can no longer compensate so as the SINR continues to worsen there is nothing that can be done to stop the transport blocks from being corrupted. In real life multiple interference sources may cause transport blocks to be lost before the UE can request a lower CQI but for these simulations we are only applying SINR for loss through the fading trace file.

The values in the fading trace file can lower the SINR at specific times and for specific PRBs. The UE responds to this by requesting lower CQI values to decrease the throughput and decrease the chances of lost data. Since we do not have any other sources of wireless loss in our simulations, we do not lose data until the SINR is low enough that the eNodeB cannot compensate. We found that applying a SINR of -100dB to a PRB was enough to cause it to fail on reception.

To create the fading trace files we use two methods. For a first round of tests a simple uniform probability distribution is used to provide initial insight into how loss impacts the LTE layers. The majority of the tests however use a Gilbert-Elliot [23] model. The Gilbert-Elliot model is a common method used to simulate packet loss. This model is depicted in **Figure 16**. There are two states to the model. One state is considered good and has low packet losses while the other state is considered bad with high packet losses. There are also probabilities that control when to stay in a state or transition to another. The exact

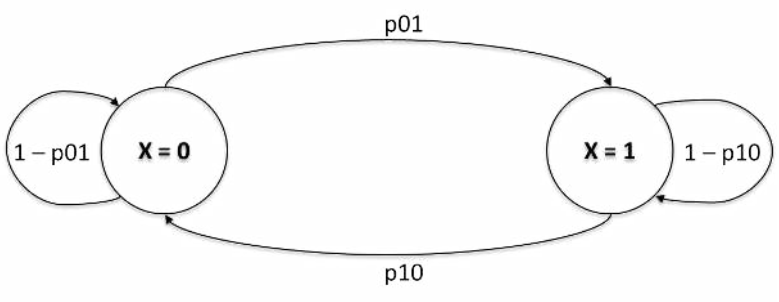


Figure 16: Gilbert-Elliot model. Figure based on [24].

probabilities used in this model is dependent on one's goals. We use a Gilbert implementation similar to the one used in [24], which also examined LTE. Here, the good state has a probability of 0 for dropping data while the bad state drops data with a probability of 100%. The exact amount of loss seen is controlled by when the model changes states. In [24] the authors specified a set of equations where knowing the total amount of data sent, the total amount of data to lose and the number and length of loss events could be used to calculate the transition probabilities.

In this work we look at loss in terms of transmission opportunities during the simulation on TTIs. There may or may not be any data sent during these TTIs. For the total number of TTIs to lose we chose the BLER goal of 10%.

Knowing how long the simulation runs gives us the total number of TTIs and the loss percent indicates the number of TTIs to lose. The only remaining variable is the number and length of loss bursts. In this case a burst of loss is any number of consecutive TTIs where any data sent is lost. The length of the burst events has an impact on LTE as they may or may not be long enough to take out the MAC layer HARQ retransmissions. During our simulations we vary the number and length of bursts to test different retransmission scenarios.

With the information on total TTIs, loss percent and burst losses to use we can use the equations in [24] to get the transition probabilities for the Gilbert-Elliot model. We define these variables as

- o_k - The number of burst losses of length k.
- a - The total number of TTIs.
- d - The total number of lost TTIs.

With these variables we calculate the probability of moving from the good to the bad state using **Equation 5** and the probability of staying in the bad state using **Equation 6**.

$$p_{01} = \frac{\sum_{k=1}^{\infty} o_k}{a} \tag{5}$$

$$1 - p_{10} = \frac{\sum_{k=1}^{\infty} (k-1) \times o_k}{d-1} \tag{6}$$

From these equations we get the probabilities of the state changes. However, while these equations take the number of gaps and gap lengths in as parameters the result from running the Gilbert-Elliot model will not match up. There is no guarantee that the state change probabilities will produce the same output as the input parameters. However our goal is to be able to get the correct overall loss rate. We examine the actual gap lengths versus the input parameters as we make fading trace files and ensure that the loss rate is correct.

For the initial tests to find the preferred settings of t-Reordering and t-StatusProhibit we use a loss rate of 10% to match the goal for BLER inside LTE. However once we find these preferred settings and fix them we create fading trace files with other error rates using both the Gilbert-Ellilot model bursty loss and comparing this to simple uniform loss rates.

4.3.6.2 RLC Settings

The RLC layer contains a few settings for both AM and UM. These settings impact how the RLC layer handles retransmissions and how long it waits before declaring data lost. The specific settings we test are:

Table 5: Suggested settings from [5] for some RLC layer timers.

Setting Name	Values
t-Reordering	0 ms, 5 ms, 10 ms, 15 ms, 20 ms, 25 ms, 30 ms, 35 ms, 40 ms, 45 ms, 50 ms, 55 ms, 60 ms, 65 ms, 70 ms, 75 ms 80 ms, 85 ms, 90 ms, 100 ms, 110 ms, 120 ms, 130 ms 140 ms, 150 ms, 160 ms, 170 ms, 180 ms, 190 ms, 200 ms
t-StatusProhibit	0 ms, 5 ms, 10 ms, 15 ms, 20 ms, 25 ms, 30 ms, 35 ms, 40 ms, 45 ms, 50 ms, 55 ms, 60 ms, 65 ms, 70 ms, 75 ms 80 ms, 85 ms, 90 ms, 95 ms 100 ms, 105 ms, 110 ms, 115 ms, 120 ms, 125 ms, 130 ms, 135 ms, 140 ms, 145 ms, 150 ms, 155 ms, 160 ms, 165 ms, 170 ms, 175 ms, 180 ms, 185 ms, 190 ms, 195 ms, 200 ms, 205 ms, 210 ms, 215 ms, 220 ms, 225 ms, 230 ms, 235 ms, 240 ms, 245 ms, 250 ms, 300 ms, 350 ms, 400 ms, 450 ms, 500 ms

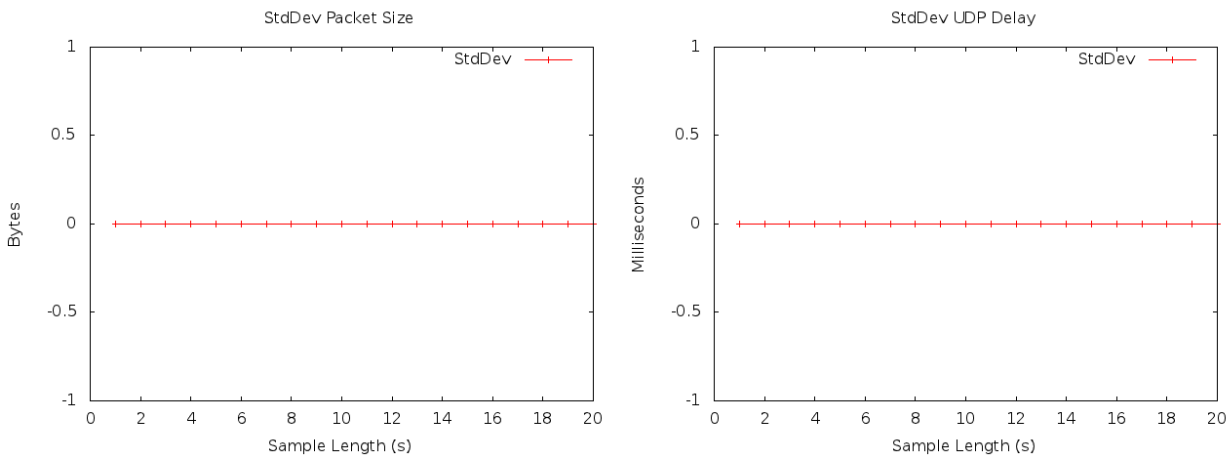
- t-Reordering - Adjustments to this timer change when the receiving side RLC entity (AM and UM) detect a lost PDU.
- t-StatusProhibit - Adjustments to this timer change the shortest time between the receiving side RLC entity sending status messages with ACKs and NACKs.

The specification for the RRC layer defines a set of suggested values for these parameters. These values are shown in **Table 5**.

4.3.7 Simulation Time

To determine the amount of time the simulation runs we look at the granularity of events and the types of applications used. In LTE, data can be transmitted every subframe (1 ms) so in a matter of seconds the simulation will produce data relating to thousands of possible transmissions and retransmissions. When examining retransmissions at the lower layer we can run experiments for a few seconds. When examining the performance at the application layer we need to run the simulation for longer.

The application should run long enough in the simulation to overcome any natural variation in its nature. For instance in TCP there is the slow start phase that starts the congestion window small. This causes an initial low bandwidth for the application using TCP (in this



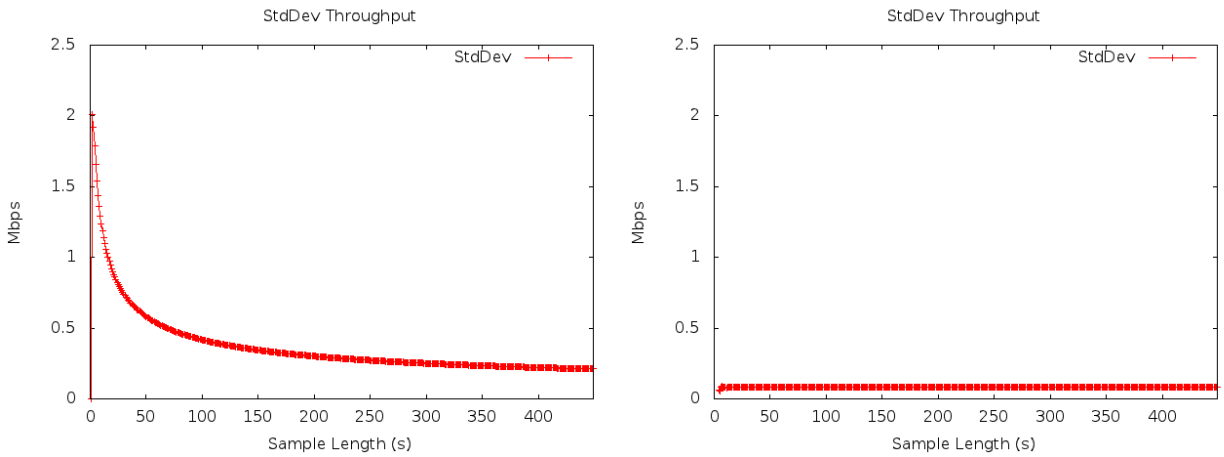
(a) UDP frame sizes.

(b) UDP delay.

Figure 17: Standard deviation for the VoIP packets.

case FTP). Another example is the MPEG video where the size of the frames is variable. If the video is showing large amounts of motion or changes then more data is sent as opposed to if the image is static. In cases like the video we have a set duration when the transmission is done. However for practical purposes we need to consider how long it takes to run the simulation. The sample video used runs for 90 minutes, which when run in simulation took around 20 hours. Rather than run the simulation for 20 hours we balance the amount of time needed to run the application to get data with how long the simulation takes. To determine how long a simulation needs to run we examine the variance of that application.

For the VoIP application we are using a constant bit rate application of only 64Kb/s. The consistent nature of this application means that we do not need to run the simulation for long to get an accurate measure of steady-state performance of its performance. We calculated the standard deviation of UDP packet sizes and reception delays. This test was run using the default wireless network setup where the UE requests a CQI value of 8 and there is no loss. The result of this test are shown in **Figure 17**. The x axis of these graphs shows how long the VoIP application needs to run for a steady state. As expected when there is no loss and with such a low throughput required the standard deviation is 0. As such we run VoIP applications for 10 seconds.



(a) Full TCP trace.

(b) TCP trace after initial startup.

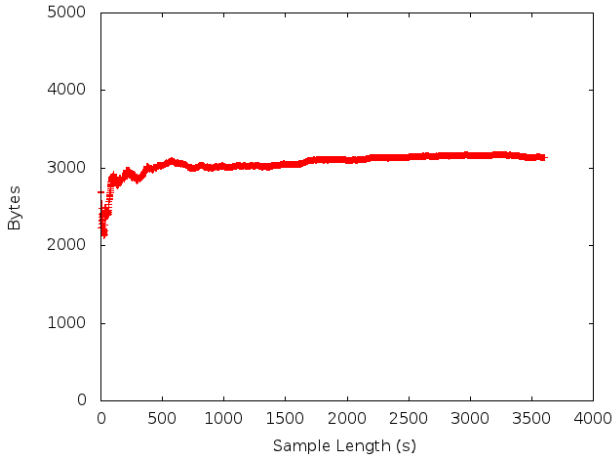
Figure 18: Standard deviation for the MPEG application.

For the FTP application the TCP throughput is of interest. When sending a file the delay is not as much of a concern as receiving the file properly so we are not as concerned with the delay of the TCP packets. To determine how long the FTP application should run we examine the standard deviation of the throughput as shown in **Figure 18**. When examining throughput we look at the data sent over time. For this example for each second of simulation time we add up all of the packets received in that second to get the number of bits received in that second. We do not average the throughput over more than one second.

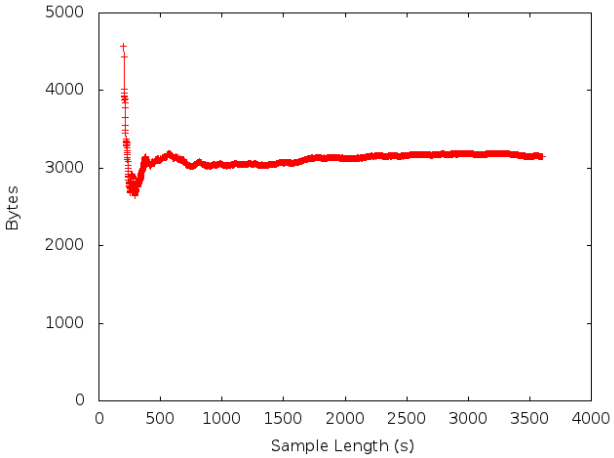
The initial variation in the application comes from the slow start phase of the TCP congestion window. Once the TCP congestion window fills the maximum bandwidth the connection allows the flow of TCP ACKs to throttle the sending rate. In **Figure 18a** the standard deviation on the throughput initially varies within 2 Mb/s before eventually dropping below 0.5 Mb/s. By starting the throughput analysis one second after the FTP application starts the the standard deviation becomes much more stable as seen in **Figure 18b**. Since the standard deviation remains stable we can run the FTP simulation for only a few seconds as long as we note the slow start discrepancy. The FTP simulations will run for 11 seconds (1 second of slow start and 10 seconds of normal operation when there is no loss).

The video application is a different situation. This application is based on a trace file of a soccer game. The frames have a variable size and spacing between transmission. Due to this variability we cannot get an accurate representation of the application with a short test. **Figure 19a** shows the standard deviation for the size of the MPEG frames and **Figure 19c** shows the delay to receive them at the UE. This test uses the same radio environment as the previous VoIP example. Here we see that the MPEG video never reaches a perfect steady state in terms of frame size or packet delay. One factor to note is that the start of the video has much more variation. It is not until after the first 500 seconds of the video trace that the standard deviation begins to normalize. This is because the beginning of the video trace is different than the rest of the video.

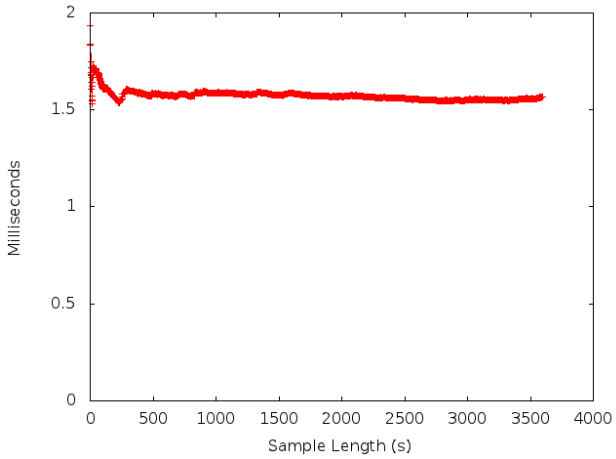
When simulating the video application we do not need to run the video specifically from start to finish. We are only interested in it as an example MPEG trace. We can therefore take advantage of this by discounting the earlier portion of the video. **Figures 19b** and **19d** shows the standard deviation for the frame size and delay when the first 200 seconds of the video are discounted. Like the full video trace the standard derivation starts to normalize after 500 seconds. In both cases (video from start and 200 seconds in) the standard deviation of the delay is less than 2 ms and the frame size is within 5,000 bytes. So for the video application we only need to run until the video is at the 500 second mark. Since starting the video 200 seconds in keeps the same standard deviation we only need to run the video between 200 and 500 seconds leaving a simulation time of only 300 seconds. This savings of 200 seconds of simulation saves us around an hour of real time.



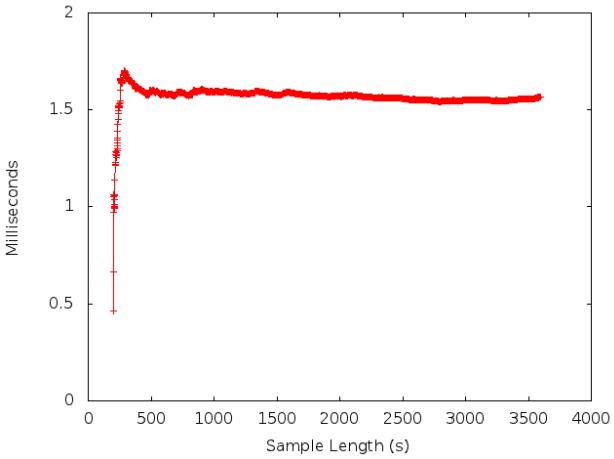
(a) MPEG frame sizes.



(b) MPEG frame sizes discounting first 200 seconds.



(c) MPEG delay.



(d) MPEG delay discounting first 200 seconds.

Figure 19: Standard deviation of frame sizes and delay for the MPEG application.

5 Results

This section looks at the results of running simulations in ns-3. **Section 5.1** examine the changes made to support NACKs in STATUS messages. **Section 5.2** is the baseline test for all applications with no wireless loss. **Section 5.2.2** covers the basics of how wireless loss impacts the applications. The results of the application simulations for VoIP, FTP and MPEG are provided in **Sections 5.3**, **5.4** and **5.5** respectively.

5.1 ns-3 LTE Additions

5.1.1 Adding NACKs to Status PDUs

As mentioned in Section 4.2 we needed to modify the LTE simulation module to support the use of NACKs in RLC AM. The first of these changes was to add support for serializing and de-serializing the NACK SNs in the RLC AM status PDU. We submitted an initial patch to the LTE module developers who wrote a test case to validate the result. This exposed some flaws in the initial patch, which we corrected. The final version of this patch passed the test case. The test case created by the developers simply input information to place in the status PDU and confirmed that the resulting bytes in the header created the correct number.

5.1.2 Detecting Sequence Numbers to NACK

The other additions to the simulator identified which PDUs had not been received and making the correct NACK requests to retrieve them. Once again the developers of the LTE module created a test case to validate this work. The test case set the loss rate and checked that the receiving RLC AM entity sent back the correct NACKs to retrieve all of the missing PDUs. There was no limit to the number of retransmissions. After multiple rounds of working with the developers and others on the discussion forum, finding bugs and suggesting fixes we finally got all of these tests to pass.

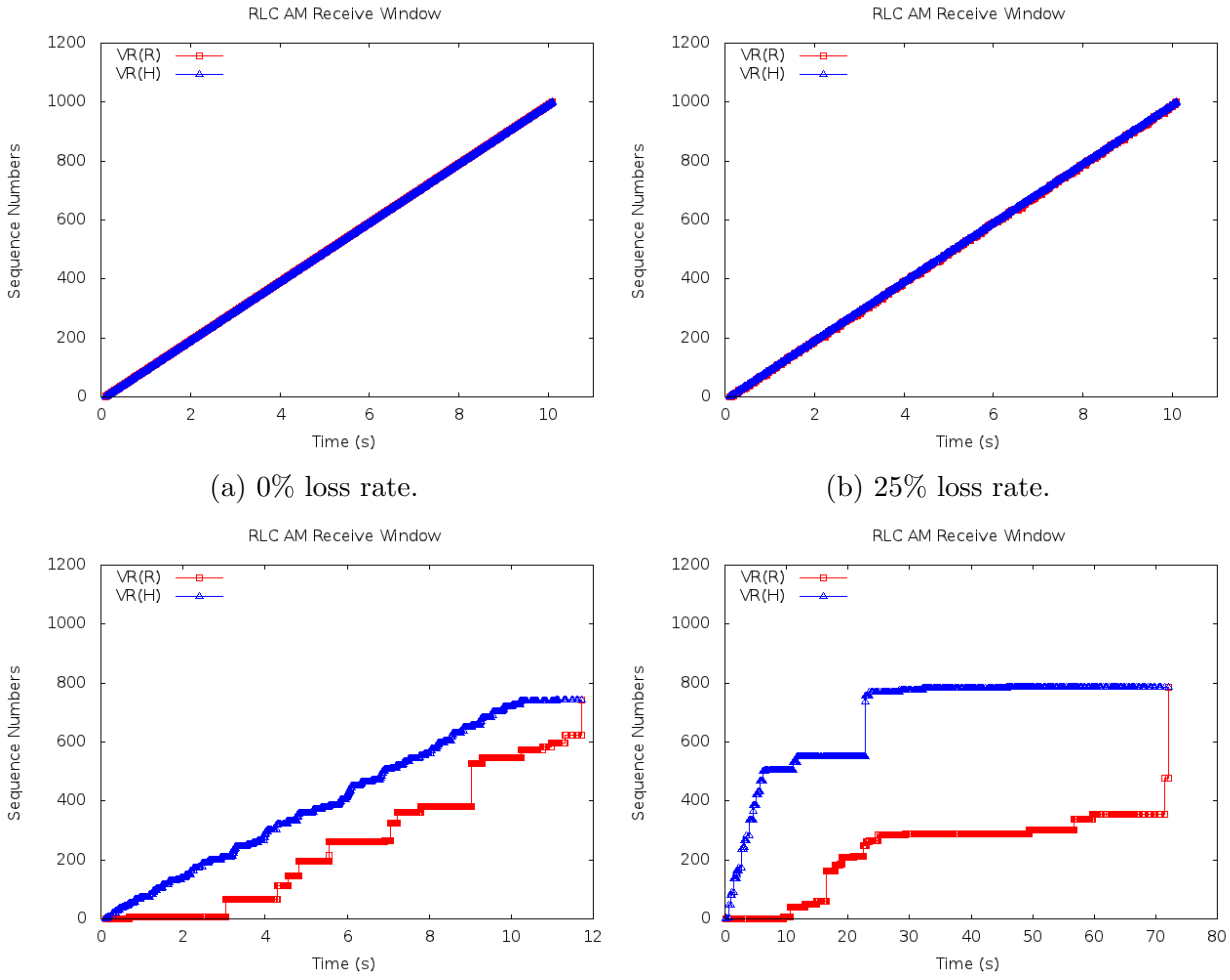
Figure 20 shows four different loss rates during the test case. The test is setup to send 1,000 RRC layer PDUs from one eNodeB to an UE. The test passes if all 1,000 PDUs make it. Since the PDUs are from the higher RRC layer the RLC layer may deliver one RRC PDU per RLC PDU or one RLC PDU may contain multiple RRC PDUs. In **Figure 20a** and **Figure 20b** the loss rate of 0% and 25% respectively produces very similar graphs. In **Figure 20a** the lower bound on the receiving window VR(R) is obstructed by VR(H), the highest seen SN. In **Figure 20b** you can see VR(R) more as there is some loss of packets.

Figures 20c and **20d** show more diversity due to the higher loss rates of 75% and 95% respectively. VR(H) stays higher than VR(R) most of the time due to so many lost packets. Fewer total RLC layer PDUs were received. As PDUs were lost the RLC layer started to pack more RRC PDUs into the RLC PDUs. This is because the loss rate set in the simulation does not take into account the radio physical layer. So the total amount of data that can be sent during each transmission opportunity is large. As PDUs are lost they are moved into the retransmission buffer. As the buffer fills the transmission opportunities have more data available to pack into the RLC PDUs. At the end of **Figures 20c** and **20d** the lower bound on the receive window finally matches the highest RLC PDU sequence number received.

5.2 Baseline

5.2.1 No Loss

The first simulation set a baseline for the VoIP and video data. This simulation had no loss and checks how the VoIP and video examples work in both UM and AM. The important results in this section is the delay experienced by the applications without any retransmissions or adjustments to other RLC settings. Future tests compare against the baseline to see how the retransmissions impact both the amount and variability of delay.



(a) 0% loss rate.

(b) 25% loss rate.

(c) 75% loss rate.

(d) 95% loss rate.

Figure 20: Validation tests of updates to LTE module

5.2.1.1 Settings

For this baseline we only use one set of values for our timers and poll flags for the RLC layer. For our original settings we chose the following:

- t-Reordering - 40 ms
- t-StatusProhibit - 20 ms

The reasoning behind this initial choice was a compromise between the ns-3 implementation and the default settings suggested by the RRC specification [5]. For t-Reordering the RRC spec recommends a value greater than 35 ms.The LTE module defaults to 20 ms for AM and 100 ms for UM so we changed this to default to 40 ms. For t-StatusProhibit RRC recommends a value greater than 0. Here the LTE module uses 20 ms.

5.2.1.2 VoIP

Figure 21 shows the baseline delay for the UDP packets with the VoIP application. Here the delay is a constant 24 ms. Of this delay, 20 ms comes from the transmission delay across the Internet link to get the data to the LTE network. Since there is no loss and the VoIP application only transmits a small amount of data the delay is the same when using both RLC AM and UM. Since there is no loss the RLC AM and UM receive windows are nearly identical.

5.2.1.3 FTP

The FTP application sends the most data it can from the server to the UE. When there is no loss and no LTE retransmissions we can expect a high throughput from this application. **Figure 22** compares the TCP transmission rate when using RLC AM and UM. Since this baseline test has no loss there is little difference for throughput performance between the two. Both AM and UM maintain a rate of around 9 Mb/s.

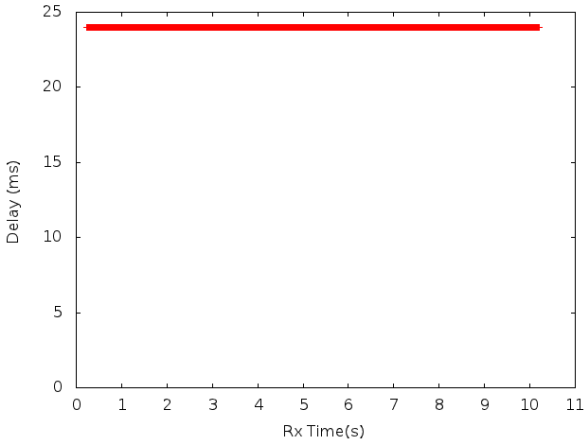


Figure 21: Baseline delay for UDP packets.

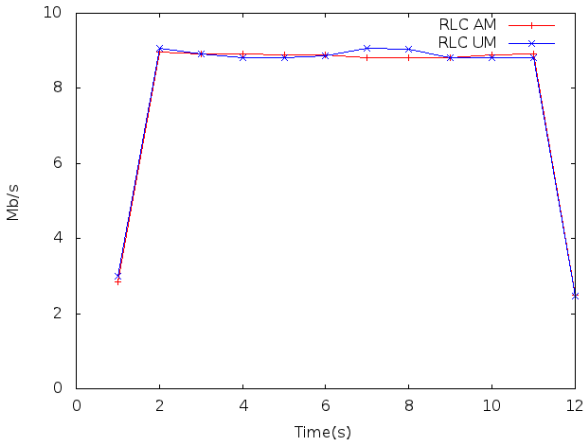


Figure 22: TCP rate seen by the UE using RLC AM and UM.

The TCP congestion window grows to fill the capacity of the link and since the sending application is keeping the sending TCP buffer full this 9 Mb/s rate seen in **Figure 22** is the highest possible TCP throughput under these simulation settings. In this simulation the radio environment causes the eNodeB to use the AMC with an MCS value of 14 (transport block index of 13). Based on **Figure 3a** , when no MIMO is used, the maximum transmission rate should be around 24 Mb/s. This maximum possible rate, however, is only achievable if the eNodeB sends a full transport block every TTI .However, when using TCP the receiver (in this case the UE) needs to send back ACKs for the received data. The scheduler on the eNodeB must therefore schedule for the UE to send its ACKs.

With an MCS value of 14, there are 100 available PRBs and the transport blocks are 3,182 bytes or 25,456 bits large. Whenever a TCP ACK needs to travel from the UE to the server one of the TTIs must be taken. For example in between time 6 and 7 seconds in this simulation the eNodeB sends 382 transport blocks on the downlink. Also during this time the UE sends 257 transport blocks on the uplink. This means that the uplink data traffic (TCP ACKs) is taking away about 6.2 Mb/s from the possible downlink throughput.

This still leaves another 9Mb/s to the theoretical max of 24Mb/s. There are some additional LTE control messages that are partially responsible for this. Another reason however is how TCP NewReno is sending data. For a 100 ms time span within time 6 - 7 seconds of this simulation the server received 44 acknowledgments. Under the default setup for TCP in ns-3, each ACK is for two TCP data packets. This occurs while TCP is in congestion avoidance so each of these ACKs New Reno increases the congestion window by less than one segment size. So within 100 ms 44 ACKs return and fewer than 44 new data packets can be sent. This helps to leave additional TTIs unused.

5.2.1.4 Video

Figure 23 shows the baseline delay for the video test. This figure shows the delay for every 20 frames received. When using AM the delay on the arriving frames varies from 24 - 46

ms. For UM the delay varies from 24 - 35 ms. Unlike the VoIP test the delay for the MPEG frames has a greater variability. In addition we can see that the overhead of the AM STATUS messages add additional delay to the frames. On average the delay on arrival of the frames under AM is about 29 ms while the average under UM is 27 ms. So while the maximum delay seen is 11 ms higher the average impact of using AM here was just 2 ms.

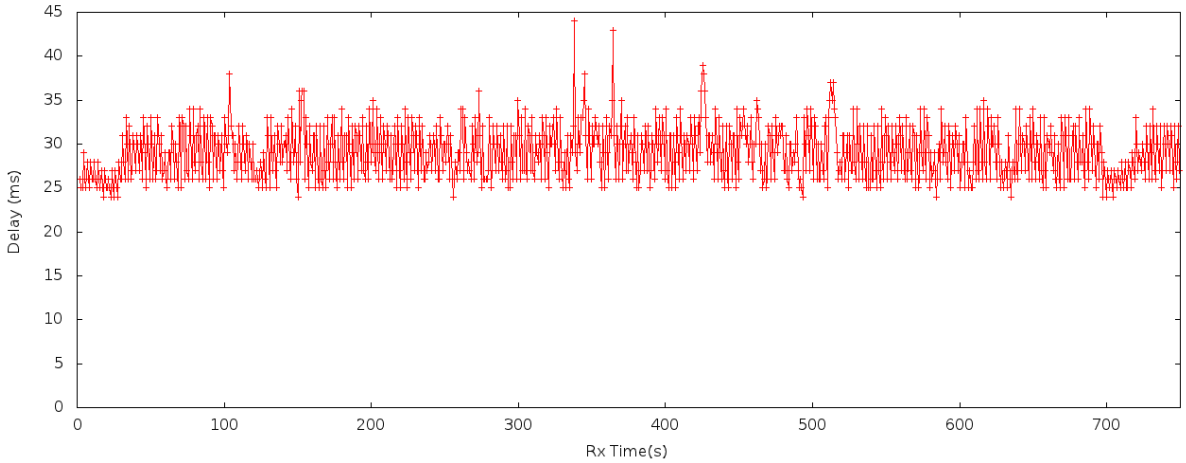
Since the baseline test has the UE requesting a CQI of 8, the eNodeB transmits with an MCS of 14. This means the eNodeB sends transport blocks of size 3,182 bytes. The average size of the MPEG frame was larger than this at 5,533 bytes. As a result, not all of the UDP packets that make up the MPEG frame can be transmitted at the same time. Some need to wait in the RLC layer transmission buffers before they can be sent. This buffering delay creates the greater variability in the UDP packet delay.

5.2.2 Basic Loss

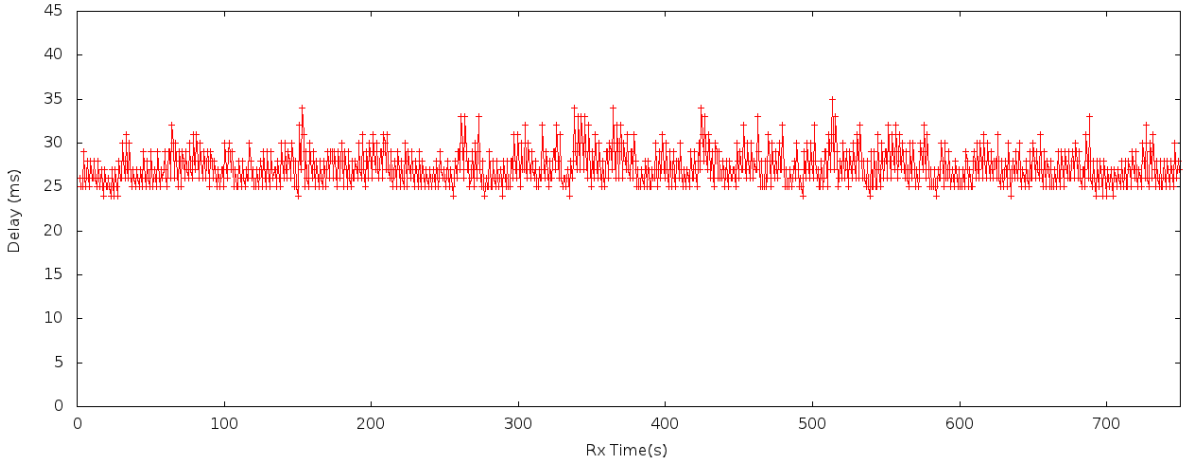
This section looks at the basics of loss events in LTE. Not all loss conditions a user may experience are emulated since radio environments are very diverse. Instead, this section applies a basic percentage of lost transport blocks and observes how this impacts the MAC, RLC and application layers.

5.2.2.1 25% Loss

First, the example of losing 25% of the transport blocks is examined. As mentioned earlier in Section 4, a fading trace file is used to decrease the SINR for specific transport blocks. **Figure 24** shows a sample from the fading trace file used for these experiments. The figure shows the SINR applied to the specific resource blocks at TTIs. The y-axis shows the resource block index, which indicates the part of the radio spectrum the resource block is transmitted in. You can see vertical yellow and black bars. The yellow bars show where we are not applying any loss while the black bars show where we are adding a SINR of -100 dBm. When loss is applied it is to all the resource blocks in the transport block, which is



(a) Baseline video frame delay over AM with no loss.



(b) Baseline video frame delay over UM with no loss.

Figure 23: Baseline video frame delay.

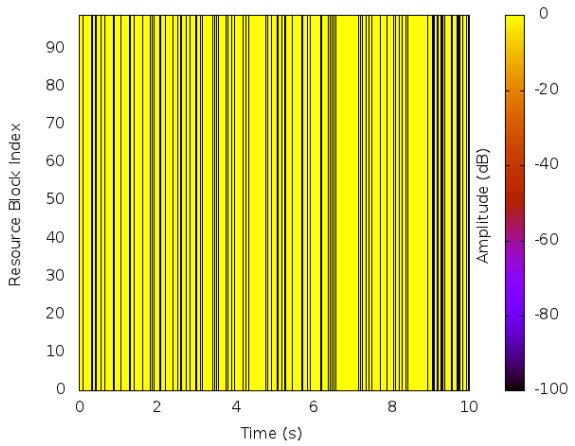


Figure 24: Sample of 25% fading trace file.

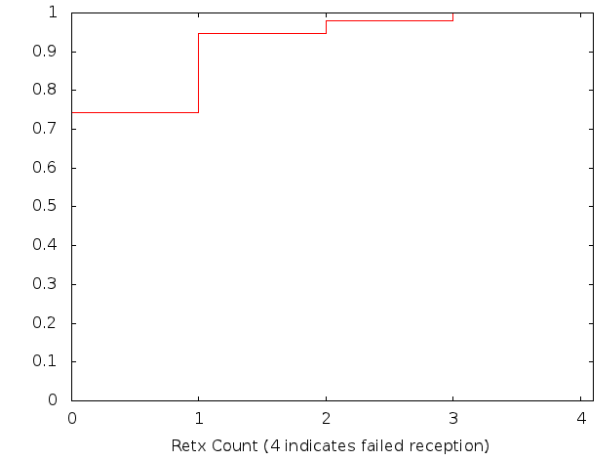
why these lines extend vertical to fill all of the resource blocks.

VoIP For the VoIP application the result of 25% loss under RLC AM and UM is shown in **Figure 25**. The cumulative distribution functions (CDFs) for the MAC layer retransmissions for AM and UM is in **Figures 25a** and **25b**. The loss rate is low enough and distributed in such a way that all lost packets are recovered by the MAC layer HARQ process. As a result these two figures are identical.

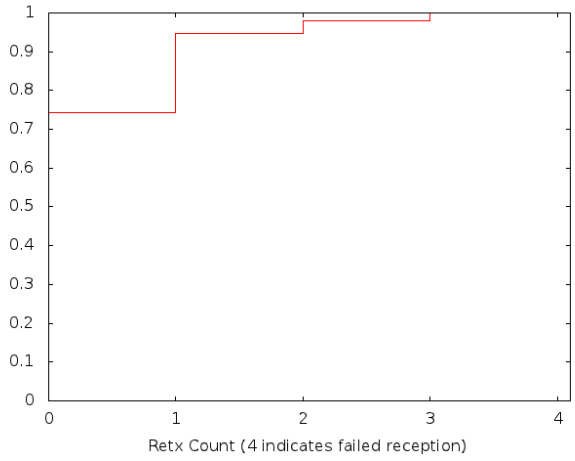
In **Figures 25c** and **25d** we have the RLC layer receive windows for AM and UM respectively. These figures show that for this instance AM and UM operate the same. SN 322 is lost in the MAC layer. When SN 323 arrives VR(R) and VR(UR) are both still at SN 321. In this case the MAC layer's HARQ process recovers SN 322 and delivers it just 1 ms after SN 323 is delivered.

Since there are no lost PDUs at the RLC layer, AM does not incur additional delays from retransmissions. Furthermore the overhead of using the AM STATUS messages is low enough that the overall delay on the VoIP application is not noticeable as seen in the delay of arrival on the UDP packets for AM and UM shown in **Figures 25e** and **25f** respectively.

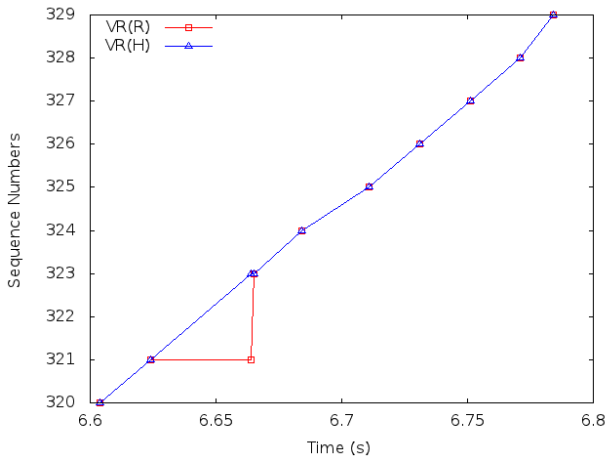
FTP While the 25% loss rate did not have a large impact on the VoIP application the FTP application is a different story. **Figure 26** shows the results of the simulation.



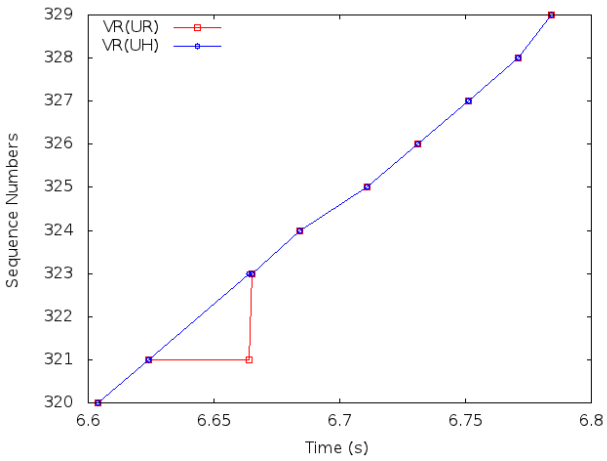
(a) CDF of HARQ retransmissions (RLC AM).



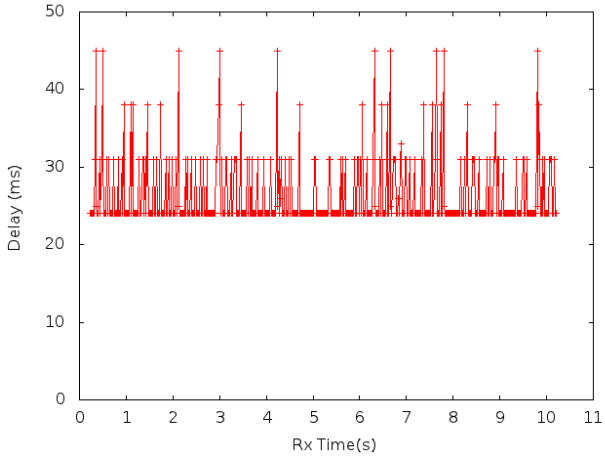
(b) CDF of HARQ retransmissions (RLC) UM.



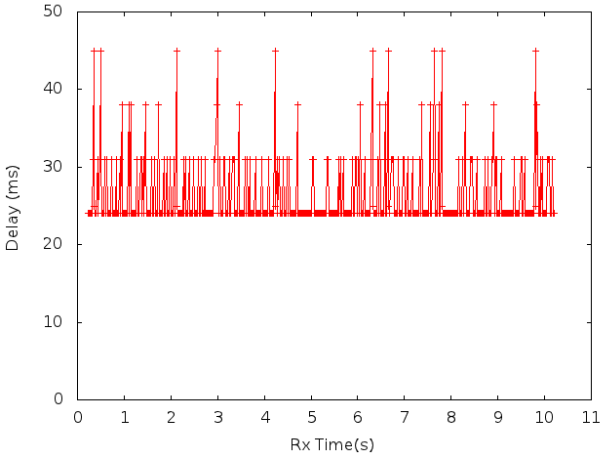
(c) RLC AM receive window.



(d) RLC UM receive window.



(e) Delay of UDP packets (RLC AM).



(f) Delay of UDP packets (RLC UM).

Figure 25: RLC AM and UM performance for VoIP with a 25% loss rate.

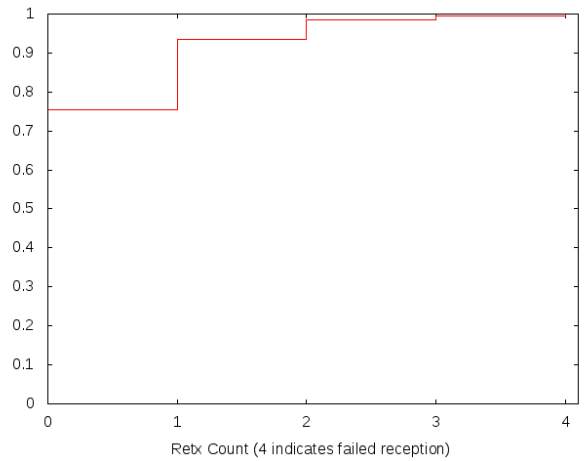
Here the MAC layer HARQ retransmissions are similar as seen in **Figures 26a** and **26b**. There are AM retransmissions in this simulation which increase the total number of data transmissions.

The throughput seen in AM and UM, shown in **Figures 26c** and **26d**, is very different. AM sees a much higher throughput than UM though both are lower than the baseline (around 9Mb/s). Since TCP adjusts its sending rate, lost packets or delayed ACKs cause the cwnd to lower or grow slowly. In this AM test, the sender experiences a retransmission time out (RTO) at 5.9 seconds. This timeout is caused by the sender not receiving any ACKs for a set amount of time. As a result TCP NewReno resets its cwnd down to one segment. This results in TCP having a lower throughput and having to regrow cwnd. These ACKs went missing due to the wireless loss.

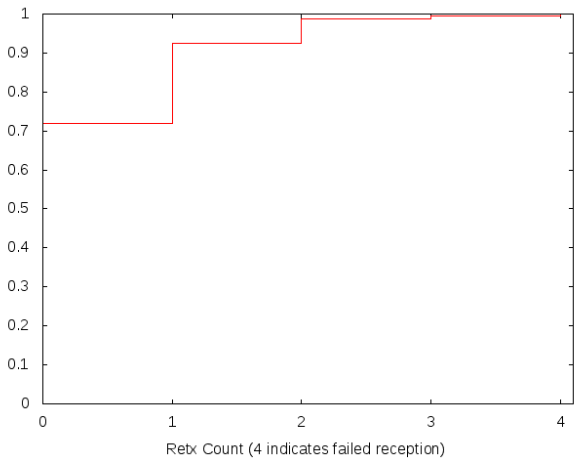
The throughput for UM is significantly lower. Here an RTO occurs at 2 seconds. This early RTO hinders the TCP throughput. In addition to this RTO, the UM test experiences 11 other RTOs. The AM simulation only experienced 2. The lack of RLC layer retransmissions results in these RTOs. This result however is not definitive. Other tests with random 25% loss settings sometimes had UM outperforming AM. Much of the results depend on exactly what TCP ACKs are delayed and when creating RTOs.

Video The delay on arrival of the video frames with a uniform 25% loss for both AM and UM is shown in **Figure 27**. Here we plot the results for every 20 frames. From these figures we can see that AM retransmissions incur a higher delay than the UM test. The average frame delay for AM was 40 ms. Not shown in the figure due to the sampling of every 20 frames is the maximum frame delay of 259 ms.

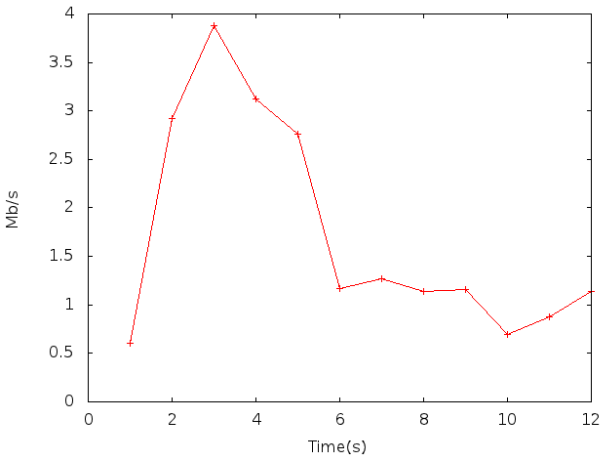
For UM the average and highest delays were 35 and 84 ms respectively. With no retransmissions in the RLC layer there was a total of 703 MPEG frames lost out of the 18,738 transmitted. These frames are considered lost if either at least one UDP packet from the frame is missing or if this frame is dependent on a missing frame. For instance, if an B frame



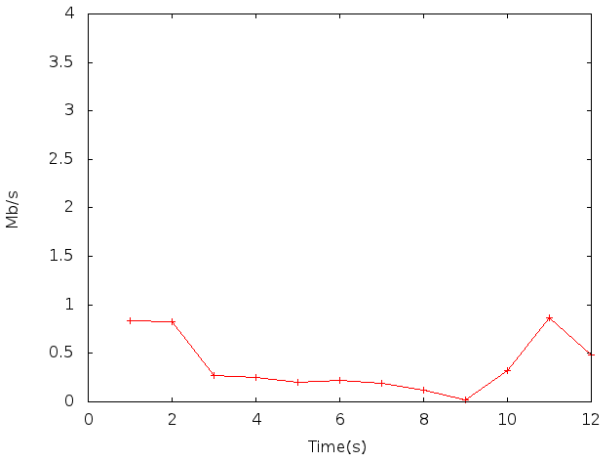
(a) CDF of HARQ retransmissions AM.



(b) CDF of HARQ retransmissions UM.

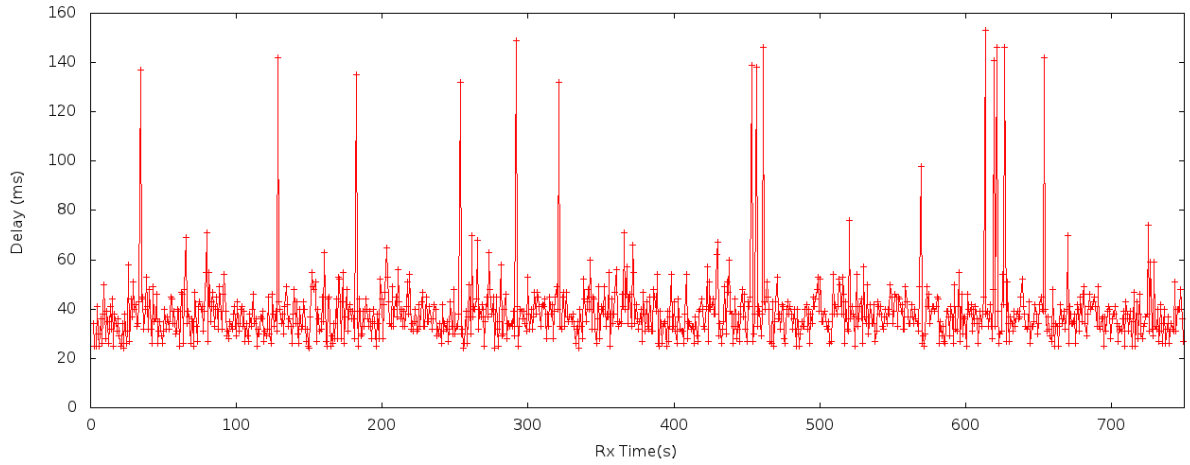


(c) TCP rate AM.

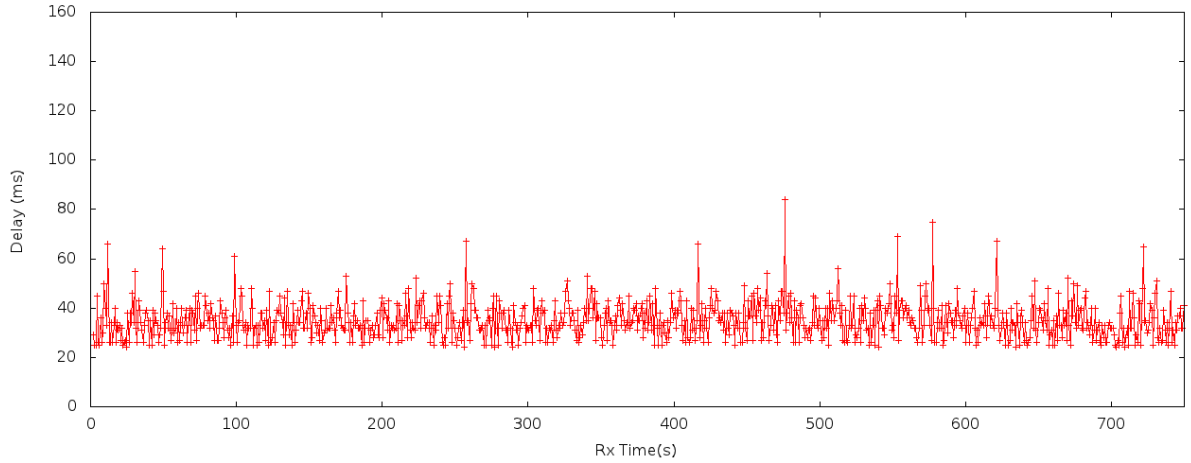


(d) TCP rate UM.

Figure 26: RLC AM and UM performance for FTP with a 25% loss rate.



(a) Video frame delay with uniform 25% loss over AM.



(b) Video frame delay with uniform 25% loss over UM

Figure 27: Video frame delay with uniform 25% loss.

is lost then only that frame is lost. If however, an I or P frame is lost then all frames that depend on them are also lost.

5.3 VoIP Simulations

This section investigates at how bursty loss events combined with modification to RLC layer settings impact the VoIP application. First the t-Reordering and t-StatusProhibit timers are adjusted to find settings that provide better performance for VoIP. Then with these timers fixed, the loss rate of the wireless link is adjusted to find if VoIP works better under AM or UM.

5.3.1 VoIP with varied RLC settings

To setup the fading trace file we select a loss rate of 10%. Additionally, we need some example gap lengths for **Equations 5** and **6**. These equations set the transition probabilities when the gap lengths are known. We need the transition probabilities to generate our gap lengths. We first select a set of input gap lengths to generate the transition probabilities and then use the Gilbert model to generate the wireless gaps to use in the simulation. For the input gap lengths we randomly select gaps between 1 - 30 ms inclusive and then randomly select the number of those gaps to match the loss percentage. We chose these gap lengths due to the RTT for the MAC layer HARQ process. We want to be sure that the gap lengths can be large enough to encompass the 3 maximum retransmissions. With these retransmissions taking 7 ms in the simulator, a gap length of 21 would be sufficient. We rounded this up to 30 to account for more input gap lengths. The output of these equations gave us the state change probabilities. While in the good state there is a 0.633% probability of transitioning to the bad state. When in the bad state there is a 6.255% probability of transitioning to the good state.

The result of this was a fading trace file that had 9.96% total loss of TTIs. The CDF in **Figure 28** comparing a uniform 10% loss, the input gap lengths to the model and the resulting gap length used to create the fading trace file.

While our simulation shows the delay of delivering UDP packets, this is not representative

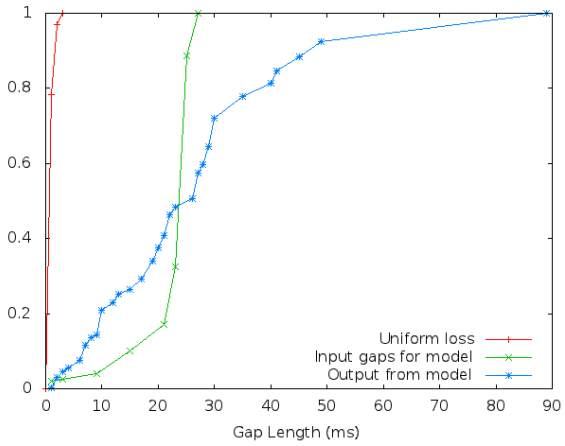


Figure 28: CDF of loss gap lengths.

of a full VoIP conversation. We used a simple simulation where a server sent the UDP packets into the LTE E-UTRA network. In a real situation there would be another VoIP client sending data into the core network that would then be directed to the LTE network. To account for the core latency of getting the UDP packets to the LTE network, we add a fixed delay to all of the packets. We use latency statistics collected in the Verizon network available from [40] to try two call scenarios. One call represents trans Atlantic and the other trans Pacific. These statistics represent the latency on Internet control message protocol (ICMP) packets that traveled through core networks to span different network ranges. This data from Verizon is for on March 2014. As the statistics shown in [40] are likely to be updated we have recorded them in **Table 6**. When calculating the MOS and delays for these tests we add the delays reported by Verizon to account for the core network delays. The simulations have a server that sends data into the LTE network over a link with a delay of 20 ms. This delay would be a part of the Verizon core network delay so 20 ms is also subtracted from the Verizon core network delays to take this into account.

To calculate the MOS we could look at the entire simulation and get the average. However, in a conversation, bursts of bad errors could cause the user to perceive the call as bad quality even when the average for the entire conversation is good. If we take any arbitrarily small unit of time to calculate the MOS, we can see results for only a few packets, which may

Table 6: Verizon core network latencies March 2014 [40]

Region	Latency (ms)
Trans Atlantic	77
Europe	14
North America	37
Intra-Japan	8
Trans Pacific	110
Asia Pacific	97
Latin America	143

not be noticeable to the user. To determine a smaller window for finding the worst MOS, we use the concept of talkspurts. When having a conversation a talkspurt is an amount of time a person speaks before stopping to allow for a response or after finishing a sentence. In these simulations only the downlink traffic is considered, which would only be one part of the conversation. In a real VoIP situation the UE in the simulation would respond with uplink traffic. It is assumed that the server sending data on the downlink would contain talkspurts of a normal conversation. Research conducted by Norwine and Murphy from Bell Laboratories [33] examined aspects of phone conversations including talkspurt length. They found that the mean length for a talkspurt was 4.14 seconds. While examining our simulation traces we average the results and get MOSs for multiple chunks of the simulation time equaling 4.14 seconds each. We do this by taking the simulation and collect MOS for every window of 4.14 seconds we can. We then report the worst MOS seen during the simulation among these talkspurts.

5.3.1.1 t-Reordering

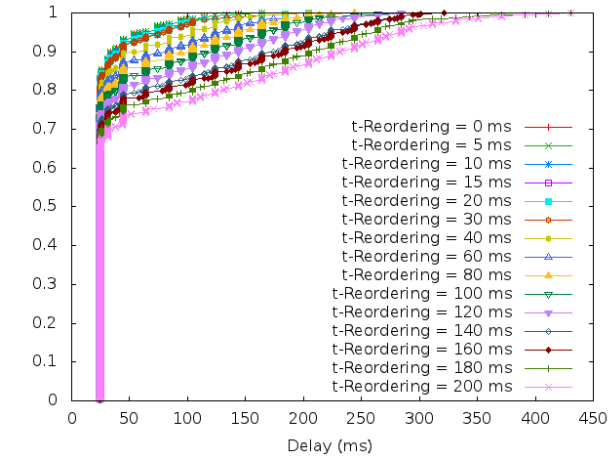
We first look at how adjusting t-Reordering impacts the VoIP application. As mentioned earlier, this timer adjusts how long to wait after the RLC layer receives data out of order from the MAC layer before considering this data to be lost. When running RLC AM the reordering timer controls when lost PDUs become visible for RLC retransmissions. In UM the timer controls how long the RLC layer will wait before abandoning reception of out of

order PDUs. While testing t-Reordering we leave the timer t-StatusProhibit set to 20 ms since that was the default in the ns-3 LTE module. Also we are testing with only a subset of the recommended timer settings specified in **Table 5** We test setting t-Reordering to values from 0 - 40 ms at increments of 5 ms, then from 40 - 200 ms at increments of 10 ms.

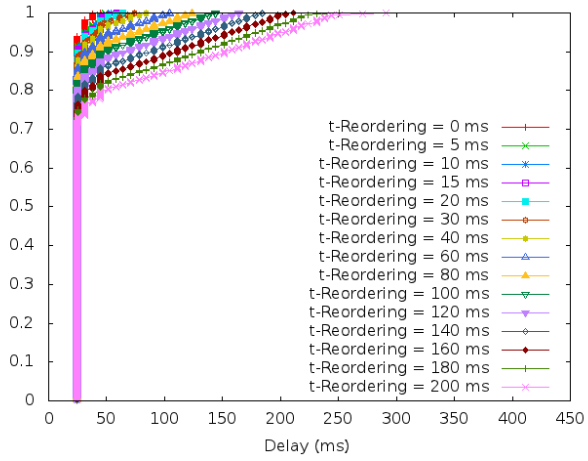
Figure 29 shows the results of adjusting the t-Reordering timer. First **Figures 29a** and **29b** provide the CDFs of the delay on all the UDP packets (just the delay in the simulation, core network not included) for AM and UM respectively. UM produces a lower delay than AM as one would expect. Also for all values of t-Reordering the CDF is similar where 60 - 70% of the packets see the same delay and the remainder follow a similar trend.

In **Figures 29c** and **29d** we have the average delay of all the UDP packets (core network included) for both the trans Atlantic and trans Pacific tests. Setting the timer t-Reordering higher increases the packet delay for both AM and UM. The AM Pacific test goes from a delay of 119 ms at t-Reordering = 0 ms to a delay of 162 ms when t-Reordering was 200 ms, which is a 36% change. However, the MOS shows that this additional delay does not greatly impact the call quality. The worst MOS scores across the talkspurts are shown in **Figures 29e** and **29f** for AM and UM. Adjusting t-Reordering does not change the MOS from a good quality. In the Pacific AM test the MOS changed from 4.38 to 4.34 when t-Reordering was 0 and 200 ms respectively.

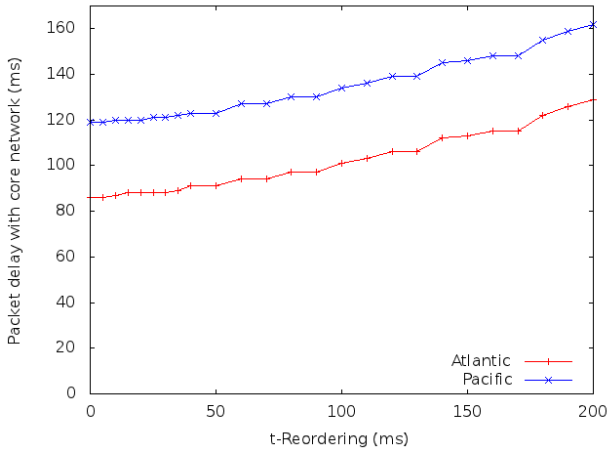
The UDP packet delays for AM and UM where t-Reordering is set to 0 and 5 ms are shown in **Figure 30**. When t-Reordering is set to 0 or 5 ms in AM there is no difference to the application as seen in **Figures 30a** and **30c**. When the same settings are used in UM there are fewer spikes when t-Reordering is 0 ms in **Figure 30b** than 5 ms in **Figure 30d**. This is due to the additional packets lost when t-Reordering is not set high enough to allow MAC layer retransmissions.



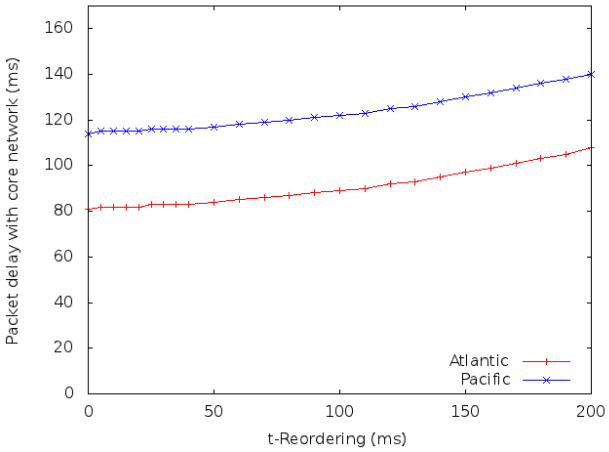
(a) UDP delay CDFs AM



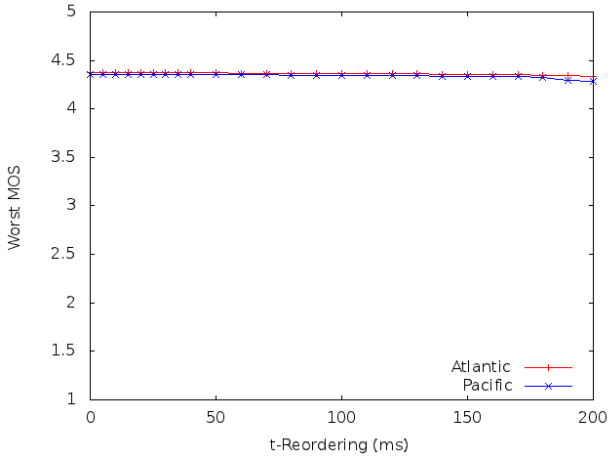
(b) UDP delay CDFs UM



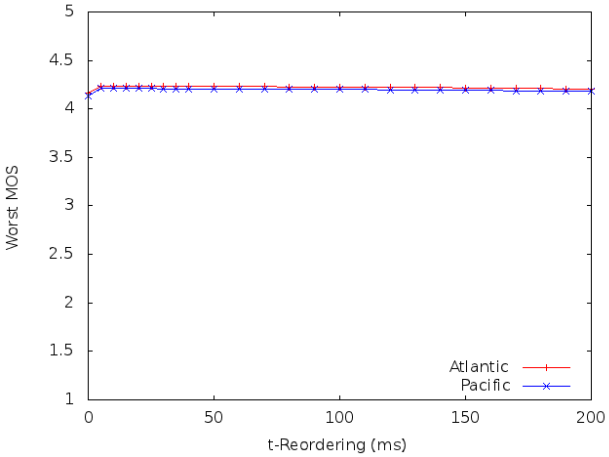
(c) UDP delay average AM



(d) UDP delay average UM

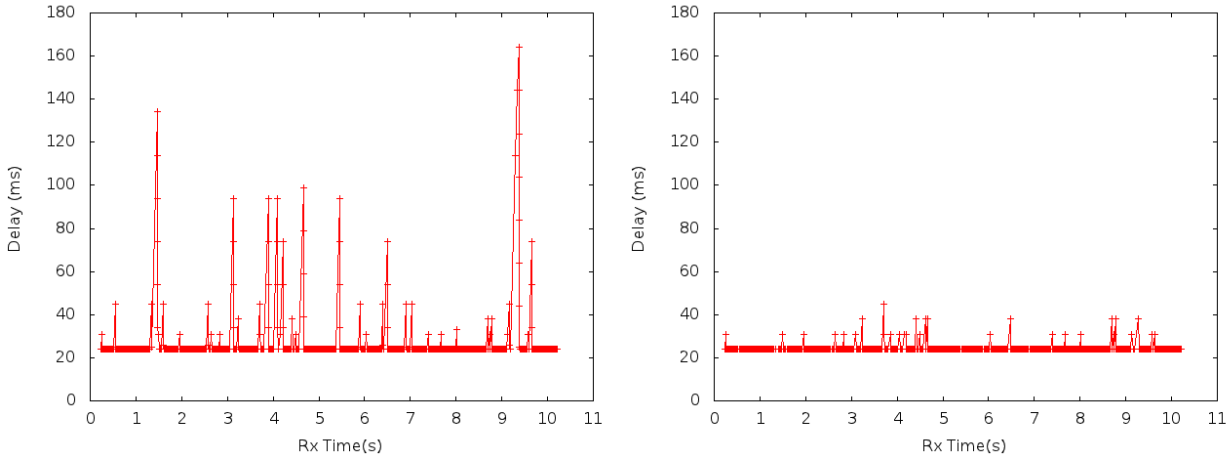


(e) Worst talkspurt MOS AM



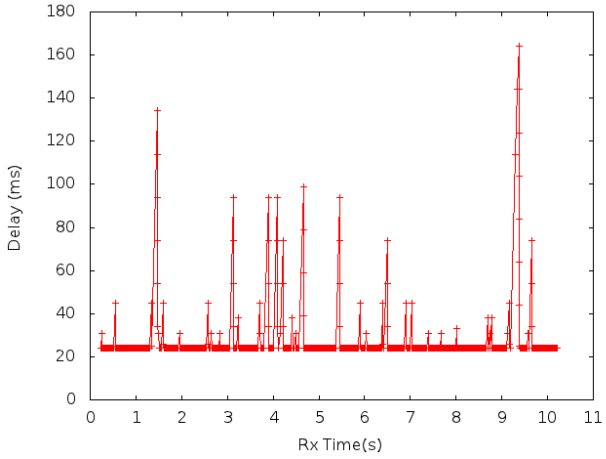
(f) Worst talkspurt MOS UM

Figure 29: VoIP t-Reordering tests.

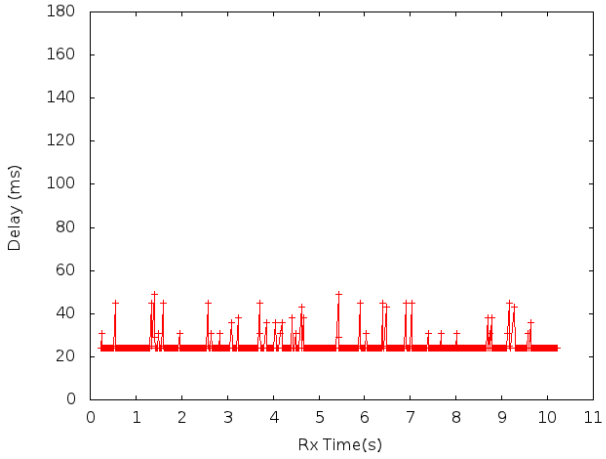


(a) AM t-Reordering = 0 ms

(b) UM t-Reordering = 0 ms



(c) AM t-Reordering = 5 ms



(d) UM t-Reordering = 5 ms

Figure 30: UDP delay when varying t-Reordering timer.

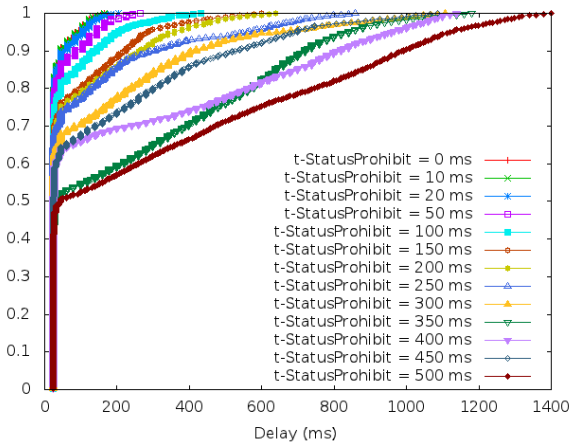
5.3.1.2 t-StatusProhibit

This section considers adjusting t-StatusProhibit and its impact on the VoIP application. This timer adjusts the minimum time a receiver using AM will wait before sending a STATUS message with ACKs and NACKs. When transport blocks cannot be recovered by the MAC layer, this timer can decrease or increase the time taken before the sender can be notified about the missing data. This timer is not the only factor in this scenario. The sender identifies in data PDUs if a STATUS message is requested. Furthermore as mentioned in the previous section, the setting on t-Reordering can change when a missing PDU can be included as a NACK in the STATUS message.

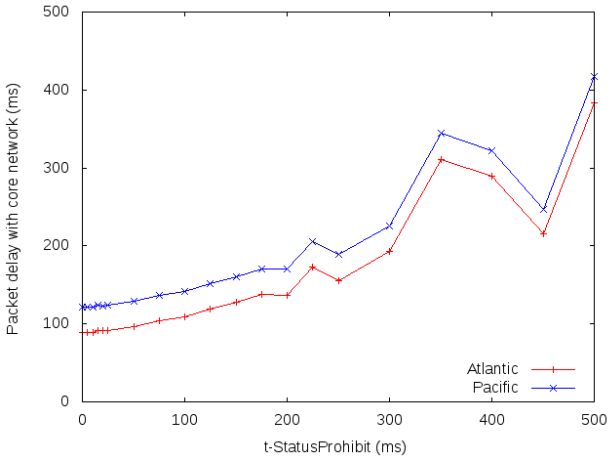
The values for t-StatusProhibit have more recommended settings as seen in **Table 5**. We again test only a subset of these possible values. This timer is tested at from 0 - 25 ms at 5 ms increments, then from 25 - 250 ms at 25 ms increments and then from 250 - 500 ms at increments of 50 ms.

The results for these tests are shown in **Figure 31**. Since t-StatusProhibit only impacts AM these results do not show any comparison to UM. The CDF shown in **31a** shows that the settings on t-StatusProhibit has a greater impact on delay compared to t-Reordering. Also while increasing t-Reordering increased the delay on packets lost consistently, here we see that some settings of t-StatusProhibit cross over other higher settings. This result is also apparent for the average UDP packet delay (with core network) in **Figure 31b** and the worst MOSs for the talkspurts in **Figure 31c**.

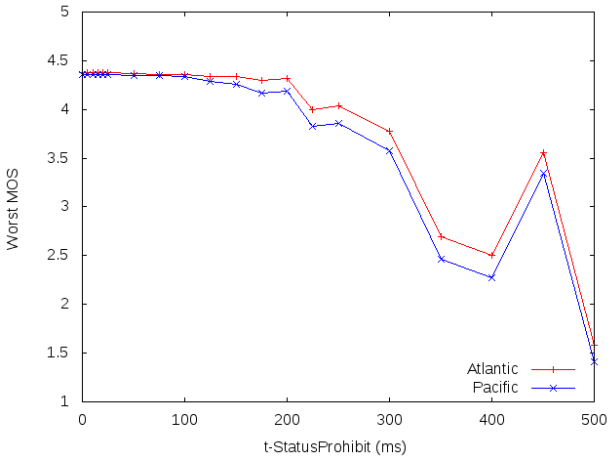
The reason why some higher values for t-StatusProhibit produce better results than lower settings is dependent on the application in use and where the wireless loss events occur. This timer reduces the total number of STATUS messages that can be sent each second as shown in **Figure 32**. The timer is started whenever a STATUS message is sent. The STATUS messages indicate what PDUs need to be retransmitted. However not all lost packets are available to be NACKed in every STATUS message. Since the MAC layer's HARQ process is attempting to recover lost data the t-Reordering timer controls when a lost PDU can



(a) UDP delay CDF for t-StatusProhibit.



(b) UDP delay average AM



(c) Worst talkspurt MOS AM

Figure 31: VoIP t-StatusProhibit tests.

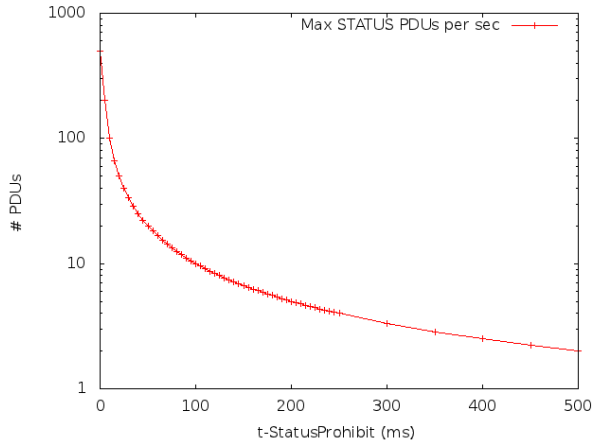


Figure 32: Maximum possible STATUS PDUs that can be sent in 1 second.

be recored as a NACK and be requested for retransmission. If the timers and loss happen just right then a t-StatusProhibit of 400 ms may send out a STATUS message just a few milliseconds before t-Reordering allows a missing PDU to be NACKed. As a result that PDU will not be requested for retransmission until at least another 400 ms have passed. If however t-StatusProhibit is set longer, to 450 ms then that PDU would be recorded in the STATUS message. When t-StatusProhibit is short there are more STATUS messages sent so the impact of having t-Reordering and t-StatusProhibit interfering is minor. However as t-StatusProhibit grows the cost of a missed NACK increases.

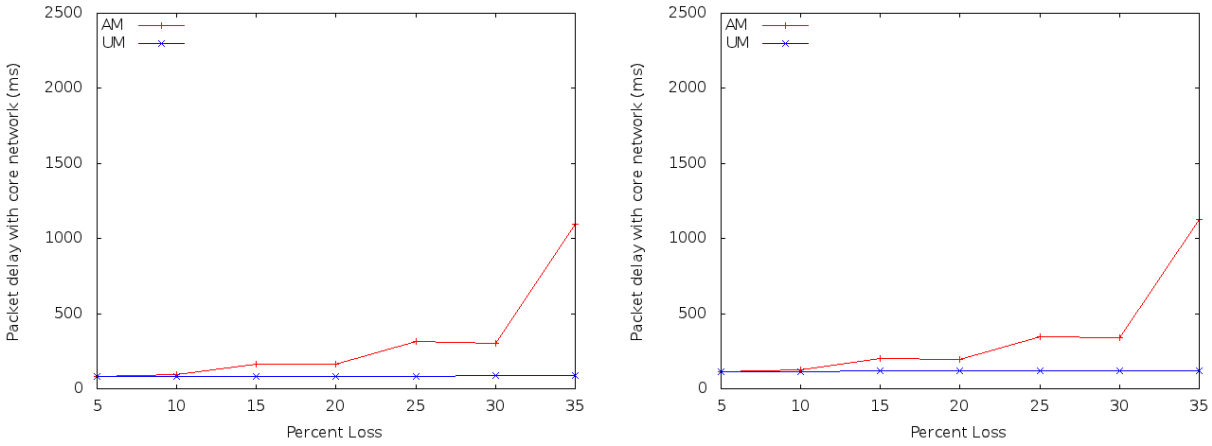
5.3.2 VoIP with AM vs UM with varied loss

We next want to fix both t-Reordering and t-StatusProhibit and adjust the amount of wireless loss using the Gilbert-Elliot model. From the previous tests we know that adjusting t-Reordering has little impact while having a large t-StatusProhibit increases the cost of missing NACKs. We set t-Reordering to 30 ms. Because we do not want to risk loosing any PDUs that could be recovered by the MAC layer. Since the timer is not started until a PDU with a higher SN is received and LTE can send data every millisecond we only need to worry about the three retransmissions. With a HARQ RTT of 8 ms we need to make sure the timer is at least 24 ms. We chose 30 ms to allow some room for starting the timer.

We set the value of t-StatusProhibit to 50 ms. From the previous tests in **Figure 31b** and **31c** we can see that a setting of 50 ms does not experience the same performance impacts as higher values. Also **Figure 32** indicates that the potential cost of having very low values could consume many TTIs leaving few for user data.

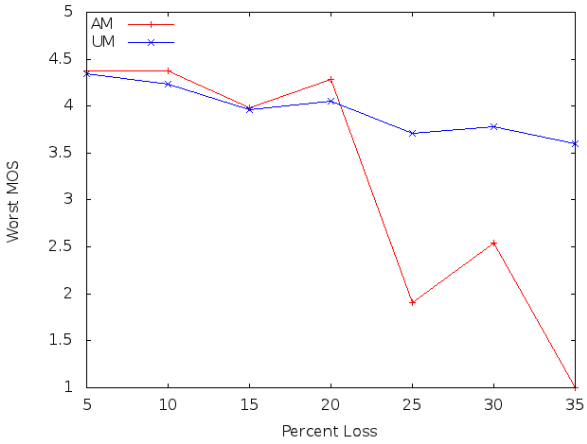
Next the Gilbert-Ellilot model was used to create fading trace files for loss rates of 5%, 10%, 15%, 20%, 25%, 30%, 35%, 40%, 45% and 50%. For loss rates 5 - 35%, we can generate the loss gap lengths using our original method where we first randomly select a set of gap lengths from 0 to 30 to find the state transition probabilities that are then used to run the Gilbert model to generate the actual fading loss trace. However for loss rates 40 - 50% these gap lengths proved too small. The resulting state transition probabilities kept producing fading trace files with total loss percentages less than their targets. So for loss rates 40 - 50% we randomly select gap lengths up to 100 as input. This then produced state transition probabilities for the Gilbert model which resulted in fading trace files that met the loss goals.

The results of these simulations are shown in **Figure 33** for the Gilbert loss model where our input gap lengths are up to size 30 while **Figure 34** shows the results for higher loss rate where we increased the gap lengths on the input data up to 100. Here for the Atlantic (left) and Pacific (right) tests we see that at low loss rates the performance of AM and UM are similar. However as the loss rates increase the extra delay in using AM out weighs the

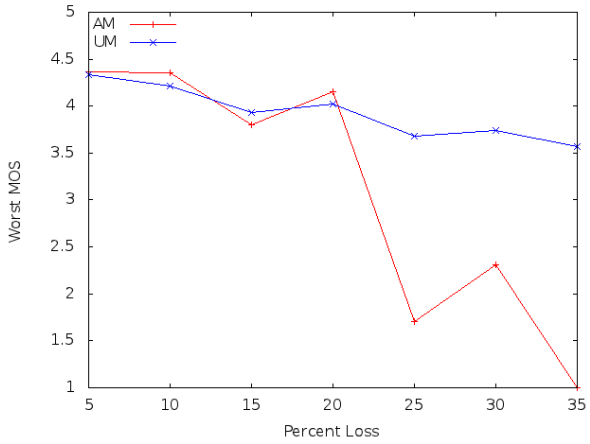


(a) UDP delay with core network Atlantic.

(b) UDP delay with core network Pacific.



(c) Worst MOS scores Atlantic.



(d) Worst MOS scores Pacific.

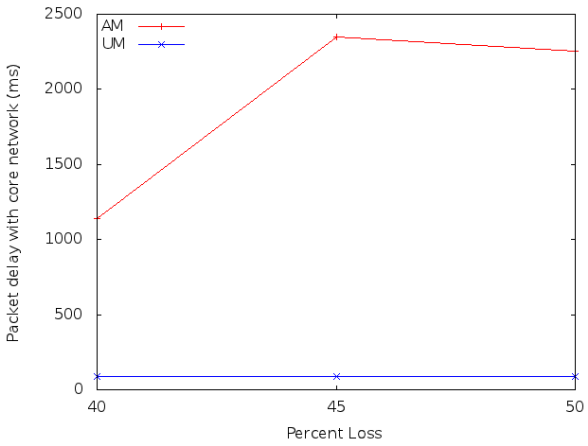
Figure 33: Results for different losses with t-Reordering = 30 ms and t-StatusProhibit = 50 ms, with input model gap lengths up to 30.

impact of higher packet loss for using UM.

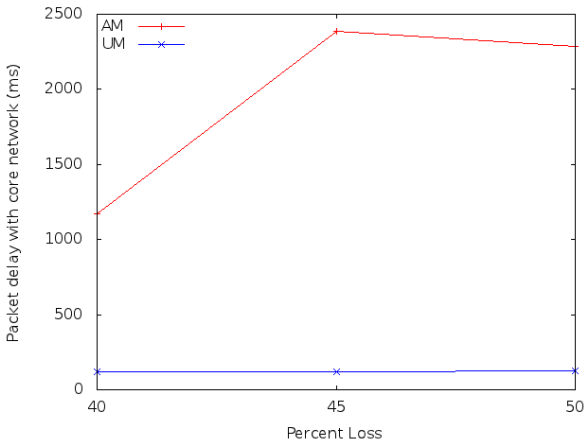
In **Figures 33c** and **33d** we see that for AM the MOS gets better from 25 - 30% loss rate. This seems odd as we would expect the MOS to get worse as AM delays longer and longer to retrieve lost data. However in **Figures 33c** and **33d** we see that the average delay does not change dramatically between loss rates of 25 and 30%.

The reason for this odd behavior is due to how the loss gaps landed during the simulation.

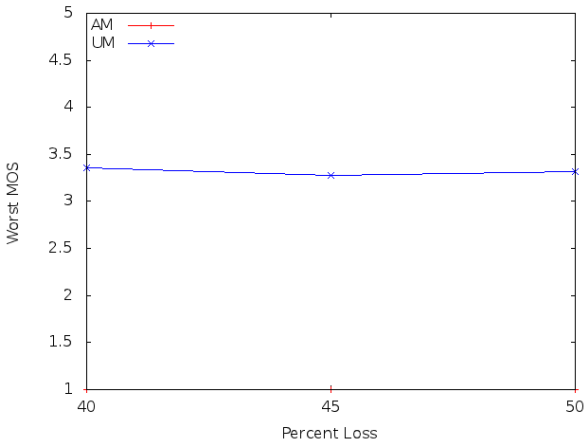
In **Figures 35a** and **35b** we see the delay on arrival of the UDP packets when running under 25 and 30% loss respectively. At the 25% loss rate there is a larger contiguous block of delayed packets from time 6 - 10. For the 30% loss rate there is no one contiguous block



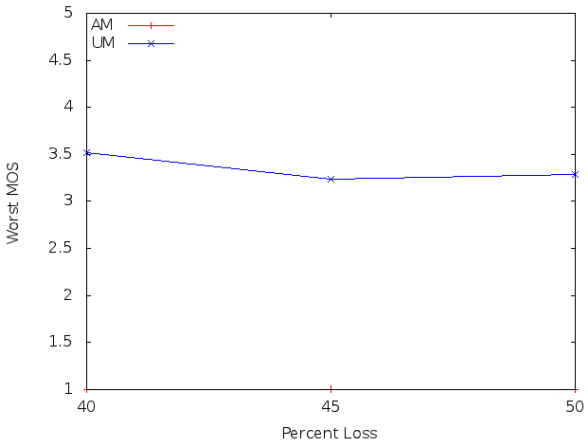
(a) UDP delay with core network Atlantic.



(b) UDP delay with core network Pacific.

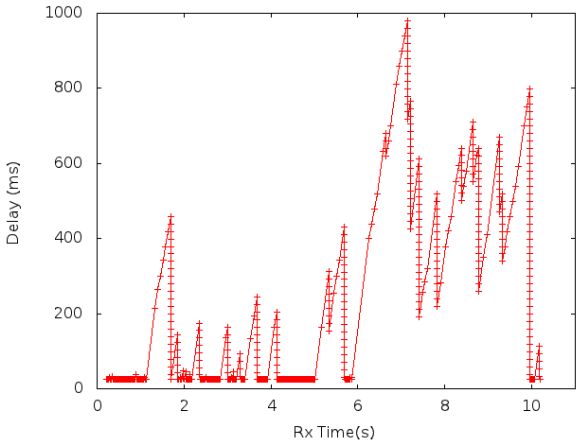


(c) Worst MOS scores Atlantic.

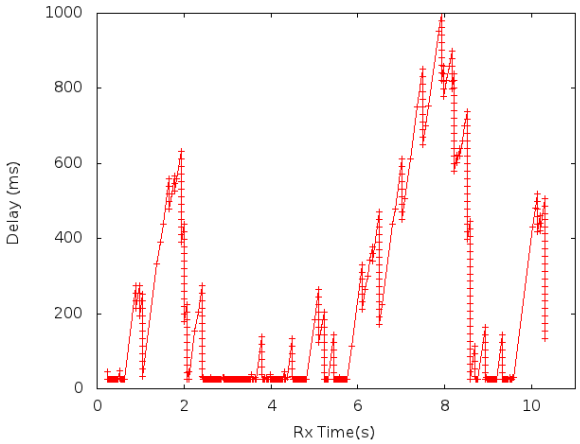


(d) Worst MOS scores Pacific.

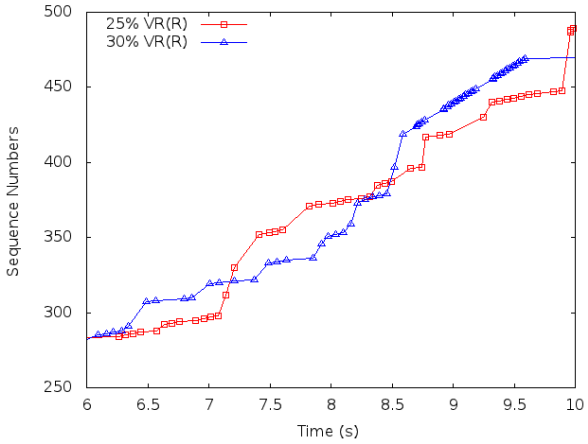
Figure 34: Results for different losses with t-Reordering = 30 ms and t-StatusProhibit = 50 ms, with input model gap lengths up to 100.



(a) UDP packet delay for 25% loss.



(b) UDP packet delay for 30% loss.



(c) RLC AM lower bound receiving window for 25 and 30% loss test.

Figure 35: 25 and 30% loss VoIP AM tests.

of long delays for so long. When we examine the worst talkbursts of 4.14 seconds this long set of contiguous delays in the 25% loss rate test produces a lower minimum MOS than the 30% test.

The reason for this large contiguous regions of delayed packets for the 25% loss test is due to how the wireless loss gaps impacted the receive window at the RLC layer. For the 25% loss test the lower bound of the receiving window remains lower for longer than in the 30% loss test. **Figure 35c** shows the lower bound for both tests RLC AM receive window for the time frame 6 - 10 seconds. At the 25% loss test there is a lower receive window from 6 to about 7.2 seconds and from about 8.5 - 10 seconds. It is only from about 7.2 to 8.5 seconds

when the 25% loss test has a higher receive window. As a result more data is being held in the RLC buffer longer in the 25% loss test compared to the 30% loss test. This difference in AM performance for 25% and 30% loss rates shows that the application is impacted not just by the amount of loss but also how the loss is shaped.

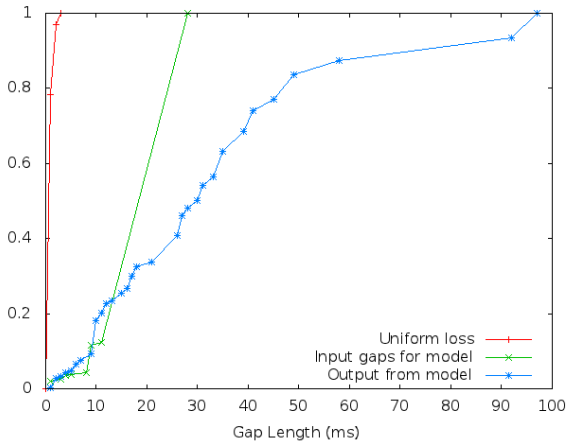


Figure 36: CDF of loss gap lengths.

5.4 FTP Simulations

In this section we look at how bursty loss events combined with modification to RLC layer settings impacts the FTP application. The first set of experiments manipulate the t-Reordering and t-StatusProhibit timers to find settings that provide better performance. Then with these timers fixed, the loss rate of the wireless link is adjusted to find if FTP works better under AM or UM.

5.4.1 FTP with varied RLC settings

Here we setup a fading trace file using the Gilbert-Elliot model. Again we set the loss rate at 10% and randomly select the input example gap lengths. As a result, when the model is in the good state there is a 0.653% chance of transitioning to the bad state. When in the bad state there is a 6.471% chance of transitioning to the good state. After running the Gilbert-Ellilot model with these state transitions the fading trace file produced an error rate of 9.999%. The CDF for the gap lengths is shown in **Figure 36**.

Table 7: FTP results AM t-Reordering

t-Reordering	Average Mb/s	Standard Dev Mb/s
0	5.2	2.2
5	4.0	1.9
10	4.6	2.2
15	5.1	2.5
20	4.4	1.9
25	2.3	1.3
30	2.3	1.3
35	2.3	1.3
40	2.5	1.3
50	4.7	1.9
60	4.4	2.0
70	5.2	1.9
80	5.3	2.3
90	5.0	2.1
100	3.4	2.0
110	4.4	2.2
120	3.9	2.2
130	4.3	2.0
140	3.7	2.1
150	3.4	1.5
160	3.6	1.7
170	3.6	1.7
180	3.6	1.7
190	4.0	1.6
200	3.6	1.7

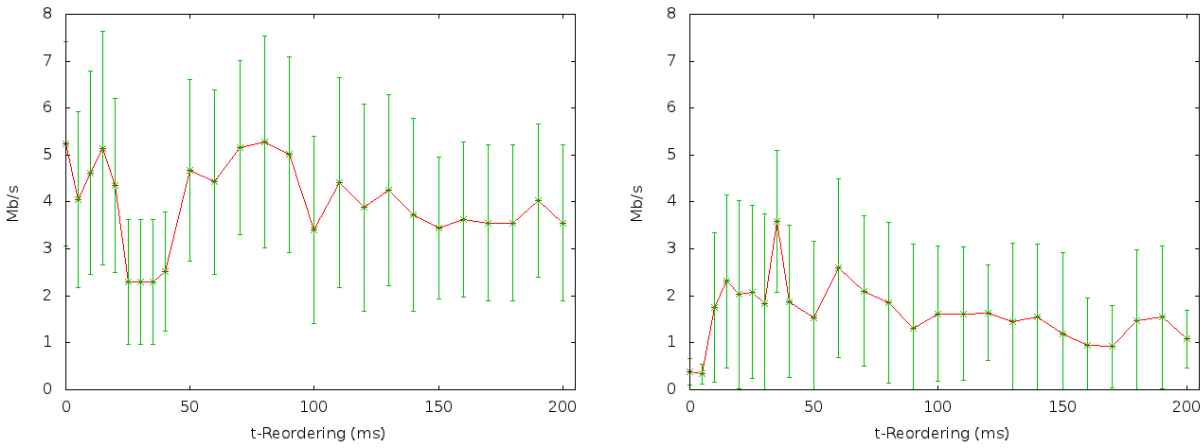
5.4.1.1 t-Reordering

The results of adjusting the timer t-Reordering for AM and UM are shown in **Tables 7** and **8** respectively. These tables list the average and standard deviation of the TCP throughput seen at the receiver. The standard deviations range from 0.2 - 2.5 Mb/s. The reason for the large standard deviation comes from TCP RTOs. When the TCP data or TCP ACKs are lost the sending server can see long delays from arriving ACKs. If this delay is too long TCP experiences RTOs and lowers cwnd to 1 segment size. TCP returns to slow start. The results for t-Reordering are also shown in **Figure 37**.

The throughput In **Figure 37** generally gets worse as the setting of t-Reordering in-

Table 8: FTP results UM t-Reordering

t-Reordering	Average Mb/s	Standard Dev Mb/s
0	0.4	0.3
5	0.3	0.2
10	1.8	1.6
15	2.3	1.8
20	2.0	2.0
25	2.1	1.8
30	1.8	1.9
35	3.6	1.5
40	1.9	1.6
50	1.5	1.6
60	2.6	1.9
70	2.1	1.6
80	1.9	1.7
90	1.3	1.8
100	1.6	1.4
110	1.6	1.4
120	1.6	1.0
130	1.5	1.7
140	1.6	1.6
150	1.2	1.7
160	0.9	1.0
170	0.9	0.9
180	1.5	1.5
190	1.5	1.5
200	1.1	0.6



(a) AM t-Reordering

(b) UM t-Reordering

Figure 37: Average FTP throughput for t-Reordering.

creases. However, there are moments, like when t-Reordering for AM is 25 - 40 ms, where the average throughput is lower than for the other settings. There is no clear setting for t-Reordering that improves both the results for AM and UM Setting t-Reordering to 50 ms produces one of the higher throughputs in the AM test. In UM this is one of the lower throughputs but it is not the lowest average throughput. Other settings for t-Reordering produced higher average throughputs but some of these have higher standard deviations, such as the AM test with t-Reordering of 80 ms with an average throughput of 5.3 Mb/s and a standard deviation of 2.3 Mb/s.

Without a clear setting for t-Reordering to improve performance we focus on lower values of t-Reordering. The reasoning for this is that t-Reordering is meant to wait for the lower MAC layer to recover lost data. With a maximum of three retransmissions and a RTT of 7 ms on the retransmissions the setting of t-Reordering should be greater than 21 ms. Having this timer set much larger than this value does not allow the MAC layer to recover additional data. t-Reordering is fixed at 50 ms for the tests with multiple loss rates.

Table 9: FTP results AM t-StatusProhibit

t-StatusProhibit	Average Mb/s	Standard Dev Mb/s
0	2.9	1.5
5	4.4	3.2
10	3.9	2.1
15	4.2	1.8
20	2.5	1.3
25	3.7	2.5
50	4.9	2.3
75	5.5	2.1
100	4.7	2.1
125	3.8	1.8
150	4.7	1.9
175	3.1	2.0
200	3.5	1.6
225	2.7	1.6
250	2.5	1.9
300	2.9	1.3
350	2.3	2.1
400	3.7	2.1
450	3.7	2.1
500	1.6	1.1

5.4.1.2 t-StatusProhibit

The result of adjusting t-StatusProhibit also has a high standard deviation as seen in **Table 9**. Here delaying STATUS messages that could recover lost data or increasing the rate of STATUS messages, removing opportunities to transmit data, negatively impact the average throughput and the standard deviation seen. The results of both the t-StatusProhibit are also shown in **Figure 38**.

In **Figure 38**, as the setting on t-StatusProhibit increases the average throughput tends to decrease. Like with the setting of t-Reordering this is not always true as some higher settings of t-StatusProhibit produces higher throughputs. Unlike the tests with t-Reordering there is one setting that produces a noticeable improvement in throughput. In the following tests with varied loss rates the timer t-StatusPrhobit is set to 75 ms.

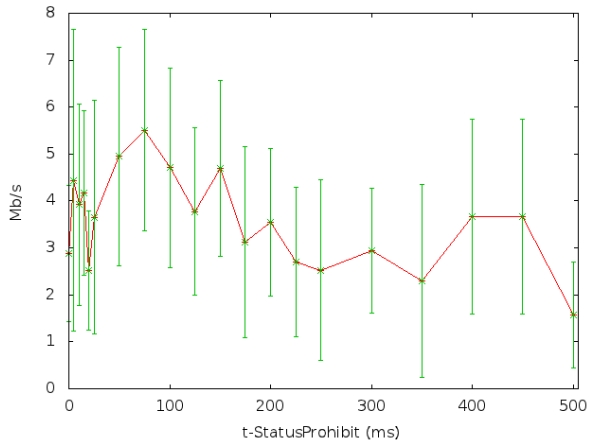


Figure 38: AM t-StatusProhibit

Figure 39: Average FTP throughput for t-StatusProhibit.

Table 10: FTP results AM multiple loss rates

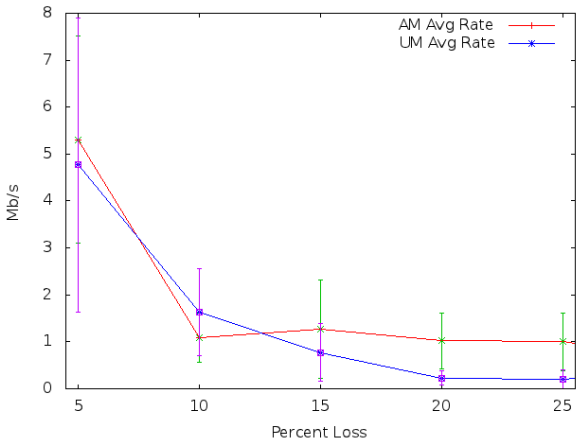
Loss Rate	Average Mb/s	Standard Dev Mb/s
5	5.3	2.2
10	1.1	0.5
15	1.3	1.0
20	1.0	0.6
25	1.0	0.6
30	0.6	0.6
35	0.6	0.4
40	0.7	0.5
45	0.3	0.2
50	0.2	0.1

5.4.2 FTP with AM vs UM with varied loss

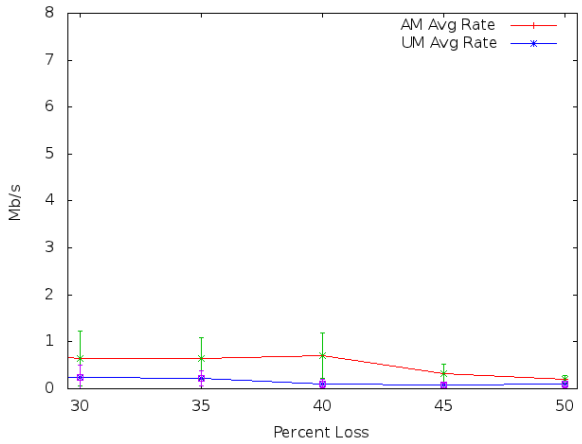
From the results in the previous test the timer t-Reordering was set to 50 ms and t-StatusProhibit was set to 75 ms. The Gilbert-Elliot model with input gap lengths of up to 30 produced fading trace files for 5, 10, 15, 20 and 25%. After this getting the loss percentage correct required larger gap lengths for input data. Like with the VoIP test the maximum possible gap length was increased to 100. From this the remaining loss sets generated were 30, 35, 40, 45 and 50%.

When the wireless loss percentage is small (5 - 10%) the MAC layer recovers most of the lost data. As a result UM can outperform AM in some cases. In **Figure 40a** the average performance for UM at 10% loss is better than AM and at 5% loss the standard deviation for the UM test does show throughputs higher than the AM standard deviation. If the wireless loss was shaped differently, like if more MAC layer retransmissions failed due to larger gap lengths then AM could perform better. As the wireless loss rate increases AM shows higher average throughput than UM. If a particular radio environment was known to have very short wireless loss gaps and a small total percentage of loss then UM could be a better choice. However, in most cases AM produces a higher throughput. The results in **Figure 40** are also shown in **Tables 10** and **11**.

For UM, shown in **Figure 40** and **Table 11**, when the loss percentage is 45 or 50%



(a) Input gap lengths maximum of 30.



(b) Input gap lengths maximum of 100.

Figure 40: Average FTP throughput for t-Reordering = 50 ms and t-StatusProhibit = 75 ms

Table 11: FTP results UM multiple loss rates

Loss Rate	Average Mb/s	Standard Dev Mb/s
5	4.8	3.1
10	1.6	0.9
15	0.8	0.6
20	0.2	0.1
25	0.2	0.2
30	0.2	0.3
35	0.2	0.2
40	0.1	0.1
45	0.1	0.0
50	0.1	0.1

very little data is received. For example, when the loss rate is 45% the data is only received for about 2 seconds out of the 10 seconds that sender was transmitting TCP data. The other data sent was lost before the congestion window had a chance to leave TCP slow start. This is why the standard deviation for UM at 45% loss is 0, there was just not enough data received to see variation.

The UE running AM at 45% loss on the other hand receives data for every second of the simulation thanks to the RLC layer retransmissions. The overall TCP throughput is low with an average of 0.3 Mb/s due to all the losses but it does receive more data than UM. Other than the average throughput being lower the fact that UM can go for seconds without receiving any data suggests that application layer connections could experience socket and connection timeouts. AM on the other hand manages to keep at least some data arriving to keep connection and socket timeouts from occurring.

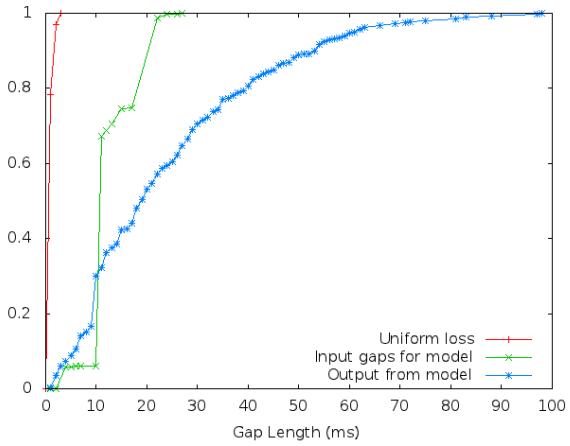


Figure 41: CDF of loss gap lengths.

5.5 MPEG Simulations

In this section we look at how bursty loss events combined with modification to RLC layer settings impacts the MPEG application. First the t-Reordering and t-StatusProhibit timers are adjusted to find settings that provide better performance for MPEG. Then with these timers fixed the loss rate of the wireless link is adjusted to find if MPEG works better under AM or UM.

5.5.1 MPEG with varied RLC settings

Here we setup a fading trace file using the Gilbert-Elliot model. Again we set the loss rate at 10% and randomly select the input example gap lengths. When in the good state in the model there is a 0.879% chance of transitioning to the bad state. When in the bad state there is a 8.783% chance of transitioning to the good state. After running the Gilbert-Ellilot model with these state transitions the fading trace file produced an error rate of 9.871%. The CDF for the gap lengths is shown in **Figure 41**.

As mentioned in **Section 4** the original goal was to run the MPEG video application for 750 simulation seconds. However in the process of doing this some problems occurred. As the loss rate went up the simulation running under AM started to throw errors. Specifically

a test in the LTE simulation module stopped the simulation because a set piece of data in some of the packet headers was not valid. This indicated to the simulator that some error occurred while writing, reading or transporting data. After examining the code it appears that a high load on AM from sending large amounts of data (the video) compared to the amount of available bandwidth (transport block size) along with the need for many retransmissions exposed this problem. Unfortunately, there was not enough time to fully debug and fix this issue. This bug did not impact the other simulations run as the test in the simulator code catches and ends any simulation that experiences the error. This error only occurred after about 450 seconds of simulation time. To avoid this error the amount of time to run the MPEG video simulation was set to 300 seconds.

The following tests show the average delay of receiving the MPEG frames. For UM we also report the percent of frames lost. A frame loss is defined as a frame having any of its UDP packets lost or any frame whose dependent frame is lost. As mentioned earlier MPEG has I, P and B frames where P frames are dependent on the previous I or P frame and B frames are dependent on the past or previous I or P frame. The sample trace file for the video sends frames in the order I, P, B, B, P, B, B, P, B, B and then repeats. Any B frame loss is just that frame. Any I or P frame loss causes losses for all following frames up to the next I or B frame preceding an arrived I frame.

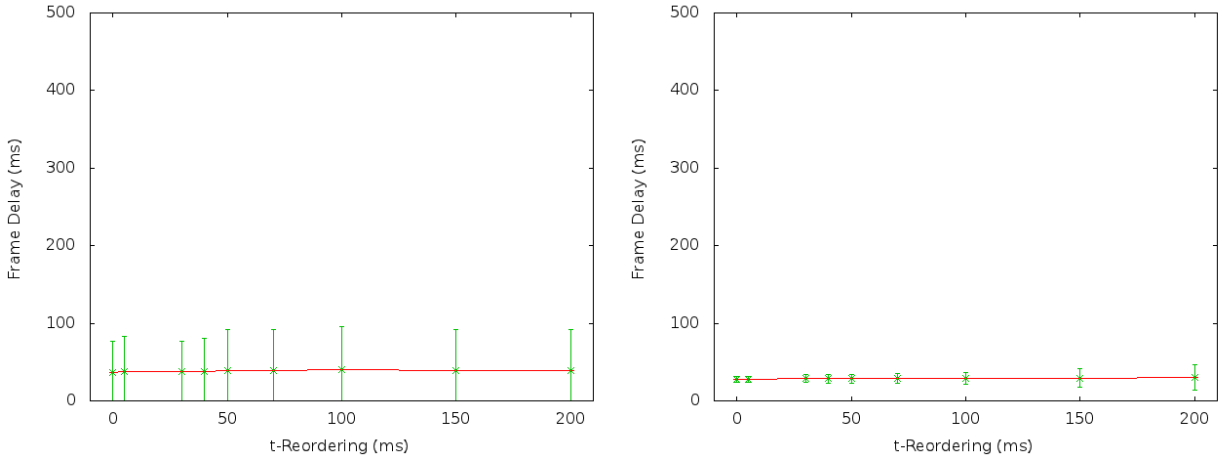
Table 12: MPEG frames lost with UM

t-Reordering (ms)	Percent Frames Loss
0	18.6
5	18.0
30	5.0
40	5.0
50	5.0
70	5.0
100	5.0
150	5.0
200	5.0

5.5.1.1 t-Reordering

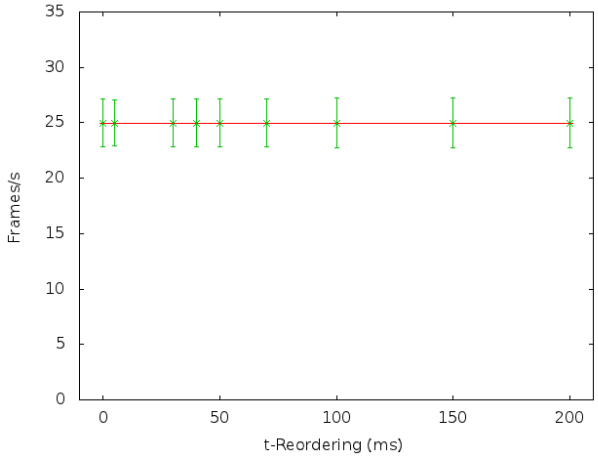
The results of adjusting t-Reordering are shown in **Figure 42**. This figure shows the average delay and frame rates along with their standard deviations for the MPEG frames. Due to the longer amount of time to run these simulations a smaller subset of t-Reordering values are tested. The average delay for both AM and UM are nearly identical, in the range of 35 - 40 ms. The average frame rate for AM remains 25 frames/s for all t-Reordering values. With UM the frame rate is lower when t-Reordering is low enough that some MAC layer retransmissions fail. The AM retransmissions result in a higher standard deviation as seen in **Figure 42a**. In terms of delay adjusting t-Reordering has little impact.

While UM has a lower standard deviation on frame delay there is the issue of lost frames. **Table 12** shows the percent of lost frames for each of the t-Reordering settings. Like the VoIP and FTP applications, if t-Reordering is set low (less than 30 ms) then the MAC layer cannot recover as much data. Here the lost UDP packets, the frames they are in and the frames dependent on them create a high of 18.6% of frames lost when t-Reordering is 0 ms. When t-Reordering is set to 30 ms or higher, the MAC layer has a chance to recover the lost data. The loss percent for this test is then only 5%. Since the setting of t-Reordering has little impact on delay and a setting of 30 ms or higher improves the percent of lost packets for UM, the tests with multiple loss percentages use t-Reordering set to 30ms.

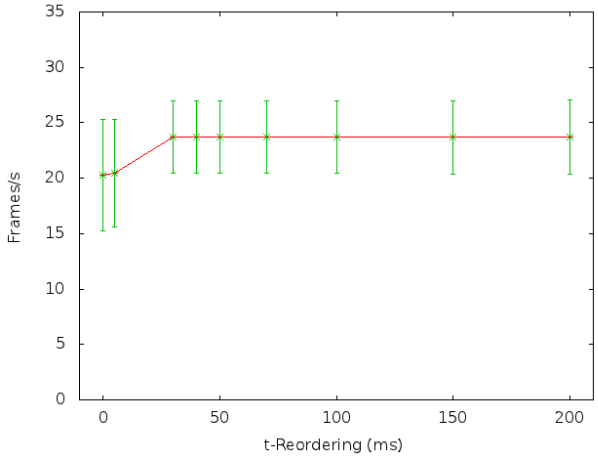


(a) Frame delay AM t-Reordering.

(b) Frame delay UM t-Reordering.

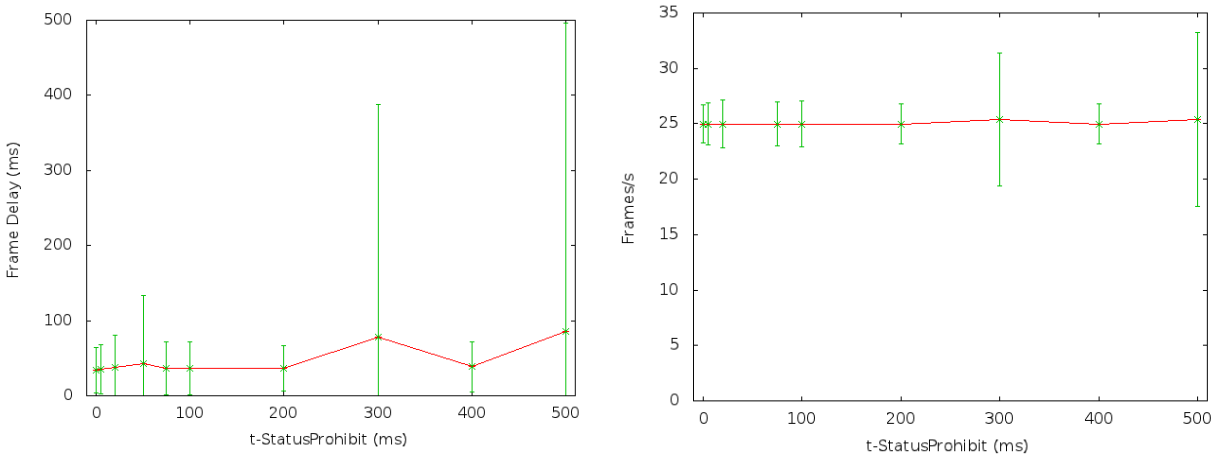


(c) Frame rate AM t-Reordering.



(d) Frame rate UM t-Reordering.

Figure 42: Average delay and frame rates for MPEG with varied t-Reordering.



(a) Frame delay t-StatusProhibit.

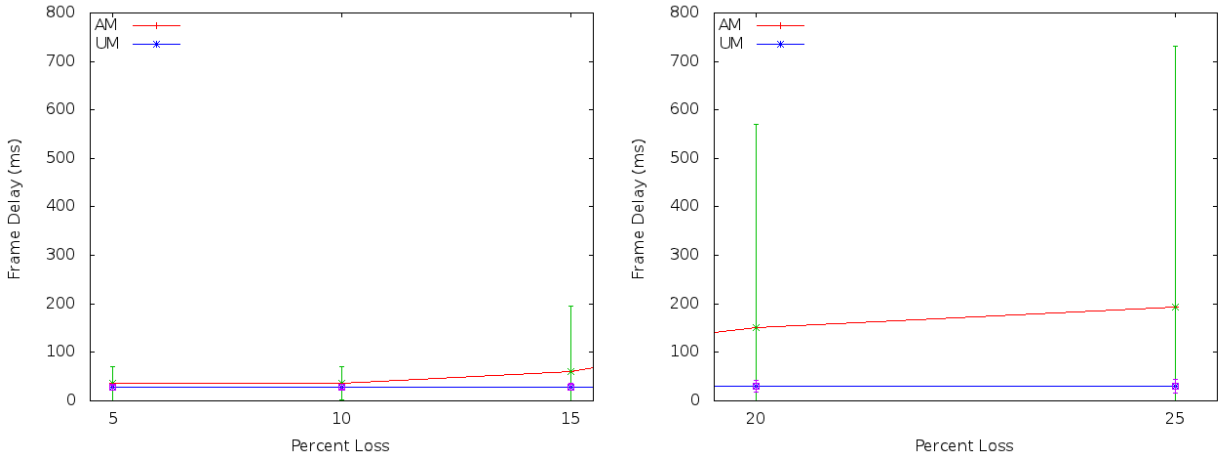
(b) Frame rate t-StatusProhibit.

Figure 43: Average delay and frame rates for MPEG with varied t-StatusProhibit

5.5.1.2 t-StatusProhibit

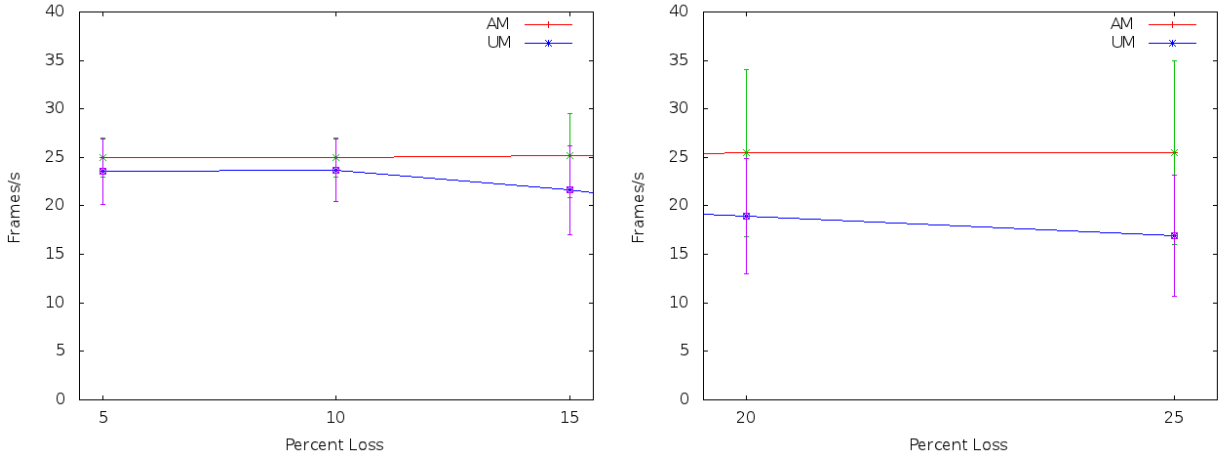
The results of adjusting t-StatusProhibit are shown in **Figure 43**. Adjusting t-StatusProhibit has a much greater impact on both delay and frame rates compared to t-Reordering. The higher the setting, the higher the delay frames experience. The average frame rate also fluctuates. As some time intervals receive fewer frames others get the retransmissions and have more frames. Similar to the tests for VoIP and FTP, setting t-StatusProhibit higher does not always cause the application performance to worsen. As with those applications the exact timing of when a lost packet can be NACKed compared to exactly when the STATUS message is sent can leave some PDUs waiting a long time for retransmissions.

Overall higher values of t-StatusProhibit add extra delay, while very low settings can cause multiple STATUS messages to take up transmission opportunities. A setting of 75 ms is a good balance on these two concerns.



(a) Input gap lengths maximum 30.

(b) Input gap length maximum 100.



(c) Input gap lengths maximum 30.

(d) Input gap length maximum 100.

Figure 44: Average delay and frame rate for t-Reordering = 30 ms and t-StatusProhibit = 75 ms.

5.5.2 MPEG with AM vs UM with varied loss

The video application is run with multiple loss rates. From the results in the previous section, t-Reordering is set to 30 ms and t-StatusProhibit to 75 ms. This test specifically uses the loss rates 5, 10 and 15%. These loss rates use input gap lengths with a maximum of 30 to produce the corresponding loss percent in the output fading loss file. The results of these tests are shown in **Figure 44**.

When the loss rate is low (5 - 10%) there is little difference between running the video over AM or UM. As the loss rate increases, however, AM experiences higher delays due to

Table 13: MPEG frames lost with UM

Percent Loss	Percent Frames Loss
5	5.6
10	5.0
15	13.3
20	24.0
25	31.9

AM retransmissions, while UM experiences lower average frame rates due to lost frames. While the average frame rate for AM remains fairly consistent, the standard deviation grows with the loss rate as some time intervals receive fewer frames that are then recovered in later intervals.

The frame loss experienced in UM is shown in **Table 13**. For low loss rates (5 - 10%) UM maintains low frame loss. As the loss rate increases the percent of frames lost also increases. At 20% wireless loss almost one quarter of all frames are either lost from missing UDP packets or lost due to the loss of dependent frames.

The takeaway from these tests is that for low loss rates (5 - 10%) there is little difference between using AM and UM. As the loss rate increases the user would perceive a degradation in video quality. With AM there would be moments where the video would slow as frames are recovered followed by either the video either fast-forwarding through the received frames to catch up in time or the playback could just drop all of the un-needed, out of time frames received. If UM is used there is little extra delay but more frames lost. For a video carrying a live conversation like a video conference, UM would only have issues from lost frames, while the retransmissions of lost data in AM would also delay frames that are arriving on time but out of order.

6 Conclusion

This work examines how retransmissions in 4G LTE impact VoIP, FTP and video applications. VoIP and video over UDP are sensitive to delay and the amount of data lost in transit. FTP is sensitive to the throughput of data. In LTE, retransmissions occur between the eNodeB and UE at both the MAC and RLC layers. In the RLC layer there are choices for either using retransmissions or not (AM and UM). In addition, the RLC layer has configuration parameters to set how long to wait for the MAC layer to recover data (t-Reordering) and the minimum time to wait in-between sending ACKs and NACKs between the eNodeB and UE.

To test the impact of 4G LTE retransmissions, we used the open source ns-3 packet network simulator, which has a LTE module. This module did not support sending NACKs in RLC AM STATUS messages. We added this capability to the simulator and sent these changes back to the developers of the module. We ran six experiments with two nodes over a range of loss and latency conditions.

In the VoIP simulations, the benefit of extra retransmissions in AM improved call quality up to a 20% loss of data on the wireless link. After this, the added delay of retransmissions was more detrimental to the total lost packets. The values for timers t-Reordering and t-StatusProhibit were set to 30 ms and 50 ms respectively. With t-Reordering at 30 ms the MAC layer has a chance to recover lost data that a lower setting could lose in UM. Higher settings to t-Reordering can delay retransmissions of lost data in AM, although this impact was small. For t-StatusProhibit at 75 ms, a STATUS message of ACKs and NACKs can only be sent at least once every 75 ms. Smaller settings did not improve performance, while higher settings greatly reduced the call quality.

The FTP simulations maintain a higher TCP throughput when using AM as opposed to UM. The additional LTE retransmissions reduce the number of TCP retransmissions required. However, large numbers of TCP RTOs cause great variability in throughputs. This

is especially true when using AM since the TCP layer must wait for the LTE retransmissions without any feedback. The RLC timers t-Reordering and t-StatusProhibit were set to 50 ms and 75 ms respectively. During testing, lower values of t-Reordering produced higher throughput though some lower settings such as 25 - 40 ms produced a much lower throughput when using AM. Similarly t-StatusProhibit also produced higher throughputs for lower settings. However, very low settings of this timer increase the number of STATUS messages, taking away from application data transmission opportunities.

The MPEG video simulations show that UM maintains a lower average delay between 27 and 30 ms with low frame loss compared to AM, with average delays between 35 and 200 ms. However, while UM maintains a lower average delay the resulting lost frames mean that UM has a lower average frame rate. When using AM the lost frames are recovered, but the video experiences a much larger variation in frame rates as losses in one time interval can lead to more frames arriving (due to retransmissions) in another. While UM experiences increasing frame loss with an increase in wireless loss, those frames that arrive without error have little delay while in AM frames that arrive correctly but out of order need to wait for the AM retransmissions. For the VoIP and FTP applications, the settings of t-Reordering and t-StatusProhibit produce lower delays for MPEG video when the timers are set within the range of tens of milliseconds. t-Reordering was set to 30 ms and t-StatusProhibit to 75 ms.

Delay sensitive applications like VoIP and UDP-based video have better performance when run over RLC UM while throughput sensitive applications like FTP perform better with the extra retransmissions of AM. Of the RLC settings examined, the timer t-Reordering is best set at least high enough to allow the MAC layer to recover all the lost data it can. The timer t-StatusProhibit is best set low enough to not adversely delay ACKs and NACKs but not so low that the network spends all of its transmission opportunities sending AM STATUS messages instead of application data. While there are other RLC retransmission settings and other types of applications, these simulations show some of the features network

providers can expect for some of the common applications they carry on their networks.

LTE networks should set the value of the RLC layer timers based on the type of application in use. This is partially possible today through the use of QoS classes. Section 2.1.2 notes that there is one RLC entity for each QoS class. Each RLC entity could set the timers t-Reordering and t-StatusProhibit, along with the choice to use AM or UM, to best handle the traffic they carry. Today traffic is mapped to these classes using traffic direction (downlink or uplink), IP address, port number and transport protocol. This granularity is not sufficient. For instance a VoIP application over UDP and a UDP MPEG video would not necessarily map to different QoS classes. What is needed in LTE is a method for applications to indicate to the LTE network layers what settings they require.

7 Future Work

Future work would in this area includes expanding on the applications, RLC layer settings used in testing and adding more features to the LTE simulator used. The video application tested used UDP packets, making the video like a live conversation setup where delay is more important that lost data. Other videos like television shows and movies where no loss is acceptable would use TCP as the underlying protocol. The default TCP congestion control mechanism used here is New Reno. The ns-3 simulator needs newer methods like BIC, CUBIC and Compound to provide more realistic simulators of what would happen in modern networks.

At the RLC layer there are more settings related to retransmissions that need testing. The timer t-PollRetransmit controls the time the sender waits before re-sending a request for an AM STATUS message. Linked to this setting are the flags pollPDU and pollByte. These flags indicate how many PDUs or bytes a node is allowed to send before requesting the AM STATUS message. These settings impact when AM uses STATUS messages and therefore impacts when PDUs can be ACKed or NACKed.

The LTE simulator in ns-3 is still under active development and could use more features that are in LTE. The RLC layer keeps a maximum retransmission count for PDUs when using AM. Currently the simulator does not use this information. Additions are needed so that the proper RRC layer control mechanism operate when a PDU is retransmitted more than the AM maximum. Also while the simulator currently supports packing multiple IP frames into a single RLC PDU it does not yet support the ability to segment these frames (encased inside PDCP layer PDUs across multiple RLC layer PDUs). This feature would reduced empty portions of transport blocks transmitted that could have fit additional data.

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A Acronyms

This section lists the acronyms used in this paper.

ACK acknowledgment 10

AM acknowledged mode 2

AMC adaptive modulation and coding 49

API application program interface 119

BLER block level error rate 5

CBR constant bit rate 44

CDF cumulative distribution function 68

cwnd congestion control window 25

CQI channel quality indicator vi

DCE direct code execution 44

EARFCN E-UTRA absolute radio frequency channel number 48

eNodeB enhanced node b 4

EPC evolved packet core 4

E-UTRA evolved universal mobile telecommunications system terrestrial radio access . . . 4

FDD frequency division duplexing 19

FEC forward error correction 10

FTP file transfer protocol iv

GBR guaranteed bit rate 18

GPRS general packet radio service 4

GSM global system for mobile communications 4

HARQ hybrid automatic repeat request iv

HTTP hypertex transfer protocol 34

ICMP Internet control message protocol 74

IP Internet protocol 4

ITU International Telecommunication Union 27

LTE Long Term Evolution iv

MAC Media Access Control 2

MCS modulation and coding scheme x

MIMO multiple input multiple output 9

MOS mean opinion score 28

MPEG movie picture experts group v

MTU maximum transmission unit 46

TCP transmission control protocol 25

TDD time division duplexing 19

TFT traffic flow template 17

TM transparent mode 10

TTI transmission time interval 5

UDP user datagram protocol 26

UE user equipment 4

UM unacknowledged mode 10

UMTS niversal Mobile Telecommunications System 4

VoIP Voice over IP iv

B CQI

The SINR of the UE has a great impact on the rate at which it can receive data and how much loss it experiences. While we cannot know what the eNodeB sends to a UE like a phone without a spectrum analyzer we can know what modulation encoding the phone asks for. In version 4.2 of the Android application program interface (API) supports getting information on LTE, including what CQI value the device reports to the tower. To get an idea of what real life signal quality a user might expect we recorded the CQI reported by a Samsung Galaxy Nexus smart phone running Android API version 4.2.2. We checked for this information every 2 minutes but we did not record data when the phone had no LTE connection. Reasons for missing connections included being connected on a 3G cellular network or having the device in an area where there was no signal.

We collected this data in Massachusetts. Specifically in the areas of Bristol, Middlesex, Norfolk, Plymouth and Worcester counties. You can see a subset of the data locations in **Figure 45** where the push-pins show some of the CQI values measured and their approximate location. During this data collection the phone connected to 81 distinct LTE cells (identified by cell ID information provided through the Android API).

The phone used to gather this data was carried and used as a normal phone with no special attention to orientation or placement. The phone was in some cases left on tables, carried by hand or in pocket. The phone was used as a normal smart phone, which included making phone calls and applications.

The result of these measurements are shown in **Figure 46**. **Table 1** shows that CQI values of 1 - 6 indicate QPSK, 7 - 9 is 16QAM and 10 - 15 is 64QAM. **Figure 46a** shows CQI measurements categorized by urban and suburban areas, where the phone was mostly stationary or moving at a walking pace, and traveling roads, where the phone was in a car traveling at speeds from 40 - 65 miles per hour. In both urban and suburban areas the phone mostly requested a CQI value of 8. While traveling on roads there was a greater variety of

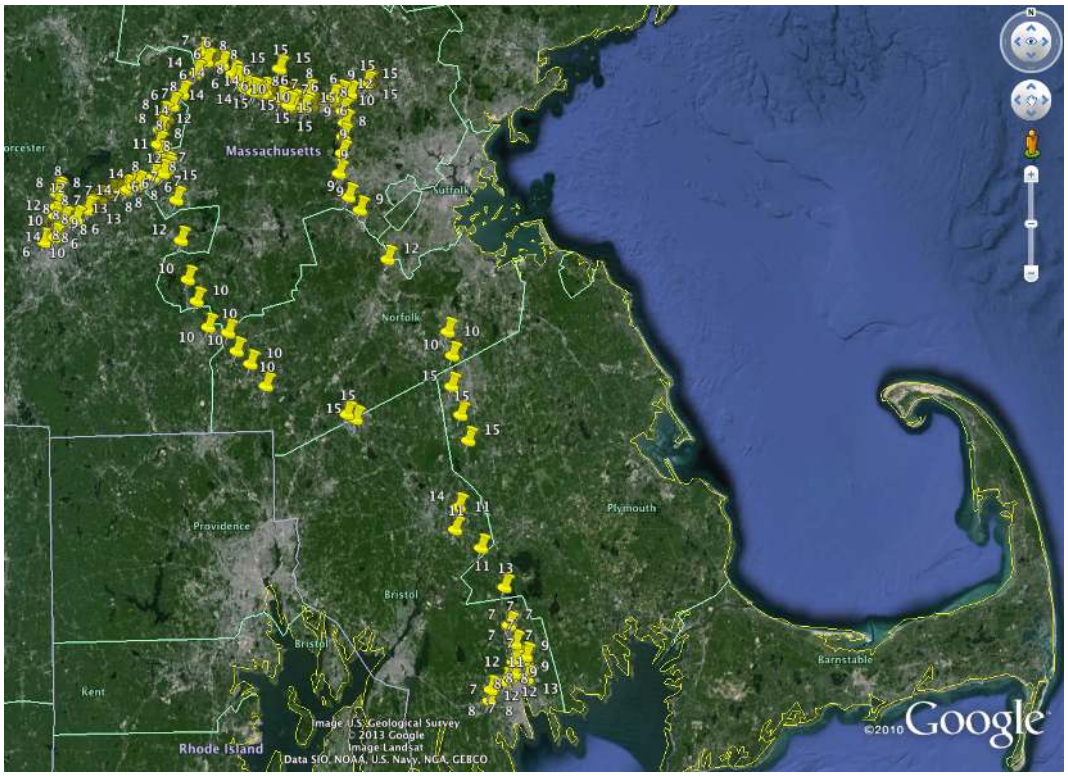
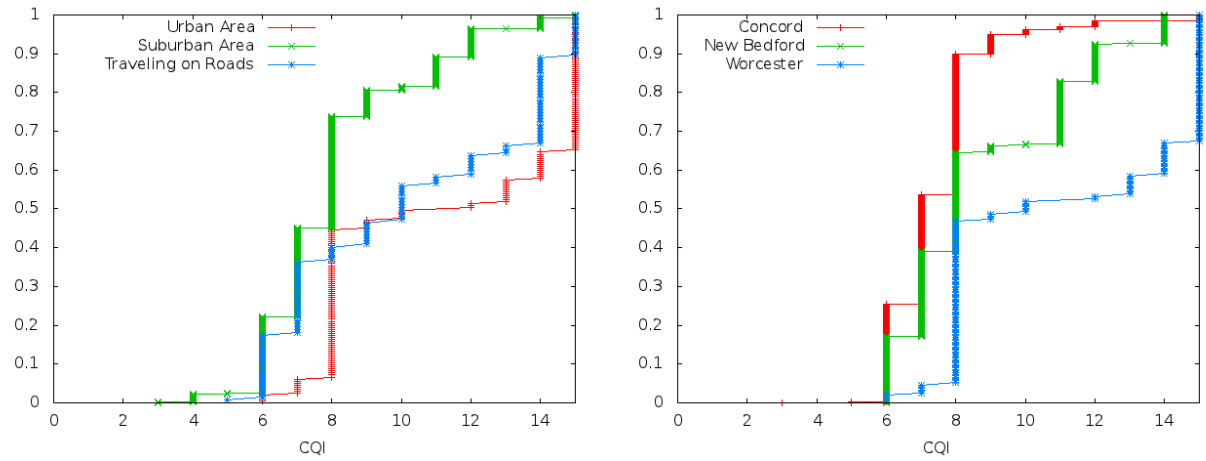


Figure 45: Map of where CQI measurements were taken.

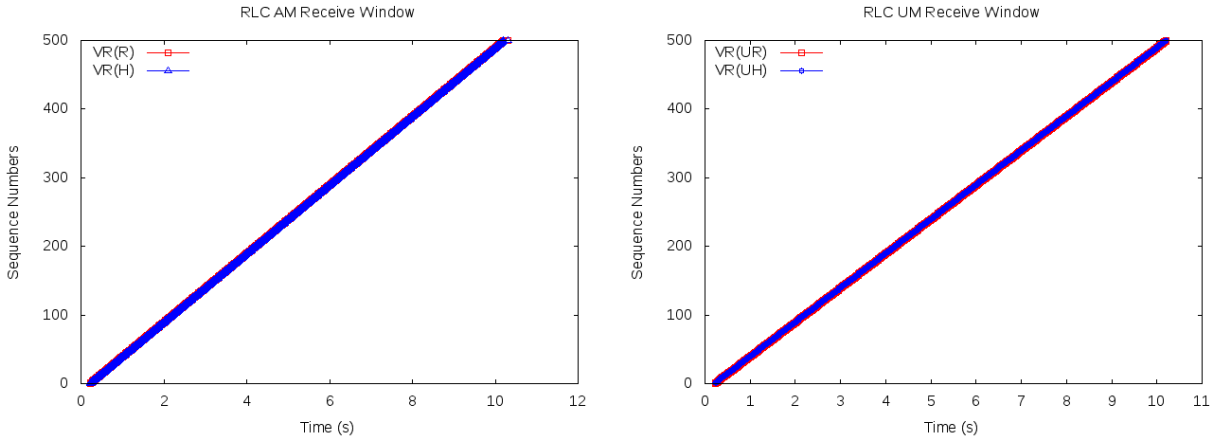
CQI values requested. **Figure 46b** lists the CQIs for the three cities and towns with the most recordings. Again, in all three locations, the phone requested a CQI of 8.



(a) CQIs for urban, suburban and traveling roads. (b) CQIs for particular cities and towns.

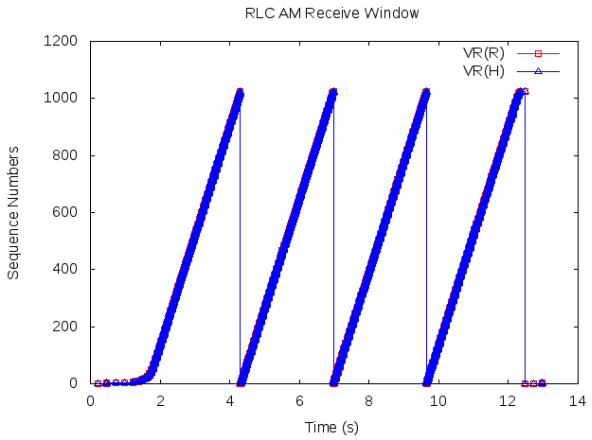
Figure 46: CQI measurements

C Additional Figures

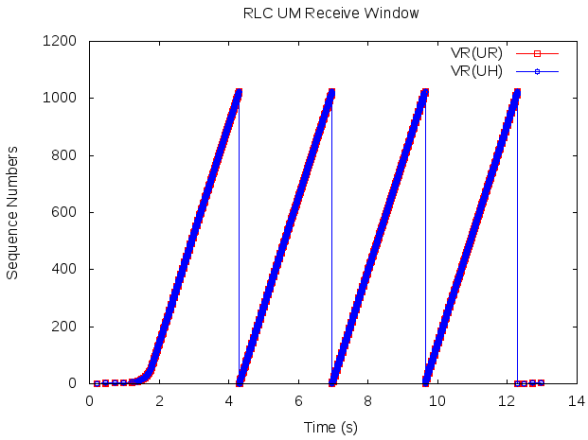


(a) Baseline RLC AM Receive Window (b) Baseline RLC UM Receive Window

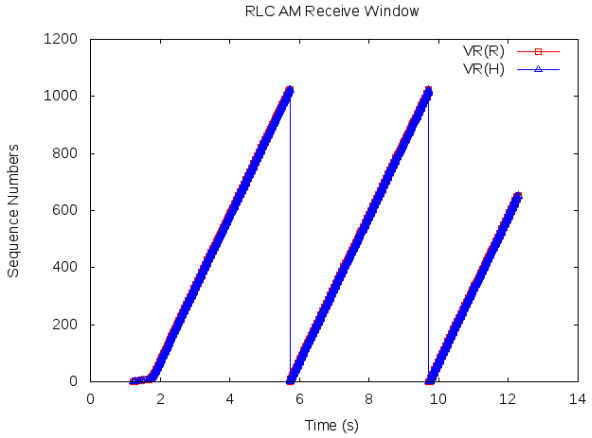
Figure 47: Baseline RLC Receive Window with VoIP



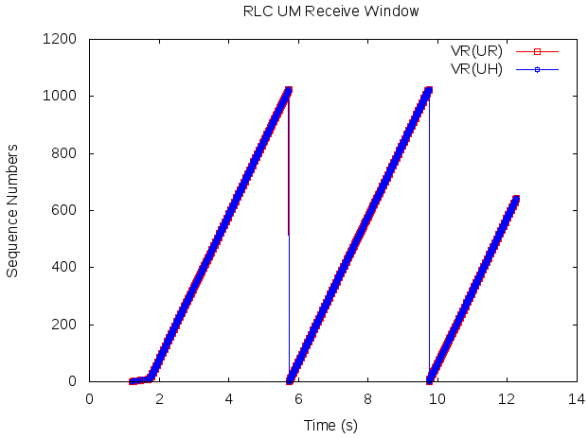
(a) RLC AM Receive Window for UE



(b) RLC UM Receive Window for UE.



(c) RLC AM Receive Window for eNodeB.



(d) RLC UM Receive Window for eNodeB

Figure 48: Baseline FTP RLC Receive Windows.

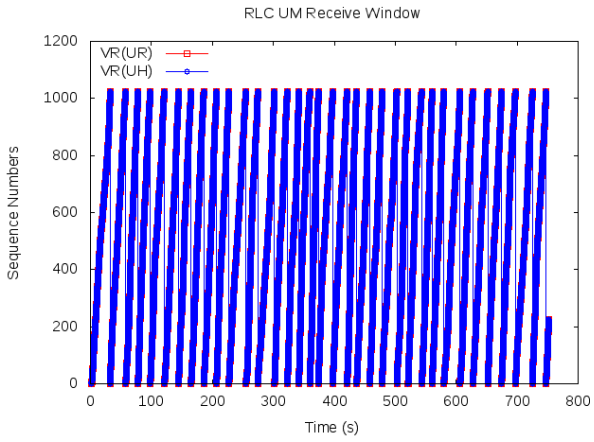
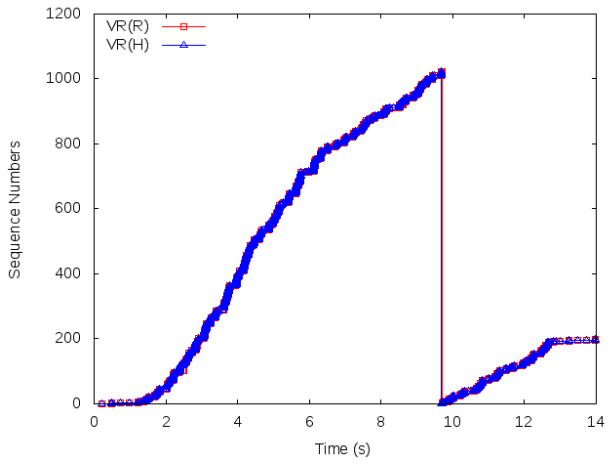
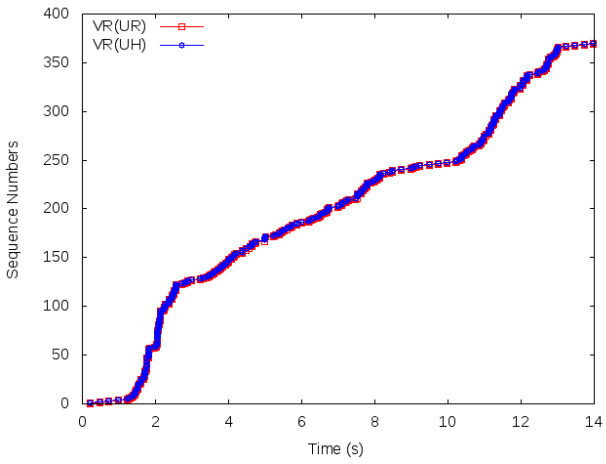


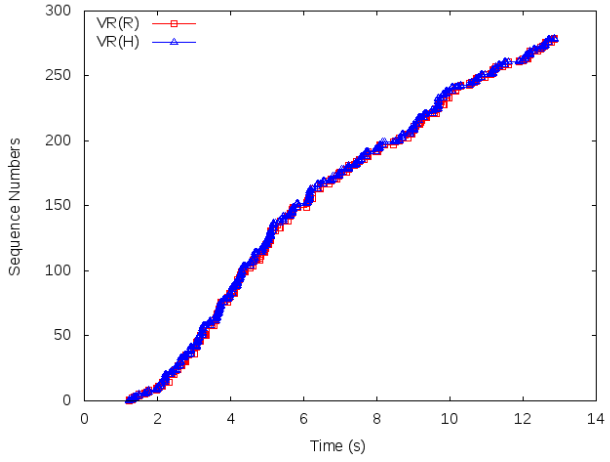
Figure 49: Baseline RLC UM receive window for Video.



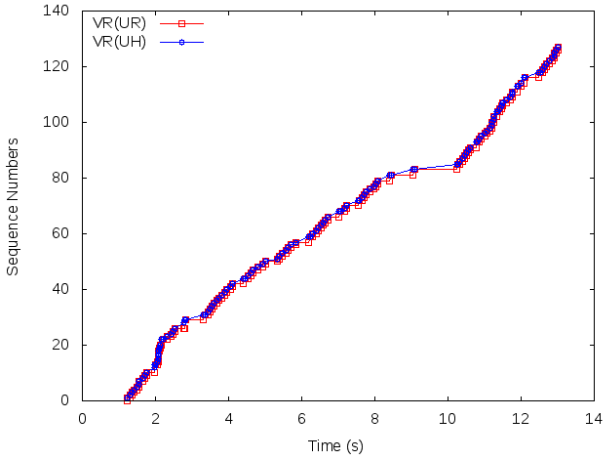
(a) UE RLC AM receive window.



(b) UE RLC UM receive window.



(c) eNodeB RLC AM receive window.



(d) eNodeB RLC UM receive window.

Figure 50: RLC performance for FTP with a uniform 25% loss rate.