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RLC Effects on Throughput in a Mobile System

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Abstract

Today wireless data communication is growing rapidly. In particular, multimedia and Internet communication by wireless systems are the most popular services. Also, the architecture of wireless networks is varying to provide such services better. In this paper, we review UTRAN (UMTS Terrestrial Radio Access Network) focusing on the RLC (Radio Link Control) sub-layer, its error recovery mechanisms and transport channels when connected to Internet. Besides, we consider data flows based on the acknowledgement mode RLC, the transparent mode MAC and simulate the effects of RLC protocol and erroneous MAC PDUs and delays, on the throughput. Finally, we evaluate the throughput versus MAC PDU retransmission times as well as the throughput efficiency versus MAC PDU retransmission times and data block lengths.

Keywords: UMTS, RLC, Internet, error recovery.

I. Introduction

UMTS (Universal Mobile Telecommunications System) is compatible with the OSI layered reference model for both UE (User Equipment) and UTRAN (Universal Terrestrial Radio Access Network) parts. UTRAN includes Node B and RNC (Radio Network Controller). The protocol architecture is subdivided into the control plane, responsible for the transmission of signaling information, and the user plane, responsible for the transmission of user data. In particular, three protocol layers are considered in the UTRAN: the Physical layer (Ll), the Data Link Layer (L2) and the Network Layer (L3). In turn, Layer 2 is split into two sub-layers, namely the Radio Link Control (RLC) and the Medium Access Control (MAC). Two other sub-layers included in Layer 2 considered in the user plane are the Packet Data Convergence Protocol (PDCP), responsible for header compression of data packets in the packet switch services, and the Broadcast/Multicast Control Protocol (BMC) that adapts the transmission for broadcast services [1, 2].

Wireless radio channels are affected by shadowing and fading. Shadowing and fading, in turn, are caused by the mobility of the subscribers. On the other hand, development of the data networks and their connections to the mobile network has introduced new attractive services in addition to customary voice services [3]. Data services include data file, video and voice transfer within the mobile network or over the Internet connection.

Internet traffic is a combination of the above traffic services and mostly uses the TCP/IP protocol suite. TCP provides end-to-end recovery and flow control. The Link Layer protocols in cellular systems operate below the TCP Layer and have complex interactions with it. TCP has been designed and tuned for networks in which segment losses and corruptions are mainly due to congestion. In wireless systems most of the errors are due to lossy media because in the wireless channels the main reason for the packet loss may be due to the high Bit Error Rate (BER) and not to the network congestion [4]. So, low efficiency of the TCP in a wireless channel is a direct result of the fact that the TCP misinterprets the packet loss resulting from high channel error rate or resulting from congestion.

In order to enhance QoS seen by the TCP layer on a wireless link, a radio link control (RLC) is generally introduced at the link layer. Typically, the RLC uses an ARQ error recovery mechanism to improve the QoS [5], [7], [9]. Besides, studies have been made on the effects of certain factors on throughput, for instance FEC (Frame Error Check) effects [6, 8], retransmission time effects CDMA2000 system [10], and BER effects in UMTS [11].

Layer 2 offers to the upper layers a service by means of the Traffic Radio Bearers (TRBs) and Signaling Radio Bearers (SRBs). The former provide the transmission of user data while the latter are intended to transfer control information that can be used in other layers. The information flow associated to a TRB or a SRB is mapped to the different types of channels depending on the position in the layered protocol architecture. They are the logical, transport and physical channels [12-14]. This channel differentiation enables a flexible architecture to allow the provision of the services by making use of different configurations of the radio interface.

Logical channels allow communication between the RLC and MAC layers and are characterized by the type of information that is being transferred across these layers. As a result, there are two logical channels for the transfer of user traffic and also five logical channels for the transfer of control information which can be either dedicated to the specific users or common to a set or to all of them. Logical channels are mapped onto transport channels by the MAC layer. Transport channels are defined between the MAC and PHY (Physical) layers, in particular, how the information from logical channels should be adapted to get access to the radio transmission medium.

In summary some functions of MAC are [15]; Hybrid Automatic Repeat Request (HARQ) functionality for HS-DSCH and E-DCH transmission, In-sequence delivery and assembly /disassembly of higher layer PDUs (Protocol Data Unit) on HS-DSCH, In-sequence delivery and assembly/disassembly of higher layer PDUs on E-DCH. Also, some functions of RLC are; Segmentation and reassembly, Concatenation, Padding, Transfer of user data, Error correction, In-sequence delivery of upper layer PDUs (Protocol Data Units), Duplicate Detection, Flow control, Sequence number check, Protocol error detection and recovery, Ciphering, SDU (Signaling Data Units) discard, Out of sequence SDU delivery, Duplication Avoidance and reordering. By combining different types of the above RLC functions three different services are provided, namely, transparent mode data transfer service (TM), unacknowledged mode data transfer service (UM) and acknowledged mode data transfer service (AM).

In section II of this paper we survey the general concepts of the transport channels. In section III we review error recovery mechanisms. In section IV we define a system model for Layer 2 to study delays and retransmission times and survey UMTS throughput as functions of retransmission times and transport block sizes (TBS). In section five we pay attention to simulation of the present system model. Finally we present some conclusions.

II. General Concepts about the Transport Channels

These channels are subdivided into dedicated transport channels (DCH and E-DCH) and common transport channels (BCH, PCH, RACH, FACH, HS-DSCH). All Transport Channels are defined as unidirectional (i.e. only uplink or downlink). This means that a UE can have simultaneously (depending on the services and the state of the UE) one or several transport channels in the downlink and one or more other transport channels in the uplink.

Some DCHs can be multiplexed and mapped onto one or several Dedicated Physical Channels (DPCH) on the physical layer. A DPCH consists of two parts, Dedicated Physical Control Channel (DPCCH) and Dedicated Physical Data Channel (DPDCH). A DPCCH carries control information which is generated internally on L1. A DPDCH carries the encoded bits of the DCH transport channels (Fig 1). Table 1 illustrates how the bit mapping is done in normal transmission mode in this layer. There are several different slot formats defined with different split of data and control bits. At establishment of a downlink DPCH, one of the permitted slot formats is selected and applied. MAC delivers Transport Block or a Transport Block Set to the physical layer every Transmission Time Interval (TTI). TTI is always a multiple of the minimum interleaving period (e.g. 10ms, the length of one radio frame). A TTI can be 10, 20, 40 or 80 ms in duration.

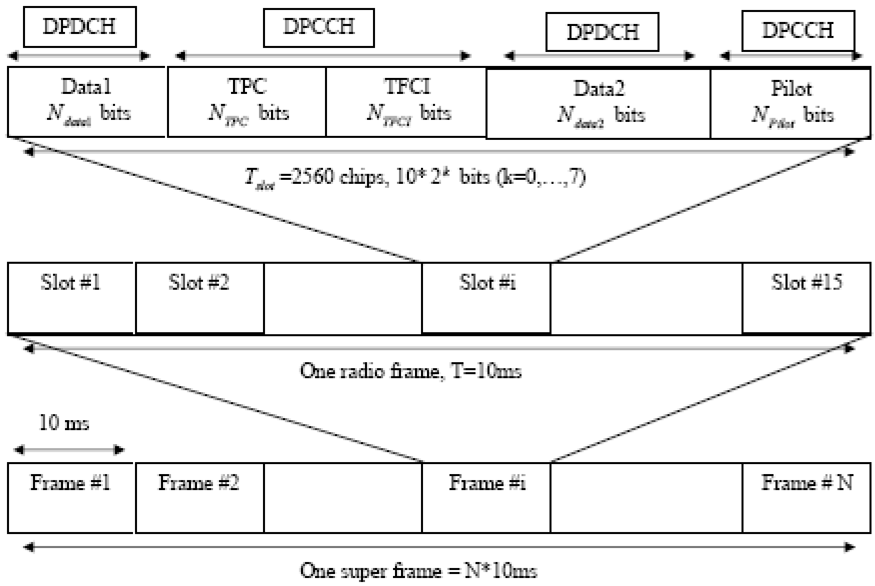


Fig. 1 Frame structure for downlink Dedicated Physical Channels DPCH [16]

Table 1. Downlink DPCH slot formats in normal transmission mode [16]

Slot Format # i	Channel Bit Rate (kbps)	Channel Symbol Rate (kbps)	SF	Bits/Slot	DPDCH Bits/Slot		DPCCH Bits/Slot			Transmitted slots per radio frame N_{Tr}
					N_{Data1}	N_{Data2}	N_{TPC}	N_{TFCI}	N_{Pilot}	
0	15	7.5	512	10	0	4	2	0	4	15
1	15	7.5	512	10	0	2	2	2	4	15
2	30	15	256	20	2	14	2	0	2	15
3	30	15	256	20	2	12	2	2	2	15
4	30	15	256	20	2	12	2	0	4	15
5	30	15	256	20	2	10	2	2	4	15
6	30	15	256	20	2	8	2	0	8	15
7	30	15	256	20	2	6	2	2	8	15
8	60	30	128	40	2	28	2	0	4	15
9	60	30	128	40	6	26	2	2	4	15
10	60	30	128	40	6	24	2	0	8	15
11	60	30	128	40	6	22	2	2	8	15
12	120	60	64	80	12	48	4	8	8	15
13	240	120	32	160	28	112	4	8	8	15
14	480	240	16	320	56	232	8	8	16	15
15	960	480	8	640	120	488	8	8	16	15
16	1920	960	4	1280	240	1000	8	8	16	15

Fig 2 shows an example in which at a certain time instances Transport Blocks are exchanged between MAC and L1 via two parallel transport channels DCH1 and DCH2. Transmission Time Interval, i.e. the time between consecutive deliveries of data between MAC and L1, is also illustrated in that figure. Transport Block (equal to a MAC PDU) is the basic unit exchanged between L1 and MAC, for L1 processing. Transport Block Set is defined as a set of Transport Blocks which are exchanged between L1 and MAC at the same time instance using the same transport channel. Data on each transport channel is organized in Transport Blocks. Depending on the requested QoS variable numbers of transport blocks with variable lengths can be transmitted in each TTI i.e. one or more TBs can be inserted into one TTS. Transport Block Size is defined as the number of bits in a Transport Block Set. In the Fig 2(a) TTI=20ms and transport block length varies TTI by TTI. In the Fig 2(b) TTI=10ms and both the length and the number of TBs varies.

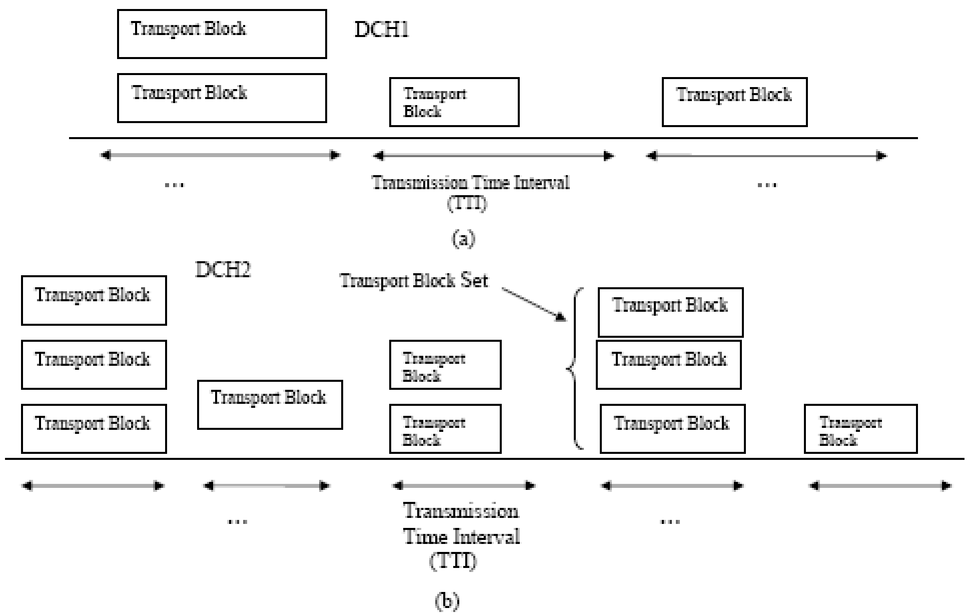


Fig. 2 Exchange of MAC PDU between MAC and L1 [17]

III. Error Recovery Mechanisms of Layer 2

The data link layer is Layer 2 of the seven-layer OSI model as well as of the five-layer TCP/IP reference model. It responds to service requests from the network layer and issues service requests to the physical layer. The data link layer is split into MAC and LLC sub-layers. The uppermost sub-layer is the Logical Link Control (LLC) one. This sub-layer multiplexes protocols running at top of the data link layer, and optionally provides flow control, acknowledgment, and error recovery. Media Access Control (MAC) is blow LLC. This refers to the sub-layer that determines who is allowed to access the media at any time (usually CSMA/CD). The Media Access Control sub-layer also determines where a frame of data ends and the next frame starts.

Error detection is the ability to detect errors caused by noise or other impairments during transmission from the transmitter to the receiver. Error correction has an additional feature that enables identification and correction of the errors. When a sender transmits a frame, it might be corrupted or lost. The data link layer at destination checks the received frame for error and uses an ARQ mechanism to send back its status to the sender. There are two ways for this mechanism; Stop and Wait ARQ and Continuous ARQ [18]. In the Stop and Wait ARQ mechanism the sender sets a timer to a definite time after sending a frame and waits for receiving an ACK message for that duration. If this message doesn’t arrive at the sender or if the timer times out, the frame will be retransmitted. This method can be implemented in Half-Duplex communication. In the Continuous ARQ mechanism the transmitter sends frames continuously and the receiver sends back ACK or NACK messages from a distinct channel (Full Duplex). The sending process continues by a number of frames specified by a window *size* even without receiving an ACK packet from the receiver. The continuous ARQ mechanism is implemented in two forms: Go-Back-N ARQ and Selective Repeat ARQ. In the first form there is a buffer at the sender in which a copy of the transmitted frames resides and will not be deleted before they are not received correctly. When a NACK(n) is received at the source, the frames will be retransmitted one after the other from the frame *n*. In the second form, when some of the received frames are not correct the receiver requests the sender to retransmit only the unsent frames. So, the receiver should be capable of reordering the received frames.

IV. System Model

Data flow mechanisms through UMTS Layer 2 are characterized by the applied data transfer modes in RLC (acknowledged, unacknowledged and transparent transmission) in combination with the data transfer type on MAC, i.e. whether or not a MAC header is required. The case where no MAC header is required is referred to as "transparent" MAC transmission. Acknowledged and unacknowledged RLC transmission modes both require a RLC header. In unacknowledged transmission, only one type of unacknowledged data PDU is exchanged between peer RLC entities. In acknowledged transmission, both data PDUs and control PDUs are exchanged between peer RLC entities. This reduces the throughput but helps the link not to be disconnected in acknowledged transmission mode relative to unacknowledged transmission mode.

There are some different combinations of data flows in Layer 2 as: transparent RLC with transparent and non-transparent MAC transmission, non-transparent RLC with transparent and non-transparent MAC transmission [13]. For lack of space we only illustrate non-transparent RLC with transparent MAC transmission in Fig 3. A number of MAC PDUs shown in the figures may comprise a transport block set. Note, however that in all cases a transport block set must not necessarily match with only one RLC SDU. The span of a transport block set can be smaller or larger than an RLC SDU (Fig. 3). The received PDUs can be reassembled by simply concatenating all RLC PDUs included in a transport block set as implied by the used transport format (TF).

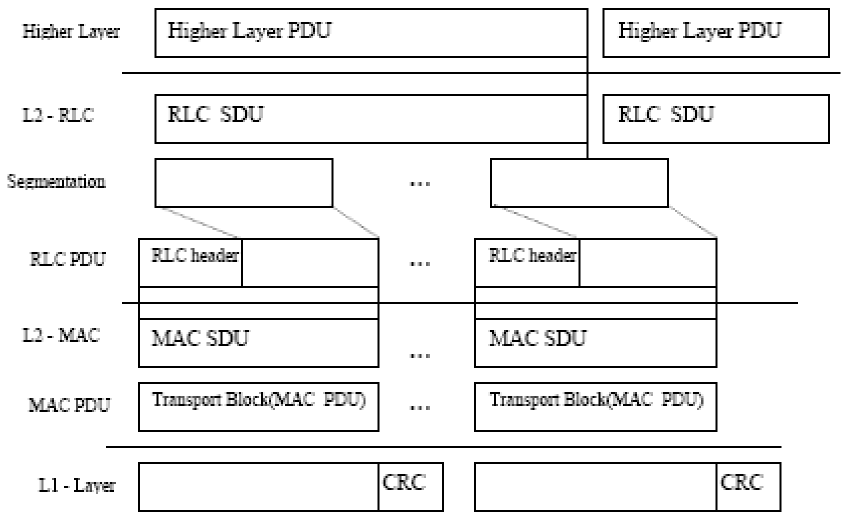
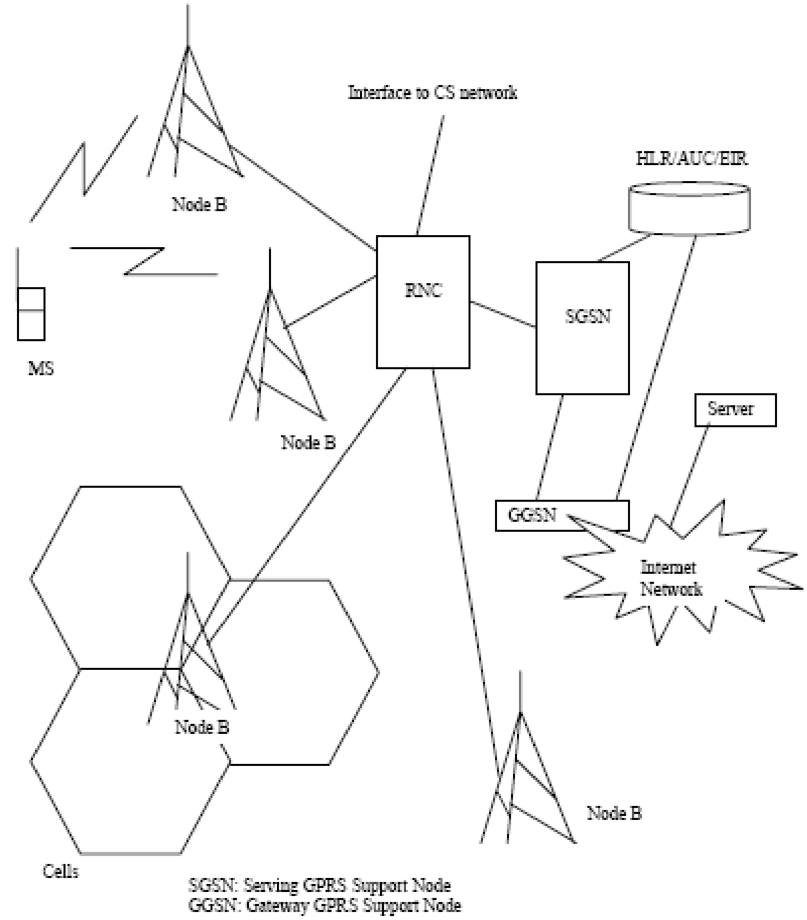


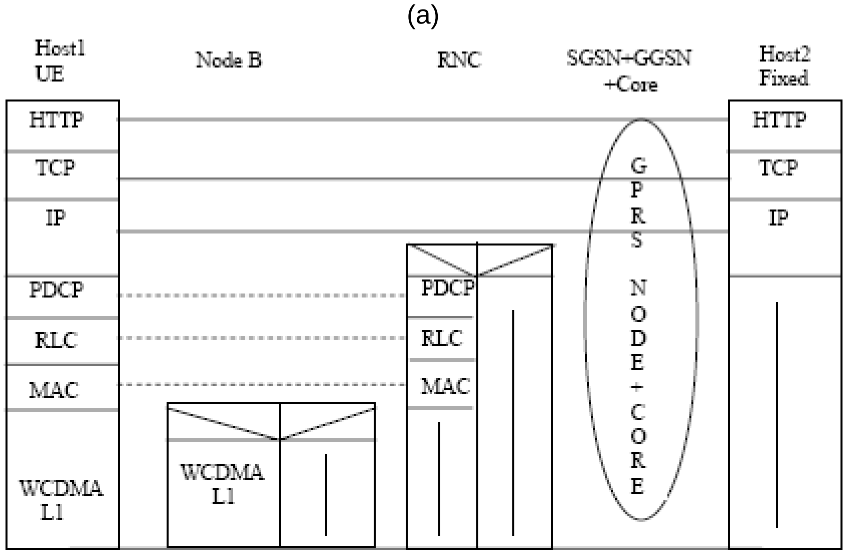
Fig. 3 Data flow for non-transparent RLC and transparent MAC [13]

Now we consider a TCP connection between two hosts such that the first link on the end-to-end path from the sender to the receiver is a wireless radio link and the second link is a wired link and connected to a server (Fig.4). In Fig 4(a) a Web browsing user has attempted to connect to a server in a public Internet network and intends to download a file from the remote server. Such a scenario is common in mobile communications. The protocol stack on the way is illustrated in the Fig 4(b). We want to evaluate effects of PDU retransmission due to packet error on the wireless link performance. We consider UE, Node B and RNC nodes in the network and assume UTRAN with AM data transfer service (Fig 4).

We assume that RLC is in acknowledged mode and MAC is in transparent mode. Therefore, RLC requires a header but MAC requires no header (Fig 3). In acknowledge transmission mode, both data PDUs and control PDUs are exchanged between peer RLC entities. We assume RLC SDU has been received in RNC from a server and converted to four AMD PDUs. Here, four RLC headers are added to them. Then, they pass through RNC/MAC by putting them into transport blocks transparently (MAC PDUs) and then pass them through the physical layer and arrive at NODE B.



(a)



(b)

Fig. 4 (a) An end-to-end system model (b) end to end protocol stack of a Web browsing user plane

Fig. 5 illustrates the timing diagram of the data transfer from the server (RNC) to the MS. In this scenario we assume that only one MAC PDU is in error in UE. Here, we have shown the processing and propagation delay times distinctly. T_{proc} is the SDU processing time needed for segmentation, T_{Iub} is the propagation delay time which is independent from TB. T_{rec} is the time after receiving the last PDU in UE and before transmitting the status PDU message [19]. We assumed RLC SDU is segmented to four L2 RLCs and the headers are added to them to constitute RLC PDUs (Figs. 3, 5). Besides, we see that RLC PDUs are the same as MAC PDUs in the transparent MAC mode (Fig 3). We assume a simple Selection Repeat ARQ protocol. In every SDU, the transmitter entity polls the receiver for a status report. According to the Selective Repeat ARQ, the receiver sends a PDU containing the status report indicating that the PDUs are received correctly and the ones to be retransmitted. When the Status PDU is received in the sender, PDUs buffered in the retransmission buffer of the sender entity are deleted or retransmitted according to the status report. Every PDU can be retransmitted at most $k_{\max}$ times. When this number is reached, the transmitter entity discards that PDU and all PDUs belonging to the same SDU [5]. Thus, when a packet encounters an error the effective throughput reduces to $d/((d+h)/R+RTT)$ in which *d* and *h* are numbers of data and header bits, respectively, and RTT is the round trip time of a PDU between RNC and UE.

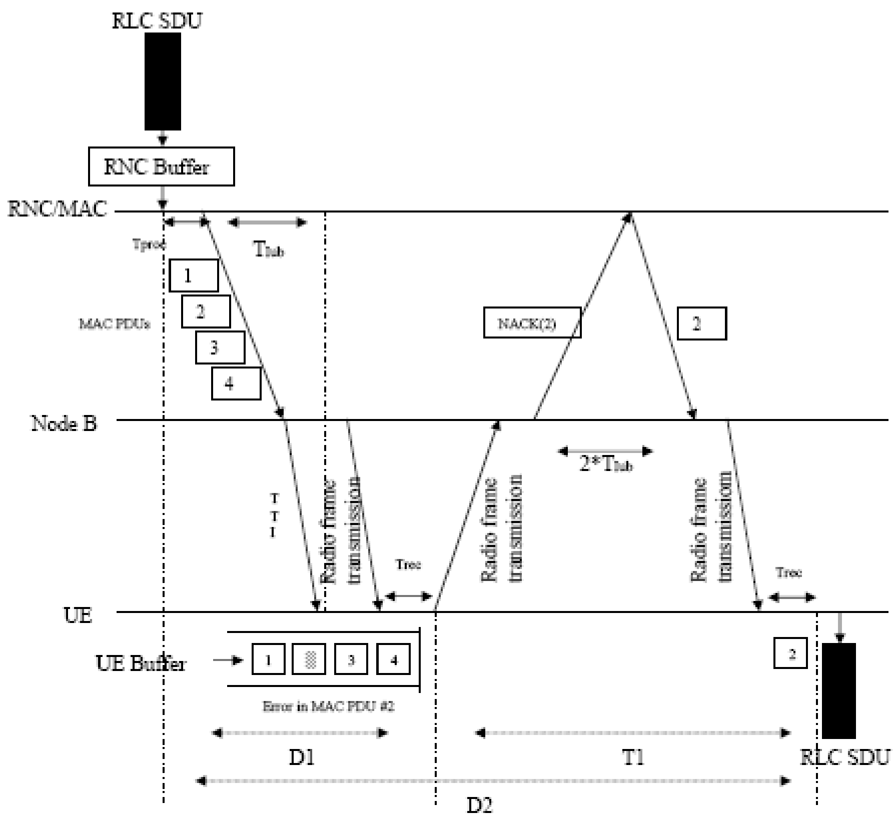


Fig. 5 Timing diagram of data transfer in RLC AM with only one error packet and one retransmission time

The delay from the time we send a NACK until a correct PDU is received is referred to as the round trip time (RTT). This is equal to the transmission time of a NACK plus 2 times the propagation delay, transmission time of a PDU and the recovery time. Assuming no error occurs, we calculate transmission time D_1 as the sum of the processing time, propagation delay T_{Iub}, PDU transmission time to the receiver and recovery time (Fig 6). If D_1 is the SDU transmission time from RNC to UE, we have

$$D_1 = T_{proc} + T_{Iub} + mTTI + T_{rec} \tag{1}$$

where m is the number of TTIs necessary to convey a RLC SDU and the header. Assuming an error occurs, T_1 equals the sum of twice the propagation delay, recovery time (T_{rec}), a NACK transmission time (Time Slot) and transmission time of a PDU as follows:

$$T_1 = 2T_{Iub} + T_{rec} + TimeSlot + NTTI \quad (2)$$

In the above formula we assumed that we can transfer a NACK by a time slot. Assuming a T_{rec} is a frame recovery (acknowledgement) time in the receiver before transmitting a status report ($T_{rec} = T_{ack}$), then we find

$$\mathrm{T}_{\mathrm{ackavg}} = \mathrm{TTI}/2 \quad (3)$$

If for a lost PDU we need *k* duplicate transmissions, then the total time of the transmission RLC PDUs and the final correct reception of MAC PDU, D_k , is:

$$D_k = D_1 + (k-1)T_1 \quad (4)$$

Now we can calculate the effective throughput as [9]

$$effective \quad throughput(k) = \frac{Correct \quad Transferred \quad Data}{D_k} \quad (5)$$

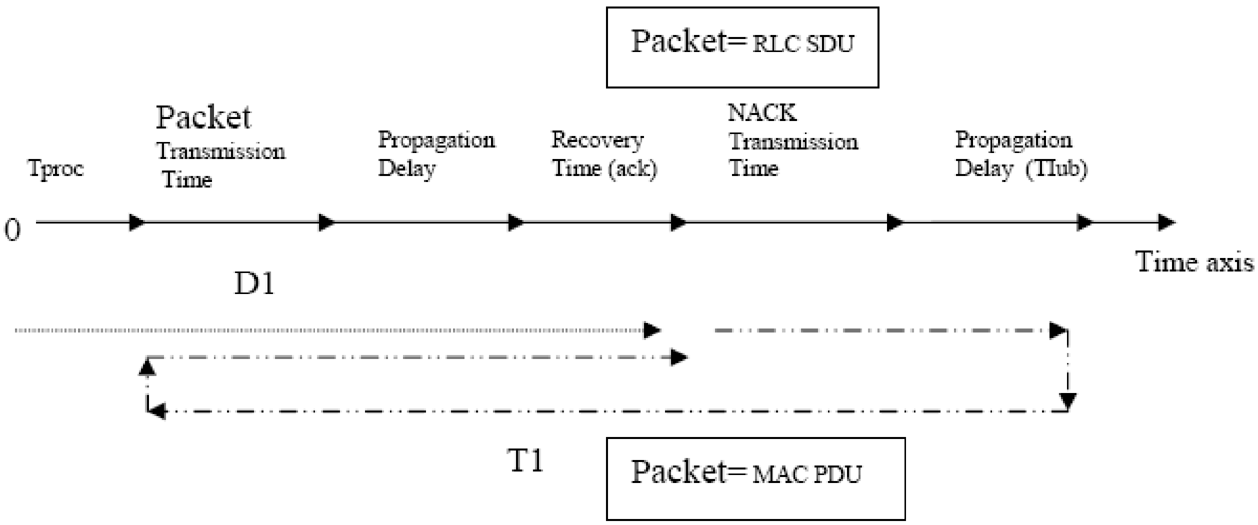


Fig. 6 Timing Diagram of a Packet transmission for only one MAC PDU error

If we define the efficiency of the protocol as

$$efficiency = \frac{effective \quad throughput}{link \quad bit \quad rate} \quad (6)$$

then we have

$$efficiency = \frac{Correct \quad Transferred \quad Data}{D_k.R} \quad (7)$$

V. Simulation

The MAC PDU size may be between 126 and 32766 bytes [17]. Now, we assume RLC SDU + headers =12600 bits and segmentation parameter = 4, then we find that RLC PDU=3150 bits which are transferred by one transport block. We also assume each of the 4 MAC PDUs contains d=3100 data bits and h = 50 header bits. In Layer 1 after CRC attachment, Turbo coding R=1/3, tail bit attachment and rate matching for forward link we found 9500 bits in the physical layer which must be transferred by a super frame (Fig 1).

For the slot format 13 in Table 1, the channel bit rate R and spreading factor SF are 240 kbps and 16, respectively. We have Nd1+Nd2 = 140 bits in each slot or 140*15=2100 bits /10ms. So, the number of frames in a super frame is N = 4.5 (we assume N=5). Besides, the transmission time duration of 9500 bits is 5*10 = 50ms and the number of TTIs necessary to convey a RLC SDU is m = 4*N (4 is the segmentation number). For the slot format 10 in Table 1, the channel bit rate R and spreading factor SF are 30kbps and 128, respectively. We have Nd1+Nd2 = 40 bits in each slot or 40*15 = 600 bits /10ms. So, the number of frames in a super frame is N=16. Besides, the transmission time duration of 9500 bits is 16*10 =160ms and the number of TTIs necessary to convey a RLC SDU is m = 4*N.
If we assume a cell radius of10km, TTI = 10ms, T_{proc} = 10ms and T_{prop} = 0.3ms and one block is in error in a SDU (Fig. 5), then the plot of the throughput of a forward link as a function of retransmission times (k) is shown in Fig. 7. We also vary the data block length and find the throughput efficiency of a forward link with i.i.d. errors versus retransmission times and data lengths as shown in Fig 8.

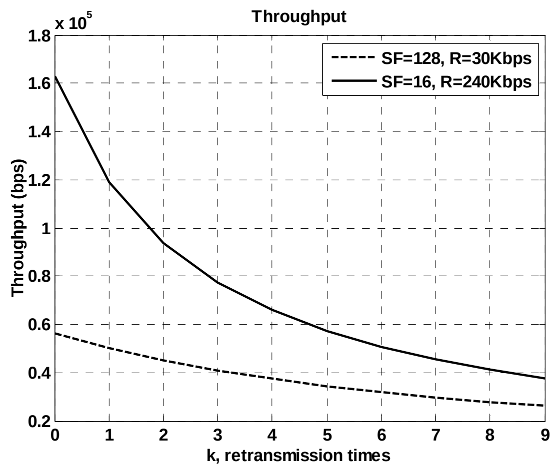


Fig. 7 Effective throughput of UMTS system versus the number of MAC PDU transmissions

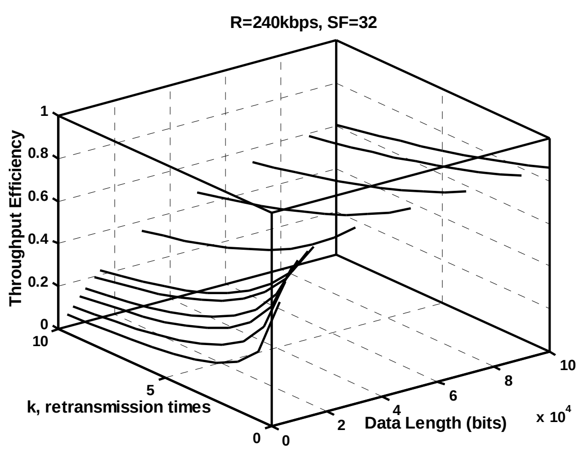


Fig. 8 Throughput efficiency of a forward link with i.i.d. errors versus retransmission times and data length

VI. Conclusion

We observe that throughput reduction occurs as a result of the increase in retransmission times of MAC PDUs and radio frames. We find 25% and 10% throughput reduction as a result of only one retransmission time in slot formats 13 and 10, respectively. These result are because of wasting the bandwidth and processing and recovery times in the RLC sub-layer and also because the link rate and the data bits in slot format 10 are less than the link rate and the data bits in slot format 13. Finally, we present the throughput efficiency versus retransmission times and data block lengths and show that the longer the data block the closer the efficiency is to 1. This is because more than one TB can be inserted into one TTI.

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