Performance Evaluation of SCTP over E-GPRS Air Interface For HTTP Traffic

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Abstract

SCTP has been originally designed to transport PSTN signaling messages over IP networks. It does not suffer from the head of line blocking issue inherent to TCP due to its multi streaming functionality. This work investigates the influence of the multi streaming when used to transport web pages in HTTP 1.0 transaction over SCTP. This solution is compared to the more classical case of HTTP 1.0 over TCP. Simulations results are obtained for a EGPRS air interface, for different cell loads and for a non congested network. Loss of data is only due to transmission errors. The results show that SCTP performs better than TCP when serveral streams are activated at SCTP level and that the multi streaming feature is not unfair for TCP.

1. Introduction

The introduction of new radio technologies such as EGPRS and UMTS eases the development of data interactive services by providing higher bit rates. Most of the data interactive services available in GPRS networks have been adopted from standard solutions of Internet. They mainly rely on the transport services of TCP which suffers from well known performances limitations when used over a radio interface [1]. SCTP can be a good candidate to transport interactive data applications over an air interface as it does not suffer from the head of line blocking issue and provides a multistreaming capacity. It is a message oriented protocol that provides the ability to convey information between two endpoints in one connection (called association in SCTP terminology). Information can be splitted among different streams. The loss of information in one stream does not affect the transmission in others streams. This feature can, for example, make it possible to transport different objects of one web page in the same association but in different streams.

This work compares the performances of SCTP over RLC/MAC to the more classical case of TCP over RLC/MAC. HTTP 1.0 is used as the data interactive application in both cases. The results obtained can be easily transposed to other data interactive applications such as WAP or Imode.

The paper is organized as follows. In section 2 a short overview of SCTP and EDGE RLC/MAC layer is given. Simulation scenario and parameters are detailed in section 3. Section 4 provides the simulation results and their interpretations. Finally, some conclusions are drawn.

2. SCTP and EDGE RLC/MAC overview

2.1 SCTP overview

SCTP is a connection oriented protocol that provides reliable transfer of messages from the upper layers (typically signalling applications). It is possible to multiplex several data transactions (for example signalling transactions) in different streams. Data transported in one stream can be delivered to upper layers independently of other streams. ARQ, flow control and congestion control are performed at the association level. Unlike TCP, SCTP keeps track of upper layer messages boundaries. Sequence numbers (called TSN in SCTP) are not counted in bytes but in messages at the association level.

SCTP encapsulates upper layer messages in packets called data chunks. Data chunks are associated with one stream. Other types of chunks are used for control purposes, for instance to acknowledge data or to setup an association. A SCTP packet can transport several chunks (as shown in figure 1).

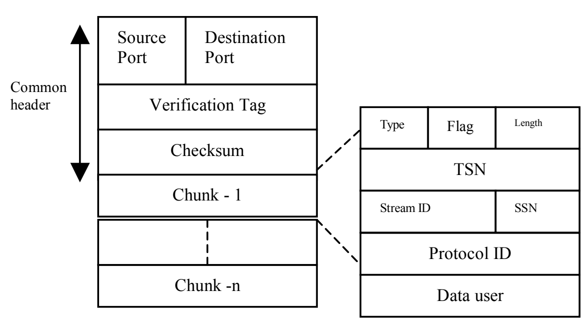

Fig 1.SCTP packet format [2]

SSN are chunk sequence numbers relative to one stream. They are used to provide in sequence delivery of the chunk contents to upper layers. Stream id identifies the stream to which a chunk belongs to.

SCTP uses a Selective acknowledgement policy. SACK chunks contain a TSN field that indicates the last chunk received correctly in sequence by the receiver and the state of the receiver window (position of the transmission gaps in the receiver window).

2.2 RLC/MAC

EGPRS RLC/MAC is an evolution of to the GPRS RLC/MAC [3]. It introduces several modifications, such as the increase of the window size and the use of incremental redundancy. It can also use a new modulation (8-PSK) and new coding schemes that increase the capacity per timeslot.

Window sizes can be chosen at the TBF setup between 64 and 1024 blocs by steps of 64. They depend also on the number of timeslots allocated to the TBF. It is also possible to report the full state of the receiver RLC receive window by using several radio blocks.

3. Description and parameters of the simulations

3.1 Simulated scenario

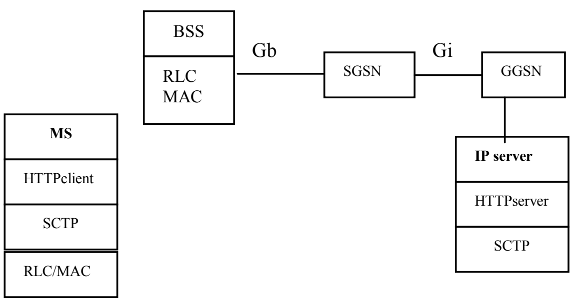

Fig 2.Simulated configuration

Simulations have been carried out with ns2 [4], considering the scenario given by figure 2 . The HTTP 1.0 traffic generator has adapted and modified from [5]. A SCTP module has been developed from previous works [6]. RLC/MAC is simulated with a ns2 module developed at ENST [7].

3.2 HTTP traffic model

HTTP 1.0 transactions generated in simulations follow the general sequence scheme described in figure 3.

Client		**Server**
Init		Init
Request page	→	RecvFirstRequest
SendFirstRequest		
RecvFirstReply	←	SendFirstReply
GenFurtherRequests		
sendFurtherRequests	→	recvFurtherRequests
recvFurtherReply	←	SendFurtherReply

Fig 3. Interaction Client/Server to download a web page

A web page is modelled as follows:

- The size of a web page is a Pareto random variable. The average size is 10Kbytes and α = 1.2,
- One page contains embedded objects which number and size are Pareto Random variables. The mean value of objects size is about 4Kbytes and α = 1.1. For the objects number the mean value is 3 and α = 1.5.

The pages generated contain at least three types of objects.

3.3 Transport protocols

Table 1 and table 2 contain the parameters used for SCTP and TCP agents in the mobiles and in the web servers.

	Mobile	**IP server**
MTU	1500 bit	1500bit
DataChunkSize	1468 bit	**1468bit**
OutStreams	3	**3**
Initial Cwnd	2	**2**
Initial Rwnd	**64Kbit**	64 Kbit
Initial RTO	1sec	1sec

Table1. SCTP agents' parameters.

	Mobile	**IP server**
MTU	1500 bit	1500bit
Packet size	1460	**1460bit**
Initial Cwnd	2	**2**
Initial Rwnd	**64Kbit**	64 Kbit
Initial RTO	1sec	1sec

Table2. TCP agents' parameters.

3.4 RLC/MAC Layer configuration and link parameters

Link adaptation is not considered in this study. Data RLC blocs are transported using a coding scheme MCS 4. Window sizes are set to 384 RLC/MAC blocs. Four timeslots are available in downlink for transmission. Simulated mobiles are 4+1.

The error model used is uniformly distributed at the block level. The Gb interface is modelled with a duplex link of 64 kb/s transmission rate and 1ms propagation delay. The Gi interface is modelled with a duplex link of 5Mb/s transmission rate and 10ms propagation delay. A uniformly distributed error model for packets (1% error rate) is also introduced in the Gi interface. Finally the network between GGSN and the server node is modelled with a duplex link of 5Mb/s transmission rate and 10ms propagation delay.

4. Results and analysis

In this section the effect of the multi streaming on the performance of HTTP 1.0 transactions is investigated. The first results are obtained for a single mobile using one and three streams to transport HTTP 1.0 transactions (section 4.1). The performances are compared to those obtained with TCP. Section 4.2 performs the same simulations than in section 4.1 for a loaded cell.

4.1. Multi streaming effect on HTTP transfer performance for a single mobile

Figure 4 compares TCP (with SACK option activated) and SCTP performances for the transfer of web pages when one single stream is activated for SCTP. For low block error rates, the two protocols have nearly the same performance. As the block error rate increases, SCTP has better performances than TCP.

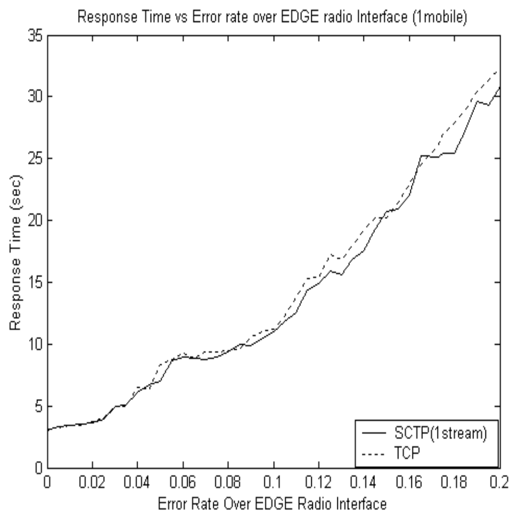

Fig 4. Response Time vs error rate in case of HTTP 1.0 over TCP and SCTP (one stream) over RLC/MAC.

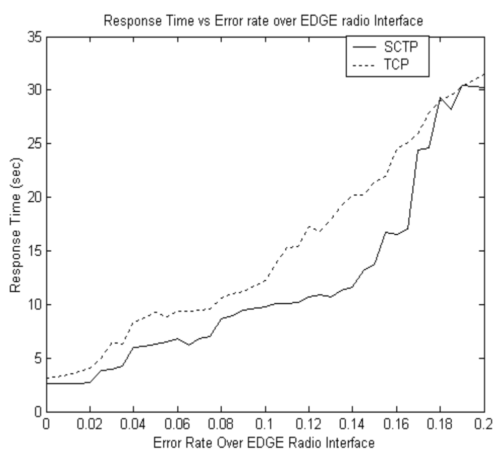

Fig 5. Response Time vs error rate in case of HTTP 1.0 over TCP/SCTP (three streams) over RLC/MAC.

Figure 5 is obtained for a SCTP configuration using three streams. SCTP performs much better than TCP. This is the effect of the multi streaming. We notice that for low error rates, we have little difference between the two transport protocols. For example for an error rate equal to 0.001, we have a difference between the response times for the two protocols less than 0.4sec . While, with higher error rates, SCTP provides better performance than TCP. For example for an error rate equal to 0.14 we have a difference between the response times for the two protocols more than 5 sec. This is due to the multi streaming facility that speeds up downloading times in case of the use of SCTP.

4.2. Multi streaming effect on HTTP transfer performance in a loaded cell

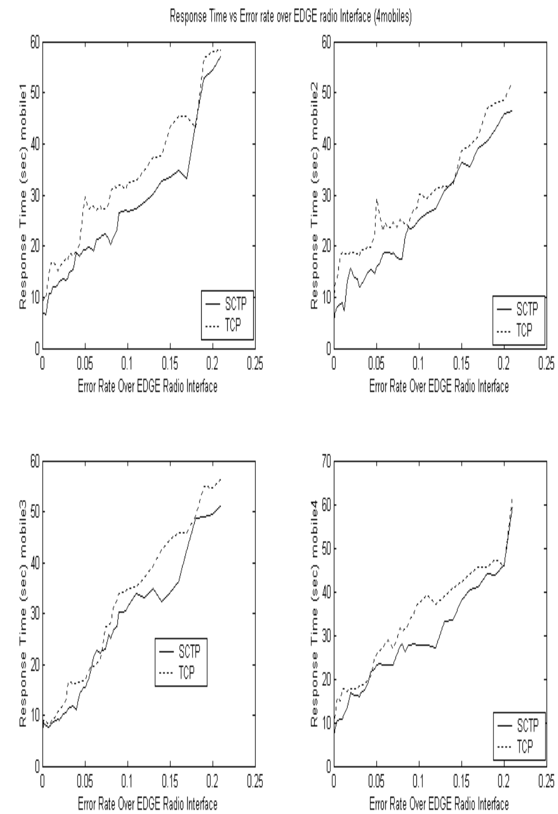

Fig 6. Response Time vs error rate in case of HTTP 1.0 over TCP/SCTP (three streams).

Figure 6 and figure 7 represent the results obtained for four active mobiles and for three streams activated for each SCTP association. SCTP exhibits better performance than TCP. This is due to the head of line blocking issue that affects TCP performance when transmission errors occur in the air interface. Transmission is blocked in the TCP transmitter until the segments in error are corrected. In the SCTP case, the transmission is blocked when transmission errors occur in one stream. The sender can continue to transmit HTTP 1.0 packets.

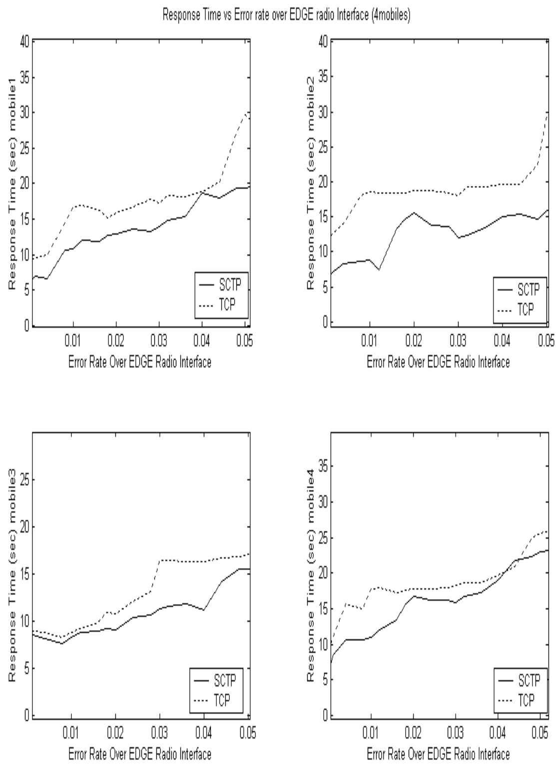

Fig 7. (zoom of figure 6)

Figure 6 and figure 7 also demonstrate another important point. SCTP and TCP behave friendly with each other, as they use the same congestion control mechanisms. The multi streaming feature does not impact on TCP connections (SCTP congestion control checks if the transmission at stream level does not generate congestion).

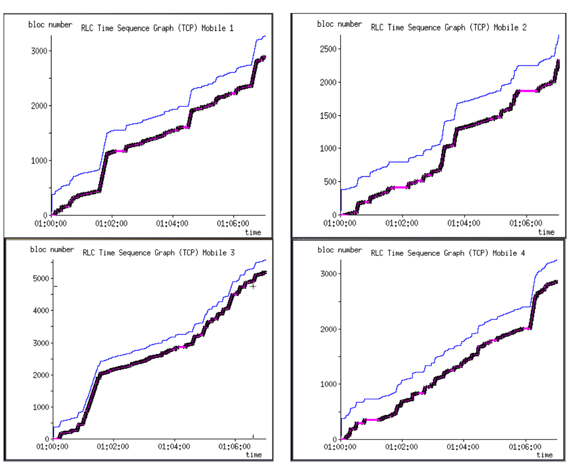

Fig 8. RLC Sequence number for HTTP 1.0 over TCP over RLC/MAC

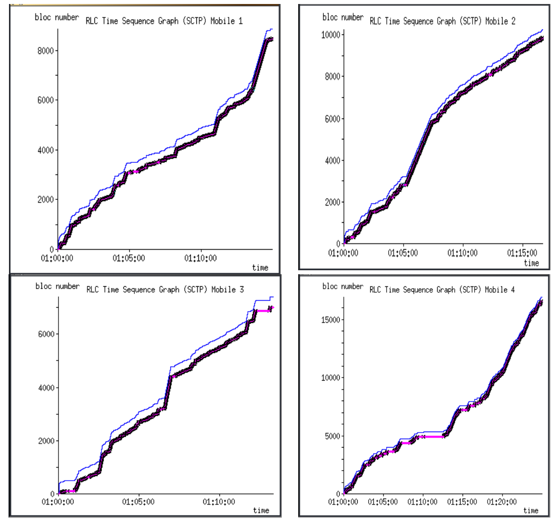

Fig 9. RLC Sequence number for HTTP 1.0 over SCTP over RLC/MAC

Figure 8 and Figure 9 show the evolution of the RLC block sequence numbers against time. Figure 8 is obtained for TCP and figure 9 for SCTP. They confirm the results observed in figure 6 and figure 7. For SCTP, it can be noticed that the RLC throughput is higer than in the TCP over RLC/MAC case. This highlights the fact that when multistreaming is used, transmission errors on the air interface can be handled more properly and avoids TCP limitations. This advantage will remain for reasonable cell loads and whenever the network is not congested. SCTP advantage is likely to disappear when the network starts becoming congested.

5. Conclusion

Several studies have already investigated the performances of SCTP in a wired context [8]. We have studied the use of SCTP protocol as a transport protocol for interactive data traffic such as HTTP 1.0 on a EGPRS network. The simulations made show that SCTP provides better response times (at HTTP level) than TCP for different block error rates on account of the multistreaming feature. This advantage is only valid for a non congested network and in that case SCTP multi streaming feature remains fair with TCP.

Further work will evaluate performances of the use of PFC (Packet Flow Context) combined with SCTP. SCTP streams can be mapped on different RLC/MAC flows to provide differentiated QoS and relative priorities between different data flows on the air interface.

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