

VoIP Packet Optimization for UMTS Networks

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Abstract: The utilization of VoIP (Voice over IP) in UMTS networks requires the establishment of adequate rules in order to maximize the efficiency of transmission over the air interface. Here we will describe a technique, based on the reduction of voice packet overhead, with the objective of minimizing the power expended during a typical conversation.

Keywords: UMTS, IMS, LTE

I. Introduction

In this work reference will be done specifically to the Packet Switching (PS) mode of transmission in UMTS systems. The PS domain, including the IMS (IP Multimedia Subsystem) [1], is intended to support multimedia traffic in general. Figure 1 presents a schematic view of the UMTS system, with the CS (Circuit Switching) and the PS domains. It is also included the Common Channel Signaling Network (SS#7).

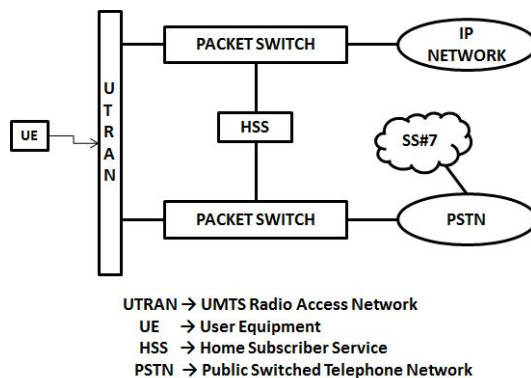


Figure 1 - Basic UMTS Network

It is of paramount importance to achieve the required QoS (Quality of Service) level in a real time IP-based communication system without compromising the necessary transmission power levels. Eliminating redundancy in the data transmission may contribute significantly to this objective.

A main issue that contributes to the efficiency's improvement of VoIP packet's transmission is the reduction of IP packet's overhead transmitted during a conversation, in agreement with the UMTS 3GPP recommendations. In fact, reducing this packet overhead by means of header compression is the fundamental mechanism adopted at the air interface to improve transmission performance in terms of power efficiency and delay [2,3].

It is important to remember the existence of two mechanisms for error control in UMTS systems: one is the checksum, based on the two bytes that appeared in a field of the UDP (User Datagram Protocol) header datagram. The UDP is the transport level for the transmission of VoIP packets. The other mechanism is the CRC (Cyclic Redundancy Check) field, of the Physical level. It must be considered that the checksum is a very simple technique, developed at the time when the processing speed was very low, due to the lack of fast processors.

The principal contribution of this work is to show that a further reduction of the overhead, without impacting the error performance, may be obtained by the adoption of the following techniques:

1. the provisory cancelation, in the air interface, of the two bytes used for the UDP checksum. Afterwards, these bytes are conveniently restored, for the normal packet routing in the fixed IP network;
2. the reduction of the CRC size from two to one byte;

The main function of the UDP checksum is to protect the IP addresses and port numbers. The calculation of the UDP checksum is a very simple procedure, with limited results, only needed where no other

error control method exists at the lower levels. It is here shown that the CRC appended by the physical layer of the UMTS protocol architecture provides the required protection, thus eliminating the need of the checksum bytes. Besides, it can be noted that at the initial phase of the call, the first packets (that contains the IP addresses and port numbers) are transmitted several times, depending of the RTT (Round Trip Time) of the system (normally seven times, for an RTT of 140ms) [4] and since the addresses will not change during the call, this constitutes an effective guarantee of address integrity. After Item 1 (Introduction), Item 2 describes the Simulation Process that allows the visualization of the solution for the problem of elimination of the redundancy associated with the transmission of VoIP packets in the air interface. Finally, item 3 shows the conclusions obtained with the application of the technique developed for this redundancy elimination.

II. Simulation Of The Proposed Technique

The transmission of compressed VoIP packets was simulated with a Matlab's program (a product of Mathworks) developed for this purpose. Figure 2 shows a simplified view of the simulator's main blocks.

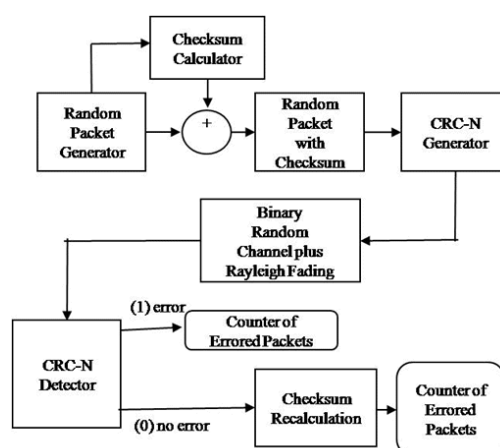


Figure 2: Diagram of the Simulator

After the insertion of the Checksum and the CRC, the output packets of the Random Packet Generator passes through a Binary Discrete Channel with Rayleigh Fading, characterized by a given value of BER (Bit Error Rate). If the output of the CRC Detector block is 0 (no error detected) the Checksum for the packet is recalculated and compared with the original one, in order to verify the occurrence of error in the packet transmitted. Thus, the CRC error detection and Checksum are performed in that order and two distinct statistics of the Frame Error Rate (FER) are computed: (i) the ratio between the number of erroneous frames detected by the CRC and the total number of transmitted frames; (ii) the ratio between the number of erroneous frames detected by the checksum and the total number of non-erroneous frames detected by the CRC (residual FER). Figures 3, 4 and 5 show the results for FER using different CRC's lengths and taking BER values ranging from 10^{-5} to 10^{-3} . Note that 3GPP considers the value of 5×10^{-4} as the maximum acceptable BER for VoIP transmission [5].

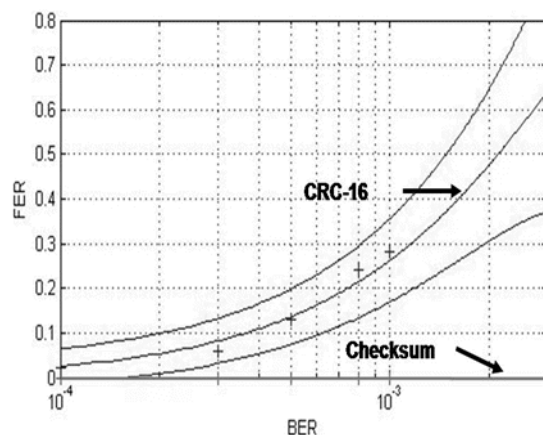


Figure 3: FER x BER for CRC-16

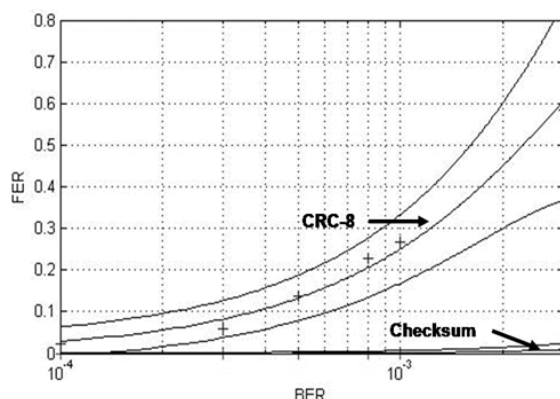


Figure 4: FER x BER for CRC-8

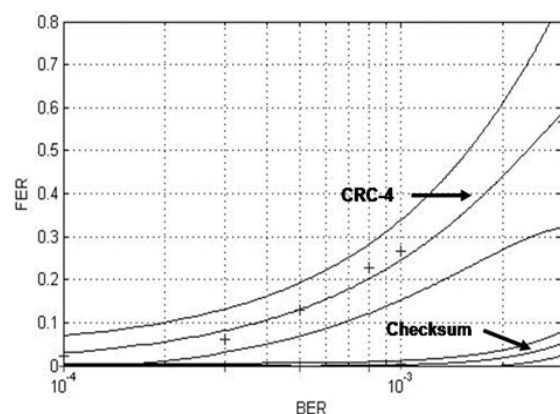


Figure 5: FER x BER for CRC-4

The results in Figures 3 to 5 allow one to assess the joint behavior of CRC and Checksum in a packet transmission over the air interface. The set of curves presented in the figures were obtained based on a simulated transmission of 10^5 packets, with a confidence interval of 95%. With a 16 bit CRC (CRC-16) no erroneous frames were detected by the checksum in the simulated channels, as shown in Figure 3, which means that CRC-16 gives a reasonable protection to the transmitted packet for the considered range of BER. Figures 4 and 5, corresponding to CRC-8 and CRC-4, show that residual erroneous packets occur only for BER values substantially above the 5×10^{-4} target point.

III. Conclusions

From the preceding discussion we may conclude that it is not reasonable to transmit the checksum in the air interface. Instead, it is better to recalculate it at the packet destination, in the UTRAN (UMTS Terrestrial Radio Access Network). The results show also that we may replace CRC-16 by CRC-8 without any significant impact in the error performance for a typical channel. Adopting this procedure, we can save 9 bytes in the air interface transmission.

It is important to emphasize that, in this situation the checksum is eliminated only in the air interface (i.e. from the UE to UTRAN and vice-versa). After the packet arrival, it is again recalculated and reintroduced.

We have to consider that the UMTS and the other mobile systems in operation, continue to receive adaptations, as a consequence of the technology evolution. In spite of the commercial introduction of new systems like LTE (Long Term Evolution), the more established systems are always receiving new adaptations, with the principal objective to become more suitable to the new requirements, for example in terms of adaptation for the introduction of new services or else to obtain an economy of resources, with the utilization of more efficient techniques.

The development discussed here had the objective to increase the efficiency of the network without introducing a new equipment. In fact, only minor adjustments in software would be required.

References

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