Chapter 5

QoS and Performance Evaluation for SIP-Based VoIP Over DMO

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ABSTRACT

In this chapter, analyses for the performance metrics that define the quality of service (QoS) of SIP-based VoIP will be introduced. SIP-based VoIP applications over Direct Mode of Operation (DMO), which behaves in a way similar to Mobile Ad-hoc Network (MANET) systems, have three main performance categories related to the QoS. These categories are the SIP signaling, voice data transmission, and MANET routing. The SIP signaling controls the VoIP calls initiation, termination, and modifications. The major QoS parameters of VoIP that are managed by SIP signaling are the registration intervals, call setup time, and call termination time. These QoS parameters are increased in MANET due to the nodes’ mobility that affects the routing calculations and the connectivity status. These necessitate mechanisms to reduce the delays in the MANET environment. The voice packets are transferred over the Real Time Protocol (RTP) which is encapsulated in the unreliable transport protocol using the User Datagram Protocol (UDP).

INTRODUCTION

In this chapter, analyses for the performance metrics that define the quality of service (QoS) of SIP-based VoIP will be introduced. SIP-based VoIP applications over Direct Mode of Operation (DMO), which behaves in a way similar to Mobile Ad-hoc Network (MANET) systems, have three main performance categories related to the QoS. These categories are the SIP signaling, voice data transmission, and MANET routing. The SIP signaling controls the VoIP calls initiation, termination, and modifications. The major QoS parameters of VoIP that are managed by SIP signaling are the registration intervals, call setup time, and call termination time. These QoS parameters are increased in MANET due to the nodes’ mobility that affects the routing calculations and the connectivity status. These necessitate mechanisms to reduce the delays in the MANET environment. The voice packets are transferred over the Real Time Protocol (RTP) which is encapsulated in the unreliable transport protocol using the User Datagram Protocol (UDP).
setup time, and call termination time. These QoS parameters are increased in MANET due to the nodes’ mobility that affects the routing calculations and the connectivity status. These necessitate mechanisms to reduce the delays in the MANET environment. The voice packets are transferred over the Real Time Protocol (RTP) which is encapsulated in the unreliable transport protocol using the User Datagram Protocol (UDP). In addition, the bandwidth consumption, delays, jitter and packet loss are QoS parameters that quantify VoIP performance. The bandwidth is dependent on the codec system used. The delays experienced by voice packets are one-way delays between the two calling ends and are affected by the routing and connectivity delays in the MANET environment. The jitter is related to the variations in the delay and the RTP tries to recover the loss for it, while the packet loss is related to the network congestion and erroneous links. A number of studies have been undertaken for these performance metrics that support the evaluation studies.

In this chapter, the simulation efforts have been carried out using GSM voice codecs to evaluate SIP call processes and QoS parameters together over MANET. The SIP signaling and QoS parameters for VoIP have been assessed on the OPNET® Modeler simulation scenarios. The simulation efforts have not considered other simultaneous applications that could influence the performance of the SIP applications to provide the effort implementations. However, the assumptions considered a background traffic with 30% to 40% of the overall bandwidth.

In this evaluation study, the results for both Static and Uniform mobility models are representing the best effort of the implemented scenarios. The results for the Static and Uniform scenarios are meeting with the evaluation results for similar scenarios over other network systems which support the reliability level of the findings of this research study. Hence, the investigated QoS parameters are considered as the benchmark values for SIP-based VoIP over AODV and OLSR MANET. These benchmarking efforts also considered the Registration Request Delays (RRD), Session Request Delays (SRD), and the Session Disconnect Delay (SDD) of the RFC 6076. These main SIP end-to-end metrics had been implemented in this research study to provide an evaluation for the SIP signaling over VoIP application between the SIP calls’ entities as represented in Figure 2. The results show that these parameters are comparable for both IPv4 and IPv6 in AODV and OLSR MANET environments. Furthermore, the simulation efforts in this research study used to design and implement number of important parameters for SIP signaling performance evaluation. These parameters are the RRD, SRD, and RDD from the RFC 6076 together with the parameters of the call registration delay, and the call setup delay.

RELEVANT RESEARCH EFFORTS

In general, limited numbers of researchers have studied and evaluated the performance of real-time applications over MANET. Most of the evaluation efforts considered Constant Bit Rate (CBR) or File Transfer Protocol (FTP) traffic with a different number of MANET nodes. For IPv4 MANET, a performance evaluation with OPNET® Modeler v14.5 for the reactive routing protocols, AODV and DSR, using GSM voice traffic, concluded that AODV has the lowest end-to-end delay and a lower network load compared with DSR (Pandey & Swaroop, 2011). Furthermore, AODV presents higher average throughput and received traffic while DSR does not scale well with large sized networks. Simulation results also showed that AODV reactive routing protocol is the best suited for MANET, while DSR recorded very poor QoS in MANET with high node capacity for GSM voice applications. However, this research did not consider VoIP applications with different mobility models (Skordoulis, et al., 2008).