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Media Access Delay and Throughput Analysis of Voice Codec with Silence Suppression on Wireless Ad hoc Network

Ruchi Deepak Shah[a], Santosh Kumar Singh[b]

[a]Research Scholar,AMET(ACADEMY OF MARITIME EDUCATION AND TRAINING) university,135, East Coast Road, Kanathur - 603112, India

[b]Head-Information Technology Department, Thakur College of Science and Commerce, Mumbai, India

Abstract

The rapid growth in internet has led to the emergence of different wireless technologies and simultaneously there has been a tough challenge to provide QoS over such wireless technologies.The demand for various real time multimedia application especially voice and video applications to be available on the internet has also increased. Multimedia applications need a lot more bandwidth and have different QoS requirements than the applications that were used in early years of the internet. The main reason for the poor quality of voice over IP is because we are transmitting the real time applications over the IP network which is mainly meant to transfer only data and not for voice transmission. The purpose of this paper is to check the performance of VOIP application under different CODEC's such as G.711, G.729 and G.723.1 over wireless network with and without silence suppression. Another issue addressed in this paper is the effect of increasing nodes on voice transmission for the different codecs. Parameters like media access delay and throughput are taken into consideration. Simulation is performed using OPNET modeler simulator.

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Keywords: VOIP; QoS; Voice Codec; Silence Suppression; OPNET

1. Introduction

The rapid growth in internet has led to the emergence of different wireless technologies and simultaneously a tough challenge to provide QoS over those wireless technologies. Wireless networking is an emerging technology that allows users to access information and services regardless of their geographic position. Wireless networks can be classified into the following two categories[3].

· Infrastructure based networks.

· Infrastructure-less (ad hoc) networks.

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Wireless ad hoc network is a collection of mobile nodes forming a temporary network where the nodes can be connected dynamically. A number of ad hoc routing protocols are available namely Destination Sequenced Distance Vector (DSDV), Dynamic Source Routing (DSR) and Ad hoc On-demand Distance Vector (AODV) routing protocols. In this paper we have simulated a wireless ad hoc network with AODV routing protocol.

The use of wireless networks has extended beyond simple text and data transmission to extremely complex voice and video networks. This led to the existence of VoIP. Voice over Internet Protocol (VoIP) is a term used for delivery of speech or voice over IP networks such as the Internet or other packet-switched networks. VOIP converts the voice signal from your telephone into a digital signal that travels over the internet then converts it back at the other end. VoIP is simply the transmission of voice conversations over IP-based networks.[4]This paper focuses on the performance of voice codecs namely G.711, G.723.1 and G.729. VoIP equipment used these compression/decompression methods for conversion of analogue audio signals to digital bit stream. These techniques reduce the required bandwidth with assured voice quality.[5]

The most popular voice coding standards for telephony include:

G.711: The most commonly used voice codec for VoIP, describes the 64 Kbps PCM coding method to transmit voice and data with quality. There are two versions of this codec according to the regions: m-law for USA and Japan, and a-law for rest of the world [5.]

G.723.1: This codec can compress the voice signals at a low bit rate and very low-bandwidth codec as compare to others[5]. This uses only 5.3 Kbps to 6.3 Kbps of bandwidth.

G.729: This low-bandwidth codec is also commonly in use that consumes only 6 to 8 Kbps for voice data transmission[5]. There are two versions of this standard, G.729 and G.729 Annex A. Both versions differ from each other w.r.t computational complexity but voice quality is as same as 32 Kbps ADPCM method[5]. The voice quality of this codec is less than G.711 codec, but this is still a good option for low bandwidth connections.

The QoS for wireless networks in this paper is measured with respect to the parameters namely media delay access and throughput

2. Network Design and Configuration

The paper aims at the performance analysis of VOIP application under different CODEC's such as G.711, G.729 and G.723.1 over wireless network. Also, silence suppression is implemented along with the voice CODECs to evaluate the performance of the codecs.

OPNET Modeler is used for simulating various scenarios. OPNET Modeler is a set of decision support tool, providing a comprehensive development environment for specification, simulation and performance analysis of communication networks, computer systems and applications. It allows us to create models, execute simulations, and analyze the output data.

The wireless ad hoc network architecture is as shown in Fig. 1. and Fig. 2. which is simulated to test the performance of the voice codec's with and without silence suppression along with varying data rate and no. of mobile nodes.

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