Performance of VoIP in Wired-Cum-Wireless Ethernet Network

Nurul I. Sarkar, School of Computing and Mathematical Sciences, Auckland University of Technology, Auckland, New Zealand

Kashif Nisar, School of Computing InterNetWorks Research Laboratory, University Utara Malaysia and School of Computing and Mathematical Sciences, Auckland University of Technology, Auckland, New Zealand

ABSTRACT

The Voice over Internet Protocol (VoIP) is a rapidly growing technology that enables transport of voice over data networks such as Ethernet Local Area Networks (LANs). This growth is due to the integration of voice and data traffic over the existing network infrastructure, low cost, and improved network management offered by the technology. This paper reports on the performance of VoIP traffic characteristics in a wired-cum-wireless Ethernet LAN. The effect of increasing the number of VoIP wireless clients, different voice codec schemes, and packet arrival distributions on system performance is investigated. Through various simulation experiments under realistic network scenarios, such as Small Office Home Office (SOHO) and campus networks, this paper provides an insight into the performance of VoIP over Ethernet LANs. Simulation results show that VoIP clients and voice codec schemes have significant effect on system performance. The authors preformed OPNET-based simulations to validate their experiments.

Keywords: Ethernet Local-Access-Networks (LANs), Network Simulator, Small Office Home Office (SOHO), Voice Codec, Voice over Internet Protocol (VoIP),

INTRODUCTION

In recent years, there has been a growing trend in real-time voice communication using an Internet protocol (IP). The Voice over Internet Protocol (VoIP) is a technology that allows users to make telephone calls over an IP data network (Internet or Intranet) instead of a traditional Public Switched Telephone Network (PSTN). Therefore, the VoIP provides a solution that merges both data and voice which gains benefits including cost savings, high quality and value added services. Today, the VoIP is becoming one of the most widely used technologies with more and more people and organisations are using VoIP systems worldwide. There are various VoIP communication software products already available on the internet: Skype, Google Talk, and Windows live messenger. All of them can provide good quality, cheaper, and even free phone calls (Theoharakis & Serpanos, 2002; Rizzetto & Catania, 1999; Goode, 2002; Yeonsik, Kakumanu, Cheng-Lin, & Sivakumar, 2007; Lucani, Badra, & Bianchi, 2007).

The VoIP is not only popular through the internet; but also a rapidly growing technology through data networks such as Ethernet LANs. The Ethernet is considered a good platform for
the VoIP (Bhumip, 2003) as Ethernet based LANs are very common in enterprises and other organizations for data networking. Therefore, there is a tremendous growth of the VoIP over both wired and wireless Ethernet LANs. This growth is due to the integration of voice and data over the existing networking infrastructure, low cost, flexibility, and improved network management offered by the technology. In addition, wireless Ethernet LANs (IEEE 802.11) allow mobile users to connect to the network from a location where network cables may not be available or may not be the best choice, such as old buildings, Hospitals, and conference rooms. Therefore, Wireless LANs (WLANs) are other important segments for VoIP deployments. The performance of VoIP in a wired-cum-wireless network is investigated in this paper.

Despite the potential benefits of the VoIP over Ethernet LANs, one of the significant challenges faced by designers of the VoIP is to provide a quality of service (QoS) to all users on the network, especially under medium-to-high traffic loads. The VoIP is a real time service; the channel access competition can result in delays or packet losses which is detrimental to real-time applications. However, the VoIP is an emerging technology that has many issues. For example, how to deploy VoIP services over existing networks is still a challenge issue for network managers, designers, planners, and engineers. Therefore, a good understanding of VoIP traffic characteristics and network performance analysis is required to assist efficient deployment of such technologies over WLANs. This paper aims to investigate the effect increasing the number of wireless clients, different codec schemes, and packet arrival distributions on system performance.

**RELATED WORK**

Voice codec generally provide a compression capability to save network bandwidth. Currently, there are various types of audio codecs available for voice applications. The simplest and most widely used codecs are G.711, G.723 and G.729 (Nguyen, Yegenoglu, Sciuto, & Subbarayan, 2001). The simplest encoder scheme is G.711 (64 kb/s). G.711 is the sample base which uses the Pulse Code Modulation (PCM). The acceptable packet loss factor of G.711 is up to 0.928%. In this paper, we consider a simulation scenario to study the performance of G.711, G.723, and G.729.

In reference (Zeadally, Siddiqui, & Kubher, 2004), the authors have studied a large scale WLAN system with about 2000 nodes and have presented most comprehensive trace of network activity. This study can help understand usage patterns in WLANs which are critical for those who develop, deploy, and manage wireless technologies as well as those who develop systems and application software for wireless networks (Salah & Alkhoraidly, 2006; Chou, 2007; Walsh & Kuhn, 2005; Huijie & Xiaokang, 2005; Van de Capelle, Van Lil, Theunis, Potemans, & Teughels, 2001; Baratvand, Tabandeh, Behboodi, & Ahmadi, 2008).

Table 1 lists the key researchers and their main contributions in the performance studies of VoIP over wired and wireless networks.

**NETWORK MODELLING AND SCENARIOS**

OPNET educational version 14.0 was used for modelling and performance studies of VoIP over WLANs. This OPNET supports the Session Initiation Protocol (SIP) only. Figure 1 shows OPNET representation of Ethernet LAN modelling. Figure 1 shows a top level view of wired and wireless Ethernet LAN modelling. Figure 2 shows a wireless LAN subnet with six wireless clients. The Application Configuration is used to setup simulation model with default parameters of VOIP application. The Profile Configuration is used to setup wired and wireless clients profile.

We study the network performance by measuring Ethernet delay, voice jitter, voice end-to-end delay, WLAN delay, WLAN throughput, and packet dropping. The total packets sent and received are also recorded. All
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