SIMULATION STUDY ON IP-QoS VIDEOCONFERENCING PERFORMANCE USING OPNET MODELER

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ABSTRACT

The deployment of Videoconferencing also known as Video and Voice over IP (VVoIP), over existing IP networks is gaining popularity these days. The Quality of Services (QOS) is one of most important areas of Internet development and support represent one of the most important features of modern multimedia networks. As the Internet originally developed for data communications is now used more and more for real-time applications. Since QoS support technologies are still objects of intensive development and fine tuning, simulations and modeling are highly required in this field. This paper presents the simulation study on IP-QoS videoconferencing performance using OPNET MODELER and to propose a method to proven that the quality of services (QoS) system is very important in Ethernet network. For this project, topology was built using the same system parameters with two different scenarios to see the comparison impact quality of services that occurred in the both scenario. Accompanying for this project, the final simulation results will show which scenario has the best performance in quality of services (QoS).

Keywords : Quality of services (QoS), Videoconferencing, Opnet Modeler.

ABSTRAK

Penyebaran sidang video juga dikenali sebagai *"Video dan Voice over IP (VVoIP)"*, melalui rangkaian yang menggunakan sistem *"Internet Protocol(IP)"*. Kualiti dalam perkhidmatan merupakan satu sistem yang paling penting dalam perkembangan internet dan merupakan salah satu ciri-ciri yang dititikberatkan dalam rangkaian multimedia yang serba moden. Kegunaan Internet pada awalnya digunakan untuk penghantaran data sahaja dan kini lebih banyak digunakan untuk aplikasi real-time. Disamping itu, QoS sistem teknologi diperlukan untuk meningkatkan kualiti perkhidmatan dalam system rangkaian. Projek ini adalah berkaitan dengan kajian simulasi terhadap prestasi penyebaran siding video IP-QoS menggunakan OPNET Modeler dan mencadangkan satu kaedah untuk membuktikan bahawa kualiti dalam perkhidmatan (QoS) sistem yang sangat penting dalam rangkaian Ethernet. Disamping itu, topologi dibina dengan menggunakan sistem yang sama parameter dengan dua senario yang berbeza untuk melihat perbandingan kesan kualiti dalam perkhidmatan (QoS) yang terjadi dalam kedua-dua senario. Keputusan akhir simulasi akan menunjukkan senario mana mempunyai prestasi terbaik dalam sistem kualiti perkhidmatan (QoS).

Kata kunci: Kualiti Perkhidmatan, Penyebaran Sidang Video, Perisian OPNET.

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OPNET	Optimized Network Engineering Tool
QoS	Quality of Services
IP	Internet Protocol
VVoIP	Video and Voice over Internet Protocol

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CHAPTER 1

Introductions

This chapter will briefly discuss on the project overview. The project background, problem statement, scope, objective and methodology will be presented in this chapter.

1.2 Project Background

The Quality of Services (QOS) is one of most important areas of Internet development. As the Internet originally developed for data communications is now used more and more for real-time applications. For this project, QOS is the ability to treat packets differently as they transit a network device, based on the packet contents and to identify which network traffic is critical traffic and appropriate resources to support those traffic streams. Accompanying for this project, we are using OPNET modeler for simulated IP-QOS videoconferencing and approach on how to deploy a popular real-time network service. To achieve an efficient deployment of videoconferencing must ensure these real-time traffic requirements can be guaranteed over new or existing IP networks.

Otherwise, we can assure the audio and video data traverse the network with minimum delay and to evaluate by measuring parameters, bandwidth, end-to-end delay, and jitter and packet loss using OPNET method. However, this system will be able to detect the latency and particularly in audio with using three fundamental concepts an affecting real-time data transmission must be considered while designing the IP network for audio and video data. The three concepts using for analysis this project is provisioning, queuing and classifying. In the manner of simulation, we can improve packet loss characteristics and allows voice data to be transmitted simultaneously along with other application services. Provide better service to selected network traffic over various technologies and allows customers to make long-distance calls at much lower rate.

1.3 Problem statement

When the Internet was first deployed many years ago, it lacked the ability to provide Quality of Service guarantees due to limits in router computing power. It therefore ran at default QoS level, or "best effort". There were four "Type of Service" bits and three "Precedence" bits provided in each message, but they were ignored. These bits were later re-defined as DiffServ Code Points (DSCP) and are largely honored in peered links on the modern Internet.

When looking at packet-switched networks, Quality of Service is affected by various factors, which can be divided into "human" and "technical" factors. Human factors include: stability of service, availability of service, delays, user information. Technical factors include: reliability, scalability, effectiveness, maintainability, Grade of Service, etc. Many things can happen to packets as they travel from origin to destination, resulting in the following problems as seen from the point of view of the sender and receiver.

The problem as seen from the point of view of the sender and receiver is:

- Dropped packets

  The routers might fail to deliver (*drop*) some packets if they arrive when their buffers are already full. Some, none, or all of the packets might be dropped, depending on the state of the network, and it is impossible to determine what will happen in advance. The receiving application may ask for this information to be retransmitted, possibly causing severe delays in the overall transmission.

- Delay

  It might take a long time for a packet to reach its destination, because it gets held up in long queues, or takes a less direct route to avoid congestion. In some cases, excessive delay can render an application such as VoIP or online gaming unusable.

- Jitter

  Packets from the source will reach the destination with different delays. A packet's delay varies with its position in the queues of the routers along the path between source and destination and this position can vary unpredictably. This variation in delay is known as jitter and can seriously affect the quality of streaming audio and/or video.

- Out-of-order delivery

  When a collection of related packets is routed through the Internet, different packets may take different routes, each resulting in a different delay. The result is that the packets arrive in a different order than they were sent. This problem requires special additional protocols responsible for rearranging out-of-order packets to an isochronous state once they reach their destination. This is especially important for video and VoIP streams where quality is dramatically affected by both latency and lack of isochronicity.

- Error

  Sometimes packets are misdirected, or combined together, or corrupted, while *en route*. The receiver has to detect this and, just as if the packet was dropped, ask the sender to repeat it.

1.4 Scope of work

There are several scopes that need to be considered in order to fulfill the requirement of this project. The main scope of this project is:

- To guarantee the requested bit rate and provide better service to selected network traffic over various technologies.
- To provide better error rates for videoconferencing, lower network transit time (latency), and decreased latency variation (jitter) and to provide predictable network service by providing dedicated bandwidth and improved packet loss characteristics.
- To provide the VVOIP system more efficient and to make long-distance calls at much lower rate.

1.5 Project objectives

For this project, we have a few objectives. The objective for this project is:

- To gain and develop knowledge about Internet Protocol (IP), Quality of services (QoS) and Videoconferencing network system.
- To obtain the knowledge for parameters herewith coding IP-QoS Videoconferencing using OPNET Modeler
- To analysis the performance of simulation project via OPNET Modeler.
- The main objectives for this project are to consideration the quality of services and make comparison between 2 scenarios with IP-QoS and without IP-QoS.

1.6 Methodology

Phase 1:

The first phases, student necessary need to meet and discuss with supervisor. After that, show the project progress, and then get advice from supervisor in order to choose the suitable software version to develop this project. Besides that, find more information from internet, books, journal, and thesis and so on. Furthermore, we would need to understand the concept and theory for the IP-QOS and videoconferencing (VVOIP).

Phase 2:

The second phase is about literature reviews for the videoconferencing. In this phase, student need to know about OPNET MODELER, videoconferencing characteristic, requirements, Internet Protocol, Quality of Services (QoS), traffic flow, error rates for videoconferencing, latency, jitter and other. Besides that, found out information about the main topology that used for this project. The literature survey was taken from journal, books, conference transcript, thesis, patent and website.

Phase 3:

The third phase is about designing the topology that used for this system. The topology was developed by using the chosen software and studies more about this software were designing the videoconferencing topology. The developed of topology is done in this phase.

Phase 4:

At phase 4, the development for simulation study on IP-QOS videoconferencing performance using OPNET modeler was continued.

CHAPTER 2

LITERATURE REVIEWS

The chapter starts with brief discussion on videoconferencing, OPNET Modeler, H.323 protocol, Internet Protocol (IP) and QoS parameters (packet loss, jitter and delay).

2.2 Videoconferencing

In videoconferencing technology, two or more people at different locations can see and hear each other at the same time, sometimes even sharing computer applications for collaboration. Videoconferencing offers possibilities for schools, colleges, and libraries to use these systems for a variety of purposes, including formal instruction, connection with guest speakers and experts, multi-school project collaboration, professional activities, and community events. Placing a video call is a lot like placing a telephone call, this system may be able to transfer files or collaborate via options such as document sharing or white boarding. [1]

2.2.1 Basic Videoconferencing Technology

Compressed video systems allow a larger audience to experience the benefits of high-quality videoconferencing at a reasonable cost. A videoconferencing system requires the audiovisual equipment, which includes a monitor, camera, microphone, and speaker, and a means of transmission. Rather than an Internet-based connection, such as that used by webcams, which have to share bandwidth with other Internet data, a compressed video system on a dedicated bandwidth provides smooth audio and video. The compressed videoconferencing may be transmitted via an ISDN (Integrated Services Digital Network) line or over IP (Internet Protocol) lines. It is an economical solution for high-quality videoconferencing. [6]

2.2.2 Connecting

The most significant distinction among videoconferencing systems is the method of transmission. Transmission is important because two systems cannot connect if they are using different transmission methods. Videoconferences can be transmitted over two protocols, H.320 - ISDN (phone) or H.323 - IP (Internet) lines. In the past, most videoconferences used ISDN lines; however, many people are now using IP connections due to cost savings. In order to connect two units using different transmission methods, a bridge must be used that will handle these mixed protocols. In an ISDN call, bandwidth is dedicated to only one videoconference, while in an IP call; bandwidth may be used to transmit for multiple uses. However, ISDN calls can be very costly since you may be making the call over a distance, in which case, long distance phone line charges apply, and ISDN lines take up 6 phone lines. Connections around the world average 384 Kbps. [3]

2.2.3 Videoconferencing-enabled IP network

Figure 2.1 illustrates a typical network infrastructure of a small- to medium-sized company residing in a high-rise building with the minimal added videoconferencing components of a H.323 gatekeeper and H.323 workstations (Recommendation H.323, 1998; Abler and Wells, 1999; Cisco Systems). The gatekeeper node handles signaling for establishing, terminating, and authorizing connections of video sessions, as well as imposing maximum bandwidth for each session. H.323 workstations or multimedia PCs have H.323 voice and video software and are equipped with a camera and a microphone. The network is Ethernet-based and has Layer-2 Ethernet switches connected by a router. The router is Cisco 2621, and the switches are 3Com Superstack 3300. All the links are switched Ethernet 100Mbps full duplex. Shared links are never suitable for real-time applications. The network shown is realistic and used as a case study only; however, our work presented in this paper can be adopted easily for larger and general networks by following the same principles, guidelines, and concepts laid out in this paper. [1]

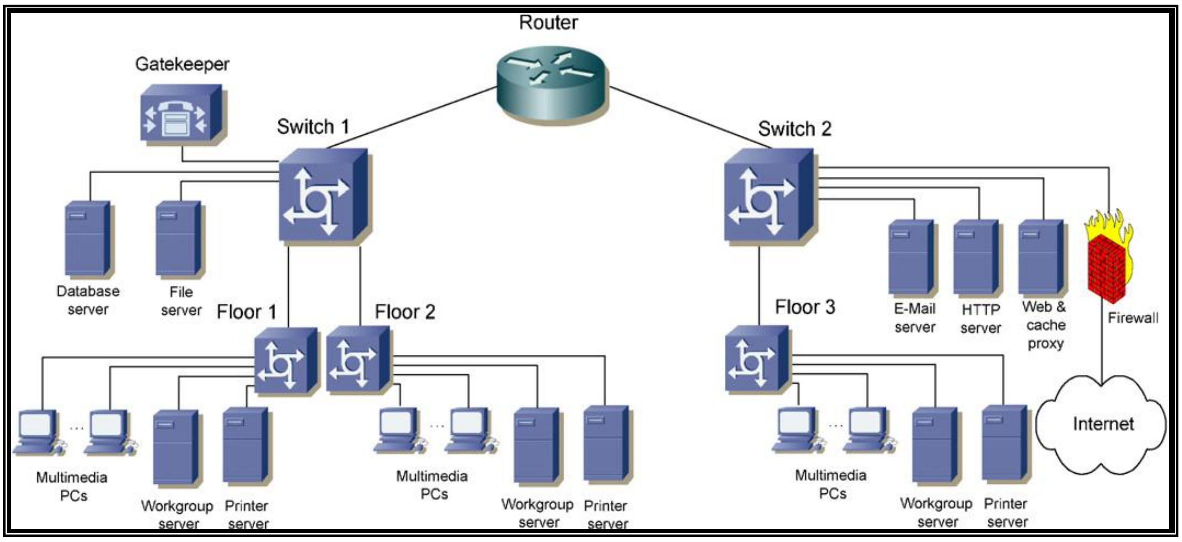

Figure 2.1: Network topology with necessary videoconferencing components. [4]

2.2.4 Traffic flow and call distribution

An important step that plays a factor in determining the number of sessions to be supported is the flow of sessions (or calls) and their distribution. Traffic flow has to do with the path that session travels through. Session distribution has to do with the percentage of sessions to be established within and outside of a floor, building, or department. For our example, we will assume that the generation of sessions is symmetric for all three floors. The intra-floor traffic will constitute 20% of overall traffic, and the other 80% will constitute inter-floor traffic. Such a distribution can be described in a simple probability tree shown in figure 2. [1]

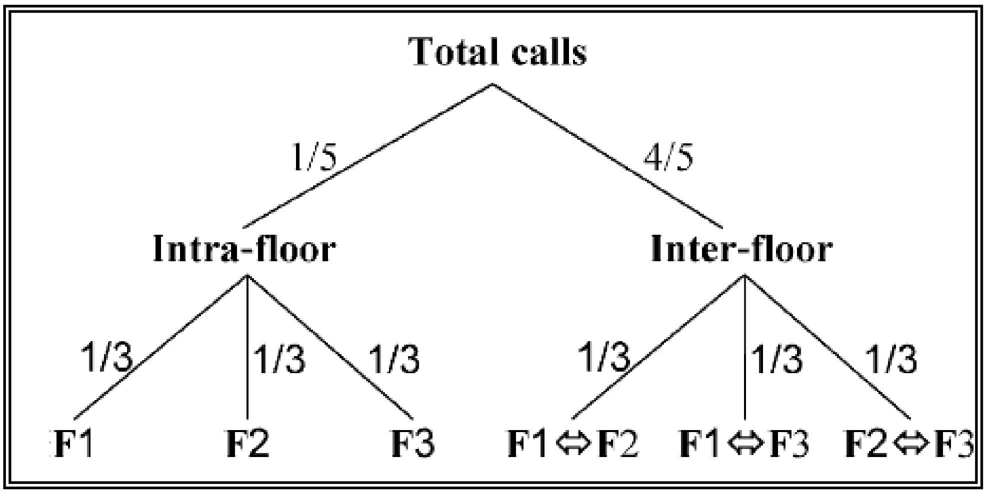

Figure 2.2: Probability tree describing session distribution. [1]

2.2.5 Additional considerations

Throughout our work, we assume voice and video calls are symmetric. We also ignore the signaling traffic generated by the gatekeeper. We consider the worst-case scenario for videoconferencing traffic. The signaling traffic involving the gatekeeper is only generated prior to the establishment of the session and when the session is finished. This traffic is relatively limited and small compared to the actual voice call traffic. In general, the gatekeeper generates no signaling traffic throughout the duration of the videoconferencing session for an already established on-going session (Goode, 2002). In order to allow for future growth, we will consider a 25% growth factor for all network elements including router, switches, and links. This factor will be taken into account in our simulation study. The rest of the paper is