DEPLOYING TRIPLE-PLAY SERVICES OVER EXISTING IP NETWORKS

BY

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A Thesis Presented to the
DEANSHIP OF GRADUATE STUDIES

KING FAHD UNIVERSITY OF PETROLEUM & MINERALS

DHAHRAN, SAUDI ARABIA

In Partial Fulfillment of the
Requirements for the Degree of

MASTER OF SCIENCE

In

COMPUTER NETWORKS

JUNE, 2010

KING FAHD UNIVERSITY OF PETROLEUM & MINERALS
DHAHRAN 31261, SUADI ARABIA

DEANSHIP OF GRADUATE STUDIES

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Dedication

I dedicate this dissertation with all of my love to my parents, my wife, my daughter, my brothers and sisters.

Acknowledgment

I thank Allah (SWT) for granting me the guidance, patience, health, and determination to successfully accomplish this work.

I would like to express my deep appreciation to my thesis advisor Dr. Khaled Salah, for his constant help, guidance, encouragement and invaluable support. Thanks are due to my thesis committee members Dr. Zubair Baig and Dr. Mohammed Sqalli for their cooperation, comments and support.

I would like to thank King Fahd University of Petroleum and Minerals for sponsoring me throughout my graduate studies. I also would like to thank Al-Hodeidah University which gives me the opportunity for completing my MSc degree in KFUPM.

Also, I would like to thank my parents, my wife and my daughter who always support me with their love, patience, encouragement and constant prayers. I would like to thank my brothers and sisters for their love and support throughout my study.

Finally, I would like to thank all my friends, mostly those who helped me in this thesis. A special thanks to Mr. Fahad Al-Haidari and Mr. Tawfik Abdo Saleh for their feedback and encouragements.

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Thesis Abstract

NAME: **Jamil Mohammed Ahmed Hamodi**

TITLE: **Deploying Triple-Play Services over Existing IP Networks**

MAJOR FIELD: **Computer Networks.**

DATE OF DEGREE: **June, 2010.**

Deployment of Triple-Play services over IP networks has been on the increase across the world particularly in North America, Europe, and Asia/Pacific. Triple-play services provide voice, data, and multimedia services as those of streamed audio, video, and IPTV. Network engineers and designers of these existing IP networks are faced with numerous challenges with regards to assessing the readiness by gauging the capacity of existing IP networks to support triple-play services. In this research work, we present an analytical and simulation approach to deploy triple-play services. The capacity of a network is gauged in terms of determining the maximum number of voice calls and audio/video streaming sessions that can be sustained while satisfying QoS requirements of all Triple-Play services, and at the same time leaving adequate capacity for future growth of the data network. We apply our approach with numerous scenarios on a typical modern medium-scale hospitality network. In particular, we consider the deployment of SIP-based VoIP, IP-Unicast of audio/video streaming VoD, and IP-Multicast AV IPTV. Our analysis is based on the principles of queuing theory, and our simulation is based on using OPNET.

خلاصة الرسالة

الإســــــم : جميل محمد أحمد حمودي

عنوان الرسالة : نشر خدمات التشغيل الثلاثي عبر شبكات الآي بي (IP) المتوفرة حالياً

التخصص : هندسة الحاسب الآلي

تاريخ التخرج: يونيو 2010

لاقى نشر خدمات التشغيل الثلاثي (Triple-Play) عبر شبكات بروتوكول الانترنت (IP) تزايدا ملحوظا في الاونه الأخيرة في جميع أنحاء العالم ولا سيما في أمريكا الشمالية وأوربا و آسيا لما يقدمه من خدمات متعددة كالوسائط المتعددة مثل ملفات الصوت والصورة والبت التلفزيوني عبر الانترنت. يواجه المهندسين ومصممين الشبكات العديد من التحديات فيما يتعلق بتقييم مدى استعداد وجاهزيه شبكات بروتوكول الانترنت لدعم خدمات التشغيل الثلاثي (Triple-Play).

يستعرض هذا البحث الخطوات اللازمة والطرق العملية لنشر خدمات التشغيل الثلاثي (Triple-Play) على شبكة آي بي (IP) قياسية بصورة ناجحة، فخطتنا البحثية اعتمدت على الطرق التحليلية والمحاكاة لدراسة حدود الإنتاجية والإعاقة الناتجة عن الشبكة. فبالنسبة للطرق التحليلية فقد استندت على نظرية الاصطفاف Queuing Theory. وأما المحاكاة اعتمدت على برنامج OPNET. ويعتبر قياسا لقدرة الشبكة من حيث تحديد الحد الأقصى لعدد المكالمات الصوتية وكذلك الحد الأقصى لتدفق ملفات الملتيميديا (audio/video streaming) والتي يمكن أن تستمر تغذيتها للشبكة دون التأثير على جودة الخدمات لجميع خدمات التشغيل الثلاثي (Triple-Play)، وفي نفس الوقت ترك القدرة الكافية للنمو في المستقبل شبكة البيانات. لقد قمنا بتطبيق هذا النهج بعده سيناريوهات على شبكة ضيافة (hospitality) نموذجية متوسطة الحجم. وعلى وجه الخصوص، نشر الصوت عبر بروتوكول الانترنت (IP) بناء على بروتوكول (SIP) وكذلك ملفات الملتيميديا (الصوت/ الصورة) بناء على بروتوكول (IP Unicast) وملفات البث التلفزيوني (IPTV) بناء على بروتوكول (IP Multicast).

CHAPTER 1

INTRODUCTION

1.1 BACKGROUND

There has been a considerable increase in Triply-Play services over IP networks, with active trials and deployment across the world including North America, Europe, and Asia/Pacific. Recently, Internet service providers have launched Triple-Play services, which consist mainly of High Speed Internet (HSI), Voice over IP (VoIP), and Internet Protocol Television (IPTV). Many network architects, managers, planners, designers, and engineers are faced with common strategic, and sometimes challenging issues to deliver Triple-Play services to their subscribers, using their existing networks and ensuring toll-quality voice and broadcast-quality video. The issues can be broken down as: how to ensure individual service quality and priority, especially quality-sensitive voice and video services, and how to provision the unique requirements of each service. Video and voice require high QoS and significant user interaction, how to integrate new customer-located Triple-Play devices like set-top boxes and voice-over-IP (VoIP) phone systems with existing systems, how to manage and support the individual services once they are provisioned [1, 2, 3]. If so, how many voice calls and video sessions can the network support before upgrading prematurely any part of the existing network

hardware? At the specified bandwidth, how many audio/video sessions can be handled without any need for additional changes in the existing IP network?

These challenging questions have led to the development of some commercial tools for testing the performance of multimedia applications in data networks. A list of the available commercial tools that support Triple-Play is highlighted in [4-14]. We summarize some of the popular tools. Considering the relationship between users' perception and network performance, EURESOM project named Jupitor II [4] has a provision to test end-to-end Quality of Service (QoS) for Network-QoS-aware applications over IP networks. Jupitor II takes into consideration various multimedia services like point-to-point and streaming videoconferencing. Major analysis has been made based on usability of 2D & 3D interface, acceptability of the user, network performance, and network-QoS management. Extra demands were made on the existing Best-Effort IP networks so as to provide sufficient quality of services to multimedia flows. In Jupitor II, QoS is defined as the overall Quality of a telecommunication service offered to the human being as user of the service, where QoS is closely related to the users' perception and expectations. Also, in Jupitor II, Network-QoS is defined as a part of network performance addressing the network engineering meaning of QoS, which has traditionally been concerned with technical network parameters such as error rates and end-to-end delay. The service provider can measure the provided Network-QoS. Jupitor II, for example, did measure both the Network-QoS parameters as well as customers' perception of QoS and figured out relationships between them.

NetIQ's Vivinet Assessor helps determine if an existing data network is capable of supporting VoIP applications. It operates by generating RTP streams to mimic VoIP traffic between pairs of hosts and summarizes the quality of these synthetic calls [5]. This product collects various measurements from the network including network load, utilization, and layer-3 path information between pairs of hosts. A software-based solution from NetIQ is available, where VoIP measurement and testing software is installed on Windows and UNIX platforms on network elements located at the edge of a network. As opposed to hardware-based solutions, a software-based solution can be limited in capabilities and less flexible, as it primarily depends on the processing power of existing network elements such as servers and workstations. Vivinet Assessor uses a cutting-edge method, based on freely distributed software agents, for calculating one-way network delay. This calculated network delay is then combined with the delay introduced by silence suppression and jitter buffers to create a complete end-to-end simulation of the actual network's delay as perceived by the end users. Vivinet Assessor also has the ability to vary the start times of simulated calls to realistically emulate call traffic patterns. Vivinet Assessor claims to support up to 2,500 calls per assessment. It also enables the definition and selection of advanced QoS parameters, including support for DiffServ and 802.1p.

VitalSuite [6] performance management software is a monitoring and measurement tool that merges various management modules of a deployed network, such as application performance management, network performance management, Real-time event delivery, and advanced reporting modules, to proactively monitor and report on the status of a

network. The suite is completely programmable, with support for multiple vendors and application services. The network performance module can help provide on-demand access to overall network performance information to help preempt problems, optimize resources and plan for maximum multimedia traffic on an existing data network.

BMC PATROL DashBoard [7] has been designed in order to analyze the impact of multimedia on the existing network. The advantage is that, this tool can quickly identify specific problems on the network that impact application performance. Along with identifying problems, it could, also, help to manage and optimize the overall flow and efficiency of the network. Being a web and Java-based network performance tool, it is friendlier and can be easily used.

A project handled by Video Development Initiative, called ViDeNet [8], explores various issues associated with globally scalable video and voice over IP. Having been developed as a test bed, it can be used to explore network management and measurement. ViDeNet acts as an interconnection scheme for H.323 gatekeepers, assists in the management of zones, and provides various directory services. Excluding H.323, ViDeNet does include support for SIP, ENUM, LDAP, and other security protocols.

Incorporating channel change responsiveness and video quality, the first QoE (Quality of Experience) test bed for triply play has been produced by Spirent technologies [9]. The triple play testing by spirant system includes: Performance of network infrastructure access devices, networks and servers, incorporates rapid validation of scalability of network elements and relations between voice, video, and data streams. In addition,

network load testing, wherein the network routers and switches performance and capacity are tested, is included. The testing includes also both multicast and unicast traffic, and confirms video and voice traffic quality, based on Quality of Experience (QoE) requirements. The data resulting is used to define network topologies, areas of concern, i.e., bottlenecks, and incorporation of best practices for enhancement of the existing infrastructure.

The main application of the Agilent Technologies Triple Play Analyzer is that it allows network professionals to troubleshoot, monitor, analyze, maintain, and optimize real-time voice, data, and video services over next generation converged IP networks. TPA (Triple Play Analyzer) provides a single window snapshot with the ability to drill down and view extensive Quality of Experience (QoE) and Quality of Service (QoS) measurements for the effect of deployment of voice, video and data on a single network infrastructure. Unlike the other testing tools, Agilent analyzer can observe the impact of one service over the other when they are simultaneously running on the network. The network analysis is achieved by studying several aspects of the observed traffic such as: identification of voice, video, and data traffic, identification of video and VoD streams in the network, identification of video and voice streams over RTP, Channel zapping, and RTSP signaling view per channel or subscriber. The data collection is non-intrusive in nature [10].

Aptixia IxLoad [11] provides a high scalable, and integrated test solution for assessing the performance of Triple Play networks and devices. It works by emulating millions of

IPTV Broadcast Video and Video on Demand (VOD) subscribers to test the Quality of Experience (QoE) that an end user may experience.

Ixload can be used to benchmark video server performance, measure the channel change performance of an IPTV deployment, measure the ability of an existing data network to carry voice and video traffic, stress test middleware devices such as encoder systems, conditional access systems, billing systems etc., measure perceived video quality as experienced by an end user, and test the performance of IPTV services such as DHCP, DNS and AAA. In addition to all of that, IxLoad can be used to generate malicious traffic to test for security and test vital aspects of the infrastructure like DNS, DHCP, RADIUS, and LDAP services.

Provided by Sunrise Telecom, the IPTV Test Suite MTT ADSLx Triple Play Modules [12] could enable network service providers to quickly verify that IPTV and VOD services are properly delivered to their end users. Some key features of this product include:

- STB emulation for IPTV and VOD services.
- MPEG video stream analysis with key metrics including packet loss, packet jitter, and latency, IPTV zap delay.
- VOD Play/Pause/Stop commands.
- PASS/FAIL pre-configured tests for service level validation.
- Test support at both ADSLx and 10/100 Ethernet interfaces.

RADVISION [13] offers tightly integrated infrastructure processing components called viaIP, for desktop and meeting room conferencing. Along with this infrastructure, it is required to procure the software management solutions such as iVIEW, which will help merge existing software like Outlook Calendar, Microsoft Windows Messenger and others with the new system environment. iVIEW provides a set of tools that could manage QoS, security, authentication, bandwidth usage, and others. Thus, this setup requires both hardware and software adjustments to our existing network.

J323 engine [14] is a Java middleware engine which provides H.323 call control services to videoconferencing and IP telephony applications through the Java Telephony API (JTAPI, 13). J323 engine provides functions for Call Control and Media Control. It offers an object oriented Java API, which can be used by developers to write their own interface or integrate their functions with it.

For the most part, the available commercial tools [4-14] use two common approaches in assessing the deployment of Triple-Play services over existing networks. One approach is based on first performing network measurements and then predicting the network readiness for supporting Triple-Play services. The prediction of the network readiness is based on assessing the health of network elements. The second approach is based on injecting real video/voice over IP traffic [15] into existing network and measuring the resulting delay, jitter, and loss.

1.2 MOTIVATION

Simulation is an important step for network design and deployment. Because network equipments are usually very expensive and not easy to be deployed, it is better to simulate network architectures before doing a real deployment. The main motivation of this research is to assess the readiness of existing IP networks to support Triple-Play services. Through the modeling and simulation, the behavior of Triple-Play network and the performance of application and services can be observed in the simulation environment. In addition, to predict the maximum number of voice calls, video sessions, and IPTV video live sessions that can be sustained by this network while satisfying QoS requirements of all Triple-Play services and leaving adequate capacity for future growth.

1.3 MAIN CONTRIBUTIONS

The research will investigate how to assess the readiness of IP networks to support Triple-Play services. Readiness will be assessed in term of maximum number of simultaneous Triple-Play sessions of voice and video that can be sustained by a given IP network while satisfying QoS requirements of all Triple-Play services and leaving adequate capacity for future growth. The primary objectives of this research can be summarized as follows:

- Research the literature and the standards on best practices on the deployment of Triple-Play services (especially those of streamed audio and video of IPTV and VoD).

- Address important engineering issues which include requirements of IPTV and VoD, traffic characterization of video and audio streams for IPTV and VoD, nature of user or viewer profile and surfing, popularity of channels to watch, number of users or viewers to support, and cost-effective leveraging of existing networks without prematurely upgrading or over provisioning the existing network.
- Determine all Triple-Play services requirements in terms of delay, bandwidth, packet loss, and jitter.
- Determine all Triple-Play hardware equipments requirement for the successful deployment, and show the appropriate placement required for these equipments.
- Configure a simulation environment in which we can simulate the VoIP supported by Session Initiation Protocol (SIP).
- Develop an analytical model based on queuing theory to determine the Triple-Play capacity of an existing IP network.
- Use OPNET to determine the Triple-Play capacity of a given IP network.
- Validate and verify results by comparing simulation results with analytical model results.

1.4 THESIS STRUCTURE

The rest of this thesis is organized as four chapters, every chapter as one paper. Chapter 2 presents the deployment of voice over IP (VoIP) over an existing hospitality networks based on Session Initiation Protocol (SIP). Chapter 3 focusing on deploying video-on-demand (VoD) and VoIP in a hospitality networks. Chapter 4 is the main contribution of this research, which presents the deployment of Triple-Play service in a hospitality network using PIM-SM multicast protocol for IPTV service. The remarks and future research are concluded in Chapter 5.

CHAPTER 2

Session Initiation Protocol (SIP)-Based VoIP Deployment over Hospitality Networks

The deployment of Voice over IP (VoIP) on data networks is emerging as a popular and promising field of research in recent times. Sustaining contemporary real-time voice transfer applications has become significantly dependant on the presence of Quality-of-Service (QoS)-compliant VoIP. Typically, the VoIP application runs as an overlay on the Session Initiation Protocol (SIP). In this chapter, as part of our contribution, we propose a SIP-based VoIP deployment on existing IP networks, to assess and prove the effectiveness of such an infrastructure to provision voice transfer services in small to medium-scale hospitality networks. In particular, the aim of this study is to predict the total number of VoIP calls that can be sustained by such a network, while satisfying QoS requirements of all network services, at the same time leaving adequate capacity for future growth of the data network. We utilize both analysis and simulation to investigate throughput and delay bounds. Our analysis is based on the principles of queuing theory. Our results are quantified through simulation using the OPNET simulation and modeling tool. A comparison of the results obtained from both analysis and simulation prove that such a VoIP deployment is in conformance of stipulated performance requirements.

2.1 INTRODUCTION

IP telephony or VoIP deployment over data networks is gaining popularity to become one of the most promising fields of research in recent times. Many network managers find it very advantageous and cost effective to deploy VoIP applications over their existing IP networks. The ease in running, managing and maintaining such applications without the need for significant modifications on the underlying network infrastructure, confirms the promise such VoIP deployments hold. However, one has to keep in mind that IP networks are best-effort networks that were designed for non-real time applications. On the other hand, for a VoIP deployment to be sustainable, it is required to have sufficient bandwidth in hand for timely delivery of packets, with low latency, jitter, and packet loss. To achieve this objective, an efficient deployment of VoIP must ensure that these real-time traffic requirements can be guaranteed over new or existing IP networks.

Since VoIP places a high demand on the underlying network resources, many network architects, managers, planners, designers, and engineers are faced with common strategic, and sometimes challenging, questions appertaining to the ability of an existing network to sustain QoS demands, the impact of VoIP on existing network services and applications, the impact of VoIP load on the service quality of currently running network services and applications, the ability of the existing networks to support VoIP, and the need to satisfy standardized QoS requirements. It is also imperative that a VoIP

deployment at the specified bandwidth is capable to handle a known number of calls without imposing significant changes to the underlying IP networks [3] [16].

The two most common signaling protocols that have been competing for VoIP application deployment over IP, over the past few years, have been H.323 and Session Initiation Protocol (SIP). The H.323 protocol was developed by the International Telecommunication Union (ITU) in 1996, and is an umbrella specification implying that it is not a protocol by itself. On the other hand, Session Initiation Protocol (SIP) was created by the Internet Engineering Task Force (IETF) in 1999 as part of the standard RFC 2543 [20-23]. Our proposed VoIP deployment for a hospitality network will use the SIP signaling protocol, considering its added features as opposed to H.323.

Several popular commercial tools have been developed in recent times to measure the end-to-end delay, jitter, and packet loss for VoIP deployments, but they differ in the way they inject voice traffic into the existing IP network. A few of these tools include EURESOM Jupitor II [4], NetIQ's Vivinet Assessor [5], BMC PATROL DashBoard [7], VitalSuite [6], RADVISION [13], J323 engine [14], ViDeNet [8], Omegon [17], CAIDA [18], NISTNet [19], and H.323 standards. These tools adopt one of two approaches to assess the deployment of VoIP over the existing network. One approach is based on predicting the readiness of the existing network to support VoIP. This prediction of the network's readiness is based on assessment of the health of individual network elements. The second approach is based on injecting real VoIP traffic into existing networks and

measuring the resulting delay, jitter, and loss. Such an approach is also referred to as network emulation [16].

For purposes of modeling and simulation of such added traffic on existing networks, the OPNET tool has gained considerable popularity. Our approach presented in this chapter predicts, prior to the purchase and deployment of VoIP equipment, the maximum number of VoIP calls that can be sustained by an existing hospitality network while satisfying QoS requirements of all network services, and at the same time leaving adequate capacity for future growth. Also, in this chapter, we attempt to address important factors appertaining to the VoIP application performance when deployed on a given hospitality network, including a study of the VoIP flow and call distribution, future growth capacity, performance thresholds, impact of VoIP on existing network services and applications, and the impact of background traffic on VoIP based on simulation.

In sharp contrast to the work in [3], in this chapter, we use the SIP protocol for modeling and simulation, as opposed to the H.323 used in [3]. Secondly, our case study is focused on hospitality networks which have a completely different distribution of calls, as compared to the case network studied in [3]. Finally, our approach measures the quality of speech traffic on a VoIP deployment on a network, using the Mean Opinion Score (MOS) metric. A decreasing MOS value indicates poor speech quality, as opposed to a higher value.

The rest of this chapter is organized as follows. In Section 2 we present work related to the deployment of VoIP on existing IP networks. Section 3 presents a brief introduction

to the Session Initial Protocol (SIP), and also gives examples of its setup using a proxy server. An elaboration of a typical network topology for a small-scale hospitality network to be used as a case study for deploying VoIP, is given in Section 4. In Section 5 we describe the practical steps to be taken prior to simulation. We simulate and perform an analysis of the obtained results in Section 6. The study is concluded in Section 7.

2.2 RELATED WORK

In this section, we describe the work done related to the deployment of VoIP over existing IP networks. It may be noteworthy to mention that not much research has been performed earlier on SIP signaling-based deployment of VoIP over an existing network. Some of the previous work, as given in [16], is based on an OPNET-based simulation approach for determining the number of VoIP calls that can be supported for a given network topology. In [3], the author presents a step-by-step methodology to deploy any new services such as VoIP. The chapter also elaborates on the use of OPNET for simulation, and how it can be used to generate VoIP traffic. Some of the analysis includes the study of VoIP flow, call distributions, future growth capacity, and background traffic. The case study used by the authors of both [3], [16] were the same, i.e. a typical network topology of a small enterprise, however, the analysis in [16] included a comparison of simulation results obtained from both MATLAB as well as OPNET. In [24], the authors presented an approach based on quantitative and qualitative

analysis of a VoIP deployment, to calculate the call capacities and also, to assess the readiness of an Ethernet network to sustain such a deployment.

The work in [25] and [26] was on the proposal of a new architecture for the deployment of SIP telephony over Multi-Protocol Label Switching (MPLS) network architectures, to overcome the issue of delays in setting up long calls, as observed in RSVP-TE enabled MPLS networks. In order to integrate the SIP protocol with the traffic engineering function of MPLS networks seamlessly, and to facilitate SIP call setup, the SIP-MPLS traffic aggregation server (TA server) was highlighted as the novelty in the new architecture. The effectiveness of such an approach was tested through OPNET simulation, for different call admission and bandwidth re-negotiation algorithms. The results were conclusively alluding towards the superiority of the TBCA-BP algorithm over others, as the most suitable call admission and bandwidth re-negotiation algorithm for TA servers. The reason stated was the satisfying performance of high bandwidth prediction accuracy, low call blocking probability, and high bandwidth utility efficiency of this algorithm.

A new analytical model based on a dimensioning framework to provide accurate estimation of the bandwidth requirement of a network, required to guarantee bounds in delay or packet loss for generalized VoIP sources (GVoIP) through Comfort Noise Generation (CNG), was proposed in [27] and [28]. The proposed scheme emulates the on-off multiplexing models for GVoIP traffic characteristics by generating traffic during off peak hours. The estimate of the bandwidth requirement in a multiplexer node is

defined in the proposed scheme as a function of the number of multiplexed GVoIP sources that exist, and the desired loss and delay performance of the scheme.

A novel multi-level profiling methodology for characterizing SIP traffic behavior was proposed earlier [48]. It characterizes VoIP services by extracting and profiling a large verity of traffic features and metrics at three different levels in a progressively refined fashion. SIP server host characterization provides a broad view of their behavior by monitoring and keeping statistics related to only the message types (request vs response) and user activity diversity, server entity characterization, which provides a functional analysis of server activities by separating their logical roles into registrar, call proxy, and so forth, and individual user characterization, which maintains more detailed profiles of individual user activities. Also, the methodology allows balancing the speed of profiling, the resource consumption, the desired sophistication of behavior characteristics, and finally the level of security to be offered, based on the specific objectives and needs of the VoIP service operator. Their analysis of SIP traffic traces obtained from an operational VoIP service, and shows that overall SIP-based VoIP traffic exhibit stable characteristics and behavior that are well captured by the statistics and features selected in their profiling methodology, thereby justifying the selection of these statistics and features.

A new QoS requirements in SIP-based for the Next Generation Network (NGN) was presented in [49]. The requirements are derived from the state of the art reflexion of both, standardization and research work. They concerned by these requirements that the

principle of resource reservation within NGN transport networks in a per-session manner, also they focused on the definition and development of mechanisms and algorithms for the appliance of virtual data pipes, the work-out and adoption of the data pipe concept within SIP-based NGN. Also they were proposed a framework for QoS control, aiming to address the scalability issues related to QoS provision in SIP-based NGN in [50], this framework approach all action required for the control of the QoS affecting media sessions is performed within the NGN service stratum (i.e., cross-strata communication is avoided). Therefore, the framework has to be provided with an integrated mechanism for collection of information on QoS affecting any ongoing and future media session.

2.3 SESSION INITIATION PROTOCOL (SIP)

In this section, we describe the SIP protocol, which we utilize for testing the effectiveness of VoIP deployments on our hospitality network. Session Initiation Protocol (SIP) is a signaling protocol created by the Internet Engineering Task Force's (IETF) Multiparty Multimedia Session Control Working Group (MMUSIC WG) in 1999. Subsequently, it was updated by SIP WG in 2002 as part of the standard RFC 3261 [29]. This protocol is used for creating, modifying, and terminating sessions with one or more participants of a VoIP session [30-32]. A SIP session can be an Internet telephone call, a multimedia conference, or a distributed computer game played by 1 or more players. The protocol defines different message types and response codes to represent different requests and response messages. Also, SIP is similar to the HyperText

Transfer Protocol (HTTP) [30] and shares some of its design principles. It adopts a request/response (client/server) architecture in which requests are generated by the client and sent to the server. The most important types of requests are: *a)* the INVITE request, that is used for inviting a user to a call, *b)* a simple acknowledgement sent from the caller to the callee is accomplished through an ACK request, and *c)* the BYE request that is used to terminate the connection between two users in a session.

SIP is a peer-to-peer protocol. The peers in a typical SIP session are called User Agents (UAs). A UA can function in one of the following roles: *a*) User agent client (UAC), which is a client application that initiates the SIP request, and *b*) User agent server (UAS), which is a server application that contacts the user when a SIP request is received, and also returns a response on behalf of the user. The physical components of the SIP network can be grouped into two categories: *clients* and *servers*. SIP clients include phones which can act as either a UAS or UAC (i.e., Softphones, Cisco SIP IP phones). The servers (gateways) provide many services, the most common being a translation function between SIP conferencing endpoints and other terminal nodes. SIP servers include Proxy servers, which are intermediate devices that receive SIP requests from a client and then forward the requests on the client's behalf. Redirect servers provide the client with information about the next hop or hops that a message should take, so that the client can contact the next hop server or UAS directly, to reach the final destination, i.e., callee. Register servers process requests from the UACs for registration of their current location in a VoIP deployment [33].

A conferencing call can be set up by the user by dialing to the Conference Server a SIP URI or an E-mail address, such as SIP: hamodi@10.80.24.186 and SIP: salah@kfupm.edu.sa. When a user initiates a call, a SIP request is sent to a SIP server (proxy or redirect). The request includes the address of the caller and the address of the intended callee. The following section provides samples of successful, point-to-point calls established using a proxy server.

The proxy server is an important component in the SIP architecture. It works in a similar way to a proxy of a HyperText Transfer Protocol (HTTP) session, or similar to other Internet Protocols. In Figure 2.1, we show a typical example of a SIP call established through an intermediary proxy server. In this example, the caller UA sends an INVITE request to the proxy server. The proxy server determines the path for routing through to the destination, and then forwards the request to the callee. The proxy server also forwards the acknowledgments of both parties. Upon receipt of the acknowledgements, a session is established between the caller and the callee. For the illustrated example, the Real-time Transfer Protocol (RTP) is used for communication between the caller and the callee [33].

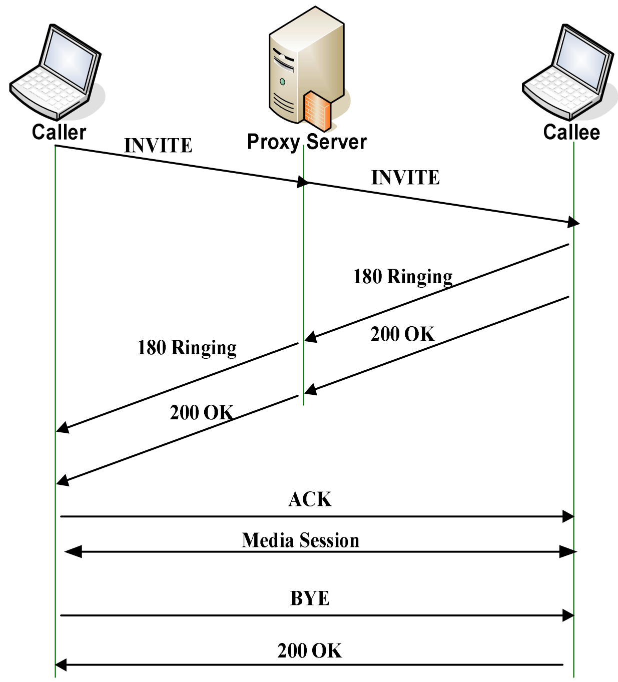


Figure 2.1 SIP call flow example with an intermediary proxy server

2.4 HOSPITALITY NETWORK

In this section, we describe our case study for VoIP deployment on a small-scale hospitality network. Such a network, which is used in several private facilities such as hotels, cruise ships and college campuses, is designed to provide entertainment, information and communication services to end-users. In Figure 2.2, we illustrate a typical hierarchical topology for this network, which consist of three layer, core layer of high-end routers and switches that are optimized for availability and performance, distribution layer of routers and switches which implement policies, and access layer which connects users via lower-end switches and wireless access point. The hierarchical topology is recommended for scalability [79]. Our network is Ethernet-based, and can

support a minimum of 200 end-users at a given time. The network spans over four floors, with each floor considered as a separate subnet consisting of a single Layer-2 switch, and a Cisco catalyst 2960, that can support at least 50 users at any point in time. The switch is connected with equipment required for customer support, such as set-top Boxes, VoIP phones, or wireless access points. Floor 1 in the topology is a lobby (possibly of a hotel), consisting of more than one access point to support several workstations in the conference room, as well as end-users at the hotel's reception. Each pair of hotel floors is connected with Layer-2 switches, namely, the Cisco catalyst 3560G, and all Layer-2 switches are connected with a backbone Layer-3 switch, namely, the Cisco catalyst 5500. In addition, the backbone Layer-3 switch is connected with four application servers. The application services running on these servers of the server farm are: E-mail, ERP, DHCP, and DNS. The network bandwidth on all links is 1Gbps with the exception of the links present within the floor subnet, which operate at 100Mbps.

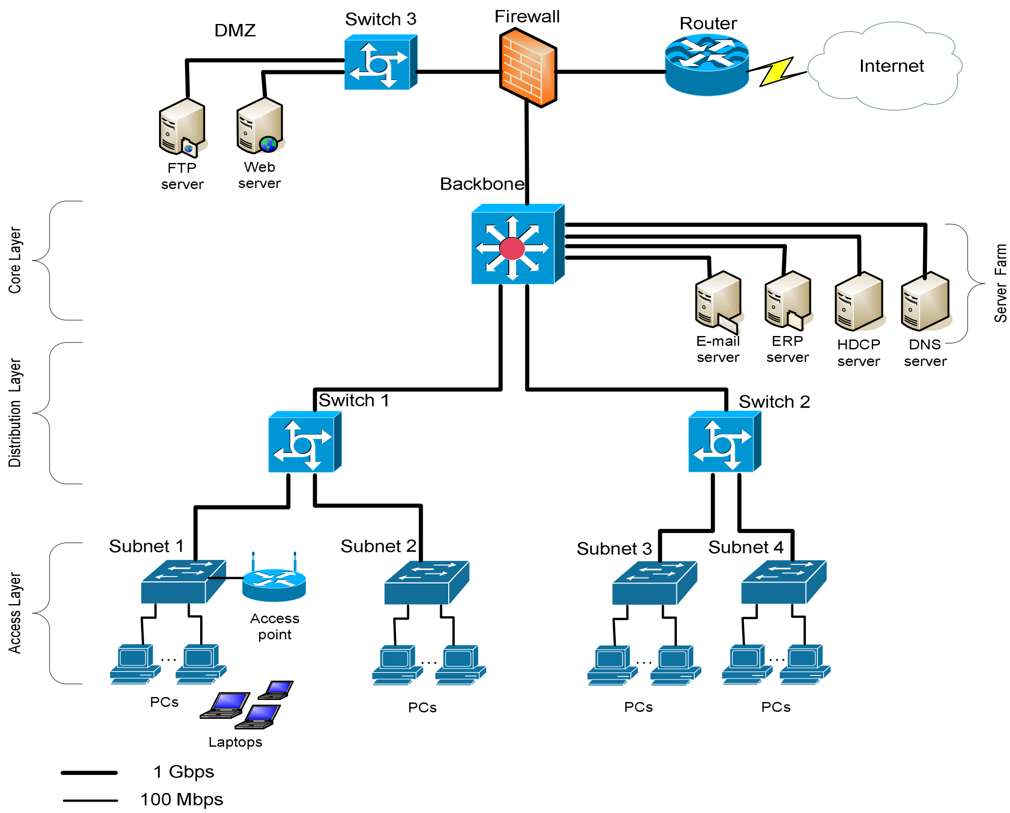


Figure 2.2 Topological diagram of a small-scale hospitality network

2.5 VOIP CASE STUDY AND ANALYSIS

In this section, we perform a case study of our hospitality network, and the VoIP deployment. We base our VoIP study of the hospitality network on analysis and simulation, to determine the maximum number of VoIP calls that can be supported by this network, under a given set of constraints. The study follows the methodology given in [3], wherein the pilot deployment of VoIP is accomplished based on the feedback obtained from analysis and simulation.

2.5.1 VoIP Traffic Characteristics, Requirements, and Assumptions

Some constraints that characterize the nature of VoIP calls include the network traffic, QoS requirements, and hardware components or devices. For our hospitality network deployment, we assume a unique call to originate from point-to-point conversation for all VoIP calls without support for conferencing. Therefore, for deploying VoIP using the SIP protocol, two components are needed, *a*) a *VoIP gateway* which is required to handle external calls, which also converts VoIP calls to/from the Public Switch Telephone Network (PSTN), and *b*) the *SIP proxy server*, which is an entity that can act as both a server and a client for the purpose of making requests on behalf of other clients. These requests can be serviced either internally or by passing them on, possibly after translation, to other servers [34].

One of the hardware requirements for deploying VoIP using SIP is a user agent (UA), which is an end point entity required for initiating and terminating sessions by exchanging requests and responses. Some of the devices that can have a built-in UA function include IP-phones, call agents, and workstations that are VoIP-enabled. There are three main performance indicators that characterize the quality of voice communications over the Internet. The first one is the *end-to-end delay,* which is also known as the mouth-to-ear delay. It is the time interval elapsed from the moment a word is spoken till the instance when the listener hears the word. There are five components that contribute to the end–to–end delay. The first component is the encoder which periodically samples the original voice signal and assigns a fixed number of bits to each sample, creating a constant bit rate stream, which in turn depends on the voice codec

employed. Table 2.1 lists the characteristics of the most popular codec's used in VoIP based on SIP [30, 38]. G.114 [30] and G.714 [39] imposes a maximum total one-way packet delay of 150ms end-to-end for VoIP applications. In [30] according to ITU-T Rec. G.114 is considered to have an acceptable delay of up to 400ms. In the G.711 standard, the delays for encoding and packetization are 1ms and 20ms, respectively. The appropriate delay for these two components for compression, is 25ms, introduced at the source. The network delay, which is the sum of the transmission, propagation, and queuing delays should not exceed 80ms, for this standard.

The second metric is the *delay jitter,* which is defined as the variation in the end–to-end delay between two consecutive packets. Ideally, it should be less than 30ms [30]. In addition, values between 30 and 75ms are also considered acceptable. The total fixed delay obtained at the receiver after the processing delays of depacketization and decoding are added, is 45ms. The third metric, i.e. the *packet loss* is considered to be acceptable only if it is less than 2% of the total transmitted packets. In Table 2.1, we enlist the characteristics of the most popular voice codec standards that may be used for our hospitality network.

Table 2.1 Characteristics of the popular voice codec

Codec	Bitrate (kb/s)	Frame (ms)	Bits per frame	Algorithmic delay (ms)	Codec delay (ms)	Compression type
G.711	64	0.125	8	0.125	0.25	PCM
G.723.1	6.3	30	189	37.5	67.5	MP-MLQ
	5.3	30	159	37.5	67.5	ACELP
G.726	16	0.125	2	0.125	0.25	ADPCM
	24	0.125	3	0.125	0.25	ADPCM
	32	0.125	4	0.125	0.25	ADPCM
G.728	16	0.625	10	0.625	1.25	LD-CELP
G.729	8	10	80	15	25	CS-ACELP
G.722	48	0.0625	3- 4	1.5	1.5625	SB-ADPCM
	56	0.0625		1.5	1.5625	SB-ADPCM
	64	0.0625		1.5	1.5625	SB-ADPCM

We adopt the G.711 codec standard with the above-listed delay and bandwidth, since this standard yields a MOS rating of around 4.3, where MOS, defined as the Mean Opinion Score, is a commonly used VoIP performance measurement metric. The MOS rating ranges on a scale of 1 to 5, with 5 being the best [40, 41]. Table 2.2 lists the MOS scores for the various codecs [38].

Table 2.2 MOS for various codec

Codec	Data Rate	MOS score
G.711	64	4.3
G.726	32	4
G.728	16	3.9
G.729	8	4
GSM	13	3.7

The traditional sample-based encoder G.711, which uses Pulse Code Modulation (PCM) to generate 8-bits samples every 0.125ms, facilitates a data rate upper bounded by 64kbps. So, the required bandwidth for a single call, in one direction, is 64kbps. The G.711 codec samples 20ms of voice per packet. Therefore, 50 such packets need to be

transmitted per second. Each packet contains 160 voice samples in order to give 8000 samples per second. Each packet is sent in one Ethernet frame. For every packet of size 160 bytes, a header needs to be added. These headers include four protocol fields: RTP + UDP + IP + Ethernet, with their respective preamble sizes added: 12 + 8 + 20 + 26. Therefore, a total of 226 bytes, or 1808 bits, needs to be transmitted 50 times per second, or 90.4 kbps, in one direction. For both directions, the required bandwidth for a single call is 100 pps or 180.8 kbps for a symmetric flow.

2.5.2 VoIP Traffic Flow and Call Distribution

It is important to determine the locations of the call endpoints, the sources and destinations, as well as their corresponding path or flow. This aids in identifying the call distribution and the calls made internally or externally. Call distribution must include percentage of calls within and outside of a floor, building, department, or organization. As a good capacity planning measure, it is recommended to base the VoIP call distribution on the busy hour traffic of phone calls for the busiest day of a week or a month. This will ensure support of the calls at all times with high QoS for all VoIP calls [3]. According to data collected from the manager of the telephony network in Al-Mana hospital, and from KFUPM telephony center, the worst case scenario for any hospitality network is the busy hour traffic of phone calls, i.e., from 8 AM to 2 PM. During this period, the internal calls constitute approximately 60% of all calls made. Figure 2.3 describes the call distribution for the hospitality network under study based on the busy hour, as well as on the projected future growth of VoIP calls. In the figure, the call

distribution is described as a probability tree. It is also possible to describe it as a probability matrix.

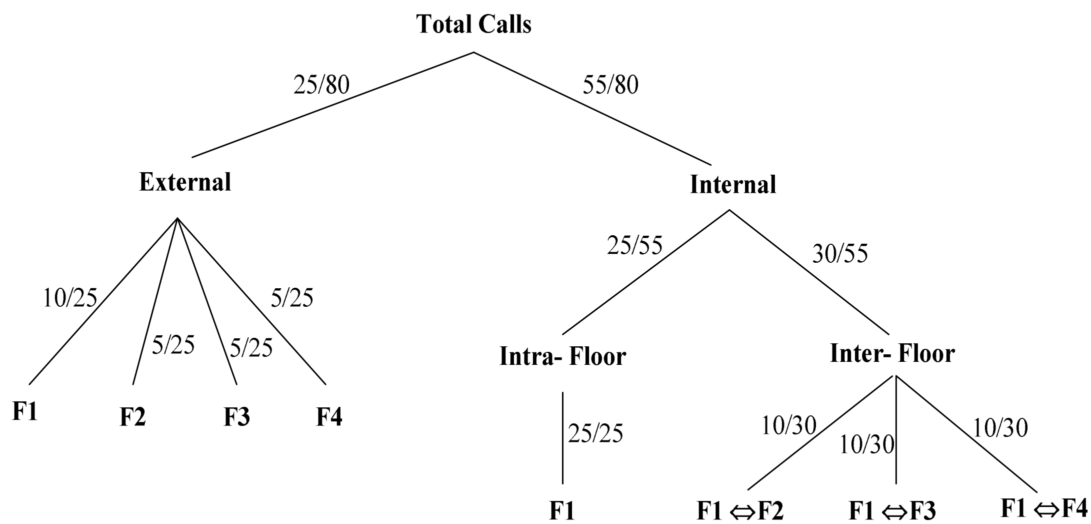


Figure 1.3 Probability tree describing the VoIP call distribution

2.5.3 Performance Thresholds and Growth Capacity

When we need to deploy any new multimedia service over a VoIP network, we need to consider the network performance thresholds or operational points for a number of key network elements. The benefit of this approach is twofold. First, the requirements of the new services to be deployed need to be satisfied. Second, adding the new service must leave the network healthy and ready for future growth.

The maximum tolerable end-to-end delay is determined by the most sensitive application to run on the network. For VoIP, this delay is found to be 150ms for our case study.

Other important performance criteria include the utilization bounds or thresholds of network resources, which are determined by factors such as current utilization, future plans, and future growth of the network. VoIP requires almost no packet loss. In the literature, 0.1 to 5% packet loss was generally asserted [42-44]. However, in [45], the required VoIP packet loss was conservatively suggested to be less than 10^{-5}.

The projected growth in users, network services, business, etc. must be taken into consideration to extrapolate the required growth capacity or the future growth factor. In our case study we reserve 25% of the available network capacity for future growth and expansion. For simplicity, we will apply this evenly to all network resources, i.e., routers, switches, and links. However, we must also keep in mind that this percentage in practice can be variable for each network resource and may depend on the current utilization and the required growth capacity.

2.5.4 Network Measurements

To seek an adequate assessment of the VoIP deployment, network measurements have to be taken over a long period of time, at least 24-hours in length. Sometimes, it is also desirable to take measurements over several days or even a week. In order to ensure good QoS at all times including peak hours, we consider the worst case scenario for network load or utilization. The peak hour is different from one network to another and it depends totally on the nature of business and the services provided by the network. Table 2.3 shows a summary of peak-hour utilization for traffic of links in both directions

connected to the router and the two switches of the network topology of Figure 2.2. These measured results will be used in our analysis and simulation study.

Table 2.3 Worst-case network measurement

link	bite rate (Mbps)	packet rate (pps)	utilization
Backbone ⇔ Switch 1	1.64	660	0.164
Backbone ⇔ Switch 2	1.04	520	0.104
Switch 1 ⇔ Floor 1	1.12	400	0.112
Switch 1 ⇔ Floor 2	0.52	260	0.052
Switch 2 ⇔ Floor 3	0.52	260	0.052
Switch 2 ⇔ Floor 4	0.52	260	0.052
Backbone ⇔ Server Farm	2.12	408	0.0212

2.5.5 Modifications based on Experiments

In this step, we assess the existing network and determine based on the existing traffic load and the requirements of the new service to be deployed, if any immediate modifications are necessary. Immediate modifications to the network may include adding and placing new servers or devices, upgrading PCs, and re-dimensioning heavily utilized links. Based on the existing traffic load discussed in subsection 2.5.4, all the links connecting the backbone switch and the Layer-2 switches and links connecting the backbone switch and the servers are underutilized. If any of the links were heavily utilized, e.g., 30-50%, the network engineer should decide to re-dimension the link to 10Gbps link at this stage.

Based on the hardware requirement for deploying VoIP service in Section 2.5.1, which describes VoIP requirements, two new hardware elements have to be added to the

existing network based on the SIP protocol, namely, a VoIP gateway and a SIP proxy server. As a network design issue, an appropriate node placement is required for these two nodes. The VoIP gateway and SIP proxy are connected to the backbone switch in order to balance the load on both distribution switches.

Figure 2.4 shows the new network topology after the incorporation of necessary VoIP components based on the SIP. As shown, the gateway and the SIP proxy server nodes for VoIP were added.

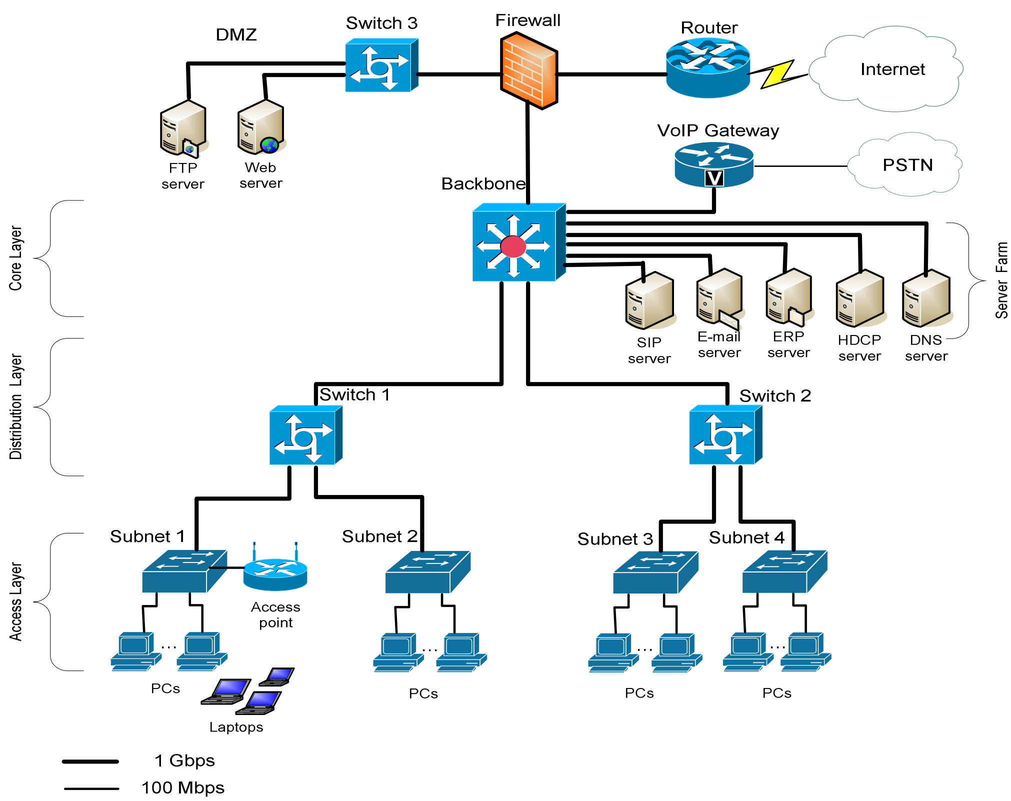


Figure 2.2 Network topology with VoIP components using SIP

2.5.6 Analysis

There are two metrics delimited by VoIP, namely, the *available bandwidth* and the *end-to-end delay*. The actual number of VoIP calls that the network can sustain is bounded by these metrics. Bandwidth bottleneck analysis is used for identifying which network element, whether it is a node or a link, restricts how many VoIP calls can be supported by the existing network. According to [3], the maximum number of calls that can be supported by a network element E_i is defined as follows,

$$MaxCalls_i = \frac{A_i(1 - growth_i)}{CallBW}, \tag{1}$$

where *CallBW* is the VoIP bandwidth for a single call imposed on element E_i, the bandwidth for one direction is given as 50 pps for switches or router and 90.4 kbps for links, as was discussed in subsection 2.5.1. The $growth_i$ is defined as the growth factor of network element E_i, and it is assumed to be 25% for our network. The parameter A_i refers to the minimum available bandwidth, and it is dependent on the capacity (C_i) of the network element.

Also, according to [3], the capacity of each element in turn is defined as the processing rate or the maximum possible transfer rate, and is dependent on the current utilization u_i. The minimum available bandwidth is therefore defined as follows:

$$A_i = (1 - u_i)C_i \tag{2}$$

The maximum number of VoIP calls that can be sustained by a network depend on the maximum number of calls that can be supported by each network element, as defined through equation (1). In addition, it is also dependent on the flow for this element E_i. So, the total number of VoIP calls can expressed as,

$$TotaCallsSupported = \min_{i=1,\ldots,N} \left[\frac{MaxCalls_i}{flow_i} \right] \quad (3)$$

As defined earlier in subsection 2.5.1, the maximum tolerable end-to-end delay for an IP telephony packet is 150 ms. The maximum number of calls that the network can sustain is bounded by this delay. The end-to-end delay for one direction of any VoIP call from a sender to a receiver can be expressed as follows as in [3]:

$$D = D_{pack} + \sum_{h \in Path} (T_h + Q_h + P_h) + D_{play} \;, \quad (4)$$

where D_{pack} is the delay due to packetization at the source. Two additional delays incurred at the source, are D_{enc} and $D_{process}$, where D_{enc} is the encoder delay of converting A/D signal into samples, and $D_{process}$ is the **PC** processing that includes encapsulation. In the G.711 codec standard, D_{pack} and D_{enc}, have values of 20 ms and 1ms, respectively. Hence, it is appropriate for our analysis to have a fixed delay of 25 ms introduced at the source, assuming a worst case scenario (i.e. peak traffic hours). D_{play} is the playback delay at the receiver, including jitter buffer delay. The jitter delay is at most 2 packets, i.e., 40ms. If the receiver's delay of $D_{process}$ is added, we obtain a total fixed delay of 45 ms at the receiver. The sum of all delays incurred in the packet network due to

transmission, queuing, and propagation going through each hop *h* in the path from the sender to the receiver, is given by $P_h + Q_h + T_h$. The propagation delay P_h is typically ignored for traffic within a LAN, but not for a WAN. For the transmission delay T_h and queueing delay Q_h we apply the queueing theory. Hence, the delay to be introduced by the network, expressed as $\sum_{h \in Path} (T_h + Q_h)$, should not exceed (150 – 25 – 45) or 80 ms.

In order to determine the maximum number of calls that can be supported by an existing network while maintaining the VoIP delay constraint, we implemented Algorithm 1, that basically determines the network capacity in terms of VoIP calls that can be sustained, by computing the number of calls based on a delay bound. Calls are added iteratively until the worst-case network delay of 80 ms is reached, at which time the algorithm stops execution. Through Algorithm 1, we present the maximum number of VoIP calls based on the voice delay constraint parameter. This parameter takes as input the number of network elements, the background traffic for every element, call flow paths, and the delay associated with processing of a packet at each of these elements. The Algorithm initializes the required bandwidth for a single unidirectional voice call in terms of packets per second (and bits/second), and returns the maximum number of voice calls that are generated by the network. The calls are added incrementally according to the voice distribution process described in subsection 2.5.2.

Algorithm 1 was implemented in MATLAB. Through the results, the maximum number of VoIP calls that could be supported on our hospitality network was found to be nearly 310. To be more precise, MATLAB results showed the number of calls that are bounded

by an 80 ms as 320 as in Figure 2.5. The limited number of calls sustainable here is due to the service rate (processing) of the network switches, and not due to the QoS requirement. However, the capacity C_i for the Layer-3 and the Layer-2 switches is 25,000 pps and 40,000 pps, respectively. After 320 calls are established, the calculated delay is negative which means that the arrival rate is greater than the service rate of either a link or a switch, thus leading to instability.

Algorithm 1: Computation of the maximum number of calls sustainable based on the VoIP delay constraint

```
Input:  n: number of network elements
        λ[1..n]: background traffic for network elements 1,2,..,n
        Delay[1..n]: delay for network elements 1,2,..,n
        P: set of call-flow paths (p), where p is a subset of {1,2,..,n}
Output: Maxcalls: maximum number of calls

        λ_VoIP ← 100 pps, or 180.8 kbps; PktSize_VoIP ← 1808 bits ;
VoIP_MaxDelay ← 80;    // maximum network delay for VoIP call in ms
MaxDelay ← 0; MaxCalls ← -1;  Delay[1..n] ← 0;

while    MaxDelay < VoIP_MaxDelay    do

   1. MaxCalls ← MaxCalls + 1
   2. Generate a call according to call distribution and let p_cc be its flow path
      // calculate VoIP packet delay for each network element after adding VoIP call
   3. for each  element i  in p_c do
               find α from the call distribution for each element i
                 λ_i ← λ_i + α * λ_VoIP
              if   i   is a link then
                   Delay_i ← (1-λ_i/2μ_i)/(μ_i-λ_i) * PktSize_VoIP
              else
                   Delay_i ← 1/(μ_i-λ_i)
              end if
      end for

   4. for each  p in P  where  p∩p_c ≠∅    do
        PathDelay(p) ← ∑Delay_i , where i  is a network element in path  p
              if    PathDelay(p) > MaxDelay     then
                        MaxDelay ← PathDelay(p)
              end if
      end for

end while
```

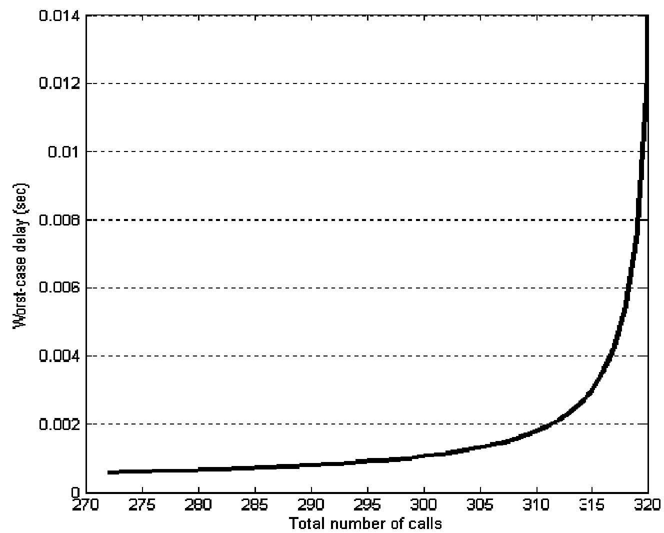


Figure 2.3 Worst case delay vs. number of VoIP calls

2.6 SIMULATION AND ANALYSIS

In this section, we describe the simulation model adopted for analyzing the effect of VoIP on our hospitality case study network. The simulation was performed for varying traffic flow and call distributions, and different simulation configurations. Our simulation approach used the popular MIL3's OPNET Modeler simulation package1, Release 14.0.A [46]. We used the OPNET Modeler to facilitate the utilization of in-built models of commercially available network elements, with reasonably accurate emulation of various real-life network configuration scenarios.

2.6.1 Simulation Setup

The topology for the hospitality case study network for our simulation is shown in Figure 2.6. The simulation model of the network has been based on the properties of SIP signaling. It may also be noted that this model of the hospitality network is an exact replica of a real-life hospitality network. The vendor-specific devices that we used for the simulation were included from OPNET's component library. In particular, we used the Cisco catalyst 5500 Layer-3 switch, for emulating the router, and a Cisco catalyst 3560G Layer-2 switch for each of the four subnet floors of the network. The VoIP gateway was modeled as a workstation, since our goal was to collect performance statistics from within the hospitality network alone. The four hospitality servers were modeled as a single Ethernet server with four operational services.

The floor LANs were modeled as individual subnets that include a Cisco catalyst 2960 Layer-2 switch and three designated Ethernet workstations to model the activities of the users of the LAN, as can be seen from Figure 2.7. One of these workstations was designated to generate the background traffic of the floor, while the other two workstations acted as second and third parties for the VoIP sessions. For instance, the Ethernet workstations for Floor 1 are labeled as F1_C1, F1_C2, and F1_C3, in Figure 2.7. F1_C1 in this case is the source of origination of the VoIP calls. The workstation F1_C2 is a sink for receiving VoIP calls, and F1_C3 is a sink as well as a source of background traffic. Please note that the model of floor LANs does not represent precisely the floor multimedia PCs or IP phones. Our simulation approach is an

automated one, as the simulation is configured to automatically keep generating 16 calls every 10 seconds as configured in the voice profile.

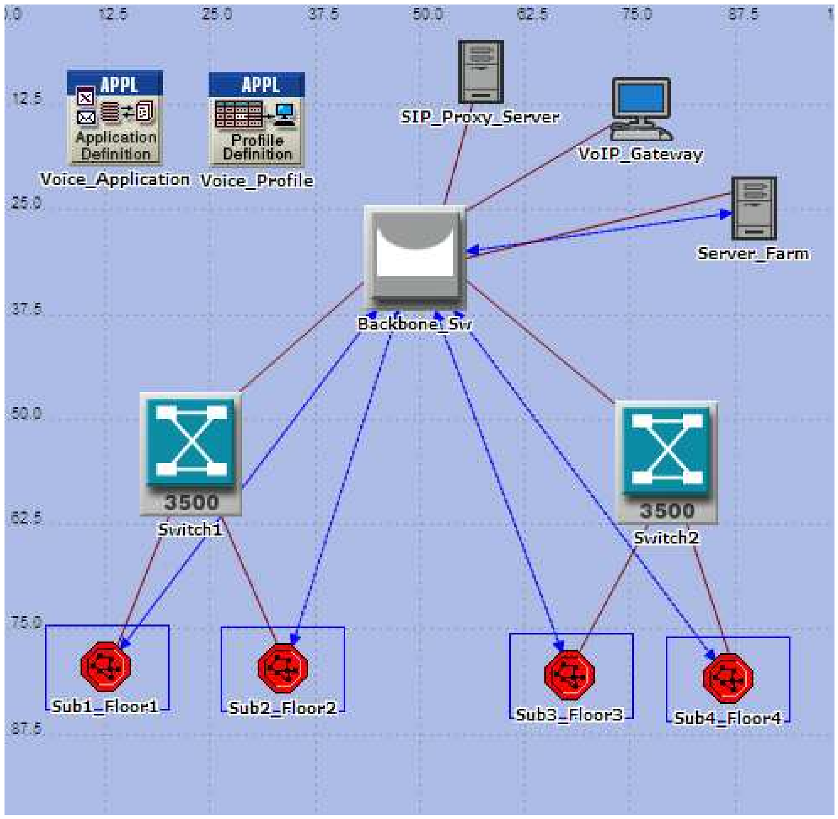


Figure 2.4 Hospitality network topology based on the SIP standard

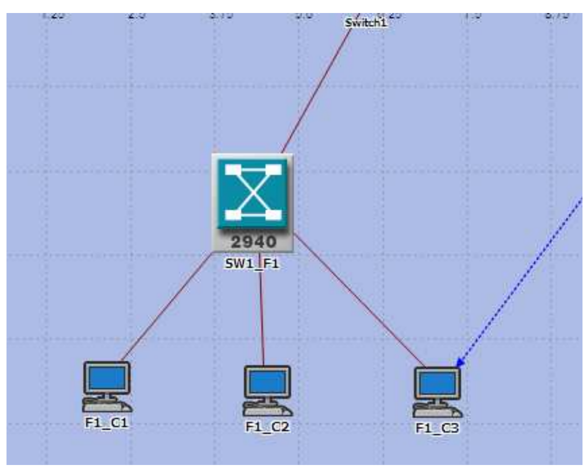


Figure 2.5 The subnet model of a single floor of the hospitality network

Our OPNET simulation was executed on a Sun E450 server. The Sun E450 server runs the Solaris 7 OS, with a SPARCV9 400 MHz processor and 1Gigabyte of memory. According to guidelines presented in [47], five simulation replications were run by feeding OPNET with different initial seeds in order to gain higher accuracy (with a narrow confidence interval) for our simulation results.

2.6.2 Simulation Results

In this subsection, we present the results obtained through simulation of the hospitality case study network of Figure 2.6. Also, we compare this results based on SIP with the results obtained from same topology based on H.323. In particular, through the results we illustrate the VoIP traffic capacity of the network, the MOS rating, and the total delay incurred through the network for the establishment of a VoIP session. Our simulation configuration runs for a total of 6 minutes, which constitutes the entire duration of the OPNET simulation. The generation of background traffic starts 40 seconds after the start time of the simulation run. We also configured the VoIP traffic to start 70 seconds into the simulation run, at which point a total of 4 VoIP bi-directional calls are added. Subsequently, every 10 seconds 16 calls are added. Our approach initiates a unique call from the end point to every other point. Figure 2.8(a) shows the total VoIP traffic that was sent, received, and dropped during the simulation. Figure 2.8(b) shows a zoomed-in version of Figure 2.8(a). From these two figures, we can observe that the last successful call was received at 4 minute and 10 seconds. At his point in the simulation, the total number of calls that were already being supported was around 300. To be more accurate,

the total number of calls that were supported was as from equation [4+ (4*60+10-70)*(16/10)] =292 calls also, as be seen in Figure 2.8(b). According to results obtained from same topology based on H.323 protocol, the last successful call received at 4 minutes and 48 seconds. Therefore, the total number of calls that were already being supported was as 316 calls as be seen in Figure 2.9(a).

Now we consider the end-to-end delay incurred by the VoIP deployment on the network. From our analysis done in the previous section, it was noted that the network delay must be below 80 msec. In Figure 2.8(c), we show the VoIP end-to-end delay for the hospitality network. As can be observed from the figure, the delay increases sharply at 4 minutes and 19 seconds, at which point, the number of calls was computed to be equal to $4+\left(\frac{4x\,60+19-70}{10}\right)x\,16=306$. Also, the results of end-to-end delay based on H.323 was seen in Figure 2.9(b), shows that the VoIP end-to-end delay increase sharply at 5 minutes, at which point, the number of calls was computed to be equal to $4+\left(\frac{5x\,60-70}{10}\right)x\,16=372$ calls.

Thirdly, we measured the MOS rating for our network, which was found to be equal to 4 for the voice codec G.711 based on SIP, as can be observed from Figure 2.8(d). Therefore, based on this simulation, the number of voice calls to be supported by the network is bounded more by the network bandwidth than by the delay. Hence, the number of the VoIP calls that the modelled hospitality network can support, without affecting the QoS, based on our simulation, was found to be upper-bounded by 300.

The results in subsection 2.6.2 show our network can support around 300 calls. In our simulation, the voice calls were added 16 calls after 10 seconds, added one call from the first floor every one second and also added one call from the other floors every five seconds. We focused on finding out the maximum number of calls that the network can sustain while satisfying the VoIP QoS at the same time, and to ensure a healthy network with normal behaviour of all network elements.

To test the robustness of our network, we performed an additional simulation run, in which 300 calls were added, all at once at the start of the simulation, say after 70 seconds. We let the simulation run for 6 minutes to reach a steady state and to examine the health of each network element. For our network, the simulation run to support 300 calls was found to be successful.

The VoIP application profiles for the designated caller workstations of the four floors need to be changed. In Figure 2.10(a), we show that each workstation on floor 1 will generate 185 calls at once, whereas in Figure 2.10(b), we show that each workstation on floors 2, 3 or 4 will generate 37 calls at once. The total number of calls that will be generated at once will therefore be equal to: 4+185+3×37= 300. In Figure 2.10(c), we illustrate the results obtained when all 300 VoIP calls are added to the network at the same time, while satisfying the QoS requirement of all network services, and leaving adequate capacity for future growth. As can be observed, the voice traffic sent is equal to the voice traffic received a round (i.e., 30,000) 1 minute into the simulation run. It can

therefore be conclusively stated that the VoIP deployment can sustain 300 calls without incurring any loss of packets.

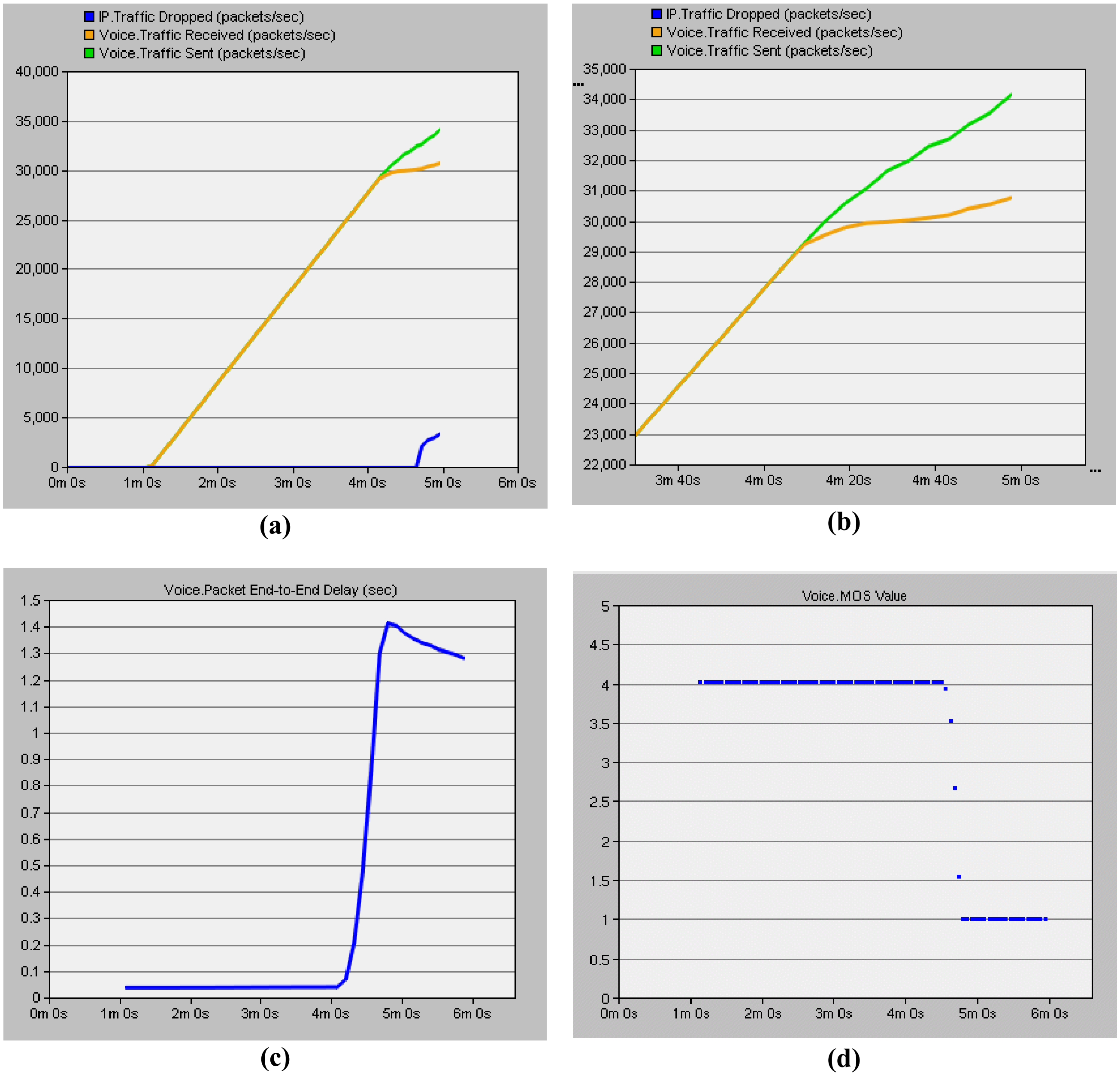
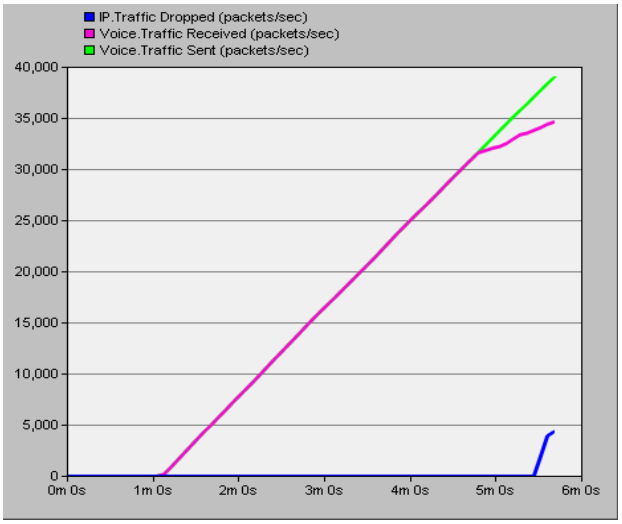
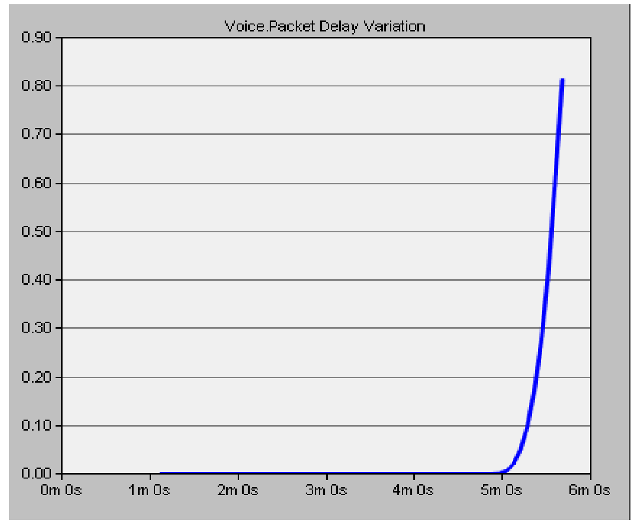


Figure 2.6 a,b)VoIP traffic, c) Delay, and d) MOS based on SIP

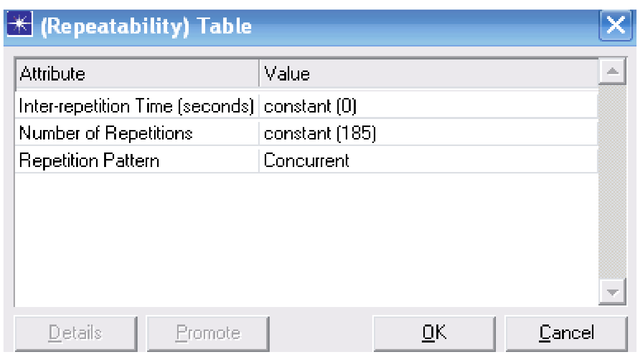


(a)

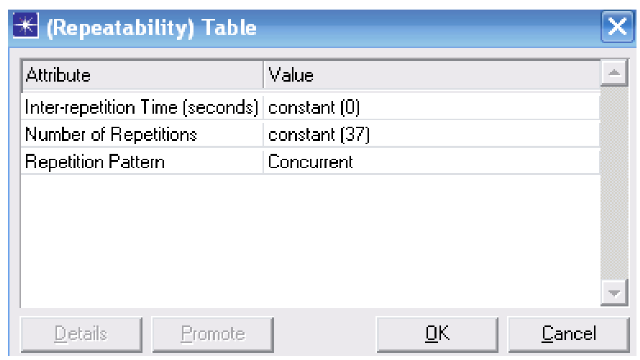


(b)

Figure 2.9 a) VoIP traffic, and b) Delay based on H.323



(a)



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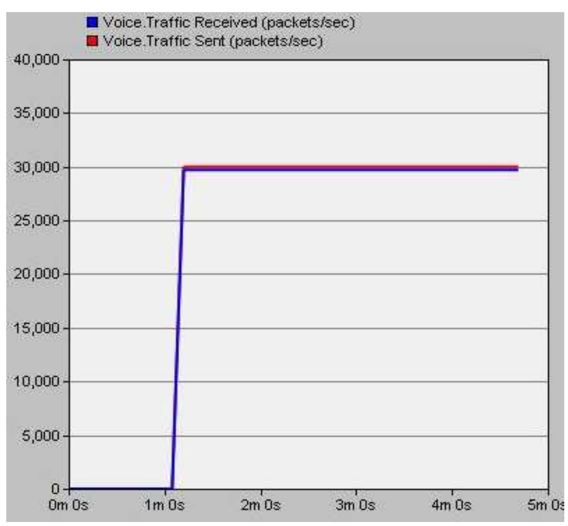


(c)

Figure 2.10 Repeatability attributes and results to generate 300 calls at same time

2.7 CONCLUSION

The chapter presented an approach on how VoIP can be deployed successfully based on the Session Initiation Protocol (SIP). Through this approach, it was intended to predict the maximum number of VoIP calls that can be sustained by a network, with a given topology, while satisfying necessary QoS requirements of all existing and new services. In addition, it was also mandated to leave enough capacity in the network for future growth. We considered a case study of deploying VoIP on a small-scale hospitality network. We initially performed a theoretical analysis of the number of VoIP calls that can be sustained on the network. We followed this exercise with actual simulation of the hospitality network in OPNET. The results from the two approaches were compared, to ascertain an upper-bound on the maximum number of VoIP calls based on SIP, that the hospitality network can sustain with given QoS constraints. It was conclusive from the results that the two results were in line and found to be close in range. The assumption of our VoIP deployment was that all calls were initiated for peer-based sessions. As part of future work, we intend to determine the maximum number of VoIP calls that can be sustain by the hospitality network, when including conference-based calls as part of the generated network traffic.

CHAPTER 3

Deploying (VoD) and (VoIP) over Hospitality Networks

Deployment of AV streaming Video-on-Demand (VoD) over existing IP networks is very limited in scope and has become one of the intense interest subjects in the research these days. In this chapter, we present an analytical approach for deploying VoIP calls and streaming Video-on-Demand (VoD) in hospitality networks. We are considering AV streaming of Video-on-Demand, and we considering IP-Unicast to deliver this streaming video. Our approach predicts the maximum number of AV streaming video sessions that can be sustained by an existing hospitality network with only 80 VoIP calls supported by this network while satisfying QoS requirements of all network services, and leaving adequate capacity for future growth. Our analysis approach is based on queuing theory, and OPNET is used for simulation to validate our analysis. Results obtained from analysis and simulation are in line and give a close match.

3.1 INTRODUCTION

IP network-based deployment offers the ability to integrate television with the other IP-based services such as VoIP and high speed Internet and that is referred to as Triple Play services. IP television covers both live TV (multicast) as well as stored video (video on

demand, or VoD) (unicast). Two cases were used to deliver streaming video to subscribe. The first case is via IP-Unicast in the case of video on demand (VoD) and the other case is via IP-Multicast in the case of live TV. The primary protocol used to deploy the video on demand is the Real Time Streaming Protocol (RTSP) [51].

Video on Demand (VoD) dedicates a single channel to each user and allows the video to be started at any time with VCR-like control (pause, rewind, fast-forward, etc.)[52]. The playback of video content requires either a set-top box which is connected to a TV, or a personal computer, which is used to decode the video content. The main codecs used to compress the video content are MPEG-2 or MPEG-4 codec.

Regarding the readiness of the exiting IP networks to support high bandwidth and time sensitive demanding real-time multimedia services, particularly those newly introduced services of streamed video and audio such that of VoD, many network architects, managers, planners, designers, and engineers are faced with common strategic, and sometimes challenging, questions. What are the QoS requirements for the new service we need to add (VoD)? How will the new service (VoD) load impact the QoS for currently running network services and applications? Will existing network support this new service and satisfy the standardized QoS requirements? If so, how many audio/video sessions can the network support before upgrading prematurely any part of the existing network hardware? At the specified bandwidth, how many audio/video sessions can be handled without any need for additional changes in the existing IP network? These challenging questions have led to the development of some commercial

tools for testing the performance of multimedia applications in data networks. A list of the available commercial tools that support Triple-Play is highlighted in [8][9][11][12][13].

All of the above available tools are commercial and they have a potential drawback. There is obviously a high cost associated with each commercial tool, as well as the complexity and learning curve. Some of these tools are too expensive to be bought by small network enterprises or institutions. Some tools that use the approach of injecting Triple-Play traffic into existing network can be very intrusive and disturbing to existing network services and applications. If the network is running time-critical services or applications, this can be an unacceptable approach. None of the commercial tools offer a comprehensive approach or methodology for successful IPTV deployment. In particular, none gives adequate prediction for the maximum number of voice and video sessions that can be supported by the network taking into account important design and engineering factors. These factors include Triple Play flow and call distribution, future growth capacity, performance thresholds, impact of voice and video traffic on existing network services and applications, and impact of the background traffic on the voice and video traffic.

In a previously related work [53], an analytical approach was presented for the deployment of desktop videoconferencing in an existing network. Authors investigated the two key performances, i.e., delay and bandwidth for videoconferencing. Also, they assessed the readiness of an existing network to support videoconferencing and predicted

the maximum number of videoconferencing sessions that can be sustained by this network while satisfying QoS requirements of all network services and leaving adequate capacity for future growth. Salah et al. [54], demonstrated how OPNET can be leveraged to assess the readiness of existing IP networks to support videoconferencing. They consider two video traces collected from Internet traffic traces, with two types of video traffic, i.e., fixed and empirical video packet sizes. Hrudey et al. [55], addressed whether Wimax access technology for streaming video application could provide comparable network performance to ADSL. They used OPNET to characterize the performance of streaming a 2-hour MPEG-4 movie to Wimax and ADSL using four performance: available bandwidth, buffering, and delay characteristics of underlying network.

In Chapter 2, an analytical approach was presented for the deployment of VoIP based on SIP protocol over an existing hospitality network. In sharp contrast to our previous work in Chapter 2, this chapter is different in significant ways. First this chapter focuses on deploying VoD services as a new service to be deployed with 80 VoIP calls supported. Second, this chapter presents an analytical approach for the deployment of VoD with 80 VoIP calls from previous chapter. Third, this work addresses the requirement for the new services needed to be deployed successfully, and predicts the maximum number of audio/video VoD sessions that can be supported by an existing hospitality networks while satisfying QoS requirements of all network services and leaving adequate capacity for future growth.

The rest of this chapter is organized as follows. Section 2 presents an overview of video content and its compression. Section 3 presents a typical network topology of a small-scale hospitality network to be used as a case study for deploying Triple play services. Section 4 describes key issues and requirements that have to be defined for VoD service as a new service to be deployed, and presents our analytical approach and algorithm for deploying successfully Triple-Play services (VoIP and VoD). Also, it describes and summarized the OPNET model and it results. Section 5 concludes our study.

3.2 VIDEO CONTENT OVERVIEW AND COMPRESSION

This section gives a brief overview of the video content and compression technology. The video information available from video services providers is referred to as the video content. The content is structured as a sequence of video frames or images that are sent or streamed to the subscriber and displayed at a constant frame rate [55]. Based on the different transmission of the streaming real-time video and their buffering requirements from the network and the client server, video content is characterized by several parameters including video format, pixel color depth, coding scheme, and frame interarrival rate.

There are numerous numbers of compression technologies developed by the International Standards Organization (ISO) and the International Telecommunication Union (ITU-T). The main advantage of this codec and compression technology is that it has reduced the required bandwidth for encoding content that will be done by reducing

the space within the frame content [55, 56]. An uncompressed SDTV signal for PAL (Phase Alternating Line) with a color encoding has a resolution 768x576 pixels and a frame rate of 25 frames/second. An RGB pixel has 3 values and the quantization is done in 256 steps (8 bits). So, the required data rate for video transmission is calculated by 768*576*25*3*8≈ 253 Mbps. Also, the sampling rate of an audio signal is about 44.1 Kbps with 16 bit quantization that required a data rate of 44.1*1000*16*2≈1.4 Mbps. In general, delivery of audio and video signals without compression would require a bandwidth of about 255 Mbps [56].

The series of MPEG specifications that are in widespread use in virtually all existing IPTV deployments have been developed by the jointly between the International Telecommunication Union (ITU-T) and the International Organization for Standards/ International Electrotechnical Commission (ISO/IEC) [57]. The ITU-T recommendation H.262/ISO/IEC MPEG-2 standard which was developed by M. Krunz and H. Hunghes in 1995 has been deployed to support IPTV and Video-on-Demand (VoD) services [58], which is used to digitize, compress, and encode TV signals. The required bandwidth for the SDTV encoded by MPEG-2 codec will be between 3 to 4 Mbps.

The introduction of ITU-T recommendation H.264 or as known as advanced video codec (AVC) has becoming interest to deploy future IPTV and VoD, which is developed by H. Koumaras, C. Skians, G. Gardikis and A. Kourtos in 2005 [59]. They proposed that a gamma distribution provides the best fit for the frame size histograms of H.264 codec video streams. H.264 (AVC) cuts the bandwidth requirement for digital video delivery to

half of the MPEG-2. It uses only between 1-2 Mbps [57][59][60]. Telecommunications companies focus their effort to deploy IPTV-DVD quality video services over DSL and the Internet by using H.264 (AVC) instead of MPEG-2.

3.3 HOSPITALITY NETWORK

A hospitality network is used in private facilities such as in hotels, cruise ships or college campuses, which offers entertainment, information and communication services designed to end-user. This section describes our case study which is a small-scale hospitality network as be seen in Figure 3.1. This Figure illustrates a typical hybrid network topology for a small-scale hospitality network. The network is Ethernet-based that can support at least 200 users. The network has four floors each floor is a subnet which is basically a Layer-2 switch, Cisco catalyst 2960, with 50 ports that can be support at least 50 users, connected with the customer-premises equipment such as set-top Box for VoD, VoIP phone, and wireless access point. Floor 1 which called as loby or known as reception has more than one access point to support more wireless in the conference room. Also, there are some users rooms in this floor. Every two floor subnets are connected with Layer-2 switches, Cisco catalyst 3560G, and these Layer-2 switches are connected with a Layer-3 backbone switch, Cisco catalyst 5500. The backbone Layer-3 switch is connected with four servers. These four servers constitute the server farm and can support E-mail, ERP, HDCP, and DNS services. All the links are 1Gbps except the links for the floor subnet which are 100Mbps.

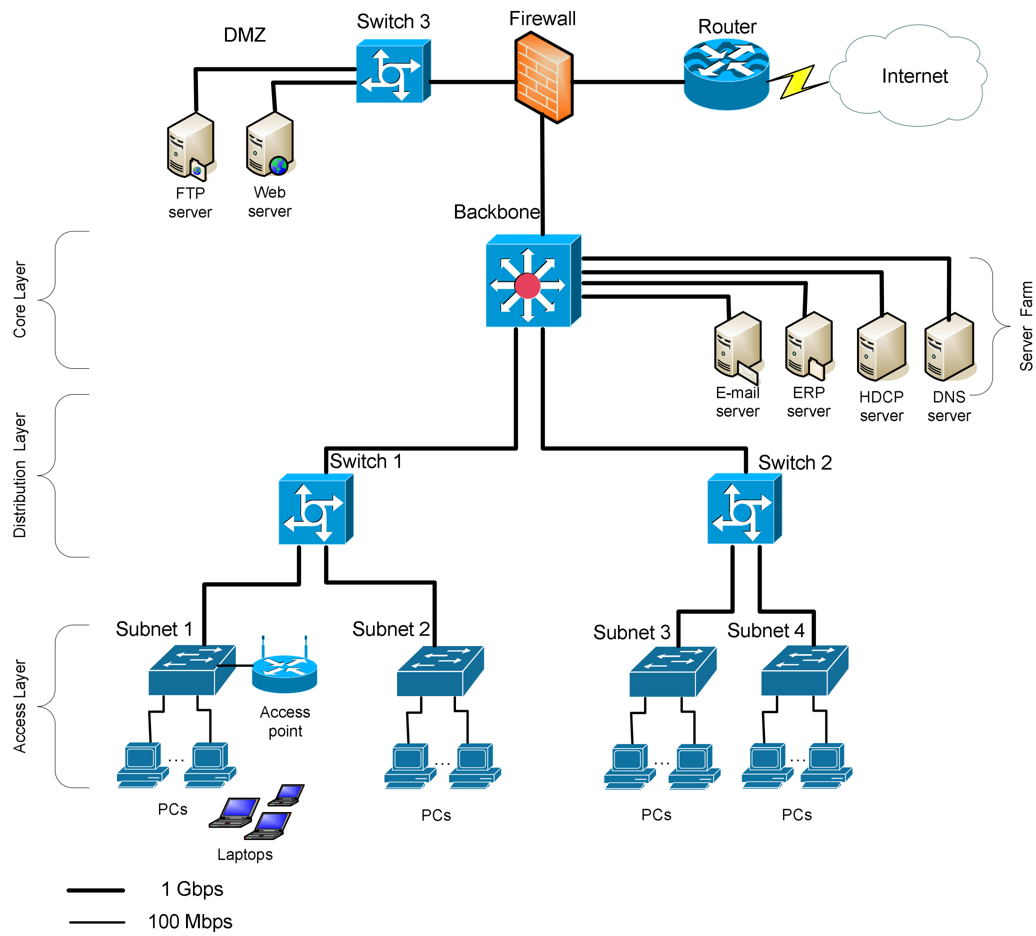


Figure 3.1 Logical diagram of a small-scale hospitality network [Chapter 2]

3.4 VIDEO-ON-DEMAND (VOD) REQUIREMENTS

We considered in Chapter 2 the requirements to deploy VoIP. Adding a new service to the topology as VoD will be needed to address and determent the requirements of this new service (VoD). In this section, we address the requirements to deploy VoD with 80 VoIP calls for our case study. These requirements need to be investigated and to be fed

into simulation for determining the maximum number of AV VoD sessions that can be supported by this network.

3.4.1 VoD Traffic Characteristics, Requirements, and Assumptions

There are some issues that come up when will be needed to deploy any services such as VoD in any network. These issues are characterizing the nature of VoD traffic, QoS requirements, and hardware components or devices. Quality of Service (QoS) is very important for deploying IPTV and VoD as it is a real-time service. However, QoS for deploying IPTV will be affected by packet loss, packet reordering, and packet faults latency, packet duplication, and jitter.

End-to-end delay. Small amount of delay does not directly affect the Quality of Experience (QoE) of IPTV. While the delay large than 1 second may result a much worse QoS toward end-user experience. Quality of Experience (QoE) is the customers' perception of how good of a job the Service Provider is doing delivering the service [61][62]. Therefore, the delay for one way must be less than 200ms. On the other hand, the end-to-end delay more than 400ms was considered to be unacceptable [77].

Packet loss ratio. There are some standard packet loss ratios given by ITU-T for classifying IPTV services. The excellent service quality of packet loss ratio is than 10^{-5} and the poor service quality of packet loss ratio is between $2*10^{-4}$ and 0.01 [77]. Therefore, a packet loss ratio above 1% is unacceptable.

Bandwidth. High-Definition Movie (HD) has been studied in our work, which has a resolution format 128X240 pixels with mean frame rates size of 34560 bytes. The mean transmission rate for this movie is 8Mbps with an interarrival rate of 30 fps. We consider also the audio frame of the TV program and movie clip which is 6 packets in an audio frame, each packet with 640 bytes in size, and the average rate per frame is 44.1 kbps.

Video equipments. The main equipments needed to deploy VoD or IPTV services are set-top boxes and a head-end server. The head-end server or known as VoD server is the source for all video content. The main functionality of set-top box (STB) is to unscramble the signal and present it on the TV [65].

3.4.2 VoD Traffic Flow and Sessions Distribution

It is important to specify the flow of sessions and their distribution to determine the maximum number of sessions that can be supported. The assumptions considered while determining the distribution, that there is a small number of rooms in the first floor and a large display screen in the reception. The number of rooms in floors 2, 3, and 4 are the same as we mentioned above in section 3.3, where every floor can support at least 50 users, and the probability of having VoD sessions from each of these floors is equal. Figure 3.2 shows the session's distribution for our network under study based on the worst busy hours and the projected future growth of video-on-demand sessions. The session's distribution is described as a probability tree as shown in Figure 3.2. 48 VoD

sessions will be distributed as 3 sessions for floor 1 and 15 sessions for each of floors 2, 3, and 4.

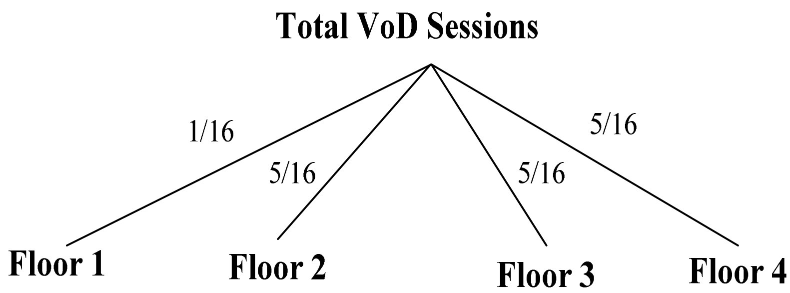


Figure 3.2 Probability tree describing the VoD sessions distribution

3.4.3 Upfront Network Assessment and Modifications

This step is very important to assess the existing network and determine, based on the existing traffic load and the requirements of the new service to be deployed, if any immediate modifications are necessary. Immediate modifications to the network may include adding and placing new servers or devices, upgrading PCs, and re-dimensioning heavily utilized links. All the links connecting the backbone switch and the Layer-2 switches and links connecting the backbone switch and the servers are underutilized. Also, the links connecting the PC's with the subnet switch are underutilized. We upgraded the link connected the set-top box to the floor switch to 1Gbps because it had heavily utilized reach to 76%.

Based on the hardware requirement for deploying VoD service in Section 3.4.1, which describes VoD requirements, the new hardware element that has to be added to the

existing network is the head end server. As a network design issue, an appropriate node placement is required for this node. The head end server should be connected to the backbone switch in the server farm in order to balance the projected load on both switches. Figure 3.3 shows the new network topology after the incorporation of necessary VoIP components based on the SIP and the new services VoD components. As shown, the head end server and STB nodes for VoD were added.

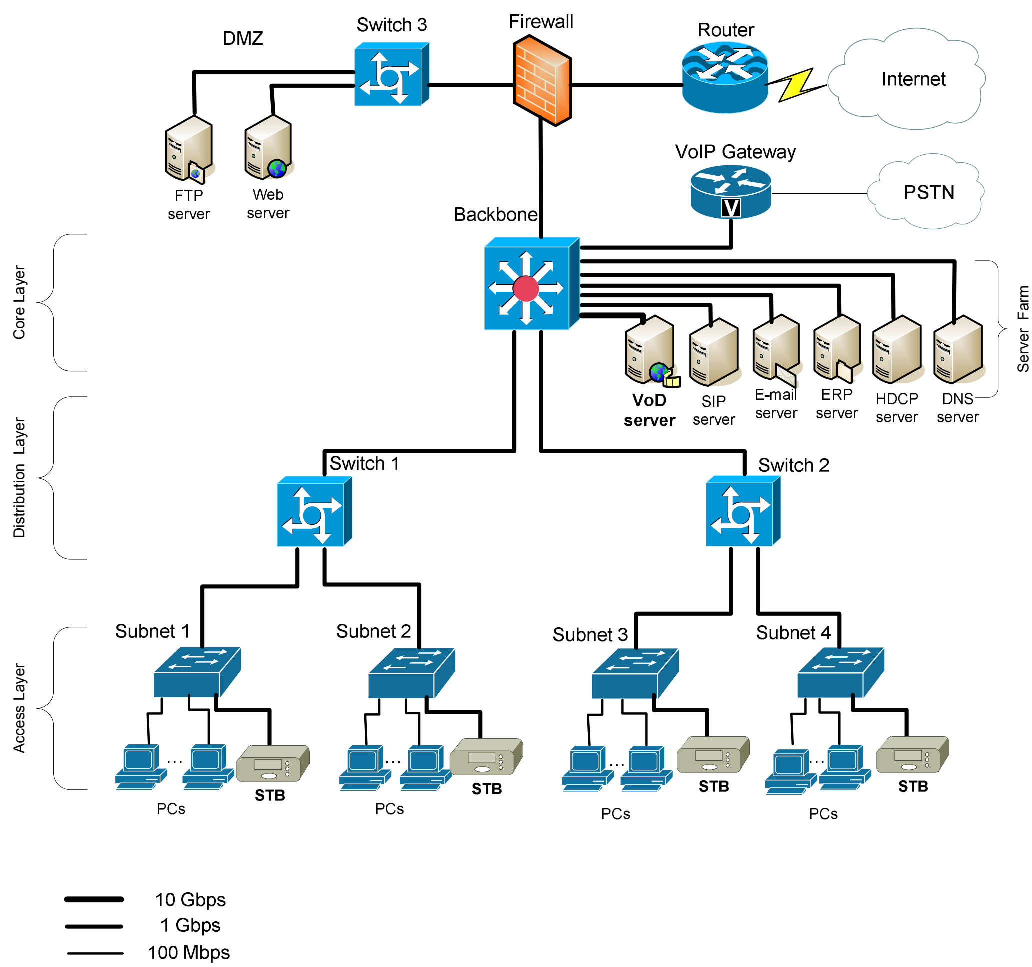


Figure 3.3 Network topology when adding VoIP and VoD components

3.4.4 Analysis

This section presents our analytical approach to deploy Video-on-Demand services with 80 VoIP calls in any hospitality network. The actual number of sessions with 80 of VoIP calls that the network can sustain is limited by two metrics. These metrics are the available bandwidth or delay as discussed in details in [3]. In order to determine the maximum number of sessions that can be supported by an existing network while maintaining a VoD delay constraint and assuming support of only 80 of VoIP calls, we develop an algorithm that basically determines network capacity in terms of audio/video sessions.

Algorithm 2 presents the overall computed maximum number of audio/video sessions based on video and voice delay constraints. It takes as input the number of network elements, the background traffic for every element, the delay for network elements, and the call flow paths, and it returns the maximum number of sessions generated by this network. It initializes with determine the required bandwidth for a single unidirectional video session and bi-directional voice call in terms of packets per second and also in terms of bits per second in line 1.

We added 80 voice calls to our network as described in lines 2 to 10. In line 3, we obtained α from the call distribution for each network element *i*. The flow percentage of voice calls for L3_Switch, Switch2 in distribution layer, and the uplink between F1_Switch and Switch1 as an example are (11/16), (3/8), and (1/2) respectively. In line 4, we multiply α by the bandwidth of voice calls and add it to λ_i for each network

element *i* and in lines from 5 to 9. We modeled the Ethernet links as *M/D/1* queues, this is appropriate since the service time for links is more of a deterministic than variable. However, the services times of the switches are not deterministic since these modeled as *M/M/1* queues.

Computing the maximum number of sessions in lines from 11 to 42, in line 13 a uniform random generator is used to generate video sessions according to session's distribution as in subsection 3.4.2. β is obtained from the sessions distribution for each network element *i*. The video traffic flow for some network elements such as L3_Switch, Swich1, and downlink from L3_Switch to Switch2 are (1), (3/8), and (10/16) respectively.

Algorithm 2: Compute maximum number of sessions based VoD and VoIP delay constraint

```
Input: n : number of network elements
       λ[1..n] : background traffic for network elements 1,2,.. n
       Delay[1..n] : delay for network elements 1,2,.. n
       P : set of call-flow paths (p) where p is a subset of {1,2,.. n}
Output: MaxSessions: maximum number of Sessions
     λ_VoIP ← 100pps, or 180.8kbps; λ_Video ← 812pps,or 9.2Mbps; PktSize VoIP ← 1808 bits;
     PktSize Video ← 11288 bits; VoD_MaxDelay← 80; VoIP_MaxDelay ← 80; VoD_MaxDelay_Found←0;
     VoIP_MaxDelay_Found ← 0; MaxSession ← -1; Delay[1..n] ← 0;

for each element i in the network do
     Find α from the call distribution for each network element i
       (α = 80 * Voice Flow).
     λ_i ← λ_i + α * λ_VoIP
     if i is a link then
         VoIP_Delay_i ← (1 − λ_i /2 μ_i ) / (μ_i - λ_i ) * PktSize VoIP
     else
         VoIP_Delay_i ← 1 / (μ_i - λ_i )
     end if
end for

while VoD_MaxDelay_Found < VoD_MaxDelay and VoIP_MaxDelay_Found < VoIP_MaxDelay do
   MaxSession ← MaxSession + 1
   Generate a session according to VoD distribution and let pc be its flow path
   for each element i in pc do
     Find β from the call distribution for each network element i
         λ_i ← λ_i + β *λ_video
     if i is a link then
         VoD_Delay_i ← (1 − λ_i /2 μ_i ) (μ_i - λ_i ) * PktSize Video
     else
         VoD_Delay_i ← 1 / (μ_i - λ_i )
     end if
   end for

   for each element i in the network do
     if i is a link then
         VoIP_Delay_i ← (1 − λ_i /2 μ_i ) / (μ_i - λ_i ) * PktSize VoIP
     else
         VoIP_Delay_i ← 1 / (μ_i - λ_i )
     end if
   end for

   for each p in P_VoD where p∩p_c ≠ ∅  do
       PathDelay(p) ← ∑ VoD_Delay_i , where i is a network element in path p
       if PathDelay(p) > VoD_MaxDelay_Found then
          VoD_MaxDelay_Found ← PathDelay(p)
       end if
   end for

   for each p in P_VoIP where p∩p_c ≠ ∅  do
       PathDelay(p) ← ∑ VoIP_Delay_i , where i is a network element in path p
       if PathDelay(p) > VoIP_MaxDelay_Found then
          VoIP_MaxDelay_Found ← PathDelay(p)
       end if
   end for
end while
```

Algorithm 2 was implemented using MATLAB and the results for the worst-case delay are plotted in Figure 3.4. Algorithm 2 basically determines the network capacity in terms of VoD sessions for the AV streaming and 80 VoIP calls that can be done by computing the number of sessions based on the delay bound. Also, algorithm 2 is applied for High-Definition movie and sessions are added iteratively until the worst-case network delay of 80 ms is reached.

Figure 3.4 shows that the maximum number of sessions that could be supported for VoD, when deploying the High-Definition Movie. The maximum number of AV streaming sessions that can be supported is about 114 sessions. This number of sessions comes after upgrading the links connected to the set-top box to 1Gbps. When the link connected to set-top box was 100Mbps, the maximum number of sessions that can be supported is about 24 sessions. It is clear that the delay for the 24 sessions is not greater than 80ms but dropped to a negative value. This situation is an indication of the instability which occurred when the arrival rate of a network element is greater than its service rate.

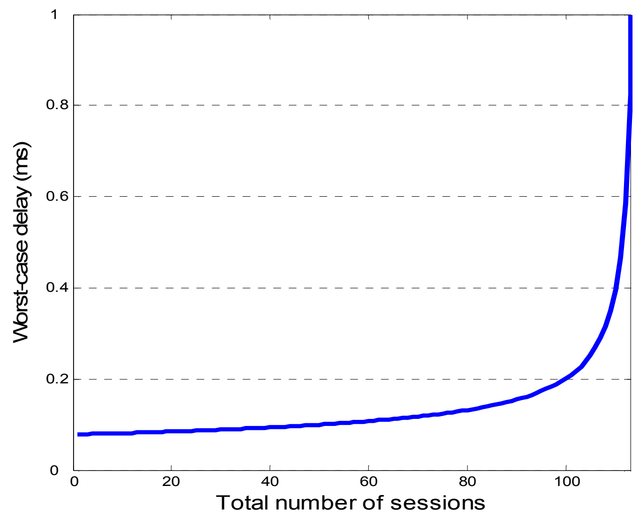


Figure 3.4 Worst case delays vs. number of video sessions

3.5 OPNET SIMULATION

This section describes in details the simulation model, traffic flow and call distribution, various simulation configurations, as well as the simulation results. Our simulation approach uses the popular MIL3's OPNET Modeler simulation package1, Release 14.0.A [46]. Because, OPNET Modeler contains a vast amount of models of commercially available network elements, and has various real-life network configuration capabilities, it makes the simulation of real-life network environment close to reality.

3.5.1 Modelling the Network

Our simulation modeling for the case study is shown in Figure 3.5. The simulation model of our case study is based on SIP signaling as mentioned in Chapter 2 and we add new services to this topology which is VoD service. In OPNET, many vendor specific models are included in the pre-defined components libraries. However, the specific devices that we need, i.e., Layer-3 switch Cisco catalyst 5500, which configure as router, and Layer-2 switch, Cisco catalyst 3560G. VoIP gateway is modeled as a workstation since we are interested only in collecting statistics inside the hospitality network. Two servers added to modeling the hospitality network. One is the head end server which is called VoD server. The VoD server is used to deploy video on demand in our case study and the other server which is known as SIP server is used to deploy VoIP based on SIP protocol as we mentioned in Chapter 2.

Floor LANs have been modeled as subnets that enclose Layer-2 switch, Cisco catalyst 2960, and four designated Ethernet workstations are used to model the activities of the LAN users, as shown in Figure 3.6. One of these workstations generates the background traffic of the floor while the other two act as parties in VoIP sessions. For example, the Ethernet workstations for Floor 1 are labeled as F1_C2, F1_C2, and F1_C3. F1_C1 is a source for sending VoIP calls. F1_C2 is a sink for receiving VoIP calls. F1_C3 is a sink and source of background traffic. F1_C4 is a sink for receiving VoD sessions. Modeling of floor LANs does not represent precisely the floor multimedia PCs or IP phones. VoIP was modeled to generate 16 calls every 10 seconds as discussed in Chapter 2, but in this

chapter modeled to stop after 80 calls added. Also, our simulation here after we added the video service is automated to generate 16 sessions every 5 seconds as configured in video profiles.

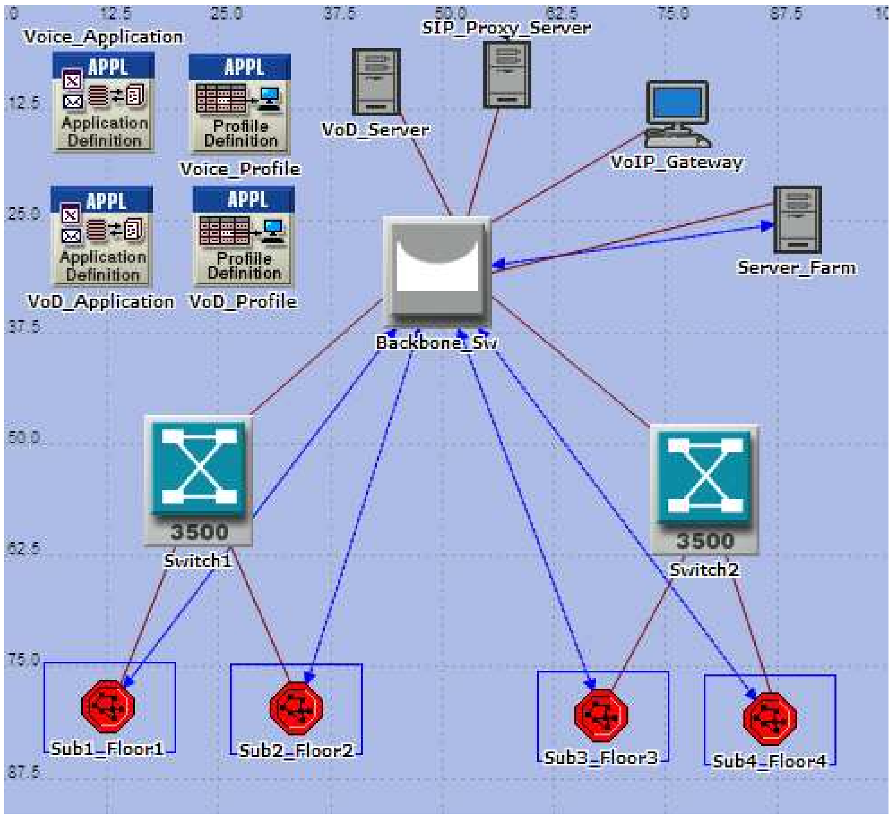


Figure 3.5 OPNET model of hospitality network with VoIP and VoD

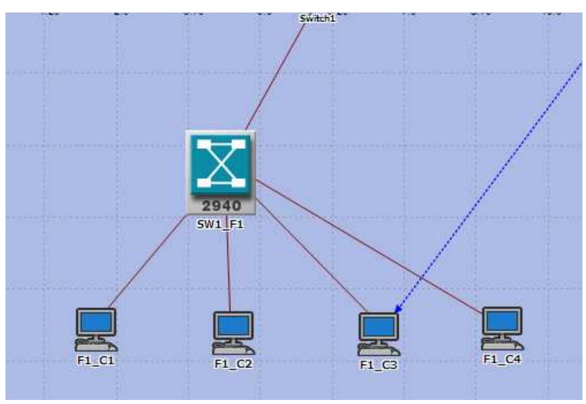


Figure 3.6 Floor subnet model for VoIP and VoD

3.5.2 Generating AV Streaming Traffic

To our knowledge, to date, the OPNET Modeler does not have built-in features to support AV Streaming or its deployment. In this section, we will explain in details how we modeled AV Streaming. The modeler profile node was configured to reflect AV streaming for HD TV. Our approach has two profiles, one for floor 1 and the other for the other floors that depended on the session's distributions as discussed in subsection 3.4.2. The profiles are configured for the HD video as shown in Figure 3.7. Their operations mode is set to simultaneous mode, with a starting time at 60 seconds, and the deviation of starting time of each application (AV) conforms to a constant of 10 seconds. The profiles are deployed to the video clients and the VoD server is configured to support the appropriate application services.

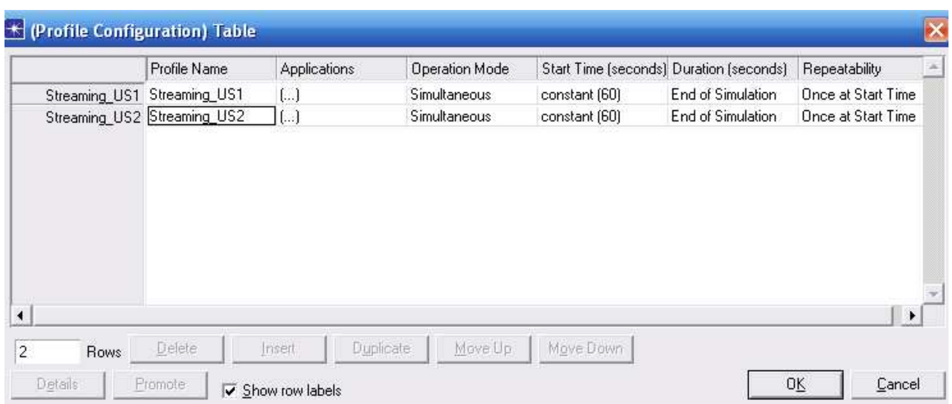
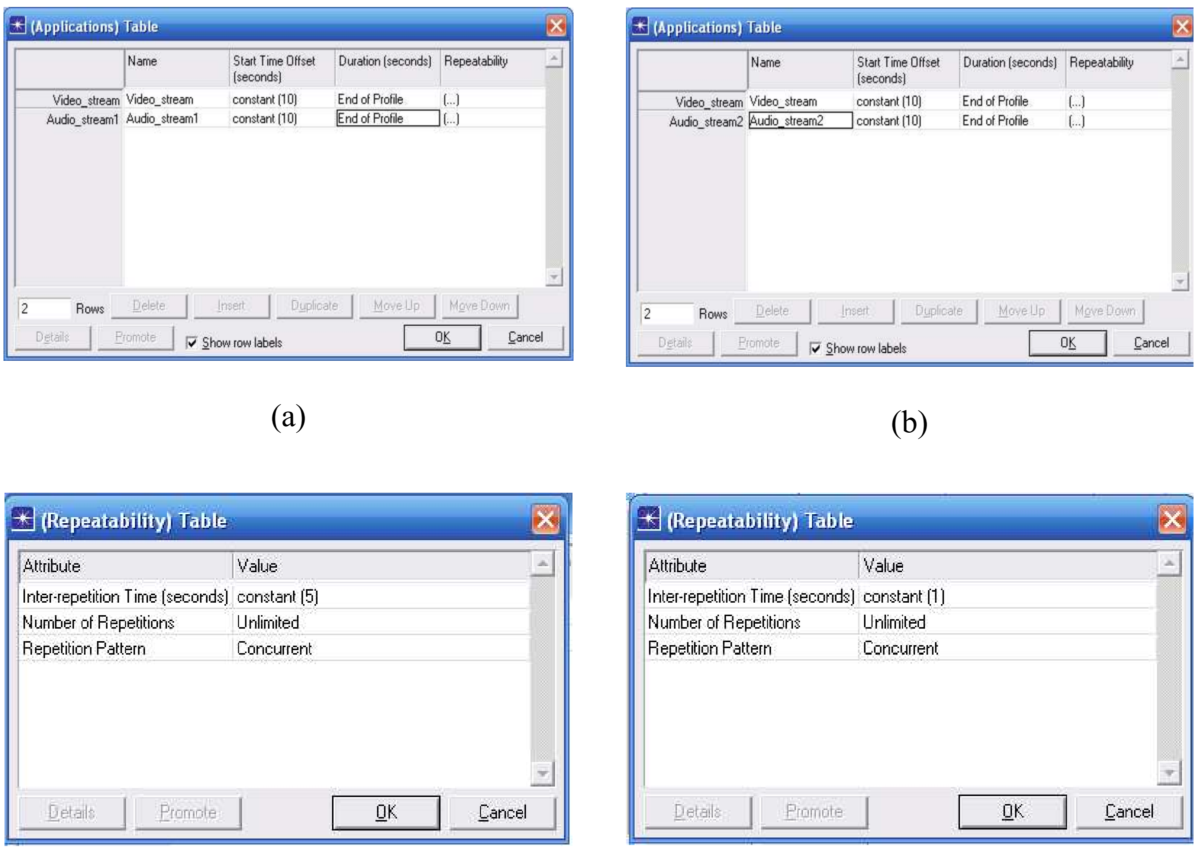


Figure 3.7 AV streaming traffic profile configurations

The configuration of the applications in the profile is shown in Figure 3.8(a,b), and it is configured to support Audio and Video at same time. Figure 3.8(a) shows the AV streaming application for Floor1 and Figure 3.8(b) for other floors. The application

repeatability for Floor1 is shown in Figure 3.8(c), where in every 5 seconds, one audio/video streaming session is added. And, the repeatability for video_stream and audio_stream application is the same. On the other hand, the application repeatability for the other floors, floor2, 3, and 4 is shown in Figure 3.8(d), where every 1 second, one audio/video streaming session is added, and the repeatability for video_stream and audio_stream application is the same.



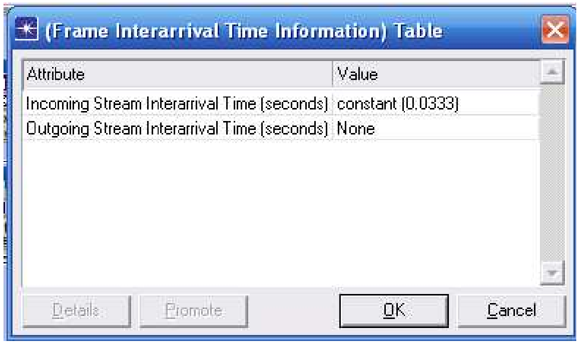
(a) (b)

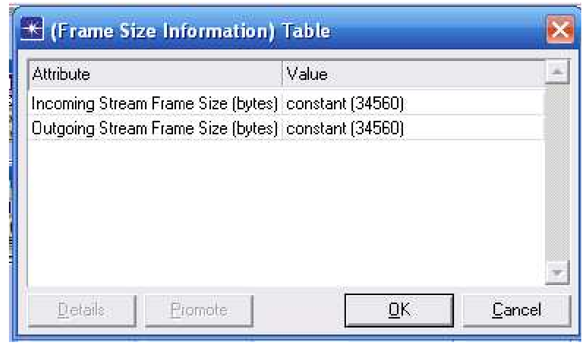
(c) (d)

Figure 3.8 Applications AV streaming in profile and repeatability

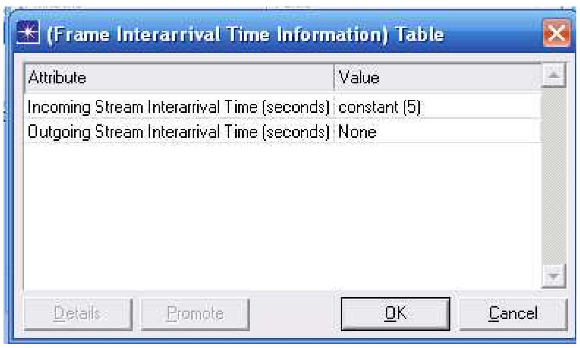
We modelled three applications for AV Streaming in the application definition. The first application for the Video Streaming which sets the incoming frame inter-arrival rate to

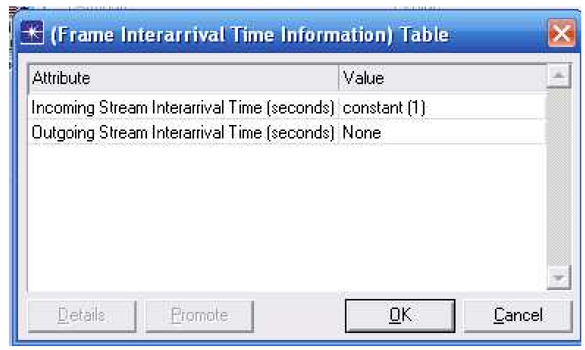
0.0333 to reflect the content encoding rate of 30 fps. And the outgoing stream interarrival time remains at none to create a unidirectional stream in Figure 3.9(a). The video trace was the High-Definition TV and refers to video having 128X240 pixels resolution. The frame size is 34560 bytes as shown in Figure 3.9(b). The average bit rate for one video streaming session is equal to 30 frames* 34560 bytes* 8 bits≈ 8 Mbps. The other two applications are modelled for the Audio streaming, the first application is modelled to add audio stream after 5 seconds as in profile for Floor 1 as shown in Figure 3.9(c). The second application is modelled to add audio stream after 1 second as in profile for the other Floors 2, 3, and 4 as shown in Figure 3.9(d). The audio traffic for one session is equal to 30 frames* 44.1 kbits≈ 1323 kbps. However, 44.1 kbps is the average bit rate for one frame audio traffic which is approximately equal to 6 packet audio in frame, every packet has 640 bytes in size [78].





(a) (b)





(c) (d)

Figure 3.9 Applications definition for AV streaming

3.5.3 Simulation Results

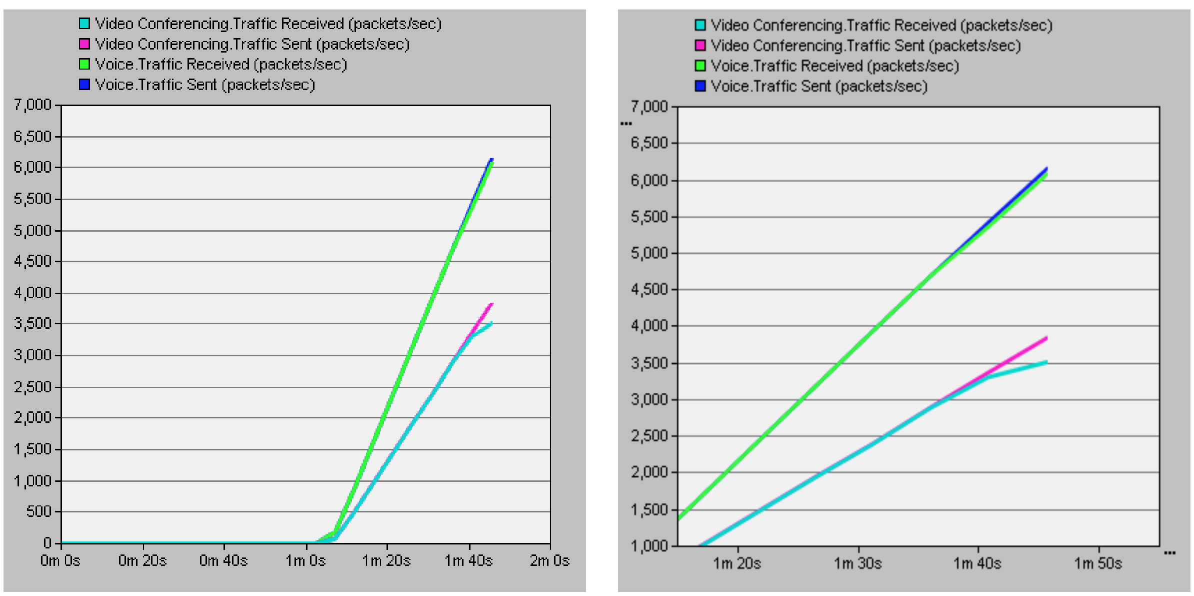
This section presents snapshots of OPNET simulation results for deploying HD AV VoD services with 80 VoIP calls. The video and voice traffic are modelled to start after 70 seconds at time of the simulation run. We discussed VoIP in Chapter 2. Voice traffic starts at 70 seconds with 4 VoIP bi-directional calls added. Then, every 10 seconds 16 calls are added. Figure 3.10(a) shows the total number of 80 VoIP calls can be supported. Our simulation in this chapter focus on deploying streaming video AV content as a new service need to deploy it in our topology. Streaming video traffic starts

at 70 seconds at which a total of 4 one way streaming video sessions are initially from the head-end server to users. Then, every 5 seconds, 16 video sessions are added. Two profiles have been configured in our simulation, and the repeatability for each application as follows. Every 1 second one video session is received by floors 2, 3, and 4, while floor 1 received only one video session each 5 seconds and that is based on the sessions distribution as described in subsection 3.4.2.

Figure 3.10(a) shows the total videoconferencing traffic for the AV streaming video trace and also the total voice (VoIP) traffic that was sent, received, and dropped. Figure 3.10(b) shows a zoom-in of Figure 3.10(a). Figure 3.10(a) and Figure 3.10(b) show the last successful video traffic that was received at 1 minute and 42 seconds. So, the maximum number of video sessions that can be supported in this case study with 80 VoIP calls is approximately equal to [4+ ((60+42)-70)*(16/5)] ≈ 107 sessions for our HD movie. Each session per user, the maximum number of users can be supported was 107 users.

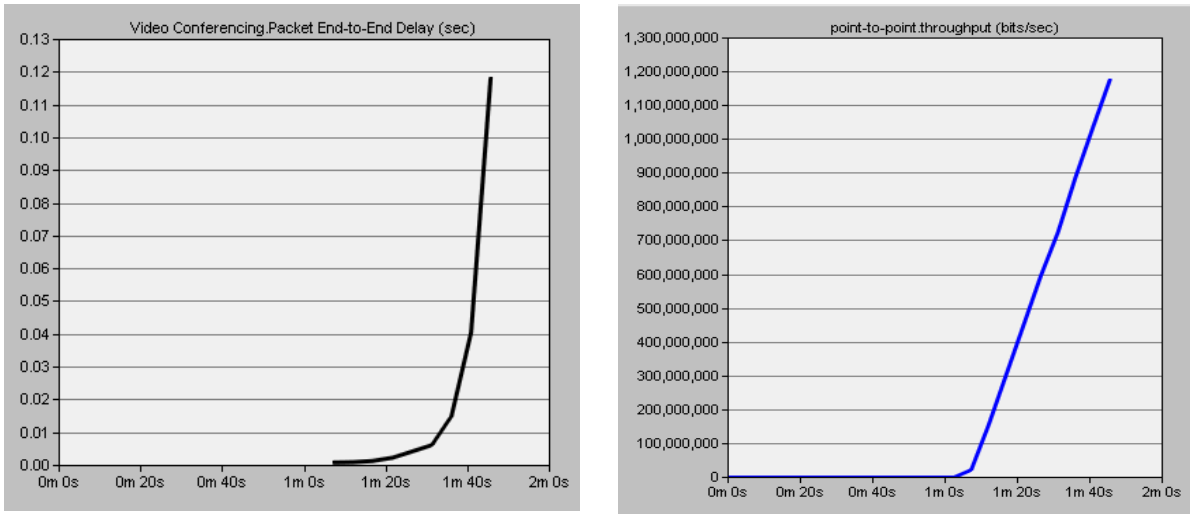
Figure 3.11(a) shows the corresponding streaming video end-to-end delay. As seen in Figure 3.11(a), the video packet end-to-end delay increases sharply after more users are added to the network. The end-to-end delay stays less than 80ms until a simulation time of 1 minute and 44 seconds as seen in Figure 3.11(a). Therefore, the number of video sessions can be computed as [4+ ((60+44)-70) *(16/5)] = 112 sessions. Therefore, the maximum number of streaming video sessions that can be supported by the network is bounded more by the network bandwidth rather than by the delay. So, the number of

audio/video sessions that can be sustained by this network based on simulation is 107 sessions. Figure 3.11(b) shows the bandwidth between the Head End sever and the L3 switch. The bandwidth increases sharply after more users are added to the network, the bandwidth at the dropped point is 1,034,313,553 bps. As we referred to in subsection 3.4.1, every streaming video has 8Mbps for video session and 1323 kbps for audio session, and the last successful session received was at 1 minute and 42 seconds which gives 107 sessions. So, the total bandwidth needed to support 107 users is equal to [107*(8Mbps+1323kbps)]≈1,032,459,264 bps. The results obtained from mathematical equation support our simulation results and the difference not much.



(a) (b)

Figure 3.10 Global videoconferencing traffic in fps of streaming video



(a) (b)

Figure 3.11 a)Video end-to-end delay, b) Bandwidth between HE server and L3-swicth

3.6 CONCLUSION

This chapter presented an analytical approach to assess network readiness for deploying video-on-demand (VoD) with VoIP on an existing IP network. A small-scale hybrid hospitality network was used as a case study. This approach predicts the maximum number of AV sessions that can be sustained with 80 VoIP calls in hospitality network. In addition, this chapter discussed many design issues pertaining to the deployment of video-on-demand service include QoS requirements, video flow, and sessions distribution.

Our approach used OPNET to verify and validate the results obtained from our analysis. The results obtained from both analysis and simulation were in line with a small difference. This difference can be contributed to the degree of accuracy between the

analytical approach and the simulation. Because the analytical approach was an approximation, the difference was 7 sessions. Also, the way to add sessions in OPNET based on the session distribution. To be precise and conservative, we consider the minimum number of sessions of the two approaches.

CHAPTER 4

Deploying IP-Multicast AV TV over Existing Hospitality Networks

Internet Protocol Television (IPTV) is seen as one of the killer services these days, which a services where digital TV signal on one side delivering by using Internet Protocol (IP). In this chapter, we present an analytical approach for deploying IP-Multicast AV TV over hospitality network. Our approach predicts the maximum number of users can be supported by hospitality networks based on IP-Multicast while satisfying QoS requirement of all network services and leaving adequate capacity for future growth. Protocol Independent Multicast- Sparse Mode (PIM-SM) was used to deploy AV IPTV, which is the popular multicast protocol used. Our analysis approach is based on queuing theory, and OPNET is used for simulation to validate our analysis. Results obtained from OPNET simulation support our results from analytical approach.

4.1 INTRODUCTION

Internet Protocol Television (IPTV) is a set of multimedia services that are distributed throughout an IP network, where the end of user receives video streams through a set-top-box (STB) connected to a broadband connection. IPTV is often combined with

services of VoD. VoD services content are not live but pre-encoded contents available at any time from servers. These services must possess an adequate level of quality of service, security, interactivity, and reliability. From the perspective of the provider, IPTV includes the video acquisition, video processed, and video secure distribution on the IP network infrastructure [51][56][66].

In Chapter 3, we proposed an analytical approach to deploy AV (VoD) on hospitality networks. The current chapter goes significantly further, providing a comprehensive study of deploying IPTV live, including insights into the characteristics and requirements of the new services (IPTV) needed to be supported in our networks. Achieving these deeper insights has been challenging because deploying IPTV (live TV) needs a multicast protocol to be deployed in our networks. In this chapter, we seek to answer the following questions about a large-scale IPTV (live TV) deployment. What are the QoS requirements for the new service we need to deploy it? Will existing network support this new service and satisfy the standardized QoS requirements? If so, how many AV streaming sessions can the network support before upgrading prematurely any part of the existing network hardware? For a popular live TV channel, how many users will be watching this channel? Which multicast protocol is the best protocol needed to deploy to support IPTV (live TV) service?

This chapter is organized as follows. In Section 2, we present some related work. In Section 3, we provide an overview of the IP multicast protocol, PIM-SM protocol. In Section 4, we present a typical network topology of a small-scale hospitality network to

be used as a case study for deploying AV IPTV (live TV) service over our case study. In Section 5, we present the IPTV QoS requirements and user characteristics. In Section 6, we present our analytical approach and algorithm for deploying successfully AV IPTV services. In Section 7, we present the OPNET simulation configuration and its results. Based on our simulation results, we outline some design guidelines and engineering decisions for the successful deployment of IPTV application over an existing IP network in Section 8. Finally, we conclude our study and identify future work in Section 9.

4.2 RELATED WORK

There is not much related work done before based on deploying IPTV channel TV popularity over an existing network. Some of these previously related works are as follows. Qiu et al. [67] analyzed the channel popularity in the context of IPTV, their analysis based on a large collection of user channel access data from a nation-wide commercial IPTV network. Also, they construct a mathematical model to capture the distribution and the time dynamics of channel popularity. They observe from their model that channel popularity is highly skewed and can be well captured by a Zipf-like distribution. Also, they observe a fair amount of channel access popularity change during a short time window, although they found that the channel popularity during moderately long time windows relatively stable.

Young et al. [68][69] reported the results and analysis of network-centric quality measurement and its impact using real-world commercial traces in various user

scenarios. The available commercial IPTV service in the reach is a video-on-demand (VoD) service using a set-top box (STB) that relies on three separate transmission techniques: best effort streaming, QoS controlled streaming, and D&P. They have measured two commercial IPTV services' traffic in four different types of residential broadband access networks: asymmetrical digital subscriber line (ADSL), cable, fiber to the building (FTTB), and fiber to the home (FTTH). It was concluded that D&P is an effective solution for IPTV deployment in the following ways: It is scalable and can deliver high-quality IPTV service even in existing best effort networks. No imminent upgrade to advanced backbone or access networks is necessary for the deployment of quality assured service. It can tolerate relatively poor network performance conditions better than the streaming solutions to support heterogeneous access networks.

Qiu [70] present a queuing model to study the effects of IPTV traffic on home networks. This model is used to ensure the QoS of IPTV. The analysis of the effects of IPTV traffic on Internet application running in home network such as web browsing was also reported. The traffic from other applications competes with the IPTV for the access bandwidth. If the total download rate exceeds the download bandwidth, packets will be dropped and hence it will degrade the video quality of IPTV. He solved this problem by giving IPTV packets priority over other packets. This mechanism can be implemented in the network layer or in the MAC layer and can be used to ensure the QoS of IPTV. Since IPTV traffic is given high priority, other Internet applications have to compete for the residual bandwidth and their performance may be affected. These applications such as HTTP, FTP, and SSH use TCP as the transport layer protocol and TCP itself has a built-

in congestion control mechanism meaning that when TCP detects that the network is congested, it will decrease its data rate. For that Qiu found that the TCP congestion control may not work well under this circumstance and proposed a new approach to improve the performance of TCP. In addition, he verified his result through ns2 simulation.

Xiao et al. [71] explained the main characteristics of the IPTV service, and reported several applications used by IPTV and indicated that IPTV might be a near revolution in the market. On the other hand, Quality of service (QoS) guarantee and traffic management is acknowledged as a technical challenge for the successful deployment of triple play services. The main purpose of traffic management is to support QoS requirements for the diverse set of triple play services, including policing, scheduling, flow control, traffic differentiation, and admission control.

Agrawal et al. [72] focused on a framework to aid planning and managing the deployment of IPTV services. The framework brings together a collection of models that capture the aggregate macroscopic behavior of representative aspects of an IPTV service that impact service performance and hence the quality of experience from the service, like the viewer profiles, zapping delay, data server blocking probabilities, etc. Some of these models were derived specifically for this framework, while others were adopted from previous studies, not related to IPTV. These models are used to map a set of external independent parameters which are assumed to be available to the service

deployment planner to a collection of performance metrics that are reflective of a particular deployment configuration.

4.3 IP MULTICAST

IP-Multicast is the most efficient approach to distribute video data from a single source to multiple receivers. Protocol Independent Multicast-Sparse Mode (PIM-SM) is the most widely deployed protocol for IPTV service in network Layer [73-75]. The PIM-SM protocol is designed on the assumption that recipient for any particular multicast group would be sparsely distributed throughout the network. It works as, if a host wants to add into a multicast group, it must join the shared tree which is also known as the RP tree (RPT, Rendezvous Point Tree). This tree relies on the central router called RP (Rendezvous Point), which receives all communications from sources and then forwards to receivers [76].

4.4 HOSPITALITY NETWORK

A hospitality network is used in private facilities such as in hotels, cruise ships or college campuses, and it offers entertainment, information, and communication services designed to end-user. This section describes our case study which is a small-scale hospitality network as shown in Figure 2.2 in Chapter 2. Figure 2.2 illustrates a typical hybrid network topology for a small-scale hospitality network. The network is Ethernet-based that can support at least 200 users. The network has four floors, each floor is a

subnet which is basically a Layer-2 switch, Cisco catalyst 2960 that can support at least 50 users, connected with the customer-premises equipment such as set-top Box for VoD, VoIP phone, and wireless access point. Floor 1 which is the lobby or known as reception has more than one access point to support more PCs in the conference room. Also, there are some users rooms in this floor. Every two floor subnets are connected with Layer-2 switches, Cisco catalyst 3560G, and these Layer-2 switches are connected with Layer-3 backbone switch, Cisco catalyst 5500. The backbone Layer-3 switch is connected with four servers. These four servers are in the server farm support E-mail, ERP, HDCP, and DNS services. All The links are 1Gbps except the links for the floor subnet which are 100Mbps.

4.5 IPTV TRAFFIC CHARACTERISTICS, REQUIREMENTS

Quality of Services (QoS) is very important for deploying IPTV as it is a real-time service. However, QoS for deploying IPTV will be affected by packet loss, packet reordering, packet faults latency, packet duplication, and jitter. There are several general QoS requirements described as follows:

Bandwidth. We used in our simulation High-Definition TV (HD) with a resolution 128X240 pixels, and the mean frame rates size of 34560 bytes. The mean transmission rate for this movie is 8Mbps with an encoding rate of 30 fps. The audio frame for the TV program and Movie clip is 6 packets in an audio frame, each 640 bytes in size, and the average rate is 44.1 kbps.

End-to-end delay. A end-to-end delay larger than 1 second may result in a much worse QoS toward end-user experience. However, if a small amount of delay occurs, it does not directly affect the Quality of Experience (QoE) of IPTV. Therefore, the delay for one way must be less than 200ms. On the other hand, the end-to-end delay more than 400ms was considered to be unacceptable [77].

Packet loss ratio. The excellent service quality of packet loss accepted ratio less than 10^{-5} and the poor service quality of packet loss ratio is between $2*10^{-4}$ and 0.01. The packet loss ratio above 1% is unacceptable [77].

Video equipments. The main equipments needed to deploy IPTV services are set-top boxes and head-end server. The head-end server or known as Video server is the source for all video content. It includes all the video streams that come from the broadcast stations, VoD servers, and so on. The Set-Top Box is a device that connects to a television, and it is responsible to receive and display digital-quality AV [65].

4.6 ANALYTICAL APPROACH

This section presents our analytical approach to deploy IP-Multicast AV TV services over a hospitality network. The actual number of users that the network can sustain is limited by two metrics. These metrics are the available bandwidth or delay as discussed in details in [3]. In order to determine the maximum number of users that can be supported by an existing network while maintaining the Video delay constraint, we develop an algorithm 3 that basically determines the network capacity in terms of

multicasting video sessions that can be sustained, by computing the number of users based on a delay bound. Users are added iteratively until the worst-case network delay of 80 ms has been reached, at which time the algorithm stops execution.

Algorithm 3 presents the overall computed maximum number of multicasting video sessions based on the video delay constraint. It takes as input the number of network elements, the background traffic for every element, the delay for network elements, the call flow paths, the multi-receivers, and the source node, and that it returns the maximum number of users supported by this network. It initializes with determine the required bandwidth for a single video session in terms of packets per second and also in terms of bits per second. Then, it builds a multicast tree based on the node level information, and includes the source with all the receivers in the tree. Algorithm 3 is implemented using MATLAB and the results for the worst-case delay are plotted in Figure 4.1, which shows that the maximum number of users that could be supported for AV TV, when deploying the High-Definition TV. The maximum number users that can be supported are about 172 users.

Algorithm 3: Compute maximum number of sessions based AV TV delay constraint

```
Input: n : number of network elements
       λ[1..n] : background traffic for network elements 1,2,.. n
       Delay[1..n] : delay for network elements 1,2,.. n
       P : set of call-flow paths (p) where p is a subset of {1,2,.. n}
       M: multi-receivers      S: source node
Output: MaxSessions: maximum number of Sessions
    λVideo ← 812pps,or 9.2Mbps; PktSize Video ← 11288 bits; Video_MaxDelay← 80;
    Video_MaxDelay_Found←0; MaxSession ← -1; Delay[1..n] ← 0;
 while Video_MaxDelay_Found < Video_MaxDelay do
     MaxSession ← MaxSession + 1
     V(T)= M ∪ {s}
     for each node v Є M do
       Generate a session and let pc be its flow path
       for each element i in pc do
            λi ← λi + λvideo
            if i is a link then
                Video_Delayi ← (1 − λi /2 μi ) (μi - λi ) * PktSize Video
              else
                Video_Delayi ←1 / (μi - λi )
              end if
        end for
     end for
     for each p in P_Video where p∩pc≠ ∅  do
         PathDelay(p)  ←∑ Video_Delayi , where i is a network element in path p
         if PathDelay(p) > Video_MaxDelay_Found then
            Video_MaxDelay_Found ←PathDelay(p)
         end if
     end for
 end while
```

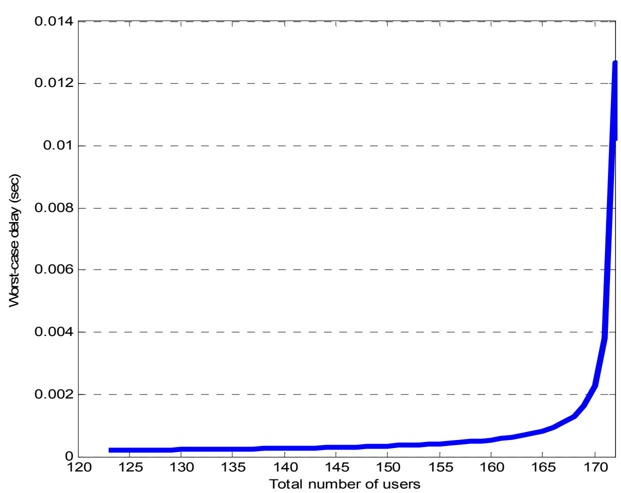


Figure 4.1 Worst case delay vs. number of users

4.7 OPNET SIMULATION

The OPNET Modeler is a commercial discrete-event simulator. It provides a global environment to model, simulate, and evaluate performance of all models of commercially available network elements, and has various real-life network configuration capabilities. This makes the simulation of real-life network environment close to reality [46]. This section describes in details the simulation model, various simulation configurations, as well as the simulation results.

4.7.1 Modeling Network

Our simulation modeling for the case study is shown in Figure 4.2. In OPNET, many vendor specific models are included in the pre-defined components libraries. However, the specific devices that we need are the Cisco catalyst 5500 Layer-3 switch which is configured as a router and the Cisco catalyst 3560G Layer-2 switch. One server is modeled as the head end server which is called video server. The head end server is responsible to run the profile which is modeled to support AV streaming. The Floor LANs are modeled as in Chapter 3.

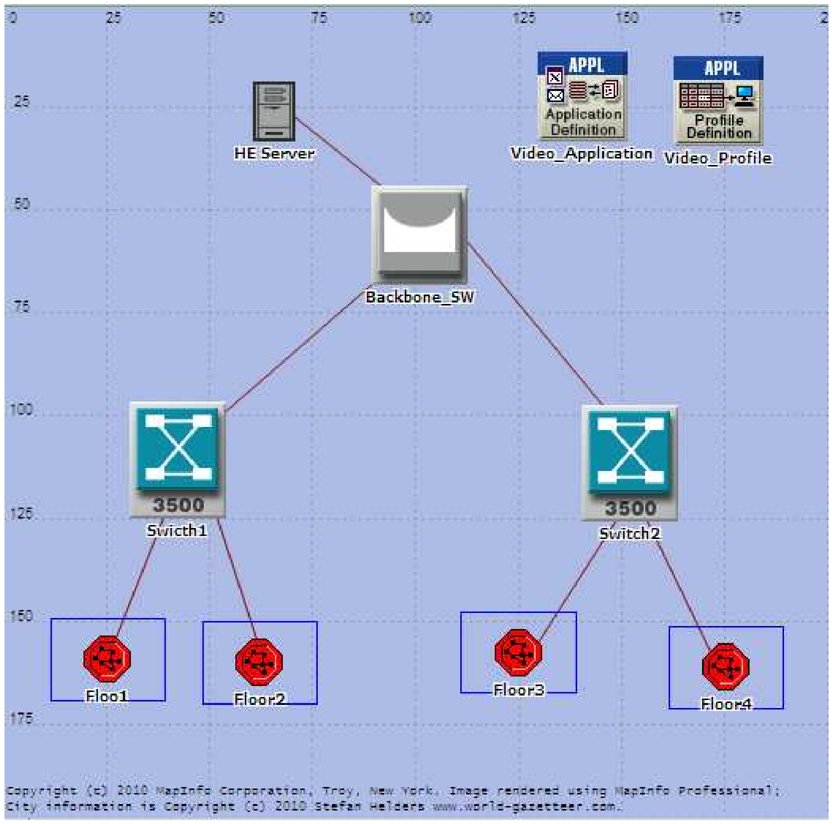


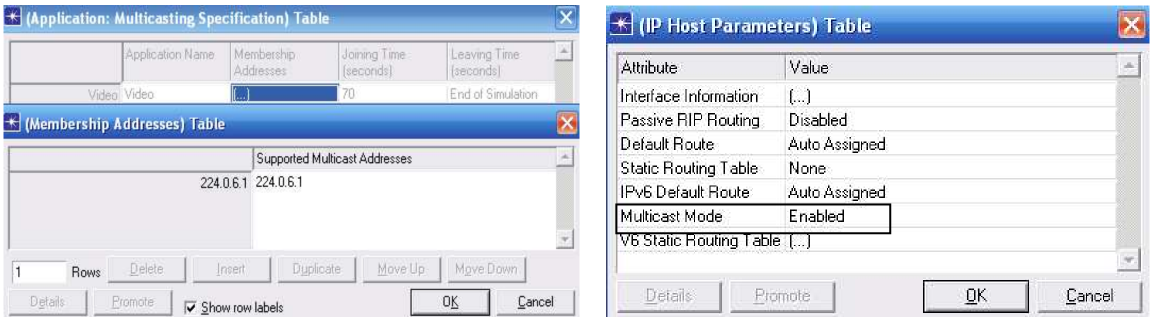
Figure 4.2 OPNET model of IPTV (live TV) hospitality network

4.7.2 Configuring IP-Multicast Protocol

PIM-SM routing protocol has been used in our simulation. This protocol can use any unicast routing information, such as the information of RIP or OSPF [76]. The configuration of this multicast protocol is as follows.

Destination Configuration. The Client destination joins an IP group address by specifying the group in the attribute application, multicast specification as seen in Figure 4.3(a). Figure 4.3(a) also shows that this client joins the membership group 224.0.6.1. Also, the IP multicasting can be enabled for each interface, and this can be configured by

changing the attributes “IP Host parameter → Multicast Mode” to enabled as seen in Figure 4.3(b).



(a) (b)

Figure 4.3 IP-Multicast for destination

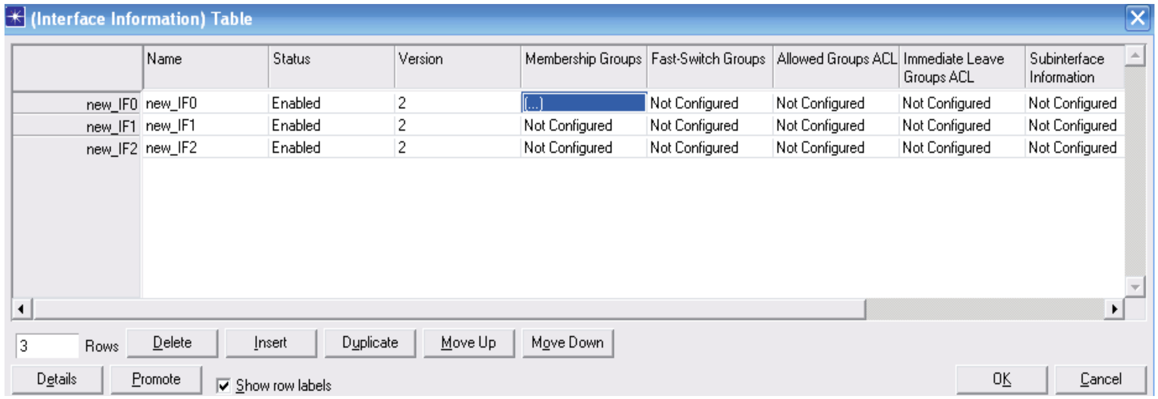
Server Configuration. A server can send the video content traffic to an IP group by specifying the destination address to be a valid class D address in the application “Destination Preferences attributes” as seen in Figure 4.4, and also the IP multicasting can be enabled same as in destination, as in Figure 4.3(b).

Figure 4.4 IP-Multicast for the HE server

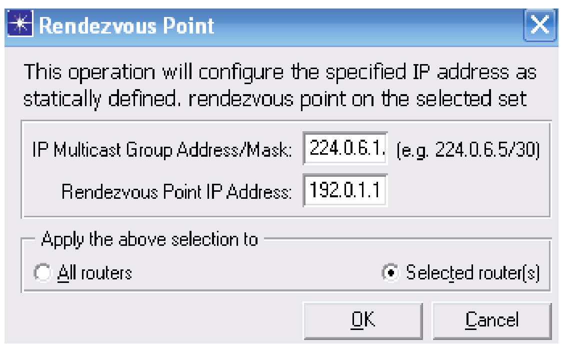
Router Configuration.

- Enable multicasting for each IP interface on routers.
    - Select all links on which IP Multicast traffic will flow through.
    - From the menu “Protocols→IP→Multicast”, select “Enable Selected Router on Selected Router Interface.”
- Create multicast group with Class D IP address and assign to it a Rendez-vous point.
    - A router can listen to a particular group by setting the group address in the IGMP static membership information table, or can do that from a menu “Protocols→IP→Multicast→Set IGMP membership group on selected Router”. This will add the specified group address in the IGMP Parameters in the IP Multicast attribute in the Router as seen in Figure 4.5(a).
    - Configure Rendzvous Point

      Selected the Router will be use as a Rendezvous Point, then select “Configure Rendezvous Point using Static RP Configuration” from the menu “Protocols→IP→Multicast”. Figure 4.5(b) shows the Static RP for our study. We set the IP Multicast RP (Rendezvous Point) to 192.0.1.1 with the IP Multicast group address 224.0.6.1. This will be seen in the “Static RP configuration” for the “PIM Parameter” in the selected Router as seen in Figure 4.5(c).



(a)



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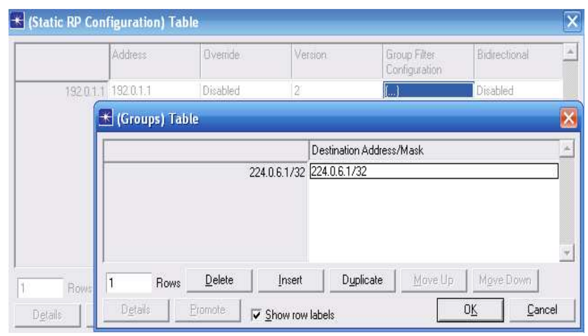


(c)

Figure 4.5 Router configuration

4.7.3 Generating AV Streaming Traffic

As we know, in a multicast technique, the sender is responsible for sending the packet to all receivers. So, the head end server is responsible to run the profile. Our profile is configured to run the audio stream in parallel with the video stream as shown in Figure 4.6.

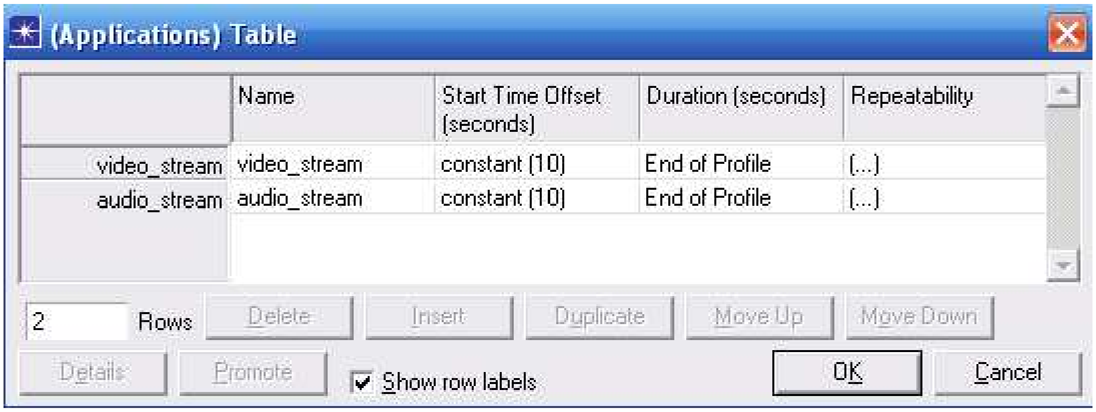
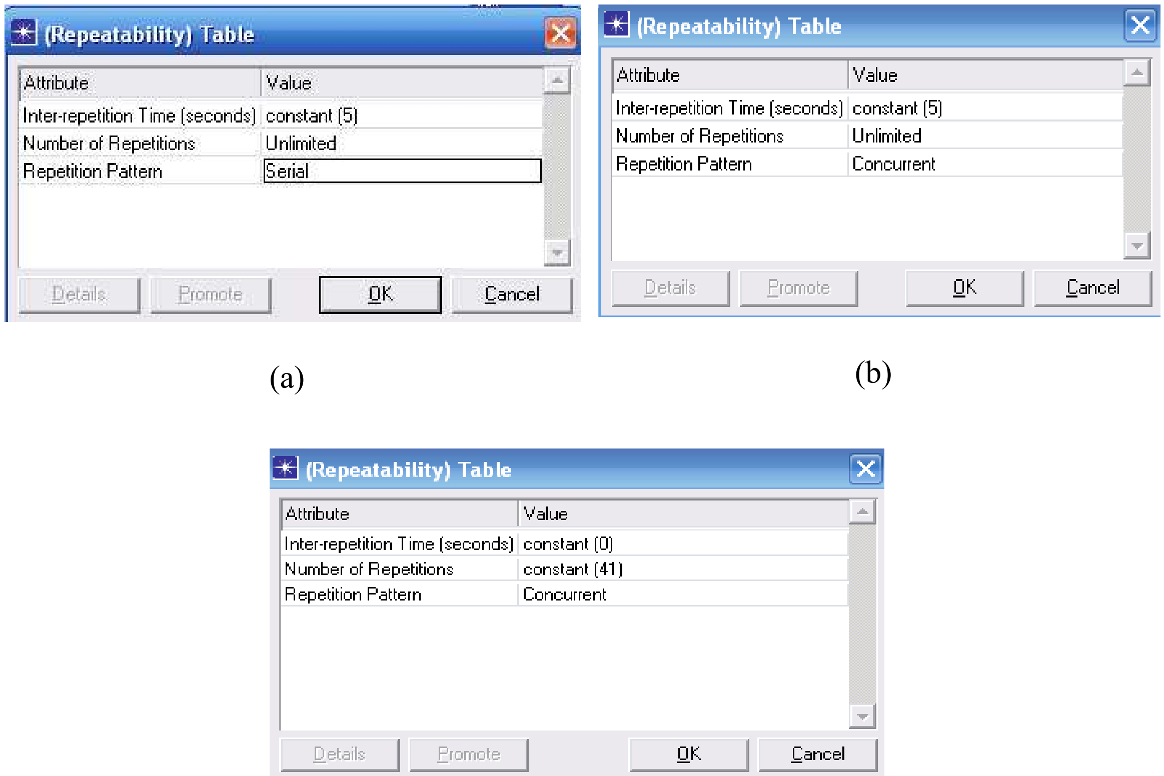


Figure 4.6 AV streaming profile configuration

To study how many users can be sustained by a network, we configured three scenarios. The first one, to deploy IPTV channel TV over our networks, we configure this scenario every 5 seconds 4 users are added. This shows how to add session to the 4 users, and from this scenario we can't reach the maximum number of users can be supported. Figure 4.7(a) shows the repeatability for this scenario. The second scenario is configured as every 20 seconds add session concurrent to 4 users, then add other session to other 4 users, and so on. Figure 4.7(b) shows the repeatability for this scenario. The third scenario has been configured to test the robustness of our network, in which 166 users will be added all at once. Figure 4.7(c) shows the repeatability for this scenario, and as we can see the inter–repetition time was constant (0), which means 166 users will be added all at once.



(a) (b)

(c)

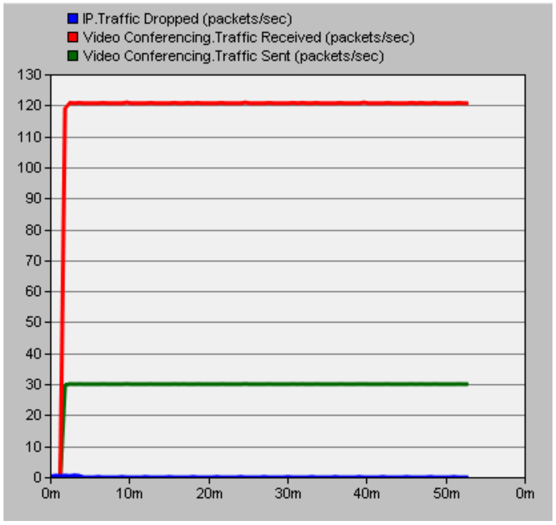
Figure 4.7 Repeatability of the application in profile

4.7.4 Simulation Results

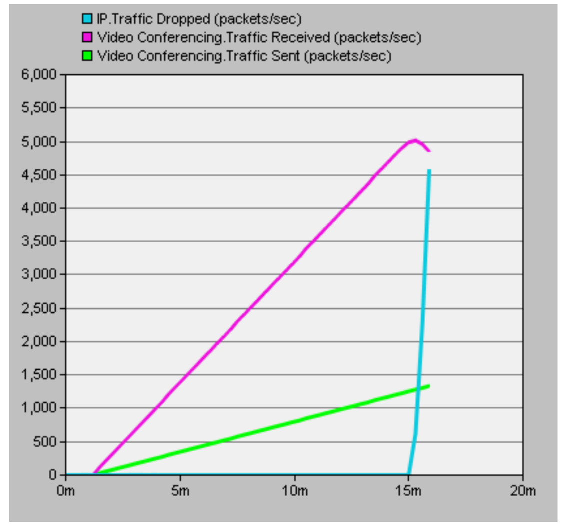
This section presents our simulation results to deploy the HDTV over existing hostility networks based on IP-Multicast. Our approach was to measure at the end of the simulation how many users can be sustained by this network. And also, to assess the readiness of our network to support IPTV while satisfying QoS requirement of all network services and leaving adequate capacity for future growth. We modeled three scenarios. The first scenario was with a profile modeled to added 4 users in serial. Since, this model adds the session to the same 4 users only. We can’t reach the maximum

number of users can be sustained. The profile in this scenario is modeled to start the AV traffic at 70 seconds of the simulation run with 4 users are added. This happens by sending the head end server session to the Backbone switch, then the Backbone switch replicate this session and send it to the four floors to support 4 users. After that, every 5 seconds a new session added to the same 4 users.

Figure 4.8(a) shows the 4 users added in the traffic received and the traffic sent one session added from the head end server, and then every 5 seconds session is added. The second scenario is modeled by adding 4 users to show the AV channel TV with 30 fps. Then, after 20 seconds another 4 users added again to see the same channel, and so on. Therefore, for that profile we reach the drop point from the router, which means we reached the maximum number of users that can be supported at the end of the simulation by our network. Figure 4.8(b) shows the traffic sent, received, and dropped for our simulation with a concurrent profile. From Figure 4.8(b), the maximum number of users that can be sustained by our network was 166 users, which supported by 41 sessions, each session for 4 users. Also, we can calculated the maximum number of users by equation [4+ (60*minutes+ seconds-70)*4/20]. The last successful user joint group was at 15 minutes. So, the maximum number of users that can be supported at this point is [4+ (60*15-70)*4/20]≈ 170 users. Results obtained from mathematical equation support our results from OPNET simulation, then after dropped point any user trying to join the group is dropped and all 166 users will show the same session. Figure 4.8(c) illustrates the results obtained when 166 users are added at same time while satisfying the QoS requirement of all network services and leaving adequate capacity for future growth.



(a)



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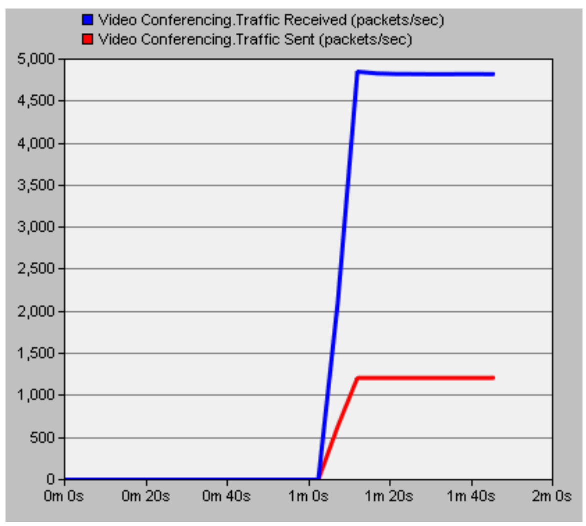

(c)

Figure 4.8 AV streaming traffic send, received, and dropped for 3 scenarios

We verified our simulation results by using some calculation as following. Our AV streaming contains audio streaming and video streaming. The average bit rate for the audio frame streaming is 44.1 kbps [78] which comes from, 6 packets in an audio frame,

each with 640 bytes in size for the TV program and Movie. So, the average bit rate for the audio stream in one session is approximately 30*44.1≈ 1323 kbps. The average bit rate needed for the HDTV is 8 Mbps. Figure 4.9 shows the total bandwidth average bit rate needed to send 41 session to support 166 users at the dropped point. The total bandwidth at 15 minutes was 372,894,828 bps. However, the bandwidth bit rate by our calculation was [41*(8 Mbps + 1323 kbps)]≈ 382,243,000 bps. The different between them is not much and the two results are close.

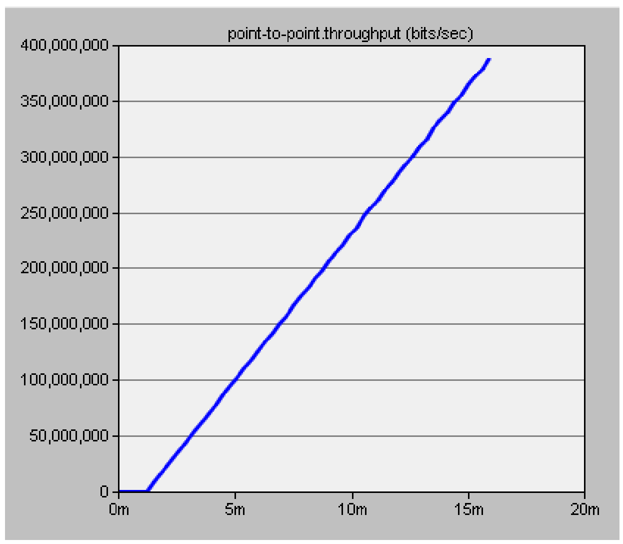

Figure 4.9 Bite Rate Bandwidth between HE server and router

Figure 4.10 shows the corresponding AV streaming End-to-End delay. The video packet end-to-end delay increases sharply after more users are added to the network. The end-to-end delay stays less than 80ms until a simulation time of 15 minutes as seen in Figure 4.10. The maximum number of users that can be supported at this point is equal to [4+(60*15-70)*4/20] = 170 users. Therefore, the maximum number of users that can be supported by the network bounded by using bandwidth less than that bounded by using

delay. Each one session supported 4 users. The maximum number of users that can be sustained by this network based on simulation is 166 users.

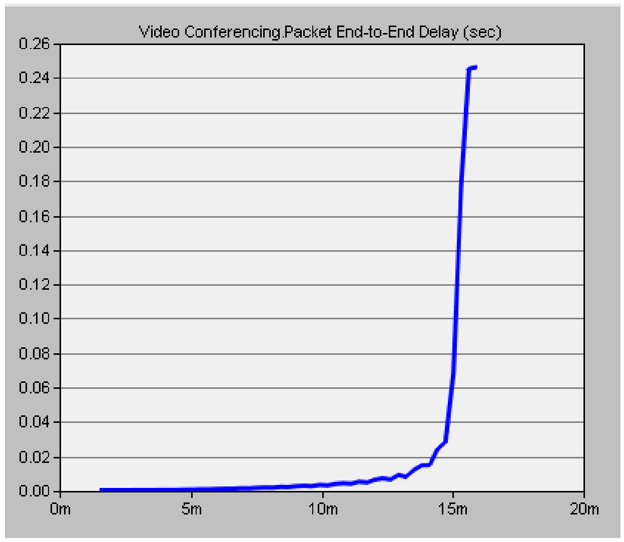

Figure 4.10 Packet end-to-end delays

4.8 DESIGN GUIDELINES AND ENGINEERING DECISIONS

By observation of the network under study, and based on our simulation and analytical results. The following network design and engineering decisions can be justified:

- The hospitality network under study, with a reserved growth factor of 25%, can safely support up to 166 users while satisfying the IPTV QoS requirements.
- For 170 users, a network delay of about 0.24 ms is encountered.
- The network capacity to support IPTV services is bounded by the network bandwidth, which support 166 users.

- The L3 switch was the primary bottleneck of this network. If we need to support more than 166 users we need to replace the L3 switch by another L3 switch with a larger capacity than the first one.

4.9 CONCLUSION

The chapter presented a simulation approach on how AV TV can be deployed successfully based on IP-Multicast. This approach was intended to predict the maximum number of users that can be sustained by a network, while satisfying the necessary QoS requirements of the new service, at the same time leaving adequate capacity for future growth of the data network. We considered a small-scale hospitality network as a case study of deploying IP-Multicast AV TV. Based on our simulation approach, a total of 166 users can be supported by the network, and based on the analytical approach, the maximum number of users that can be supported was 172 users. The difference between the two approaches was 6 users because the analytical approach was an approximation. As part of our future work, we intend to study the channel distribution on the hospitality network, considering the channel time zapping. Also, we intend to study assessing the network to support a new interactive service such as video games. Another thing we intend to study is the deployment of Quadruple-play services, which means the deployment of Triple-Play services in wireless networks.

CHAPTER 5

Conclusion and Future work

Deploying Triple-play services over IP networks has been increasing all over the world including North America, Europe, and Asia/Pacific. Our goal in this thesis was to predict the maximum number of voice and audio/video sessions that can be sustained by this network while satisfying QoS requirements of all Triple-Play services and leaving adequate capacity for future growth. In this chapter we present a summary of our major contributions in this thesis work to study the deployment of Triple-Play services over hospitality networks. It also gives indications of future research directions.

We presented an analytical and simulation approaches to deploy Triple-play services over hospitality networks. In particular, we considered the deployment of SIP-based VoIP, IP-Unicast of audio/video streaming VoD, and IP-Multicast AV IPTV. First, we present an analytical and simulation approaches to deploy VoIP based on SIP. The maximum number of VoIP calls obtained from the analytical approach and simulation were in line and close match, and supported at the end of simulation around 300 calls. Second, we present an analytical and simulation approaches to deploy AV streaming VoD based on IP-Unicast with 80 VoIP calls from the first part, the difference between the two approaches was 7 sessions. This difference can be contributed to the degree of

accuracy between the analytical approach and the OPNET simulation. The total number of sessions supported based on the analytical approach was 114 whereas based on the simulation approach it was 107 sessions. Finally, we present an analytical and simulation approaches to deploy IP-Multicast AV TV over hospitality networks. Results obtained from these approaches were in line and close match. Based on the simulation approach, a total of 166 users can be supported. However, based on the analytical approach 172 users can be supported.

The work presented in this thesis opens a new area for further research. The followings are some future directions.

- In our work, we studied only unique calls from a floor to another. We intend to study and conclude on the maximum number of VoIP calls sustainable on the hospitality network, when conference-based calls are included as part of the generated network traffic.
- In our work, we studied Triple-Play services. We intend to deploy Quadruple Play which means deploying Triple-Play services over wireless.
- In this work. We deployed only one channel for deploying IPTV and this not in reality, we intend to deploy channel distribution with considering the channel time zapping.
- Develop a GUI analytical tool to determine triple play capacity of an existing network.
- The introduction of new services such as video games opens another possibility for research.

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