الجامعة الاسلامية العالمية ماليزيا
INTERNATIONAL ISLAMIC UNIVERSITY MALAYSIA

VOICE OVER IP USING RSVP AND DIFFSERV

BY

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A DISSERTATION SUBMITTED IN PARTIAL FULFILLMENT OF THE REQUIREMENT FOR THE DEGREE OF
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ABSTRACT OF THE THESIS

Internet promises to deliver real-time, two-way synchronous voice traffic, as Internet telephony or Voice over IP (VoIP). The success of the Internet in this area depends on whether it provides the QoS requirement of a high-quality telephony service over an IP network. RSVP and Diffserv are two different techniques that can be used for achieving QoS of voice flows are explained in detail. The performance of the two distinct architectural models is studied in terms of throughput, end-to-end delay and jitter using Network Simulator version-2 (NS-2). The study proposes a per-EF aggregate admission control at ingress of Diffserv domain to protect the served voice flows from being affected by new voice flows that are over the bandwidth allocation of the EF aggregation. The implementation of the admission control shows superior performance of the proposed scheme.

ملخص البحث

يتعهد الإنترنت بتوفير خدمات التراسل الآنية باتجاهين كتراسل الخدمات الهاتفية بواسطة بروتوكولات الإنترنت (IP). ويعتمد نجاح الإنترنت في هذا المجال على إمكانية توفير الخدمات النوعية (QoS) للخدمات الهاتفية ذات النوعية العالية عبر شبكات الـ(IP).

تمثل كل من الـ(RSVP) والـ(Diffserv) تكنولوجيات بالإمكان استخدامها لتوفير الخدمات النوعية المطلوبة للمراسلات الهاتفية. وقد تم دراسة كفاءة التكنولوجيات أعلاه من خلال نماذج لدراسة عوامل التدفق والإعاقة وتغيير الإعاقة باستعمال برنامج محاكاة الشبكات NS-2.

وتقترح الدراسة "السيطرة على سماح" بدخول محمل مسارات من نوع EF في مدخل منطقة الـ Diffserv لضمان عدم تأثر المكالمات الهاتفية قيد الخدمة بواسطة مكالمات إضافية خارج نطاق الحزمة المخصصة.

وتبين نتائج الدراسة باستخدام "السيطرة على السماح بالدخول" كفاءة عالية جداً للأسلوب المقترح.

APPROVAL PAGE

I certify that I have supervised and read this study and that in my opinion it conforms to acceptable standards of scholarly presentation and is fully adequate, in scope and quality, as a thesis for the degree of Master of Science in Computer and Information Engineering.

Sufyan Al Irhayim
Supervisor

I certify that I have supervised and read this study and that in my opinion it conforms to acceptable standards of scholarly presentation and is fully adequate, in scope and quality, as a thesis for the degree of Master of Science in Computer and Information Engineering.

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Ahmad Faris Ismail
Dean, Kulliyyah of Engineering

DECLARATION

I hereby declare that this thesis is the result of my own investigations, except where otherwise stated. Other sources are acknowledged by footnotes giving explicit references and a bibliography is appended.

Name: Qahhar Muhammad Qadir

Signature .... Date ....

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To my beloved parents and all my sisters and brothers…

Without whom this thesis would not has been done.

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LIST OF ACRONYMS

AF PHB	Assured Forwarding Per Hop Behaviour
AS	Autonomous Systems
ASDR	Approximate State-Dependent Routing
ATM	Asynchronous Transfer Mode
BA	Behaviour Aggregate
BAC	Behaviour Aggregate Classifier
BE	Best Effort
CL	Controlled Load
Diffserv	Differentiated Service
DPO	Direct Path Only
DSCP	Differentiated Service Code Point
EF PHB	Expedited Forwarding Per Hop Behaviour
FCFS	First Come First Service
FIFO	First In First Out
FTP	File Transfer Protocol
GPRS	General Packet Radio Service
GS	Guaranteed Service
IETF	Internet Engineering Task Force
Intserv	Integrated Service
IP	Internet Protocol
ISP	Internet Service Provider
ITU.T	International Telecommunications Unit. Telecommunications section
LAN	Local Area Network
LSP	Label Switch Path
MC	Multifields Classifier
MPLS	MultiProtocol Label Switching
NS-2	Network Simulator-version 2
OTcl	Object-oriented Tool Command Language
PSTN	Public Switched Telephone Network
RED	Random Early Detection
RFC	Request For Comments
RSVP	Resource reSerVation Protocol
SAPF	Shortest Available Path First
SDR	State-Dependent Routing
SLA	Service Level Agreement
SMTP	Simple Mail Transfer Protocol
SNMP	Simple Network Management Protocol
SPF	Shortest Path First
ST2	Stream Transport Protocol – Version 2
TCA	Traffic Conditioning Agreement
TCP/IP	Transport Control Protocol/Internet Protocol
TCS	Traffic Conditioning Specification
TE	Traffic Engineering
VLL	Virtual Leased Line
VPN	Virtual Private Network
WAPF	Widest Available Path First
WRR	Weighted Round Robin

CHAPTER 1

INTRODUCTION AND LITERATURE SURVEY

1.1 Introduction

Internet has seen enormous growth in both the number of users and in the demand for new services from applications. Today, traditional telephone carriers, Service Providers, and large corporations face the challenge of integrating into a single packet network.

The adoption of the Internet in various areas has surpassed other technologies such as radio, television, and personal computer. The Internet is becoming a convenient and cost-effective medium for collaboration, education, e-commerce, entertainment, and telephony. Internet convergences data, voice, and video in one common medium. Internet traffic is growing due to the increasing amount of various services provided by the Internet.

Internet promises to deliver real-time, two-way synchronous voice traffic, called Internet telephony or Voice over IP (VoIP). There has been substantial interest in recent past in migrating telephony service away from circuit-switched networks onto an IP-based packet-switched network infrastructure as it is more cost effective. Since a telephony service requires stringent bounds on end-to-end packet delay, jitter, and loss, the success of the Internet in this area depends on whether it provides the QoS requirement of a high-quality telephony service over an IP network.

Nevertheless, IP telephony provides a number of benefits compared to the Public Switched Telephone Network (PSTN). Beside the integration of voice, data, and fax, VoIP can provide many other facilities such as: sound grading, video telephony, unified messaging, low cost voice calls, real-time billing, remote teleworking, enhanced teleconferencing, etc. A major challenge for achieving this, is the Internet itself. Since the Internet was designed for non real-time data communications, without having QoS in mind, it poses several technical challenges before the Internet can be successfully used for carrying these telephony services. Loss, delay and jitter are some of these challenges.

The 21st century has shown that the Internet is experiencing a remarkable growth in all aspects: the number of users, the number of hosts, the amount of traffic, the number of links, the bandwidth of individual links, and the variety of applications' services. This rapid pace of growth brought about two main challenges; queue management and forwarding at routers and switches.

The increasing demand for Internet and multimedia services is straining the existing telecommunications infrastructure. The Internet relies on the protocol layer IP (3rd layer in TCP/IP suite) that enables interoperation of various network technologies, but provides only the simple QoS, point-to-point best effort data delivery in which, all user packets compete equally for network resources. IP network has been sufficient until recently when the usage and popularity of IP network soared placing a significant burden on limited network resources, such as bandwidth and buffer space resulting in heavy congestion. It became evident that some QoS of IP network needs to be developed to offer differing levels of treatment to user packets. Heavy network utilization causes traffic congestion that causes variable queuing delays and congestion losses on IP networks, which are not efficient for real-time and mission-critical applications. Real-time communications services over the Internet, needs a new architecture to meet these required quality of services. Schemes to reduce delay, loss, jitter and provide bandwidth on demand are required. A high number of voice flows needs effective flow management. Per aggregation management must be applied instead of costly per-flow management. A capacity planning and provisioning mechanisms are required to make the probability of blocking offered calls be fairly small to ensure that the network has adequate capacity to handle the expected traffic volume, and efficient routing mechanisms are required to ensure that the offered traffic is routed over the network in a manner that makes efficient use of available network capacity.

For achieving the above requirements, researchers are working on increasing the Internet raw bandwidth, developing resource reservation protocols, developing new Internet communications models and architecture, and developing adaptive and scalable transmission and presentation mechanisms to maximize QoS with limited network resources.

A number of research papers, experiments, and prototypes appeared in the last few years to extend the internet architecture and protocols to support not only QoS over IP networks, but to control the sharing of bandwidth on a particular link among different traffic classes as well. Internet Engineers Task Force (IETF) RFC 1633 defined the components and requirements of Integrated Services (Intserv), then IETF RFC 2475 defined Differentiated Services (Diffserv) and RFC 3031 defined MultiProtocol Label

Switching (MPLS) models. These extensions are necessary to meet the growing need for real-time services for various multimedia applications, while maintaining the fundamental service model of the Internet. Intserv uses Resource reSerVation Protocol (RSVP) as signalling protocol to reserve the necessary resources along the path as a way for achieving QoS. Diffserv is another IP QoS architecture based on packet marking with a so-called codepoint that allows packets to be prioritised at each node according to user requirements. It can provide scalable multi-class of services in IP network solving the scalability problem of RSVP in traditional Intserv and offering QoS on aggregations of flows, in contrast to Intserv, which was a per-flow basis. MPLS as an advanced forwarding scheme, convergences the connection oriented forwarding techniques and the Internet's routing protocols based on labelling and establishing Label Switched Path (LSP). Furthermore, MPLS provides gigabit forwarding, network scalability and Traffic Engineering (TE).

Issues related to QoS over IP networks have attracted a lot of researchers attention recently. Many conferences and even journals are exclusively dedicated to issues related to this subject. Most of these papers address specific issues, but a few touch on the performance of delay-sensitive traffic such as voice over Intserv/RSVP and Diffserv.

One of the recent papers by Mishra (2000) examines how the capacity management and routing mechanisms used in IP networks can be augmented to support an IP telephony service. It evaluates the performance of two proposed distinct architectural models for the service. In the first model, it simulates an enhanced IP network to support IP Intserv transport and QoS routing, while the second simulates a normal IP network which can support an overlay Virtual Private Network (VPN) with dedicated capacity for the VoIP service. The performance of these architectural models and their associated routing policies [Shortest Path First (SPF), Shortest Available Path First (SAPF), and Widest Available Path First (WAPF) in the first model and Direct Path Only (DPO), Success to the Top (STT), State-Dependent Routing (SDR), and Approximate State-Dependent Routing (ASDR) in the second model] were evaluated using simulation. The simulation results realize the QoS achievement of a high-quality telephony service over IP network and make efficient use of network capacity.

Another interesting study, Priggouris et al. (2000) simulated two QoS enabling schemes (Intserv and Diffserv) to enhance the General Packet Radio Service (GPRS) architecture. The simulation results of the two schemes were used to suggest solutions in order to overcome the lack of GPRS QoS differentiation basis of IP flows. This can serve as an alternative to the IP address of a mobile station, and the use of IP tunnels of the GPRS core network, which makes the applicability of IP QoS schemes troublesome. Its proposed solutions to the problems include the establishment of QoS reservations across the GPRS core network, and the required signalling enhancements, and suggested particular modifications in the components of the GPRS architecture. An important conclusion from the simulated model of the Intserv architecture of the GPRS cellular infrastructure showed that the proposed Intserv has good scalability, even for a large number of users.

Techniques and approaches required to support guaranteed end-to-end QoS for multimedia communications over the Internet are provided in Lu (2000). The article identifies the main requirements for multimedia communications such as low end-to-end delay, jitter, loss, and bandwidth. It discusses the main categories of multimedia applications and the proposed IETF architectures for providing QoS such as Intsev, Diffserv and MPLS are studied in details.

The efficiency of IP network for IP telephony can be related to the volume of voice traffic carried with deterministically guaranteed quality, and the amount of network resources used. A simulation study Baldi and Risso (2000) found that the efficiency of the network in the presence of best effort traffic is maximum when the percentage of best effort traffic prevails and the number of nodes on the path of voice calls is small. An IP network carrying compressed voice is compared to circuit-switching carrying PCM (64kb/s) encoded voices. Some design choices affecting IP telephony efficiency are discussed also.

A detailed explanation of the various methods used to achieve QoS over IP network is given by Xiao and Ni (1999). Each of Intserv, RSVP, Diffserv, MPLS and Constraint Based Routing and how they differ from, related to, work with each other to deliver QoS on the Internet are described in detail. It argues that the Diffserv alone is not sufficient for providing QoS, thus a framework for emerging Internet QoS using the above components is presented. The framework consists of Intserv/RSVP and Diffserv at the Transport layer, constraint based routing at the network layer, and MPLS at the link layer. It also describes how MPLS and constraint based routing may be added to Diffserv architecture to meet QoS requirements for better performance. Finally, a comparison between ATM network and network with routers that implement Diffserv and MPLS is given.

The implementation of RSVP protocol may not incur much overhead over existing IP networks. An experimental study for an industry-strength RSVP implementation on a commercial IP router is given by Neogi et al. (1999). An IP router testbed of five routers organized in a square-like topology in a commercial site is installed for the performance experiments and traffic statistics collection. The article recommends that the RSVP's control messages do not incur significant overhead in terms of processing delay and bandwidth consumption, while control messages may be sent at higher priorities. The study has proven that the performance overhead of real-time packet scheduling is nonnegligible in the presence of a large number of real-time connections.

Achieving specific QoS requirements of individual flows by managing available resources to aggregated flows is studied by Yeom and Reddy (2001). This situation leads to unequal bandwidth sharing within aggregated flows due to different Round Trip Time (RTTs) within aggregated flows. Yeom and Reddy (2001) proposes two new aggregated marking algorithms, called IN-Fair and Bandwidth Fair marking for managing the contracted bandwidth among the individual flows of an aggregation. Both algorithms maintain state of individual flows such as individual marking rate, current sending rate at the edge of the network and their effective utilization in adaptively marking packets of individual flows to meet their QoS requirements. Resource need to be transferred to the edge of the network on the sender's side for improving the service provided to receiving-insensitive application and achieving QoS requirements of individual flows.

Finally, Jeong et al. (2001) has proposed a set of router-based QoS mechanisms including queue policy, resource reservation and marking to support UDP/TCP applications over the Internet. It proposed a network architectural framework including service specification and QoS mechanisms of integration of RSVP/Intserv and Diffserv for achieving end-to-end QoS with scalability for UDP/TCP applications. The edge routers define the IP packet header field for packet state information. This is used to classify packets into three services; rate-guarantee service (RGS) which supports CLS in the Intserv region, for UDP traffic, Better-than-Best-Effort Service (BBES) for TCP traffic and Best-Effort Service (BES) for other flows. Buffer space is allocated within the core network to provide lossless rate guarantees to the QoS-UDP class, and simple congestion detection/notification and rate adaptation mechanisms are used to provide equal bandwidth for TCP flows. The edge router maintains the traffic profile for each flow. The border router maintains only a single traffic profile/meter per aggregate traffic class to provide the scalability. A proposed lightweight intra-domain signalling and a simple admission control in the core network may provide resource reservation without maintaining per-flow state in the core routers. Per-aggregation traffic metering is done at the border routers for each QoS-UDP and TCP to ensure that incoming flows are in compliance with their agreed-upon rates, while per-flow metering is done at the edge routers for each QoS-UDP and TCP flows separately.

1.2 Thesis Statement

There has been a great interest in migrating telephony and other delay sensitive traffic to IP-based networks; however, the lack of QoS caused delay, jitter and unpredictable behaviour.

IETF proposed some architectures that can provide QoS to traffic over IP networks. Diffserv EF and RSVP can provide a certain QoS for voice traffic over IP network. Both of these models are promising contenders for implementation to bring telephony and other delay-sensitive services to IP networks. However, each of these two models can provide the QoS to the IP network in a different way. RSVP provides each flow with the required bandwidth. EF PHB is a suitable technique for voice transmission over the Internet due to its low delay, jitter and loss. On the other hand, RSVP does not guarantee the required delay and jitter and it is not a scalable architecture. Traffic aggregation model used by Diffserv adds some unpredictability. Per-aggregate treatment of Diffserv and lack of per-aggregate admission control make the ingress of a Diffserv domain examine the aggregate arrival of EF flows even if they are out of the EF allocated bandwidth. The active EF flows are affected by the new EF flows, causing decrease in throughput and increase in delay of the active EF flows, which is not acceptable for voice transmission.

Implementing a simple per-aggregate admission control at ingress of Diffserv domain solves the above problems by simply blocking the new EF calls without increasing the complexity of the ingress if the available resources are not sufficient for them. To overcome the weakness of each model and keep the advantages of each of them, the current trend is to use Diffserv supplemented by some of the resource reservation capabilities of RSVP (Roy, 2000). In this case, the Intserv model can be successfully deployed in the edge part of the network without compromising scalability.

We use the computer simulation to evaluate the performance of two distinct architectural models for QoS. Network simulator version-2 (ns-2) is used to create

Diffserv and RSVP domains and to implement Diffserv EF PHB and RSVP. The extensibility and free of charge availability make ns to be used widely by many researchers around the world. They can download it free from the Internet and add their new algorithms to the extensible background engine using C++. Topologies for each Diffserv and RSVP have been simulated in different conditions to find out the advantages and weakness of each of them for carrying voice.

The thesis studies the issue of QoS over IP networks with a particular focus on time-sensitive traffic such as voice over IP. It examines performance characteristics such as end-to-end delay, packet loss, jitter and bandwidth availability for various IETF proposed architectural models with particular emphasis on Diffserv and RSVP. Detailed architectural models were constructed for both protocols using simulation. It suggests a per-aggregate admission control to be implemented at ingress of Diffserv domain. It protects the in place calls from being affected by the unallowable new calls.

The thesis is organized as follows: chapter one is the introduction to the subject and literature survey. A theoretical explanation of each of IETF Diffserv and Intserv models is given in chapters two and three separately respectively. Chapter four is the simulation work of each of them. Chapter five comprises their results and discussion. Finally, chapter six concludes the thesis and recommends for further work.