

الآية

بسم الله الرحمن الرحيم

﴿اللَّهُ نُورُ السَّمَاوَاتِ وَالْأَرْضِ مَثَلُ نُورِهِ كَمَشْكَاةٍ فِيهَا ۚ مَصْبَاحُ الْمَصْبَاحِ
فِي زُجَاجَةٍ الزُّجَاجَةُ كَأَنَّهَا كَوْكَبٌ دُرِّيٌّ ۚ يُوقَدُ مِنْ شَجَرَةٍ مُبَارَكَةٍ زَيْتُونَةٍ لَا
شَرْقِيَّةٍ وَلَا غَرْبِيَّةٍ يَكَادُ زَيْتُهَا يُضِيءُ وَلَوْ لَمْ تَمْسَسْهُ نَارُ نُورٍ ۚ عَلَى نُورٍ
يَهْدِي اللَّهُ لِنُورِهِ مَنْ يَشَاءُ ۚ وَضَرَبَ اللَّهُ الْأَمْثَالَ لِلنَّاسِ وَاللَّهُ بِكُلِّ شَيْءٍ
عَلِيمٌ﴾

صدق الله العظيم

[سورة النور: ٣٥]

Acknowledgement

First of all, I am deeply grateful to Allah, my Lord, who gave me the power and confidence to pursue this work and complete my MSc studies.

I would like to be appreciative for help and support of the very kind people around me, especially I would like to thankful to my supervisor Dr. Rashid A. Saeed his greater guidance and for technical support as well in this project. Thanks for him to make it possible for successful completion of this project.

I would especially thank the electronic engineering school staff.

Table of contents

الاية.....	i
Acknowledgement.....	ii
Table of contents.....	iii
List of figures.....	v
Abbreviations.....	vi
Abstract.....	vii
المستخلص.....	viii
Chapter One: Introduction	
1.1 Background.....	1
1.2 Problem Statement.....	3
1.3 Objective.....	3
1.4 Research Methodology.....	3
1.5 Project Scope.....	4
Chapter Tow: Related work	
2.1 Introduction.....	6
2.2 Voice over IP (VoIP).....	6
2.2.1 VoIP Network Components.....	7
2.2.2 VoIP Signaling Protocols.....	8
2.2.3 VoIP Challenges.....	11
2.4 VoIP over Transmission Protocols.....	12
2.4.1 VOIP over TCP.....	12
2.4.2 VOIP over UDP.....	15
2.4.3 VOIP over RTP.....	16

2.5 Previous Studies.....	19
2.6 Quality of Service (QoS).....	20
2.7 VoIP QoS Factor.....	21
2.7.1 Throughput.....	22
2.7.2 Packet Loss.....	23
2.7.3 End To End Delay.....	23
2.7.4 Jitter.....	26
2.8 Summary.....	27
Chapter Three: Methodology	
3.1 Introduction.....	28
3.2 Simulation Scenario.....	28
3.2 Design Module.....	28
3.2.1 Ns-2 Implementation.....	29
3.2.2 Filtering the Trace File With AWK.....	30
3.2.3 Parsing and Plotting With MATLAB.....	32
3.2.4 Simulation to Measure QoS.....	33
3.3 Summary.....	37
Chapter Four: Results and Discussion	
4.1 Introduction.....	38
4.2 Throughput	39
4.3 Packet Loss.....	41
4.4 End-To-End delay.....	43
4.5 Jitter.....	45
4.6 Summary.....	47
Chapter Five: Conclusions and Recommendations	
5. Conclusions and Recommendations	48
References.....	49

List of figures

Figure (1-1): Methodology flow chart.....	4
Figure (2-1): VoIP protocols over TCP/IP stack.....	9
Figure (2-2): H.323 Header.....	10
Figure (2-3): TCP header.....	12
Figure (2-4): UDP header.....	15
Figure (2-5): RTP Header.....	17
Figure (3-1): NS-2 implementation of a VoIP network.....	29
Figure (3-2): NS-2 trace file.....	31
Figure (3-1-a): Simulation flowchart (throughput).....	33
Figure (3-1-b): Simulation flowchart (packet loss).....	34
Figure (3-1-c): Simulation flowchart (delay).....	35
Figure (3-1-d): Simulation flowchart (jitter).....	36
Figure (4-1): Throughput from node (0) to node (4).....	39
Figure (4-2): Throughput from node (4) to node (0).....	40
Figure (4-3): Packet loss from node (0) to node (4)	41
Figure (4-4): Packet loss of from node (4) to node (0)	42
Figure (4-5): End-to-End Delay from node (0) to node (4)	43
Figure (4-6): End-to-End Delay from node (4) to node (0)	44
Figure (4-7): Jitter from node (0) to node (4)	45
Figure (4-8): Jitter from node (4) to node (0)	46

Abbreviations

CGI	Common Gateway Interface
Codecs	Coders/Decoders
CPL	Call Processing Language
HTTP	Hyper-text Transfer Protocol
IETF	Internet Engineering Task Force
IP	Internet Protocol
LAN	Local Area Network
MGCP	Media Gateway Control Protocol
MPLS	Multi-Protocol Label Switching
NS-2	Network Simulation Version 2
PBX	Private Branch eXchange
PC	Personal Computer
PSTN	Public Switching Telephone Network
QoS	Quality of Service
RTP	Real-time Transport Protocol
RSVP	Reservation Protocols
SCTP	Stream Control Transmission Protocol
SIP	Session Initiation Protocol
SMTP	Simple Mail Transfer Protocol
SS7	Signaling system 7
TCP	Transport Control Protocol
UDP	User Datagram Protocol
VoIP	Voice over Internet Protocol
WAN	Wide Area Network

Abstract

In recent years, Voice over IP (VoIP) has gained a lot of popularity and become an industry favorite over Public Switching Telephone Networks (PSTN) with regards to voice communication. The project consists of creating a VoIP network and testing for its known faults. Through this project wanted to get a better understanding of the underlying layers of the network and see if and where improvements can be made. In implementation stage the RTP packets for VOIP applications had been sent and compared with TCP/UDP packets to obtain results which are mainly related to QoS factors. The attained result approved that RTP consider to be batter to reduce a packet loss than UDP, and also approved that UDP/RTP are most reliable because they had a very small delay and jitter in contrast with TCP. Hence, find that, UDP/RTP are more balance and prefer than TCP in real-time applications such as VOIP.

المستخلص

في السنين الحالية كسب تطبيق نقل الصوت عبر بروتوكول الانترنت انتشاراً واسعاً واصبح المفضل في الصناعة من شبكات تبديل الهواتف العامة مع الاخذ في الاعتبار إتصالات الت الصوت. المشروع يتكون من انشاء شبكة لنقل الصوت عبر بروتوكول الانترنت واختبار اعطالها المعروفة . من خلال هذا نريد الحصول علي فهم واضح لطبقات الشبكة ولنري اين يمكن التحسين وكيف . في مرحلة التطبيق . حزم بروتوكول الزمن الحقيقي لتطبيقات الصوت عبر بروتوكول الانترنت تم ارسالها ومقارنتها مع حزم بروتوكول بيانات المستخدم وحزم بروتوكول التحكم في النقل للحصول علي نتائج متعلقة اساساً بعوامل كفاءة الخدمة. اثبتت النتيجة المحرزة ان بروتوكول الزمن الحقيقي يعتبر الافضل لتقليل فقدان الحزم مقارنة ببروتوكول بيانات المستخدم واثبتت ايضاً انها الاثنان اكثر اعتمادية لان لديهما زمن تأخير صغير جداً مقارنة ببروتوكول التحكم في النقل. وهكذا وُجد ان بروتوكول الزمن الحقيقي و بروتوكول بيانات المستخدم اكثر موازنةً وتفضيلاً مقارنةً ببروتوكول التحكم في النقل.