

Implementation of Voice over Internet Protocol (VoIP) in UUM Campus

Feras Zen Alden

UNIVERSITI UTARA MALAYSIA

2011

**Implementation of Voice over Internet Protocol (VoIP) in UUM
Campus**

**A thesis submitted to the Academic Dean Office in partial fulfillment
of the requirement for the degree Master of Science
(Information Communication Technology)**

Universiti Utara Malaysia

By

Feras Zen Alden

Copyright © Feras Zen Alden, 2011. All rights reserved

PERMISSION TO USE

In presenting this project of the requirements for a Master of Science in Information Communication Technology (MSc. ICT) from Universiti Utara Malaysia, I agree that the University library may make it freely available for inspection. I further agree that permission for copying of this project paper in any manner, in whole or in part, for scholarly purposes may be granted by my supervisor(s) or in their absence, by the Dean of Graduate School. It is understood that any copying or publication or use of this project or parts thereof for financial gain shall not be allowed without my written permission. It is also understood that due recognition shall be given to me and to Universiti Utara Malaysia for any scholarly use which may be made of any material from my project paper.

Request for permission to copy or make other use of materials in this project, in whole or in part, should be addressed to:

Dean of Research and Postgraduate Studies

College of Arts and Sciences

Universiti Utara Malaysia

06010 UUM Sintok

Kedah DarulAman

Malaysia

ABSTRACT

Currently, users (student, staff and lecturer) at UUM still using traditional mechanism to contact with each other using phone call and messenger contact. In this case, the users need to have an Internet access in order to communicate with the other side. In both cases, using phone call or Internet has some limitations in terms of cost. This paper proposes the Voice over Internet Protocol (VoIP) system that can help users at UUM campus to freely communicate by using this VoIP technique. In the other side, this proposed system also helps to increase the effectiveness of using the Internet bandwidth; since the users can communicate with each other without the need to have an Internet access. Instead, they can contact with each other using the current Local Area Network (LAN) at UUM. Thus, this system can let the users to contact the destination user anywhere anytime in the coverage area of UUM.

Acknowledgments

In the Name of Allah, the Most Gracious and Most Merciful

All Praises to Allah for his guidance and blessing for giving me the strength and perseverance to complete this study. I would foremost like to thank my beloved family, for providing me with the opportunity to pursue my goals and for their love and affection, which has helped me along my study stages and through all my life. I would like to express my deepest gratitude to my supervisors Mr. Adib M.Monzer Habbal and Mr. Khuzairi bin Mohd Zaini for them guidance, instructions, and his advices that have enabled me to complete my project properly. Last my thanks would go to my all friends and classmates who gave me their help and shared with me their knowledge.

Feras Zen Alden /30-6-2011

TABLE OF CONTENTS

PERMISSION OF USE.....	i
ABSTRACT.....	ii
ACKNOWLEDGEMENT.....	iii
TABLE OF CONTENTS.....	iv
LIST OF FIGURE.....	vii
LIST OF TABLE.....	Viii
LIST OF ABBREVIATION.....	ix
CHAPTER ONE: INTRODUCTION.....	1
1.0. Background.....	1
1.1. Problem Statement.....	2
1.2. Research Questions.....	3
1.3. Research Objectives.....	3
1.4. Significance of the Study.....	3
1.5. Scope and Limitation of the Study.....	4
1.6. Organization of the Report.....	5
CHAPTER TWO: LITERATURE REVIEW.....	6
2.0. Introduction.....	6
2.1. Definition of VoIP System.....	6
2.2. Background of VoIP Technology	6
2.3. Benefits of VoIP.....	7
2.4. VoIP Protocols.....	8
2.4.1. H.323 Overview.....	8
2.4.1.1.Basic Call Flow	9
2.4.2. SIP Overview	11
2.4.2.1. Basic Call Flow	11
2.4.3. RTP& RTCP Overview	13
2.4.4. TCP Overview.....	14
2.4.5. UDP Overview.....	15
2.4.6. How Does VoIP Technique Work?	15

2.5. VoIP Planning and Implementation.....	16
2.6. Related Works	18
2.7. Chapter Summary	20
CHAPTER THREE: RESEARCH METHODOLOGY.....	21
3.0. Introduction.....	21
3.1. Research Methodology.....	21
3.1.1. The Awareness of the Problems.....	22
3.1.2. Suggestions.....	23
3.1.2.1. Hardware requirements of the system.....	23
3.1.2.2. Software requirements of the system.....	23
3.1.3. Implementation	23
3.1.3.1. Analysis.....	24
3.1.3.2. Construction.....	24
3.1.3.3. Testing.....	25
3.1.4. Evaluation.....	25
3.1.5. Conclusion.....	25
3.2. Summary.....	25
CHAPTER FOUR: SYSTEM ANALYSIS AND DESIGN.....	26
4.0. Introduction.....	26
4.1. Design stage	26
4.1.1. Network Design	26
4.1.2. Types Of Service	27
4.1.3. Call Flow	28
4.2. Implementation Stage	29
4.2.1. Server Configuration	30
4.2.1.1. Define SIP User	33
4.2.1.2. Dial Plan Scripts	37
4.2.1.3. Application Testing.....	37
4.2.2. User Configuration	38
4.2.2.1. Choose VoIP Softphone Program.....	39
4.2.2.2. Testing VoIP System By Creating SIP Account	41
4.3. Summary.....	43

CHAPTER FIVE: SYSTEM EVALUATION.....	44
5.0. Introduction.....	44
5.1. User Interview	44
5.2. Evaluation Process Of The Application	44
5.2.0. Evaluation Of Demographic Data	45
5.2.1. Evaluation Of Application Usefulness	47
5.2.2. Evaluation Of The Application Ease Of Use.....	49
5.2.3. Evaluation Of The Application Satisfaction	51
5.3. Summary	53
CHAPTER SIX: CONCLUSION OF THE STUDY.....	54
6.0. Introduction	54
6.1. Project Conclusion	54
6.2. Study Contributions	55
6.3. Limitation Of The Study	55
6.4. Future Works	56
REFERENCES.....	57
APPENDIX A	60
APPENDIX B	61
APPENDIX C	71

LIST OF FIGURES

Figure 2.1: Basic Call Flow	9
Figure 2.2: Basic Call Flow	12
Figure 2.3: Connectivity Options	18
Figure 3.1: The research methodology	22
Figure 4.1: Proposed UUM Network Architecture For VoIP Implementation...	27
Figure 4.2: Call flow between two users at UUM campus	29
Figure 4.3: Menu select Screen	31
Figure 4.4: Build Menu select	32
Figure 4.5: Finishing the installation	33
Figure 4.6: Dial Plan Show	35
Figure 4.7: Applications Testing	38
Figure 4.8: ZoIPer Interface	40
Figure 4.9: Option ZoIPer window	41
Figure 4.10: Add SIP account	41
Figure 4.11: Field of account	42
Figure 5.1: The distribution of the respondents' age as a pie chart	46
Figure 5.2: The distribution of the respondents' degree as a pie chart	46
Figure 5.3: Bar chart of the question four of the usefulness	48
Figure 5.4: Bar chart of the question one of the application ease of use	50
Figure 5.5: Bar chart of the question one of the application satisfaction	52

LIST OF TABLES

Table 5. 1: Distribution of demographic data	45
Table 5.2: Descriptive statistics of application usefulness for all items	47
Table 5.3: Distribution of question three in terms of application's usefulness	48
Table 5.4: Descriptive statistics related to the application's ease of use	49
Table 5.5: Distribution of question one in terms of application's ease of use	50
Table 5.6 : Descriptive statistics related to the application's satisfaction	51
Table 5.7: Distribution of question one in terms of application's satisfaction	52

LIST OF APPREVIATION

VoIP	Voice over IP
LAN	Local Area Network
WLA	Wireless Local Area Network
PSTN	Public Switched Telephone Network
IP	Internet Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
RTP	Real Time Transport Protocol
RTCP	Real Time Transport Control Protocol
IP PBXs	IP Private Branch Exchange
ACD	Automatic Call Distribution
IVR	Interactive Voice Response
SIP	Session Initiation Protocol
PBX	Private Branch Exchange
MGCP	Media Gateway Control Protocol
ITU	International Telecommunication Union

CHAPTER ONE

INTRODUCTION

1.0. Background

Nowadays, in light of current developments in the field of communication networks and due to the urgent need to send data in the shortest time and the lowest possible cost, the VoIP technology has been found. According to (Zhang, Hillenbrand, Müller, 2005), VoIP is a technology to transport voice communication over IP network such as Internet. This technology provides the capability of making phone calls over the packet switched networks instead of traditional circuit switched networks. With the development of the Internet, VoIP over a Wireless Local Area Network (WLAN) is expected to become an important Internet application. In addition, this technology is becoming a popular service on the Internet platform and many VoIP protocols have been proposed since it was first developed, (Wang, Liew & Fellow, 2005).

VoIP can greatly reduce the telephone call costs comparing with the traditional Public Switched Telephone Network (PSTN) system, due to the low cost features of the Internet usage, so it is expected that the VoIP may completely replace the circuit switched PSTN system in the future. Therefore, VoIP is attracting more users from the traditional telephone communication area, and more companies try to invest in the development and usage of the VoIP systems (Zhang, Hillenbrand, Müller, 2005). Moreover, VoIP is one of the fastest growing Internet applications at this time. It has two fundamental advantages compared with voice over traditional telephone networks. First, by using advanced voice-compression techniques and bandwidth

The contents of
the thesis is for
internal user
only

REFERENCES

Ahson, S.& Ilyas, M. (2009). VoIP Handbook Applications, Technologies, Reliability, and Security. United State of America: CRC Press printed

Baset, S. A. & H. Schulzrinne (2004). An analysis of the skype peer-to-peer internet telephony protocol. USA: Arxiv preprint .

Desantis, M. (2008). Understanding Voice over Internet Protocol (VoIP). [Electronic version]. US-CERT, 1-5.

Gillispie, P. J. (2005). VoIP and IP Telephony: Planning for Convergence in State Government. USA: NASCIO.

Gokhale, S.S., & Jijun, Lu. (2005) *Signaling performance of SIP based VoIP: a measurement-based approach*. Paper presented at Global Telecommunications Conference, New York: Department of Computer Sciences & Engineering.

Hallock, J. (2004). "A Brief History of VoIP." Evolution.

Hoffer, J., George, J., & Valacich, J. (2002). *Modern Systems Analysis and Design*. New Jersey, USA: Prentice Booking.

Ivar, J., Magnus, C., Patrik, J., & Gunnar, O. (1993). *Object-Oriented Software Engineering*. England: Addison Wesley.

Kelly, T. (2005). VoIP For Dummies. Indiana: Wiley Publishing, Inc.

Kojima, J., Ôshita, M., Kaneko, K., & Maruyama, T. (2000). Corporate Network System Using VoIP Technology. [Electronic version]. Hitachi, 189-193.

Liu, Z. (2002). *Object-Oriented Software Development with UML*. Macau, China: International Institute for Software Technology.

Nokia. (2004). *Advantages of SIP for VoIP*. Canada: Nokia.

Patrick, P. (2009). *Voice Over Ip Security*. USA: Cisco Press.

Rosenberg, J., J. Weinberger, et al. (2003). STUN-Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs), RFC 3489, March, from <http://www.ietf.org/rfc/rfc3489.txt>.

Rosenberg, J., H. Schulzrinne, et al. (2002). "SIP: Session Initiation Protocol (RFC 3261)." Internet Engineering Task Force, from, <http://www.ietf.org/rfc/rfc3489.txt>.

Schulzrinne, H., S. Casner, et al. (1996). RTP: A transport protocol for real-time applications, rfc 1889, January.

Stevens, W. R. and G. R. Wright (1994). *TCP/IP Illustrated*. USA: Addison-wesley.

Vaishnavi, V., & Kuechler, W. (2004). Design Research in Information Systems. Retrieved 3, 18, 2011, from: <http://desrist.org/design-research-in-information-systems/>

Van Megelen, J., L. Madsen, et al. (2007). Asterisk: the future of telephony. USA: O'Reilly Media.

Wang, Liew., & Fellow, V. (2005). Solutions to performance problems in VoIP over a 802.11 wireless LAN. *IEEE Transactions on Vehicular Technology*, 54(1), 366-384.

Zhang, G., Hillenbrand, M., & Müller, P. (2005) Facilitating the Interoperability among Different VoIP Protocols with VoIP Web Services. In Proceedings of the First International Conference on Distributed Frameworks for Multimedia Applications, (DFMA'05). Kaiserslautern, Germany: IEEE Computer Society. 39 – 44.