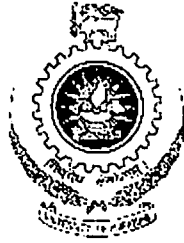


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**IMPLEMENTATION OF A FULL-FEATURED IP  
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**Submitted in partial fulfillment for the degree of Masters of  
Engineering in Electronics and Telecommunication**

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The work presented in this dissertation has not been submitted  
for the fulfilment of any other degree



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## **DEDICATION**

I dedicate this thesis to my Central collage Kekirawa , University of Peradeniya and University of Moratuwa for the guidance given to me at all times to achieve my goals and targets and providing me with the post graduate course that I received.

It is with reverence and respect that I remember my parent and teachers at this junction of my life for giving me the best of energy and enabling me to get the best possible education.



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# CONTENTS

Abstract	i
List of Figures	ii
List of Tables	iv
List of Appendixes	v
List of Abbreviations used	vi
<b>1 INTRODUCTION</b>	<b>1</b>
1.1 Vox and H.323	1
1.2 H.323 Overview	2
1.3 H.323 Architectural overview	2
1.3.1 Terminal	3
1.3.2 Gateway	3
1.3.3 Gatekeeper	4
1.3.4 Multipoint Control Unit (MCU)	4
1.4 Importance of standardise IP telephony system	4
1.4.1 Cost and the technical competitiveness of the system	4
1.4.2 Compatibility with the multi-vendor hardware	6
1.4.3 Replacing and extending existing service	6
1.4.4 Provision for future development	6
1.5 System Implementation	6
1.5.1 Software selection	8
1.5.2 Hardware selection	8
1.6 Organization of this thesis	8
<b>2 H.323 STANDARDS, PROTOCOLS, AND ELEMENTS</b>	<b>10</b>
2.1 H.323 standards	10
2.1.1 Call signalling and control	10
2.1.2 Audio/Video/Fax codecs	10
2.1.3 H.323 related standards	11
2.2 Protocols	12
2.2.1 Transport protocols	12
2.2.2 Signalling protocols	14
2.2.3 Media related protocols	16
2.2.4 Security related protocols	16
2.2.5 Supplementary services related protocols	16
2.2.6 Protocols related to real time data transmission	17
2.3 Order of protocols used in call processing	18
2.4 Elements and their functions	19
2.4.1 Terminal	19
2.4.2 Gateways	20
2.4.3 Gatekeeper	21
2.4.4 Multiple Control Unit	25

<b>3</b>	<b>SIGNALLING AND SIGNALLING CHANNELS</b>	<b>26</b>
3.1	RAS (Registration Admission and Status)	26
	3.1.1 Gatekeeper discovery	28
	3.1.2 End-point registration	29
	3.1.3 End-point location	32
	3.1.4 Admission, bandwidth changes, status, and disengage	32
	3.1.5 Access tokens	33
	3.1.6 Alternate gatekeeper procedures	33
	3.1.7 Usage information reports	33
	3.1.8 call credit related capabilities	33
3.2	Call signalling	34
	3.2.1 Call setup	34
	3.2.2 Call clearing	38
3.3	Call control signalling	41
3.4	Signalling channel and routing	44
	3.4.1 Call signalling channel routing	45
	3.4.2 Control channel routing	47
<b>4</b>	<b>SYSTEM OPERATION AND IMPLEMENTATION ARCHITECTURE</b>	<b>50</b>
4.1	Deployment architecture and operation	50
	4.1.1 System architecture	50
	4.1.2 System operation	51
4.2	Implementation requirements	55
	4.2.1 Gatekeeper	55
	4.2.2 Gateways	56
	4.2.3 Terminals	57
<b>5</b>	<b>SYSTEM INSTALLATION</b>	<b>58</b>
5.1	Basic libraries and Gateway card installation	58
	5.1.1 Pwlib	58
	5.1.2 Openh323	58
	5.1.3 Quicknet card drivers	58
5.2	Application software installation	58
	5.2.1 Gatekeeper	58
	5.2.2 Gateways	60
	5.2.3 Modification of FXO Gateway	60
	5.2.4 Terminals	61
<b>6</b>	<b>QUALITY OF SERVICE MANAGEMENT</b>	<b>63</b>
6.1	Latency	63
	6.1.1 Cause of latency	64
	6.1.2 Managing latency	72
6.2	Jitter	73
6.3	Echo	74



<b>7</b>	<b>FUTURE WORK</b>	<b>75</b>
7.1	Supplementary services	75
7.2	Interfacing the system with modern voice networks	75
	<b>REFERENCES</b>	<b>77</b>



## ABSTRACT

Traditionally, voice traffic has been carried on circuit-switched networks. However, in recent years great interest has been generated in carrying voice over a variety of nontraditional packet-switched networks. These techniques are known generally as Voice over Packet (VoP). The most popular implementation of VoP has been VoIP on the IP-based Internet. Other VoP technologies comprise those such as voice over Frame Relay (VoFR), Voice Over ATM (VoATM) ,and voice over Digital Subscriber Loop (VoDSL). IP is the dominating technology in end user level deployments, carrying millions of minutes of voice traffic today.

This thesis describes the implementation of an Internet Telephony (Voice over IP, VoIP) system within the University's voice and data networks.

The deployment of VoIP can reduce costs by combining all types of traffic onto a single network infrastructure, eliminating the need to maintain and pay for several different services. The University has its internal telephone network implemented via several PABXs, and its computer network consisting of several departmental networks. Implementation of the VoIP system enables the integration of the two, and through that, the extension of telephone facilities to a larger group of people, as well as the development of value-added services. ITU-T Recommendation H.323 is the most widely used standard facilitating VoIP.

Further this details the hardware and the software aspects of the designed H.323-based VoIP system, and their integration for implementation in the university-wide network.



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## LIST OF FIGURES

Figure 1-1	H.323 Architectural overview	3
Figure 1-2	Portion of deployed system	7
Figure 2-1	Protocols related to OSI model	11
Figure 2-2	Format of RTP header field	13
Figure 2-3	RTCP structure	14
Figure 2-4	Q.931 header structure	15
Figure 2-5	Basic component of H.323 terminal equipment	19
Figure 2-6	H.323 Gateway architecture	21
Figure 2-7	Presence of Gatekeeper in H.323 zone	21
Figure 2-8	MCU architecture	25
Figure 3-1	Gatekeeper auto discovery	29
Figure 3-2	End-point registration	31
Figure 3-3	Basic call setup no gatekeeper	36
Figure 3-4	Both end-points, registered same gatekeeper-direct call signalling	37
Figure 3-5	Both end-points registered same gatekeeper and route call signalling	38
Figure 3-6	End-point initiated call clearing	40
Figure 3-7	Gatekeeper initiated call clearing	41
Figure 3-8	Gatekeeper routed call signalling	46
Figure 3-9	Direct end-point call signalling	47
Figure 3-10	Direct H.245 control channel connection between end-points	48
Figure 3-11	Gatekeeper routed Control channel connection between end-points	49
Figure 4-1	Basic system architecture	51
Figure 4-2	Registration and admission	52
Figure 4-3	Accounting system for pre-determined call duration	53
Figure 4-4	Call processing	54
Figure 5-1	Netmeeting configuration	61
Figure 6-1	Quality perception vs latency	63
Figure 6-2	Part of deployed IP telephony system	64
Figure 6-3	Gateway processing	65
Figure 6-4	DSP voice compression sub system	66
Figure 6-5	Framing process	66

Figure 6-6	Anatomy of an IP telephony packet	70
Figure 7-1	Resource reservation for point to point connection	76



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## LIST OF TABLES

Table 1-1	Hardware and software cost comparison	5
Table 2-1	Description of signalling standards	10
Table 2-2	Description of codec standards	10
Table 2-3	Description of H.323 standards	11
Table 2-4	Description of RTP fields	12
Table 2-5	Description of RTCP fields	13
Table 2-6	Description of Q.931 fields	15
Table 2-7	Order of protocols and intended functions in call processing	18
Table 3-1	Description of basic RAS messages	27
Table 3-2	Description of RAS Q.931/H.225.0 message format	34
Table 3-3	H.323 message formats	42
Table 6-1	Voice code frame sizes	67
Table 6-2	Compression method and MOS score	68
Table 6-3	Sample latency budget	73



## LIST OF APPENDIXES

Appendix I	Installation steps and command	I
Appendix II	Configuration files, commands and parameter references	III
Appendix III	Boot scripts	XXX
Appendix IV	Scripts for supplementary services	XXXIV
Appendix V	Code modifications	XLIX



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## LIST OF ABBREVIATIONS

ACF	Admission Confirmation
APE	Application Protocol Entity
ARJ	Admission Reject
ARQ	Admission Request
BCF	Bandwidth Change Confirmation
B-ISDN	Broadband Integrated Services Digital Network
BRJ	Bandwidth Change Reject
BRQ	Bandwidth Change Request
CAS	Channel Associated Signalling
CED	Called Terminal Identification Tone
CID	Conference Identifier
CIF	Common Intermediate Format
CNG	Calling Tone
DBR	Deterministic Bit Rate
DCF	Disengage Confirmation
DNS	Domain Name System
DRQ	Disengage Request
DSVD	Digital Simultaneous Voice and Data
DTMF	Dual-Tone Multi Frequency
FAS	Facility Associated Signalling
GCC	Generic Conference Control
GCF	Gatekeeper Confirmation
GID	Global Call Identifier
GIT	Generic Identifier Transport
GK	Gatekeeper
GQOS	Guaranteed Quality of Service
GRJ	Gatekeeper Reject
GRQ	Gatekeeper Request
GSTN	General Switched Telephone Network
GW	Gateway
HDLC	High Level Data Link Control
HTTP	Hypertext Transfer Protocol
ID	Identifier
IP	Internet Protocol
IRQ	Information Request
IRR	Information Request Response
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
LAN	Local Area Network
LCF	Location Confirmation
LRJ	Location Reject
LRQ	Location Request
MC	Multipoint Controller

MCS	Multipoint Communications System
MCU	Multipoint Control Unit
MG	Media Gateway
MGC	Media Gateway Controller
MP	Multipoint Processor
MTU	Maximum Transmission Unit
N-ISDN	Narrow-band Integrated Services Digital Network
NACK	Negative Acknowledge
NFAS	Non-facility Associated Signalling
NNI	Network-to-Network Interface
NSAP	Network Layer Service Access Point
OLC	H.245 OpenLogicalChannel message
PBN	Packet Based Network
PDU	Packet Data Unit
PRI	Primary Rate Interface
QCIF	Quarter CIF
QOS	Quality of Service
RAS	Registration, Admission and Status
RAST	Receive and Send Terminal
RCF	Registration Confirmation
RIP	Request in Progress
RRJ	Registration Reject
RRQ	Registration Requestb
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SCN	Switched Circuit Network
SCR	Service Control Response
SSRC	Synchronization Source Identifier
TCP	Transport Control Protocol
TGW	Trunking Gateway
TSAP	Transport layer Service Access Point
UCF	Unregister Confirmation
UDP	User Datagram Protocol
UNI	User-to-Network Interface
URJ	Unregister Reject
URQ	Unregister Request