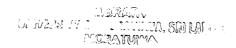


UNIVERSITY OF MORATUWA SRI LANKA

IMPLEMENTATION OF A FULL-FEATURED IP TELEPHONY SYSTEM





Submitted in partial fulfillment for the degree of Masters of Engineering in Electronics and Telecommunication

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UM Thesis coll

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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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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 ABBREVIATIONS

ACF Admission Confirmation
APE Application Protocol Entity

ARJ Admission Reject ARO 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
DRO 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

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Standardization Sector

LAN Local Area Network
LCF Location Confirmation

LRJ Location Reject
LRQ Location Request
MC Multipoint Controller

MCS Multipoint Communications System

Multipoint Control Unit MCU

MG Media Gateway

MGC Media Gateway Controller **Multipoint Processor** MP

Maximum Transmission Unit **MTU**

N-ISDN Narrow-band Integrated Services Digital Network

NACK Negative Acknowledge

Non-facility Associated Signalling **NFAS** Network-to-Network Interface NNI

NSAP Network Layer Service Access Point H.245 OpenLogicalChannel message OLC

Packet Based Network PBN Packet Data Unit PDU PRI Primary Rate Interface

Quarter CIF QCIF

QOS Quality of Service

RAS Registration, Admission and Status

Receive and Send Terminal **RAST** RCF **Registration Confirmation**

RIP Request in Progress RRJ Registration Reject RRQ Registration Requestb

Real Time Control Protocol **RTCP**

Real Time Protocol University of Moratuwa, Sri Lanka. RTP Switched Circuit Network

SCN

Service Control Response SCR

SSRC Synchronization Source Identifier

TCP **Transport Control Protocol**

TGW Trunking Gateway

Transport layer Service Access Point **TSAP**

UCF Unregister Confirmation User Datagram Protocol UDP UNI User-to-Network Interface

URJ Unregister Reject Unregister Request **URQ**