Web Call SDK

A Dissertation submitted to the Royal Institute of Technology (KTH) in partial fulfillment of the requirements for the degree of Master of Science

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Abstract

This is a skeleton for KTH theses. More documentation regarding the KTH thesis class file can be found in the package documentation.

To my parents, LI Chongzhi and WU Wei, who have guided me through life and encouraged me to follow my own path, and to my wife, MEI Dan, for being waiting for me and keeping faith in me.

Contents

Acknowledgements

... acknowledgement text ...

Part I

Theory

Chapter 1

Introduction

1.1 Background

IP telephone (VoIP) technology enables the communications between Internet users and the endpoints in PSTN circuit switched (CS) networks. As we know, Session Initiation Protocol (SIP) is widely adopted as a signaling protocol for VoIP communications. Because of its simplicity, power and extensibility, it has also been selected by the Third Generation Partnership Project (3GPP) as a major component of IP Multimedia Subsystem (IMS) for the evolved UMTS core network. Based on IMS/SIP network technology, Web Call Example Application integrated call functions into Web containers. This presents a simple way to implement the communication convergence of Web, IMS/SIP network, and CS networks. It does not require the installation of the plug-in clients on the browser or other special client software as most VoIP services. Compare to other customer-oriented VoIP products, Web Call Example Application offers solutions for developers. It integrated the SIP call function into Web container as Java EE components, providing the example of the convergences between SIP/IMS and the Web with existing Internet Web technology on Java EE Web container.

1.2 Problem

Chapter 2

Requirement

... some text ...

Some reference [1] RFC3261 [2]

Chapter 3

Study

Chapter 4

Solution

Part II

Implementation

Chapter 5

Web Call Architecture

Chapter 6

SIP Call Component

Chapter 7

Web Application

Chapter 8

Web Service Interface

Chapter 9

Web Service Client

Chapter 10

Conclusion

Bibliography

[1] S. Deorowicz and A. Skórczyński. *LEd documentation.* 2004. [cited at p. 9]

[2] Schulzrinne H. Camarillo G. Johnston A. Peterson J. Sparks R. Handley M. Rosenberg, J. and E. Schooler. SIP: Session initiation protocol. June 2002. [cited at p. 9]

Appendices

Appendix A

Appendix title

... some text ...

List of Symbols and Abbreviations

Abbreviation	Description	Definition
3GPP	3rd Generation Partnership Project	page ??
3pcc	Third Party Call Control	page ??
CS	Circuit Switched	page ??
GSM	Global System for Mobile communications	page ??
IMS	IP Multimedia Subsystem	page ??
IP	Internet Protocol	page ??
ISUP	ISDN User Part	page ??
JDK	Java Development Kit	page ??
JRE	Java Runtime Environment	page ??
JVM	Java Virtual Machine	page ??
PSTN	Public Switched Telephone Network	page ??
SDP	Session Description Protocol	page ??
SIP	Session Initiation Protocol	page ??
VoIP	Voice over Internet Protocol	page ??

List of Figures

List of Tables

Index