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# Unified Communication and WebRTC

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**Title:** Unified Communication and WebRTC  
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**Problem description:**

Web Real-Time Communication (WebRTC) offers application developers the ability to write rich, real-time multimedia application (e.g. video chat) on the web, without requiring any plugins, downloads or installations. WebRTC is also currently the only existing soon-to-be standardized technology on the market to create horizontal cross-platform communication services, encompassing smartphones, tablets, PCs, laptops and TVs, which adds value for both consumers and enterprises. WebRTC gives operators the opportunity to offer telephony services to more devices, such as PCs, tablets and TVs. This thesis considers how WebRTC can enhance the existing echo-systems for telephony and messaging services by providing the end-user rich application client.

It will also covers research about different solutions to implement WebRTC to cooperate with existing telephony services like hosted virtual Private Branch Exchange (PBX) services.

A prototype of WebRTC deployment based on different rich communication scenarios will be implemented along with this thesis. Some corresponding test and evaluation will be fulfilled in this prototype.

Research about advanced WebRTC usability in telephony and messaging services will be covered in this thesis by the feedback of the WebRTC prototype

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## Abstract

For the development of traditional telephony echo-systems, the cost of maintenance traditional telephony network is getting higher and higher but the number of customer does not grow rapidly any more since almost every one has a phone to access the traditional telephony network. WebRTC is an Application Programming Interface (API) definition drafted by the World Wide Web Consortium (W3C) that supports browser-to-browser applications for voice calling, video chat, and Peer-To-Peer (P2P) file sharing without plugins.[Wik14] “This technology, along with other advances in HyperText Markup Language 5 (HTML5) browsers, has the potential to revolutionize the way we all communicate, in both personal and business spheres.”[ABJ13]

As network operators aspect, WebRTC provides many opportunities to the future telecommunication business module. For the users already have mobile service, operator can offer WebRTC service with session-based charging to the existing service plans. Messaging APIs can augment WebRTC web application with Rich Communication Services (RCS) and other messaging services developers already know and implement. Furthermore, since WebRTC is a web based API, then the implementation of Quality of Service (QoS) for WebRTC can provide assurance to users and priority services (enterprise, emergency, law enforcement, eHealth) that a WebRTC service will work as well as they need it to. WebRTC almost provide network operator a complete new business market with a huge amount of end-users.

As an end-user aspect, WebRTC provides a much simpler way to have real-time conversation with another end-user. It is based on browser and internet which almost personal or enterprise computer already have, without any installation and plugins, end-user can have exactly the same service which previous stand-alone desktop client provides. By the system this thesis will cover, the end-user even can have the real-time rich communication service with multiple kinds of end-users.

This thesis will cover the research about how to apply WebRTC technology with existing legacy Voice over Internet Protocol (VoIP) network.

**Keywords :** WebRTC, AngularJs, Nodejs, SIP, WebSocket, Dialogic XMS

## Preface

WebRTC is quite popular topic in the web development field since the massive usage and development of HTML5 web application on the internet. The initial purpose of this web API is to provide the browser client the ability to create real-time conversation between each other. After many WebRTC based application come out the market, it is quite normal to think about how to integrate these kind of web application with the current legacy telephony network as the next big step for this technology. The requirement of this process is not only from the traditional telephony operator but also the normal end-users. The approach to achieve this goal is the main purpose of this thesis.

Research about current WebRTC technology usage and development of a WebRTC prototype system are the two main parts of this thesis. The prototype system is implemented by regarding to the research of WebRTC integrated with legacy telephony network.

Current status of WebRTC technology, WebRTC business use cases, analysis of different possible WebRTC implement solutions and WebRTC system architecture will be covered in this thesis. Some research regarding with the development of WebRTC prototype system will be covered in this thesis as well.

The prototype described in this thesis is implemented to cooperate with existing legacy VoIP network services through Session Initiation Protocol (SIP) server and PBX service. It will provide most of essential functions which are included in the legacy telephony business, besides other communication functions used on web. Moreover, some analysis and discussion about the feedback of the prototype will be covered in this thesis.

The prototype will be implemented in programming language Javascript for both client front-end and server back-end by using the AngularJs framework and Nodejs framework mainly. The approach and reason to choose these framework and programming language will be expounded in the later chapter in this thesis.

## Acknowledgment

# Contents

List of Figures	v
List of Tables	vii
List of Algorithms	ix
List of Acronyms	xiii
1 Introduction	1
References	3



# List of Figures



# List of Tables



# List of Algorithms









# List of Acronyms

**API** Application Programming Interface.

**HTML5** HyperText Markup Language 5.

**P2P** Peer-To-Peer.

**PBX** Private Branch Exchange.

**QoS** Quality of Service.

**RCS** Rich Communication Services.

**SIP** Session Initiation Protocol.

**VoIP** Voice over Internet Protocol.

**W3C** World Wide Web Consortium.

**WebRTC** Web Real-Time Communication.



# Chapter 1

## Introduction



# References

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