Analy~~ze~~sis and Implementation of third party call control in Web Call SDK

Shanbo Li[1]

KTH - Royal Institute of Technology

SE-100 44 Stockholm

Sweden

shanbo@kth.se

Abstract

Session initiation Protocol (SIP) is widely used in VoIP commutation. Web Call SDK is a project from the Developer Program of Ericsson is Mobility World. It integrates call functions into Web containers. This paper ~~firstly~~ illustrates the current problem of the Web Call SDK. Then it shows how the Third Party Call Control (3PCC) mechanism is used in the new solution.

fist

use a serif font

justify paragraph

Categories and Subject Descriptors

J.m [Computer Applications]: Miscellaneous

General Terms

Performance, Design, Experimentation

Keywords

Session initiation Protocol, Voice over IP, Third Party Call Control

[1] Shanbo Li is a Java Developer in Ericsson Developer Program, Mobility World. He is in charge of Web Call SDK project.

the

via gateways

use a serif font

1 Introduction

1.1 Background

IP telephone (VoIP) technology enables the communications between Internet users and ~~the~~ endpoints in the PSTN/circuit switched (CS) networks. As we know, the 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 the IP Multimedia Subsystem (IMS) for the evolved UMTS core network.

Web Call SDK ~~from~~ Ericsson's includes a high-level API that can be used to implement SIP call control. Based on IMS/SIP network technology, Web Call SDK integrates call functions into Web containers. This presents a simple way to implement ~~the~~ communication convergence of Web, IMS/SIP networks and CS networks. It does not require the installation of the plug-in clients on the browser or other special client software as do most VoIP services.

spell out

to be used by

Compared to other customer-oriented VoIP products, Web Call SDK offers solutions ~~for~~ developers. It integrates the SIP call function into Web container as Java EE components, providing ~~the~~ example ~~of the~~ convergence between SIP/IMS and the Web with existing Internet Web technology on Java EE Web container.

for

Are they different or the same?

Are these the same?

1.2 Overview

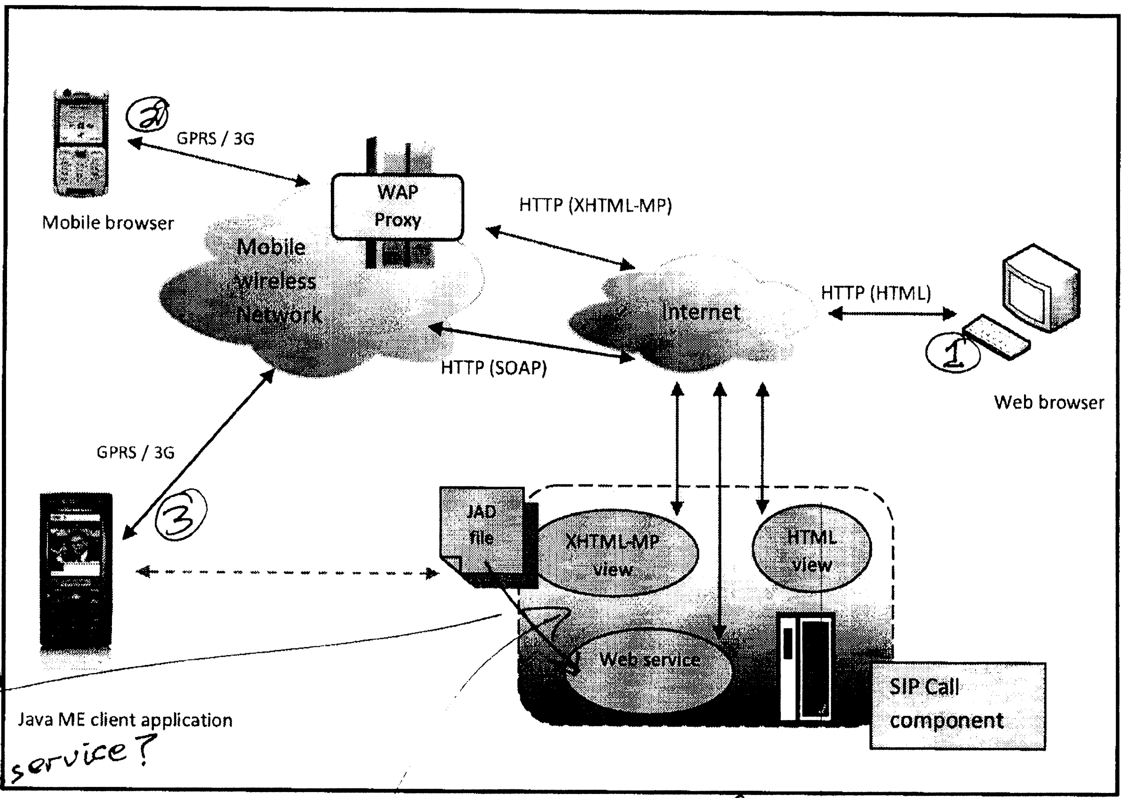
The Web Call Kit is a software development kit (SDK) consisting of a SIP Control Component Library, an Integrated Web Applications and a Java ME - Web Service Client application. The purpose of the Web Call Controller Kit is to provide a high level API to implement SIP call control, and demonstrate how to integrate a SIP call component in a web container such as Java Beans and web services. It also includes a Java ME application example to access a web service which makes phone-to-phone SIP calls. This SDK can help Java application developers ~~to~~ understand how the SIP call component library works and how Java EE and Java ME applications can be developed upon it. This SDK can be used for connecting to a SIP based network such as the IMS core network.

1.2.1 Network ~~Overview~~

Figure 1(a) ~~Network overview~~ shows the general network architecture for a service provider with the integrated web application. The application runs on a Java EE web container and resides outside the operator's mobile network.

the

Highlight this in the figure



Isn't this JAD file from the web service?

Figure 1 Network overview

where exactly is this?

In the front-end of the application, there are three types of access.

1. Web users can use web browsers to visit the web site via ~~the~~ their ISP on the internet. The web application provides HTML version for web users.

2. Mobile users can use WAP browsers to visit the web site via their mobile operator's network. The web application provides XHTML-MP versions for mobile users having either bigger or smaller screens.

3. A ~~More~~ more advanced mobile user, with JSR 172 support, can download and install the Java ME client application, and access the web services via their mobile operator's network.

The interaction between the second form of access and the third one is that: the integrated web application dynamically generates the Java ME client application profile after the WAP ~~users~~ user logs in.

How do the WAP & 3G Jave ME relate? Does the Java ME user have to go via WAP? Why?

2 Problem

2.1 Current Web Call SDK

Currently, Web Call SDK manually setups and manages sessions, and meanwhile, act as media communication proxy.

The session control module handles the SIP session initiation and the coordination between sessions. the SIP control component only implemente~~d~~s the phone-to-phone SIP call function. Thus, the primary primitive operation of ~~SIP~~ TP control component is establish~~ment of~~ing a session between endpoints A and B. Establishment of this session is orchestrated by a third party, the SIP controller.

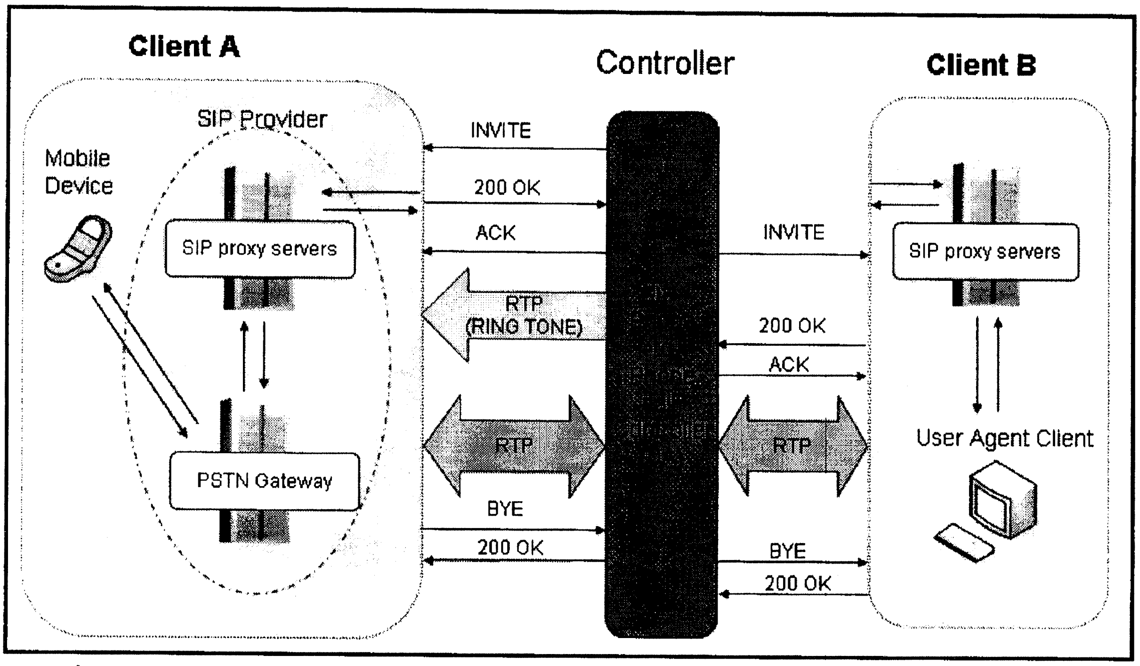


Figure 2: Use case of Web Call SDK (Relay Call)

The basic case flow is shown in Figure 2. The controller first sends an INVITE to A. A's phone rings, and A answers. This results in a 200 OK that is sent to session initiator, controller. The controller needs to send the ACK. This part is the typical SIP session setup steps.

After the session between the controller and client A has been established, the controller starts to send ring tone to client A, indicating that it is waiting for B to answer the phone.

Once the controller receives a 200 OK from client A, it start another session initiation with client B by sending an INVITE to B.

What causes the controller to send the INVITE to A?

traffic

proxying

Can you quantify this?

4x RTP packets per session

After B answers the phone, controller receives 200 OK from client B. It will stop sending ring tone to A, and start RTP packets ~~proxy~~ between A and B.

2.2 Problems of current solution

it does in an

2.2.1 The load ~~of~~ controller

The controller here acts ~~as a~~ back to back user agent (B2BUA). ~~It~~ receives the RTP flow from one client and transfers to another client. It ~~is~~ the same ~~as~~ the opposite direction. That means all of the RTP will go through the controller. So the load ~~of~~ controller will be~~come much~~ heavier ~~with~~ the number increasing of concurrent users. A powerful host is needed to handle the RTP flows. ~~But~~ the design goal of Web Call SDK ~~is just~~ a tool kit that can easily integrate ~~to~~ a web site. ~~And~~ current mechanism of session control will decrease the performance of whole web site.

as

into

However,

Unfortunately, this

was simply

2.2.2 The latency of phone call

As can be seen from the session flow, the RTP goes from one client to controller via a SIP provider and then the controller transfer the RTP to another client via the SIP provider again. ~~It~~ means that ~~one~~ direction of RTP will go through SIP provider twice. So the latency of phone call via SIP provider will be double of normal calls. Furthermore, the performance of single line transfer will decrease as mentioned in section 2.2.1. The latency of phone-to-phone call via Web Call SDK test turned to be more then 3 seconds. It is quite unacceptable.

This

that

Why so high? Note: 1.5 sec. for regular calls

Explain

2.2.3 Reliability

Lack of

Since

thus

~~All~~ of RTP flows go through the controller, ~~so~~ if the controller crashes, all of calls will be interrupted immediately.

(excessive latency)

3 Solution

A solution is needed urgently to solve current problem. The main reason ~~of~~ two problems in section 2 is the RTP flows have to go via controller. So ~~the~~ solution should find a way to avoid the RTP and establish the phone call ~~inside SIP network~~.

this

for

that

the

a

proxy

directly between A and B

3.1 Third Party Call Control

3.1.1 Introduc~~e~~ of third party call control

tion

In the traditional telephony context, third party call control allows one entity (which we call the third party call controller) to set up and manage a communications relationship between two or more other parties. Third party call control (referred to as 3pcc) is often used for operator services (where an operator creates a call that connects two participants together) and conferencing. [1]

for

3.1.2 Third party call control in Web Call SDK

the

the SIP RFC

Third party call control is possible using only the mechanisms specified within RFC 3261 ~~SIP: Session Initiation Protocol~~. [1, 2] ~~And~~ The most significant advantage of third party call control is the controller only need to handle SIP message transfer and leaves the RTP flow for ~~SIP~~ the ISP ~~service provider~~.

The message transfer simply only happens at the start ~~initial~~ of the call or at the end while of call. All the controller needs to do is ~~just~~ send several messages between the two clients. ~~As~~ the RTP flow is established directly between clients~~, on direction of communication only go through the SIP provider once~~. Just like a normal SIP call. ~~And~~ As a result the latency just depends on the service quality of ~~SIP provider~~ ISP.

Drop the background

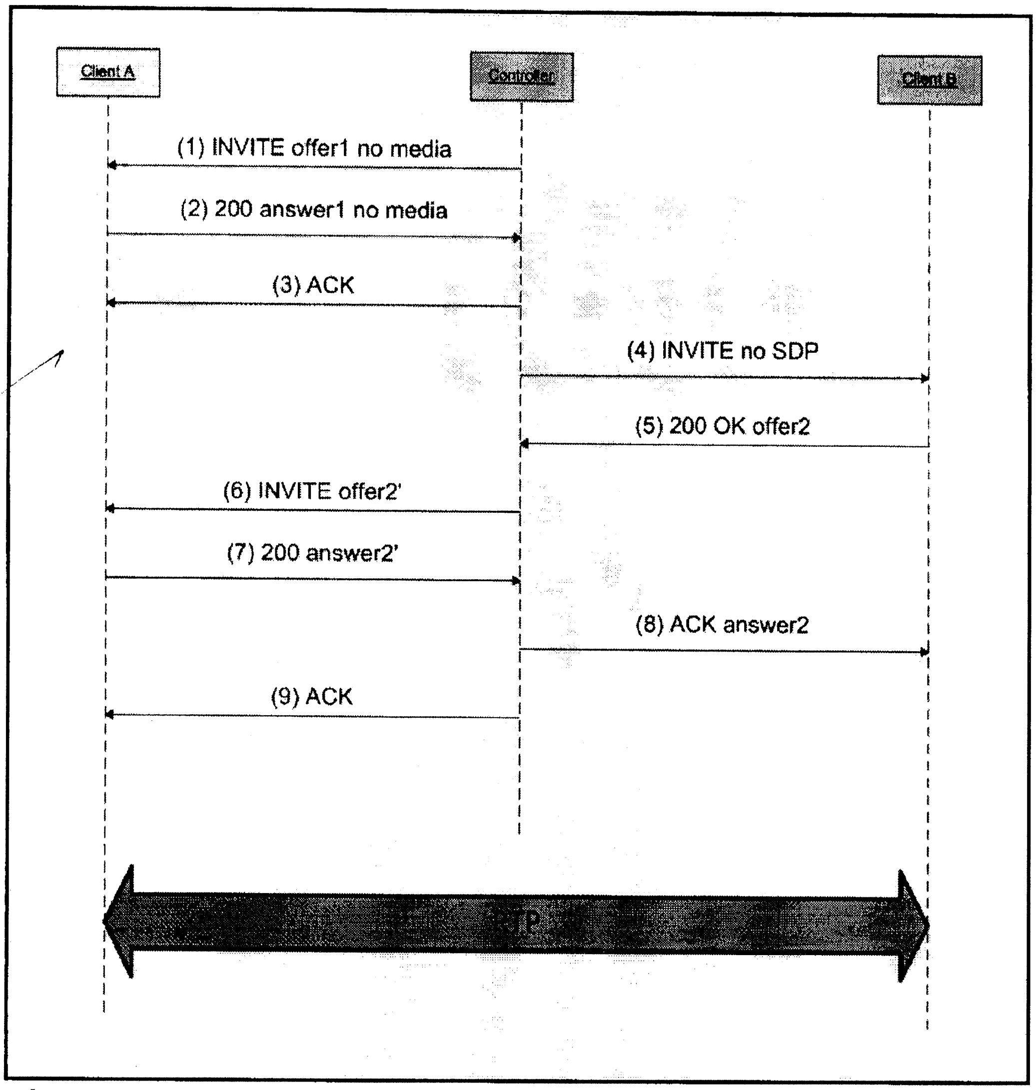


Figure 3: Flow of Third Party Call Control

Put the Figure 3 after this text

3.2 Design

3.2.1 3PCC Call Establishment

As shown in Figure 3, our 3PCC call establishment follows the Flow IV described in RFC 3725[1]. First, an INVITE(1) is send out from controller to client A. ~~Its~~ Its SDP has no media field, ~~it~~ this means the RTP will be established later by a re-INVITE. After the INVITE(1), client A will return a 200 OK with no m field in SDP. Controller will acknowledge this by an ACK(3). Then, the controller will send an ~~INVENT~~ INVITE(4) without SDP to client B. ~~And~~ Client B will answer this ~~INVENT~~(4) by sending out a 200 OK (5). After controller gets the 200 OK (5), it will construct a suitable SDP according the SDP in the 200 OK(5) and send a re-~~INVENT~~(6) to client A. Client A now can reply a 200 OK(7) with SDP and the controller will send an ACK(8) to client B according the 200 OK(7) from client. And finally, send an ACK(9) to client A. The RTP can start now directly between A and B.

3.2.2 Architecture

Figure 4 shows the new architecture of SIP Component of Web Call SDK. It uses an Abstract Factory Pattern to produce the call controller, ~~so~~ thus developers can easily switch from one kind of controller to another.

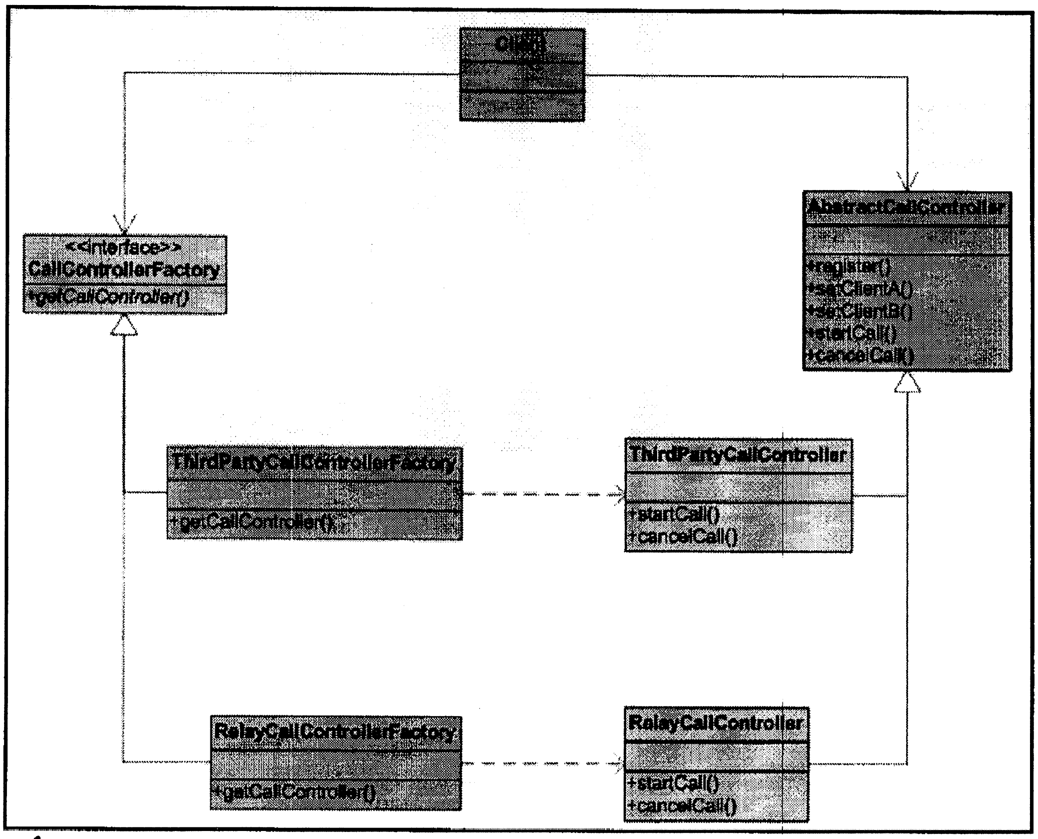


Figure 4 Architecture of SIP component of Web Call SDK

This architecture separates different ~~implements~~ of call controller so it is easy to manage and update the code. Since ThirdPartyCallController and RelayCallController both extend AbstractCallController. Some of public method can be defined in AbstractCallController, such as register(), setClientA(), and setClientB(). ~~And~~ the two ~~implements~~ can establish the SIP call session ~~with~~ their own way. The information ~~of~~ controller as well as client A and client B are pre-configured when construct ~~of~~ CallControllerFactory. So developers ~~don't~~ need to input the clients twice when ~~swich~~ from one controller to another.

Web Call SDK uses MjSip v1.6[3] toolkit as the lower level SIP stack and implemented phone-to-phone session control.

In the implementation of Third Party Call Controller, the trickiest part is the flow control. In a general ~~INVENT of~~ SIP, a Three-Way Handshake is ~~enough~~.[8] But in the 3PCC implantation of Web Call SDK, since the initial ~~of~~ session is controller and it aims to establish a phone-to-phone call between two users, ~~it needs to send~~ and handle 9 SIP messages. According to RFC3725 we customize the origin field from 200 OK offer2(9) and send new SDP within ~~INVENT~~ offer2'.

Finally the implementations; the; the; switching; about the; do not; by the; the; sufficient; INVITE; ing; this requires sending; initiated by the; INVITE

4 Evaluation

This section presents results of experiments of latency ~~of~~ SIP phone-to-phone call which established by Web Call SDK. We ran Web Call SDK both in simulation environment and live experiments. The result ~~appears~~ that the third party call significantly decrease the latency of phone-to-phone call.

for; are; the; a; a; show; approach

what about the load?

4.1 Simulation

We use Ericsson Service Development Studio (SDS) 4.0 as the SIP service provider. [4] It is the only fully comprehensive tool for development and end-to-end testing of both the client and server side of new convergent all-IP (IMS) applications.

We use two clients, Express Talk [5] and PhonerLite [6]. They are both very good SIP client. Express Talk register ~~with~~ A which means client A ~~at Ericsson SDS~~, and PhonerLite register ~~with~~ B which ~~indicates~~ client B.

s; s; s; as; means it is; as; it is; Both register with the SDS registrar.

In this simulation environment evaluation, we tried three use cases. The first one is direct client to client call ~~via Ericsson SDS~~. The second test use controller as a Back to Back User Agent (Relay Call) which is the original solution of Web Call SDK. The last is third party call control.

The test result ~~and cooperation~~ ~~can be seen from~~ Table 1. We do not have a professional test tool of duration in hand and we use a stop watch to count the latency. So the figures in table are not accuracy ~~enough~~ but it is enough to see the differences.

are shown in; very; the

why not use wireshark or other tool to capture the packets?

are and the

From the se result s, we ~~can~~ easily conclude that ~~compare with~~ the direct~~ly~~ ~~endpoints~~ call, Third party call control ~~is~~ much better than the relay call.

	Directly Phone-to-phone Call	Relay Call	Third Party Call control
Establish	1s	6s	5s
RTP Flow	Directly	Transfer at controller	Directly
Latency	Less than 1s	More than 2s	Less than 1s

Table 1: Test Result s of SIP Phone Call in Ericsson's SDS Simulation Environment.

4.2 Real IMS network

We plan to test and verify our Third Party Call Control solution ~~remotely~~ on a real Ericsson IMS core and application servers at our IMS Expert Center. Ericsson Mobility World's IMS Expert Center lab [7] in Montreal, Canada is open~~s~~ ~~for the~~ to developers all over the world and it is the best way to prove our solution.

Unfortunately, due to the full schedule of IMS Expert Center, we can only test it in the end of May, 2008

Can you quantite this?

5 Conclusions

After we implement ed the Third Party Call Control in Web Call SDK, the load on the controller server side is greatly reduced. Once the session is set up between end points, the controller only needs to manage the SIP session state. The RTP ~~transferring would~~ packets flow is then ~~be~~ peer-to-peer between the endpoints. Even if the controller crash es, the call will not be interrupted, because the~~y~~ parties are ~~using~~ directly exchanging RTP.

However, ~~since~~ the third party call control in ~~session initiation protocol~~ SIP is just a specification, ~~And~~ most ~~of~~ SIP servicers and gateways ~~don't~~ do not yet support the 3PCC specification ~~yet~~. ~~That is way we keep~~ Thus we retain the back to back user agent solution as ~~This~~ solution ~~follow the most common protocol~~ is directly supported in SIP, ~~so~~ there for the Web Call SDK can be used anywhere.

6 Future Work

There are some tasks and features in Web Call SDK that are left for future development because of time limitations.

concerning most

This is a list of future tasks ~~of~~ third party call control and ~~much~~ of them will follow the instructions of RFC3725.

Error Handling

does not consider any

The

While establish the ~~call~~, errors may happen both from client A or client B. Current solution ~~haven't consider the~~ errors. So it may lead ~~an~~ establishment failed, but without correct error message. After implement the error handling, ~~one~~ can easily found where is the exact place which error accord as well as why there is error.

call the

the

Is finding it important or only a behaving properly?

Cancel Call

ed not a

Due to the limit~~ation of~~ time, we hav~~en't~~ implemented ~~the~~ call cancel or test it. This will not be hard, because the controller can ~~just~~ send a simple BYE to both ~~side of~~ clients.

Is it so simple? simply ed

7 Reference:

[1]. J. Rosenberg, J. Peterson, H. Schulzrinne,G. Camarillo. *RFC 3725 Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP).* April 2004.

[2]. J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler. *RFC 3261 SIP: Session Initiation Protocol.* June 2002.

[3]. CreaLab. MjSIP. http://www.mjsip.org/

should have date or date of last access

[4]. Ericsson. Ericsson Service Development Studio 4.0 . http://www.ericsson.com/mobilityworld/sub/open/technologies/ims_poc/tools/sds_40

[5]. NCH Software. Express Talk. http://www.nch.com.au/talk/

[6]. Heiko Sommerfeldt. PhonerLite. http://www.phonerlite.de/

[7]. Ericsson. Ericsson Mobility World's IMS Expert Center lab. http://www.ericsson.com/mobilityworld/sub/open/technologies/ims_poc/testarea.html

[8]. Gonzalo Camarillo. SIP Demystified. McGraw-Hill. 2001