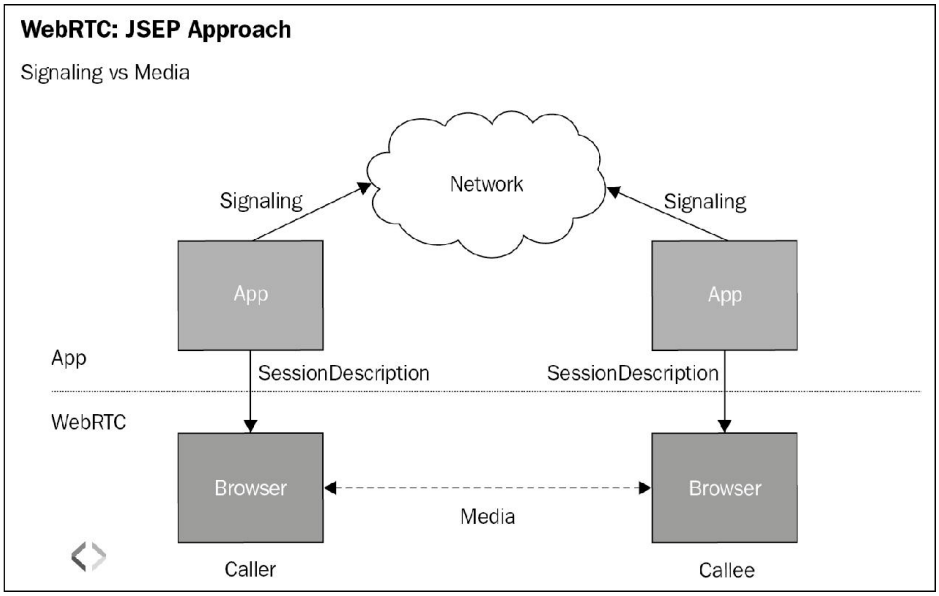
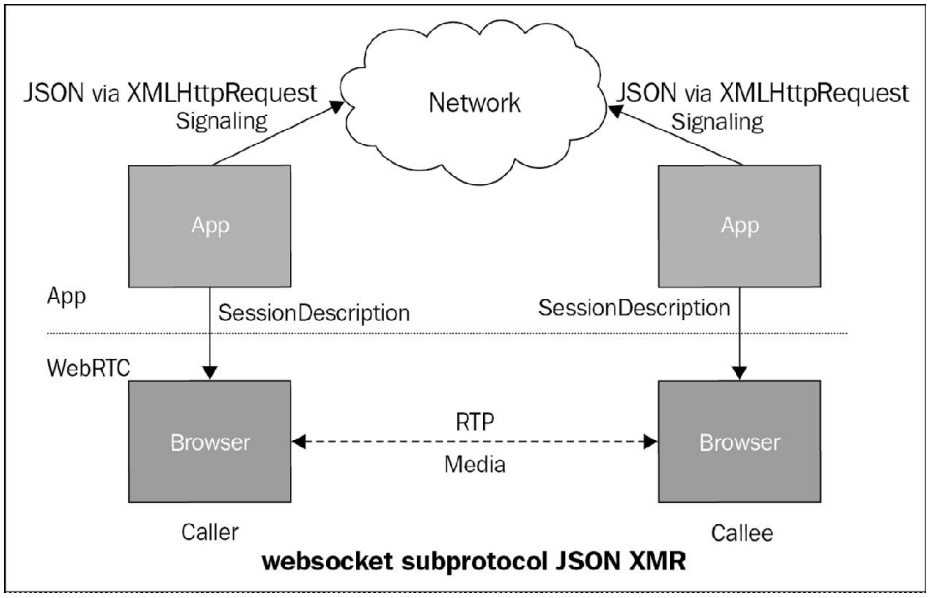
1

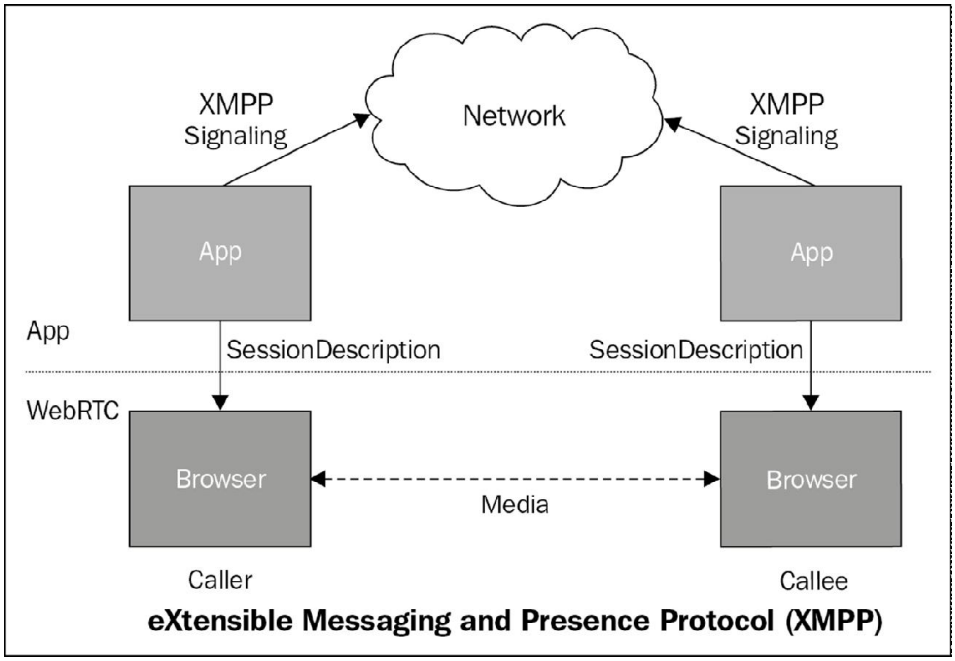
Running WebRTC With and Without SIP

JavaScript Session Establishment Protocol (JSEP)



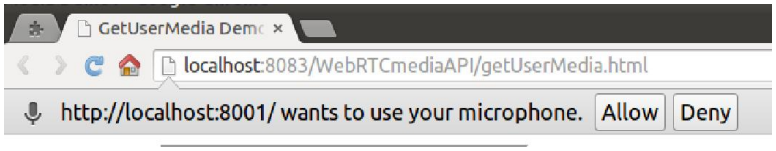
The signal and media planes

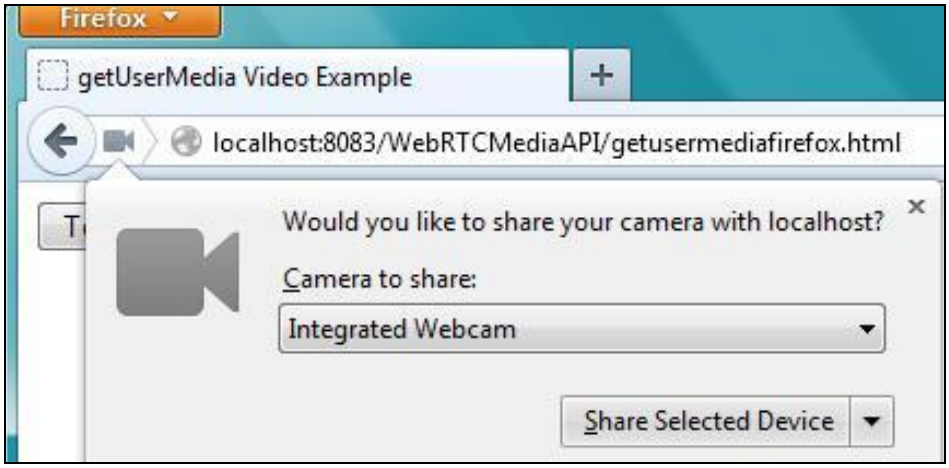


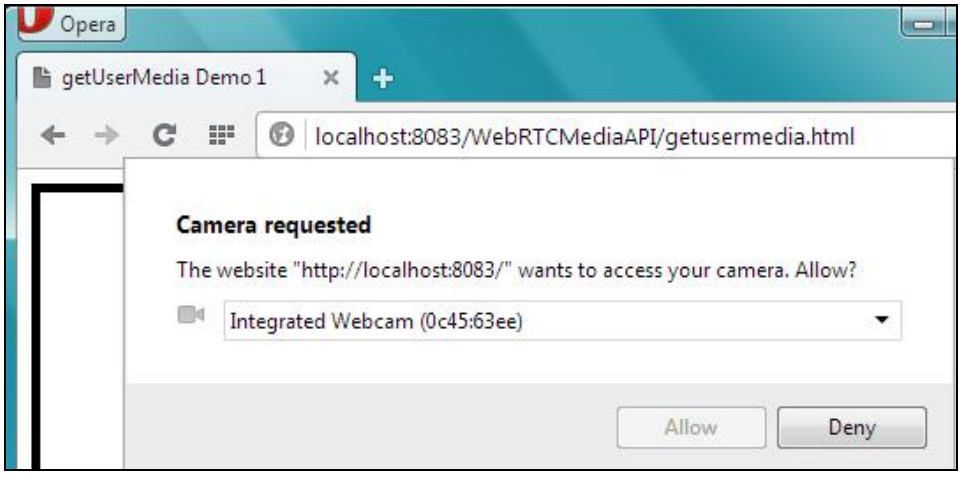


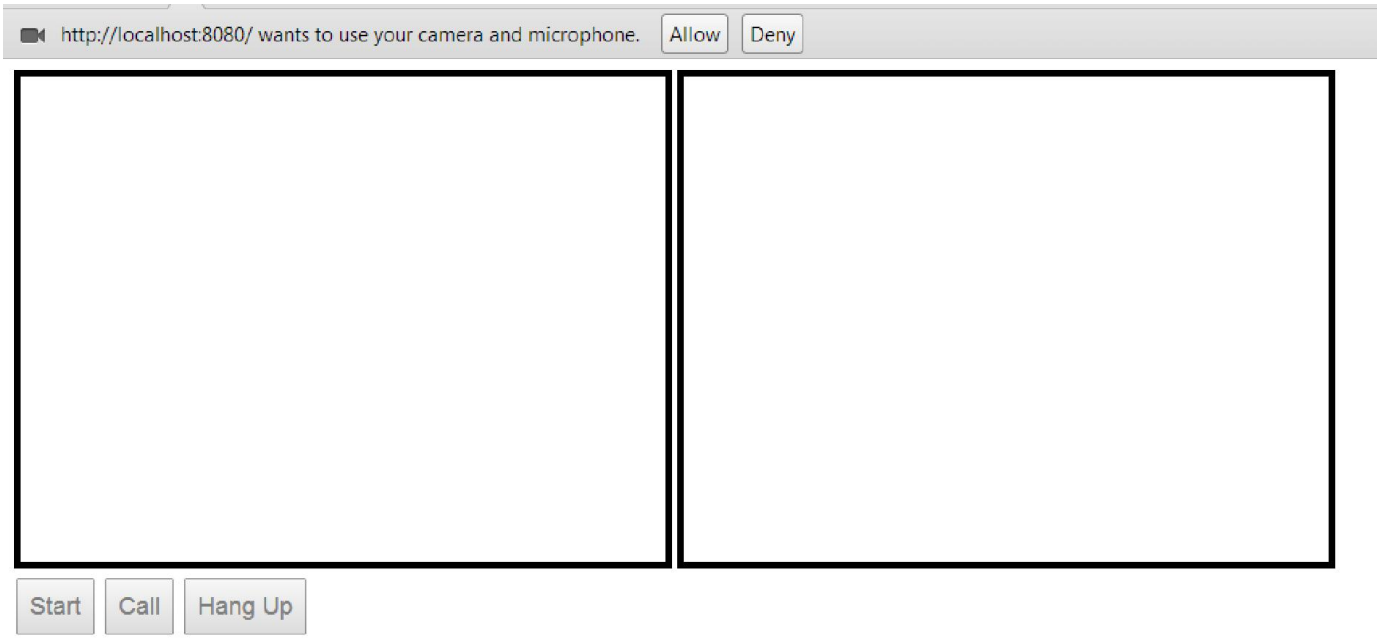
Running WebRTC without SIP

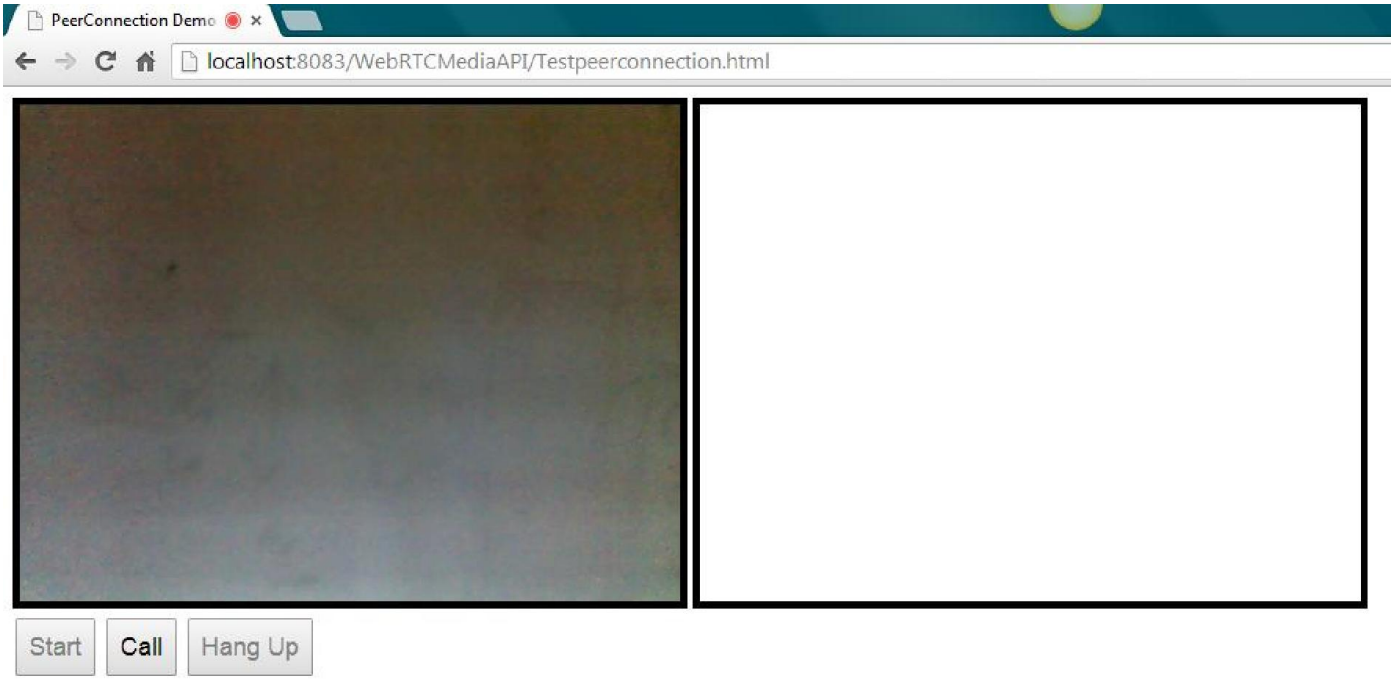
Sending media over WebSockets

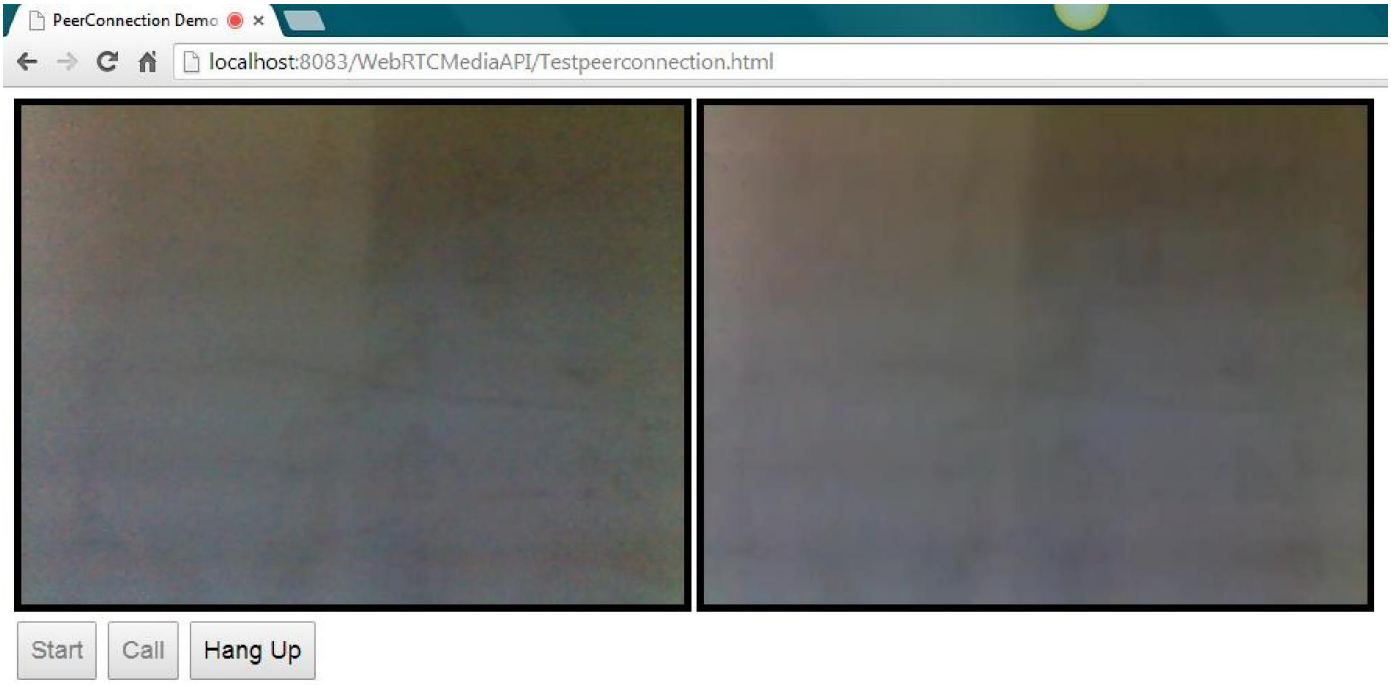


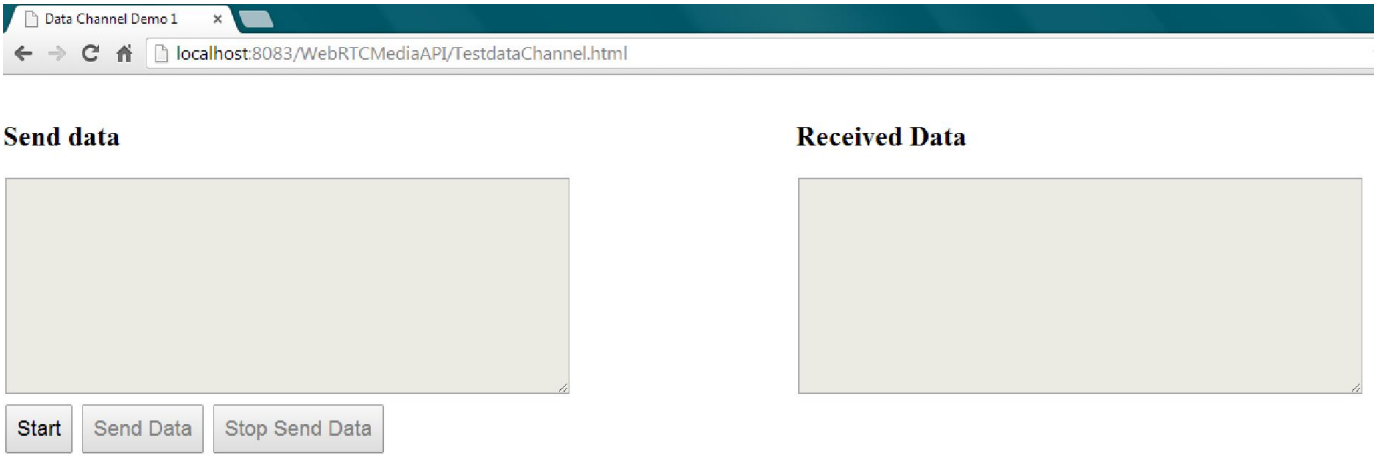





http://localhost:8080/ wants to use your camera and microphone.
Allow
Deny
Start
Call
Hang Up


PeerConnection Demo
localhost:8083/WebRTCMediaAPI/Testpeerconnection.html
Start
Call
Hang Up


PeerConnection Demo
localhost:8083/WebRTCMediaAPI/Testpeerconnection.html
Start
Call
Hang Up


Data Channel Demo 1
localhost:8083/WebRTCMediaAPI/TestdataChannel.html
Send data
Received Data
Start
Send Data
Stop Send Data

Send data

Hello ,

This a Datachannel test , to exchange a set of text between two peers .

Start | Send Data | Stop Send Data

Received Data

Send data

Hello ,

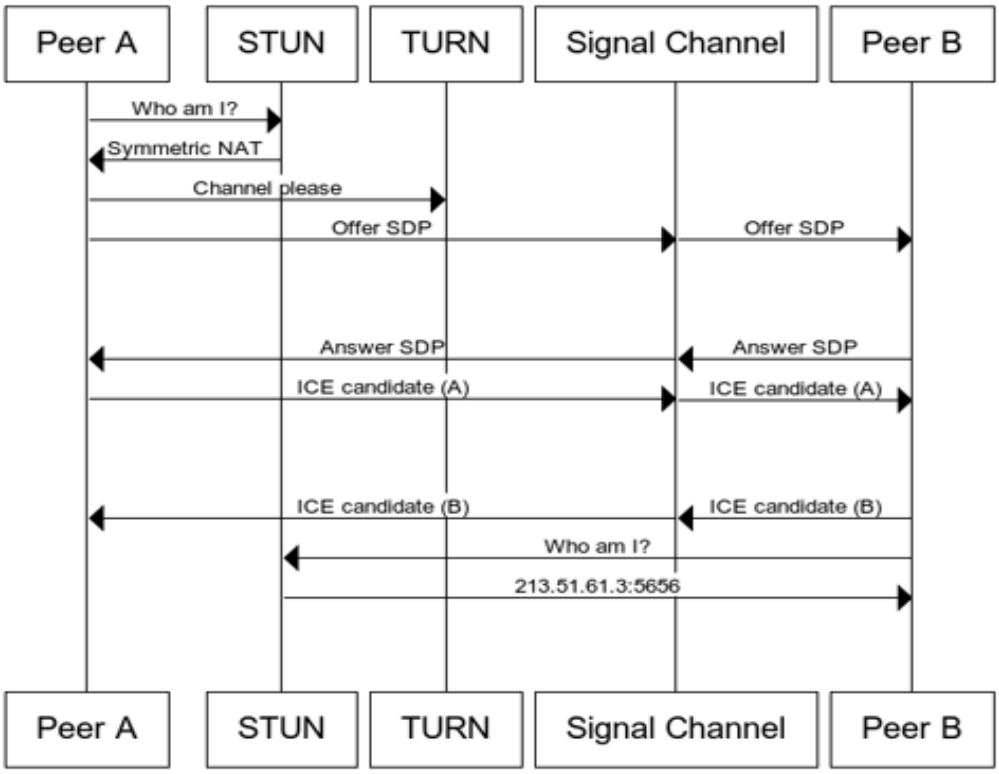
This a Datachannel test , to exchange a set of text between two peers .

Start | Send Data | Stop Send Data

Received Data

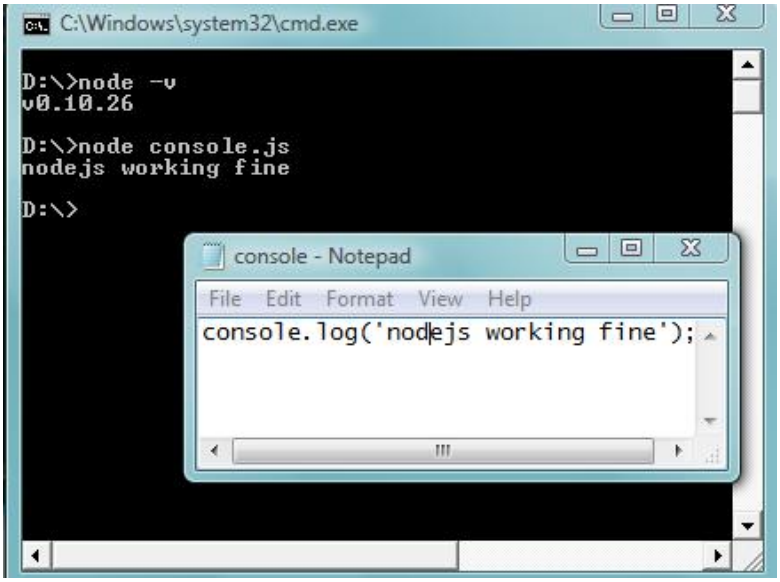
Hello ,

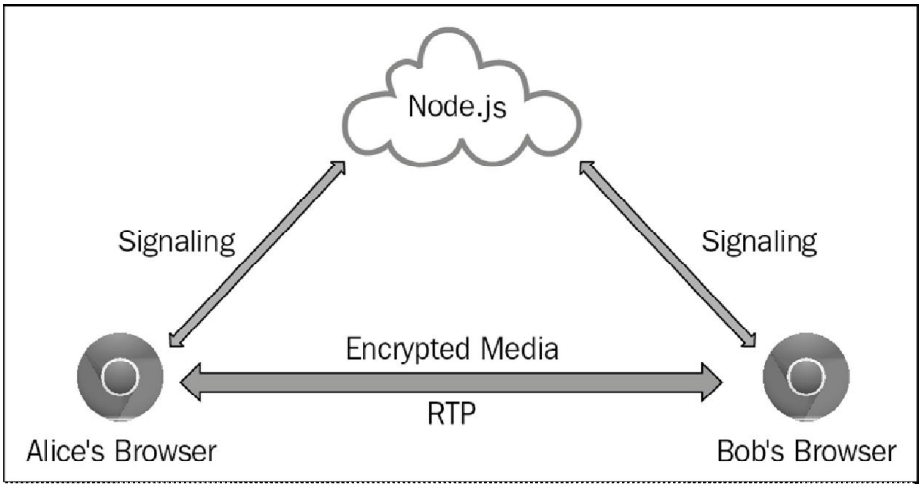
This a Datachannel test , to exchange a set of text between two peers .


Peer A
STUN
TURN
Signal Channel
Peer B
Who am I?
Symmetric NAT
Channel please
Offer SDP
Offer SDP
Answer SDP
Answer SDP
ICE candidate (A)
ICE candidate (A)
ICE candidate (B)
ICE candidate (B)
Who am I?
213.51.61.3:5656
Peer A
STUN
TURN
Signal Channel
Peer B

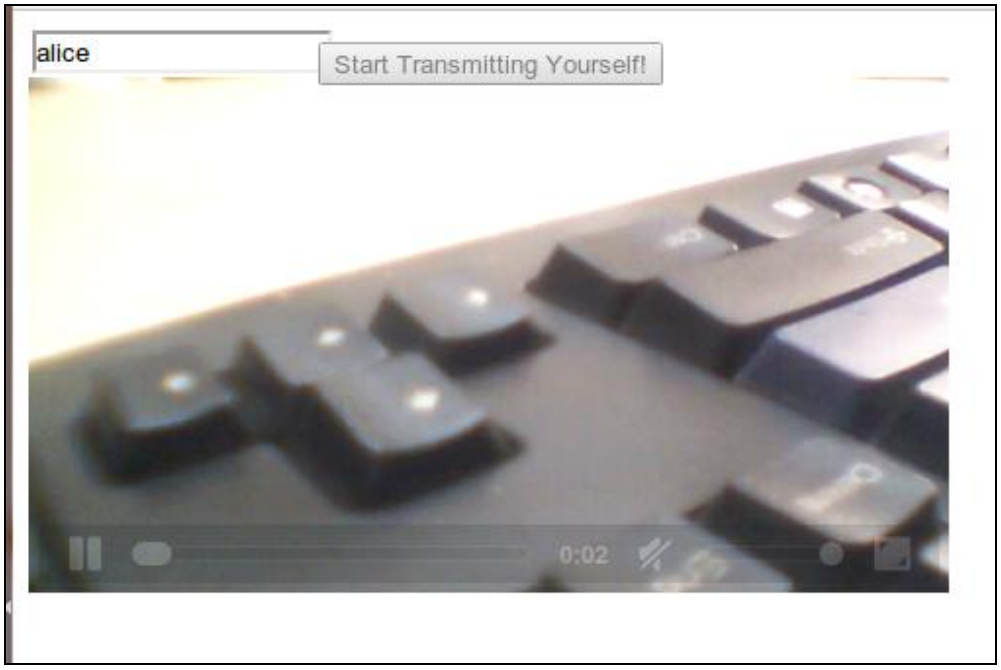
WebRTC through WebSocket signaling servers

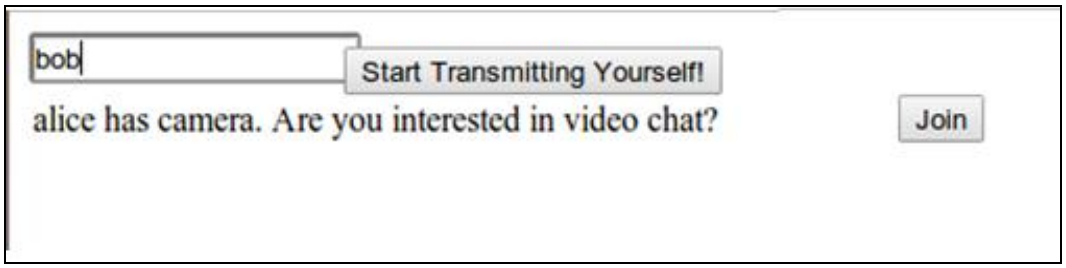
Node.js

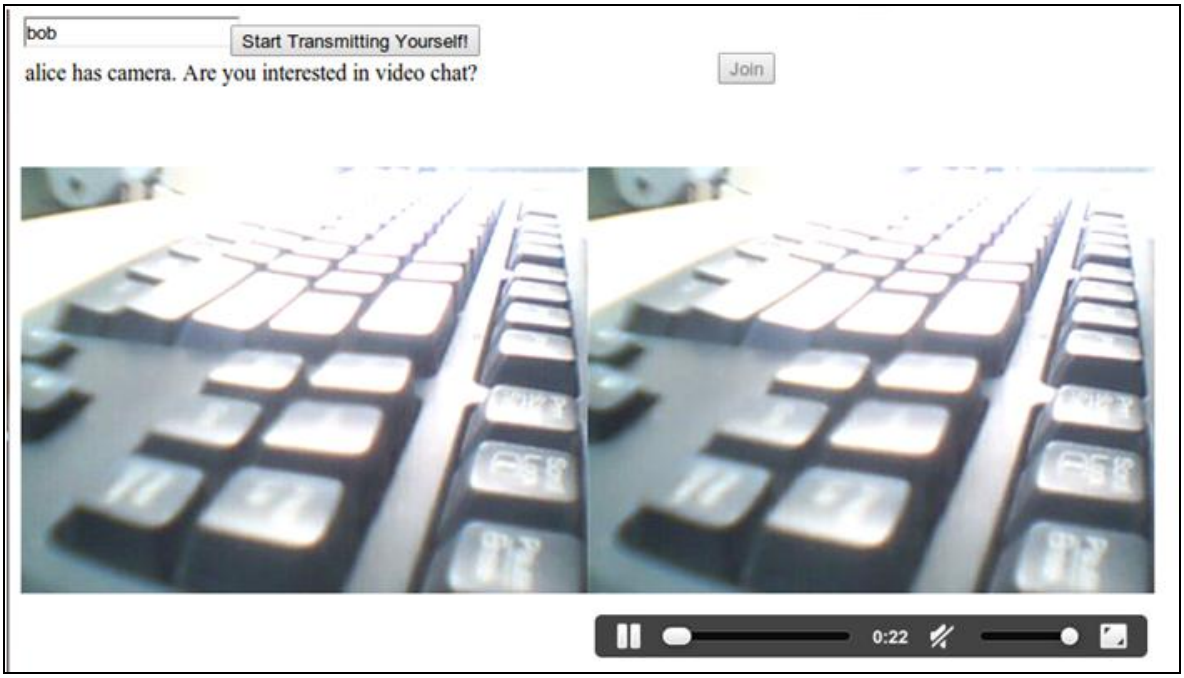




Making a peer-to-peer audio call using node.js for signaling

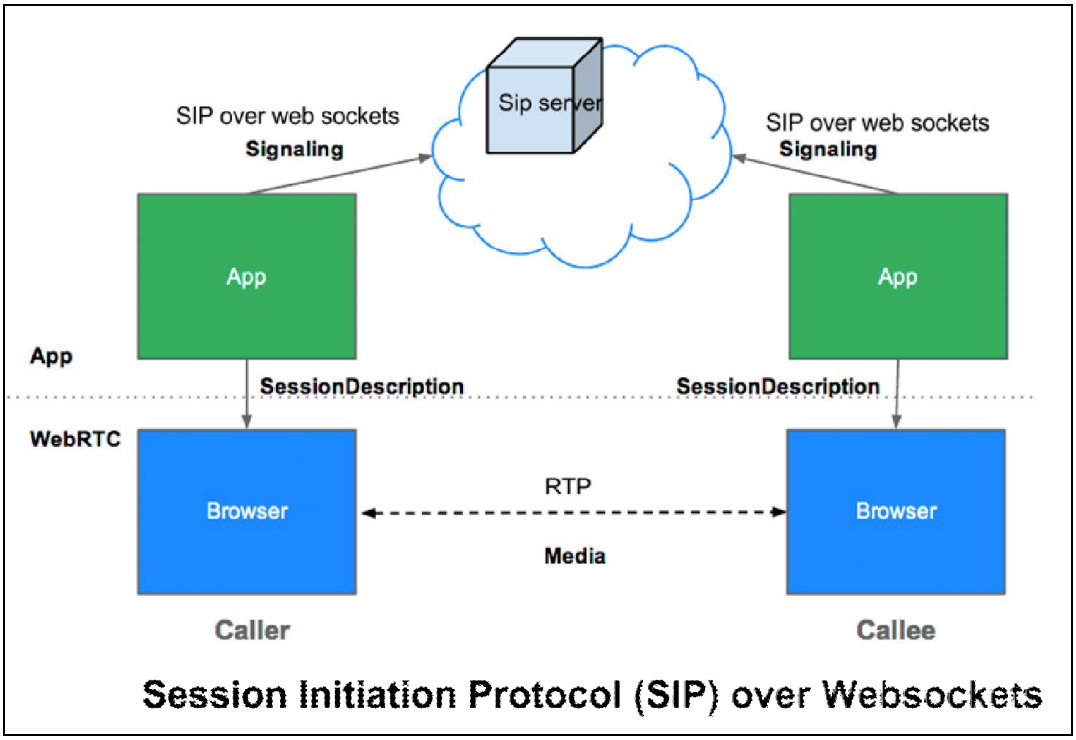




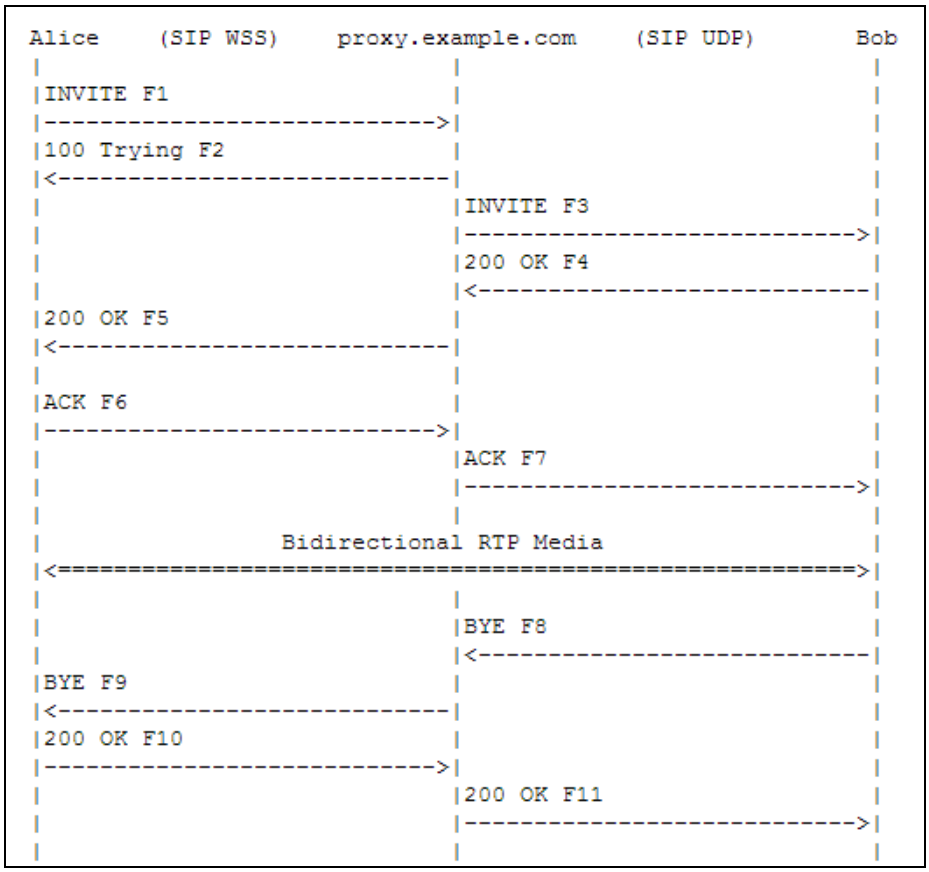

bob
Start Transmitting Yourself!
alice has camera. Are you interested in video chat?
Join
0:22

Running WebRTC with SIP

Session Initiation Protocol (SIP)



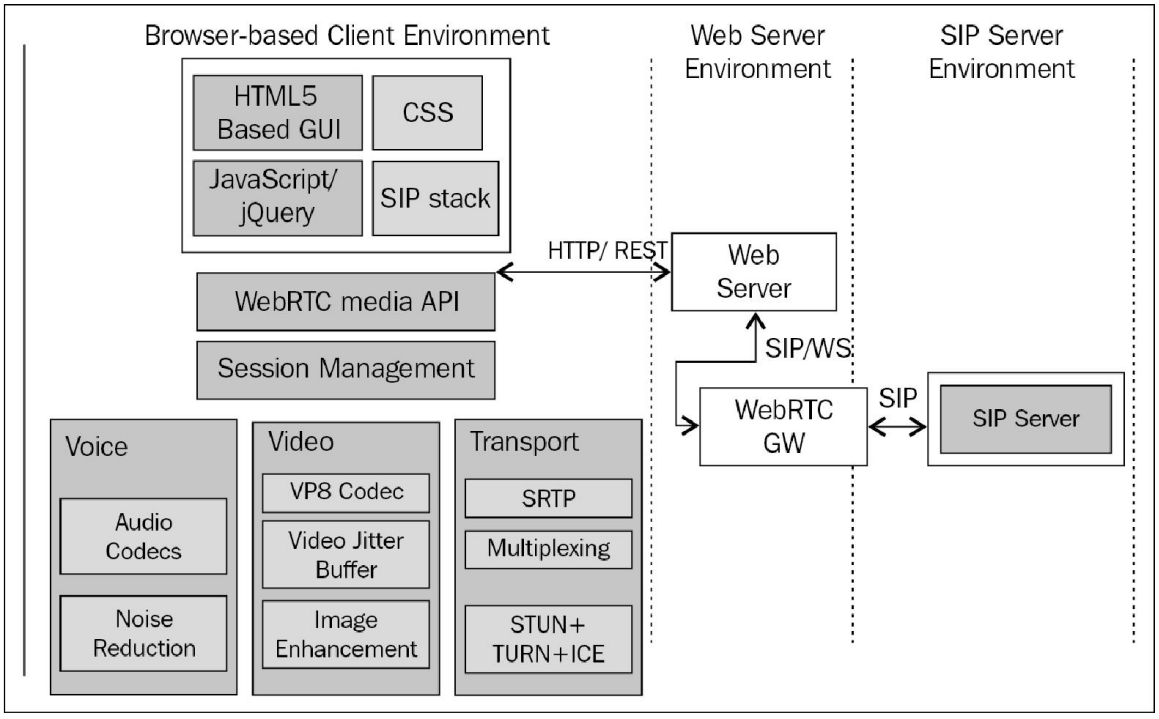

```
Alice     (SIP WSS)     proxy.example.com
|                                    |
|HTTP GET (WS handshake) F1          |
|----------------------------------->|
|101 Switching Protocols F2          |
|<-----------------------------------|
|                                    |
|REGISTER F3                         |
|----------------------------------->|
|200 OK F4                           |
|<-----------------------------------|
|                                    |
```


Alice (SIP WSS) proxy.example.com (SIP UDP) Bob
INVITE F1
100 Trying F2
INVITE F3
200 OK F4
200 OK F5
ACK F6
ACK F7
Bidirectional RTP Media
BYE F8
BYE F9
200 OK F10
200 OK F11

2

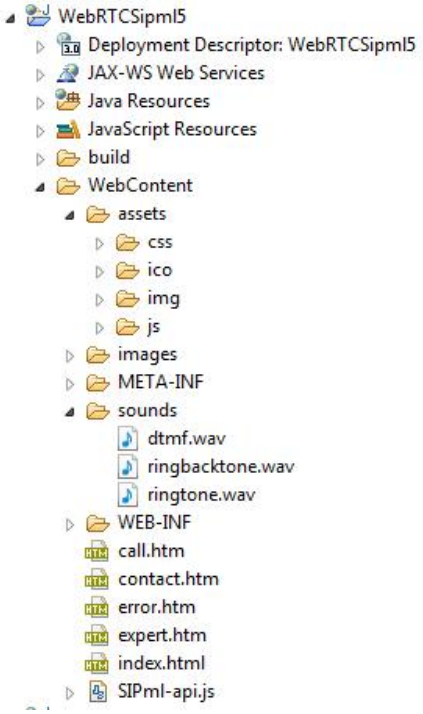
Making a Standalone WebRTC Communication Client

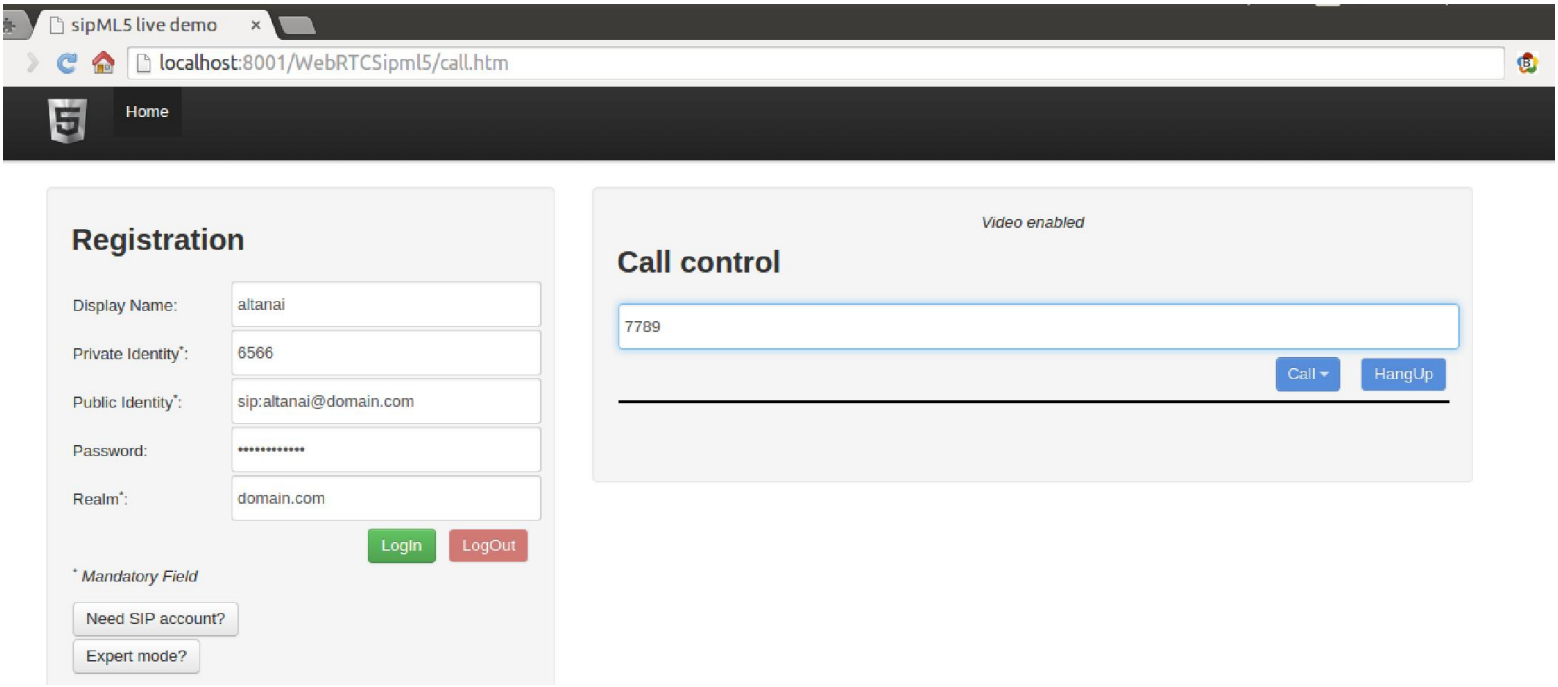
Description of WebRTC client-server model

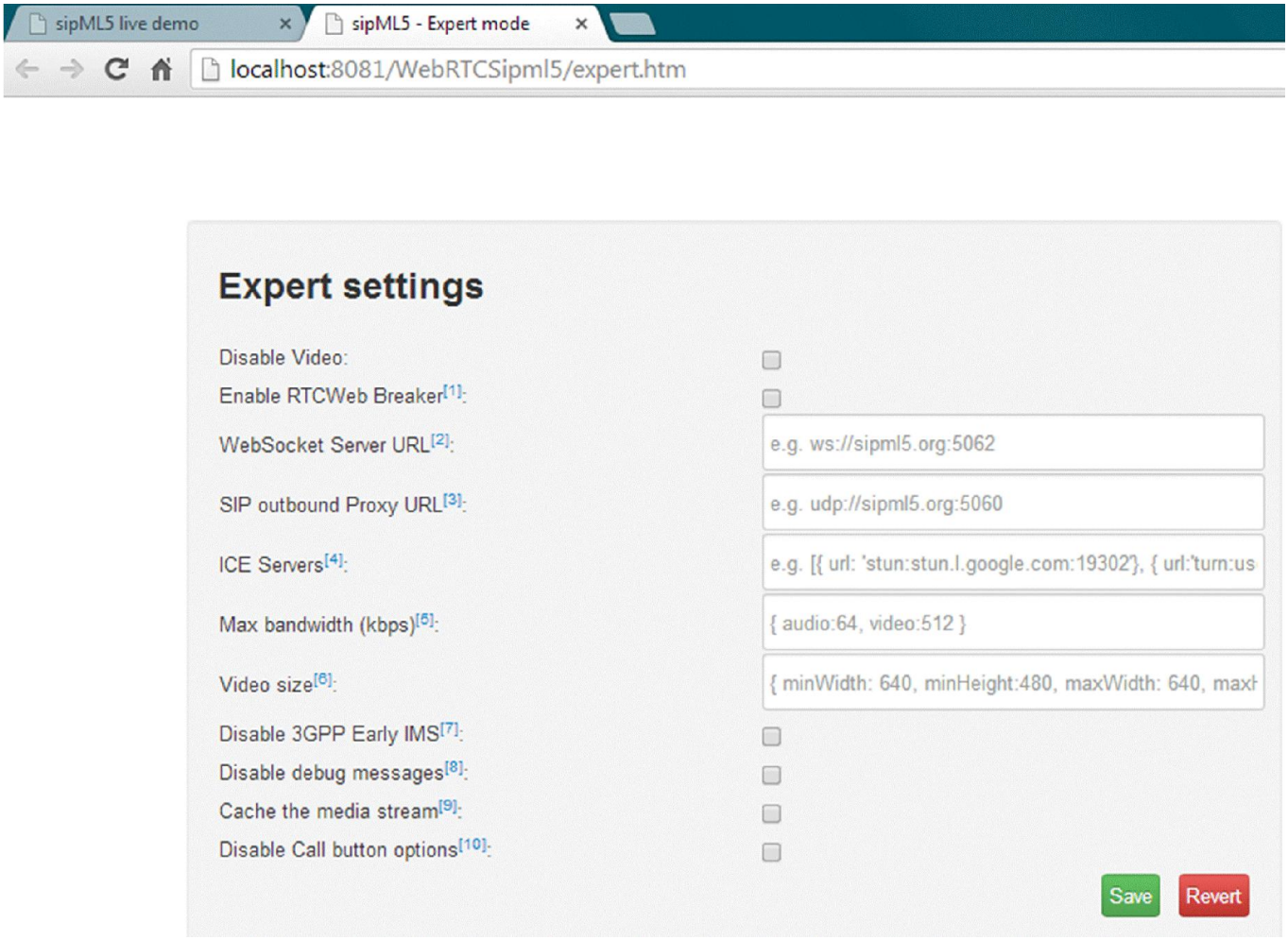


sipML5 WebRTC Client

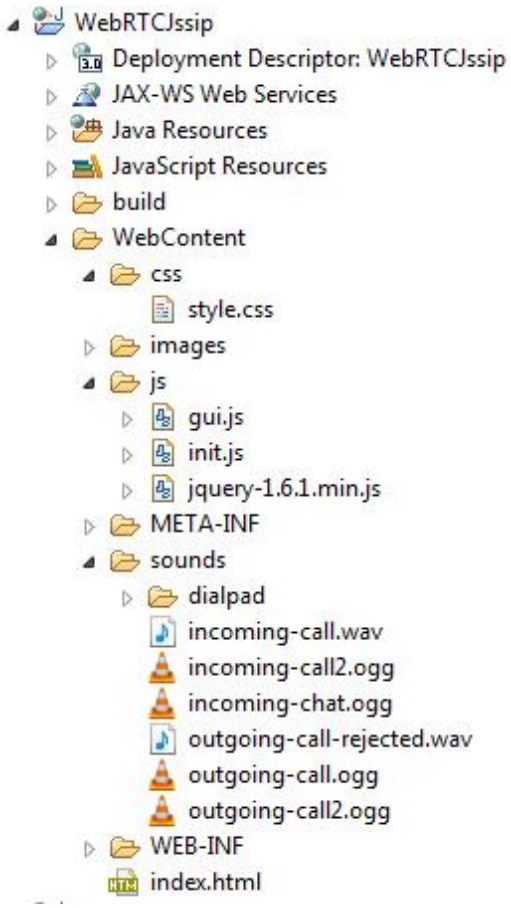
```
D:\workspacenativewebrtc\apache-tomcat-7.0.50\bin>set JAVA_HOME=C:\Program Files
\Java\jdk1.6.0_06

D:\workspacenativewebrtc\apache-tomcat-7.0.50\bin>startup.bat
Using CATALINA_BASE:   "D:\workspacenativewebrtc\apache-tomcat-7.0.50"
Using CATALINA_HOME:   "D:\workspacenativewebrtc\apache-tomcat-7.0.50"
Using CATALINA_TMPDIR: "D:\workspacenativewebrtc\apache-tomcat-7.0.50\temp"
Using JRE_HOME:        "C:\Program Files\Java\jdk1.6.0_06"
Using CLASSPATH:       "D:\workspacenativewebrtc\apache-tomcat-7.0.50\bin\bootst
rap.jar;D:\workspacenativewebrtc\apache-tomcat-7.0.50\bin\tomcat-juli.jar"
D:\workspacenativewebrtc\apache-tomcat-7.0.50\bin>
```



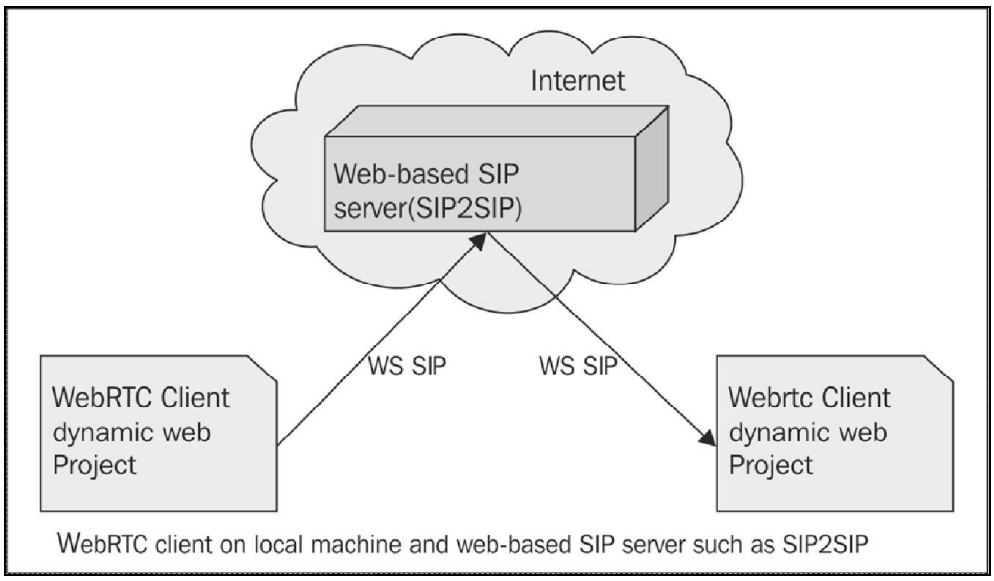

sipML5 live demo
localhost:8001/WebRTCSipml5/call.htm
Home
Registration
Display Name:
altanai
Private Identity*:
6566
Public Identity*:
sip:altanai@domain.com
Password:
Realm*:
domain.com
LogIn
LogOut
* Mandatory Field
Need SIP account?
Expert mode?
Video enabled
Call control
7789
Call
HangUp

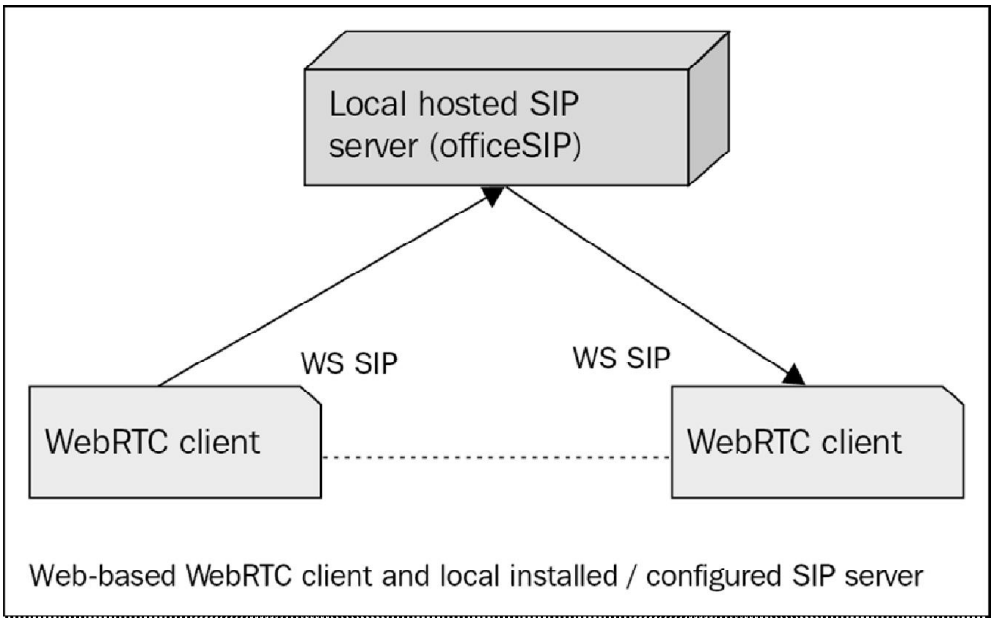

sipML5 live demo
sipML5 - Expert mode
localhost:8081/WebRTCSipml5/expert.htm
Expert settings
Disable Video:
Enable RTCWeb Breaker[1]:
WebSocket Server URL[2]:
e.g. ws://sipml5.org:5062
SIP outbound Proxy URL[3]:
e.g. udp://sipml5.org:5060
ICE Servers[4]:
e.g. [{ url: 'stun:stun.l.google.com:19302'}, { url:'turn:us
Max bandwidth (kbps)[5]:
{ audio:64, video:512 }
Video size[6]:
{ minWidth: 640, minHeight:480, maxWidth: 640, maxH
Disable 3GPP Early IMS[7]:
Disable debug messages[8]:
Cache the media stream[9]:
Disable Call button options[10]:
Save
Revert

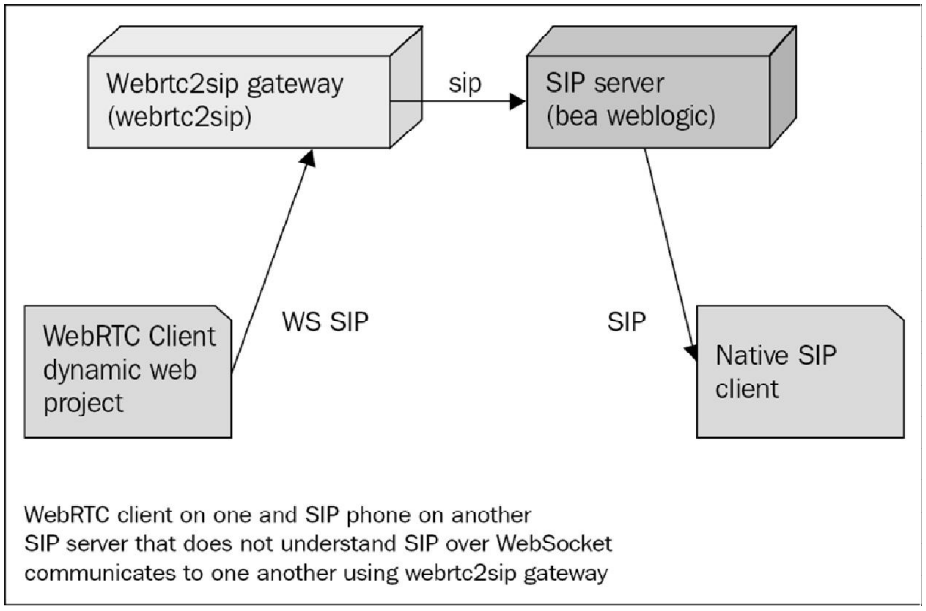
The jsSIP WebRTC client



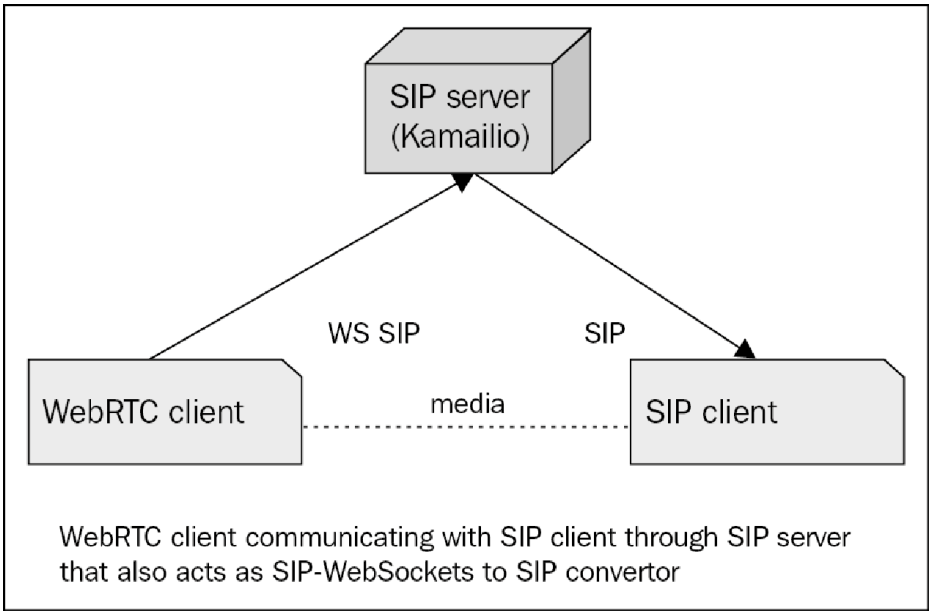
SIP servers


Internet
Web-based SIP server(SIP2SIP)
WS SIP
WS SIP
WebRTC Client dynamic web Project
Webrtc Client dynamic web Project
WebRTC client on local machine and web-based SIP server such as SIP2SIP


Local hosted SIP server (officeSIP)
WS SIP
WS SIP
WebRTC client
WebRTC client
Web-based WebRTC client and local installed / configured SIP server

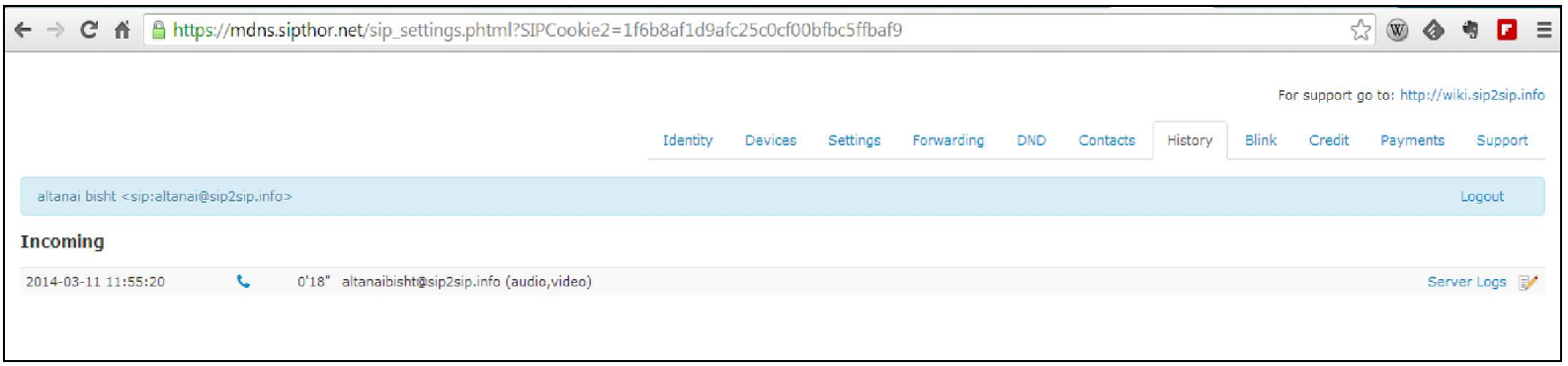


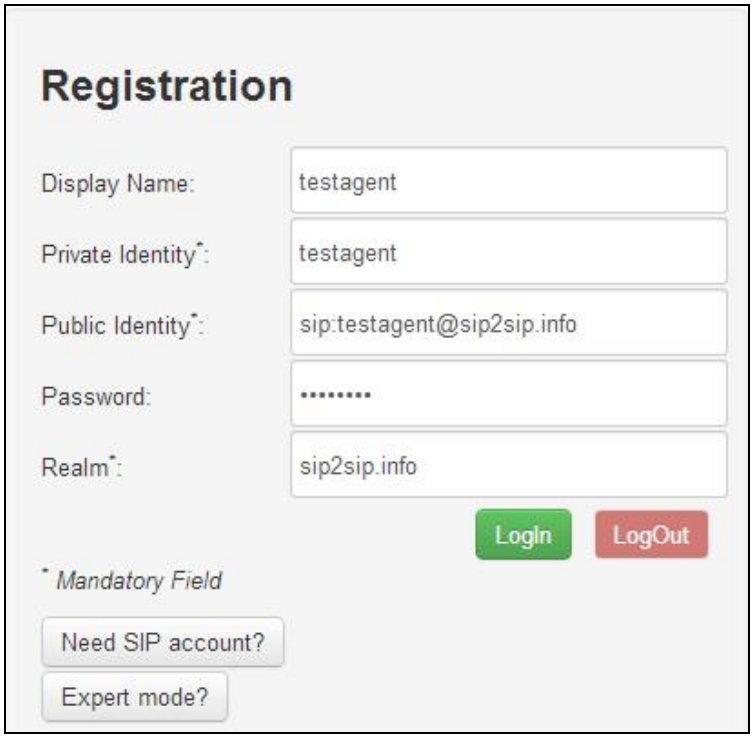
WebRTC client on one and SIP phone on another SIP server that does not understand SIP over WebSocket communicates to one another using webrtc2sip gateway



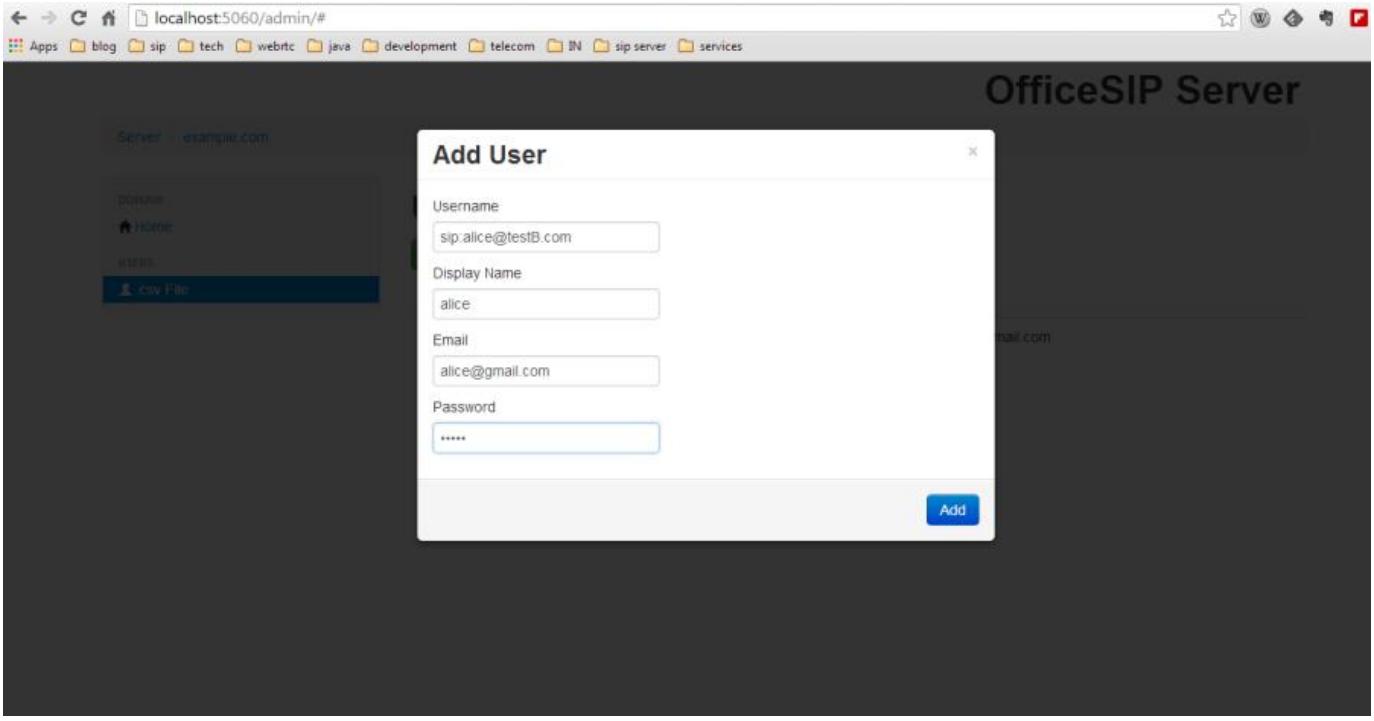
WebRTC client communicating with SIP client through SIP server that also acts as SIP-WebSockets to SIP convertor

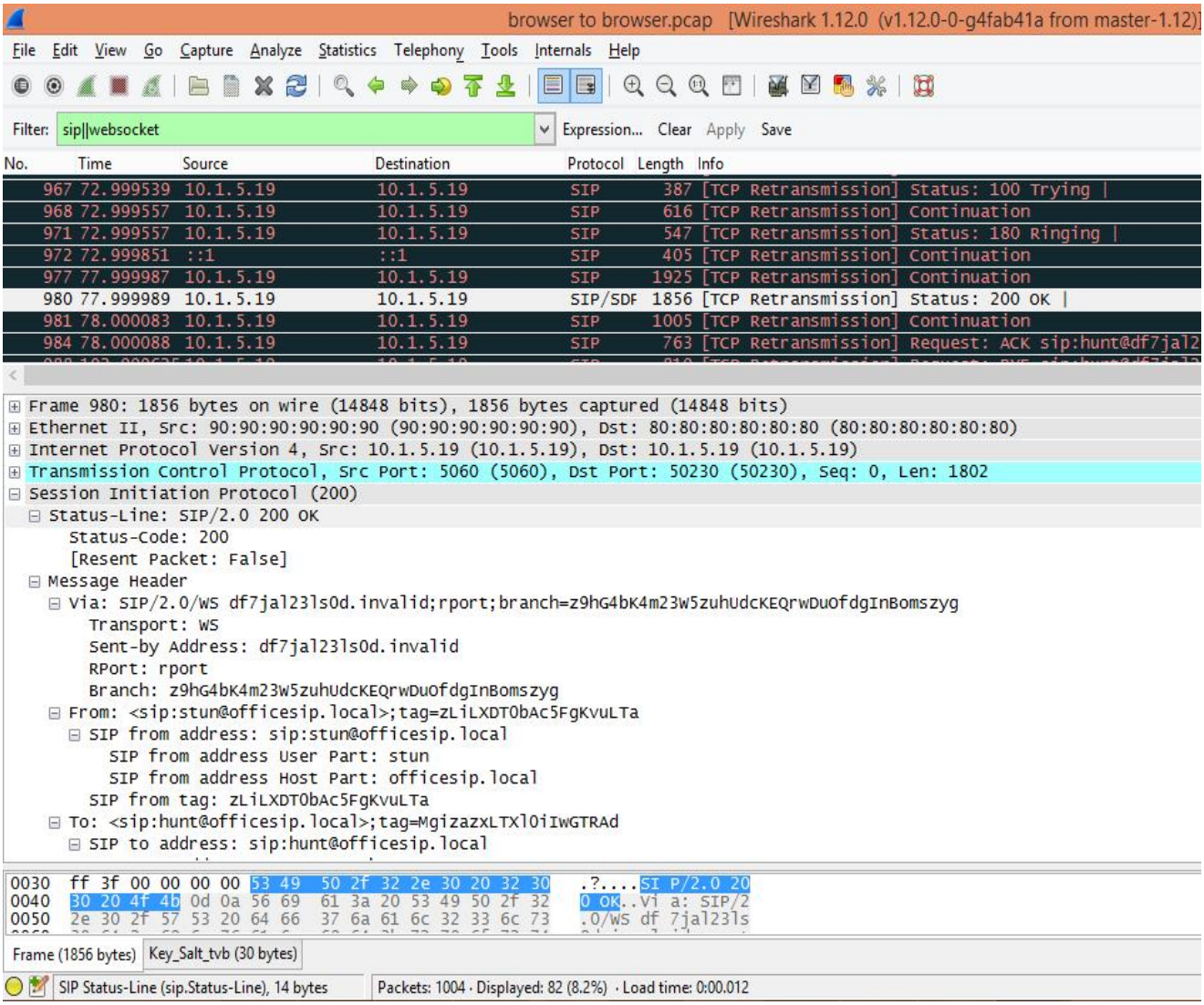
SIPWS to SIPWS



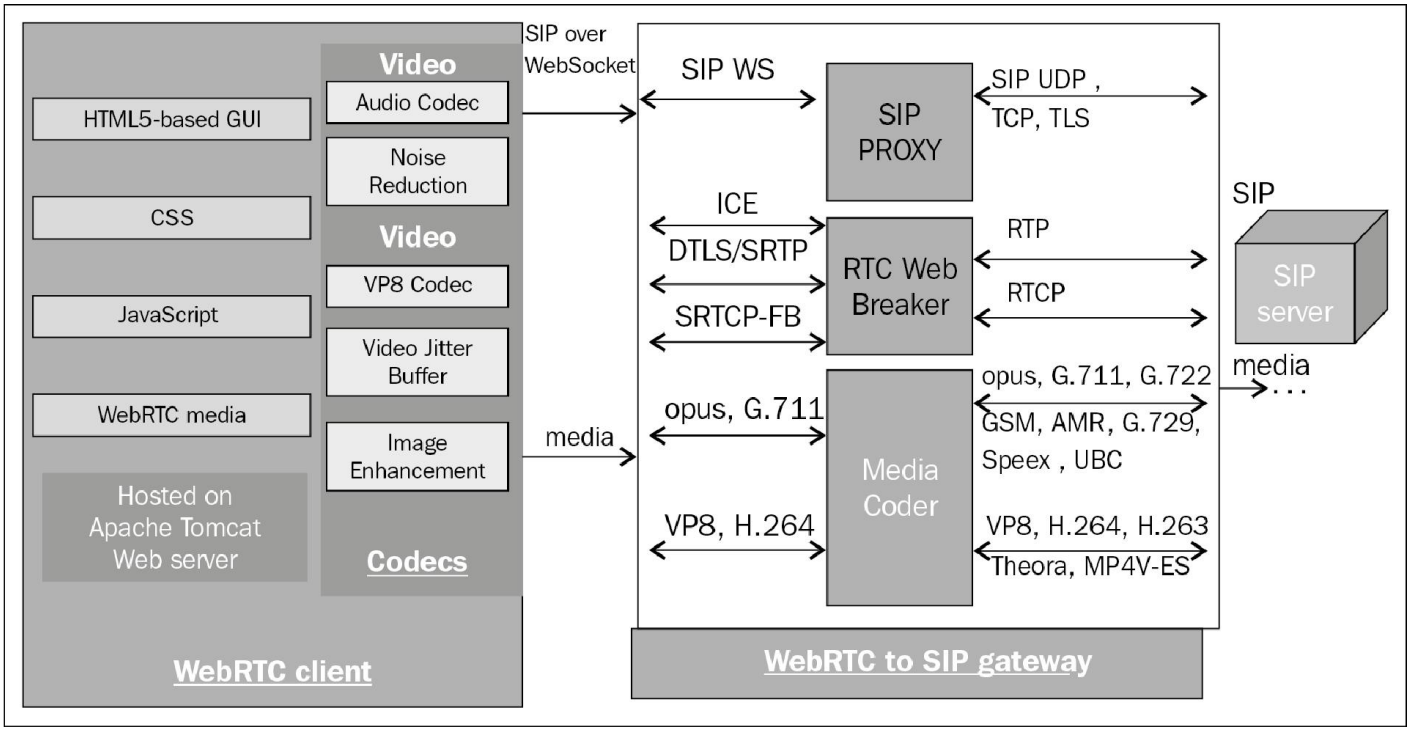


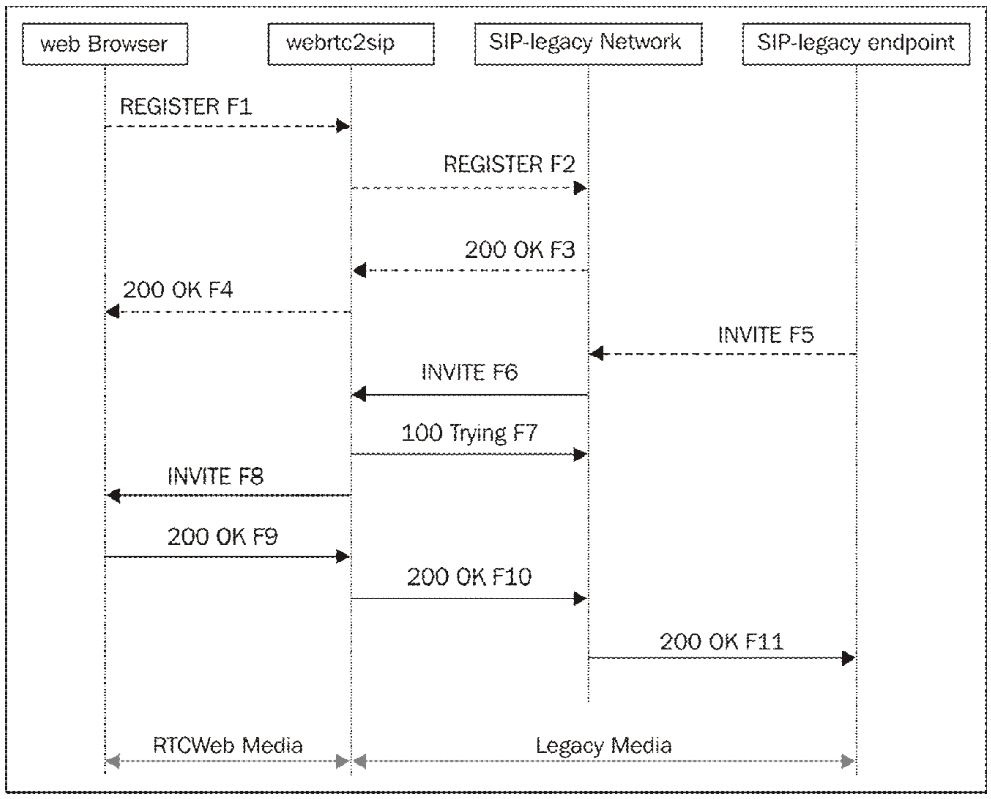
OfficeSIP Server Setup
SiP
SERVER
Welcome to the OfficeSIP Server Setup Wizard
The Setup Wizard will install OfficeSIP Server on your computer. Click Next to continue or Cancel to exit the Setup Wizard.
Back
Next
Cancel


localhost:5060/admin/#
Apps
blog
sip
tech
webrtc
java
development
telecom
IN
sip server
services
OfficeSIP Server
Add User
Username
sip:alice@testB.com
Display Name
alice
Email
alice@gmail.com
Password
Add


browser to browser.pcap [Wireshark 1.12.0 (v1.12.0-0-g4fab41a from master-1.12)]
File Edit View Go Capture Analyze Statistics Telephony Tools Internals Help
Filter: sip||websocket
Expression... Clear Apply Save
No. Time Source Destination Protocol Length Info
967 72.999539 10.1.5.19 10.1.5.19 SIP 387 [TCP Retransmission] Status: 100 Trying |
968 72.999557 10.1.5.19 10.1.5.19 SIP 616 [TCP Retransmission] Continuation
971 72.999557 10.1.5.19 10.1.5.19 SIP 547 [TCP Retransmission] Status: 180 Ringing |
972 72.999851 ::1 ::1 SIP 405 [TCP Retransmission] Continuation
977 77.999987 10.1.5.19 10.1.5.19 SIP 1925 [TCP Retransmission] Continuation
980 77.999989 10.1.5.19 10.1.5.19 SIP/SDP 1856 [TCP Retransmission] Status: 200 OK |
981 78.000083 10.1.5.19 10.1.5.19 SIP 1005 [TCP Retransmission] Continuation
984 78.000088 10.1.5.19 10.1.5.19 SIP 763 [TCP Retransmission] Request: ACK sip:hunt@df7jal2
Frame 980: 1856 bytes on wire (14848 bits), 1856 bytes captured (14848 bits)
Ethernet II, Src: 90:90:90:90:90:90 (90:90:90:90:90:90), Dst: 80:80:80:80:80:80 (80:80:80:80:80:80)
Internet Protocol Version 4, Src: 10.1.5.19 (10.1.5.19), Dst: 10.1.5.19 (10.1.5.19)
Transmission Control Protocol, Src Port: 5060 (5060), Dst Port: 50230 (50230), Seq: 0, Len: 1802
Session Initiation Protocol (200)
Status-Line: SIP/2.0 200 OK
Status-Code: 200
[Resent Packet: False]
Message Header
Via: SIP/2.0/WS df7jal23ls0d.invalid;rport;branch=z9hG4bK4m23W5zuhUdcKEQrwDuOfdgInBomszyg
Transport: WS
Sent-by Address: df7jal23ls0d.invalid
RPort: rport
Branch: z9hG4bK4m23W5zuhUdcKEQrwDuOfdgInBomszyg
From: <sip:stun@officesip.local>;tag=zLiLXDTObAc5FgKvuLTa
SIP from address: sip:stun@officesip.local
SIP from address User Part: stun
SIP from address Host Part: officesip.local
SIP from tag: zLiLXDTObAc5FgKvuLTa
To: <sip:hunt@officesip.local>;tag=MgizazxLTXl0iIwGTRAd
SIP to address: sip:hunt@officesip.local
0030 ff 3f 00 00 00 00 53 49 50 2f 32 2e 30 20 32 30 .?....SI P/2.0 20
0040 30 20 4f 4b 0d 0a 56 69 61 3a 20 53 49 50 2f 32 0 OK..Vi a: SIP/2
0050 2e 30 2f 57 53 20 64 66 37 6a 61 6c 32 33 6c 73 .0/WS df 7jal23ls
Frame (1856 bytes)
Key_Salt_tvb (30 bytes)
SIP Status-Line (sip.Status-Line), 14 bytes
Packets: 1004 · Displayed: 82 (8.2%) · Load time: 0:00.012

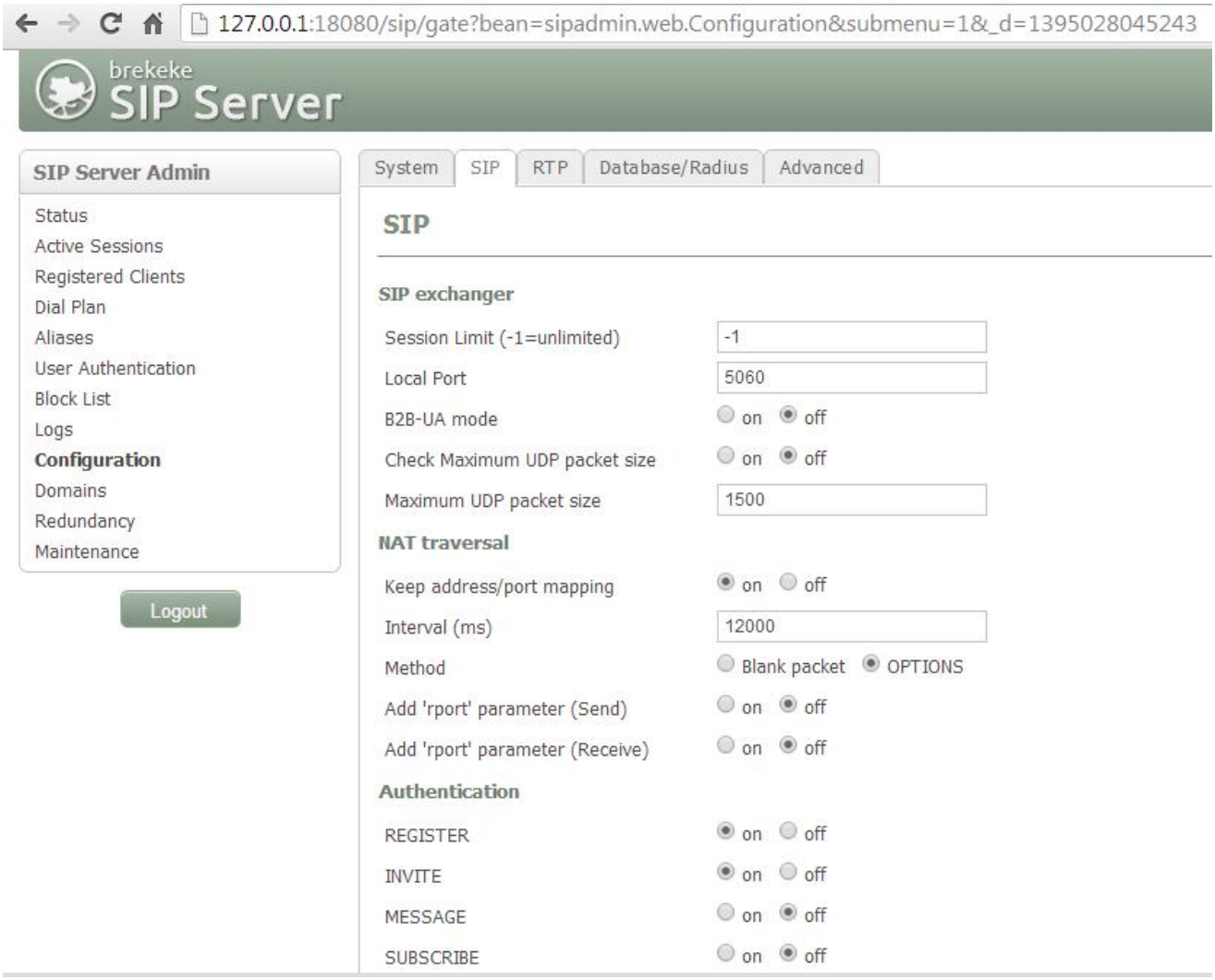
The gateway to convert SIP over WebSocket to native SIP




web Browser
webrtc2sip
SIP-legacy Network
SIP-legacy endpoint
REGISTER F1
REGISTER F2
200 OK F3
200 OK F4
INVITE F5
INVITE F6
100 Trying F7
INVITE F8
200 OK F9
200 OK F10
200 OK F11
RTCWeb Media
Legacy Media

```
config.status: creating tinySIP/tinySIP.pc
config.status: creating bindings/Makefile
config.status: creating bindings/tinyWRAP.pc
config.status: creating plugins/audio_opensles/Makefile
config.status: creating plugins/audio_opensles/plugin_audio_opensles.pc
config.status: creating config.h
config.status: executing depfiles commands
config.status: executing libtool commands
configure:

************************************************************************
*                              CONGRATULATIONS
************************************************************************
Cross Compilation:   no
Target OS:           Linux
Host setup:          x86_64-unknown-linux-gnu
Install prefix:      /usr/local
Compiler:            gcc

Enable GPL:          yes
Enable Non Free:     yes

FFmpeg:              no
VP8 video codec:     yes
OPUS audio codec:    no
ILBC audio codec:    no
G.729 audio codec:   no -> yes
GSM audio codec:     no
AMR audio codec:     no -> yes
SPEEX audio codec:   yes
G.722 audio codec:   yes
G.711 audio codec:   yes

YUV:                 no
SPEEX DSP:           yes

SSL:                 yes
DTLS-SRTP:           yes
DTLS:                yes

SRTP:                yes

WebRTC:              Enabled(no): AEC(no), NS(no)

Monotonic timers:    yes
RESOLV:              no

DEBUG:               no

Report any issue at https://groups.google.com/group/doubango
```

```
checking for TINYMSRP... yes
checking for TINYRTP... yes
checking for TINYIPSEC... yes
checking for libxml2... /usr
checking for libpthread... /usr
checking that generated files are newer than configure... done
configure: creating ./config.status
config.status: creating Makefile
config.status: creating config.h
config.status: config.h is unchanged
config.status: executing depfiles commands
configure:

********************************************************************************
                                CONGRATULATIONS
********************************************************************************
Host setup:
Install prefix:             /opt/webrtc2sip
Compiler:                   gcc

DOUBANGO
     MIN_VER:               2.0.898
     TINYSAK:               yes
     TINYNET:               yes
     TINYHTTP:              yes
     TINYSIP:               yes
     TINYDAV:               yes
     TINYSDP:               yes
     TINYSIGCOMP:           yes
     TINYMEDIA:             yes
     TINYMEDIA:             yes
     TINYSMS:               yes
     TINYMSRP:              yes
     TINYRTP:               yes
     TINYIPSEC:             yes

LIBXML2_INCLUDE:            /usr/include/libxml2
LIBXML2_LIB:                /usr/lib

LIBPTHREAD_INCLUDE:         /usr/include
LIBPTHREAD_LIB:             /usr/lib

Report issues at https://groups.google.com/group/doubango

--------------------------------------------------------------------------------
Next steps
  1) run 'make' to build the source
  2) run 'make install' to install
--------------------------------------------------------------------------------


altanai@tcs:~/Downloads/webrtc2sip-read-only$ sudo make
make  all-am
```

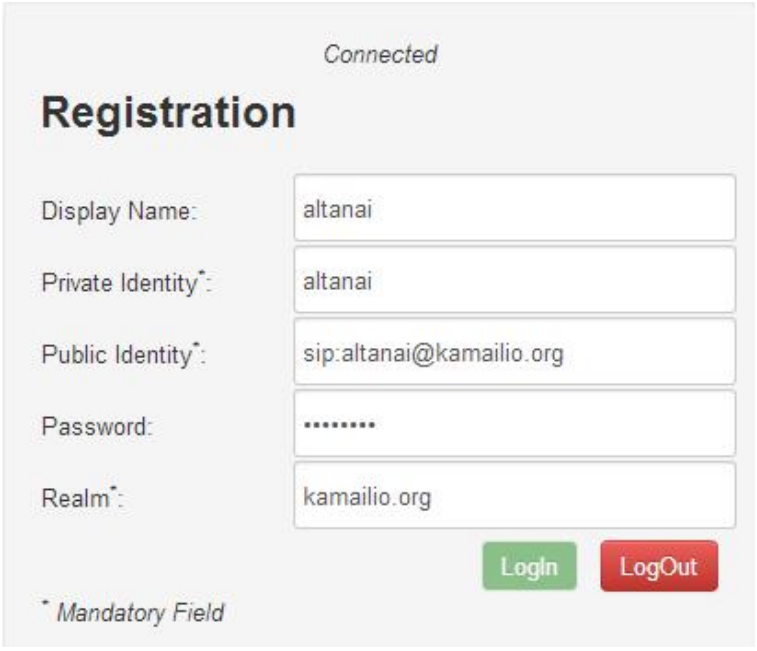

127.0.0.1:18080/sip/gate?bean=sipadmin.web.Configuration&submenu=1&_d=1395028045243
brekeke
SIP Server
SIP Server Admin
Status
Active Sessions
Registered Clients
Dial Plan
Aliases
User Authentication
Block List
Logs
Configuration
Domains
Redundancy
Maintenance
Logout
System
SIP
RTP
Database/Radius
Advanced
SIP
SIP exchanger
Session Limit (-1=unlimited)
-1
Local Port
5060
B2B-UA mode
on
off
Check Maximum UDP packet size
on
off
Maximum UDP packet size
1500
NAT traversal
Keep address/port mapping
on
off
Interval (ms)
12000
Method
Blank packet
OPTIONS
Add 'rport' parameter (Send)
on
off
Add 'rport' parameter (Receive)
on
off
Authentication
REGISTER
on
off
INVITE
on
off
MESSAGE
on
off
SUBSCRIBE
on
off

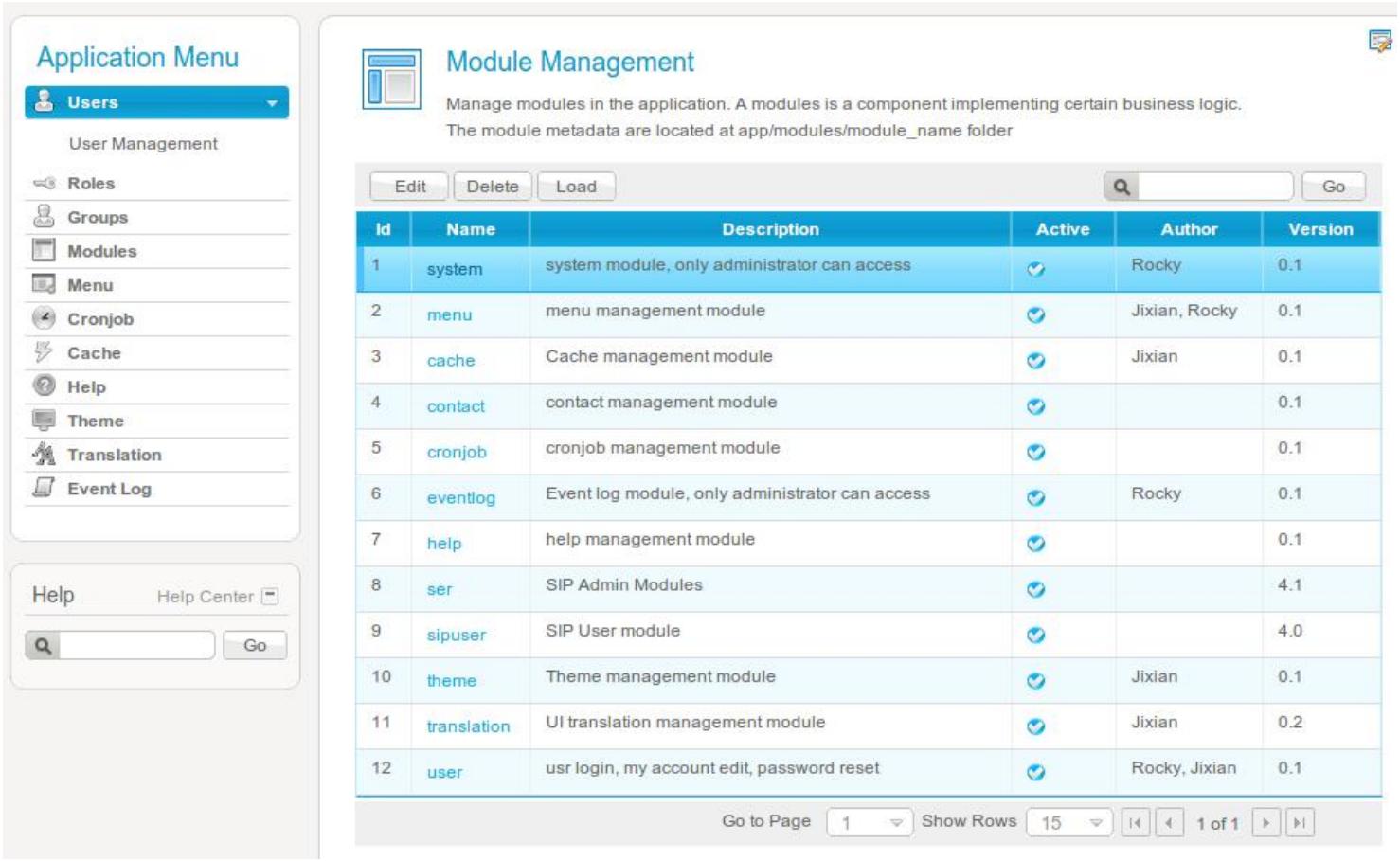
```
altanai@altanai-ThinkCentre-M91P:~$ git clone git://git.sip-router.org/kamailio
kamailio
Cloning into 'kamailio'...
remote: Counting objects: 154816, done.
remote: Compressing objects: 100% (36188/36188), done.
remote: Total 154816 (delta 110885), reused 151152 (delta 107667)
Receiving objects: 100% (154816/154816), 38.25 MiB | 173 KiB/s, done.
Resolving deltas: 100% (110885/110885), done.
altanai@altanai-ThinkCentre-M91P:~$ cd kamailio
altanai@altanai-ThinkCentre-M91P:~/kamailio$ pwd
/home/altanai/kamailio
altanai@altanai-ThinkCentre-M91P:~/kamailio$ git checkout -b 4.1 origin/4.1
Branch 4.1 set up to track remote branch 4.1 from origin.
Switched to a new branch '4.1'
altanai@altanai-ThinkCentre-M91P:~/kamailio$ make cfg; make all; make install
target architecture <x86_64>, host architecture <x86_64>
making config...
rm -f modules.lst
make --no-print-directory modules.lst
saving modules list...
generating autover.h ...
CC (gcc) [kamailio]            action.o
CC (gcc) [kamailio]            atomic_ops.o
CC (gcc) [kamailio]            basex.o
CC (gcc) [kamailio]            bit_count.o
CC (gcc) [kamailio]            bit_scan.o
CC (gcc) [kamailio]            cfg_core.o
CC (gcc) [kamailio]            cfg_parser.o
CC (gcc) [kamailio]            core_cmd.o
CC (gcc) [kamailio]            counters.o
CC (gcc) [kamailio]            crc.o
CC (gcc) [kamailio]            daemonize.o
CC (gcc) [kamailio]            data_lump.o
CC (gcc) [kamailio]            data_lump_rpl.o
CC (gcc) [kamailio]            dns_cache.o
CC (gcc) [kamailio]            dns_func.o
CC (gcc) [kamailio]            dprint.o
CC (gcc) [kamailio]            dset.o
CC (gcc) [kamailio]            dst_blacklist.o
CC (gcc) [kamailio]            endianness.o
CC (gcc) [kamailio]            error.o
```

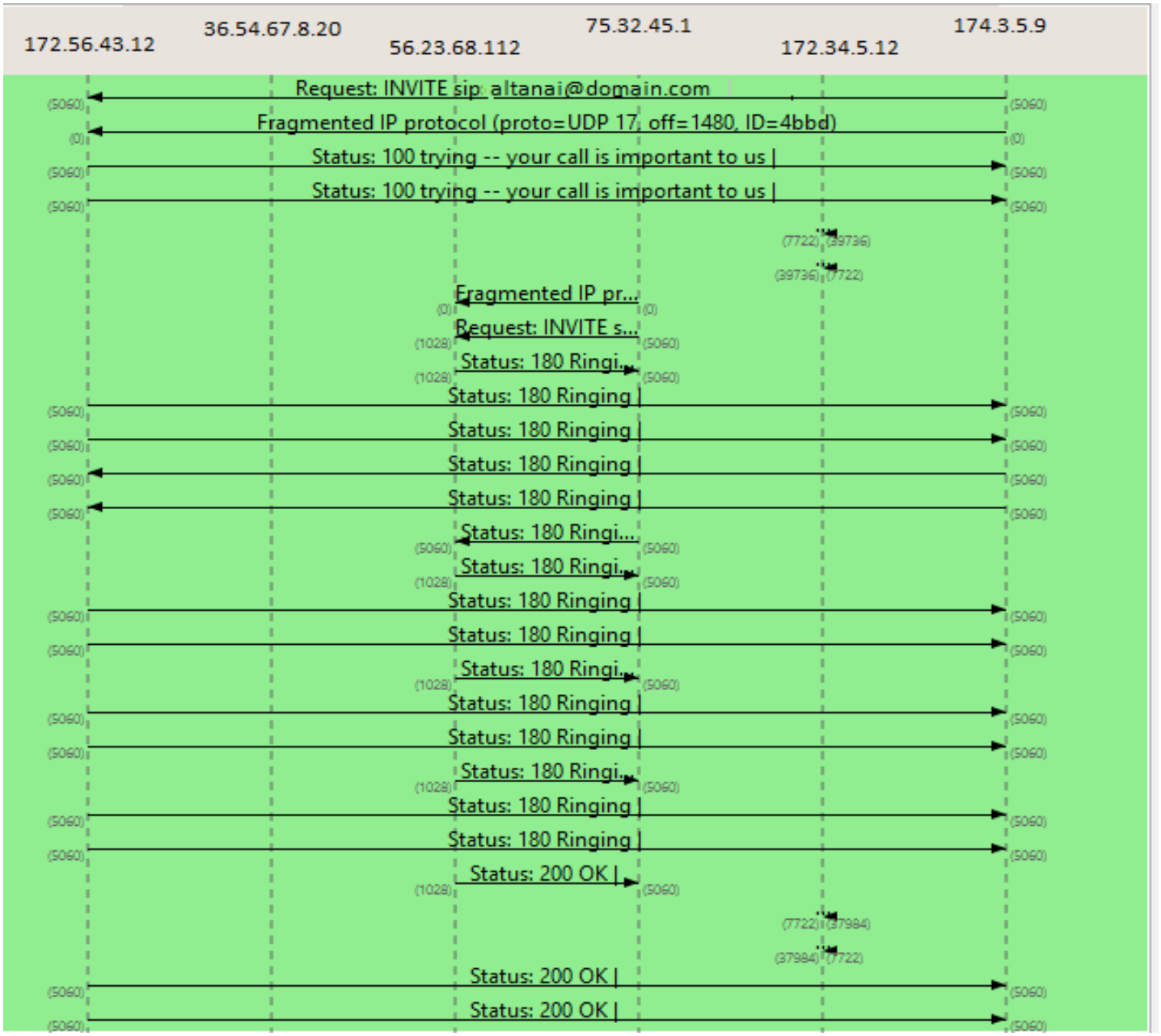
```
altanai@tcs:/usr/sbin$ ls
aa-status               e2freefrag              named-compilezone       select-default-ispell
accept                  e4defrag                named-journalprint      select-default-wordlist
accessdb                fdformat                netscsid                sendmail
acpid                   filefrag                NetworkManager          service
addgroup                gconf-schemas           newusers                setvesablank
add-shell               genrandom               node                    smtp-sink
adduser                 gnome-menus-blacklist   nodeusers               smtp-source
alsactl                 groupadd                nologin                 split-logfile
anacron                 groupdel                nsec3hash               tarcat
apparmor_status         groupmod                ntpdate                 tcpd
aptd                    grpck                   ntpdate-debian          tcpdchk
arp                     grpconv                 ownership               tcpdmatch
arpaname                grpunconv               pam-auth-update         tcpdump
arpd                    grub-bios-setup         pam_getenv              toshsat1800-irdasetup
aspell-autobuildhash    grub-install            pam_timestamp_check     try-from
avahi-autoipd           grub-mkconfig           paperconfig             tunelp
avahi-daemon            grub-mkdevicemap        pm-hibernate            tzconfig
avivotool               grub-mknetdir           pm-powersave            ufw
ax25-node               grub-probe              pm-suspend              unity-greeter
bccmd                   grub-reboot             pm-suspend-hybrid       update-alternatives
biosdecode              grub-set-default        popcon-largest-unused   update-apt-xapian-index
bluetoothd              guest-account           popularity-contest      update-binfmts
chat                    hciattach               postalias               update-ca-certificates
check_forensic          hciconfig               postcat                 update-catalog
checkgid                hciemu                  postconf                update-cracklib
chgpasswd               hplj1000                postdrop                update-default-aspell
chpasswd                hplj1005                postfix                 update-default-ispell
chroot                  hplj1018                postfix-add-filter      update-default-wordlist
ck-log-system-restart   hplj1020                postfix-add-policy      update-dictcommon-aspell
ck-log-system-start     hpljP1005               postkick                update-dictcommon-hunspell
ck-log-system-stop      hpljP1006               postlock                update-fonts-alias
console-kit-daemon      hpljP1007               postlog                 update-fonts-dir
cpgr                    hpljP1008               postmap                 update-fonts-scale
cppw                    hpljP1505               postmulti               update-grub
cracklib-check          htcacheclean            postqueue               update-grub2
cracklib-format         httxt2dbm               postsuper               update-grub-gfxpayload
cracklib-packer         iconvconfig             pppconfig               update-gsfontmap
cracklib-unpacker       inetd                   pppd                    update-icon-caches
```

```
altanai@tcs:/usr/local/sbin$ sudo kamailio start
[sudo] password for altanai:
loading modules under /usr/local/lib64/kamailio/modules_k/:/usr/local/lib64/kamailio/modules/
 0(15261) INFO: tls [tls_init.c:385]: init_tls_compression(): tls: init_tls: disabling compression...
Listening on
             udp: 14.96.130.114:5060
             udp: 14.96.130.114:6060
             tcp: 14.96.130.114:5060
             tcp: 14.96.130.114:6060
             tcp: 14.96.130.114:443
Aliases:
             tcp: tcs.com:443
             tcp: tcs.com:6060
             tcp: tcs.com:5060
             udp: tcs.com:6060
             udp: tcs.com:5060
             *: 14.96.130.114:*

 0(15261) INFO: <core> [tcp_main.c:4836]: init_tcp(): init_tcp: using epoll_lt as the io watch method (auto
 detected)
 0(15263) INFO: rr [../outbound/api.h:54]: ob_load_api(): Failed to import bind_ob
 0(15263) INFO: rr [rr_mod.c:159]: mod_init(): outbound module not available
 0(15263) INFO: usrloc [hslot.c:53]: ul_init_locks(): locks array size 512
 0(15263) INFO: cfgutils [cfgutils.c:784]: mod_init(): no hash_file given, disable hash functionality
 0(15263) INFO: auth [auth_mod.c:350]: mod_init(): auth: qop set, but nonce-count (nc_enabled) support disa
bled
 0(15263) WARNING: tls [tls_mod.c:273]: mod_init(): WARNING: tls: mod_init: tls support is disabled (set en
able_tls=1 in the config to enable it)
 0(15263) INFO: <core> [udp_server.c:176]: probe_max_receive_buffer(): INFO: udp_init: SO_RCVBUF is initial
ly 212992
 0(15263) INFO: <core> [udp_server.c:227]: probe_max_receive_buffer(): INFO: udp_init: SO_RCVBUF is finally
 425984
 0(15263) INFO: <core> [udp_server.c:176]: probe_max_receive_buffer(): INFO: udp_init: SO_RCVBUF is initial
ly 212992
 0(15263) INFO: <core> [udp_server.c:227]: probe_max_receive_buffer(): INFO: udp_init: SO_RCVBUF is finally
 425984
altanai@tcs:/usr/local/sbin$ 12(15285) INFO: ctl [io_listener.c:225]: io_listen_loop(): io_listen_loop:  us
ing epoll_lt io watch method (config)
```

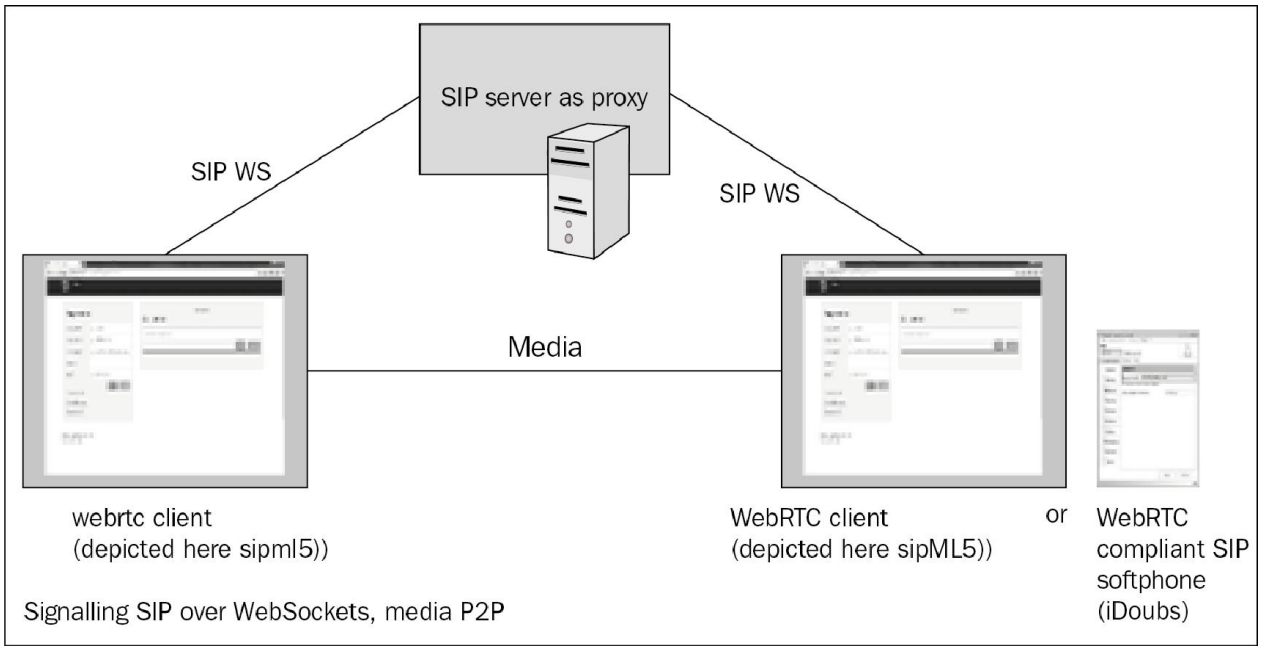
```
altanai@homeMachine:/usr/local/sbin$ ls
kamailio kamcmd kamctl kamdbctl
altanai@homeMachine:/usr/local/sbin$ sudo ./kamctl add alice
alice
new user ‘alice’ added
```


Connected
Registration
Display Name:
altanai
Private Identity*:
altanai
Public Identity*:
sip:altanai@kamailio.org
Password:
Realm*:
kamailio.org
LogIn
LogOut
* Mandatory Field


Application Menu
Users
User Management
Roles
Groups
Modules
Menu
Cronjob
Cache
Help
Theme
Translation
Event Log
Help
Help Center
Go
Module Management
Manage modules in the application. A modules is a component implementing certain business logic.
The module metadata are located at app/modules/module_name folder
Edit
Delete
Load
Go
Id
Name
Description
Active
Author
Version
1 system system module, only administrator can access Rocky 0.1
2 menu menu management module Jixian, Rocky 0.1
3 cache Cache management module Jixian 0.1
4 contact contact management module 0.1
5 cronjob cronjob management module 0.1
6 eventlog Event log module, only administrator can access Rocky 0.1
7 help help management module 0.1
8 ser SIP Admin Modules 4.1
9 sipuser SIP User module 4.0
10 theme Theme management module Jixian 0.1
11 translation UI translation management module Jixian 0.2
12 user usr login, my account edit, password reset Rocky, Jixian 0.1
Go to Page 1
Show Rows 15
1 of 1

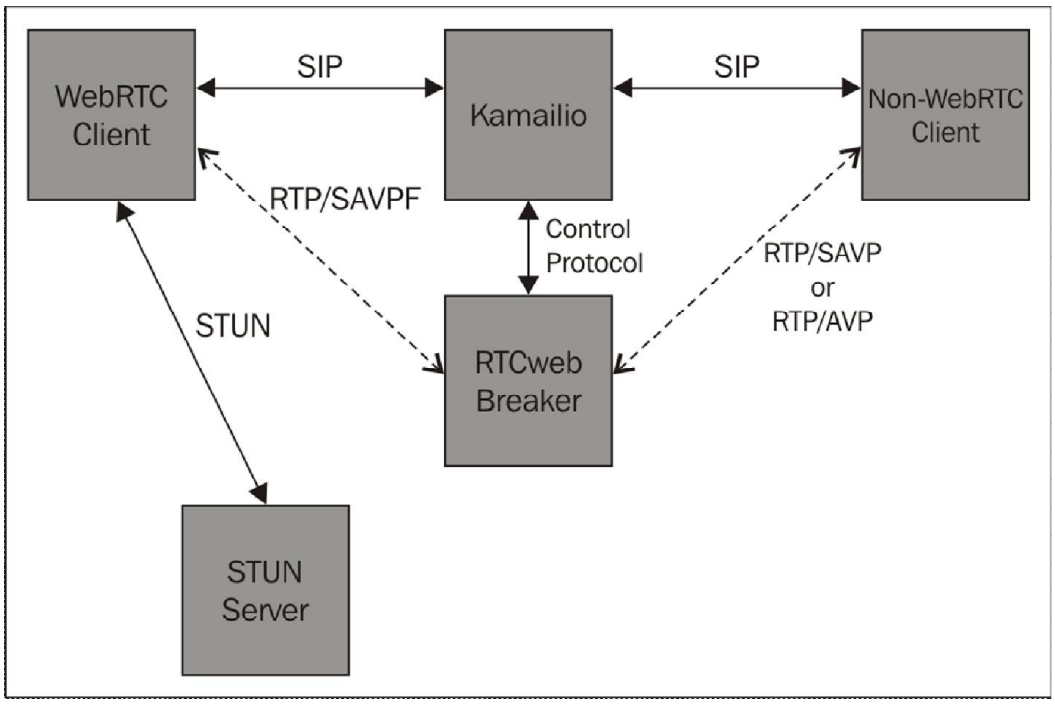

172.56.43.12
36.54.67.8.20
56.23.68.112
75.32.45.1
172.34.5.12
174.3.5.9
Request: INVITE sip: altanai@domain.com
Fragmented IP protocol (proto=UDP 17, off=1480, ID=4bbd)
Status: 100 trying -- your call is important to us |
Status: 100 trying -- your call is important to us |
Fragmented IP pr...
Request: INVITE s...
Status: 180 Ringi...
Status: 180 Ringing |
Status: 180 Ringing |
Status: 180 Ringing |
Status: 180 Ringing |
Status: 180 Ringi...
Status: 180 Ringi...
Status: 180 Ringing |
Status: 180 Ringing |
Status: 180 Ringi...
Status: 180 Ringing |
Status: 180 Ringing |
Status: 180 Ringi...
Status: 180 Ringing |
Status: 180 Ringing |
Status: 200 OK |
Status: 200 OK |
Status: 200 OK |

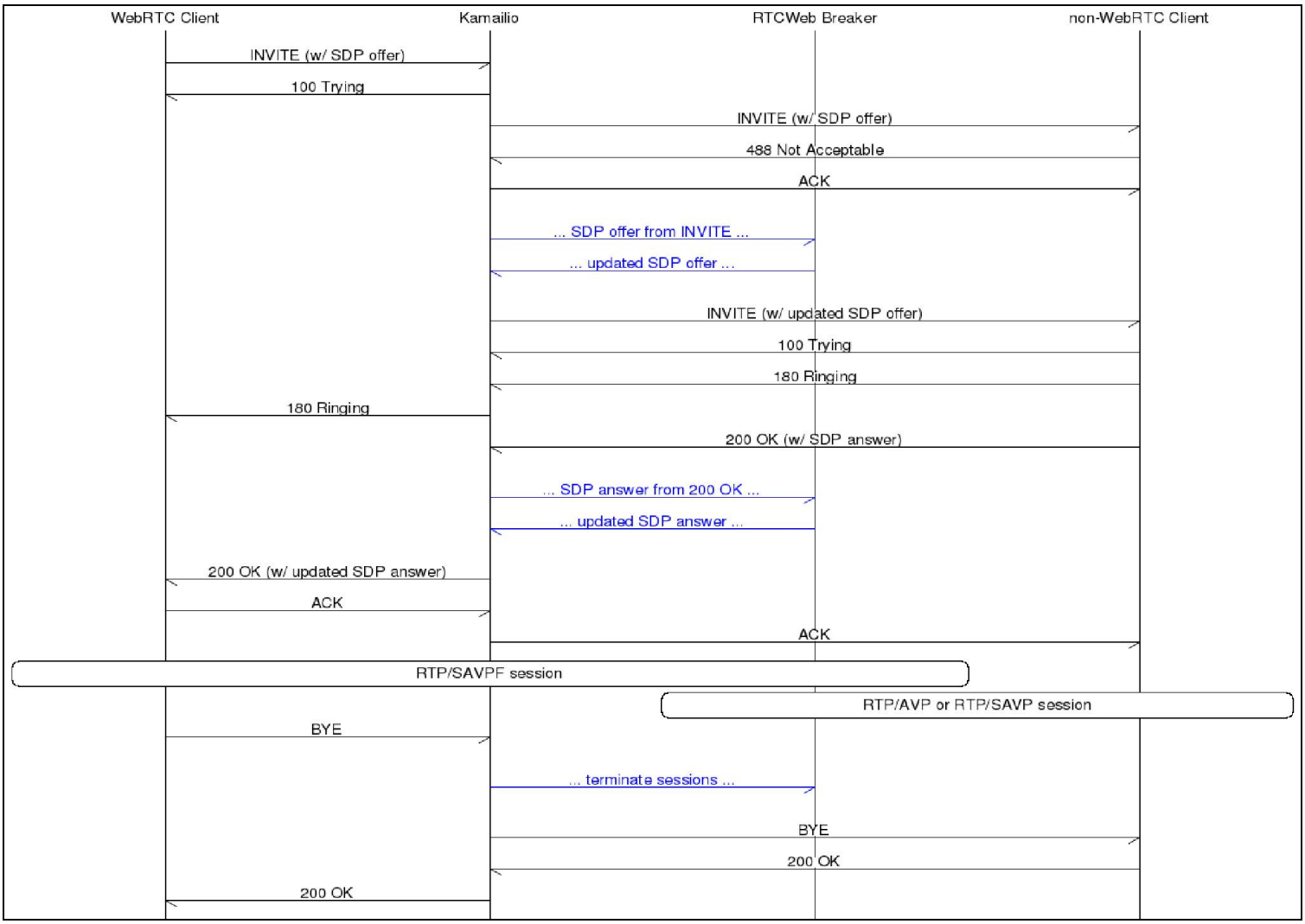
Limitations of existing setup

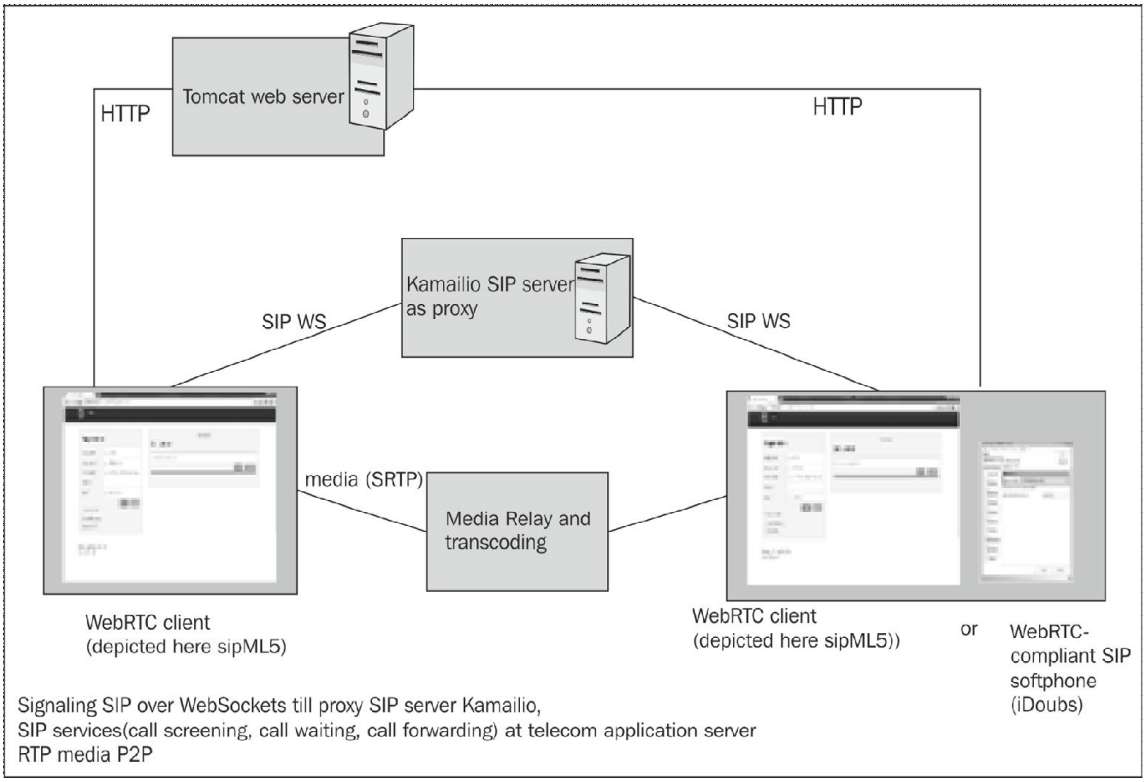


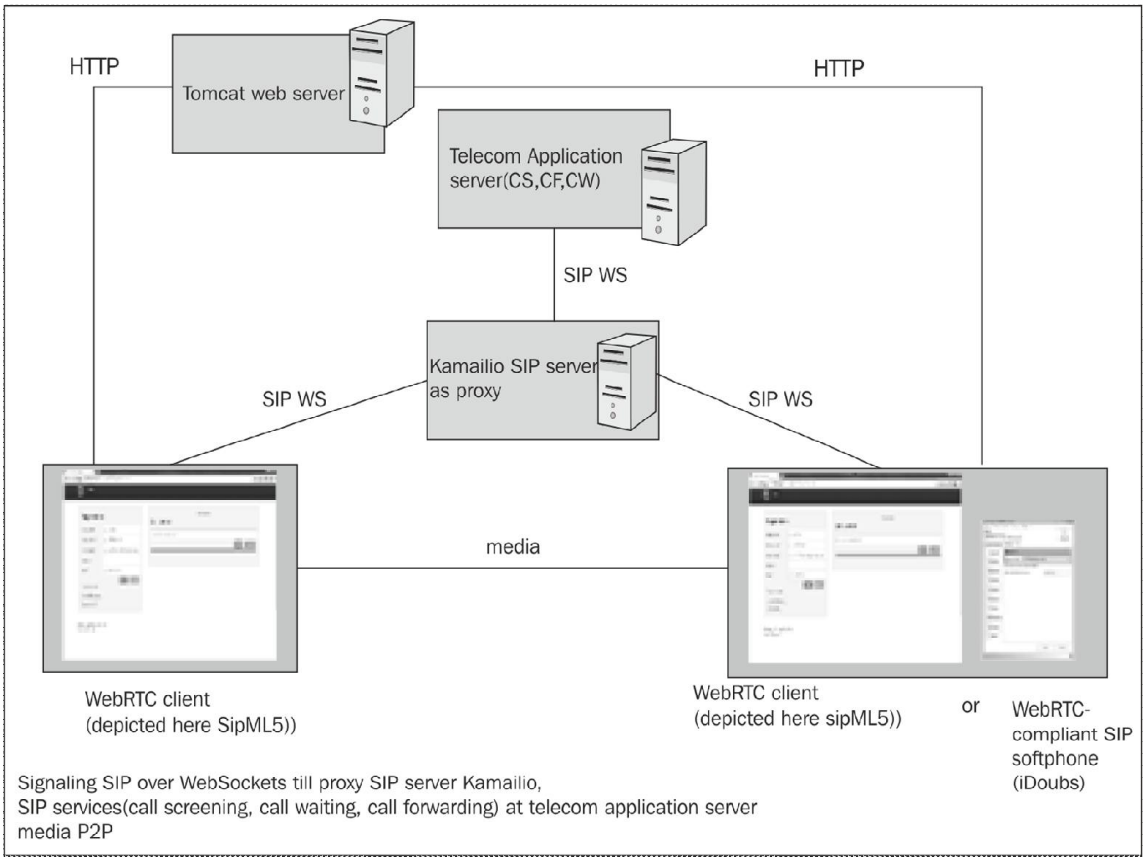
Firewall and NAT issues

Media transcoding



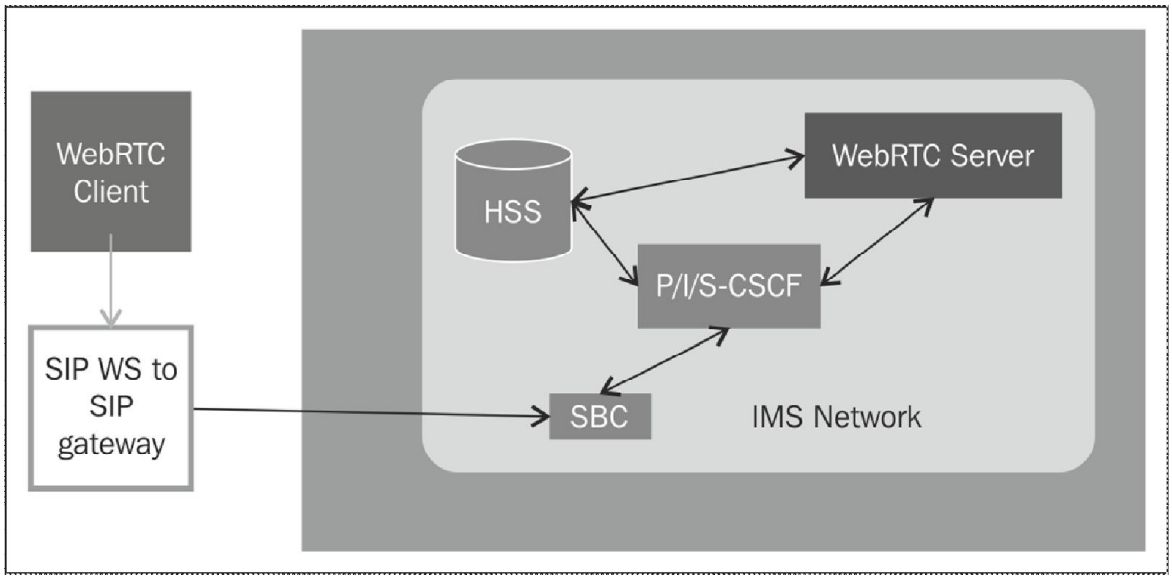

WebRTC Client
Kamailio
RTCWeb Breaker
non-WebRTC Client
INVITE (w/ SDP offer)
100 Trying
INVITE (w/ SDP offer)
488 Not Acceptable
ACK
... SDP offer from INVITE ...
... updated SDP offer ...
INVITE (w/ updated SDP offer)
100 Trying
180 Ringing
180 Ringing
200 OK (w/ SDP answer)
... SDP answer from 200 OK ...
... updated SDP answer ...
200 OK (w/ updated SDP answer)
ACK
ACK
RTP/SAVPF session
RTP/AVP or RTP/SAVP session
BYE
... terminate sessions ...
BYE
200 OK
200 OK

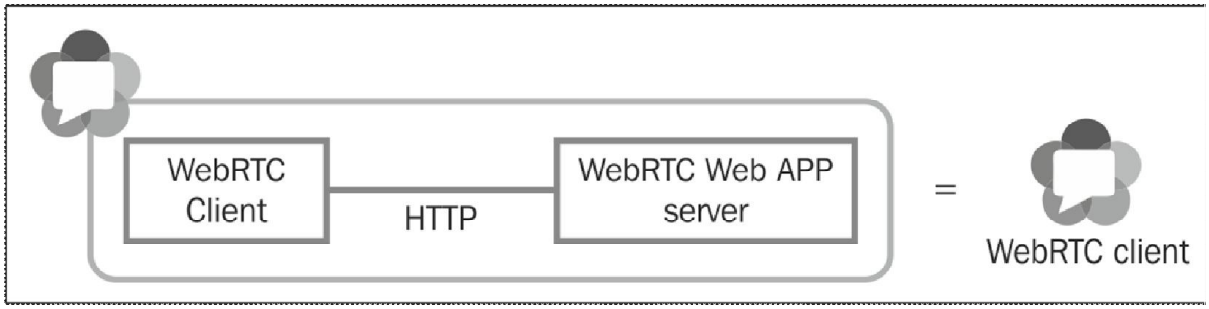

Tomcat web server
HTTP
HTTP
Kamailio SIP server
as proxy
SIP WS
SIP WS
media (SRTP)
Media Relay and
transcoding
WebRTC client
(depicted here sipML5)
WebRTC client
(depicted here sipML5))
or
WebRTC-
compliant SIP
softphone
(iDoubs)
Signaling SIP over WebSockets till proxy SIP server Kamailio,
SIP services(call screening, call waiting, call forwarding) at telecom application server
RTP media P2P


HTTP
Tomcat web server
HTTP
Telecom Application server(CS,CF,CW)
SIP WS
Kamailio SIP server as proxy
SIP WS
SIP WS
media
WebRTC client
(depicted here SipML5))
WebRTC client
(depicted here sipML5))
or
WebRTC-compliant SIP softphone
(iDoubs)
Signaling SIP over WebSockets till proxy SIP server Kamailio,
SIP services(call screening, call waiting, call forwarding) at telecom application server
media P2P

3
WebRTC with SIP and IMS

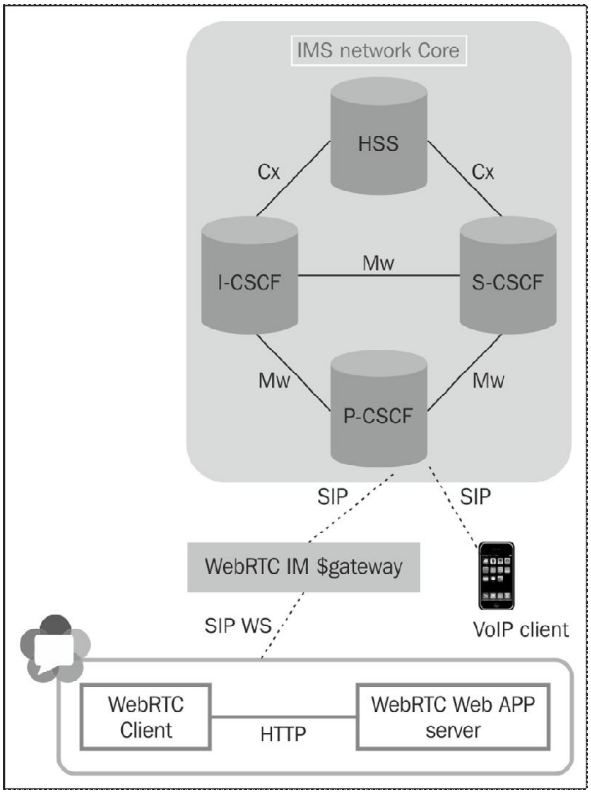
Interaction with core IMS nodes





The IP Multimedia Subsystem core

OpenIMS Core




```
java version "1.7.0_40"
Java(TM) SE Runtime Environment (build 1.7.0_40-b43)
Java HotSpot(TM) 64-Bit Server VM (build 24.0-b56, mixed mode)
```

```
add-imscore-user_newdb.sh  ecscf.sh       icscf.sh        killser   open-ims.dnszone  persist_pg.sql  scscf.xml  tls_prepare.sh
cfg.dox                    FHoSS          icscf.sql       lrf.cfg   pcscf.cfg         README.PCC      ser_ims    trcf.cfg
configurator.sh            fhoss.sh       icscf.thig.cfg  lrf.sh    pcscf.sh          remove_sems.sh  stopser    trcf.sh
dbdump.sh                  icscf.cfg      icscf.thig.sh   mgcf.cfg  pcscf.xml         scscf.cfg       TGPPGq.xml
ecscf.cfg                  icscf_pg.sql   icscf.xml       mgcf.sh   persist_my.sql    scscf.sh        TGPPRx.xml
```

```
altanai@tcs:/opt/OpenIMSCore/ser_ims/cfg$ sudo ./configurator.sh
Domain Name:tcs.com
IP Adress:10.1.5.20
File to change ["all" for everything, "exit" to quit]:all
changing: ecscf.cfg icscf.cfg icscf_pg.sql icscf.sql icscf.thig.cfg icscf.xml lrf.cfg mgcf.cf
st_pg.sql scscf.cfg scscf.xml TGPPGq.xml TGPPRx.xml trcf.cfg
```

```
mysql> show tables
    -> ;
+-------------------------+
| Tables_in_hss_db        |
+-------------------------+
| aliases_repository_data |
| application_server      |
| capabilities_set        |
| capability              |
| charging_info           |
| cx_events               |
| dsai                    |
| dsai_ifc                |
| dsai_impu               |
| ifc                     |
| impi                    |
| impi_impu               |
| impu                    |
| impu_visited_network    |
| imsu                    |
| preferred_scscf_set     |
| repository_data         |
| sh_notification         |
| sh_subscription         |
| shared_ifc_set          |
| sp                      |
| sp_ifc                  |
| sp_shared_ifc_set       |
| spt                     |
| tp                      |
| visited_network         |
| zh_uss                  |
+-------------------------+
27 rows in set (0.00 sec)
```

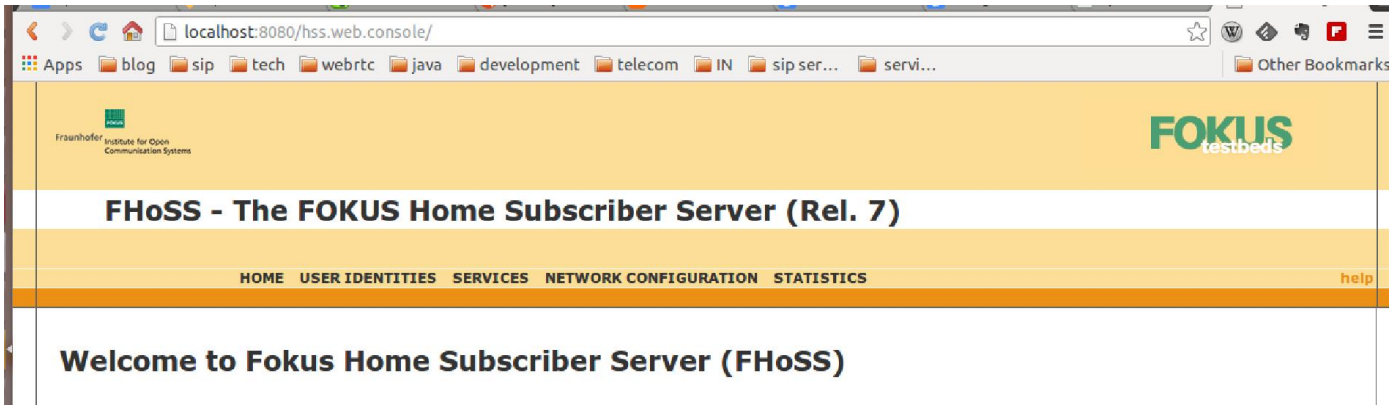
```
mysql> select * from impi;
+----+---------+--------------+-------+-------------+--------------------
---+--------------+-----------------+-----------------------+
| id | id_imsu | identity     | k     | auth_scheme | default_auth_scheme
er | zh_uicc_type | zh_key_life_time | zh_default_auth_scheme |
+----+---------+--------------+-------+-------------+--------------------
---+--------------+-----------------+-----------------------+
|  4 |       1 | alice@tcs.com | alice |         127 |                  1
   |            0 |             3600 |                     1 |
|  2 |       2 | bob@tcs.com   | bob   |         255 |                  1
   |            0 |             3600 |                     1 |
+----+---------+--------------+-------+-------------+--------------------
---+--------------+-----------------+-----------------------+
2 rows in set (0.00 sec)
```

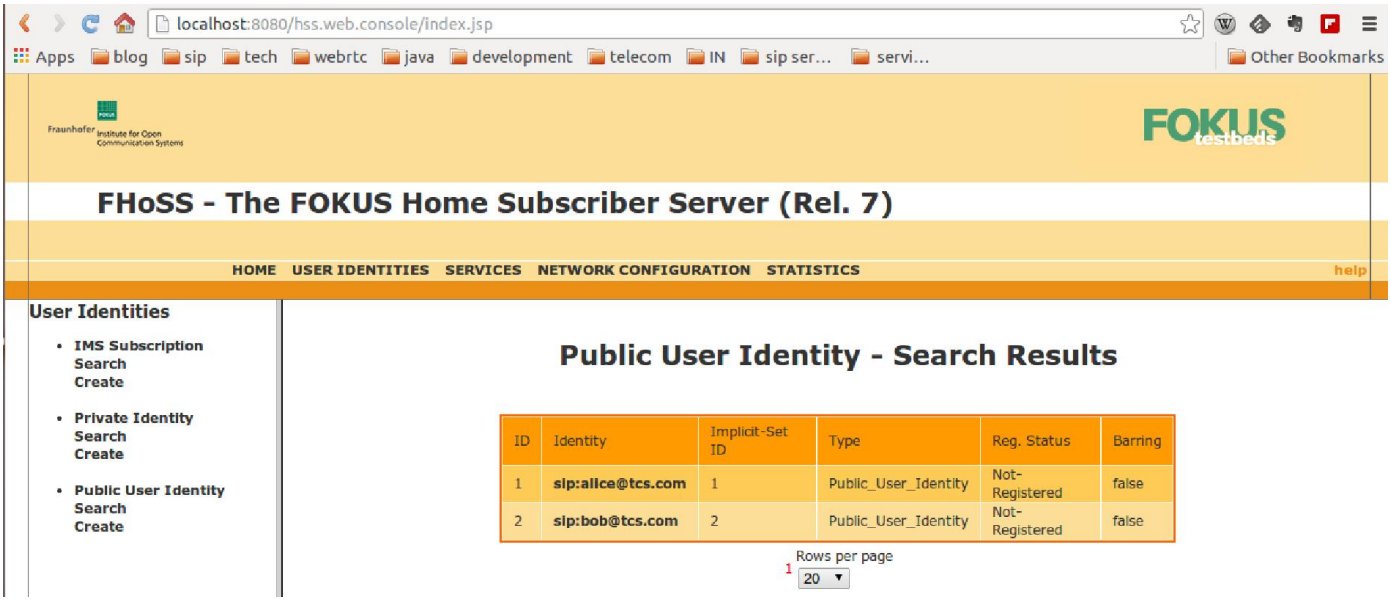
```
./pcscf.sh: line 3: setkey: command not found
Listening on
            udp: 10.1.5.20 [10.1.5.20]:4060
            tcp: 10.1.5.20 [10.1.5.20]:4060
Aliases:
            tcp: tcs.com:4060
            udp: tcs.com:4060
            *: pcscf.tcs.com:4060

 0(14564) init_tcp: using epoll_lt as the io watch method (auto detected)
 0(14564) Maxfwd module- initializing
 0(14564) INFO:P-CSCF:mod_init: Initialization of module
 0(14564) INFOP-CSCF:mod_init: E-CSCF uri is sip:ecscf.tcs.com:7060
 0(14564) DBG:P-CSCF:mod_init: Can not import load_client_rf. This module might
require client_rf module.
 0(14564) DBG:P-CSCF:mod_init: Usage of the charging info in the pcscf module wi
ll be disabled, no charging correlation possible.
 0(14564) INFOP-CSCFmod_init:E-CSCF uri is sip:ecscf.tcs.com:7060
 0(14564) INFO: udp_init: SO_RCVBUF is initially 212992
 0(14564) INFO: udp_init: SO_RCVBUF is finally 425984
 2(14566)  1(14565)  3(14567) INFO:P-CSCF:mod_init: Initialization of module in
child [2]
INFO:P-CSCF:mod_init: Initialization of module in child [3]
 0(14564) INFO:P-CSCF:mod_init: Initialization of module in child [0]
INFO:P-CSCF:mod_init: Initialization of module in child [1]
 4(14568) INFO:P-CSCF:mod_init: Initialization of module in child [4]
 6(14570) 10(14574)  7(14571) INFO:P-CSCF:mod_init: Initialization of module in
child [8]
 8(14572) INFO:P-CSCF:mod_init: Initialization of module in child [6]
 9(14573) 11(14575) INFO:P-CSCF:mod_init: Initialization of module in child [-4]

 5(14569) INFO:P-CSCF:mod_init: Initialization of module in child [7]
INFO:P-CSCF:mod_init: Initialization of module in child [-1]
INFO:P-CSCF:mod_init: Initialization of module in child [5]
INFO:P-CSCF:mod_init: Initialization of module in child [-1]
 5(14569) INF:P-CSCF:----------  Registrar Contents begin --------
 5(14569) INF:P-CSCF:----------  Registrar Contents end ----------
 5(14569) INF:P-CSCF:----------  Subscription list begin ---------
 5(14569) INF:P-CSCF:----------  Subscription list end -----------
 5(14569) INF:P-CSCF:----------  Registrar Contents begin --------
 5(14569) INF:P-CSCF:----------  Registrar Contents end ----------
 5(14569) INF:P-CSCF:----------  Subscription list begin ---------
 5(14569) INF:P-CSCF:----------  Subscription list end -----------
 5(14569) INF:P-CSCF:----------  Registrar Contents begin --------
 5(14569) INF:P-CSCF:----------  Registrar Contents end ----------
 5(14569) INF:P-CSCF:----------  Subscription list begin ---------
 5(14569) INF:P-CSCF:----------  Subscription list end -----------
^Y 5(14569) INF:P-CSCF:----------  Registrar Contents begin --------
 5(14569) INF:P-CSCF:----------  Registrar Contents end ----------
 5(14569) INF:P-CSCF:----------  Subscription list begin ---------
 5(14569) INF:P-CSCF:----------  Subscription list end -----------
```

```
altanai@tcs:/opt/OpenIMSCore$ sudo ./icscf.sh
[sudo] password for altanai:
Listening on
           udp: 10.1.5.20 [10.1.5.20]:5060
           tcp: 10.1.5.20 [10.1.5.20]:5060
Aliases:
           *: tcs.com:*
           *: icscf.tcs.com:*

 0(14729) init_tcp: using epoll_lt as the io watch method (auto detected)
 0(14729) Maxfwd module- initializing
 0(14729) INFO:I-CSCF:mod_init: Initialization of module
 0(14729) INF:I-CSCF:icscf_db_get_nds: Loaded 1 trusted domains
 0(14729) INF:I-CSCF:icscf_db_get_capabilities: Loaded 2 capabilities for 1 S-CSCFs (0 invalid entries in db)
 0(14729) Twofish encryption ready
 0(14729) INFO:cdp:cdp_init(): CDiameterPeer initializing
 0(14729) Diameter Peer Config:
 0(14729)       FQDN    : icscf.tcs.com
 0(14729)       Realm   : tcs.com
 0(14729)       VendorID: 10415
 0(14729)       ProdName: CDiameterPeer
 0(14729)       AcceptUn: [X]
 0(14729)       DropUnkn: [X]
 0(14729)       Tc      : 30
 0(14729)       Workers : 4
 0(14729)       QueueLen: 8
 0(14729)       ConnTime: 5
 0(14729)       TranTime: 5
 0(14729)       SessHash: 128
 0(14729)       DefAuthT: 60
 0(14729)       MaxAuthT: 300
 0(14729)       Peers   : 1
 0(14729)               FQDN:  hss.tcs.com       Realm: tcs.com          Port: 3868
 0(14729)       Acceptors : 1
 0(14729)               Port:  3869      Bind: 10.1.5.20
 0(14729)       Applications : 4
 0(14729)               Auth ID:  16777216       Vendor: 10415
 0(14729)               Auth ID:  16777216       Vendor: 4491
 0(14729)               Auth ID:  16777216       Vendor: 13019
 0(14729)               Auth ID:  16777216       Vendor: 0
 0(14729)       Supported Vendors : 0
 0(14729)       Routing Table :
 0(14729)               DefaultRoute: [  10] hss.tcs.com
 0(14729) INFO: udp_init: SO_RCVBUF is initially 212992
 0(14729) INFO: udp_init: SO_RCVBUF is finally 425984
 1(14731)  2(14732) INFO:I-CSCF:mod_init: Initialization of module in child [2] receiver child=1 sock=10.1.5.20:5060
INFO:I-CSCF:mod_init: Initialization of module in child [1] receiver child=0 sock=10.1.5.20:5060
 3(14733) INFO:I-CSCF:mod_init: Initialization of module in child [3] receiver child=2 sock=10.1.5.20:5060
 4(14734) INFO:I-CSCF:mod_init: Initialization of module in child [4] receiver child=3 sock=10.1.5.20:5060
 0(14729) INFO:I-CSCF:mod_init: Initialization of module in child [0] main
 0(14729) INFO:cdp:cdp_child_init(): CDiameterPeer starting ...
 6(14736) INFO:I-CSCF:mod_init: Initialization of module in child [-1] timer
```

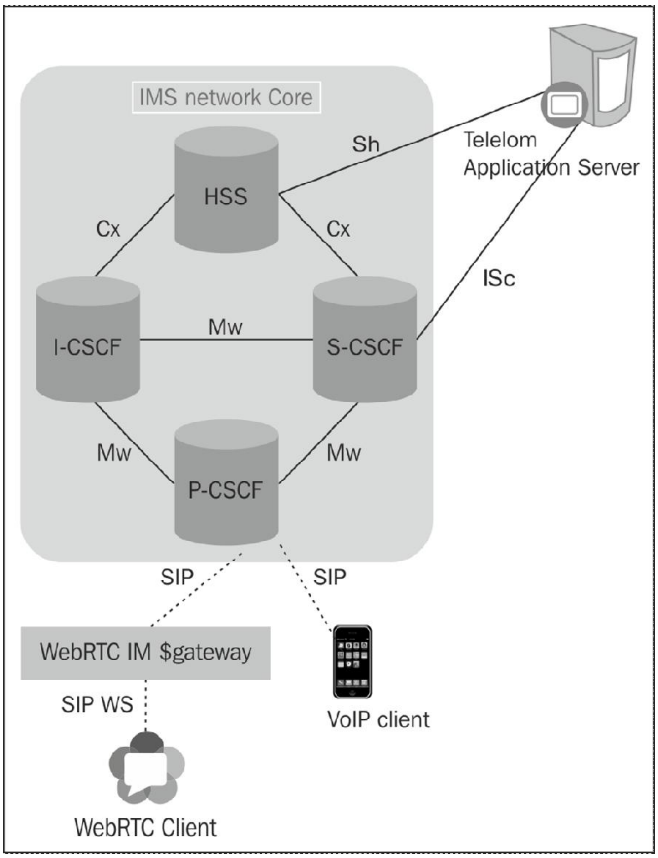
```
altanai@tcs:/opt/OpenIMSCore$ sudo ./scscf.sh
[sudo] password for altanai:
Listening on
            udp: 10.1.5.20 [10.1.5.20]:6060
            tcp: 10.1.5.20 [10.1.5.20]:6060
Aliases:
            tcp: tcs.com:6060
            udp: tcs.com:6060
            *: scscf.tcs.com:6060

 0(14868) init_tcp: using epoll_lt as the io watch method (auto detected)
 0(14868) Maxfwd module- initializing
 0(14868) INFO:S-CSCF:mod_init: Initialization of module
 0(14868) INFO:ISC: - init
 0(14868) INFO:cdp:cdp_init(): CDiameterPeer initializing
 0(14868) Diameter Peer Config:
 0(14868)       FQDN    : scscf.tcs.com
 0(14868)       Realm   : tcs.com
 0(14868)       VendorID: 10415
 0(14868)       ProdName: CDiameterPeer
 0(14868)       AcceptUn: [X]
 0(14868)       DropUnkn: [X]
 0(14868)       Tc      : 30
 0(14868)       Workers : 4
 0(14868)       QueueLen: 8
 0(14868)       ConnTime: 5
 0(14868)       TranTime: 5
 0(14868)       SessHash: 128
 0(14868)       DefAuthT: 60
 0(14868)       MaxAuthT: 300
 0(14868)       Peers   : 1
 0(14868)               FQDN:  hss.tcs.com       Realm: tcs.com          Port: 3868
 0(14868)       Acceptors : 1
 0(14868)               Port:  3870      Bind: 10.1.5.20
 0(14868)       Applications : 4
 0(14868)               Auth ID:  16777216       Vendor: 10415
 0(14868)               Auth ID:  16777216       Vendor: 4491
 0(14868)               Auth ID:  16777216       Vendor: 13019
 0(14868)               Auth ID:  16777216       Vendor: 0
 0(14868)       Supported Vendors : 0
 0(14868)       Routing Table :
 0(14868)               DefaultRoute: [  10] hss.tcs.com
 0(14868) INFO: udp_init: SO_RCVBUF is initially 212992
 0(14868) INFO: udp_init: SO_RCVBUF is finally 425984
 1(14869)  2(14870) INFO:S-CSCF:mod_child_init: Initialization of module in child [1]
INFO:S-CSCF:mod_child_init: Initialization of module in child [2]
 3(14871)  4(14872) INFO:S-CSCF:mod_child_init: Initialization of module in child [4]
 0(14868) INFO:S-CSCF:mod_child_init: Initialization of module in child [0]
 0(14868) INFO:ISC: - child init [0]
 0(14868) INFO:cdp:cdp_child_init(): CDiameterPeer starting ...
 7(14875) INFO:S-CSCF:mod_child_init: Initialization of module in child [1001]
 8(14876) INFO:S-CSCF:mod_child_init: Initialization of module in child [1002]
```

```
altanai@tcs:/opt/OpenIMSCore/FHoSS/deploy$ sudo ./startup.sh
Building Classpath
Classpath is lib/xml-apis.jar:lib/xercesImpl.jar:lib/xerces-2.4.0.jar:lib/xalan-2.4.0.jar:lib/tomcat-util.jar:lib/tomcat-http.jar:lib/tomcat-coyote.jar:lib/struts.jar:lib/servlets-default.jar:lib/servlet-api.jar:lib/naming-resources.jar:lib/naming-factory.jar:lib/mysql-connector-java-3.1.12-bin.jar:lib/mx4j-3.0.1.jar:lib/log4j.jar:lib/junit.jar:lib/junitee.jar:lib/jta.jar:lib/jsp-api.jar:lib/jmx.jar:lib/jdp.jar:lib/jasper-runtime.jar:lib/jasper-compiler-jdt.jar:lib/jasper-compiler.jar:lib/hibernate3.jar:lib/FHoSS.jar:lib/ehcache-1.1.jar:lib/dom4j-1.6.1.jar:lib/commons-validator.jar:lib/commons-modeler.jar:lib/commons-logging.jar:lib/commons-logging-1.0.4.jar:lib/commons-lang.jar:lib/commons-fileupload.jar:lib/commons-el.jar:lib/commons-digester.jar:lib/commons-collections-3.1.jar:lib/commons-beanutils.jar:lib/cglib-2.1.3.jar:lib/catalina-optional.jar:lib/catalina.jar:lib/c3p0-0.9.1.jar:lib/base64.jar:lib/asm.jar:lib/asm-attrs.jar:lib/antlr-2.7.6.jar::log4j.properties:..
2014-04-13 14:50:28,389 INFO  de.fhg.fokus.hss.main.TomcatServer - startTomcat Tomcat-Server is started.
2014-04-13 14:50:29,375 WARN  org.apache.catalina.connector.MapperListener - registerEngine Unknown default host: 127.0.0.1
2014-04-13 14:50:30,547 INFO  de.fhg.fokus.hss.web.servlet.ResponseFilter - init Response Filter Initialisation!
2014-04-13 14:50:30,988 INFO  de.fhg.fokus.hss.main.TomcatServer - startTomcat WebConsole of FHoSS was started !
2014-04-13 14:50:32,753 WARN  org.hibernate.impl.SessionFactoryObjectFactory - addInstance InitialContext did not implement EventContext
2014-04-13 14:50:32,791 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - <init> Bean style constructor called, don't forget to configure!
2014-04-13 14:50:32,795 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - configure FQDN: hss.open-ims.test
2014-04-13 14:50:32,795 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - configure Realm: open-ims.test
2014-04-13 14:50:32,795 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - configure Vendor_ID : 10415
2014-04-13 14:50:32,795 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - configure Product Name: JavaDiameterPeer
2014-04-13 14:50:32,796 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - configure AcceptUnknwonPeers: true
2014-04-13 14:50:32,796 INFO  de.fhg.fokus.diameter.DiameterPeer.DiameterPeer - configure DropUnknownOnDisconnect: true
2014-04-13 14:50:32,802 INFO  de.fhg.fokus.hss.main.HSSContainer - waitForExit
Type "exit" to stop FHoSS!
```



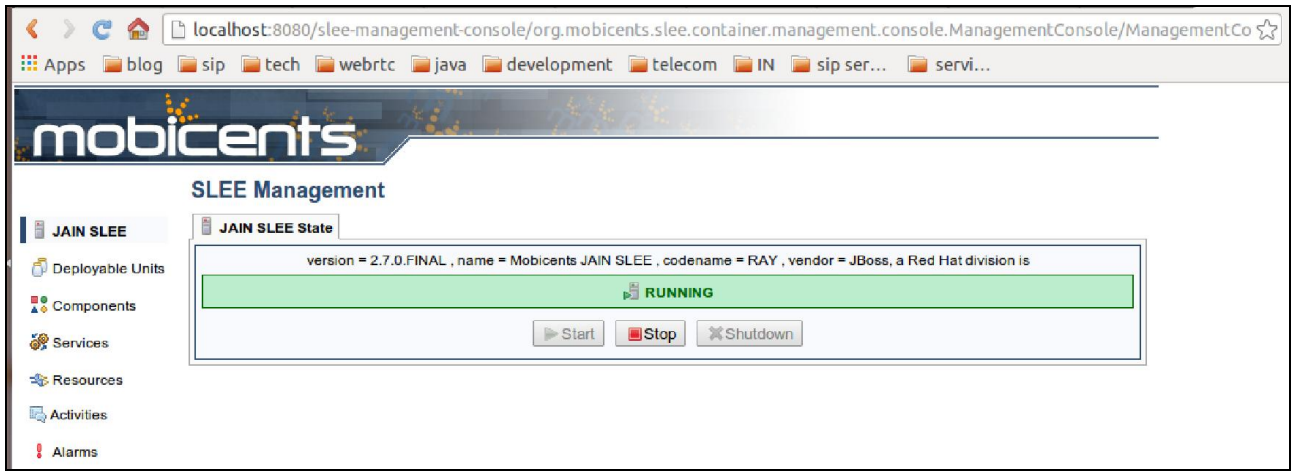

localhost:8080/hss.web.console/index.jsp
Apps
blog
sip
tech
webrtc
java
development
telecom
IN
sip ser...
servi...
Other Bookmarks
Fraunhofer Institute for Open Communication Systems
FOKUS testbeds
FHoSS - The FOKUS Home Subscriber Server (Rel. 7)
HOME
USER IDENTITIES
SERVICES
NETWORK CONFIGURATION
STATISTICS
help
User Identities
IMS Subscription
Search
Create
Private Identity
Search
Create
Public User Identity
Search
Create
Public User Identity - Search Results
ID
Identity
Implicit-Set ID
Type
Reg. Status
Barring
1
sip:alice@tcs.com
1
Public_User_Identity
Not-Registered
false
2
sip:bob@tcs.com
2
Public_User_Identity
Not-Registered
false
Rows per page
1
20

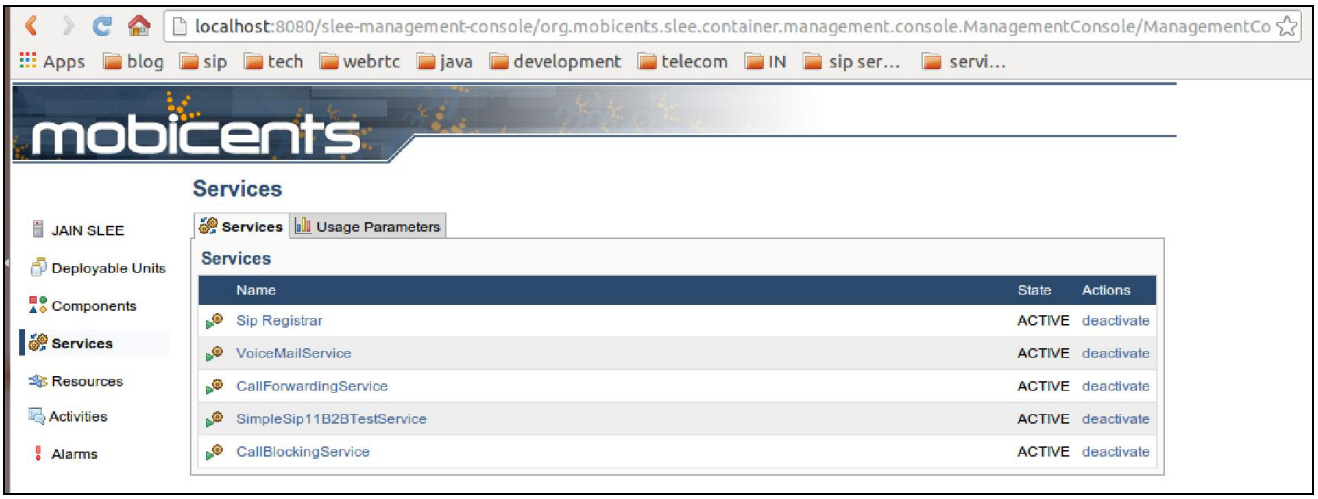
The Telecom server

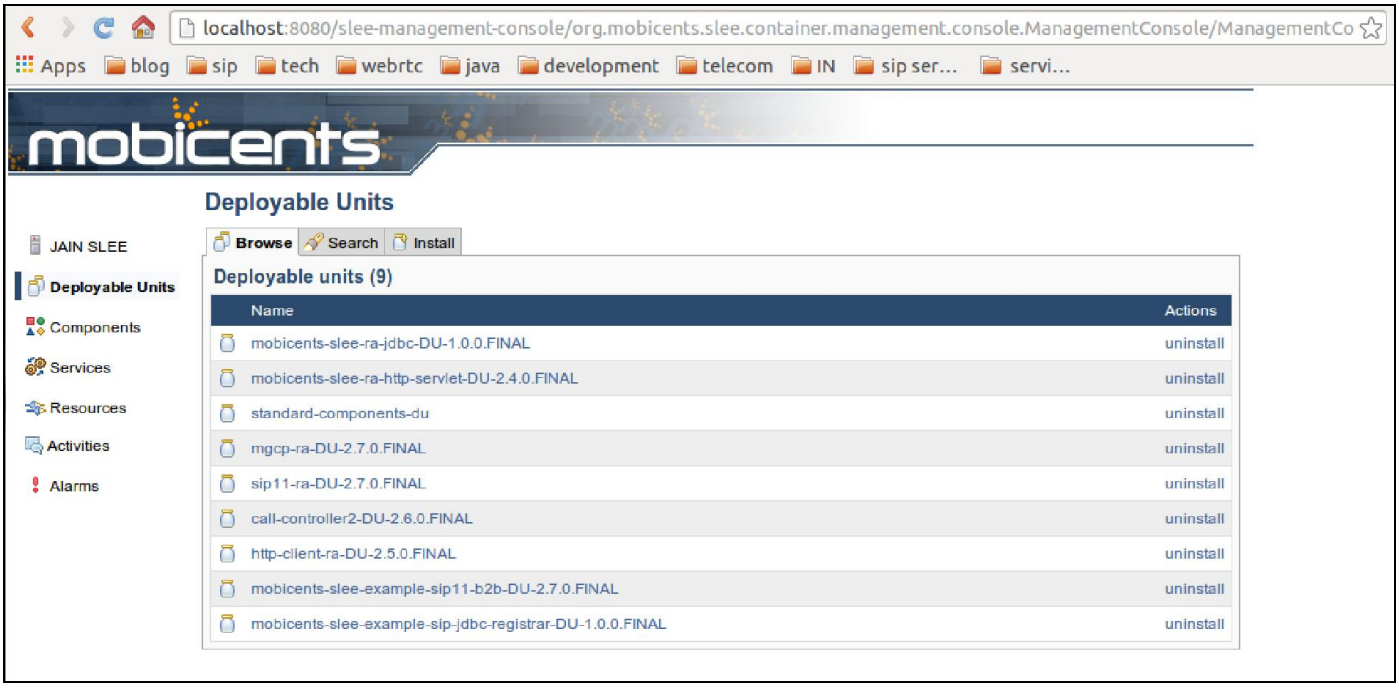


Mobicents Telecom Application Server

```
altanai@tcs:~/mobicents-jainslee-2.7.0.FINAL-jboss-5.1.0.GA/jboss-5.1.0.GA/bin$ sudo ./run.sh
=========================================================================

  JBoss Bootstrap Environment

  JBOSS_HOME: /home/altanai/mobicents-jainslee-2.7.0.FINAL-jboss-5.1.0.GA/jboss-5.1.0.GA

  JAVA: /usr/lib/jvm/java-7-openjdk-amd64/jre/bin/java

  JAVA_OPTS: -Dprogram.name=run.sh -Xms128m -Xmx512m -XX:MaxPermSize=256m -Dorg.jboss.resolver.warning=true -Dsun.rmi.dgc.
rval=3600000 -Dsun.rmi.dgc.server.gcInterval=3600000 -Djava.net.preferIPv4Stack=true

  CLASSPATH: /home/altanai/mobicents-jainslee-2.7.0.FINAL-jboss-5.1.0.GA/jboss-5.1.0.GA/bin/run.jar

=========================================================================

11:46:12,399 INFO  [ServerImpl] Starting JBoss (Microcontainer)...
11:46:12,400 INFO  [ServerImpl] Release ID: JBoss [The Oracle] 5.1.0.GA (build: SVNTag=JBoss_5_1_0_GA date=200905221634)
11:46:12,400 INFO  [ServerImpl] Bootstrap URL: null
11:46:12,401 INFO  [ServerImpl] Home Dir: /home/altanai/mobicents-jainslee-2.7.0.FINAL-jboss-5.1.0.GA/jboss-5.1.0.GA
```

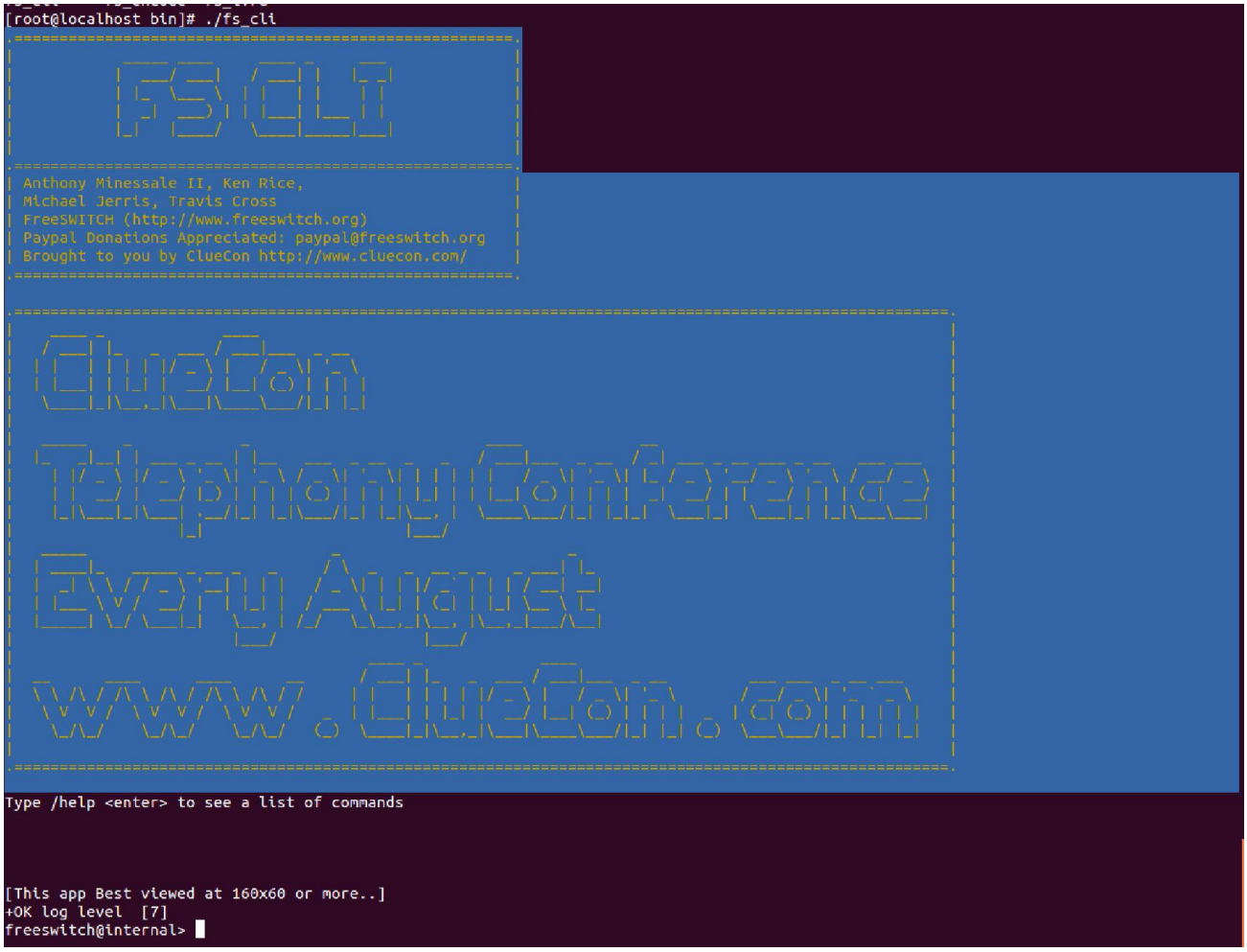



localhost:8080/slee-management-console/org.mobicents.slee.container.management.console.ManagementConsole/ManagementCo
Apps
blog
sip
tech
webrtc
java
development
telecom
IN
sip ser...
servi...
mobicents
Services
JAIN SLEE
Deployable Units
Components
Services
Resources
Activities
Alarms
Services
Usage Parameters
Services
Name
State
Actions
Sip Registrar
ACTIVE
deactivate
VoiceMailService
ACTIVE
deactivate
CallForwardingService
ACTIVE
deactivate
SimpleSip11B2BTestService
ACTIVE
deactivate
CallBlockingService
ACTIVE
deactivate

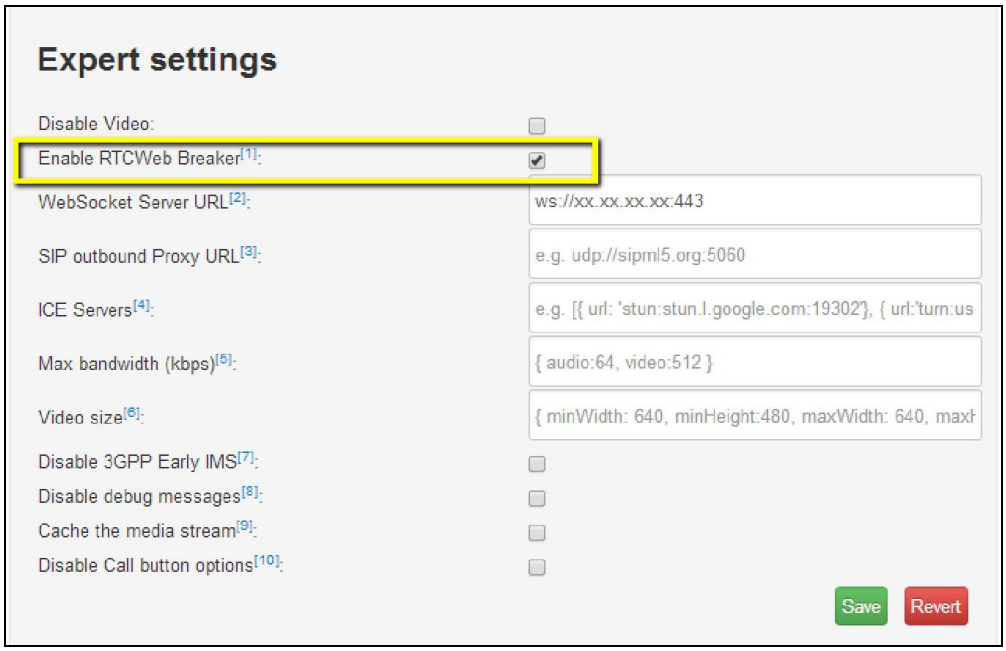

localhost:8080/slee-management-console/org.mobicents.slee.container.management.console.ManagementConsole/ManagementCo
Apps
blog
sip
tech
webrtc
java
development
telecom
IN
sip ser...
servi...
mobicents
Deployable Units
JAIN SLEE
Deployable Units
Components
Services
Resources
Activities
Alarms
Browse
Search
Install
Deployable units (9)
Name
Actions
mobicents-slee-ra-jdbc-DU-1.0.0.FINAL
uninstall
mobicents-slee-ra-http-servlet-DU-2.4.0.FINAL
uninstall
standard-components-du
uninstall
mgcp-ra-DU-2.7.0.FINAL
uninstall
sip11-ra-DU-2.7.0.FINAL
uninstall
call-controller2-DU-2.6.0.FINAL
uninstall
http-client-ra-DU-2.5.0.FINAL
uninstall
mobicents-slee-example-sip11-b2b-DU-2.7.0.FINAL
uninstall
mobicents-slee-example-sip-jdbc-registrar-DU-1.0.0.FINAL
uninstall

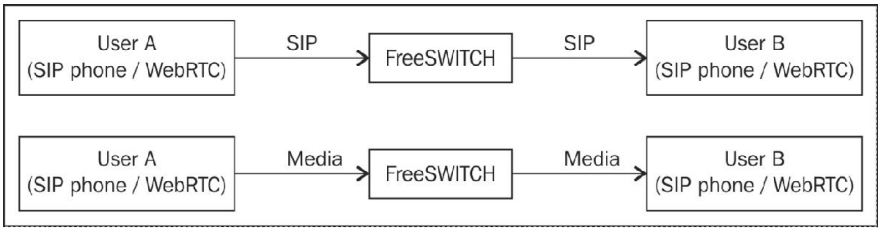
The Media Server

The FreeSWITCH Media Server

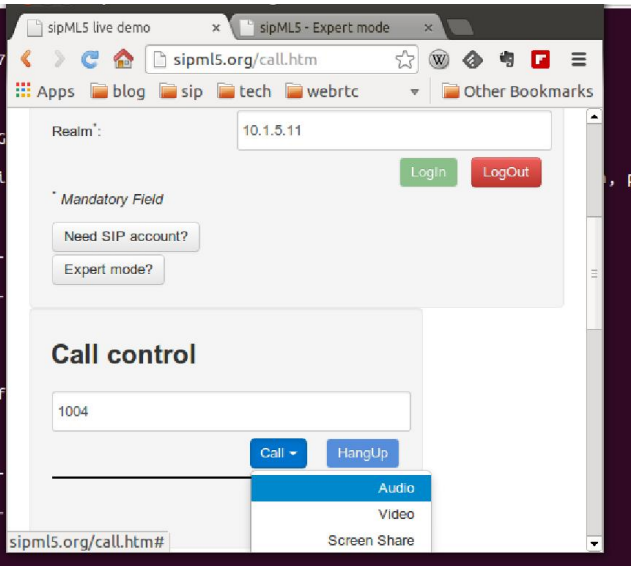


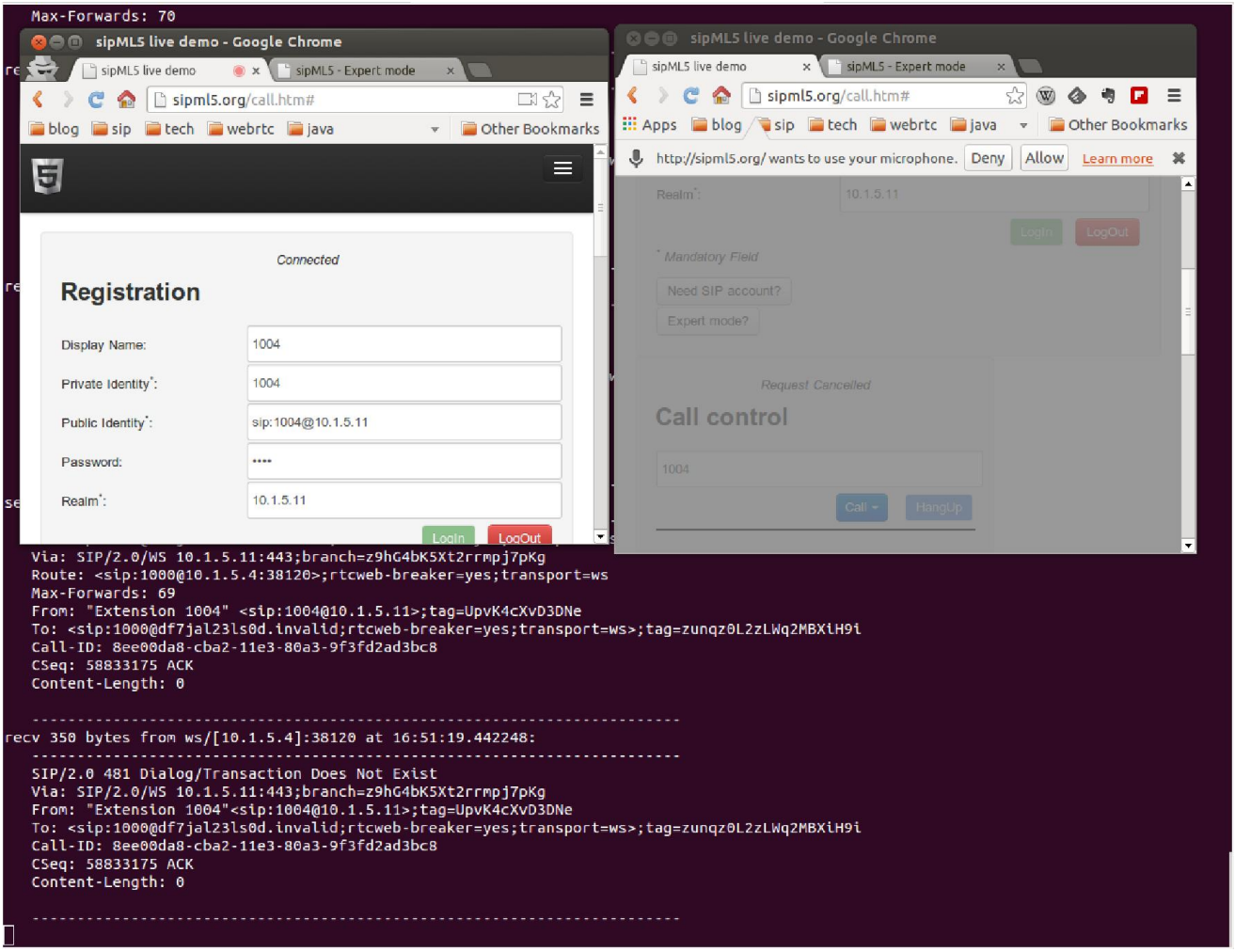
```
freeswitch@internal> sofia status profile sipinterface_1 reg

Registrations:
=================================================================================================
Call-ID:        b9f521ab-f06c-39a0-25a7-7d5e924d837b
User:           1018@10.1.5.2
Contact:        "1018" <sip:1018@df7jal23ls0d.invalid;rtcweb-breaker=yes;transport=ws;fs_nat=yes;fs_path=sip%3A1018%4010.1.5.11%3A4950
%3Brtcweb-breaker%3Dyes%3Btransport%3Dws>
Agent:          IM-client/OMA1.0 sipML5-v1.2014.04.18
Status:         Registered(WS-NAT)(unknown) EXP(2014-04-29 18:16:45) EXPSECS(173)
Host:           tcs.com
IP:             10.1.5.11
Port:           49502
Auth-User:      1018
Auth-Realm:     10.1.5.2
MWI-Account:    1018@10.1.5.2

Call-ID:        b91b5167-3b61-a1d6-a2cf-d74c5095ea63
User:           1004@10.1.5.2
Contact:        "1004" <sip:1004@df7jal23ls0d.invalid;rtcweb-breaker=yes;transport=ws;fs_nat=yes;fs_path=sip%3A1004%4010.1.5.10%3A3571
%3Brtcweb-breaker%3Dyes%3Btransport%3Dws>
Agent:          IM-client/OMA1.0 sipML5-v1.2014.04.18
Status:         Registered(WS-NAT)(unknown) EXP(2014-04-29 18:18:09) EXPSECS(257)
Host:           tcs.com
IP:             10.1.5.10
Port:           35715
Auth-User:      1004
Auth-Realm:     10.1.5.2
MWI-Account:    1004@10.1.5.2

Total items returned: 2
=================================================================================================
```

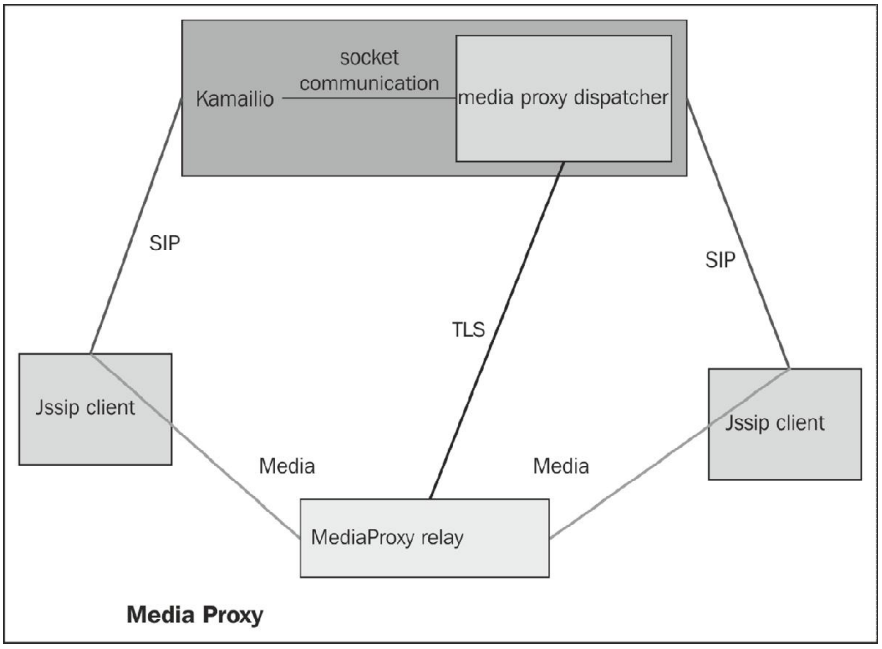


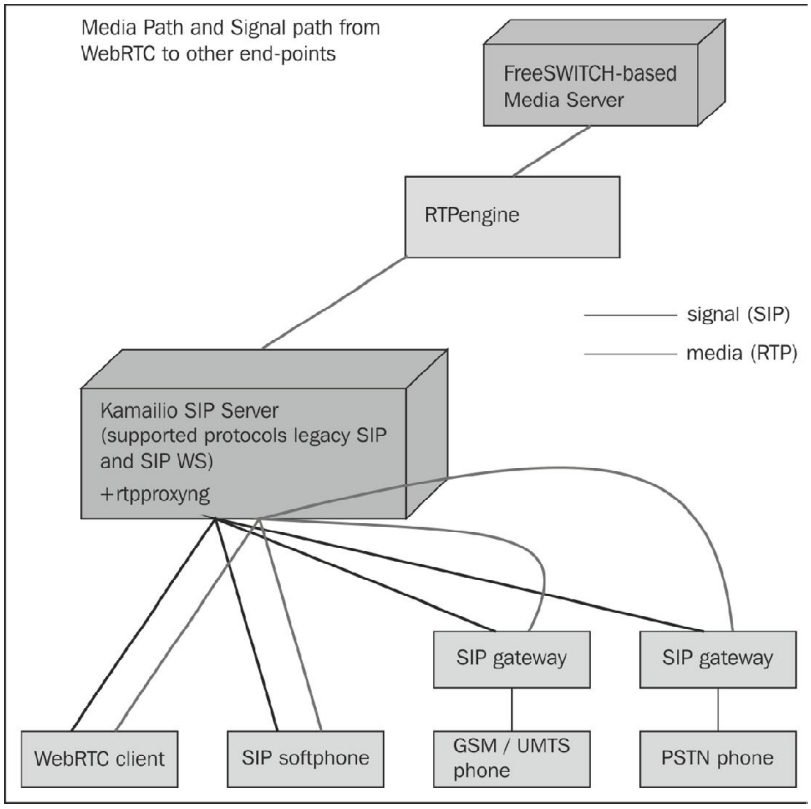


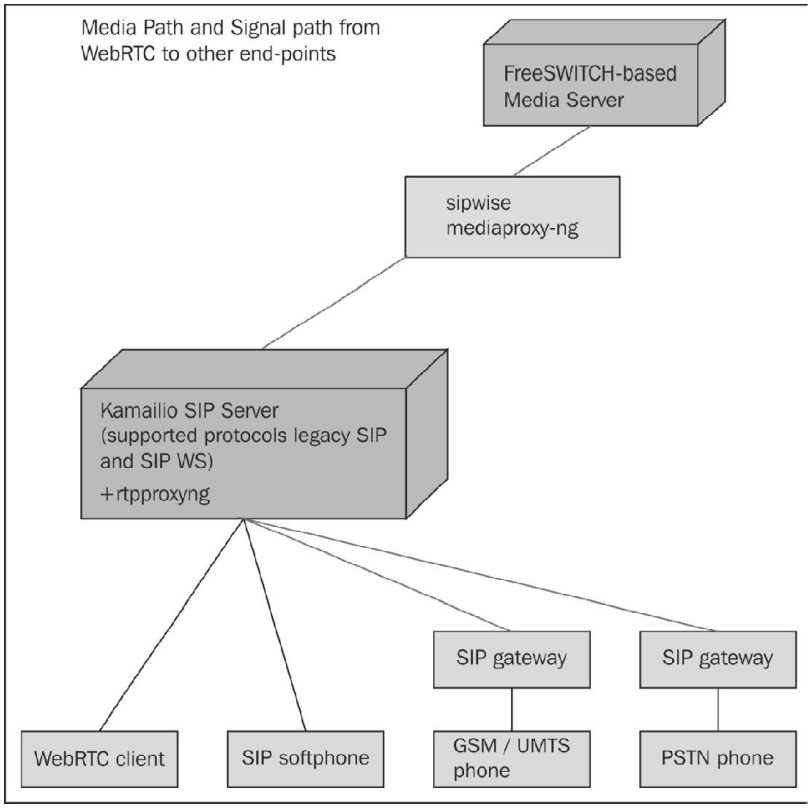

```
   From: <sip:mod_sofia@10.1.5.11:9999>;tag=cBXt6Fm4a9U3N
   To: <sip:1019@10.1.5.11>
   Call-ID: aa39e060-cb9d-11e3-bd65-9f3fd2ad3bc8_58f18dc2-2930-578b-497
   CSeq: 58824950 OPTIONS
   Contact: <sip:mod_sofia@10.1.5.11:9999>
   User-Agent: The 2600hz Project
   Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REG
   Supported: path, replaces
   Allow-Events: talk, hold, conference, presence, as-feature-event, di
resence.winfo, message-summary, refer
   Content-Length: 0

   ------------------------------------------------------------------
recv 302 bytes from ws/[10.1.5.4]:37579 at 16:16:17.749021:
   ------------------------------------------------------------------
   SIP/2.0 405 Method Not Allowed
   Via: SIP/2.0/WS 10.1.5.11:443;branch=z9hG4bKa8XjQtQNSvBej
   From: <sip:mod_sofia@10.1.5.11:9999>;tag=aSa92SjXgQFye
   To: <sip:1000@10.1.5.11>
   Call-ID: aa39ccba-cb9d-11e3-bd63-9f3fd2ad3bc8_870807cc-58a5-07c7-74f
   CSeq: 58824948 OPTIONS
   Content-Length: 0

   ------------------------------------------------------------------
recv 302 bytes from ws/[10.1.5.4]:37639 at 16:16:17.749395:
   ------------------------------------------------------------------
   SIP/2.0 405 Method Not Allowed
   Via: SIP/2.0/WS 10.1.5.11:443;branch=z9hG4bKBHQBSN8rp510D
```



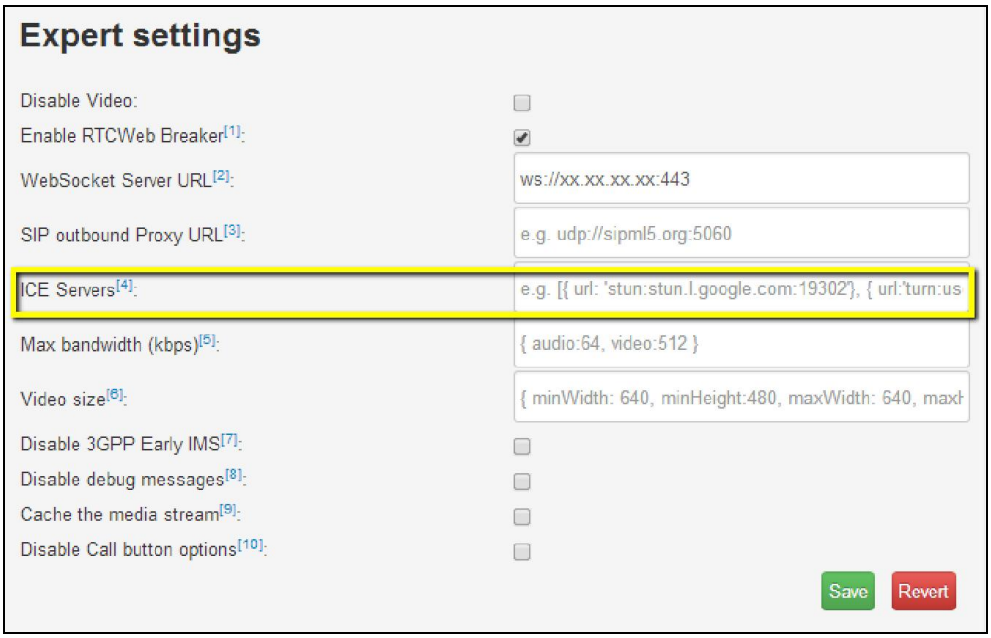

Max-Forwards: 70
sipML5 live demo - Google Chrome
sipML5 live demo
sipML5 - Expert mode
sipml5.org/call.htm#
blog sip tech webrtc java Other Bookmarks
Connected
Registration
Display Name: 1004
Private Identity*: 1004
Public Identity*: sip:1004@10.1.5.11
Password: ••••
Realm*: 10.1.5.11
LogIn
LogOut
sipML5 live demo - Google Chrome
sipML5 live demo
sipML5 - Expert mode
sipml5.org/call.htm#
Apps blog sip tech webrtc java Other Bookmarks
http://sipml5.org/ wants to use your microphone. Deny Allow Learn more
Realm*: 10.1.5.11
LogIn LogOut
* Mandatory Field
Need SIP account?
Expert mode?
Request Cancelled
Call control
1004
Call HangUp
Via: SIP/2.0/WS 10.1.5.11:443;branch=z9hG4bK5Xt2rrmpj7pKg
Route: <sip:1000@10.1.5.4:38120>;rtcweb-breaker=yes;transport=ws
Max-Forwards: 69
From: "Extension 1004" <sip:1004@10.1.5.11>;tag=UpvK4cXvD3DNe
To: <sip:1000@df7jal23ls0d.invalid;rtcweb-breaker=yes;transport=ws>;tag=zunqz0L2zLWq2MBXiH9i
Call-ID: 8ee00da8-cba2-11e3-80a3-9f3fd2ad3bc8
CSeq: 58833175 ACK
Content-Length: 0
recv 350 bytes from ws/[10.1.5.4]:38120 at 16:51:19.442248:
SIP/2.0 481 Dialog/Transaction Does Not Exist
Via: SIP/2.0/WS 10.1.5.11:443;branch=z9hG4bK5Xt2rrmpj7pKg
From: "Extension 1004"<sip:1004@10.1.5.11>;tag=UpvK4cXvD3DNe
To: <sip:1000@df7jal23ls0d.invalid;rtcweb-breaker=yes;transport=ws>;tag=zunqz0L2zLWq2MBXiH9i
Call-ID: 8ee00da8-cba2-11e3-80a3-9f3fd2ad3bc8
CSeq: 58833175 ACK
Content-Length: 0


socket
communication
Kamailio
media proxy dispatcher
SIP
SIP
TLS
Jssip client
Jssip client
Media
Media
MediaProxy relay
Media Proxy

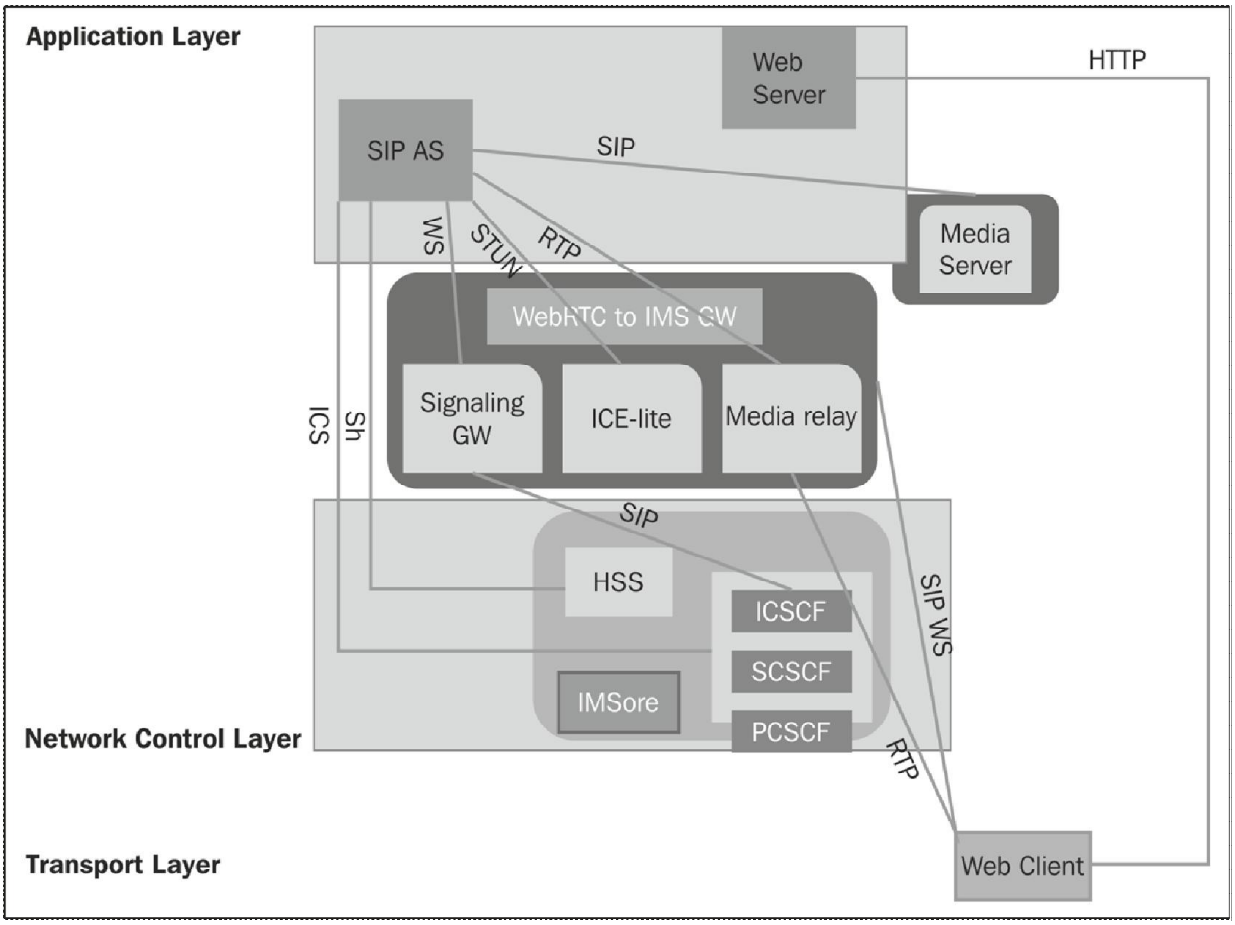

Media Path and Signal path from
WebRTC to other end-points
FreeSWITCH-based
Media Server
RTPengine
signal (SIP)
media (RTP)
Kamailio SIP Server
(supported protocols legacy SIP
and SIP WS)
+rtpproxyng
SIP gateway
SIP gateway
WebRTC client
SIP softphone
GSM / UMTS
phone
PSTN phone


Media Path and Signal path from
WebRTC to other end-points
FreeSWITCH-based
Media Server
sipwise
mediaproxy-ng
Kamailio SIP Server
(supported protocols legacy SIP
and SIP WS)
+rtpproxyng
SIP gateway
SIP gateway
WebRTC client
SIP softphone
GSM / UMTS
phone
PSTN phone

WebRTC over firewalls and proxies


Expert settings
Disable Video:
Enable RTCWeb Breaker[1]:
WebSocket Server URL[2]:
ws://xx.xx.xx.xx:443
SIP outbound Proxy URL[3]:
e.g. udp://sipml5.org:5060
ICE Servers[4]:
e.g. [{ url: 'stun:stun.l.google.com:19302'}, { url:'turn:us
Max bandwidth (kbps)[5]:
{ audio:64, video:512 }
Video size[6]:
{ minWidth: 640, minHeight:480, maxWidth: 640, maxH
Disable 3GPP Early IMS[7]:
Disable debug messages[8]:
Cache the media stream[9]:
Disable Call button options[10]:
Save
Revert

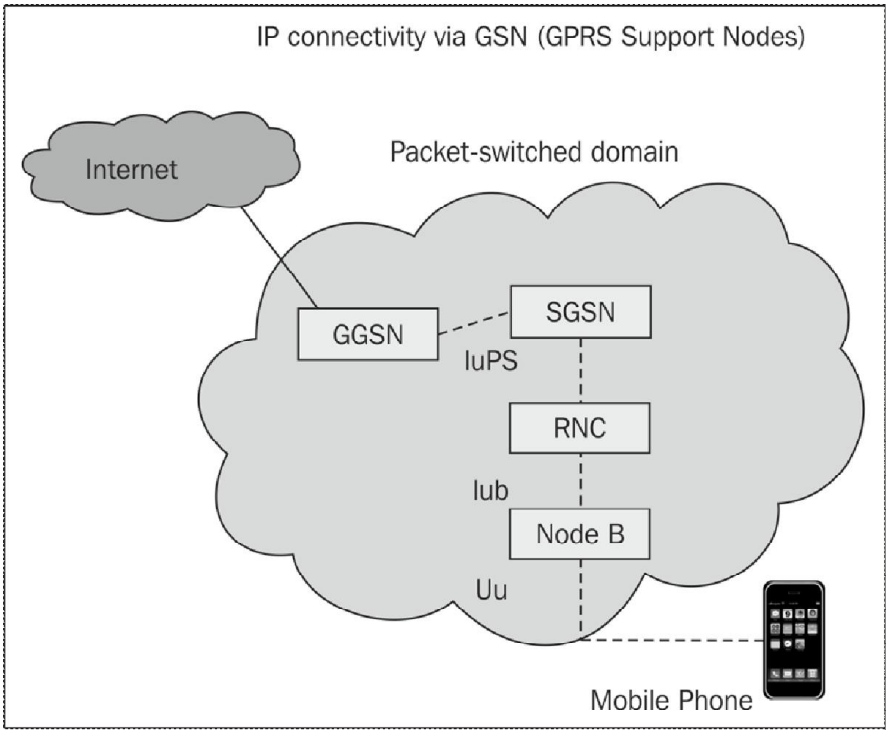
The final architecture for WebRTC to IMS integration

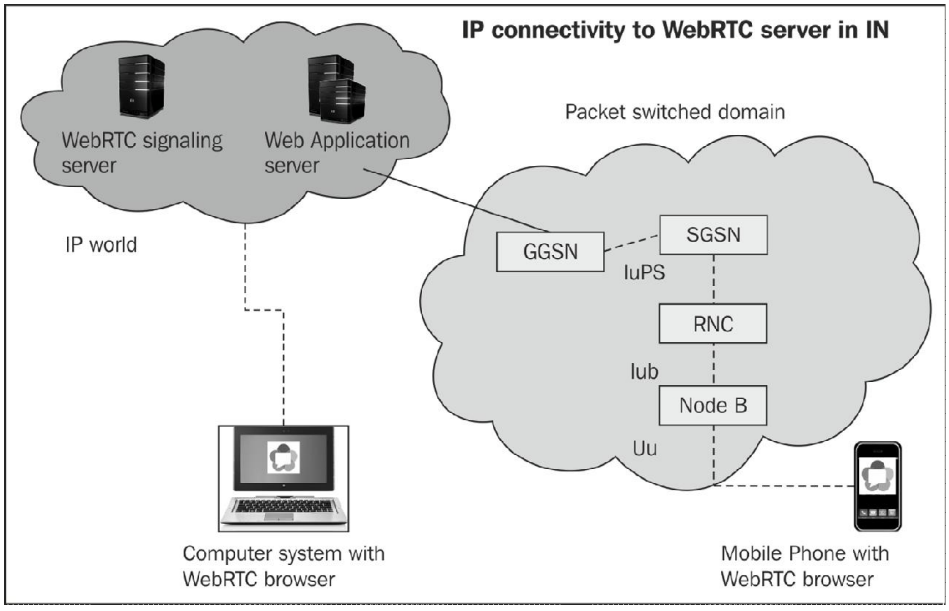


4

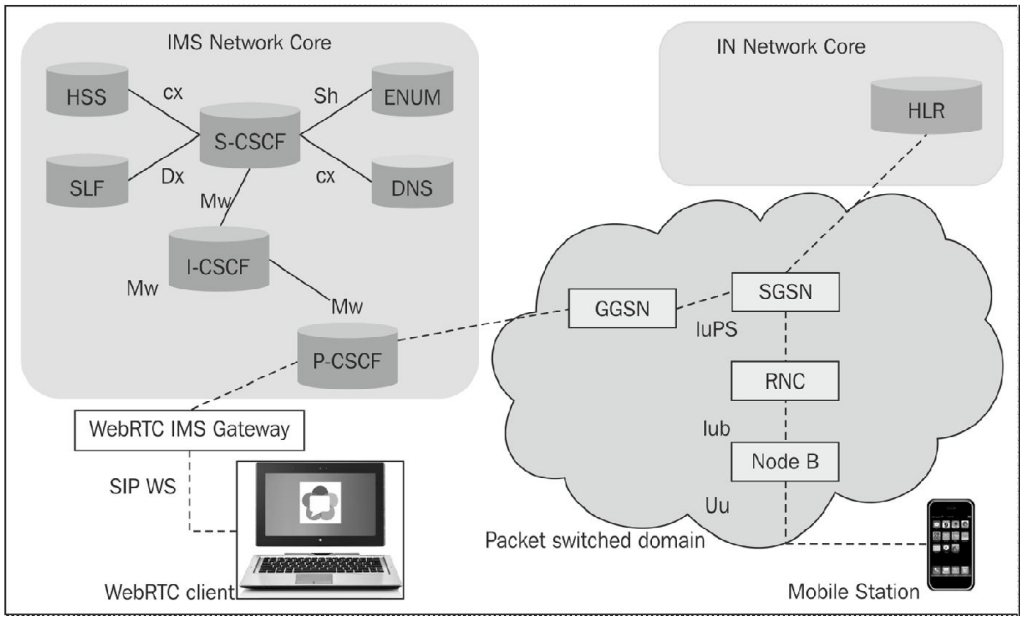
WebRTC Integration with Intelligent Network

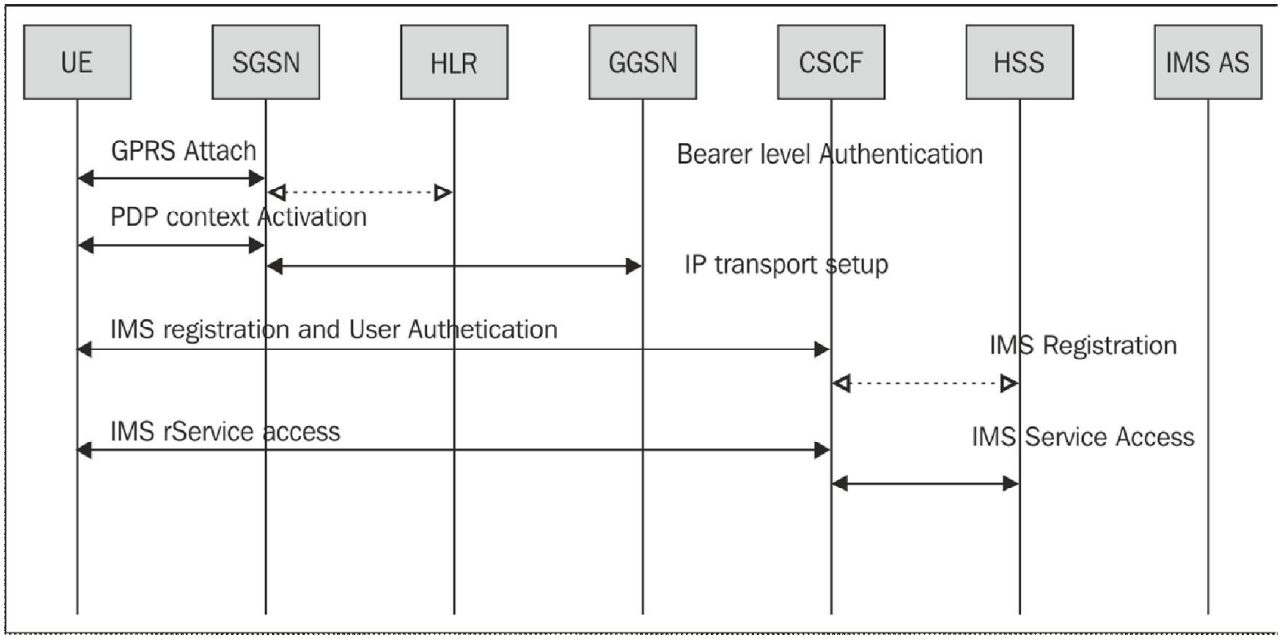
From mobile to WebRTC client through GPRS

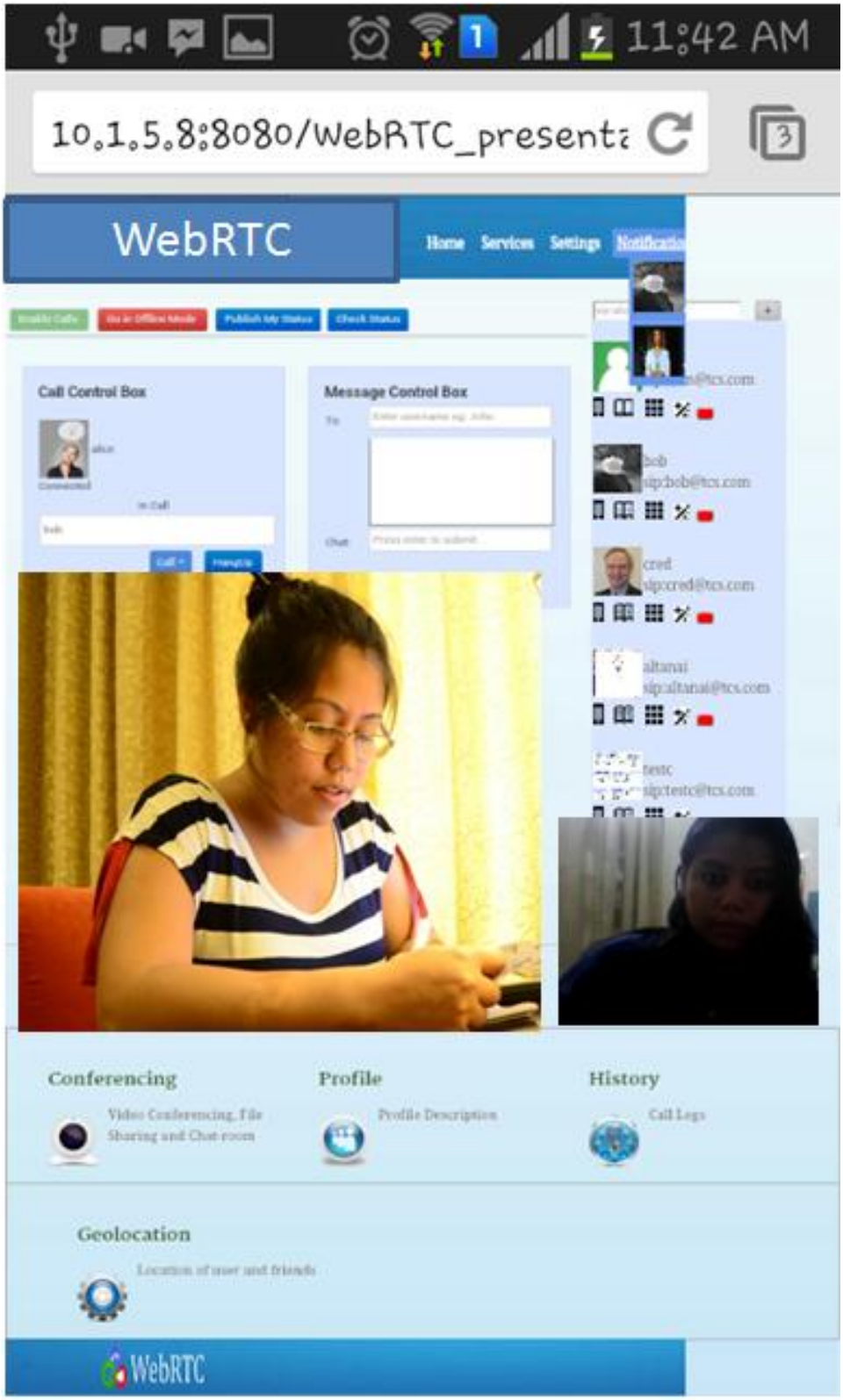




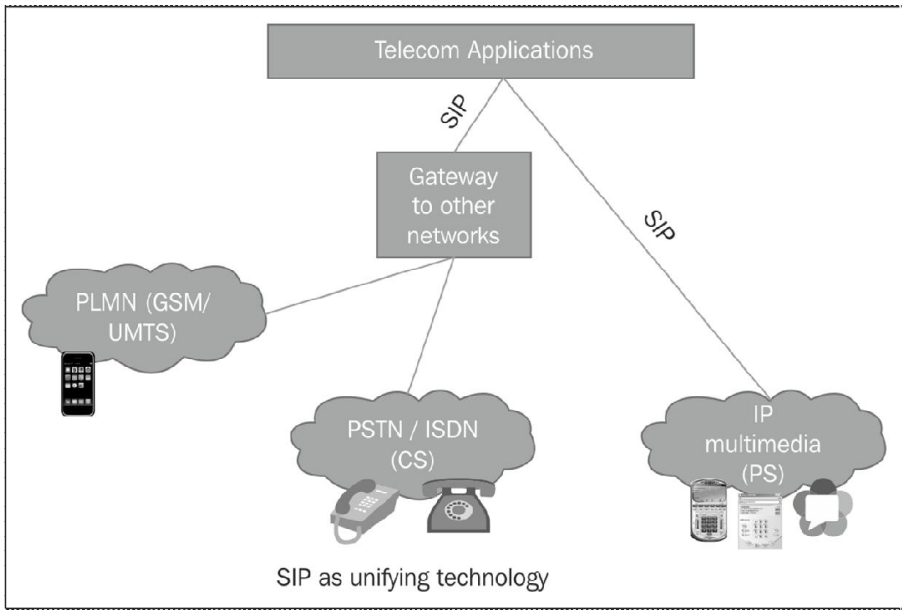
IMS connectivity to Gateway GPRS Support Node



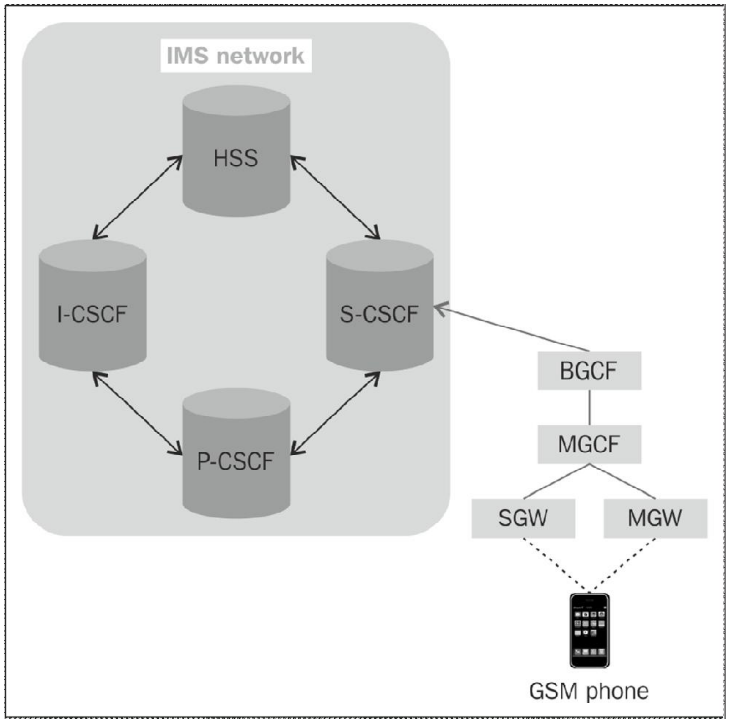

UE
SGSN
HLR
GGSN
CSCF
HSS
IMS AS
GPRS Attach
Bearer level Authentication
PDP context Activation
IP transport setup
IMS registration and User Authetication
IMS Registration
IMS rService access
IMS Service Access


11:42 AM
10.1.5.8:8080/WebRTC_presenta
WebRTC
Home
Services
Settings
Call Control Box
Message Control Box
Conferencing
Video Conferencing, File Sharing and Chat room
Profile
Profile Description
History
Call Logs
Geolocation
Location of user and friends
WebRTC

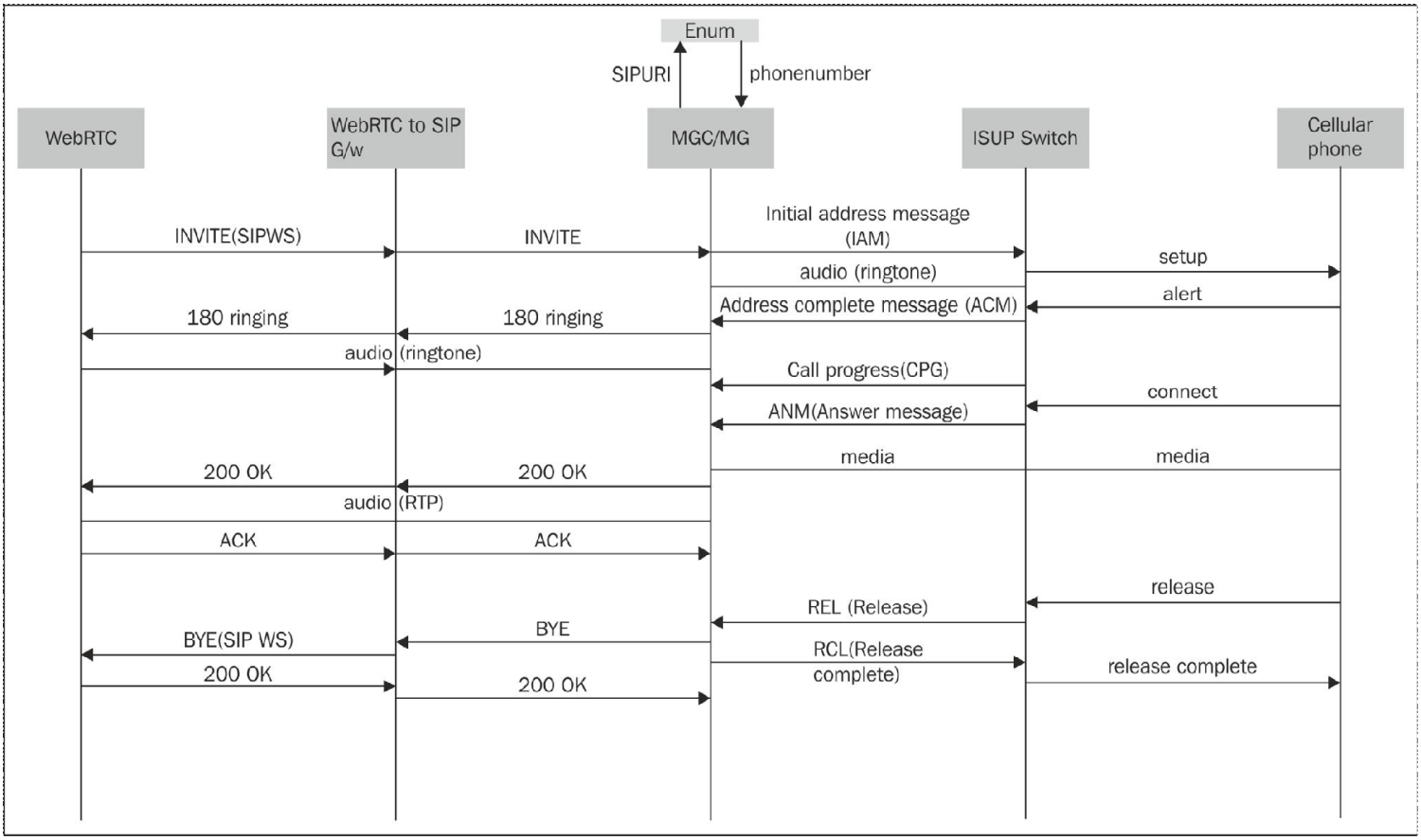
From mobiles to WebRTC client through Voice Network



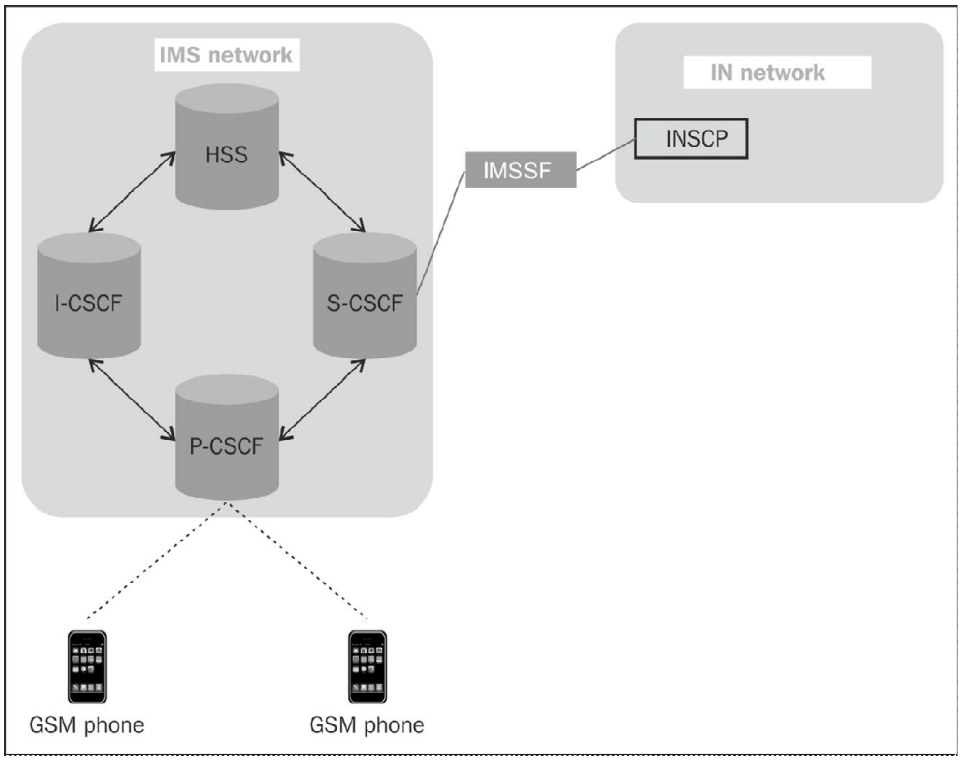
SIP as unifying technology



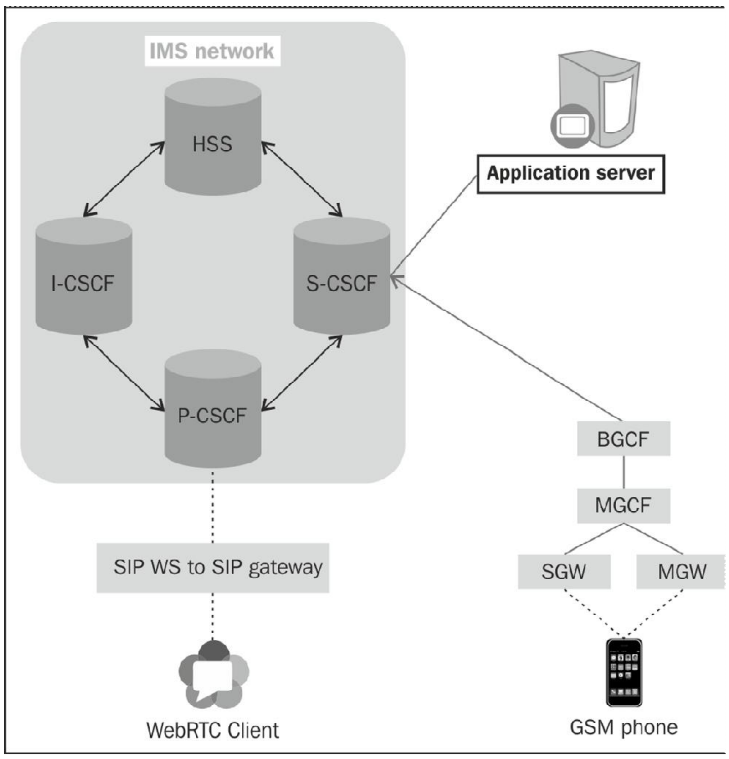
GSM phone


Enum
SIPURI
phonenumber
WebRTC
WebRTC to SIP G/w
MGC/MG
ISUP Switch
Cellular phone
INVITE(SIPWS)
INVITE
Initial address message (IAM)
setup
audio (ringtone)
alert
Address complete message (ACM)
180 ringing
180 ringing
audio (ringtone)
Call progress(CPG)
connect
ANM(Answer message)
media
media
200 OK
200 OK
audio (RTP)
ACK
ACK
release
REL (Release)
BYE
BYE(SIP WS)
RCL(Release complete)
release complete
200 OK
200 OK

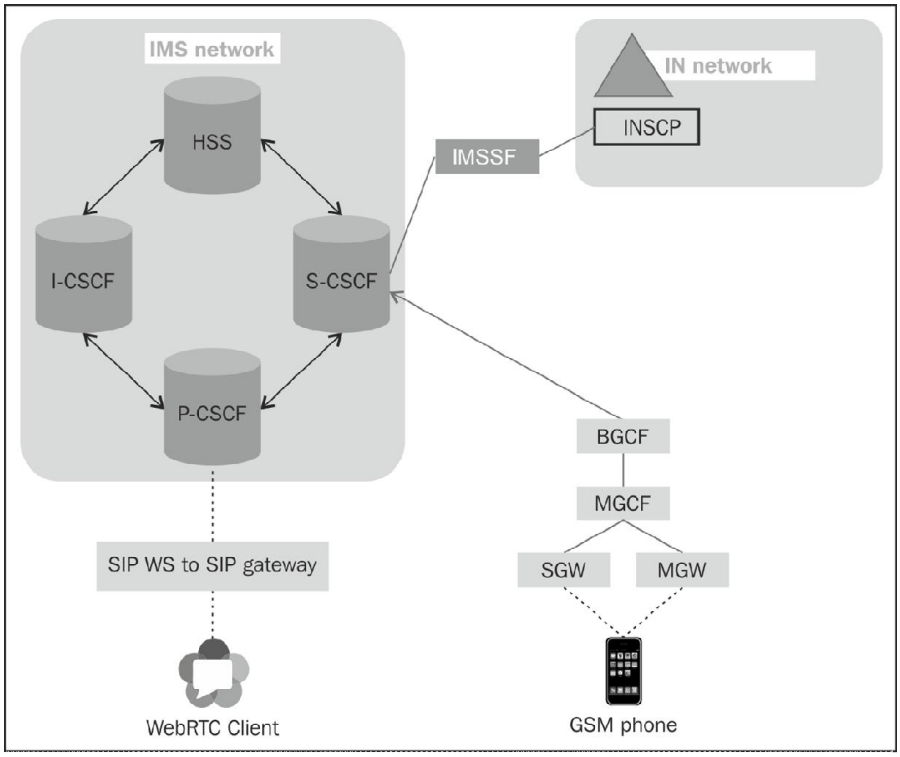
Call processed with the IN service logic

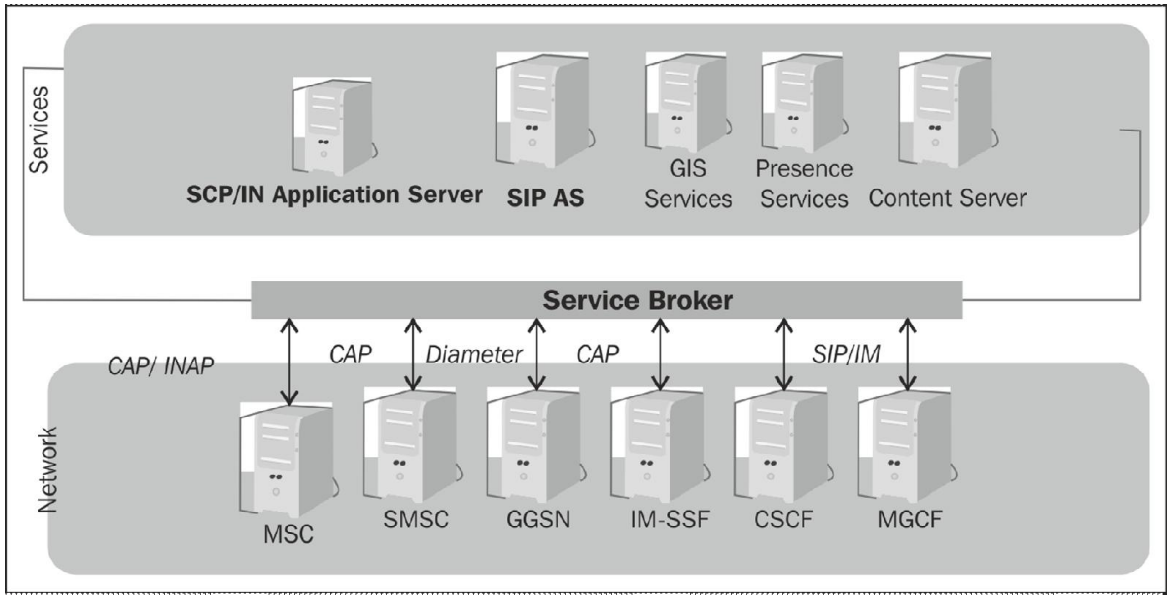


The WebRTC client's communication with the GSM phone through IMS

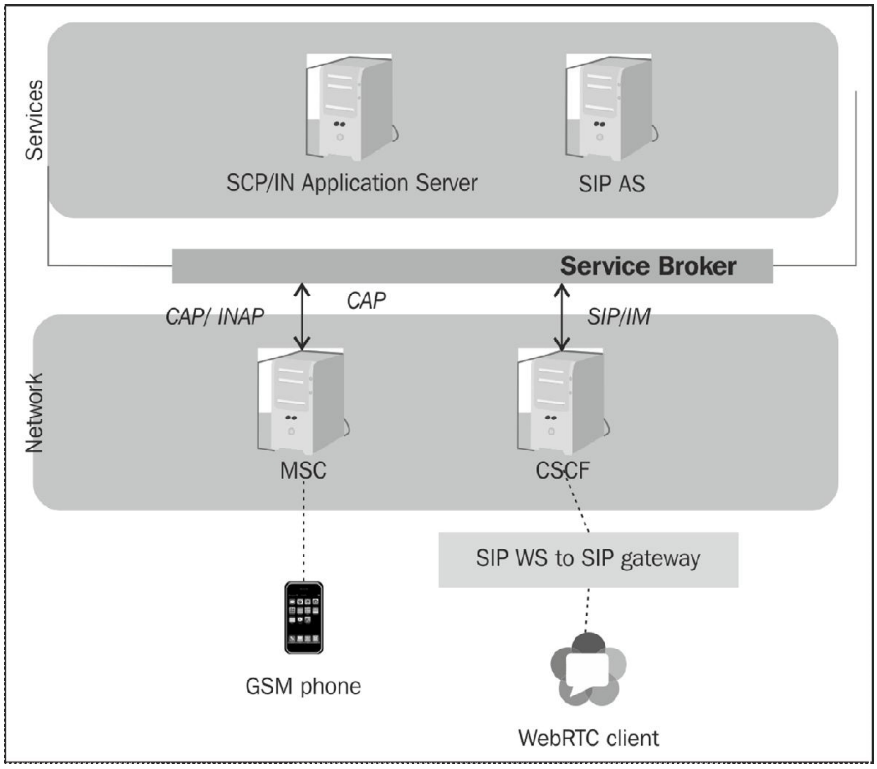


The WebRTC client's communication with a GSM phone with IN services




Services
SCP/IN Application Server
SIP AS
GIS Services
Presence Services
Content Server
Service Broker
CAP/ INAP
CAP
Diameter
CAP
SIP/IM
Network
MSC
SMSC
GGSN
IM-SSF
CSCF
MGCF

The services broker for endpoints, WebRTC in IMS to GSM phone in Intelligence Networks

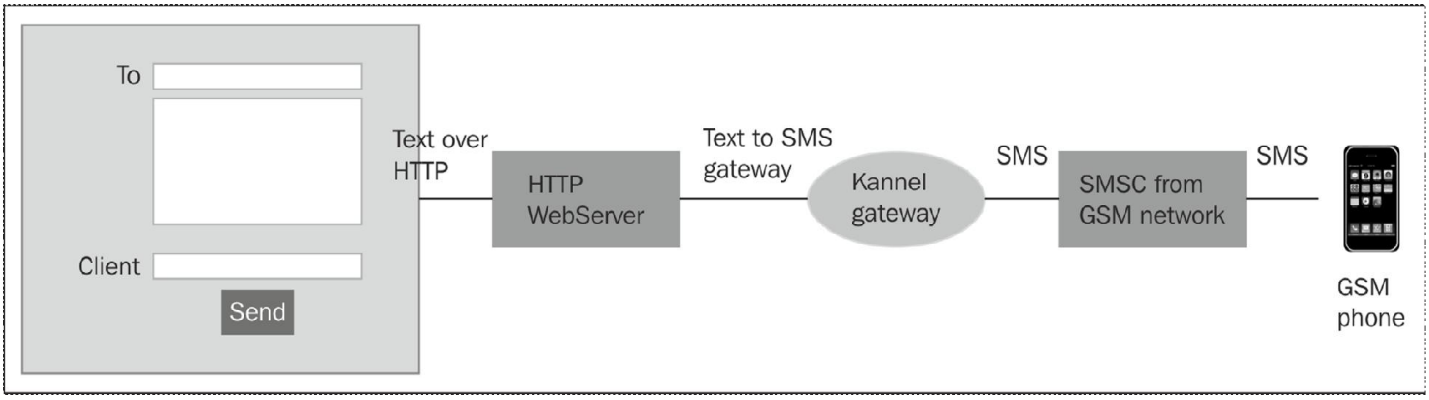


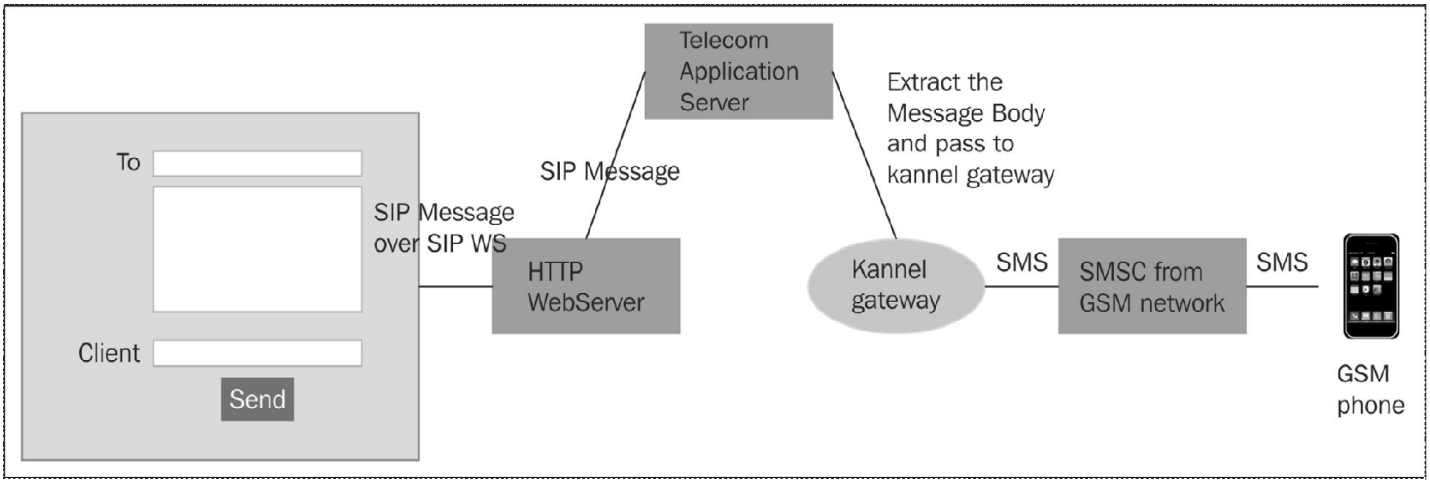
The WebRTC client's SIP messages to SMS in a GSM phone (SMSC)

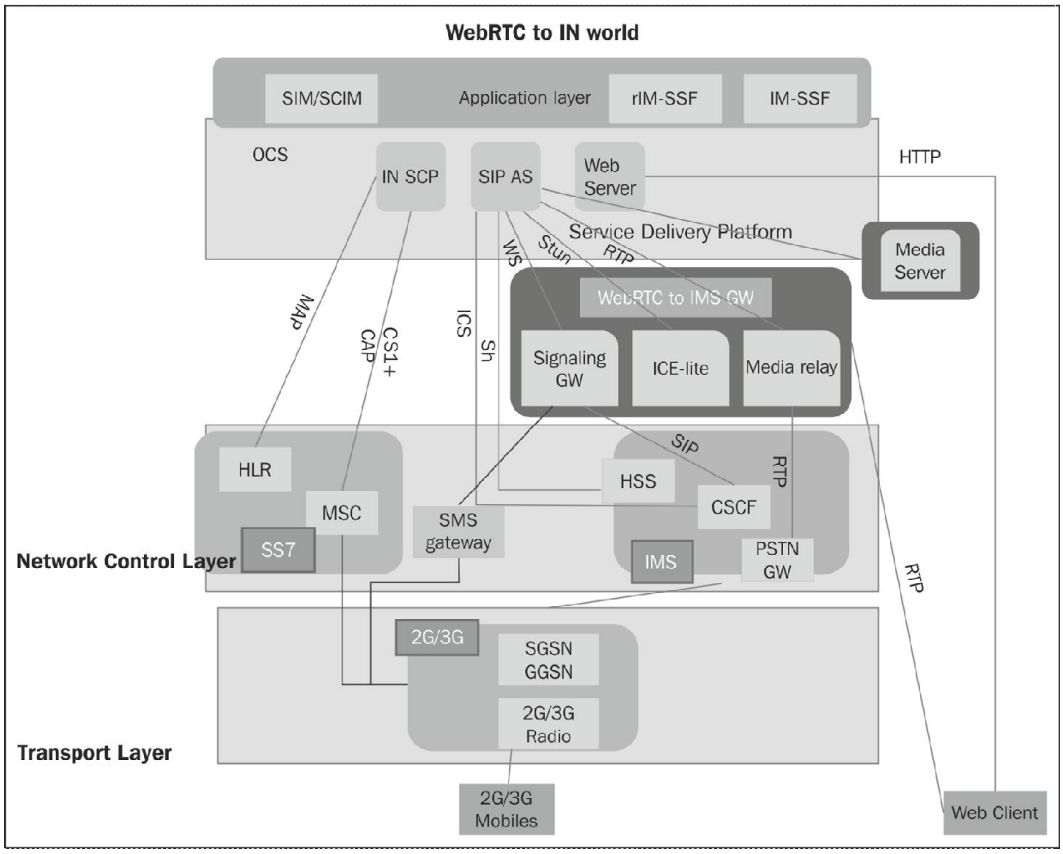
The Kannel gateway

```
altanai@tcs:~/kannel/gateway-1.4.3$ configure

Configuring for Kannel gateway version 1.4.3 ...

Running system checks ...
checking build system type... x86_64-unknown-linux-gnu
checking host system type... x86_64-unknown-linux-gnu
checking for gcc... gcc
checking for C compiler default output file name... a.out
checking whether the C compiler works... yes
checking whether we are cross compiling... no
checking for suffix of executables...
checking for suffix of object files... o
checking whether we are using the GNU C compiler... yes
checking whether gcc accepts -g... yes
checking for gcc option to accept ISO C89... none needed
checking for gcc option to accept ISO C99... -std=gnu99
checking for a BSD-compatible install... /usr/bin/install -c
checking for ranlib... ranlib
checking for bison... bison -y
checking for flex... flex
checking lex output file root... lex.yy
checking lex library... -lfl
checking whether yytext is a pointer... yes
checking for ar... ar
checking for convert... /usr/bin/convert
checking for perl... /usr/bin/perl
checking for inline... inline
checking for special C compiler options needed for large files... no
checking for _FILE_OFFSET_BITS value needed for large files... no
checking how to run the C preprocessor... gcc -std=gnu99 -E
checking for grep that handles long lines and -e... /bin/grep
checking for egrep... /bin/grep -E
checking for ANSI C header files... yes
checking for sys/types.h... yes
checking for sys/stat.h... yes
checking for stdlib.h... yes
checking for string.h... yes
checking for memory.h... yes
checking for strings.h... yes
checking for inttypes.h... yes
checking for stdint.h... yes
checking for unistd.h... yes
checking size of short...
```

```
altanai@tcs:~/kannel/gateway-1.4.3$ make
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_alog.o -c gw/bb_alog.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_boxc.o -c gw/bb_boxc.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_http.o -c gw/bb_http.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_smscconn.o -c gw/bb_smscconn.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_store.o -c gw/bb_store.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_store_file.o -c gw/bb_store_file.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_store_spool.o -c gw/bb_store_spool.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/bb_udp.o -c gw/bb_udp.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/dlr.o -c gw/dlr.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/dlr_mem.o -c gw/dlr_mem.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/dlr_mysql.o -c gw/dlr_mysql.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/dlr_oracle.o -c gw/dlr_oracle.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/dlr_pgsql.o -c gw/dlr_pgsql.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/dlr_sdb.o -c gw/dlr_sdb.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/heartbeat.o -c gw/heartbeat.c
gcc -std=gnu99 -D_REENTRANT=1 -I. -Igw -g -O2 -D_XOPEN_SOURCE=600 -D_BSD_SOURCE -D_LARGE_FILES= -I/usr/incl
ude/libxml2 -o gw/html.o -c gw/html.c
```

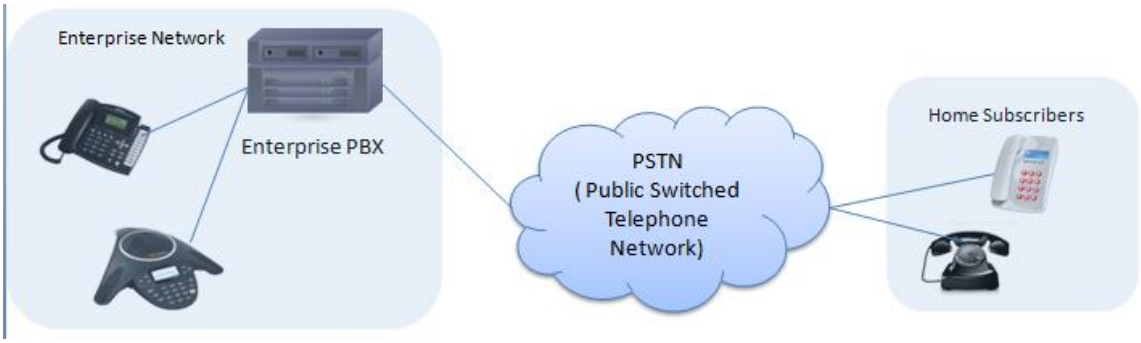



Telecom
Application
Server
Extract the
Message Body
and pass to
kannel gateway
To
SIP Message
SIP Message
over SIP WS
HTTP
WebServer
Kannel
gateway
SMS
SMSC from
GSM network
SMS
Client
Send
GSM
phone

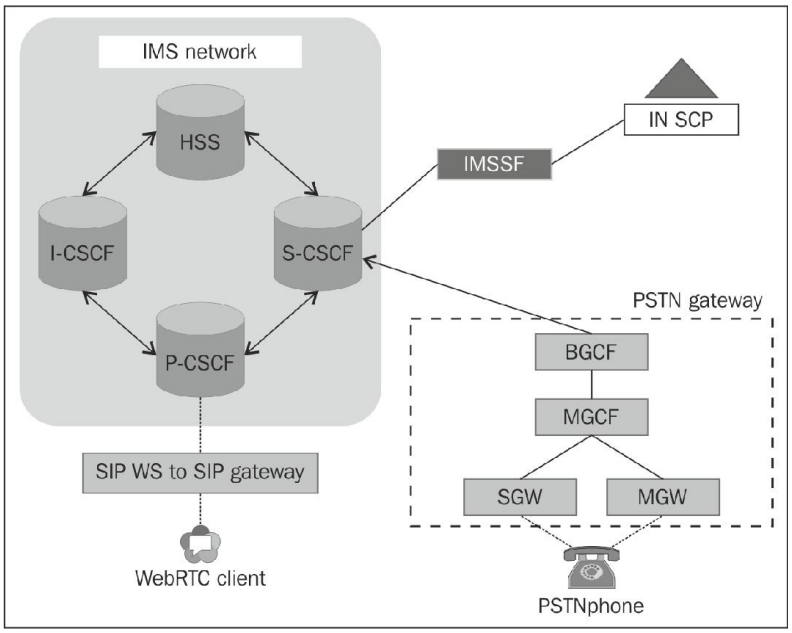

WebRTC to IN world
SIM/SCIM
Application layer
rIM-SSF
IM-SSF
OCS
IN SCP
SIP AS
Web
Server
HTTP
Service Delivery Platform
Media
Server
WS
Stun
RTP
WebRTC to IMS GW
Signaling
GW
ICE-lite
Media relay
Map
CS1+
CAP
ICS
Sh
SIP
RTP
HLR
MSC
SS7
SMS
gateway
HSS
CSCF
IMS
PSTN
GW
Network Control Layer
RTP
2G/3G
SGSN
GGSN
2G/3G
Radio
Transport Layer
2G/3G
Mobiles
Web Client

5

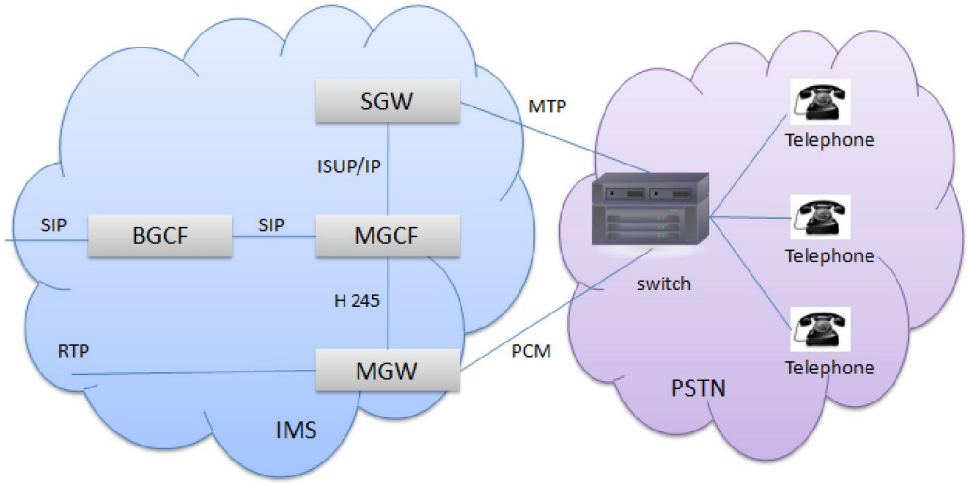
WebRTC Integration with PSTN



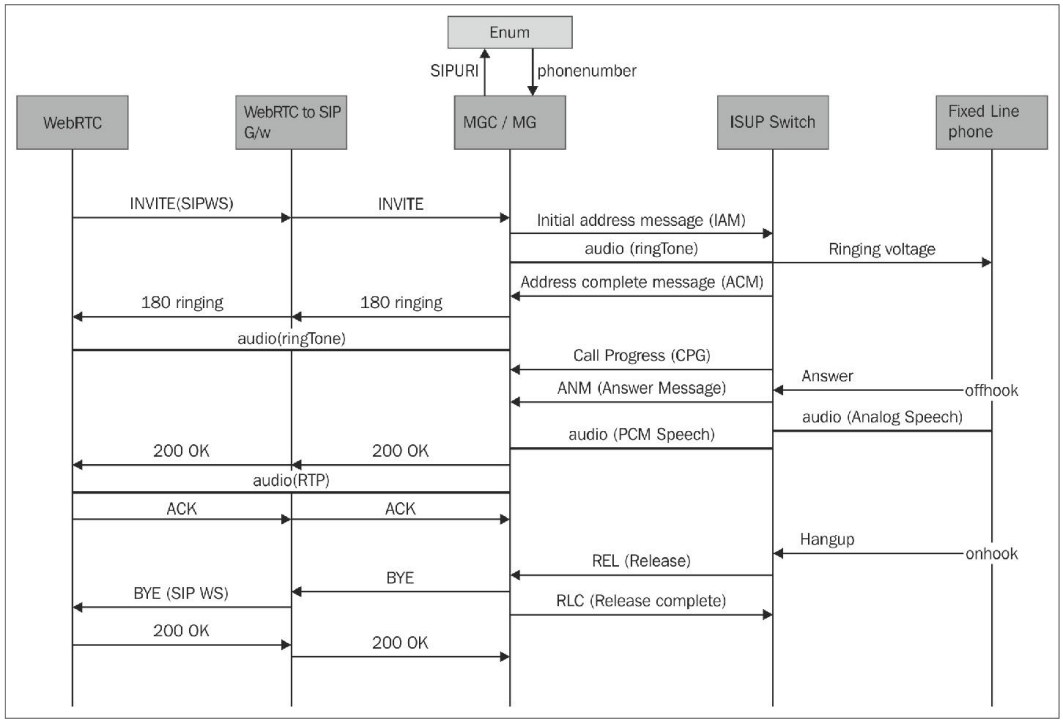
WebRTC connectivity to the PSTN



The PSTN gateway

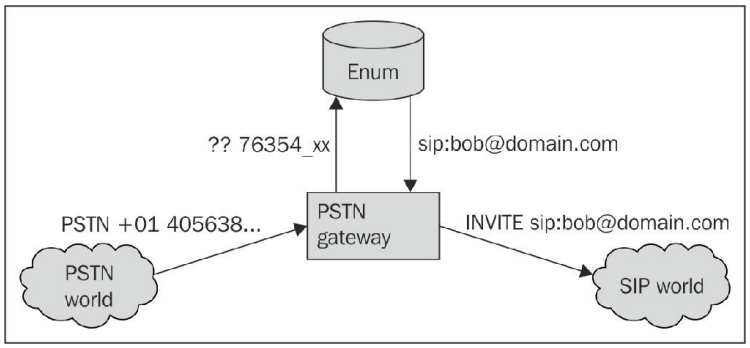


The PSTN connectivity to IMS via PSTN gateways



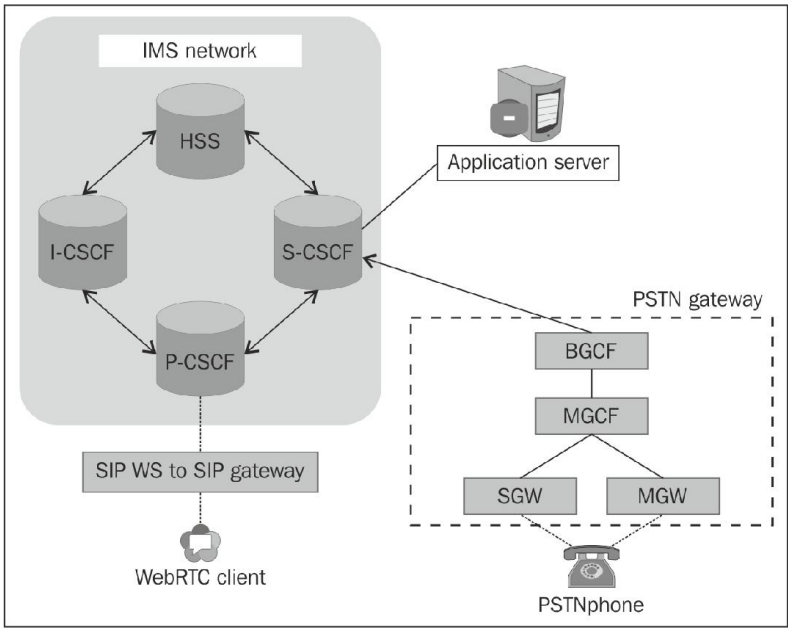
The challenges in connecting the WebRTC world to the PSTN landscape

Address mapping

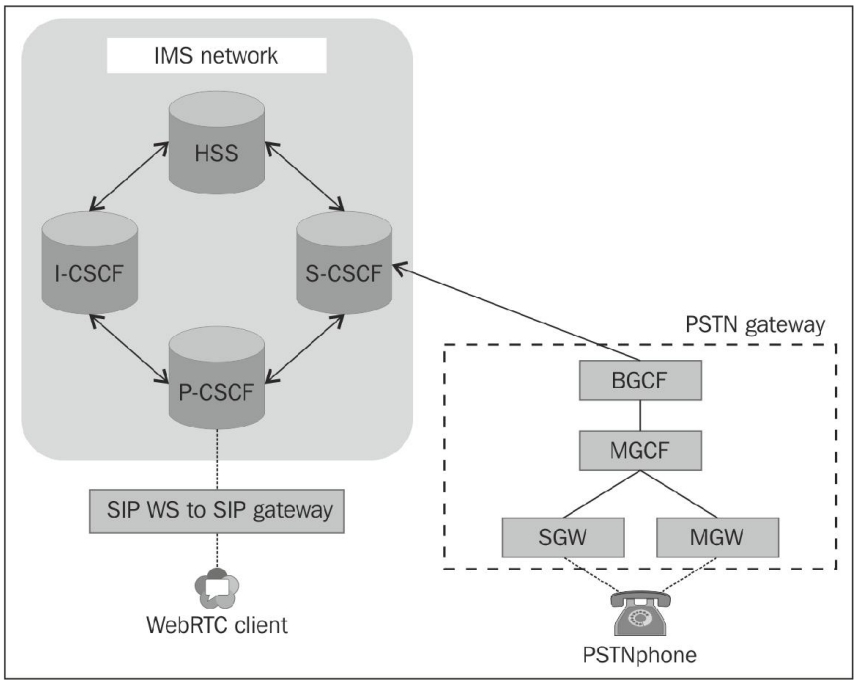


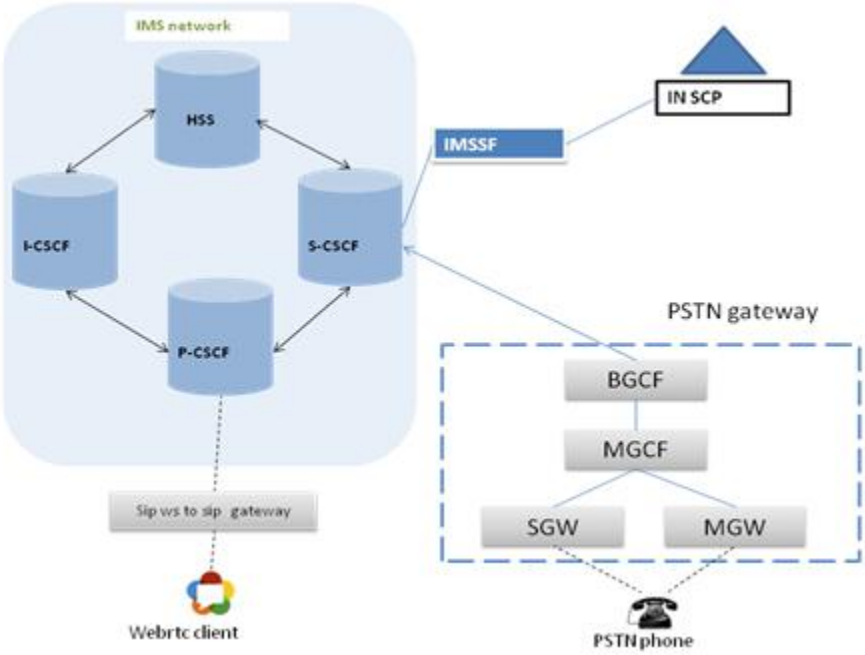
The service logic

SIP service logic through application server

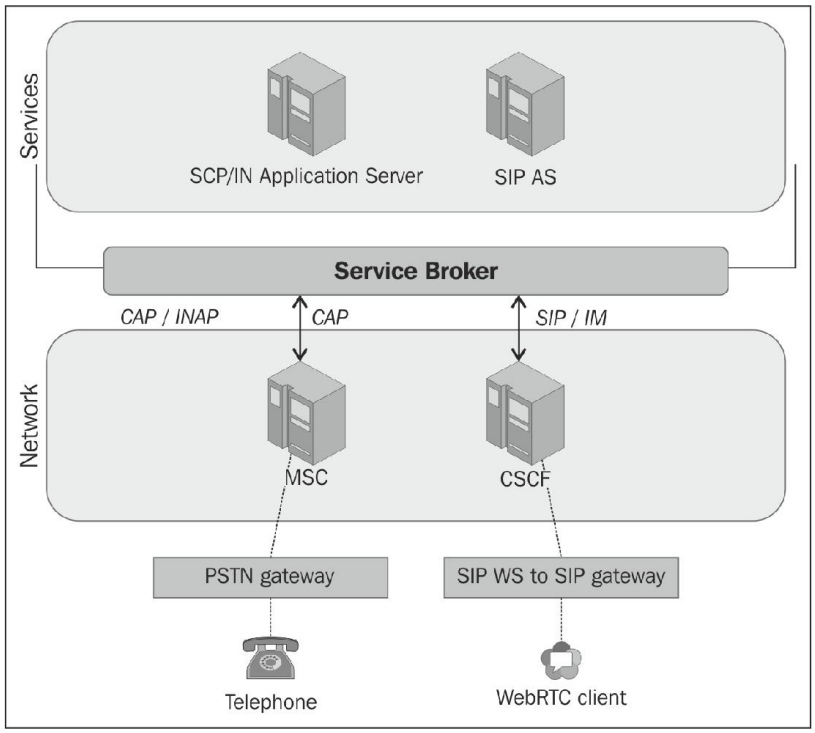


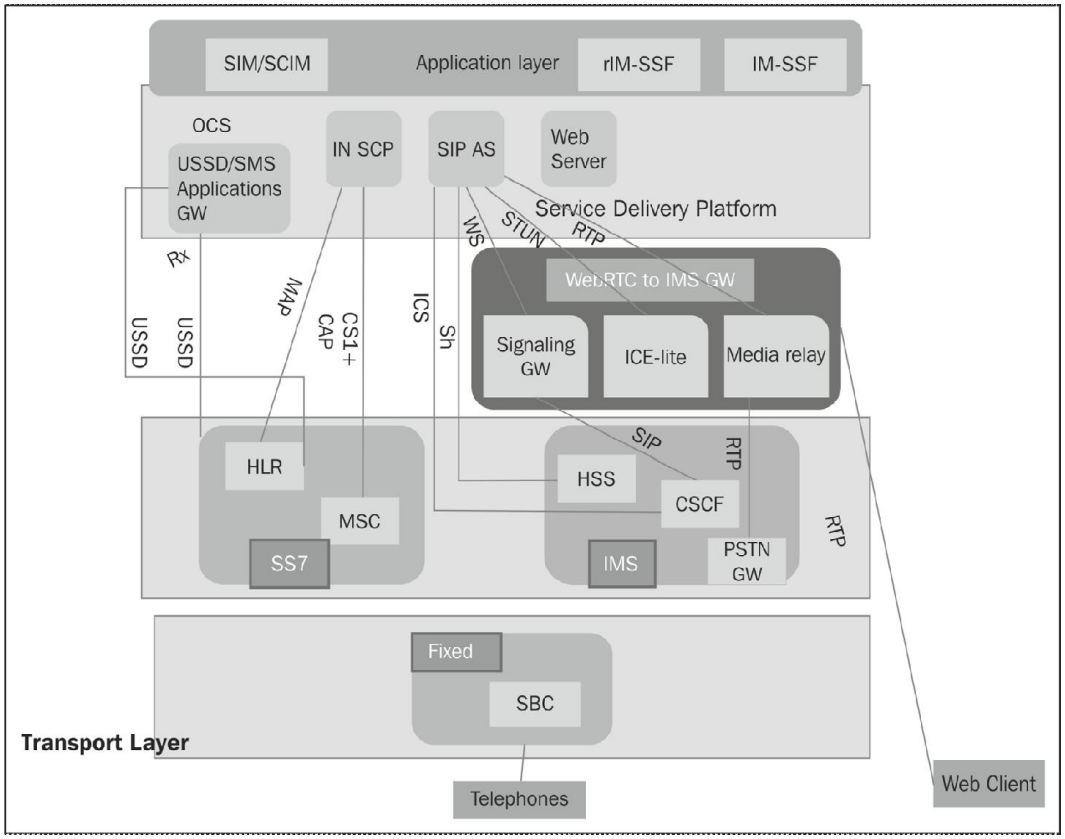
IN services via IMSSF




IMS network
HSS
I-CSCF
S-CSCF
P-CSCF
IMSSF
IN SCP
PSTN gateway
BGCF
MGCF
SGW
MGW
Sip ws to sip gateway
Webrtc client
PSTN phone

The Service Broker for the orchestration of services

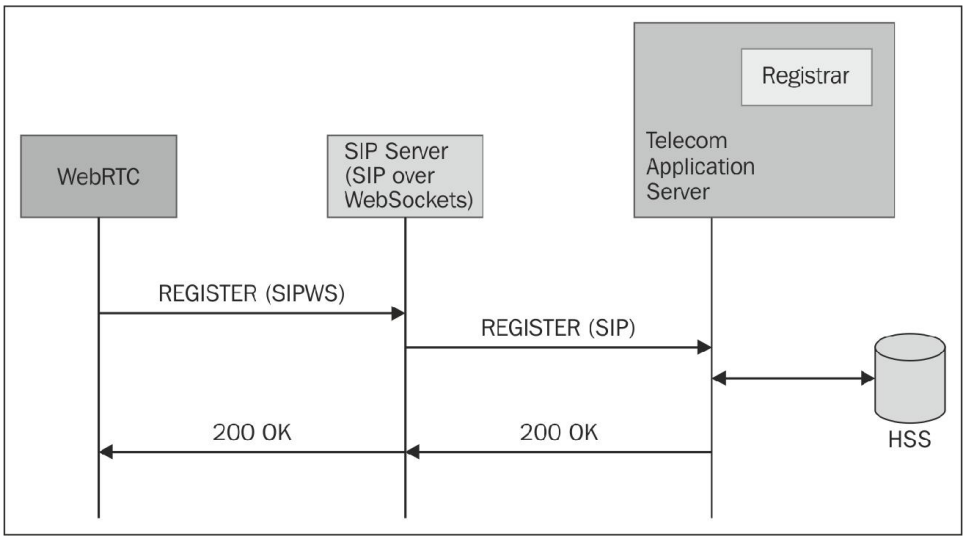


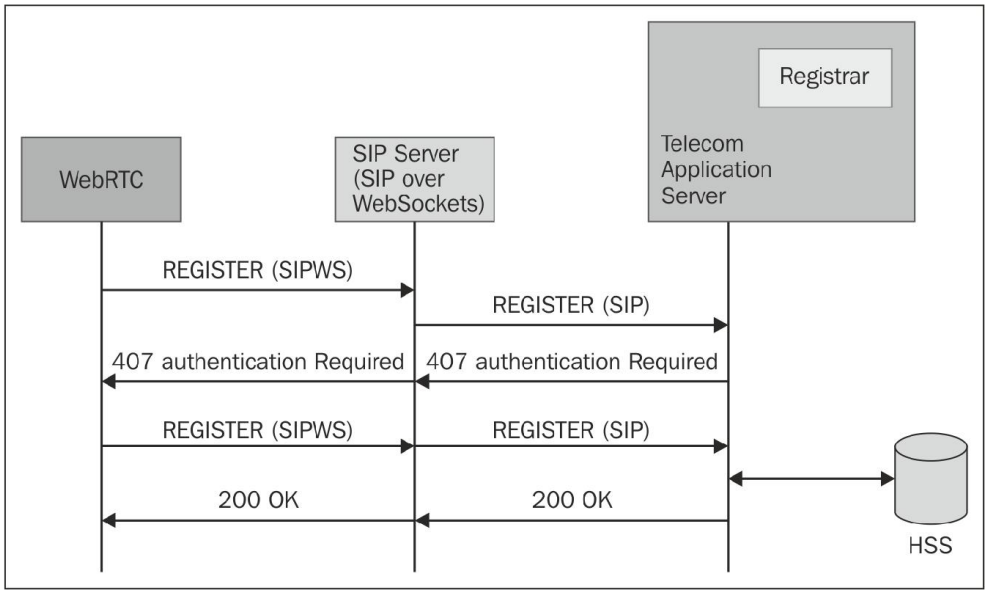

SIM/SCIM
Application layer
rIM-SSF
IM-SSF
OCS
USSD/SMS Applications GW
IN SCP
SIP AS
Web Server
Service Delivery Platform
WS
STUN
RTP
WebRTC to IMS GW
Signaling GW
ICE-lite
Media relay
Rx
USSD
USSD
MAP
CS1+ CAP
ICS
Sh
HLR
MSC
SS7
HSS
CSCF
IMS
PSTN GW
SIP
RTP
RTP
Fixed
SBC
Transport Layer
Telephones
Web Client

6
Basic Features of WebRTC over SIP

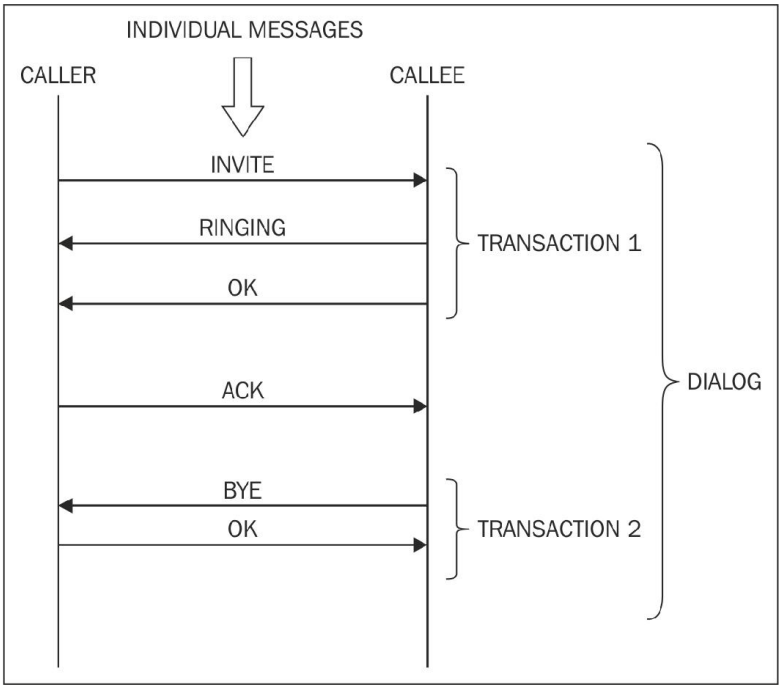
SIP services

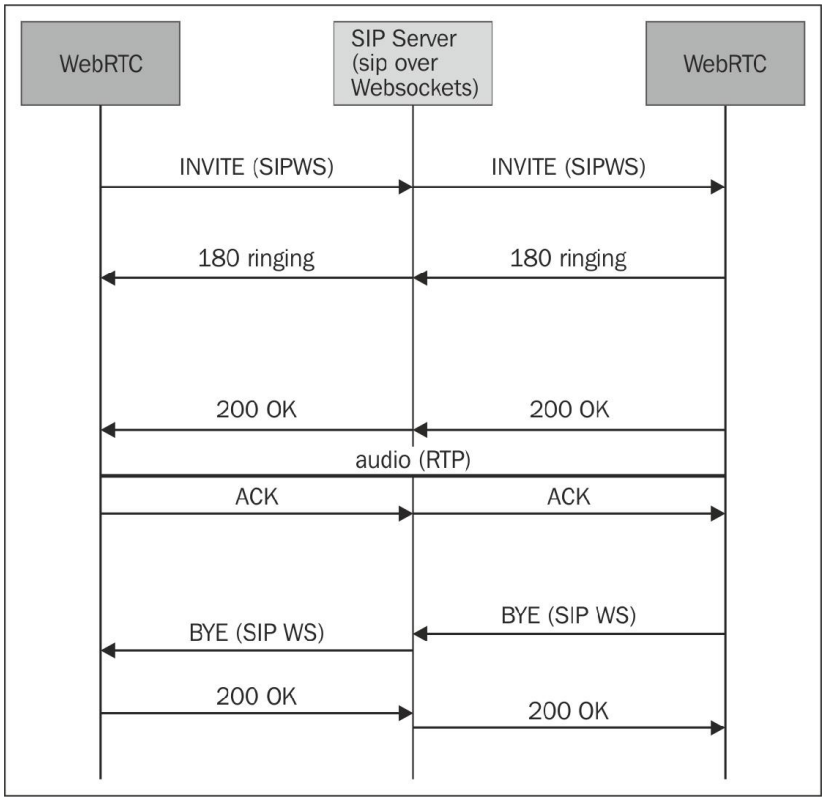
Registering a SIP client

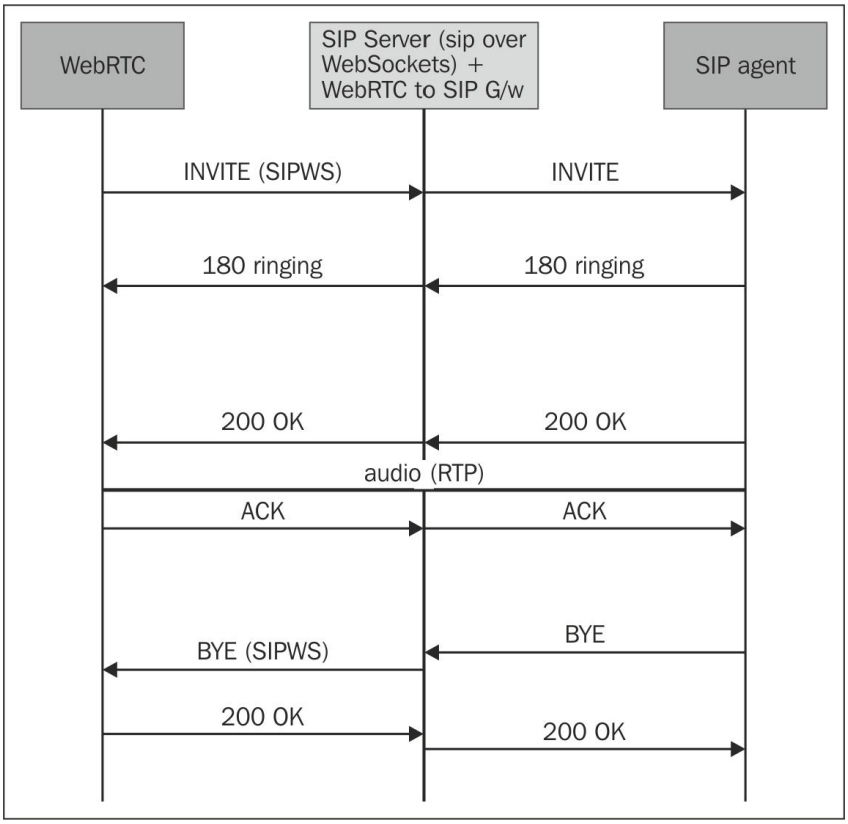




Making audio and video calls using SIP

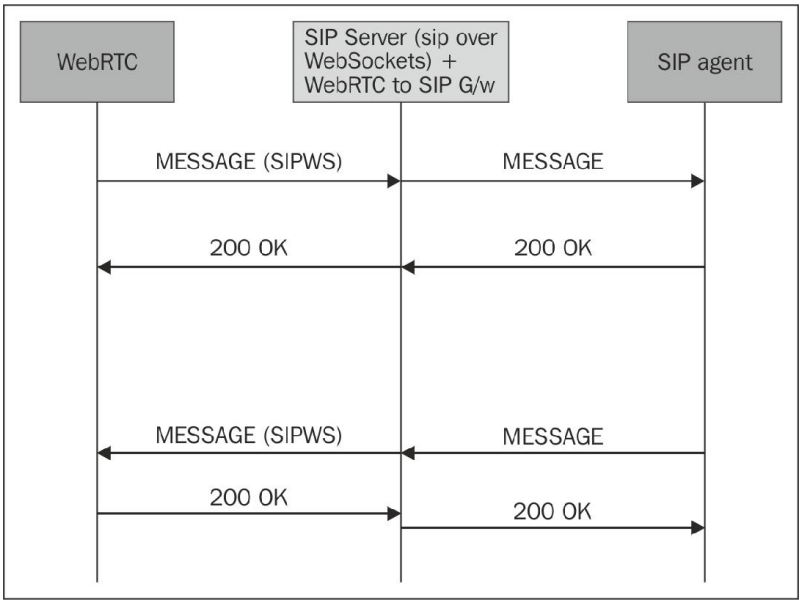



WebRTC
SIP Server (sip over Websockets)
WebRTC
INVITE (SIPWS)
INVITE (SIPWS)
180 ringing
180 ringing
200 OK
200 OK
audio (RTP)
ACK
ACK
BYE (SIP WS)
BYE (SIP WS)
200 OK
200 OK

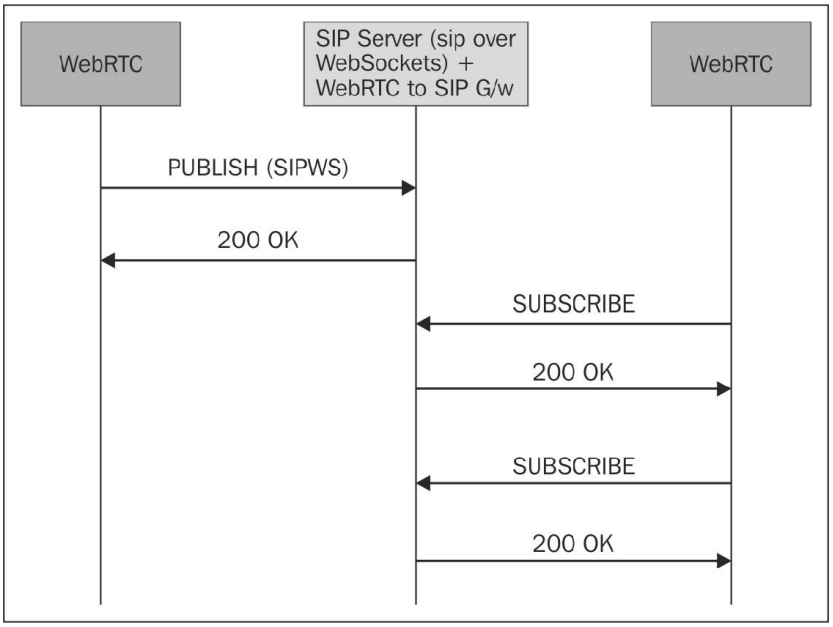

WebRTC
SIP Server (sip over WebSockets) + WebRTC to SIP G/w
SIP agent
INVITE (SIPWS)
INVITE
180 ringing
180 ringing
200 OK
200 OK
audio (RTP)
ACK
ACK
BYE
BYE (SIPWS)
200 OK
200 OK

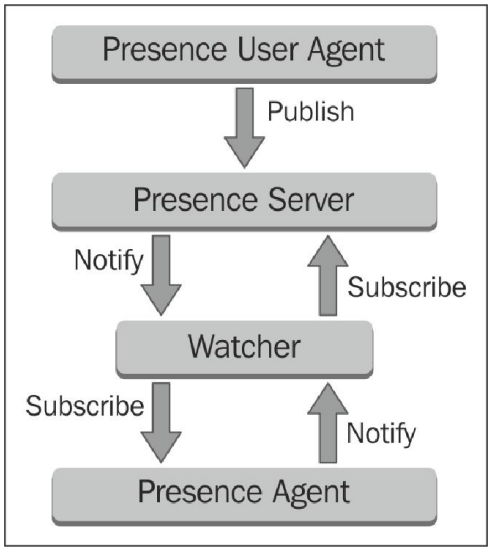
http://webrtc.domain.com/ wants to access your camera and microphone
Allow
Deny

Text Chat using SIP

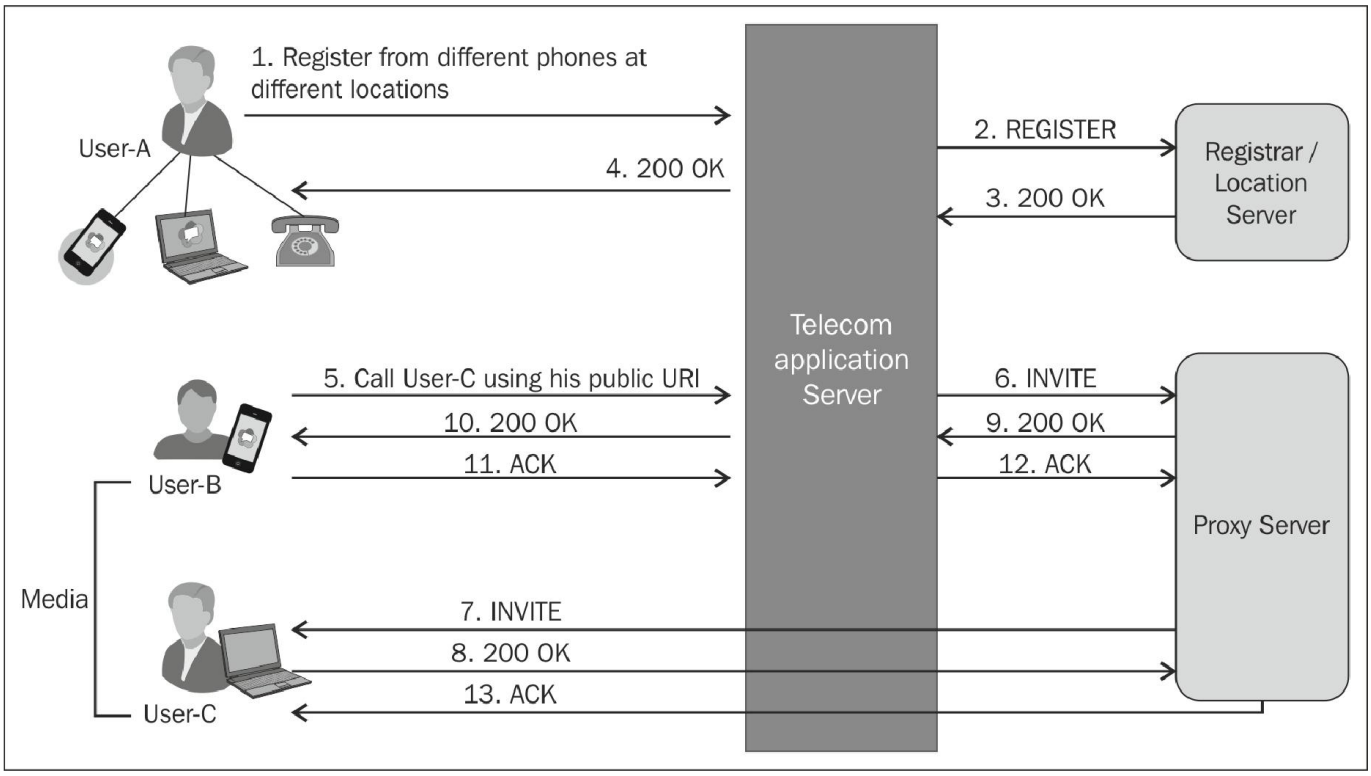


Obtaining the online/offline status of users using SIP

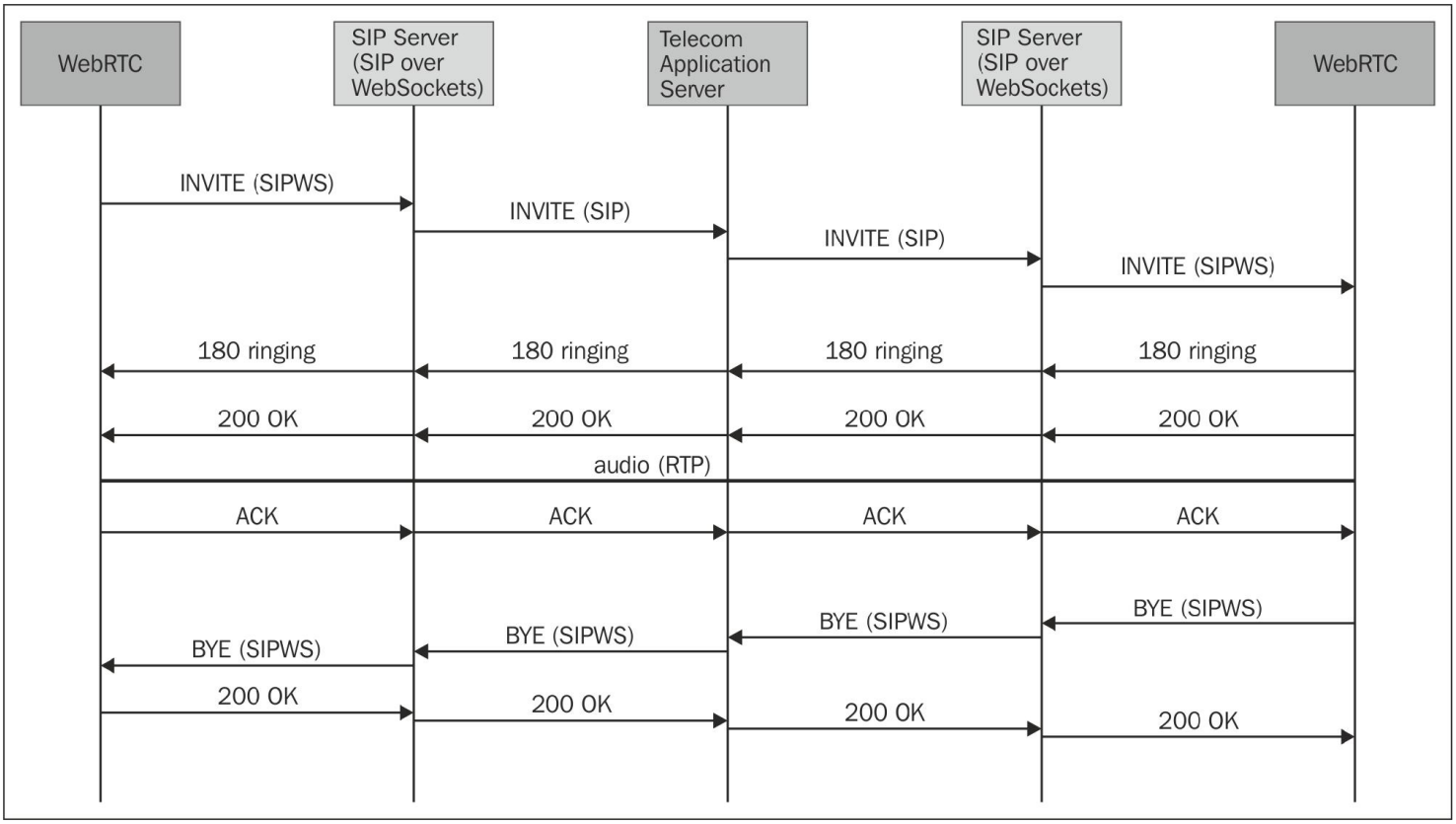



Presence User Agent
Publish
Presence Server
Notify
Subscribe
Watcher
Subscribe
Notify
Presence Agent

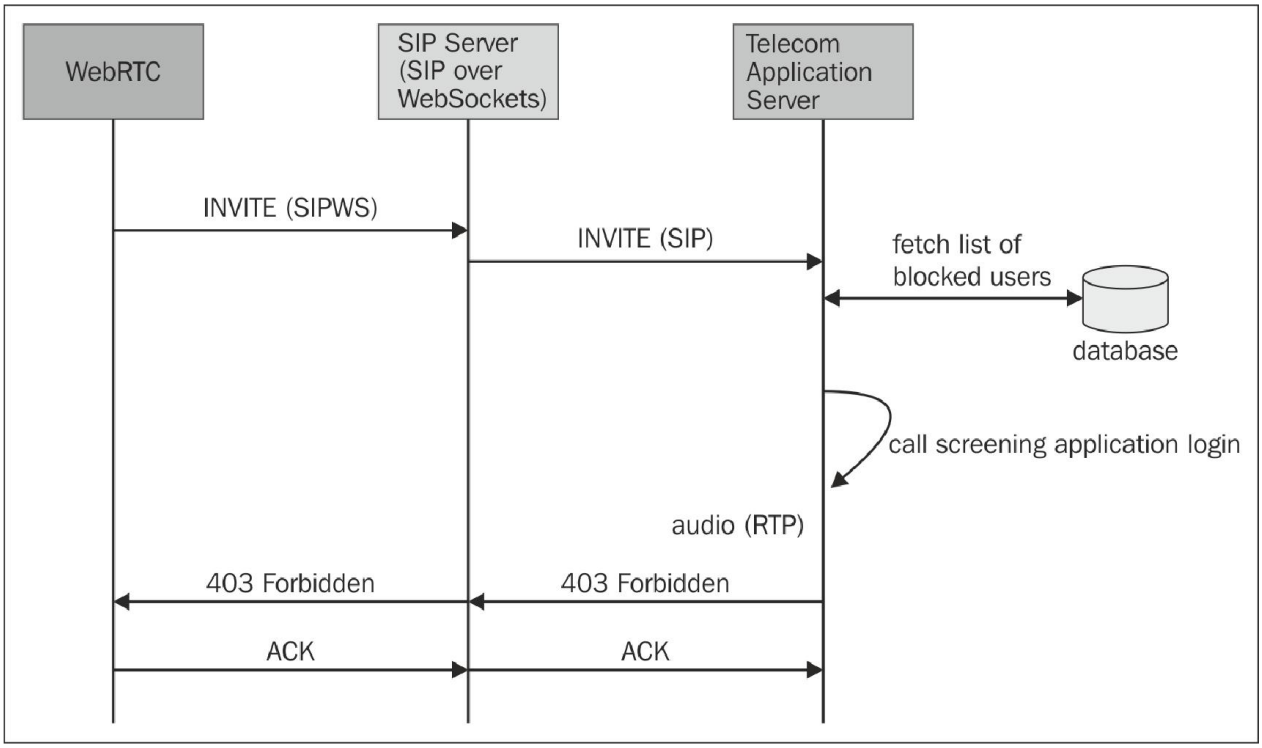
Services in the Application Server



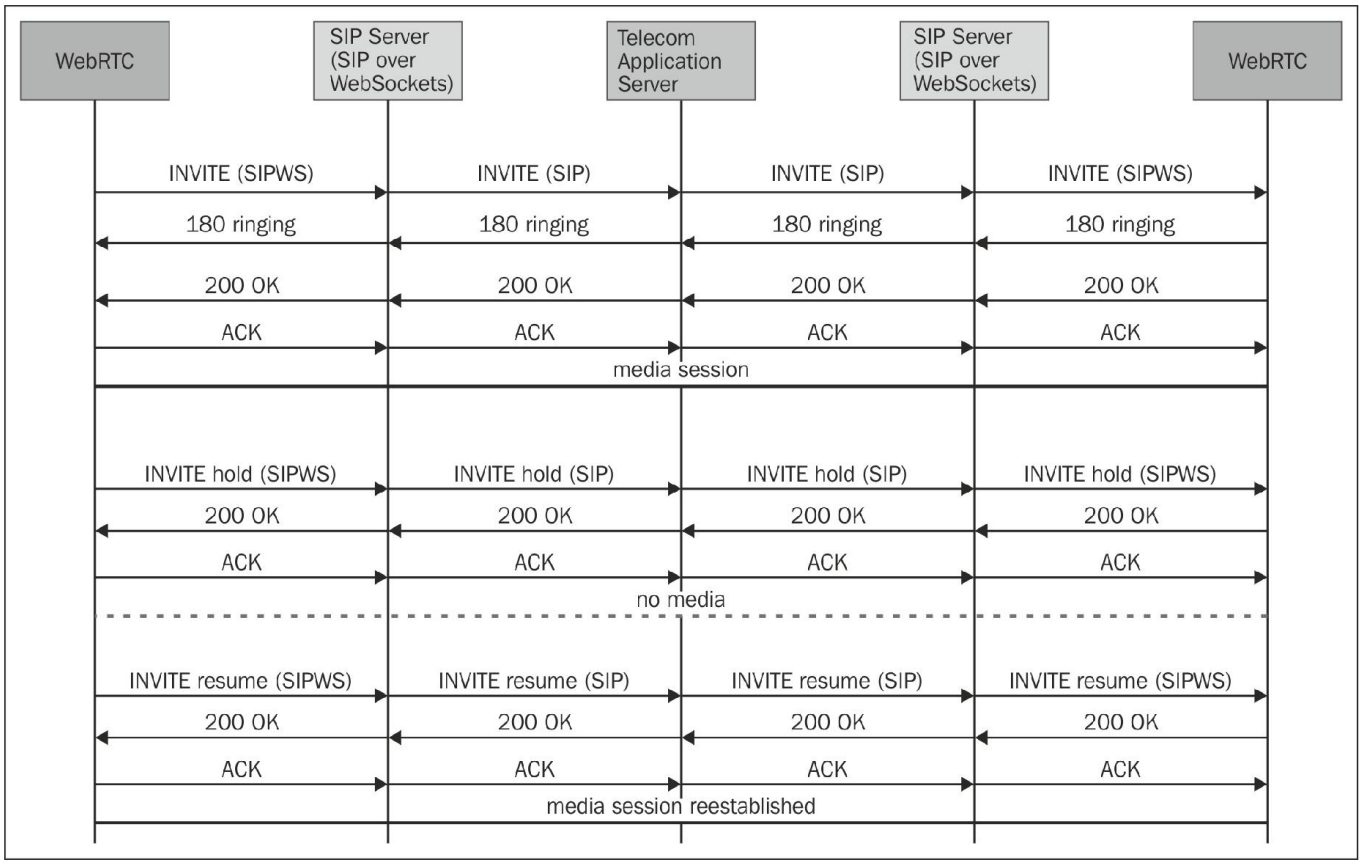
Back-to-back user agent



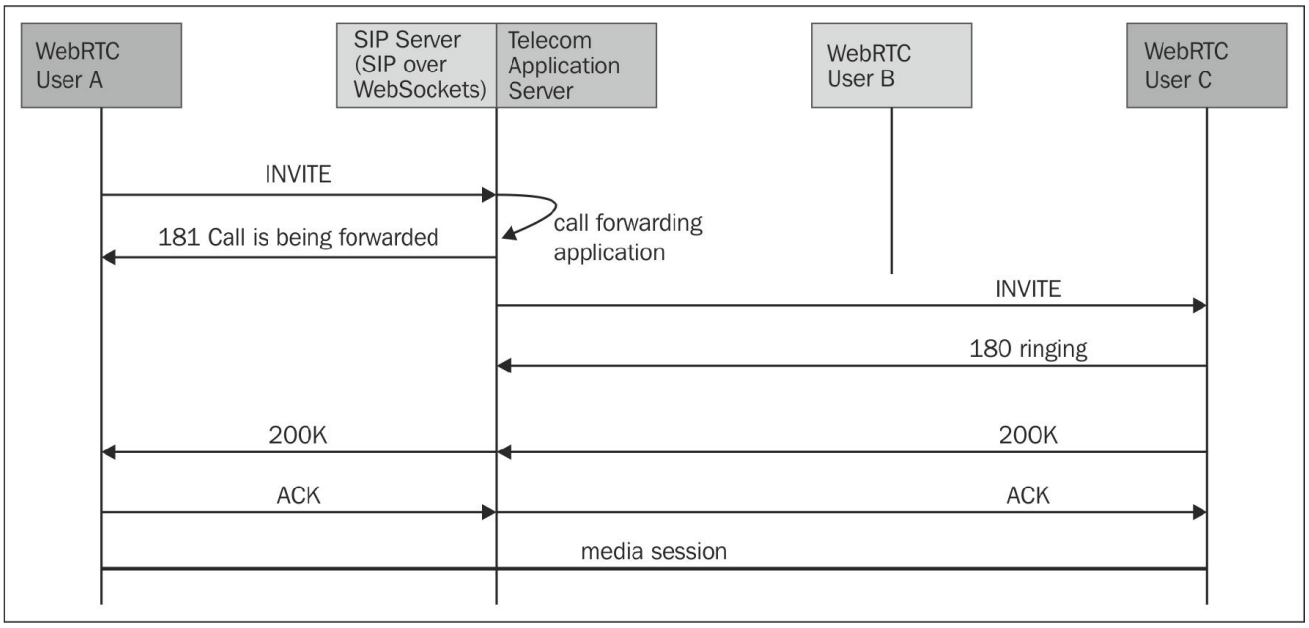
Call screening

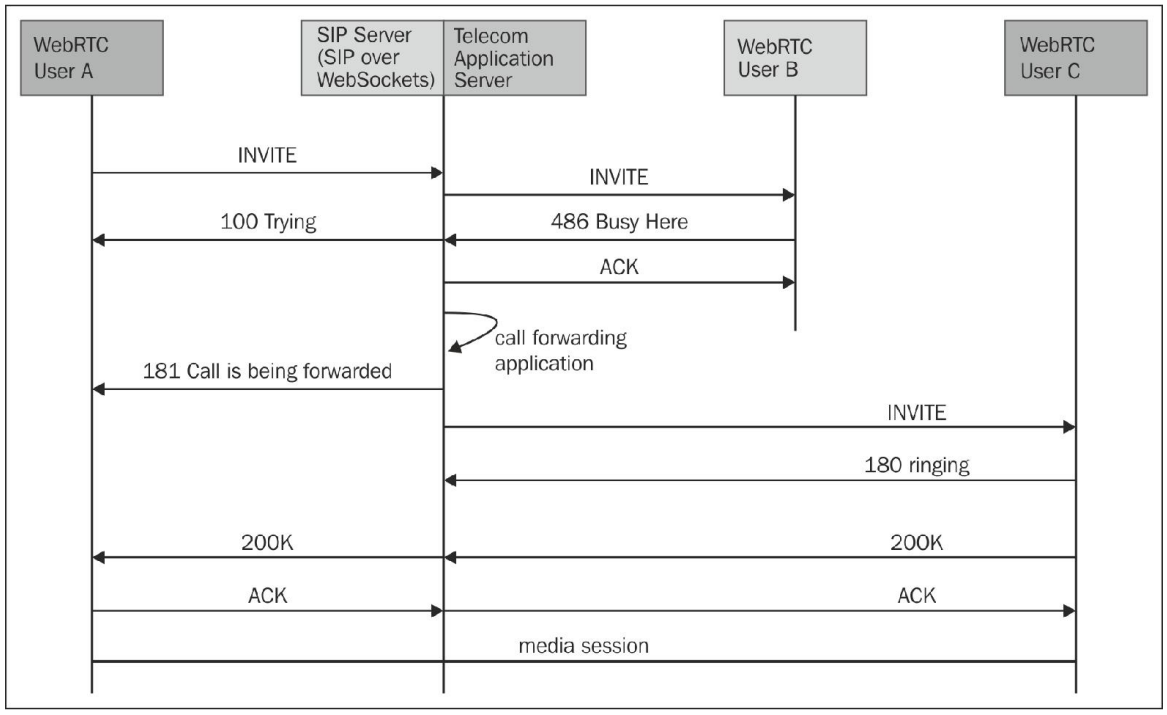


Call hold/resume

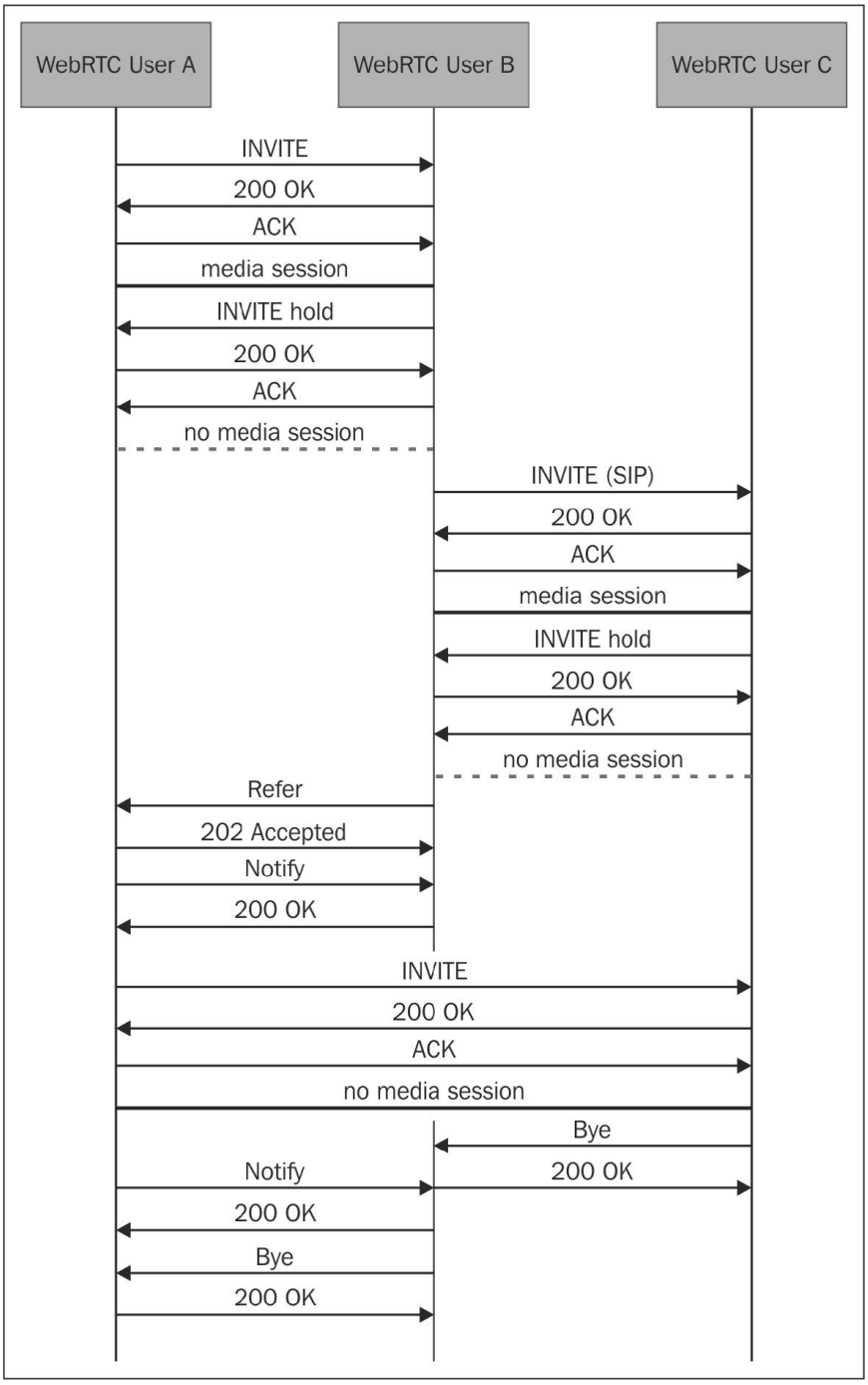


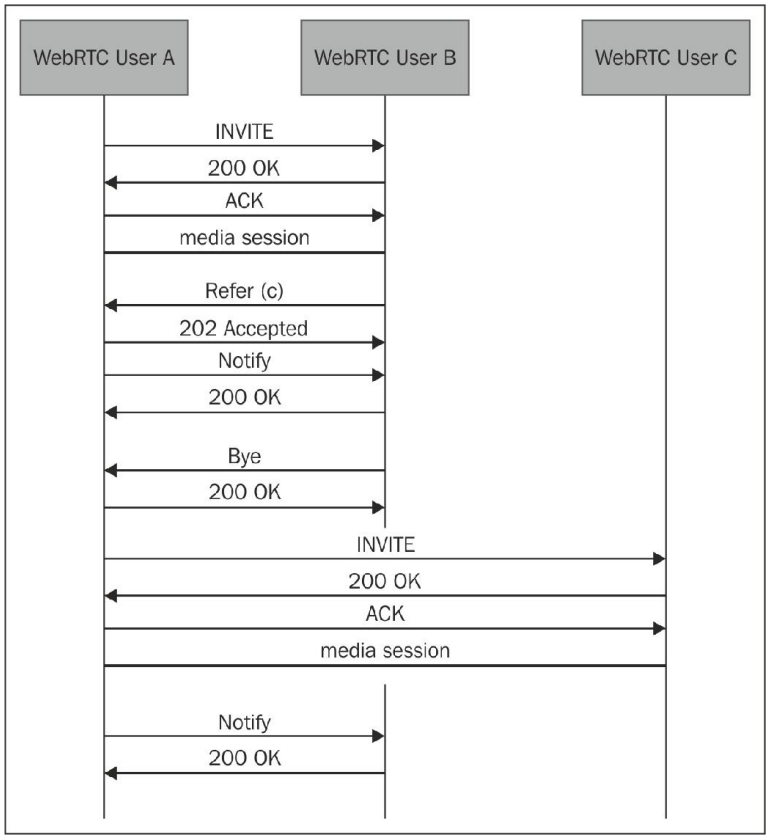
Call forwarding



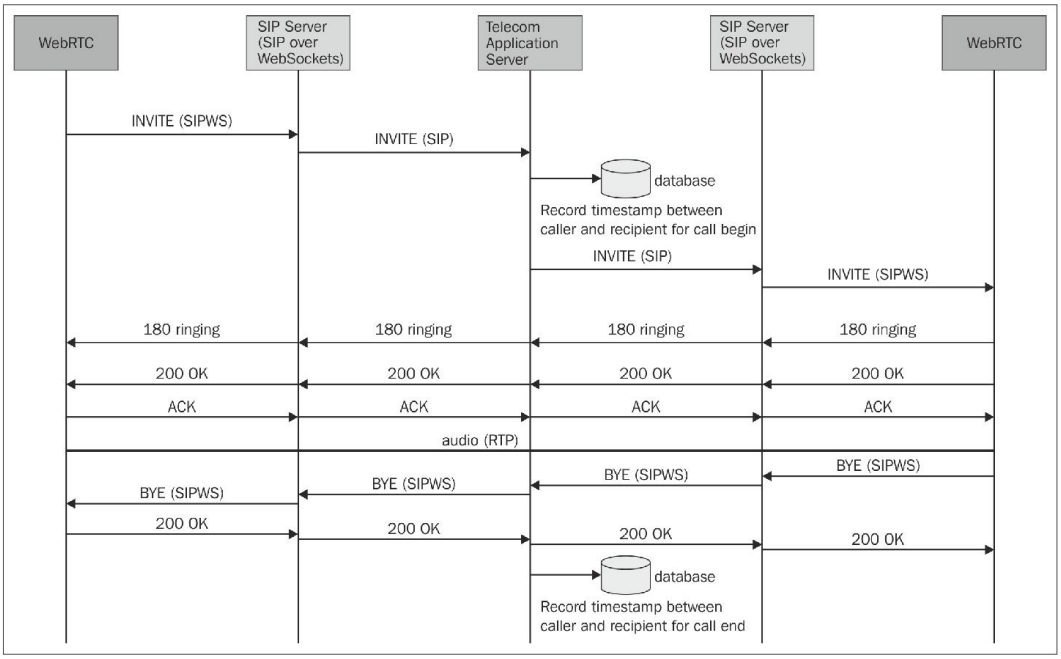


Call transfer


WebRTC User A
WebRTC User B
WebRTC User C
INVITE
200 OK
ACK
media session
INVITE hold
200 OK
ACK
no media session
INVITE (SIP)
200 OK
ACK
media session
INVITE hold
200 OK
ACK
no media session
Refer
202 Accepted
Notify
200 OK
INVITE
200 OK
ACK
no media session
Bye
Notify
200 OK
200 OK
Bye
200 OK

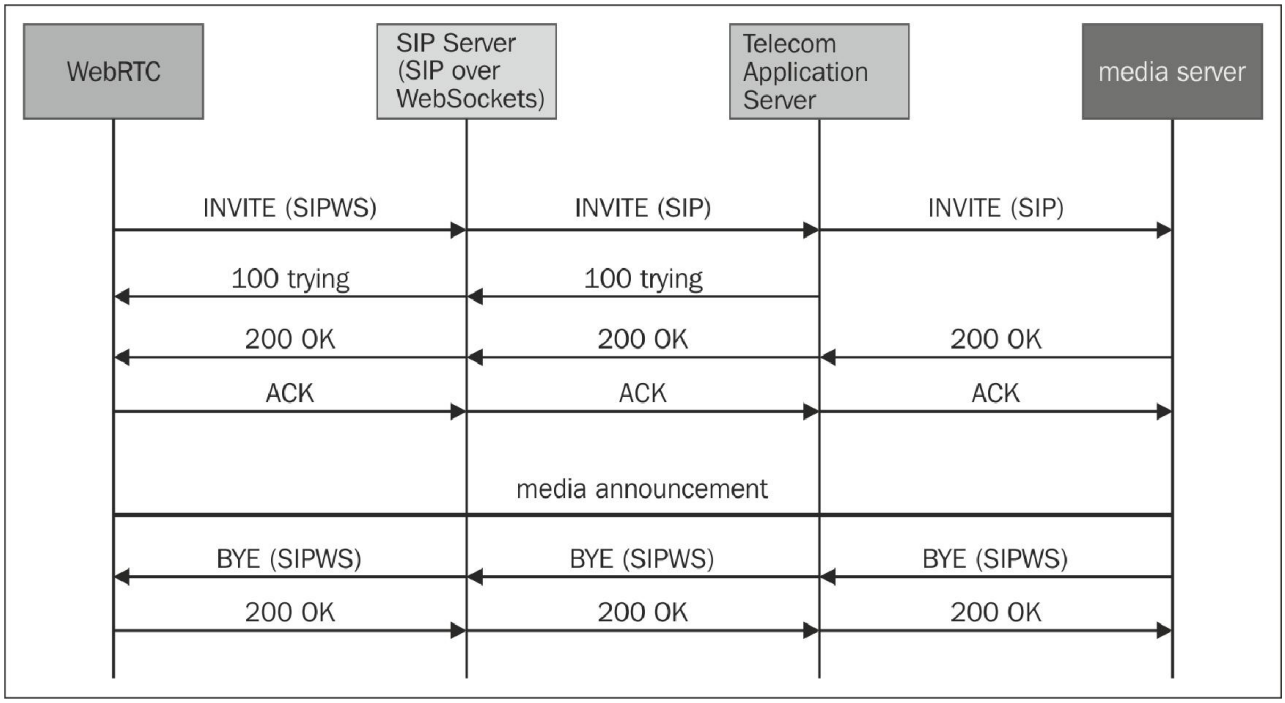

WebRTC User A
WebRTC User B
WebRTC User C
INVITE
200 OK
ACK
media session
Refer (c)
202 Accepted
Notify
200 OK
Bye
200 OK
INVITE
200 OK
ACK
media session
Notify
200 OK

Generation of call log for tracking

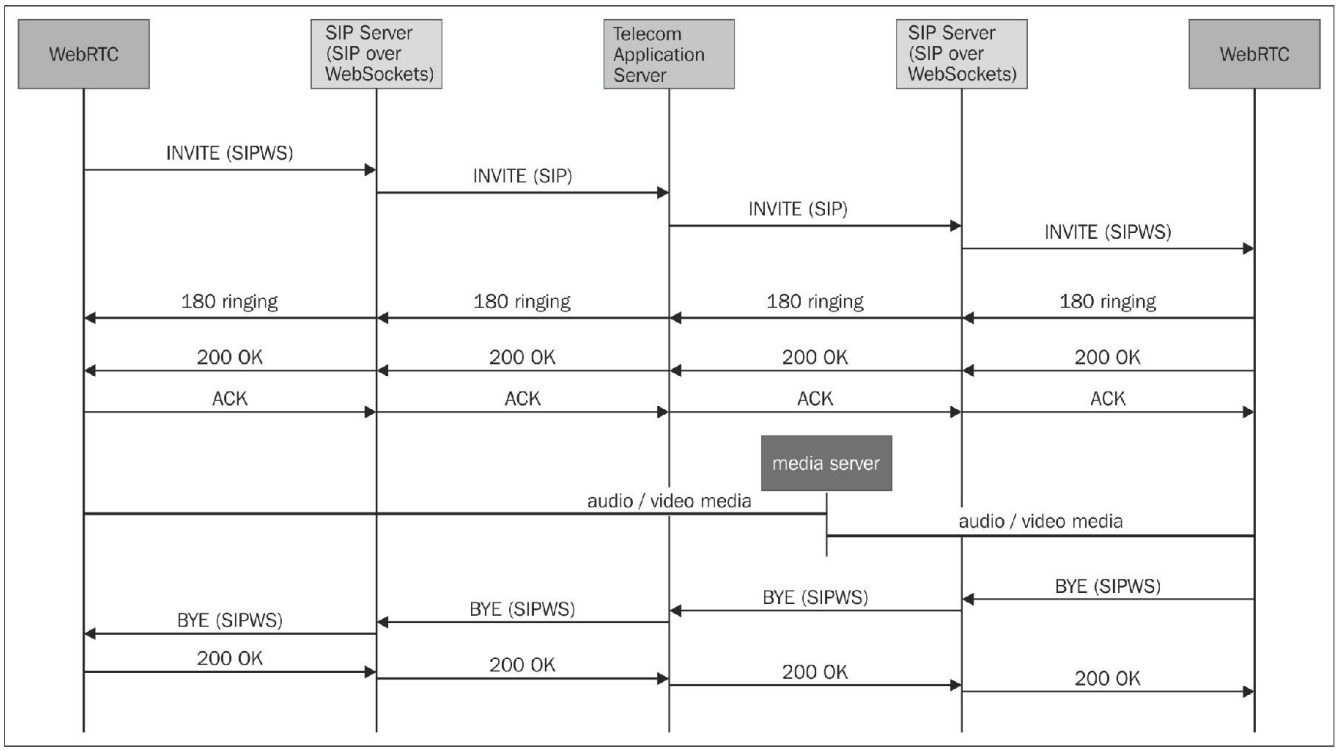


Media Server-based features

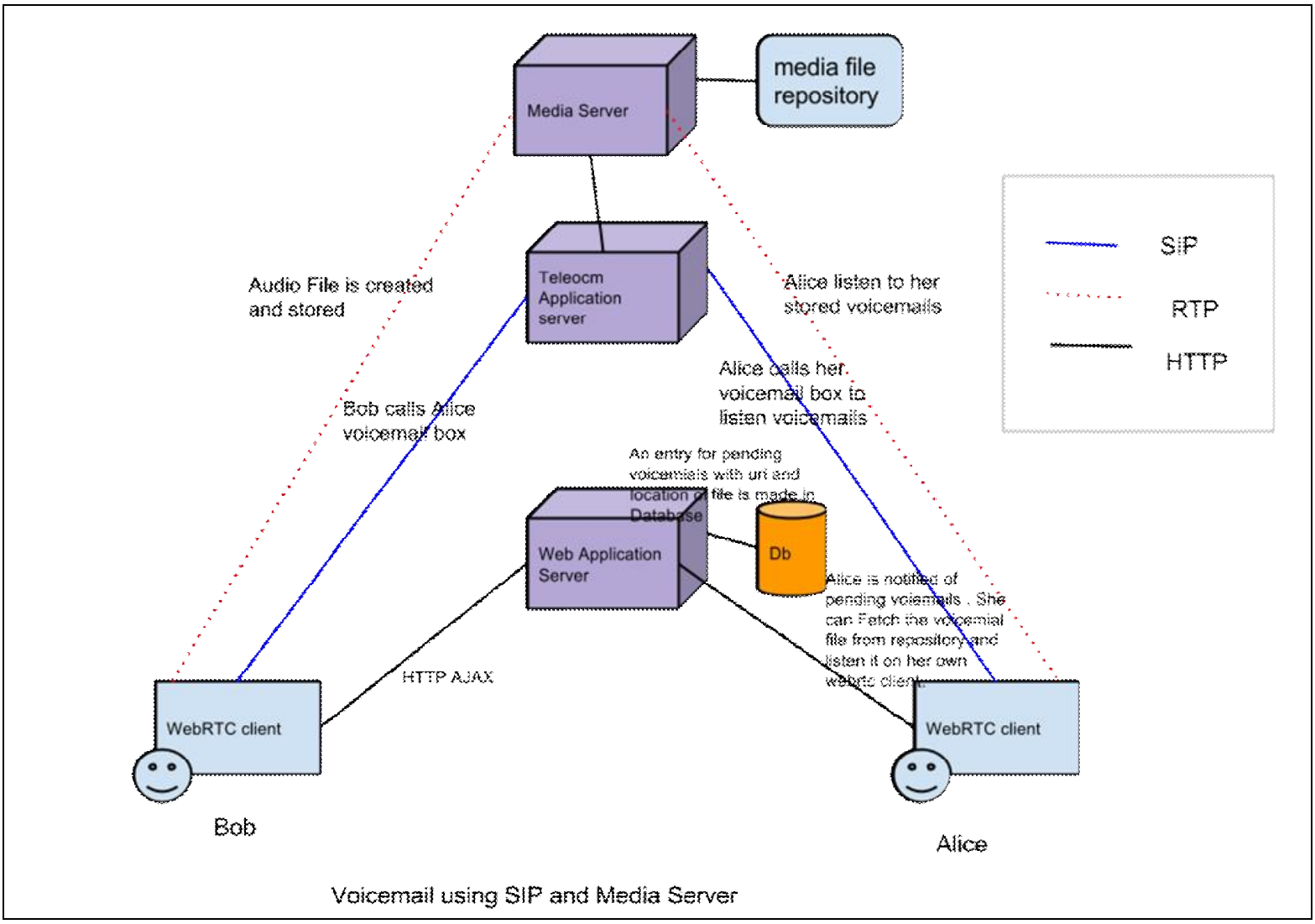
Announcement



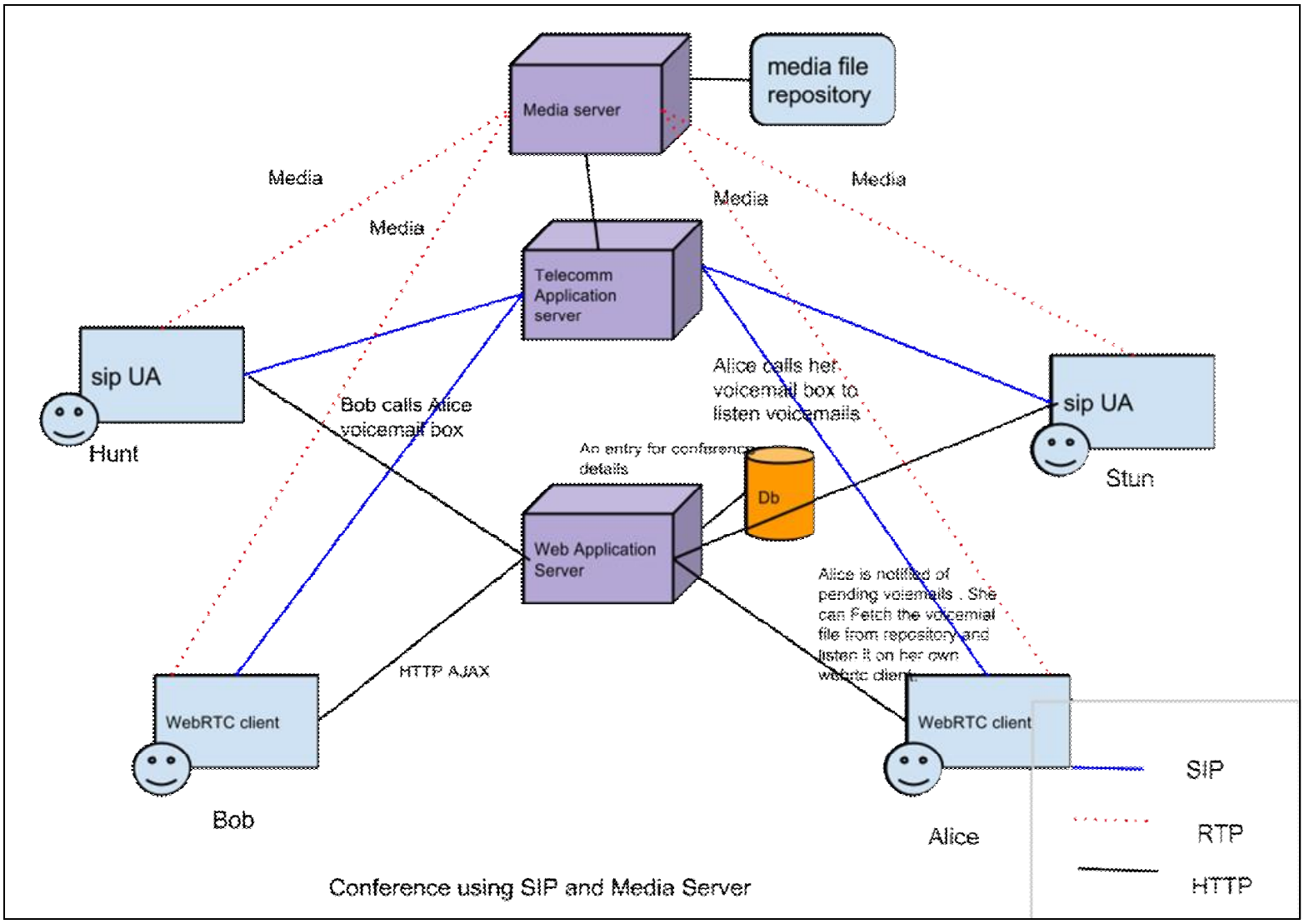
Media relay


WebRTC
SIP Server (SIP over WebSockets)
Telecom Application Server
SIP Server (SIP over WebSockets)
WebRTC
INVITE (SIPWS)
INVITE (SIP)
INVITE (SIP)
INVITE (SIPWS)
180 ringing
180 ringing
180 ringing
180 ringing
200 OK
200 OK
200 OK
200 OK
ACK
ACK
ACK
ACK
media server
audio / video media
audio / video media
BYE (SIPWS)
BYE (SIPWS)
BYE (SIPWS)
BYE (SIPWS)
200 OK
200 OK
200 OK
200 OK

Voicemail

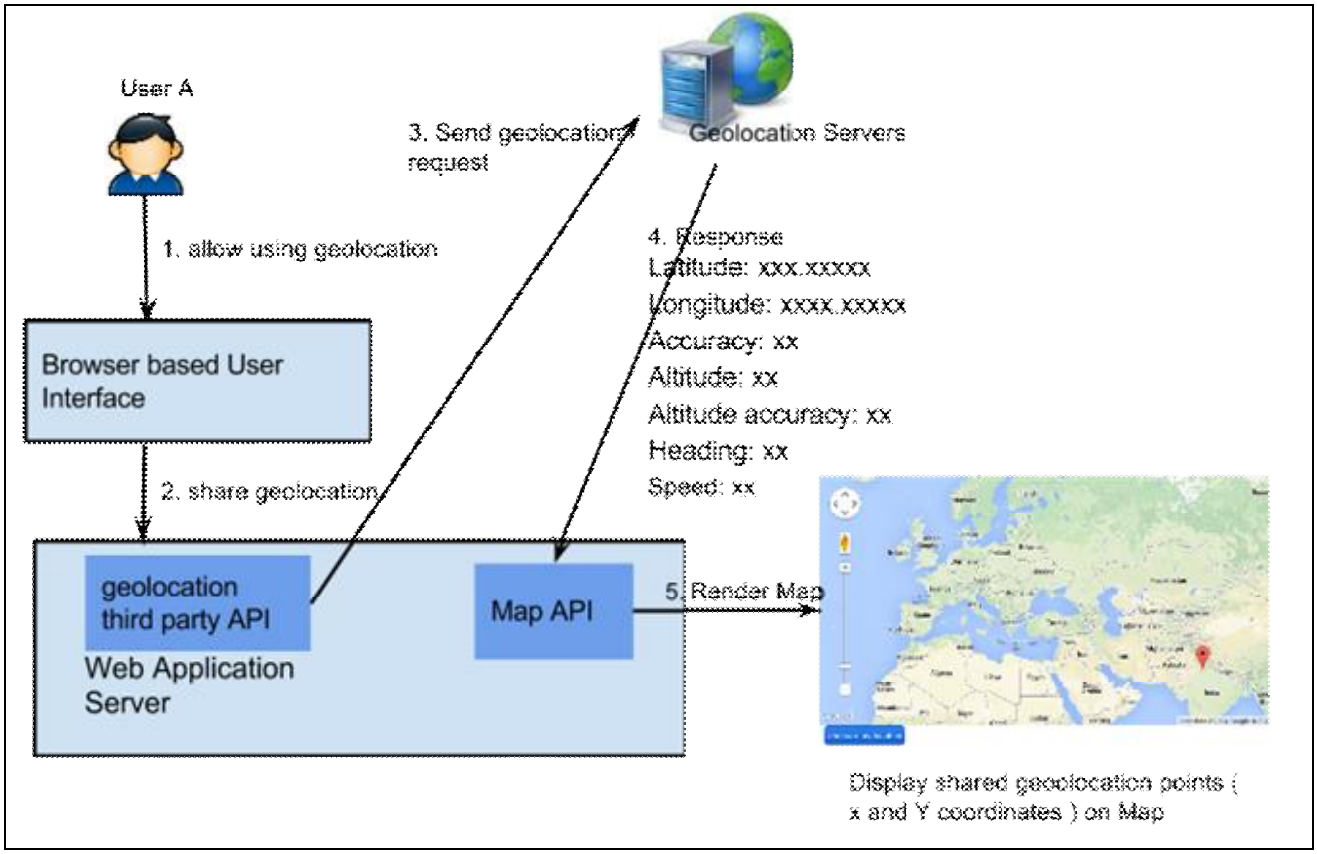

Media Server
media file repository
Teleocm Application server
Audio File is created and stored
Alice listen to her stored voicemails
SIP
RTP
HTTP
Bob calls Alice voicemail box
Alice calls her voicemail box to listen voicemails
An entry for pending voicemails with url and location of file is made in Database
Web Application Server
Db
Alice is notified of pending voicemails . She can Fetch the voicemial file from repository and listen it on her own webrtc client
HTTP AJAX
WebRTC client
WebRTC client
Bob
Alice
Voicemail using SIP and Media Server

Conferencing

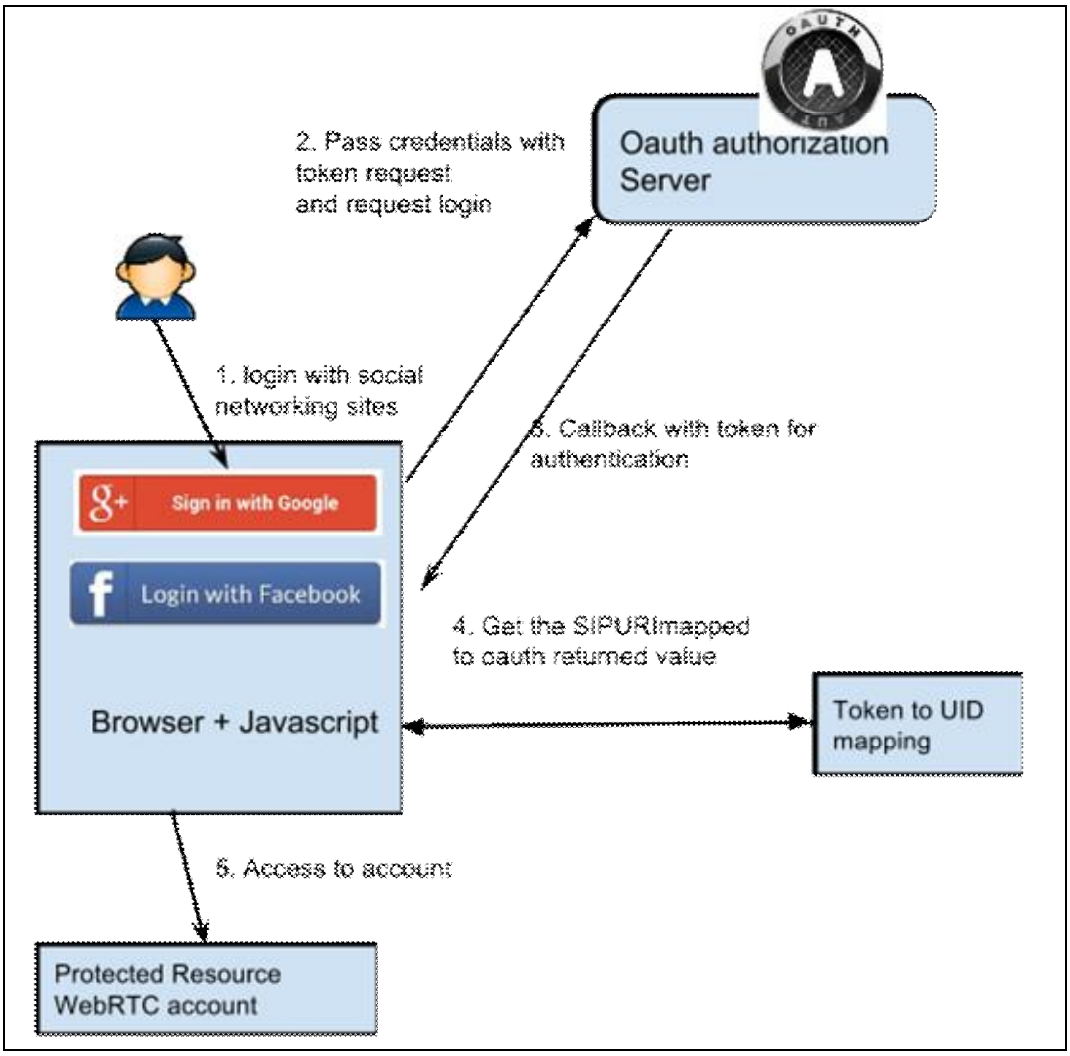


Features of a web application

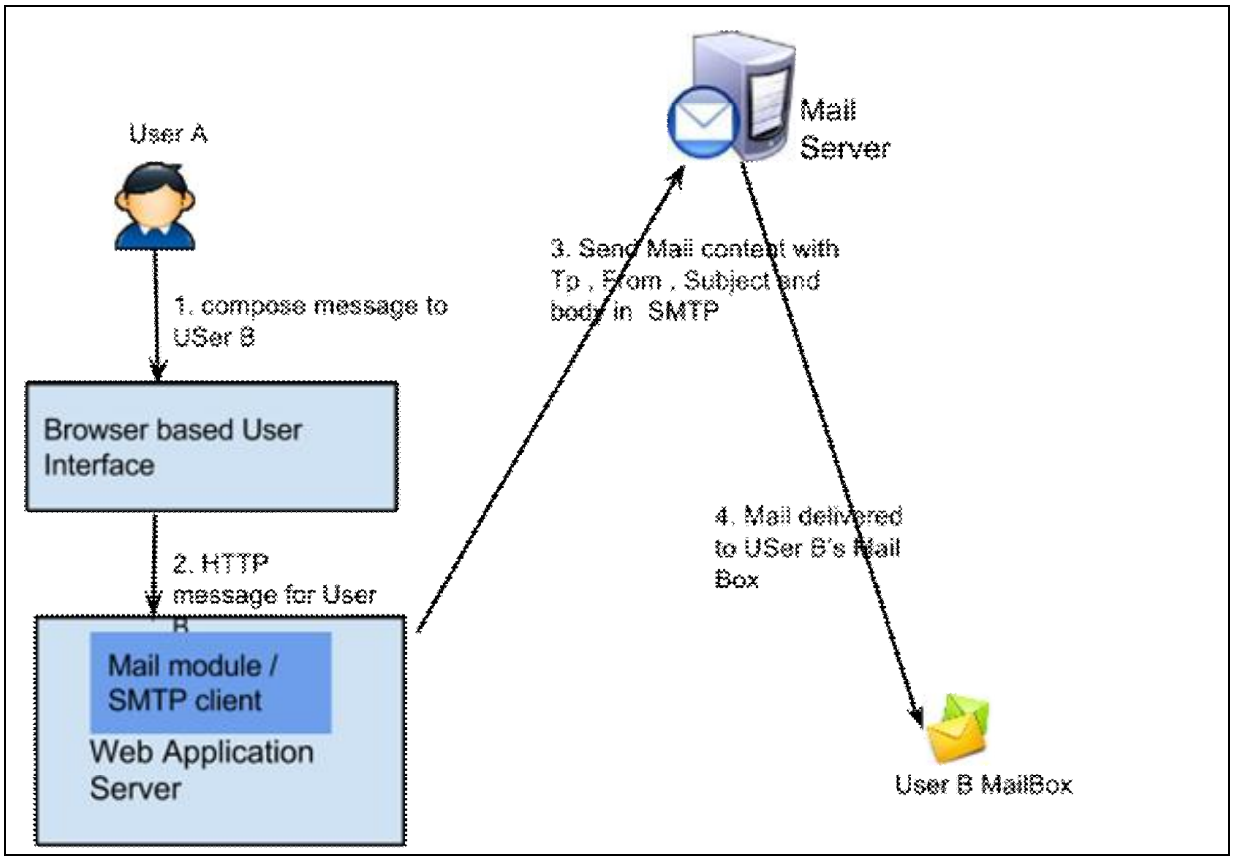
Geolocation



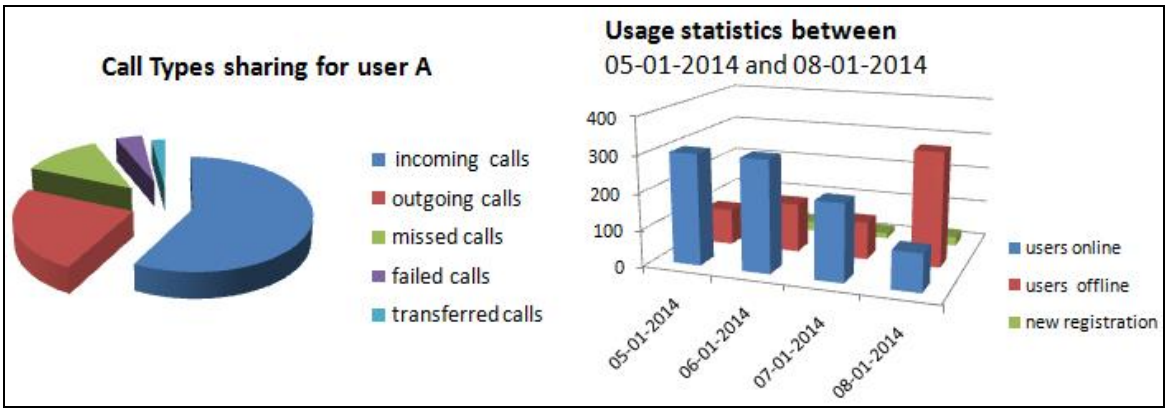
Authenticating users with OAuth



Delivering an instant message as a mail



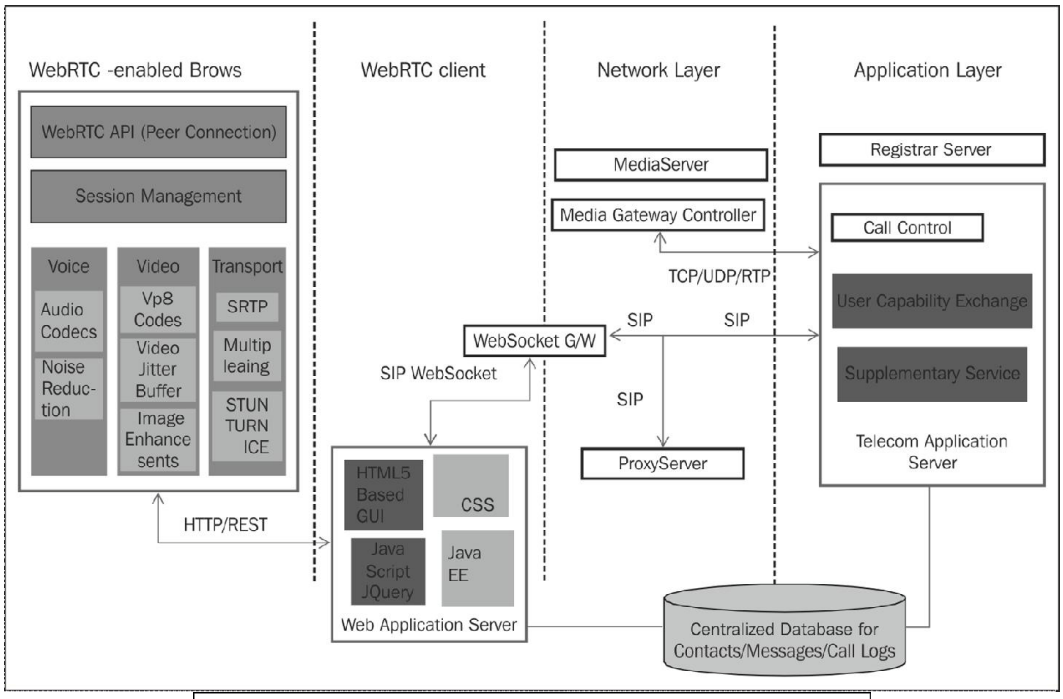
The admin console

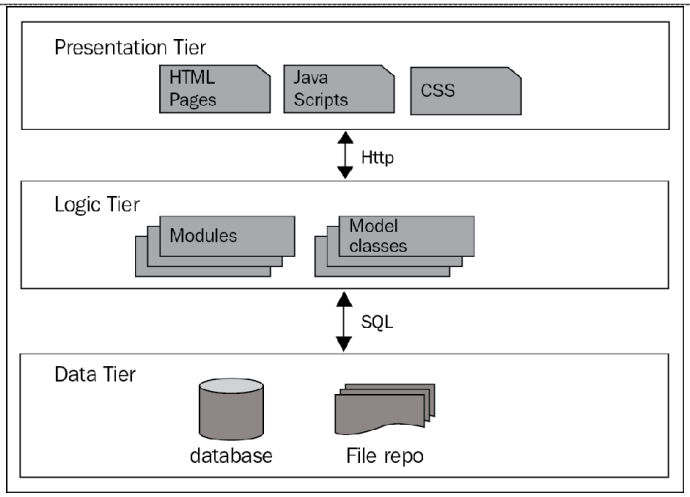


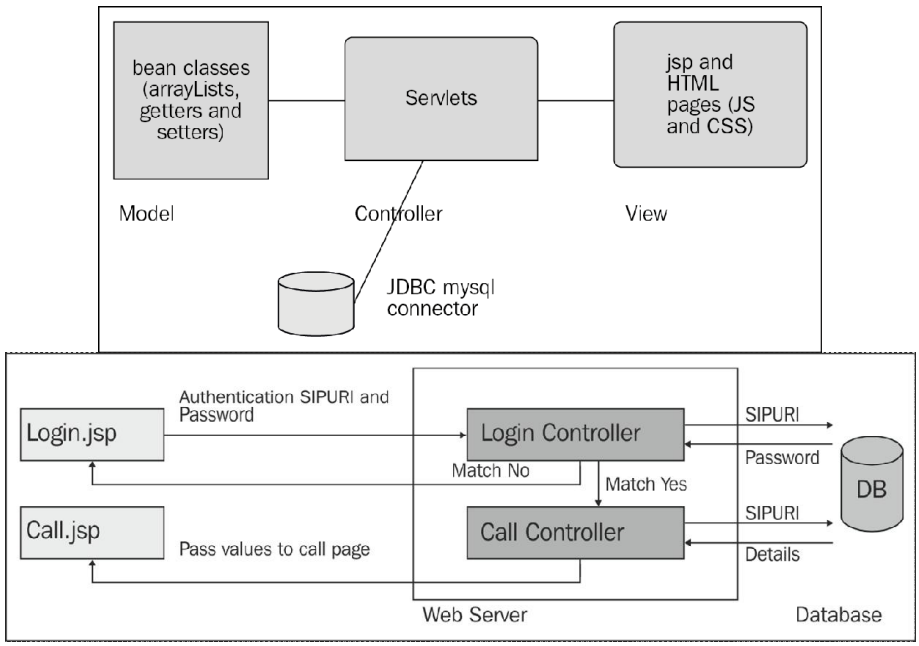
7

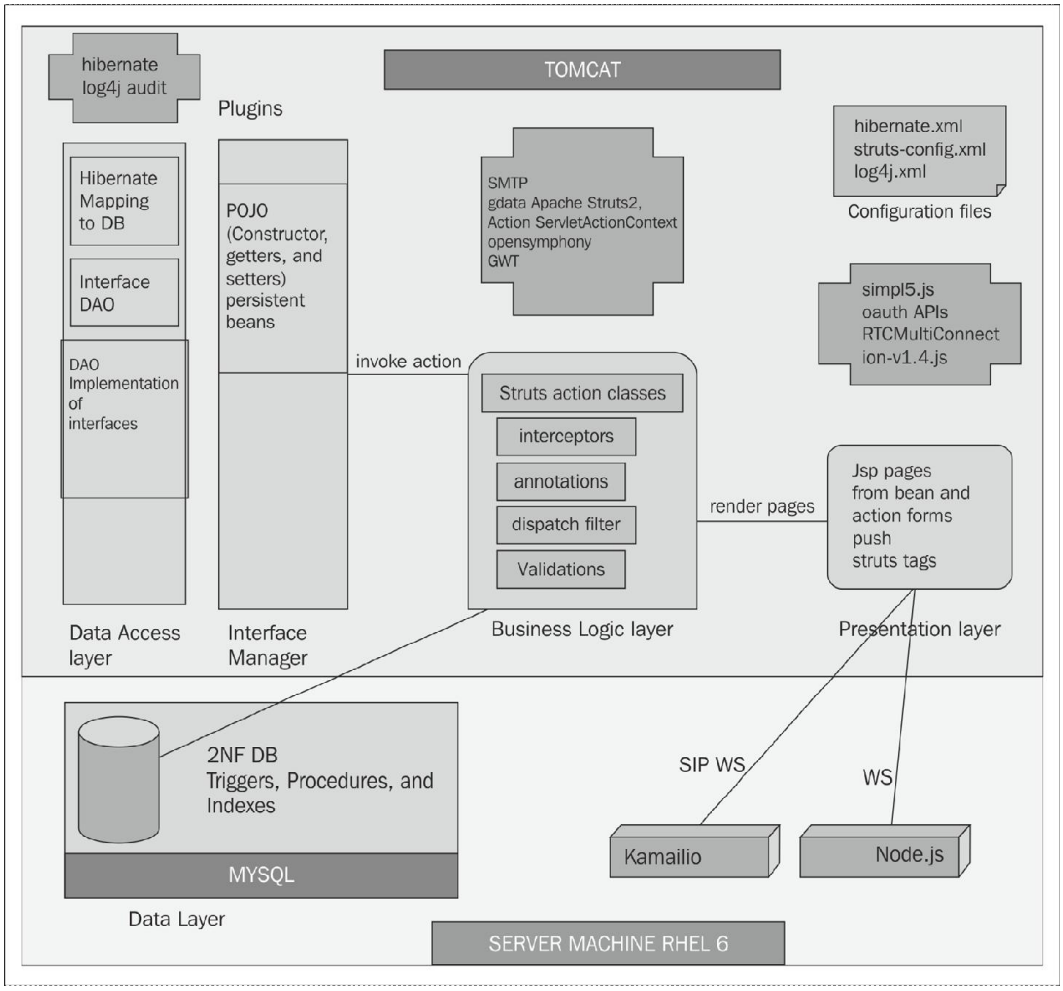
WebRTC with Industry Standard Frameworks

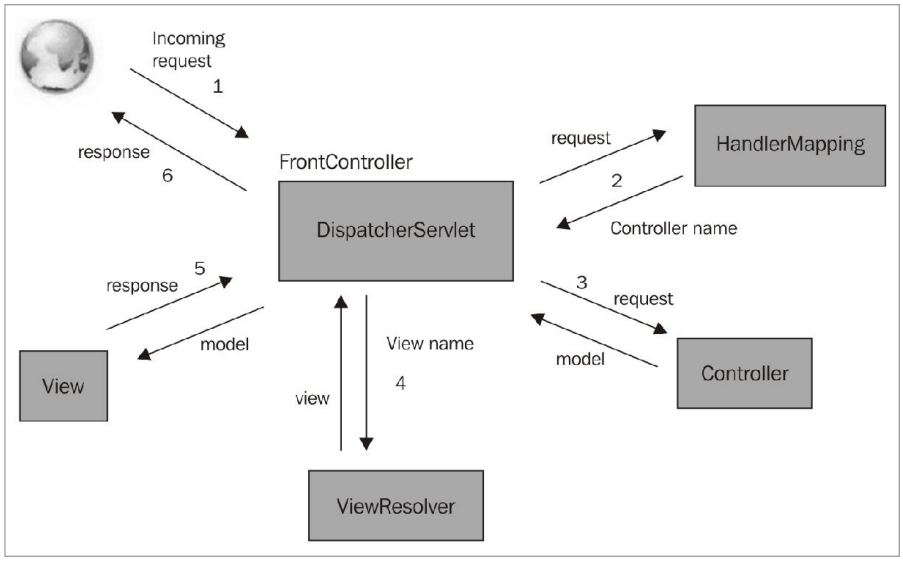
The Multitier architecture

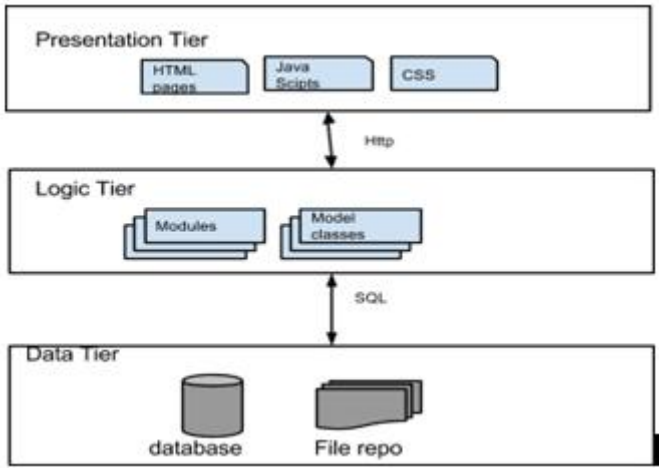





bean classes (arrayLists, getters and setters)
Servlets
jsp and HTML pages (JS and CSS)
Model
Controller
View
JDBC mysql connector
Authentication SIPURI and Password
Login.jsp
Login Controller
SIPURI
Password
Match No
Match Yes
DB
Call.jsp
Pass values to call page
Call Controller
SIPURI
Details
Web Server
Database

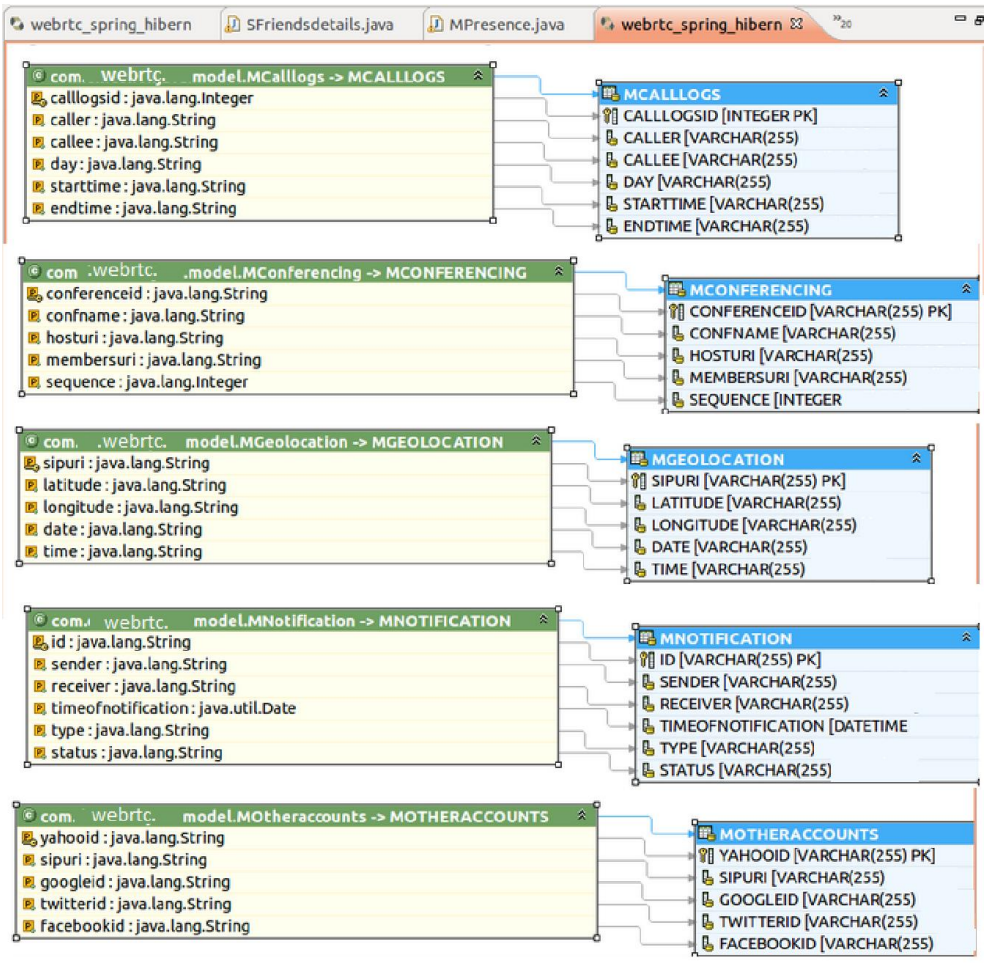

hibernate
log4j audit
TOMCAT
Plugins
hibernate.xml
struts-config.xml
log4j.xml
Configuration files
Hibernate Mapping to DB
Interface DAO
DAO Implementation of interfaces
POJO (Constructor, getters, and setters) persistent beans
SMTP
gdata Apache Struts2,
Action ServletActionContext
opensymphony
GWT
simpl5.js
oauth APIs
RTCMultiConnect
ion-v1.4.js
invoke action
Struts action classes
interceptors
annotations
dispatch filter
Validations
render pages
Jsp pages
from bean and
action forms
push
struts tags
Data Access layer
Interface Manager
Business Logic layer
Presentation layer
2NF DB
Triggers, Procedures, and Indexes
MYSQL
Data Layer
SIP WS
WS
Kamailio
Node.js
SERVER MACHINE RHEL 6


Incoming
request
1
response
6
FrontController
DispatcherServlet
request
2
Controller name
HandlerMapping
response
5
model
View
view
View name
4
3
request
model
Controller
ViewResolver

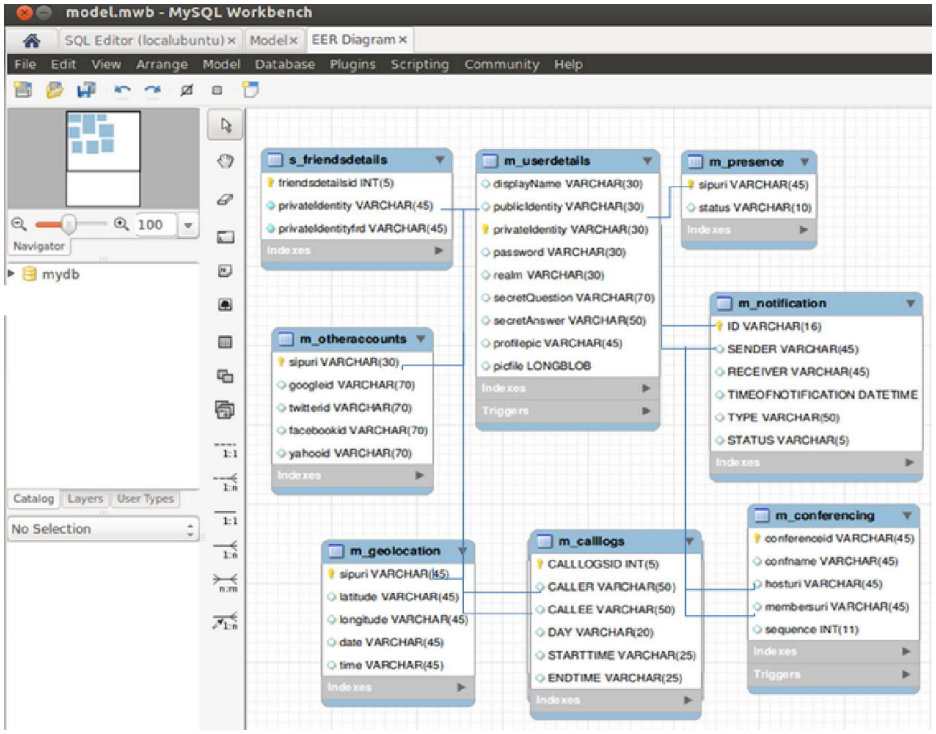

Presentation Tier
HTML pages
Java Scipts
CSS
Http
Logic Tier
Modules
Model classes
SQL
Data Tier
database
File repo

The design of a WebRTC client

The Class diagram



The Entity Relationship model

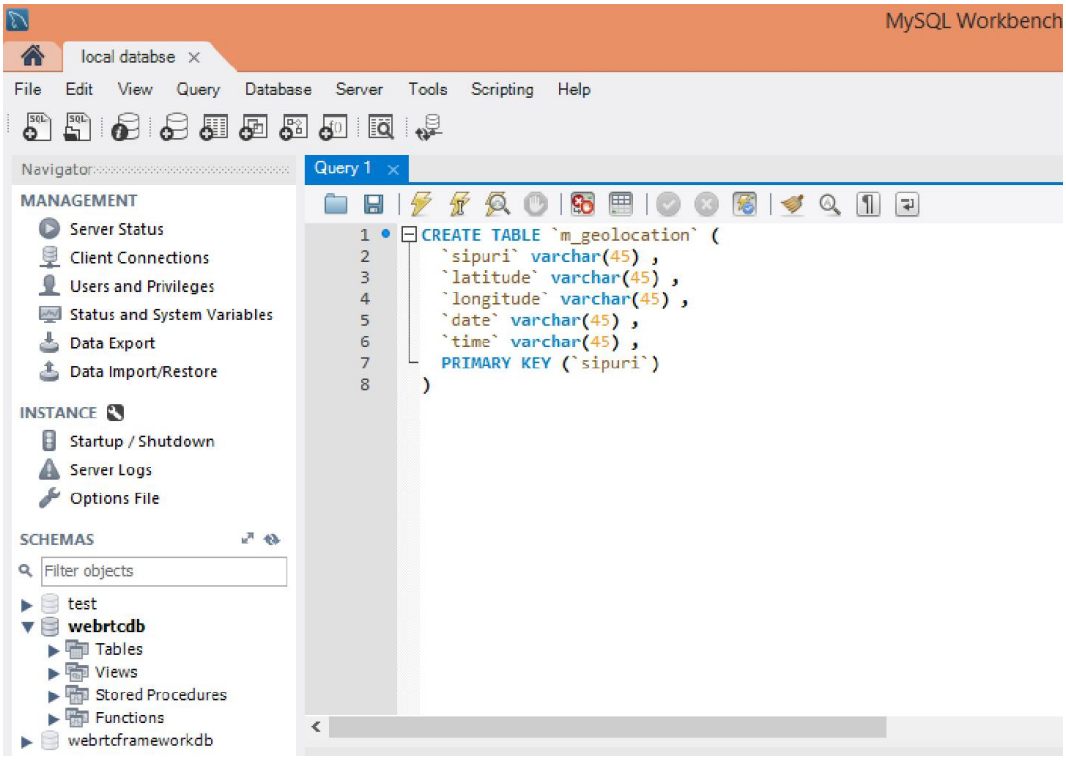


The environment setup

Java Runtime Environment (JRE)

Integrated Development Environment with Java Enterprise Edition (EE)

Databases

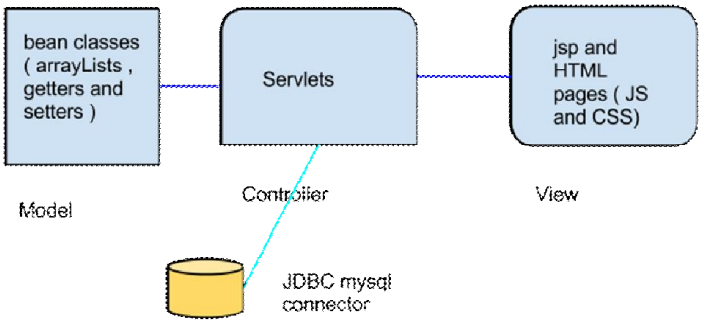

MySQL Workbench
local databse
File Edit View Query Database Server Tools Scripting Help
Navigator
MANAGEMENT
Server Status
Client Connections
Users and Privileges
Status and System Variables
Data Export
Data Import/Restore
INSTANCE
Startup / Shutdown
Server Logs
Options File
SCHEMAS
Filter objects
test
webrtcdb
Tables
Views
Stored Procedures
Functions
webrtcframeworkdb
Query 1
1 CREATE TABLE `m_geolocation` (
2 `sipuri` varchar(45) ,
3 `latitude` varchar(45) ,
4 `longitude` varchar(45) ,
5 `date` varchar(45) ,
6 `time` varchar(45) ,
7 PRIMARY KEY (`sipuri`)
8)

The web application server

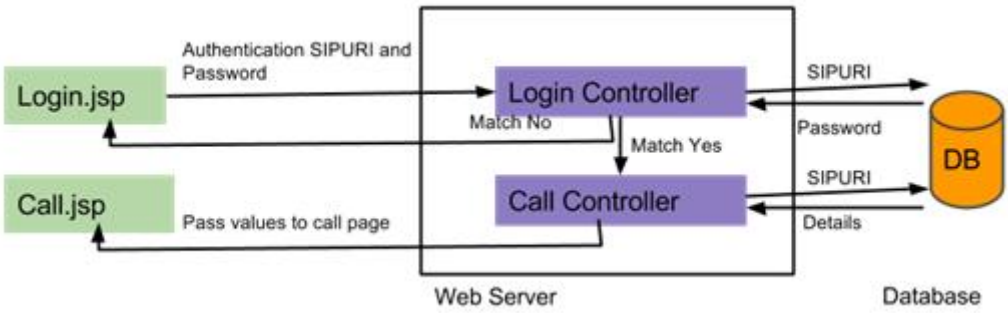
The web application infrastructure

JSP- / Servlet-based WebRTC web project

Programming the JSP- / Servlet-based web project structure



Development of modules



The User Account module

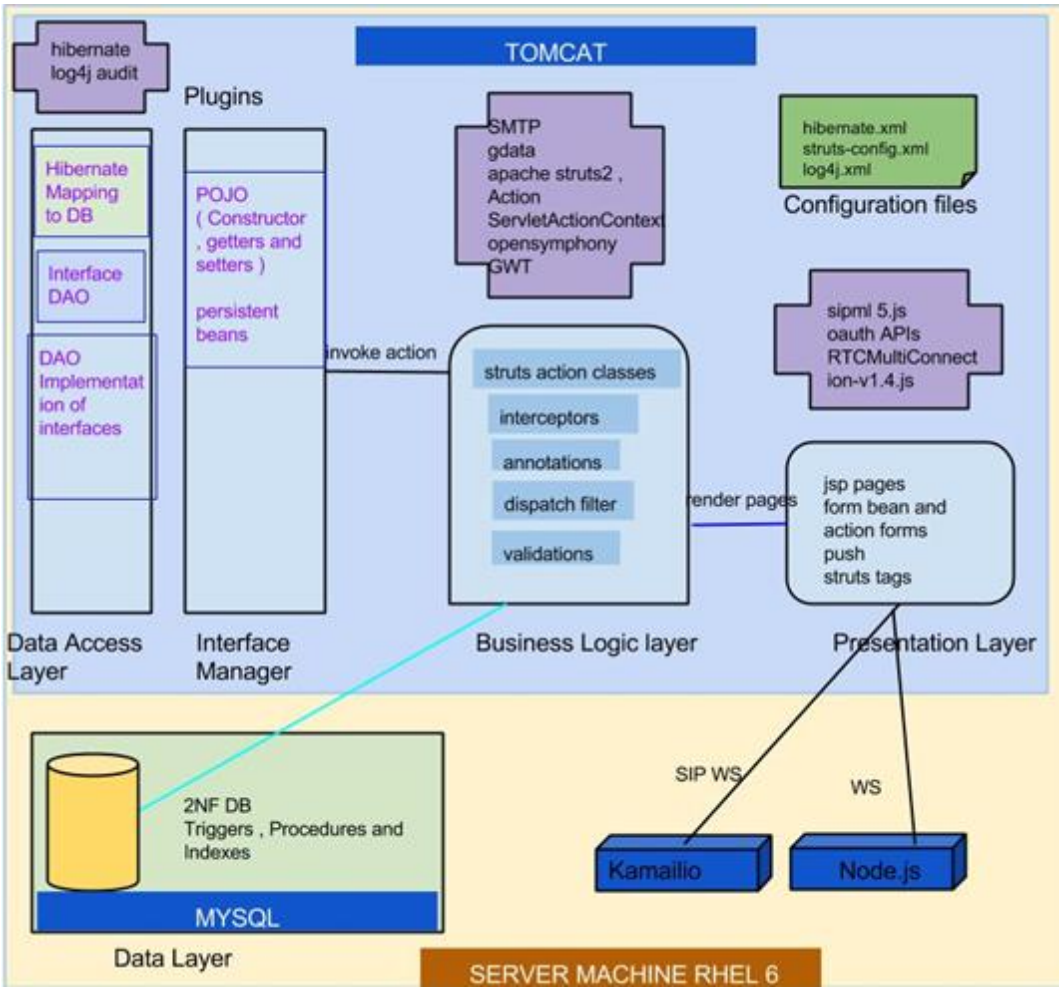
The Communication module

The Phonebook module

- WebCommunicatorv1
  - Deployment Descriptor: WebCommunicator
  - JAX-WS Web Services
  - Java Resources
    - src
      - com.webrtc.bean
        - FriendBean.java
        - Login.java
      - com.webrtc.controller
        - Controller.java
        - LoginController.java
        - PresenceServlet.java
      - com.webrtc.dao
        - MyLoginDao.java
      - com.webrtc.util
        - DbConnection.java
    - Libraries
  - JavaScript Resources
  - build
  - WebContent
    - assets
    - css
    - META-INF
    - scripts
    - sounds
    - WEB-INF
    - Dashboard.jsp

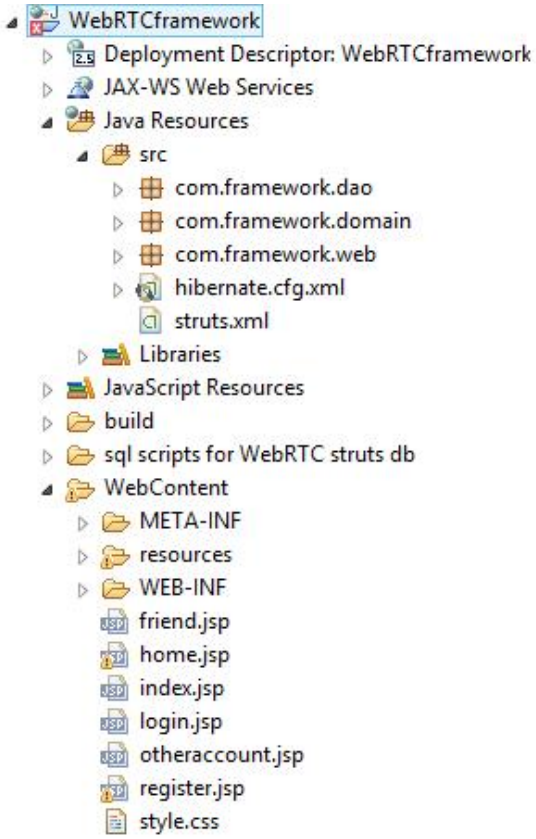
Struts- / Hibernate-based WebRTC web project

Programming the Struts- / Hibernate-based web project structure



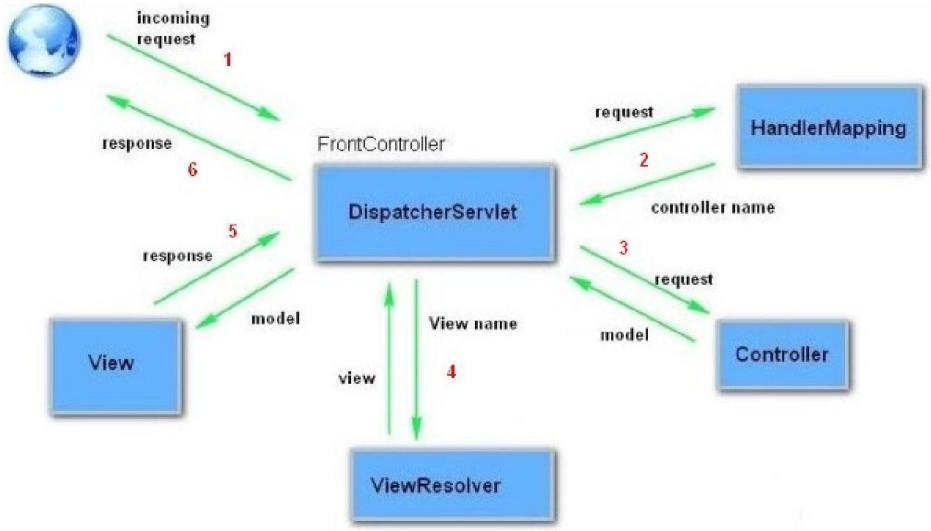
Development of modules

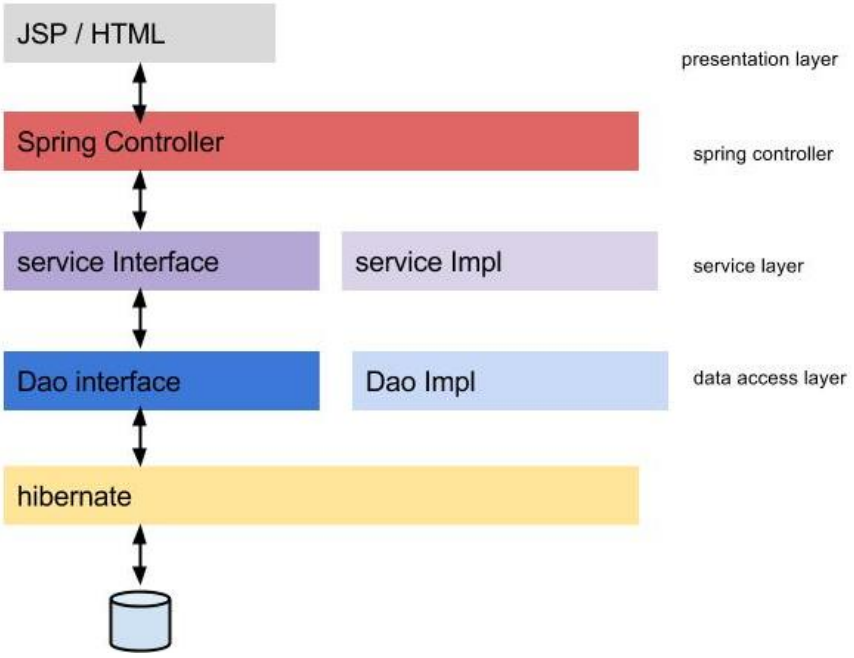
The OtherAccount module

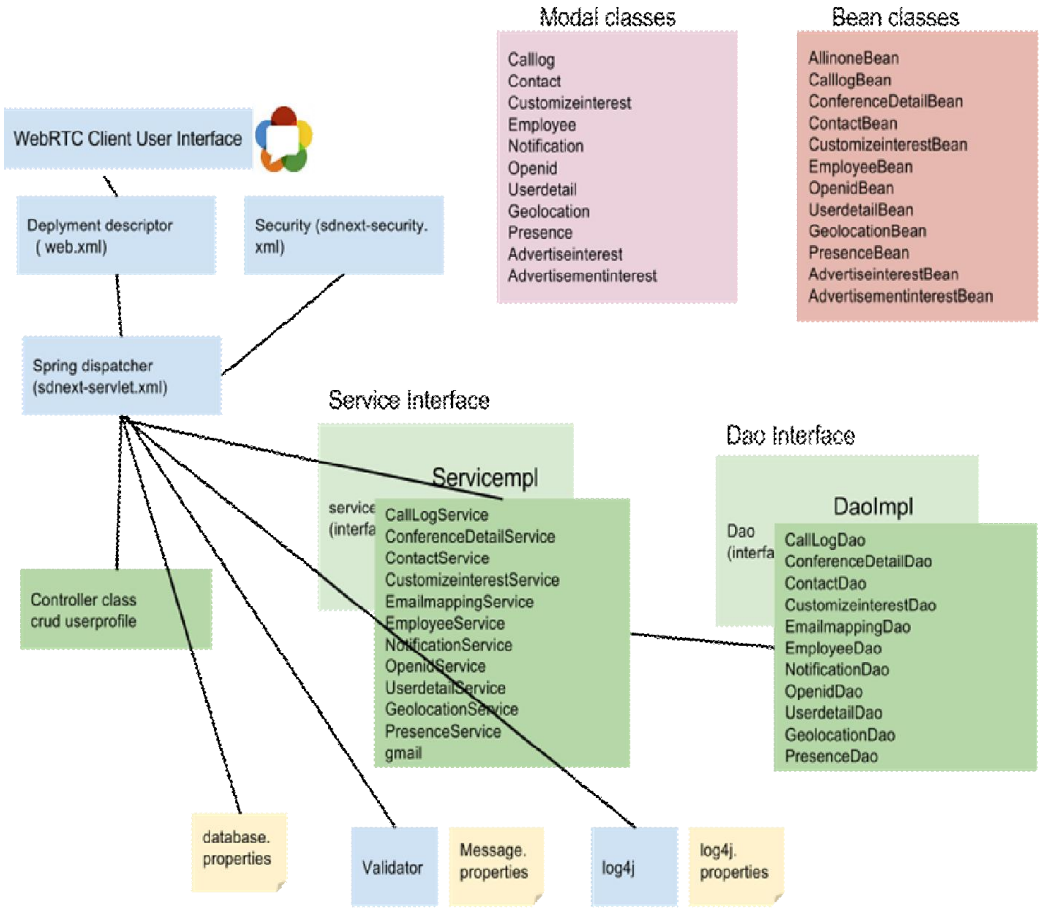

WebRTCframework
Deployment Descriptor: WebRTCframework
JAX-WS Web Services
Java Resources
src
com.framework.dao
com.framework.domain
com.framework.web
hibernate.cfg.xml
struts.xml
Libraries
JavaScript Resources
build
sql scripts for WebRTC struts db
WebContent
META-INF
resources
WEB-INF
friend.jsp
home.jsp
index.jsp
login.jsp
otheraccount.jsp
register.jsp
style.css

Spring 3 MVC-based WebRTC web project

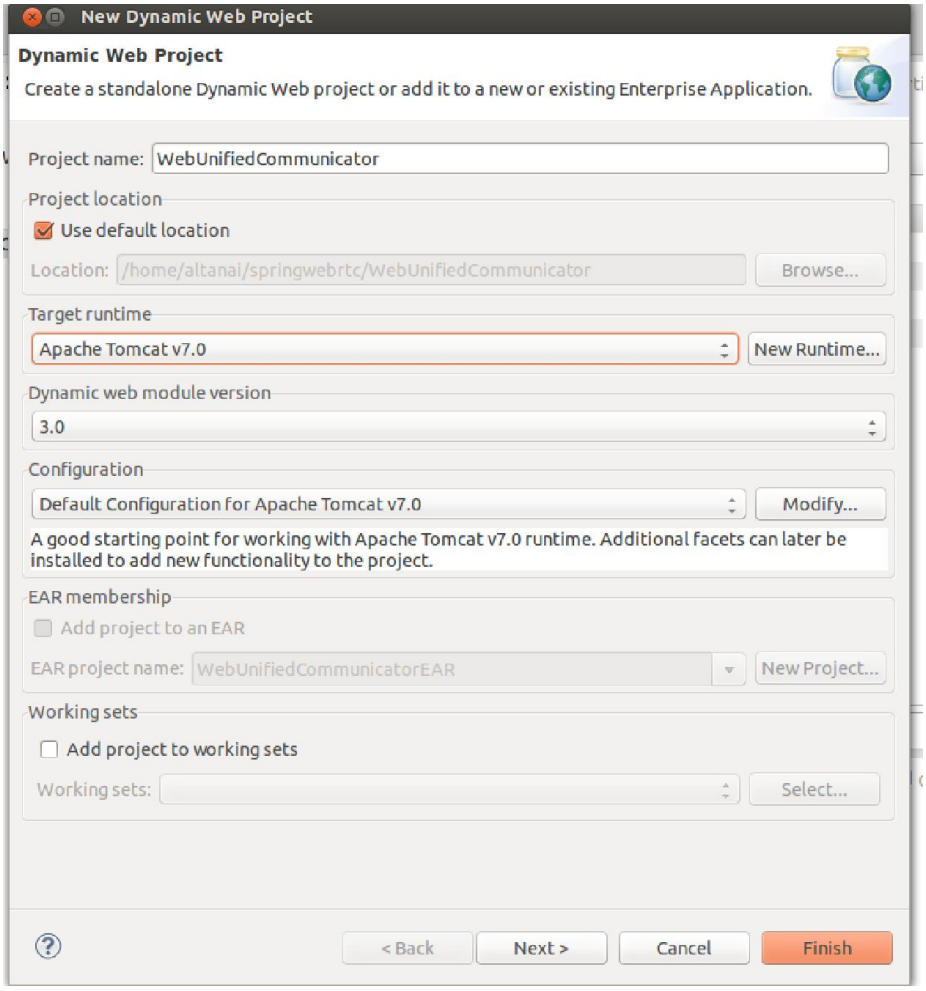
Programming the Spring 3 MVC web project structure



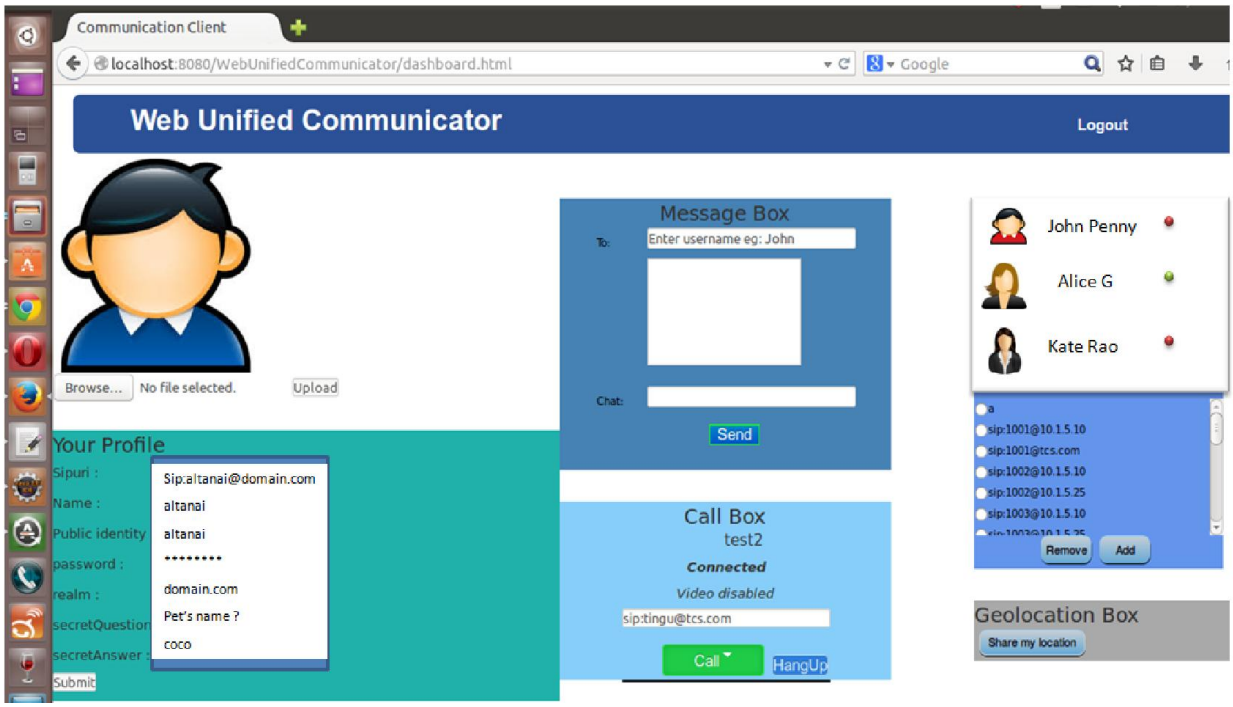


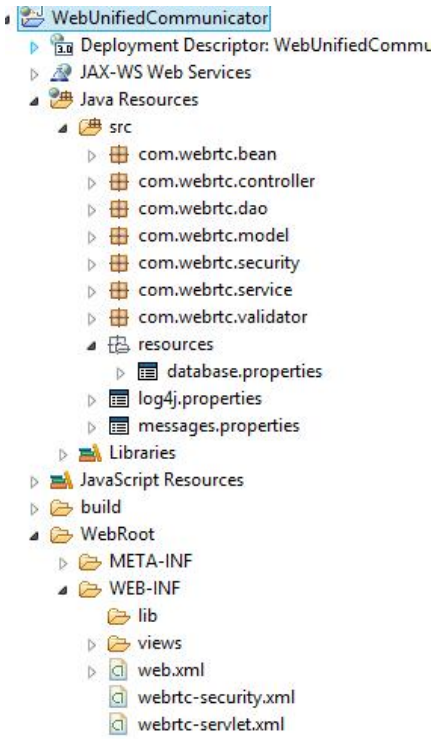

Modal classes
Calllog
Contact
Customizeinterest
Employee
Notification
Openid
Userdetail
Geolocation
Presence
Advertiseinterest
Advertisementinterest
Bean classes
AllinoneBean
CalllogBean
ConferenceDetailBean
ContactBean
CustomizeinterestBean
EmployeeBean
OpenidBean
UserdetailBean
GeolocationBean
PresenceBean
AdvertiseinterestBean
AdvertisementinterestBean
WebRTC Client User Interface
Deplyment descriptor (web.xml)
Security (sdnext-security. xml)
Spring dispatcher (sdnext-servlet.xml)
Service Interface
Servicempl
service (interfa
CallLogService
ConferenceDetailService
ContactService
CustomizeinterestService
EmailmappingService
EmployeeService
NotificationService
OpenidService
UserdetailService
GeolocationService
PresenceService
gmail
Dao Interface
DaoImpl
Dao (interfa
CallLogDao
ConferenceDetailDao
ContactDao
CustomizeinterestDao
EmailmappingDao
EmployeeDao
NotificationDao
OpenidDao
UserdetailDao
GeolocationDao
PresenceDao
Controller class crud userprofile
database. properties
Validator
Message. properties
log4j
log4j. properties

Development of modules



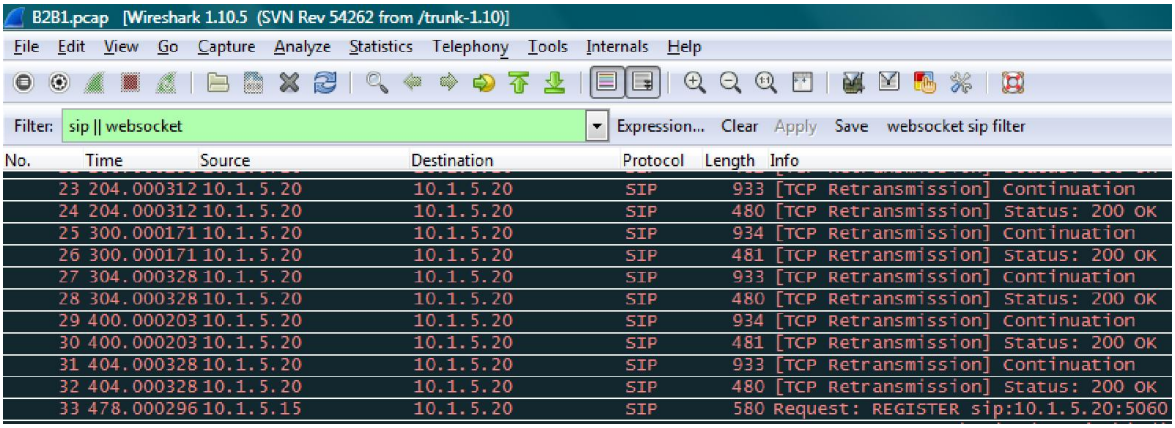
The Geolocation module


Communication Client
localhost:8080/WebUnifiedCommunicator/dashboard.html
Google
Web Unified Communicator
Logout
Browse...
No file selected.
Upload
Your Profile
Sipuri :
Name :
Public identity
password :
realm :
secretQuestion
secretAnswer :
Submit
Sip:altanai@domain.com
altanai
altanai
••••••••
domain.com
Pet's name ?
coco
Message Box
To:
Enter username eg: John
Chat:
Send
Call Box
test2
Connected
Video disabled
sip:tingu@tcs.com
Call
HangUp
John Penny
Alice G
Kate Rao
a
sip:1001@10.1.5.10
sip:1001@tcs.com
sip:1002@10.1.5.10
sip:1002@10.1.5.25
sip:1003@10.1.5.10
Remove
Add
Geolocation Box
Share my location



Testing

Testing the signal flow



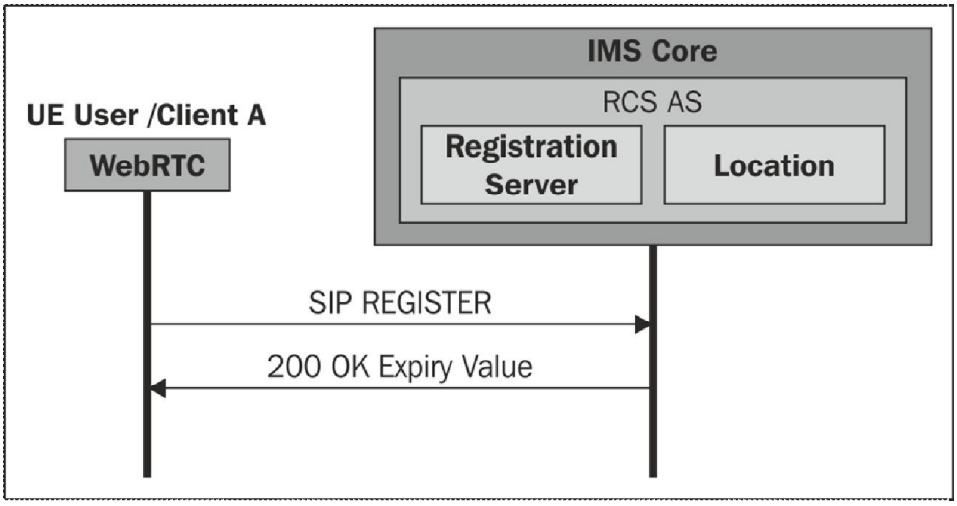
8

WebRTC and Rich Communication Services

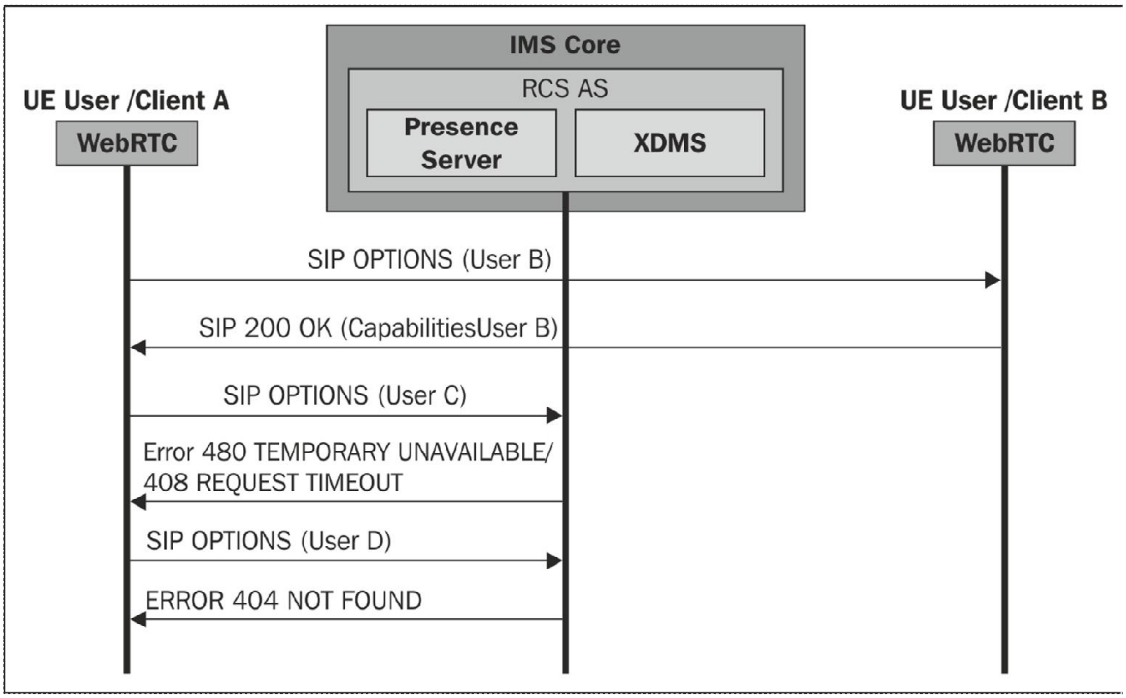
Rich Communication Services

RCS specifications

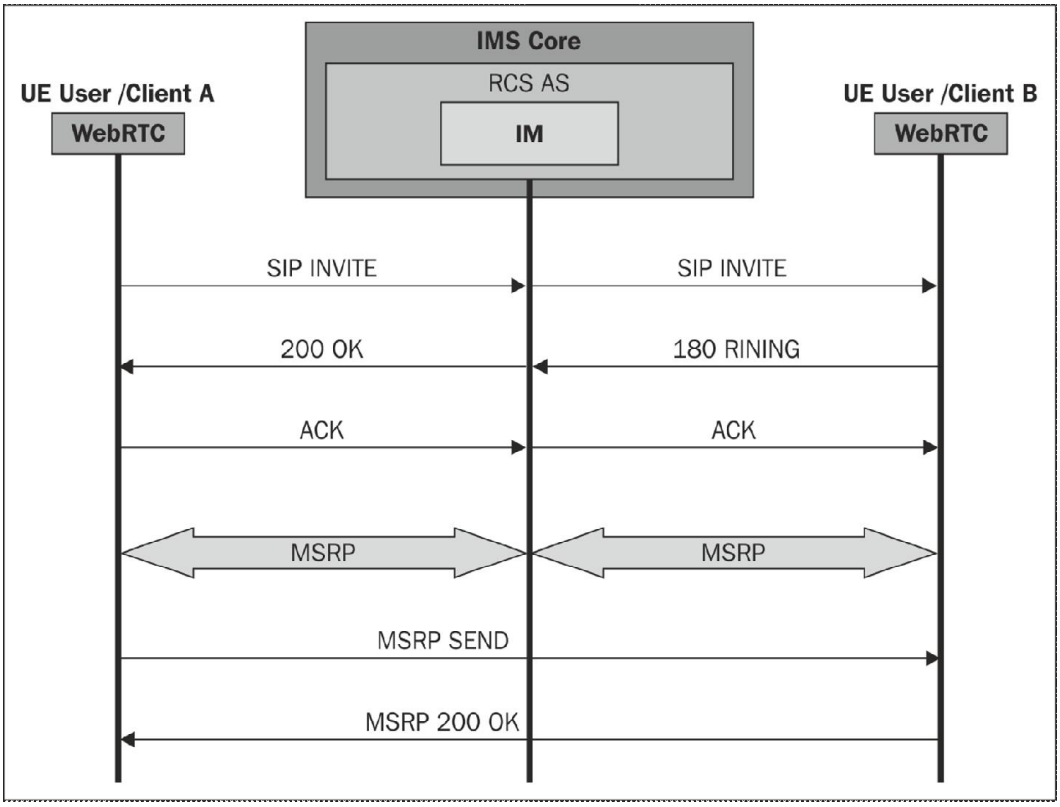
Service discovery by an RCS-enabled device

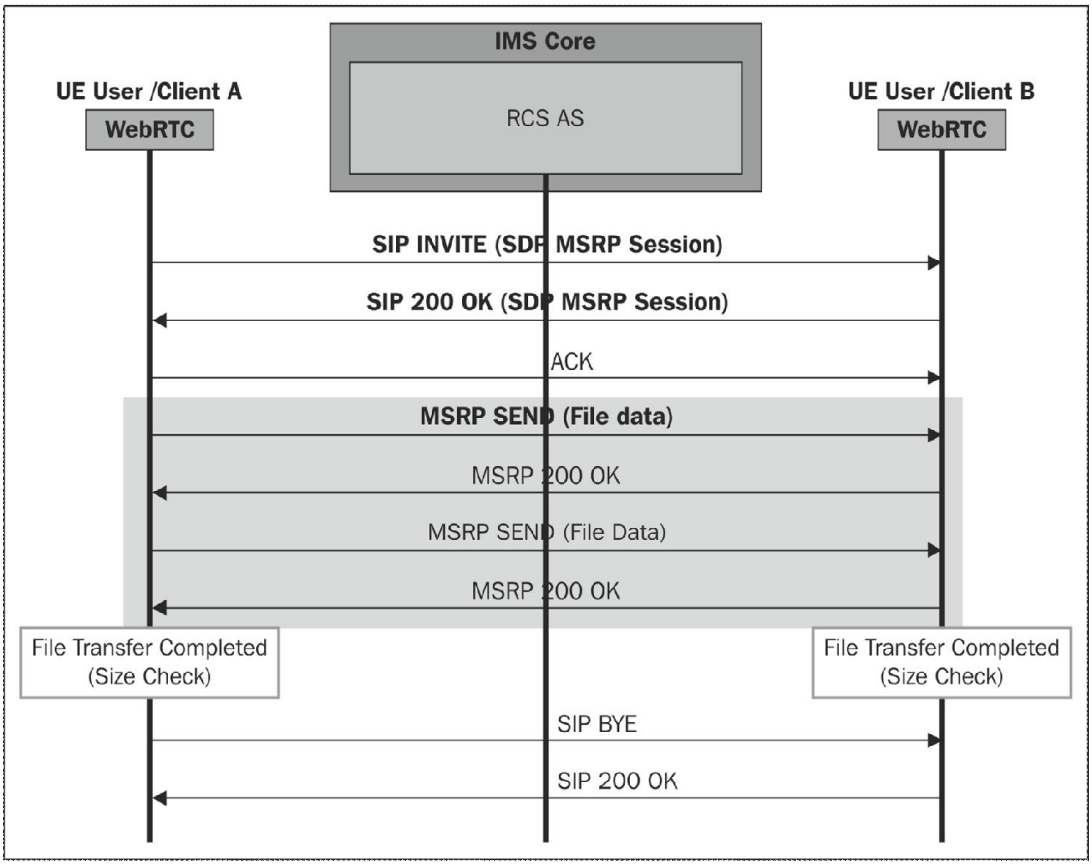


User capability exchange

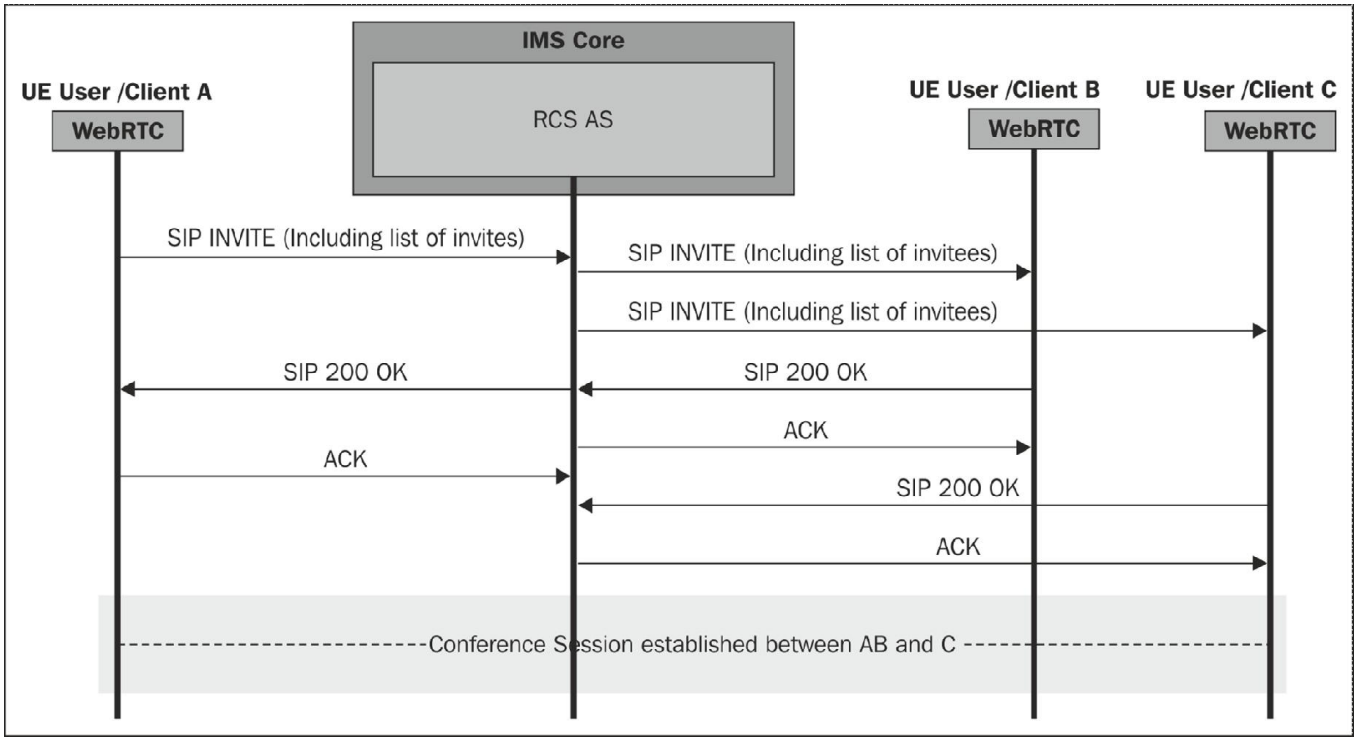


Chats with multimedia sharing

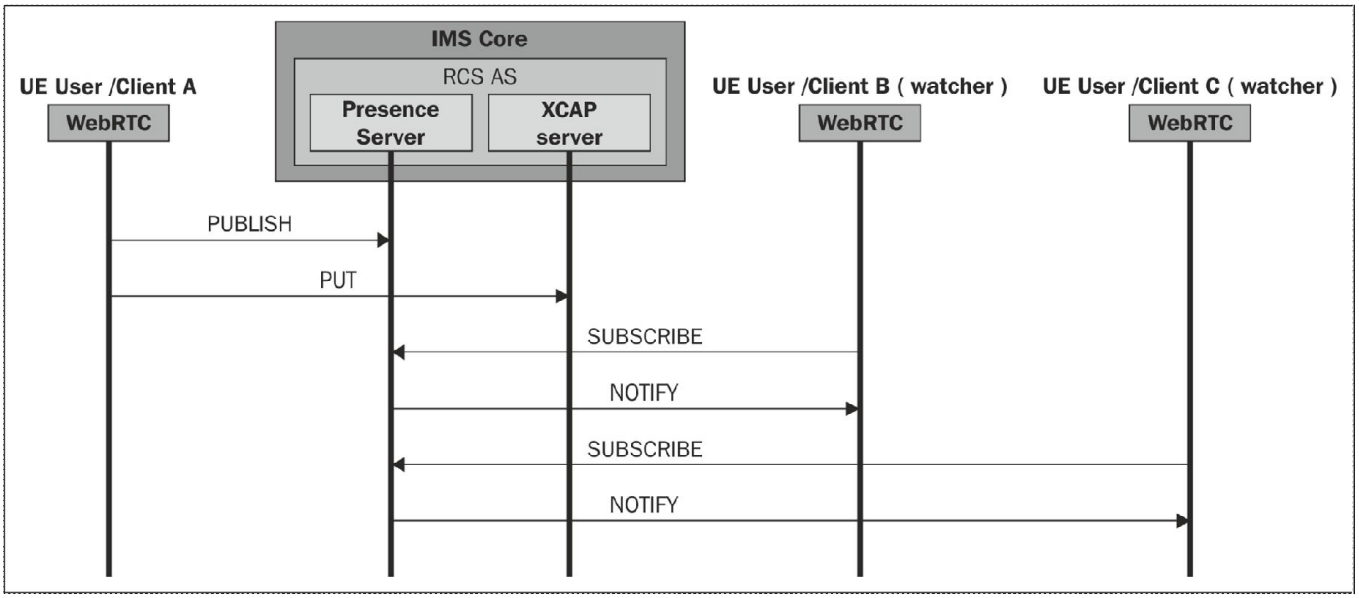



IMS Core
RCS AS
UE User /Client A
WebRTC
UE User /Client B
WebRTC
SIP INVITE (SDP MSRP Session)
SIP 200 OK (SDP MSRP Session)
ACK
MSRP SEND (File data)
MSRP 200 OK
MSRP SEND (File Data)
MSRP 200 OK
File Transfer Completed (Size Check)
File Transfer Completed (Size Check)
SIP BYE
SIP 200 OK

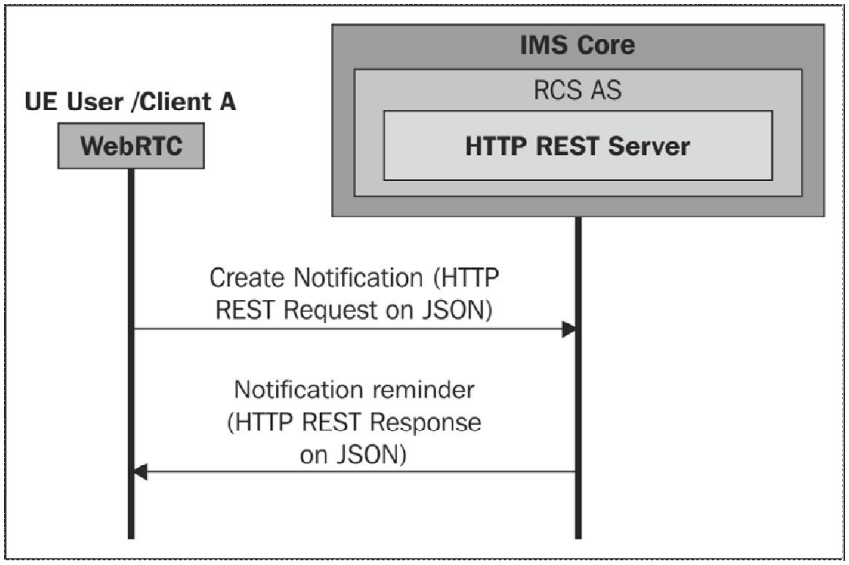
Group chat in a conference session



User availability through XCAP

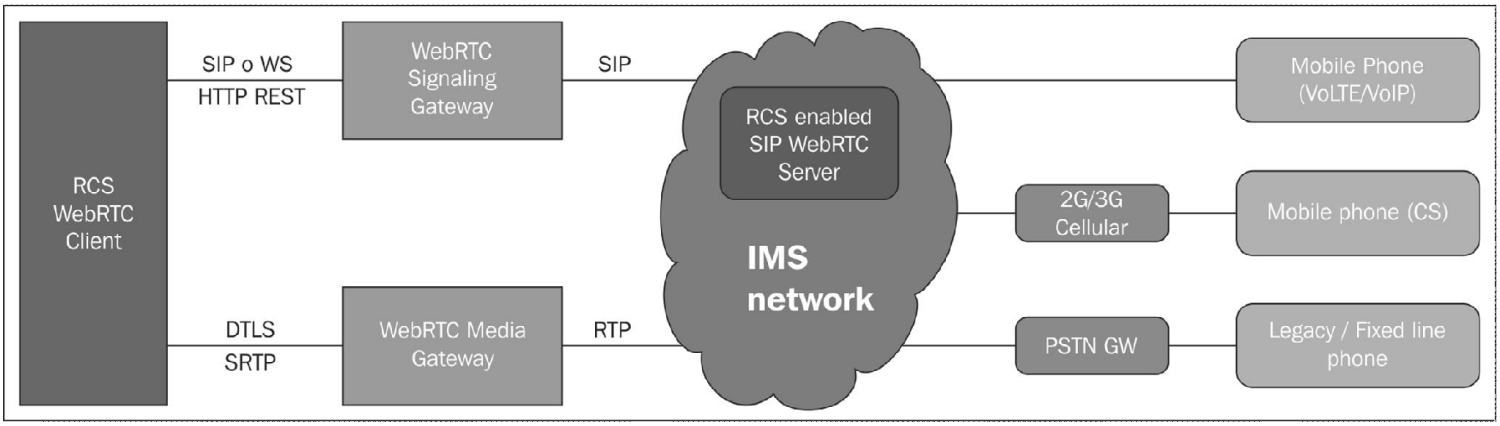


REST-based notifications



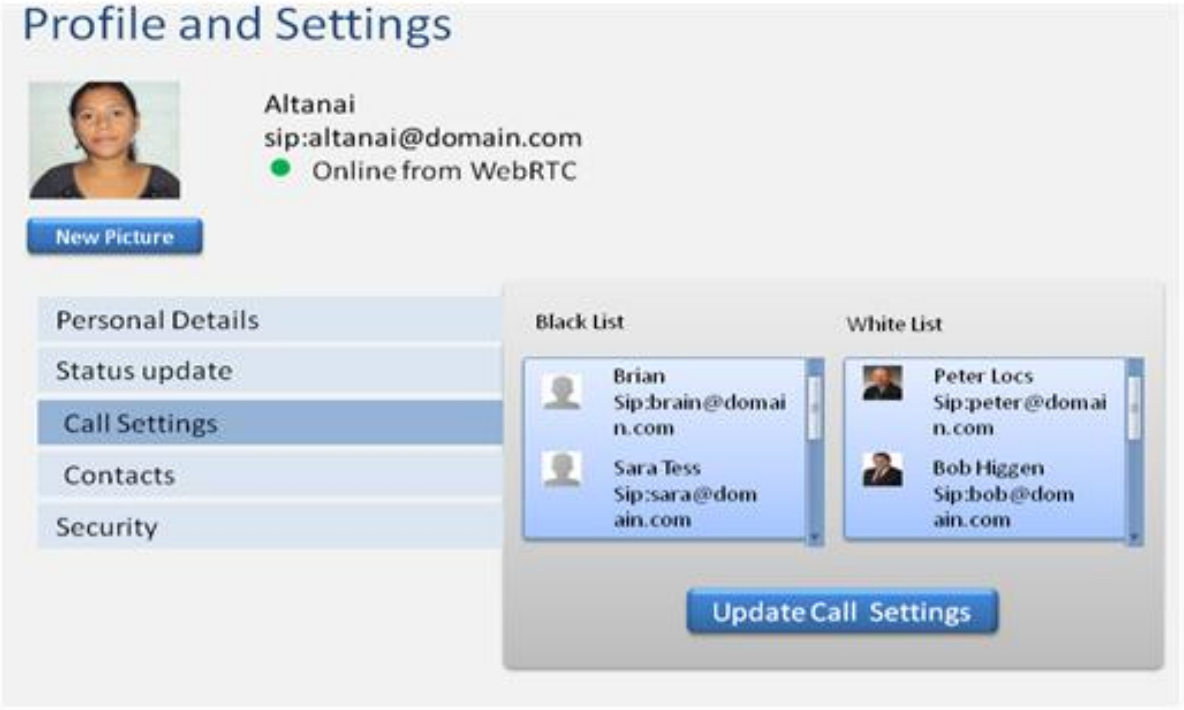
Interoperability and interworking

The RCS ecosystem and WebRTC



RCS services in WebRTC

User profile



Integration with social networks

Profile and Settings

Altanai
sip:altanai@domain.com
Online from WebRTC

New Picture

- Personal Details
- Status update
- Call Settings
- Contacts
- Security

Import contacts from Facebook

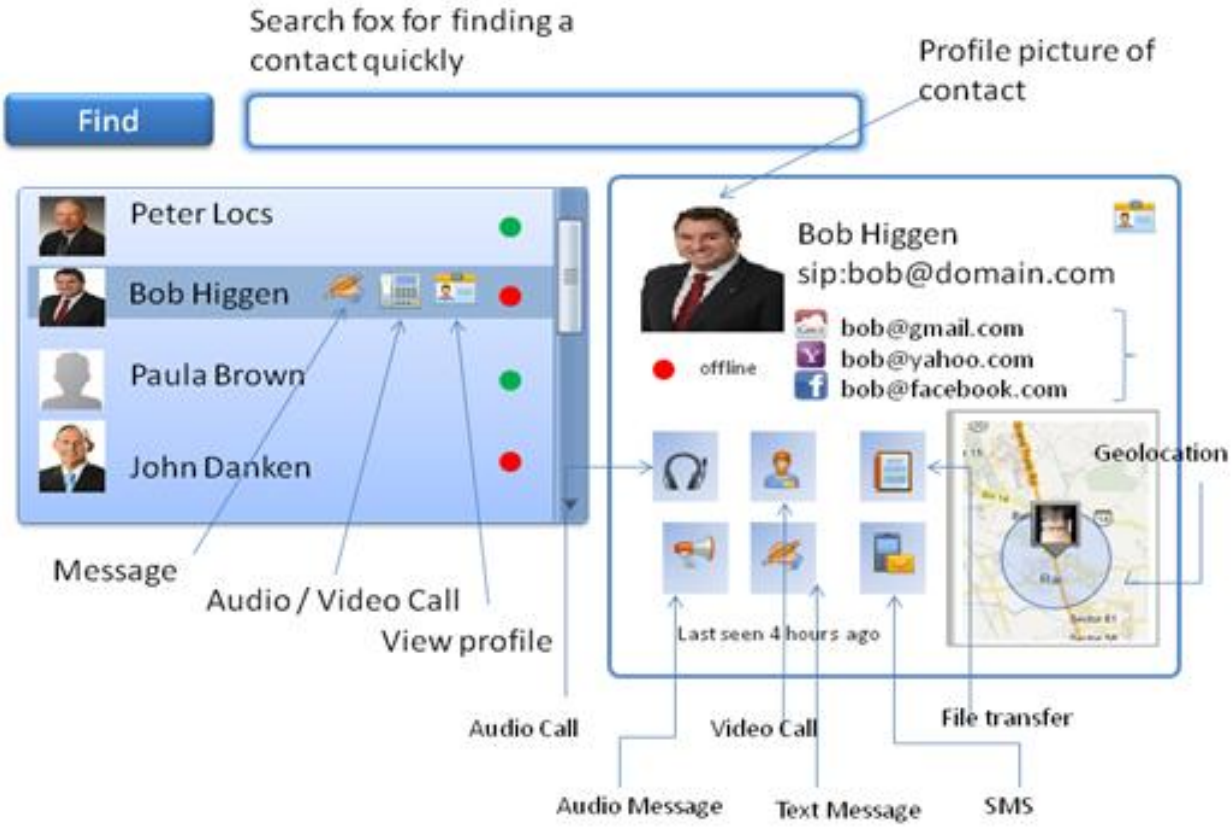
Import contacts from Google

Import contacts from Yahoo

Import contacts from .csv file

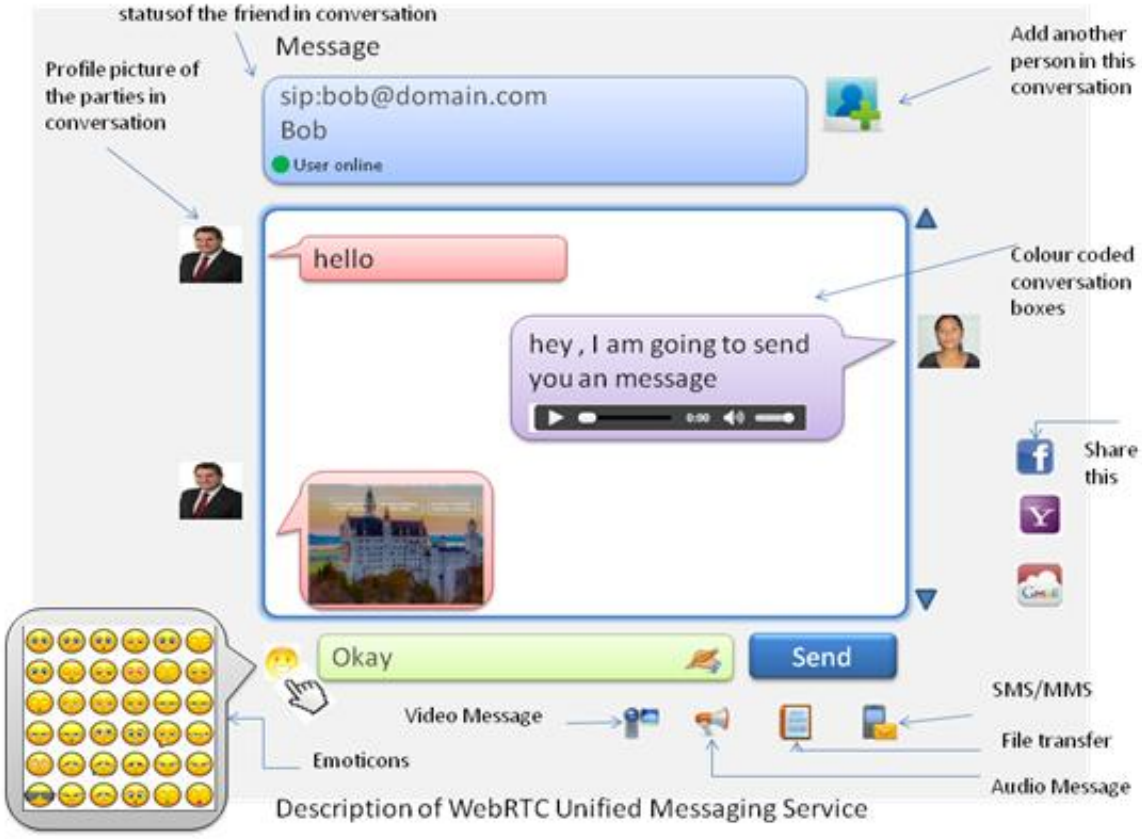
Goto Phonebook

The enhanced phonebook



User capabilities and Presence

Unified messaging box



Group Messaginig
sip:bob@domain.com (Bob Higgens)
sip:peter@domain.com (Peter Locs)
sip:altanai@domain.com (Altanai)- Me
Hello everyone lets begin with project status
Peter locs
Hello , sure Peter . Bob please share the plants location
Bob Higgens
Okay
Send

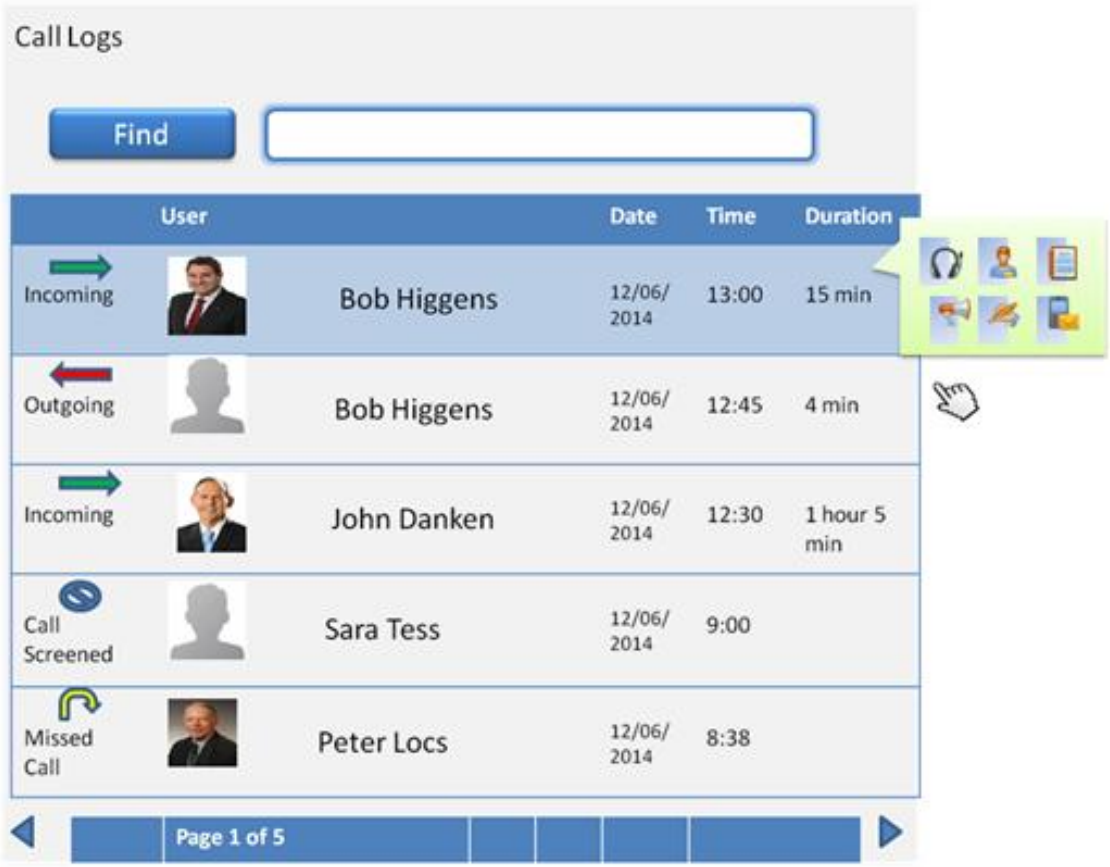
Message history

Rich calls

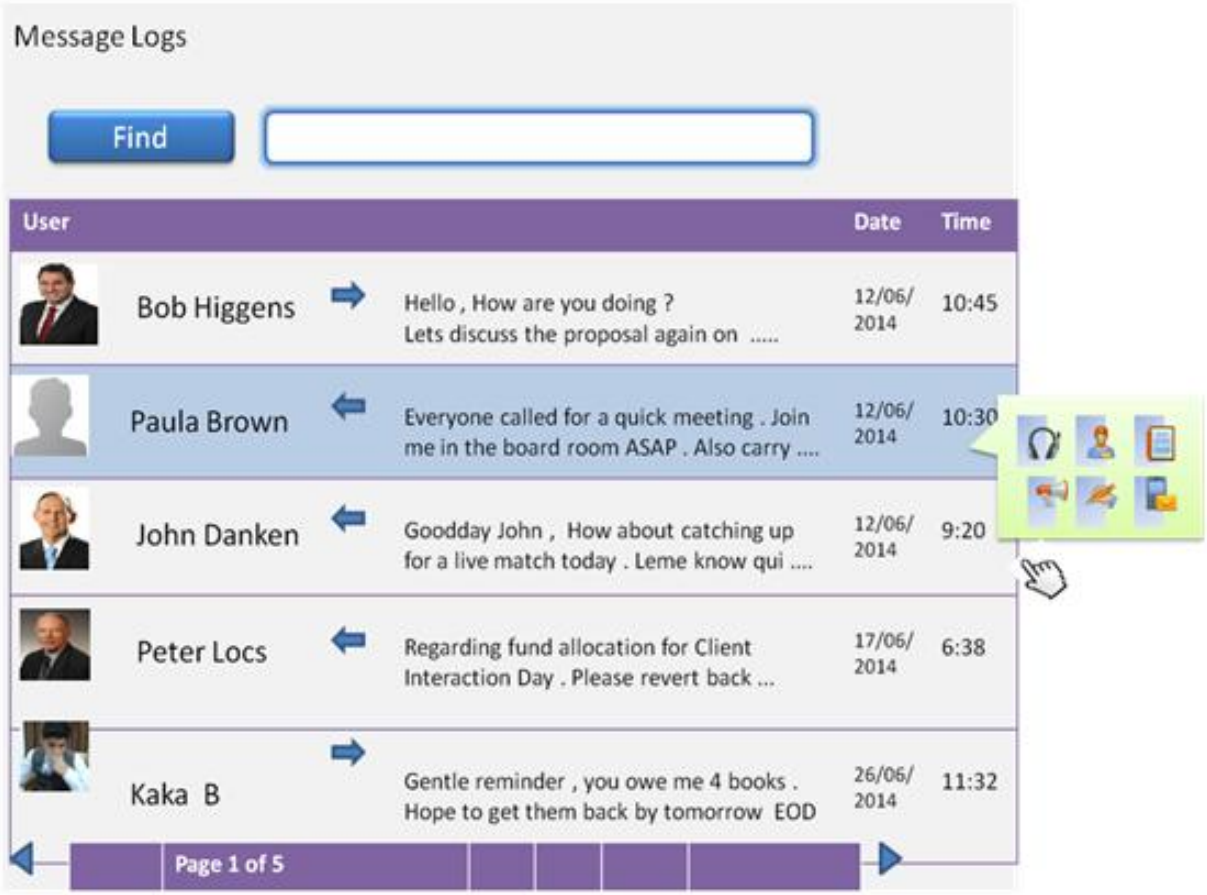
Video Call
sip:bob@domain.com
Bob Higgens
sip:bob@domain.com
User online
Call
Hold/Resume
Hangup
In Call

Desktop Sharing
sip:bob@domain.com
Bob Higgens
sip:bob@domain.com
User online
Call
Hold/Resume
Hangup
In Call

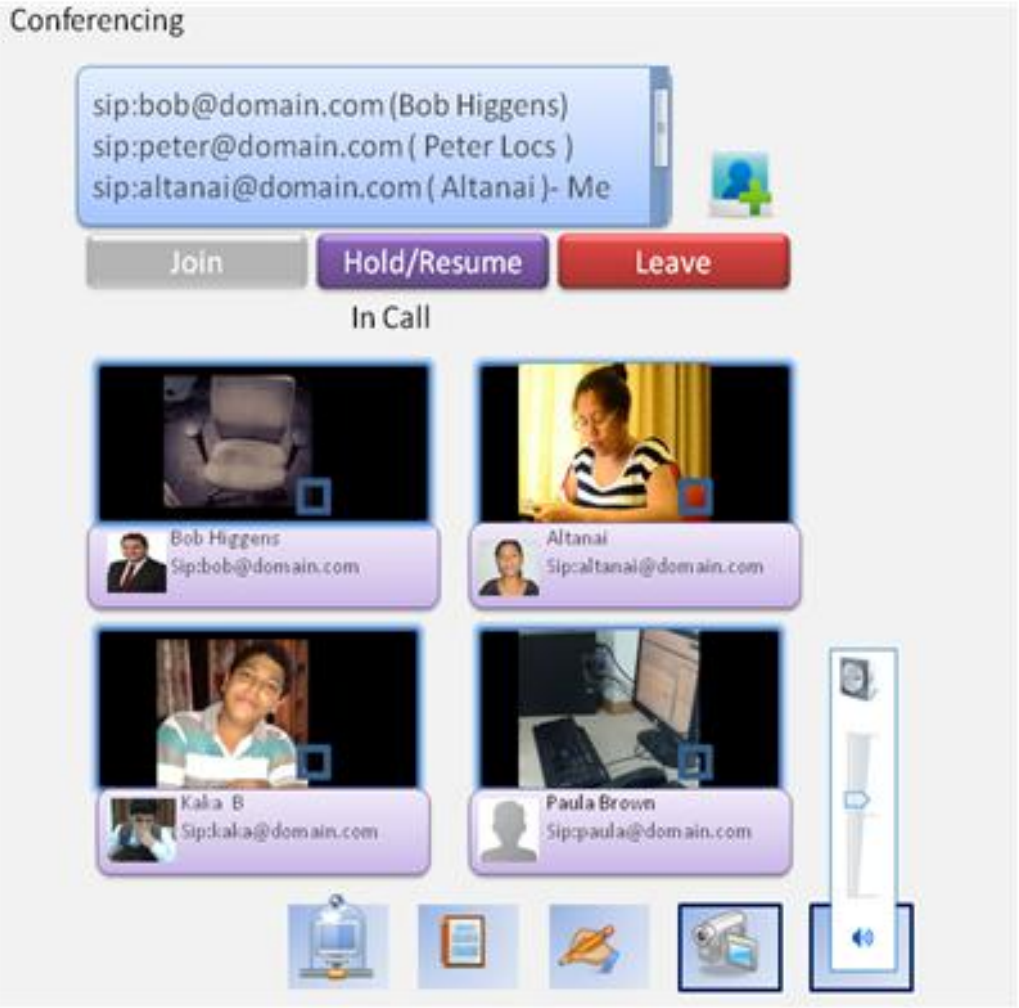
Call logs


Call Logs
Find
User
Date
Time
Duration
Incoming
Bob Higgens
12/06/2014
13:00
15 min
Outgoing
Bob Higgens
12/06/2014
12:45
4 min
Incoming
John Danken
12/06/2014
12:30
1 hour 5 min
Call Screened
Sara Tess
12/06/2014
9:00
Missed Call
Peter Locs
12/06/2014
8:38
Page 1 of 5

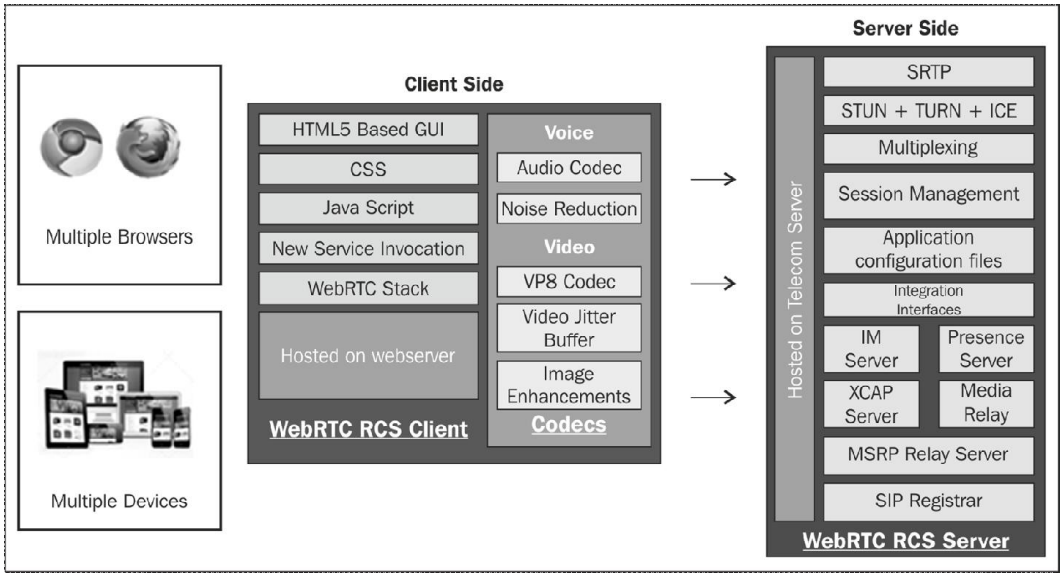
Message history



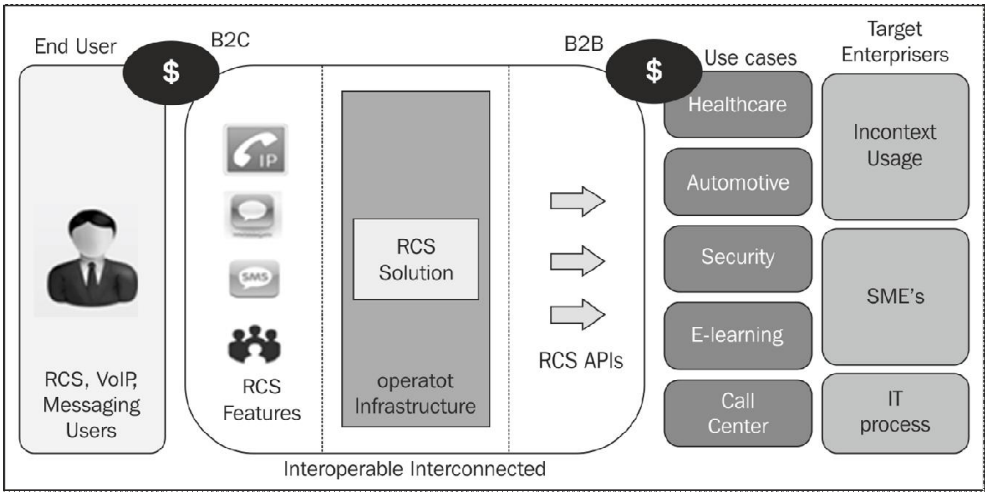
Multiparty conferencing



WebRTC architecture with RCS modules

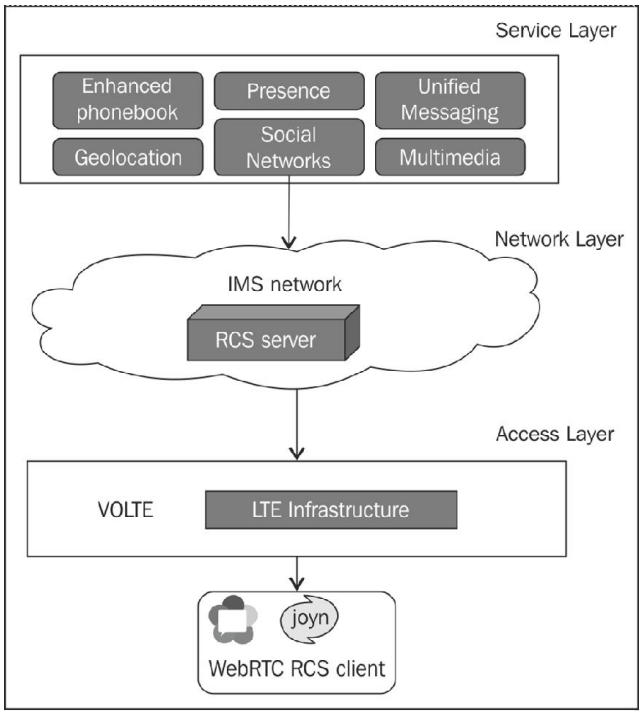


Telecom operator's benefit derived from RCS



Voice over LTE

Combination of WebRTC, VOLTE, and RCS

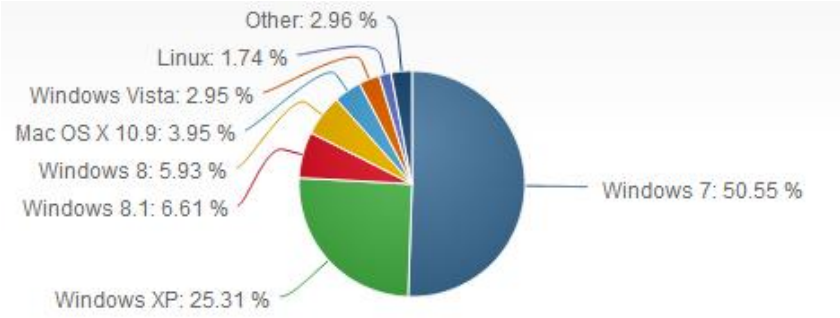


9

Native SIP Application and Interaction with a WebRTC Client

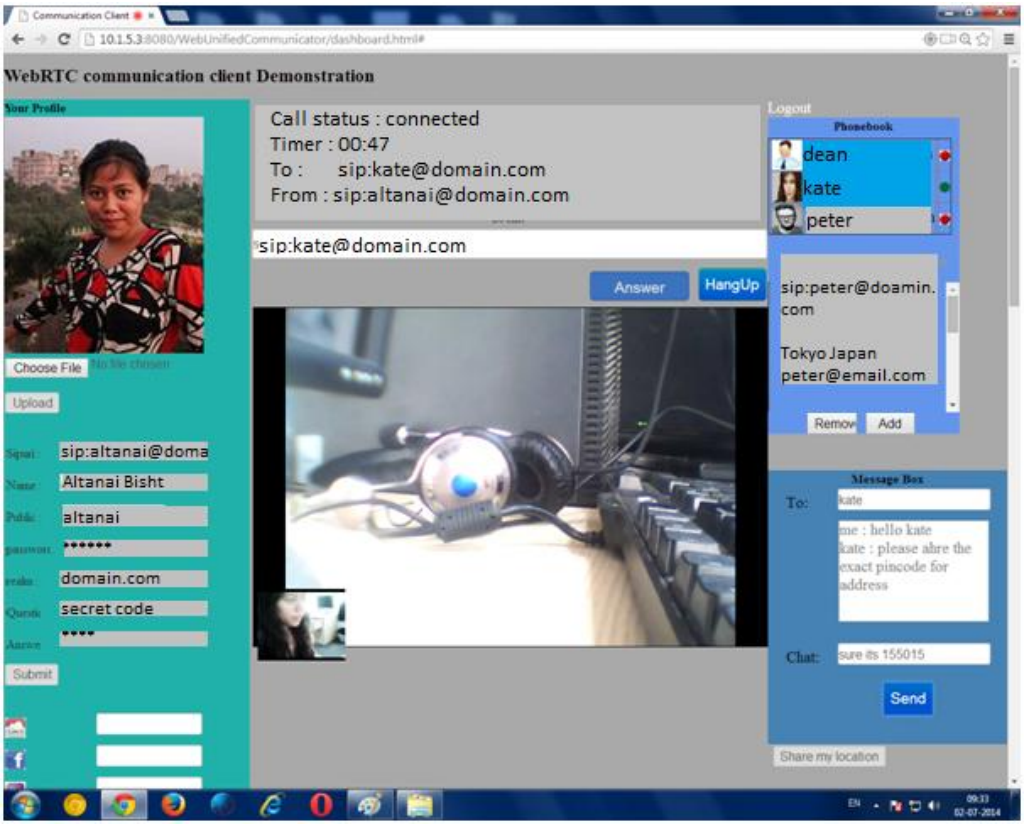
Support for WebRTC in various operating systems

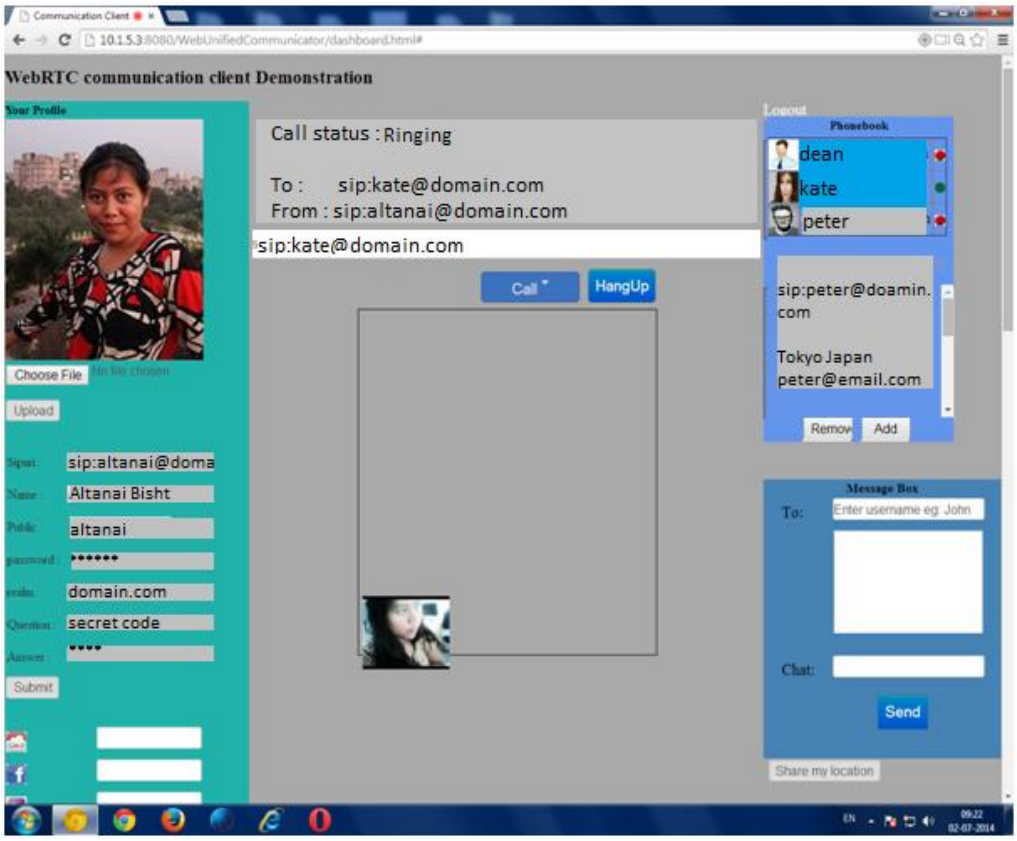
Windows OS



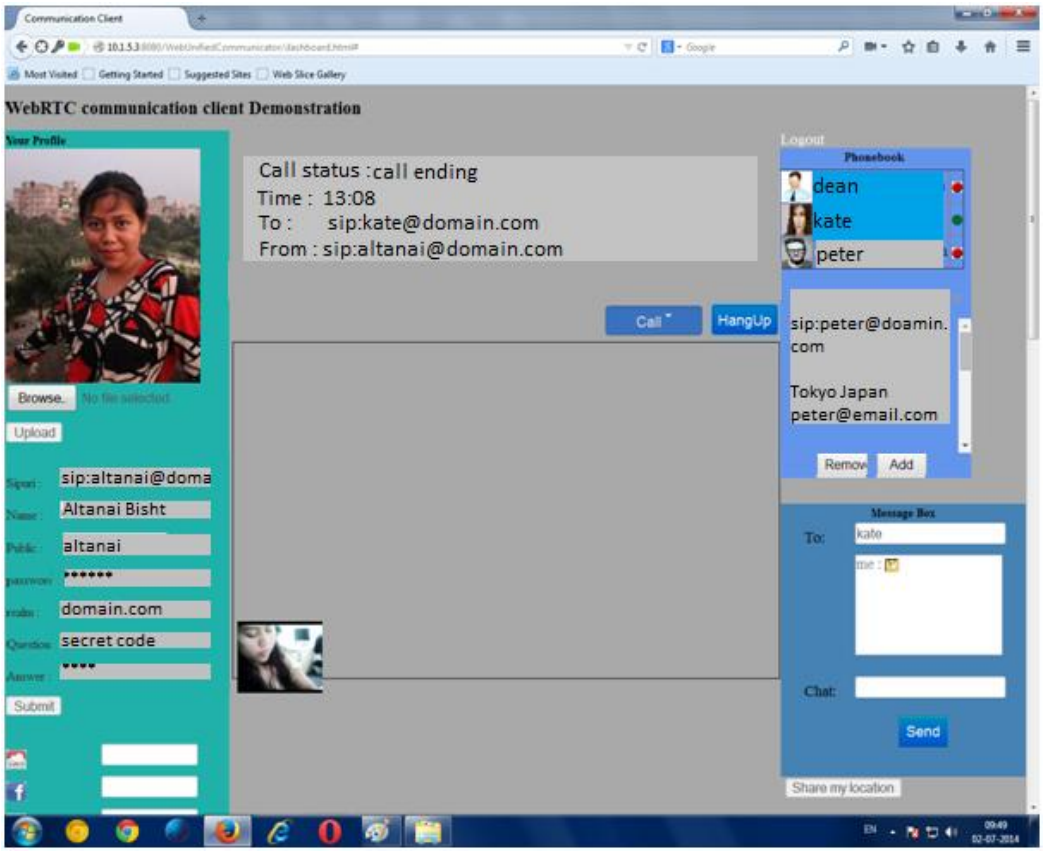
Native browser support for WebRTC client

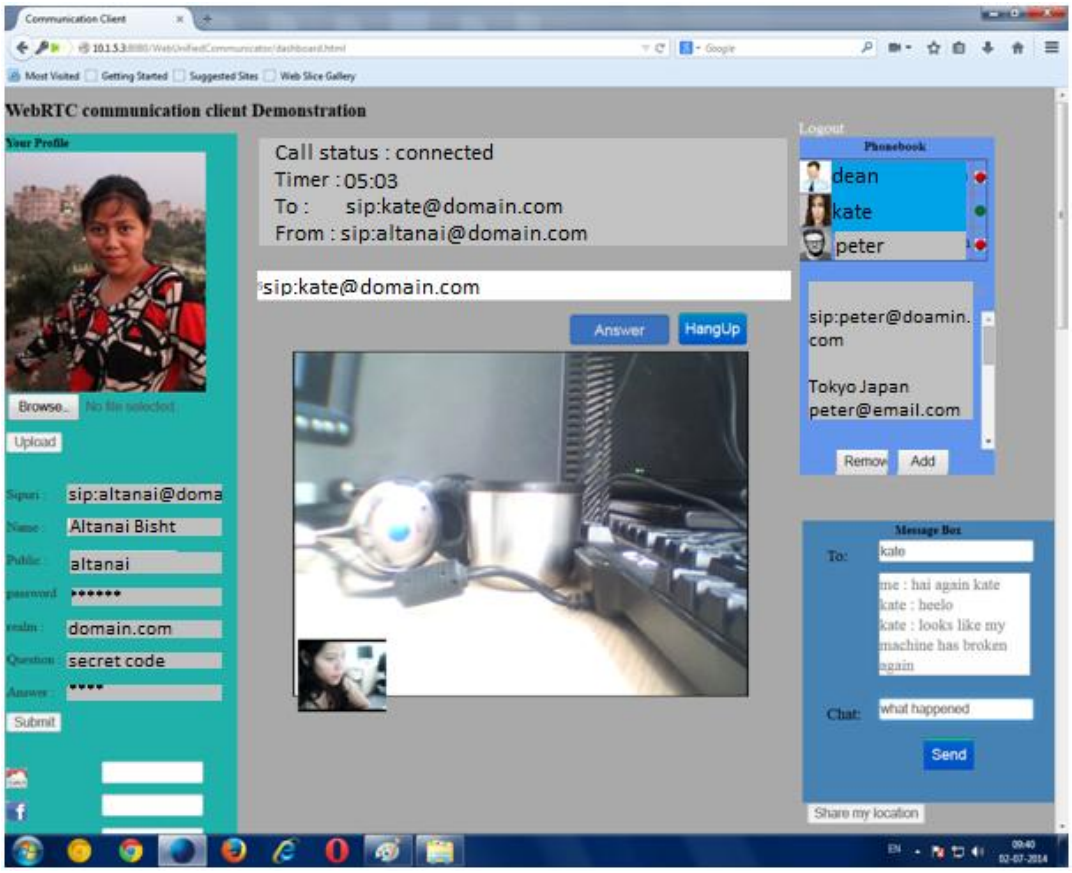
The Google Chrome browser support for WebRTC client


Communication Client
10.1.5.3:8080/WebUnifiedCommunicator/dashboard.html#
WebRTC communication client Demonstration
Your Profile
Choose File
Upload
sip:altanai@doma
Altanai Bisht
altanai
domain.com
secret code
Submit
Call status : connected
Timer : 00:47
To : sip:kate@domain.com
From : sip:altanai@domain.com
sip:kate@domain.com
Answer
HangUp
Logout
Phonebook
dean
kate
peter
sip:peter@doamin.com
Tokyo Japan
peter@email.com
Remov
Add
Message Box
To:
kate
me : hello kate
kate : please ahre the exact pincode for address
Chat:
sure its 155015
Send
Share my location

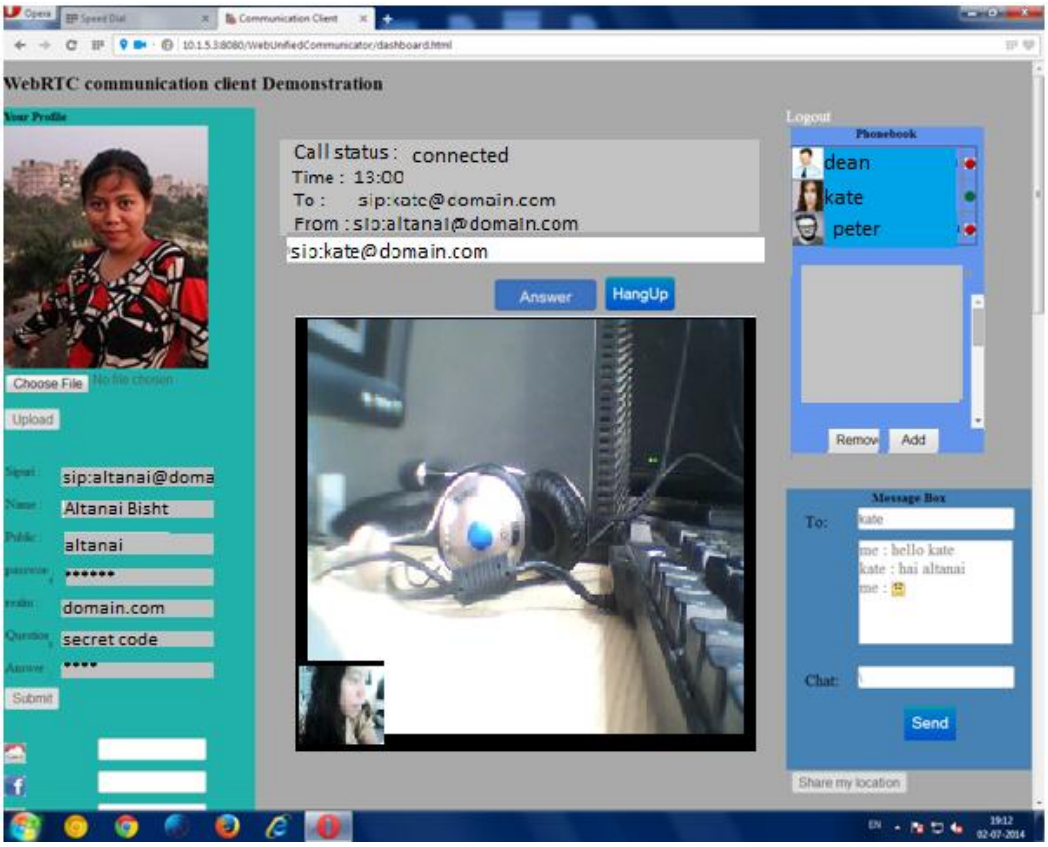


The Mozilla browser support for WebRTC client


Communication Client
WebRTC communication client Demonstration
Your Profile
Browse..
No file selected
Upload
Sipuri :
sip:altanai@doma
Name :
Altanai Bisht
Public :
altanai
password :
domain.com
Question :
secret code
Answer :
Submit
Call status :call ending
Time : 13:08
To : sip:kate@domain.com
From : sip:altanai@domain.com
Call
HangUp
Logout
Phonebook
dean
kate
peter
sip:peter@doamin.
com
Tokyo Japan
peter@email.com
Remov
Add
Message Box
To:
kate
Chat:
Send
Share my location



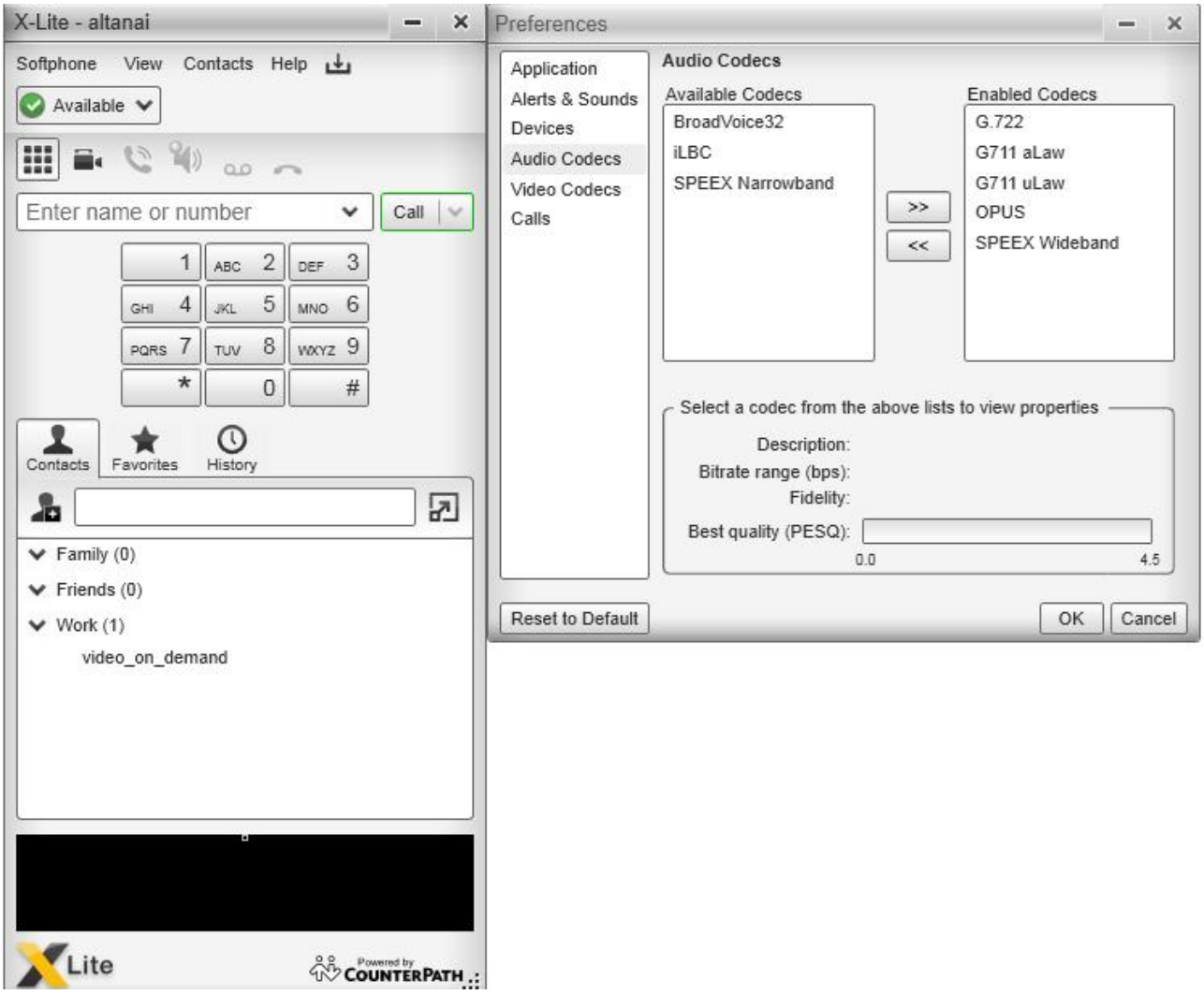
The Opera browser support for WebRTC client



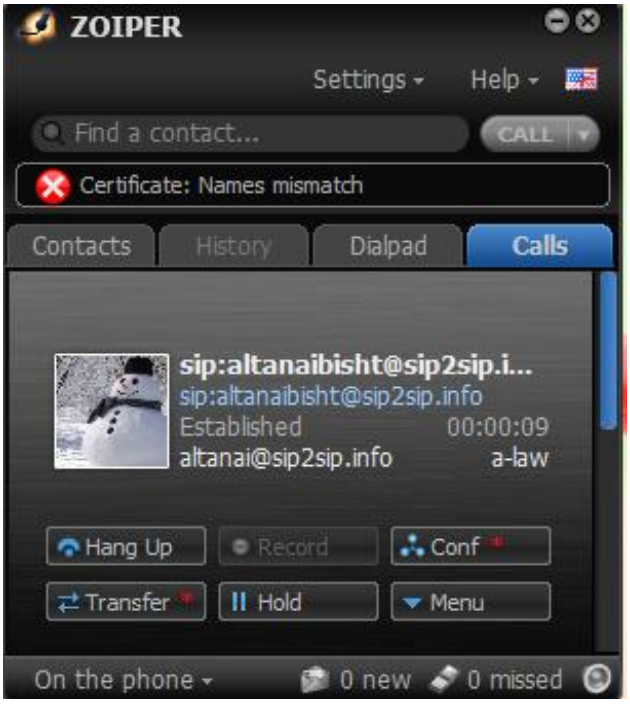
SIP softphones capable of interacting with WebRTC clients

X-Lite

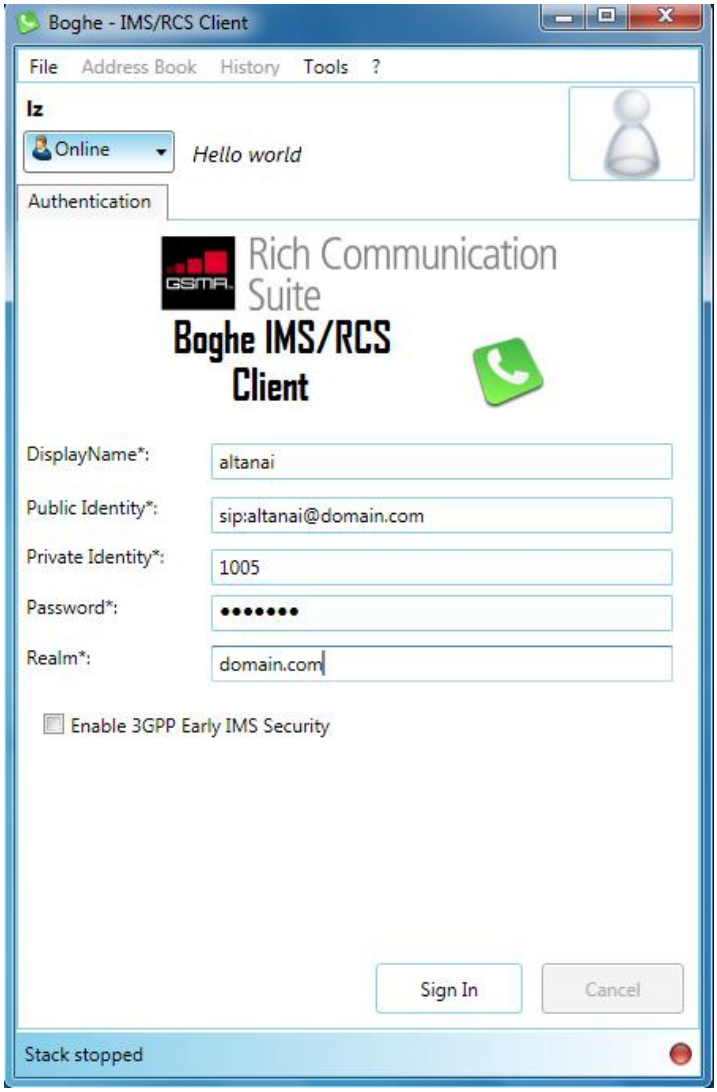
Video
Detach
Not Sending Video
Start
Enter phone number
sip:kate@domain.com
X-Lite
FLASH
MUTE
Powered by
COUNTERPATH
Calls & Contacts
Detach
On the Phone
Calls
Missed Calls
Received Calls
Dialed Calls
Blocked Incoming Calls
Contacts
Calls
Browser


X-Lite - altanai
Softphone View Contacts Help
Available
Enter name or number
Call
1 ABC 2 DEF 3
GHI 4 JKL 5 MNO 6
PQRS 7 TUV 8 WXYZ 9
* 0 #
Contacts Favorites History
Family (0)
Friends (0)
Work (1)
video_on_demand
Lite
Powered by COUNTERPATH
Preferences
Application
Alerts & Sounds
Devices
Audio Codecs
Video Codecs
Calls
Audio Codecs
Available Codecs
BroadVoice32
iLBC
SPEEX Narrowband
>>
<<
Enabled Codecs
G.722
G711 aLaw
G711 uLaw
OPUS
SPEEX Wideband
Select a codec from the above lists to view properties
Description:
Bitrate range (bps):
Fidelity:
Best quality (PESQ):
0.0
4.5
Reset to Default
OK
Cancel

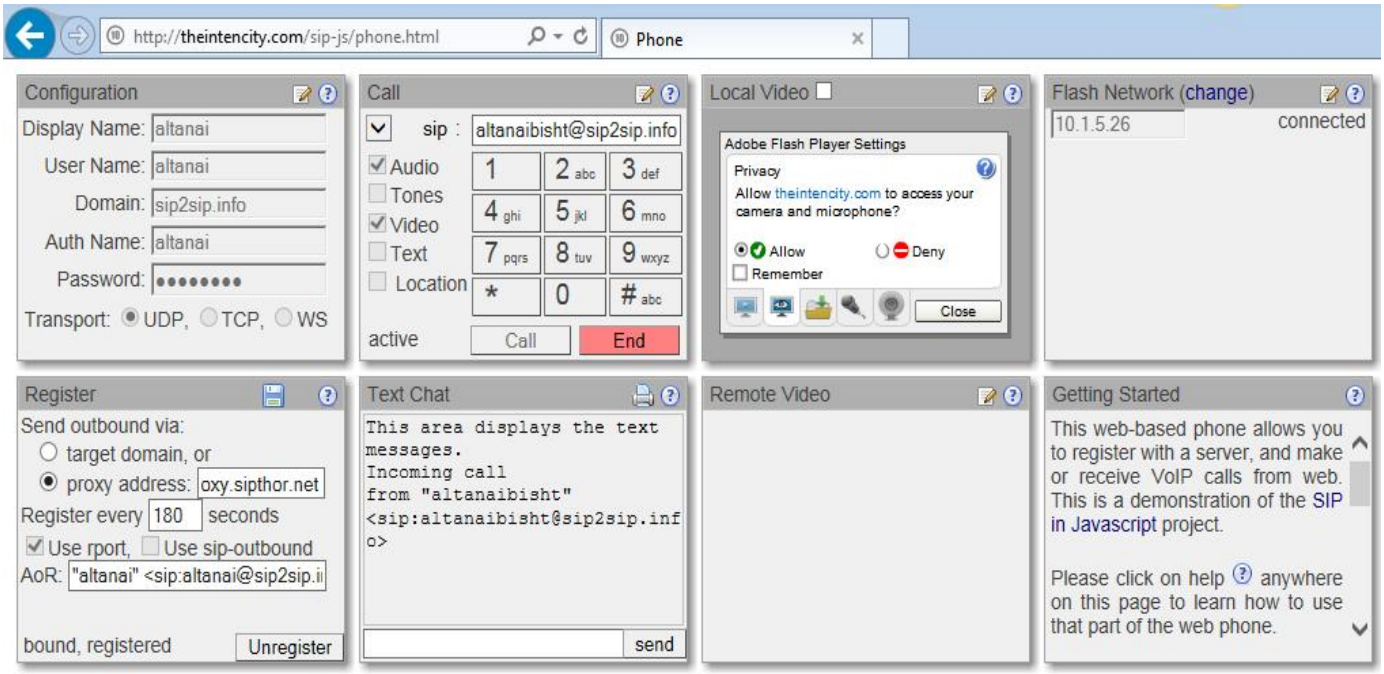
Zoiper


ZOIPER
Settings
Help
Find a contact...
CALL
Certificate: Names mismatch
Contacts
History
Dialpad
Calls
sip:altanaibisht@sip2sip.i...
sip:altanaibisht@sip2sip.info
Established
00:00:09
altanai@sip2sip.info
a-law
Hang Up
Record
Conf
Transfer
Hold
Menu
On the phone
0 new
0 missed

Boghe


Boghe - IMS/RCS Client
File
Address Book
History
Tools
?
lz
Online
Hello world
Authentication
Rich Communication Suite
Boghe IMS/RCS Client
DisplayName*:
altanai
Public Identity*:
sip:altanai@domain.com
Private Identity*:
1005
Password*:
Realm*:
domain.com
Enable 3GPP Early IMS Security
Sign In
Cancel
Stack stopped

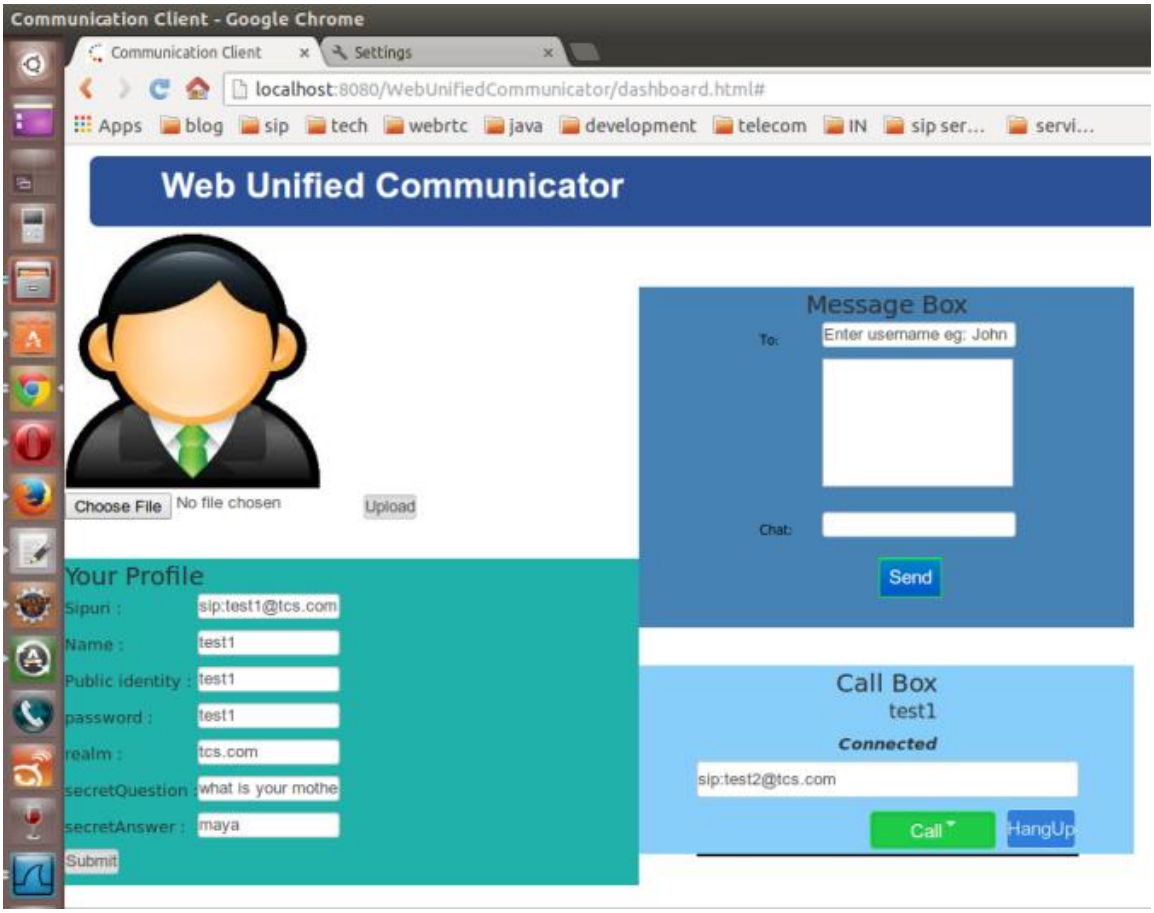
WebRTC unsupported browser interacting with WebRTC client

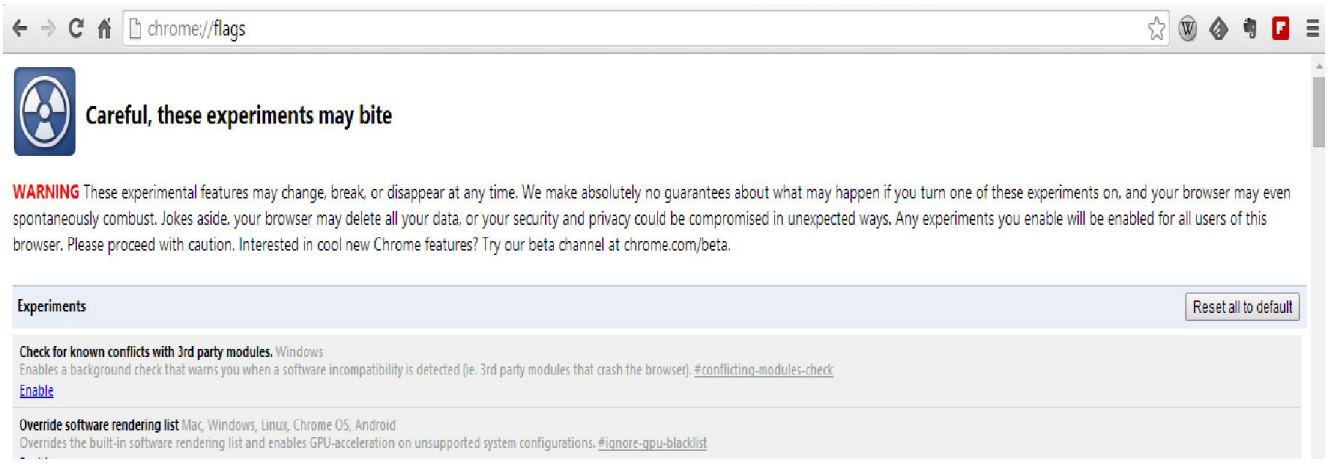


Linux OS

Native browser support for WebRTC client

The Google Chrome browser support for WebRTC client


Communication Client - Google Chrome
Communication Client
Settings
localhost:8080/WebUnifiedCommunicator/dashboard.html#
Apps
blog
sip
tech
webrtc
java
development
telecom
IN
sip ser...
servi...
Web Unified Communicator
Choose File
No file chosen
Upload
Your Profile
Sipuri :
sip:test1@tcs.com
Name :
test1
Public identity :
test1
password :
test1
realm :
tcs.com
secretQuestion :
what is your mothe
secretAnswer :
maya
Submit
Message Box
To:
Enter username eg: John
Chat:
Send
Call Box
test1
Connected
sip:test2@tcs.com
Call
HangUp


chrome://flags
Careful, these experiments may bite
WARNING These experimental features may change, break, or disappear at any time. We make absolutely no guarantees about what may happen if you turn one of these experiments on, and your browser may even spontaneously combust. Jokes aside, your browser may delete all your data, or your security and privacy could be compromised in unexpected ways. Any experiments you enable will be enabled for all users of this browser. Please proceed with caution. Interested in cool new Chrome features? Try our beta channel at chrome.com/beta.
Experiments
Reset all to default
Check for known conflicts with 3rd party modules. Windows
Enables a background check that warns you when a software incompatibility is detected (ie. 3rd party modules that crash the browser). #conflicting-modules-check
Enable
Override software rendering list Mac, Windows, Linux, Chrome OS, Android
Overrides the built-in software rendering list and enables GPU-acceleration on unsupported system configurations. #ignore-gpu-blacklist

The Mozilla browser support for WebRTC client

SIP softphones capable of interacting with WebRTC clients

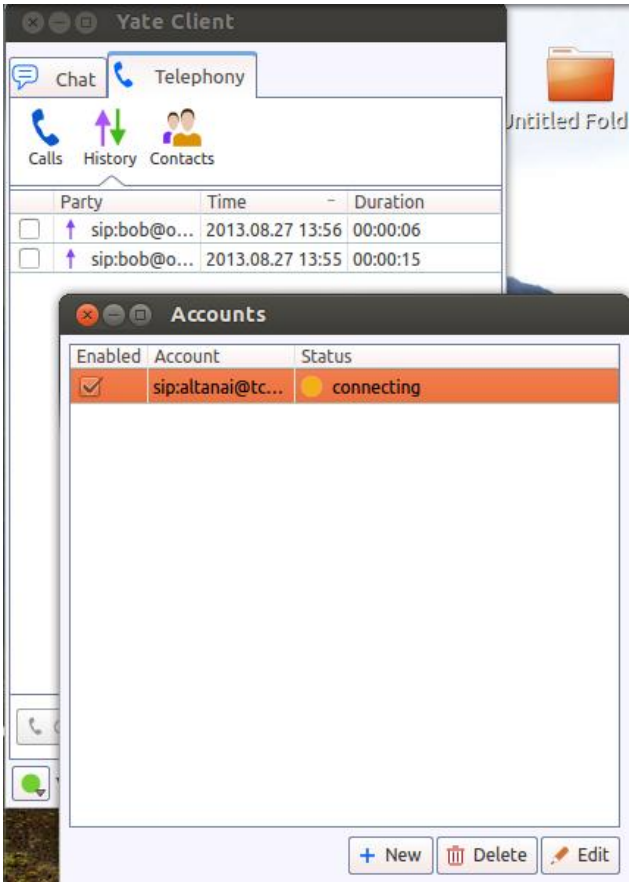
Kapanga

Kapanga
The Media over IP Softphone
[MyConfig1398342995.cfg] (ip to ip)
Calling sip:altanai@domain.com
sip:altanai@domain.com
LINE 1
LINE 2
LINE 3
1
2 ABC
3 DEF
4 GHI
5 JKL
6 MNO
7 PQRS
8 TUV
9 WXYZ
*
0 OPER
#
CONF
HOLD
XFER
REC
FAX
CLEAR
DIR

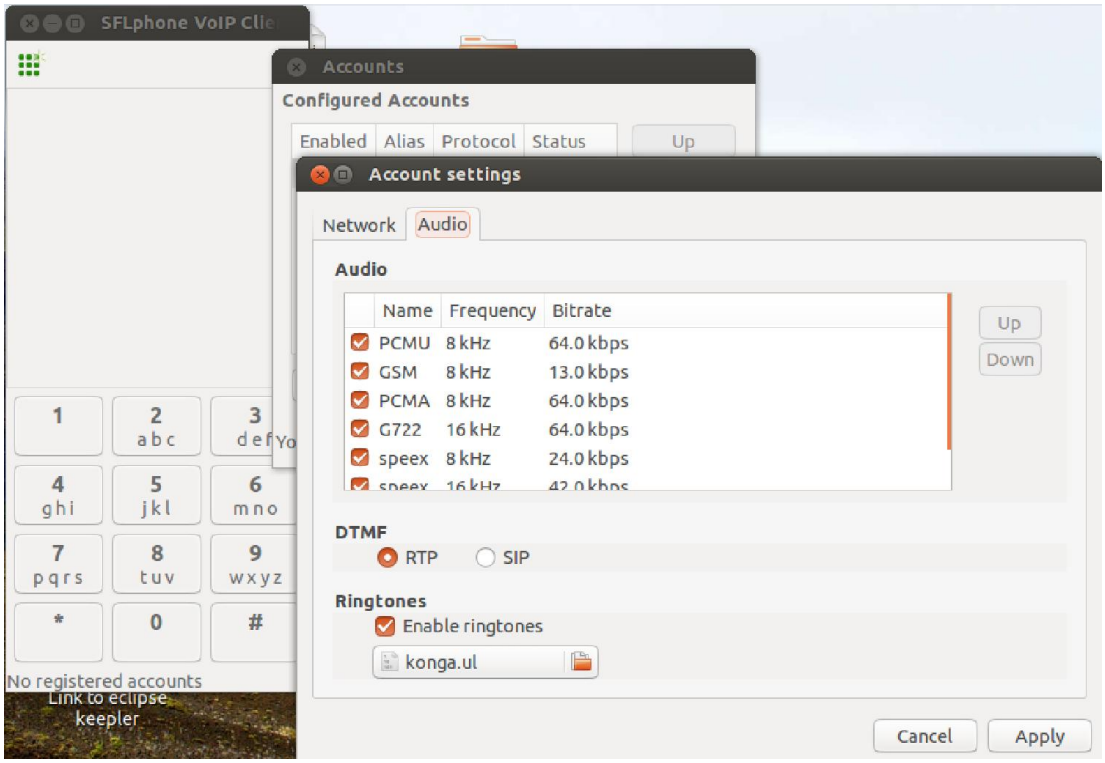
Linphone

Linphone
SIP address or phone number:
sip:peter@domain.com
Contacts
Recent calls
Keypad
Call #1
Calling...
00::00::00
sip:peter@domain.com
My current identity:
sip:altanai@domain.com
Contacting

Yate

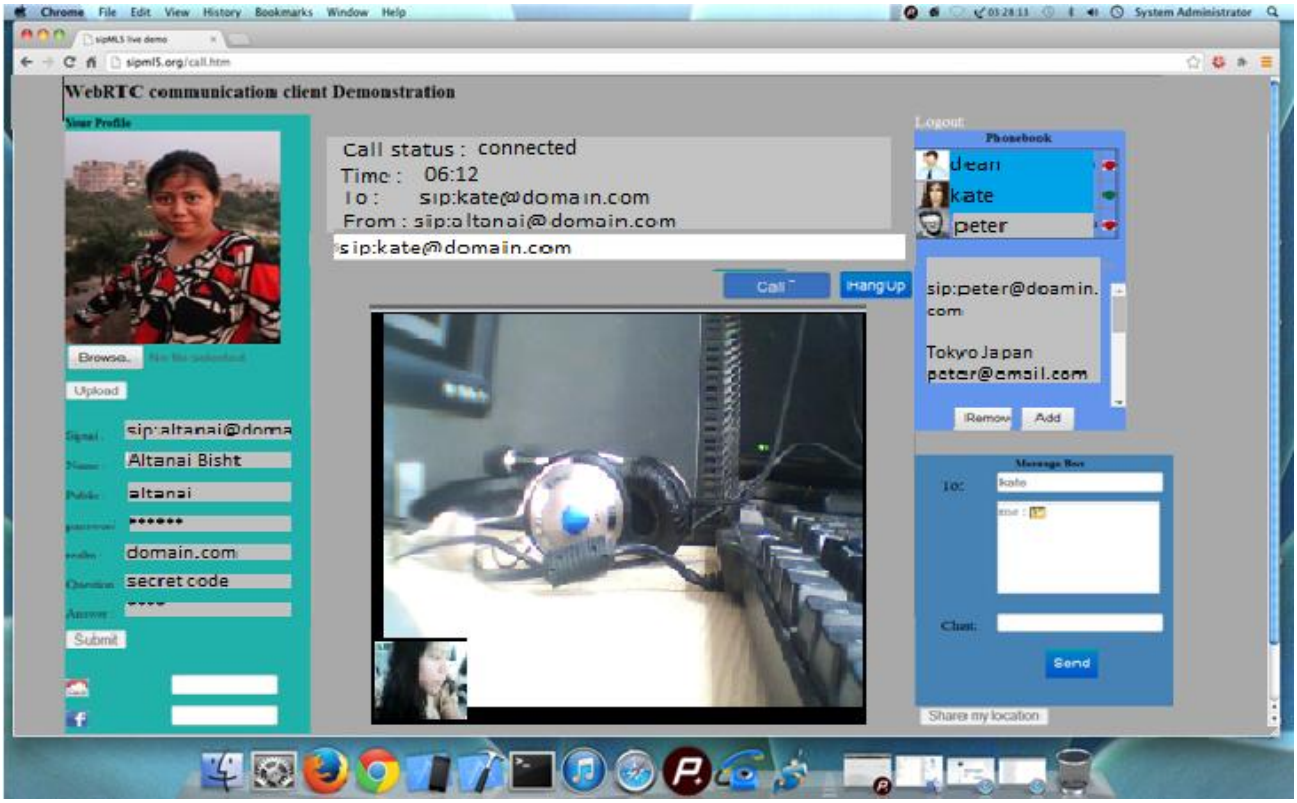

Yate Client
Chat
Telephony
Calls
History
Contacts
Party
Time
Duration
sip:bob@o...
2013.08.27 13:56
00:00:06
sip:bob@o...
2013.08.27 13:55
00:00:15
Accounts
Enabled
Account
Status
sip:altanai@tc...
connecting
New
Delete
Edit

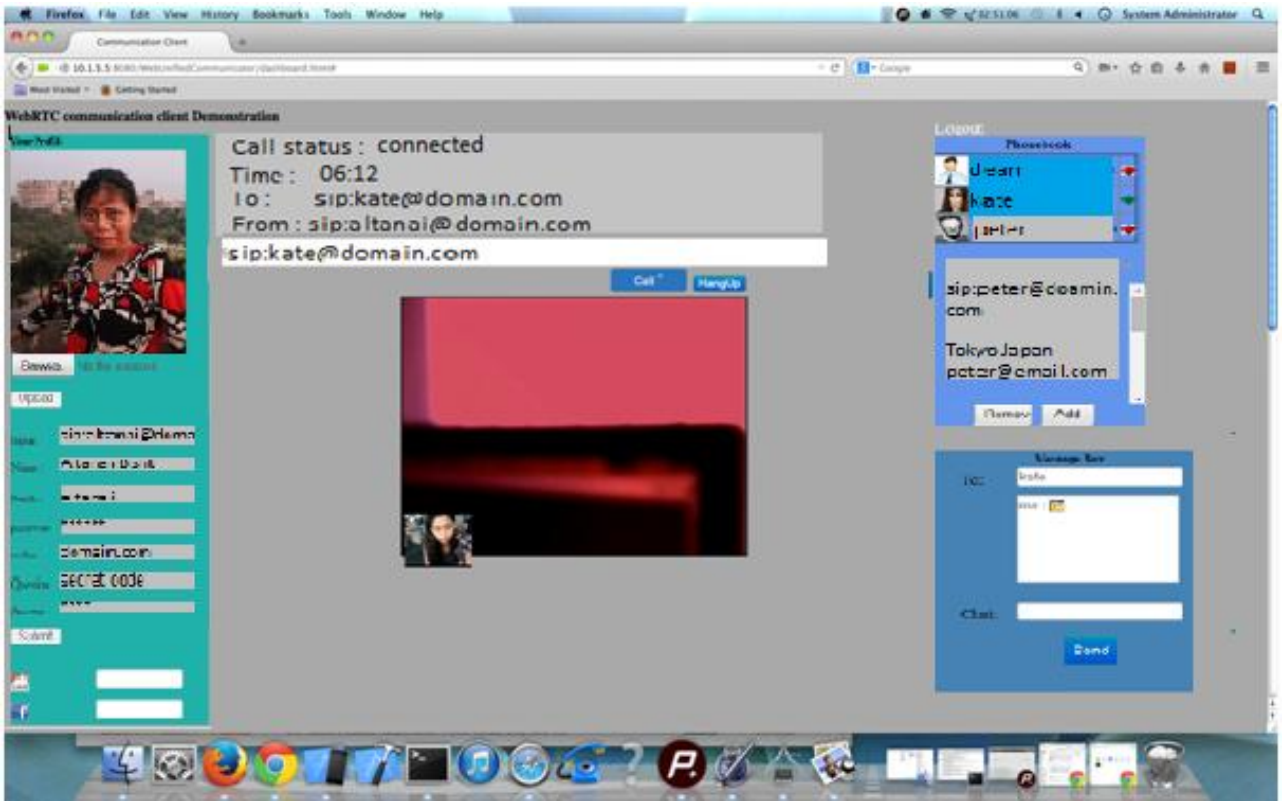
SFL


SFLphone VoIP Clie
Accounts
Configured Accounts
Enabled
Alias
Protocol
Status
Up
Account settings
Network
Audio
Audio
Name
Frequency
Bitrate
PCMU 8 kHz 64.0 kbps
GSM 8 kHz 13.0 kbps
PCMA 8 kHz 64.0 kbps
G722 16 kHz 64.0 kbps
speex 8 kHz 24.0 kbps
Up
Down
DTMF
RTP
SIP
Ringtones
Enable ringtones
konga.ul
1
2 abc
3 def
4 ghi
5 jkl
6 mno
7 pqrs
8 tuv
9 wxyz
*
0
#
No registered accounts
Link to eclipse keepler
Cancel
Apply

Mac OS

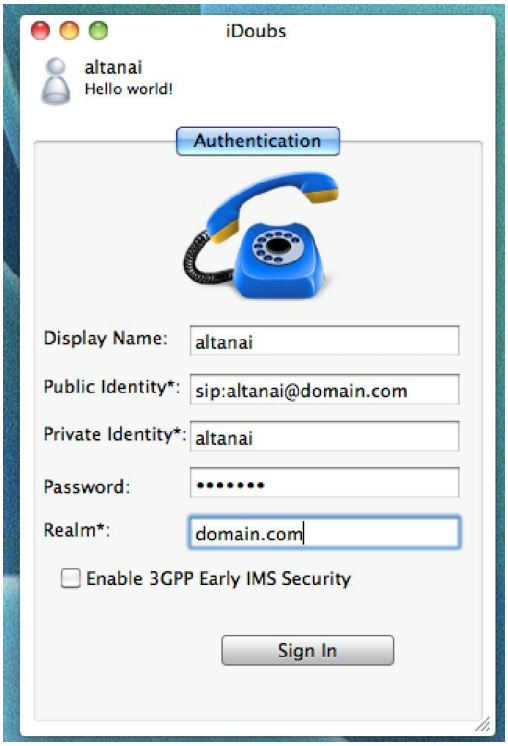
Native browser support for WebRTC clients





SIP softphones capable of interacting with WebRTC clients

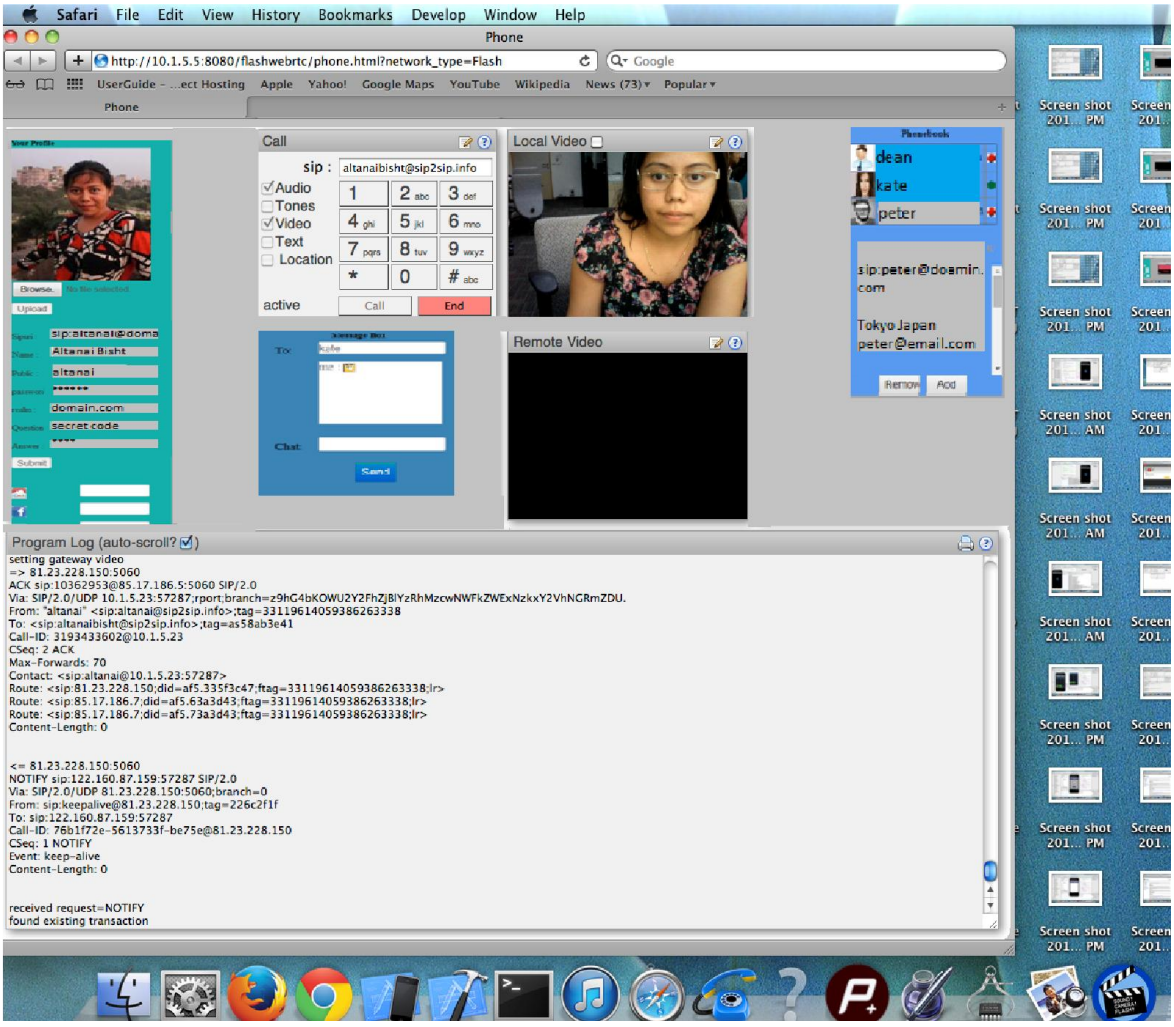
iDoubs


iDoubs
altanai
Hello world!
Authentication
Display Name:
altanai
Public Identity*:
sip:altanai@domain.com
Private Identity*:
altanai
Password:
Realm*:
domain.com
Enable 3GPP Early IMS Security
Sign In

Jitsi

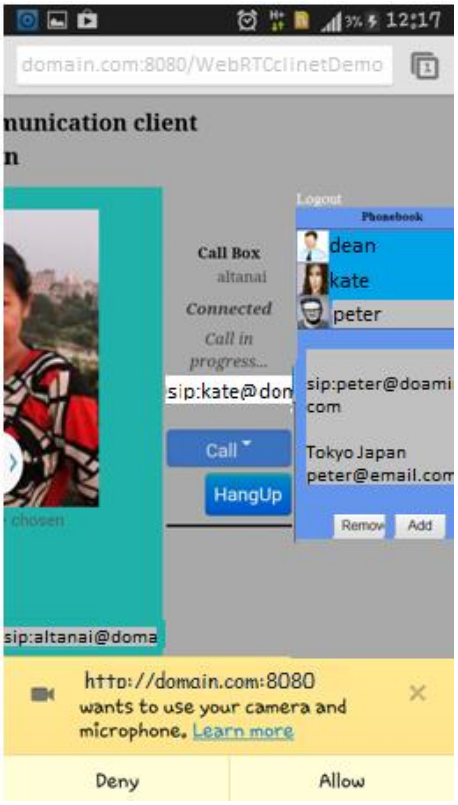
Jitsi
altanai
Online
sip:alice@domain.com
No matching contacts found. Press Ctrl+Enter to call "sip:alice@domain.com" or use the buttons below.
Add contact
Call contact
00:00:00 | Call
alice (alice@domain.com)
Initiating call
Jitsi
Applications

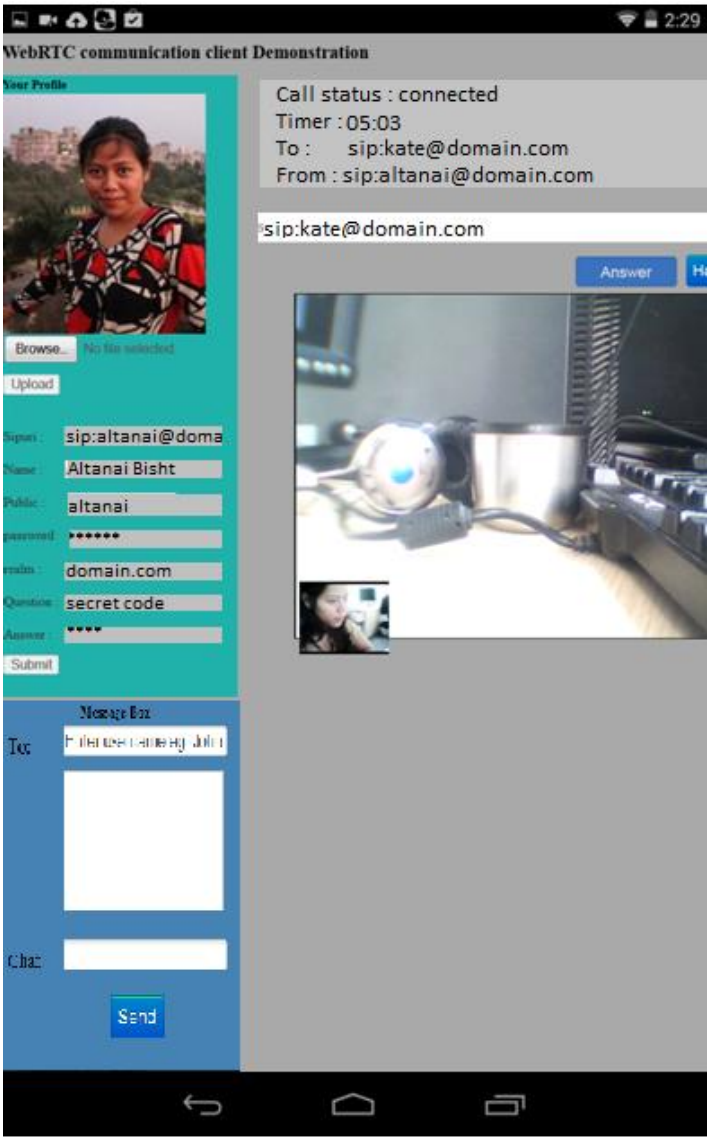
WebRTC unsupported browser interacting with WebRTC client

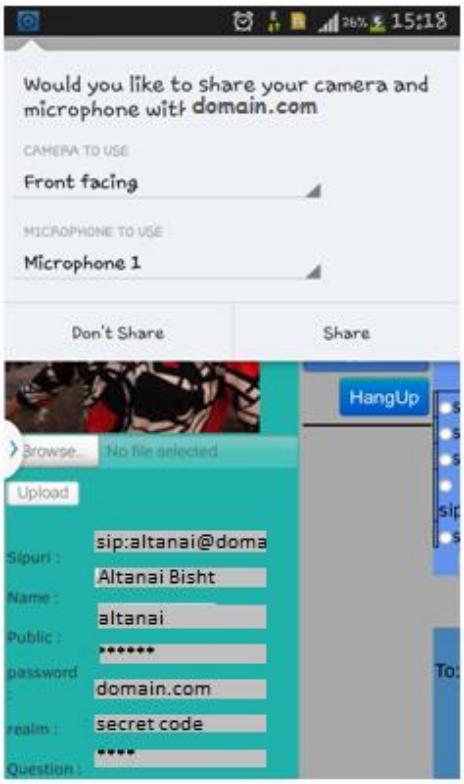


Android OS for mobiles

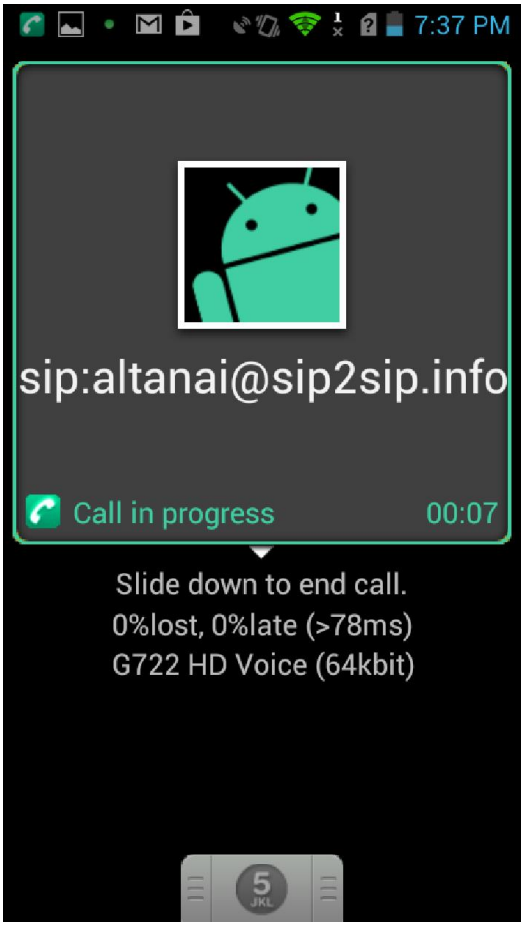
Native browser support for WebRTC client

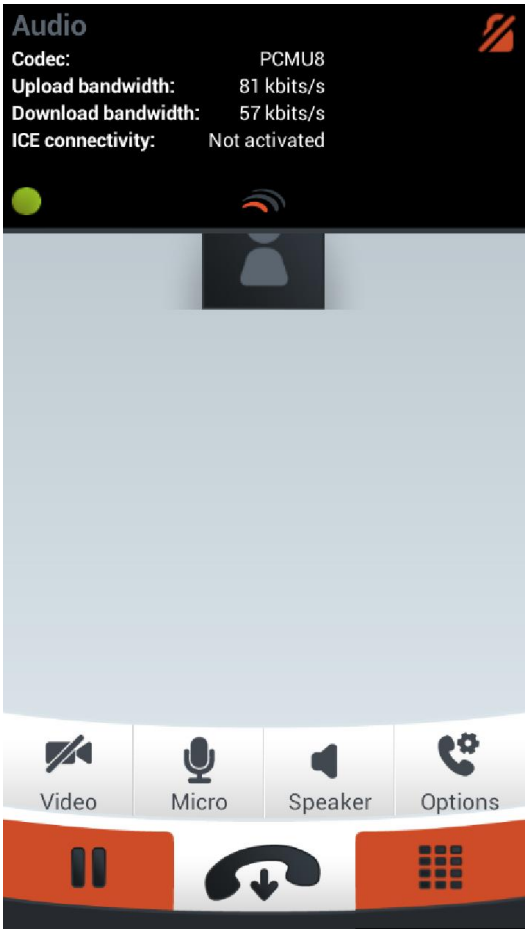



2:29
WebRTC communication client Demonstration
Your Profile
Browse...
No file selected
Upload
sip:altanai@doma
Altanai Bisht
altanai
domain.com
secret code
Submit
Message Box
To:
Chat:
Send
Call status : connected
Timer : 05:03
To : sip:kate@domain.com
From : sip:altanai@domain.com
sip:kate@domain.com
Answer


15:18
Would you like to share your camera and microphone with domain.com
CAMERA TO USE
Front facing
MICROPHONE TO USE
Microphone 1
Don't Share
Share
HangUp
Browse...
No file selected
Upload
Sipuri :
sip:altanai@doma
Name :
Altanai Bisht
Public :
altanai
password :
domain.com
realm :
secret code
Question :

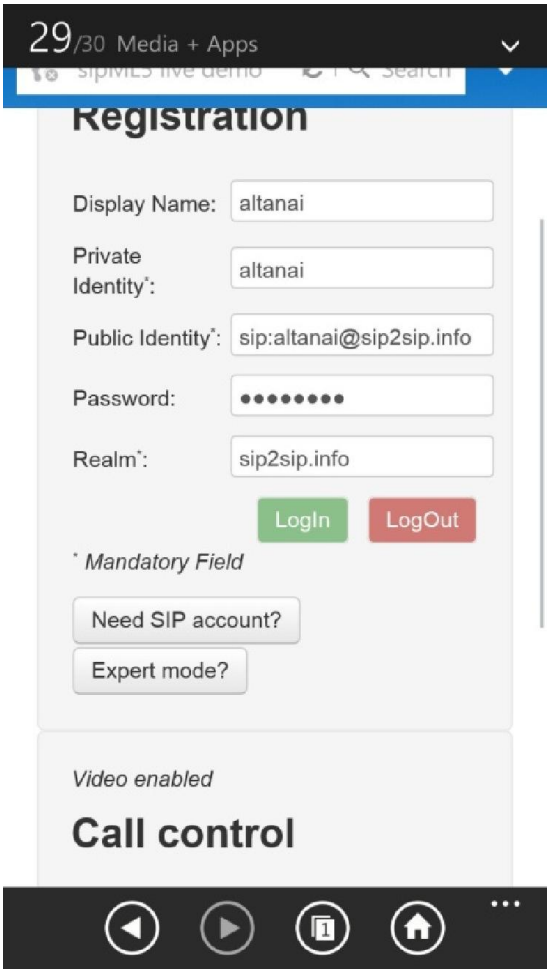
Android phone/tablet SIP applications capable of interacting with WebRTC clients




Audio
Codec: PCMU8
Upload bandwidth: 81 kbits/s
Download bandwidth: 57 kbits/s
ICE connectivity: Not activated
Video
Micro
Speaker
Options

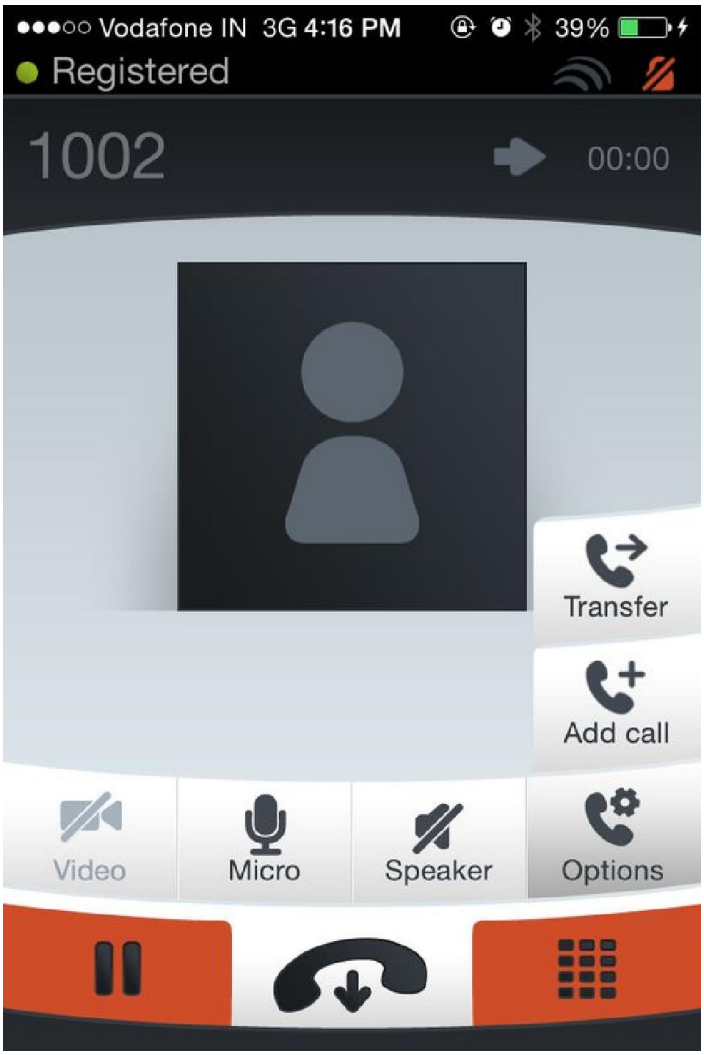
Developing a lightweight Android SIP application

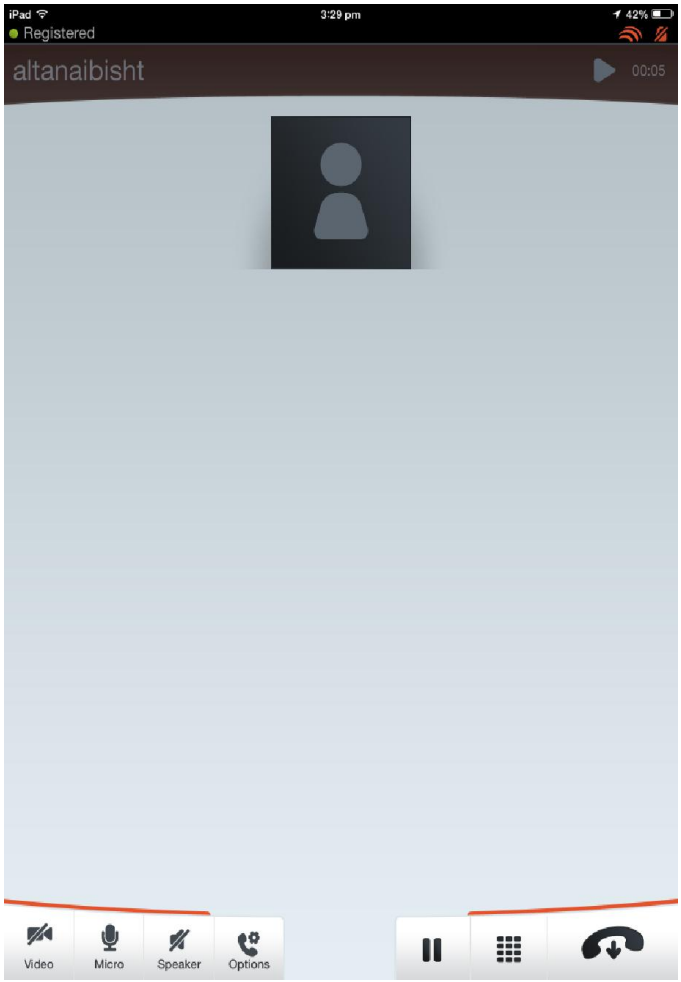
Windows OS for Mobiles



Apple iPhone

iPhone /iPad IP applications interacting with WebRTC Client




iPad
3:29 pm
42%
Registered
altanaibisht
00:05
Video
Micro
Speaker
Options

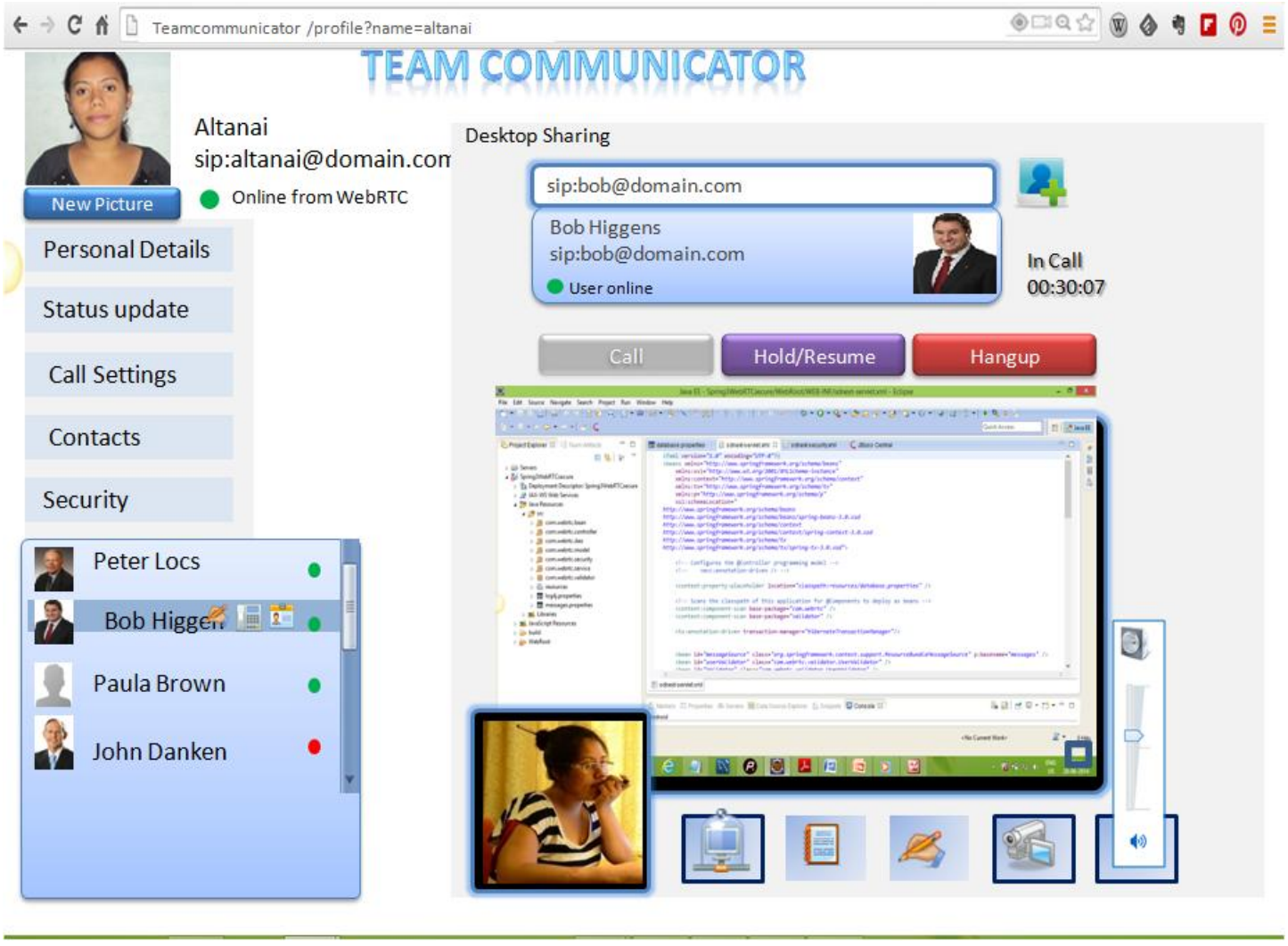
10

Other WebRTC Use Cases

Unified communicator

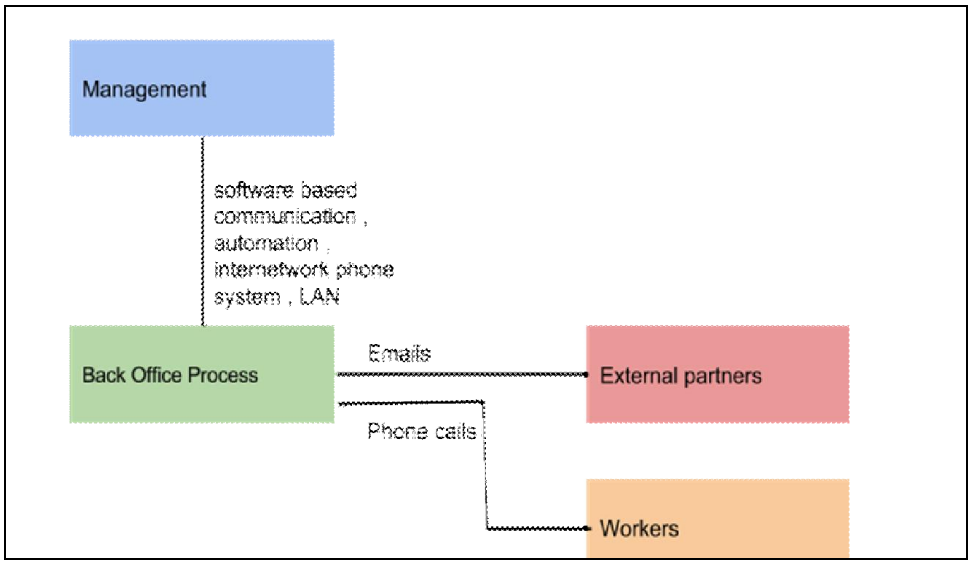
Team communicator

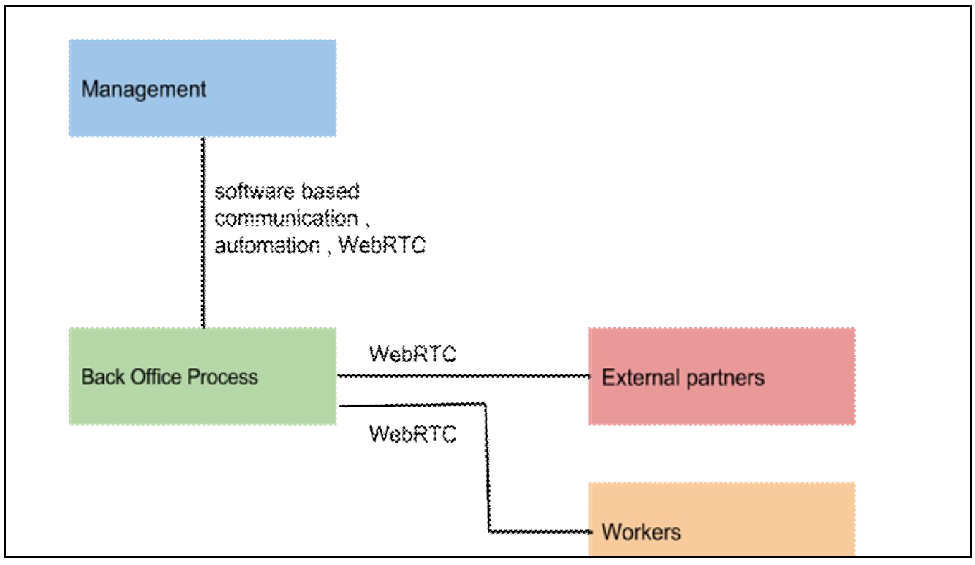
Teamcommunicator /profile?name=altanai
TEAM COMMUNICATOR
New Picture
Altanai
sip:altanai@domain.com
Online from WebRTC
Personal Details
Status update
Call Settings
Contacts
Security
Peter Locs
Bob Higgen
Paula Brown
John Danken
Conferencing
sip:bob@domain.com (Bob Higgens)
sip:peter@domain.com (Peter Locs)
sip:altanai@domain.com (Altanai)- Me
In Call
00:36:02
Join
Hold/Resume
Leave
In Call
Bob Higgens
Sip:bob@domain.com
Altanai
Sip:altanai@domain.com
Kaka B
Sip:kaka@domain.com
Paula Brown
sip:paula@domain.com


Teamcommunicator /profile?name=altanai
TEAM COMMUNICATOR
Altanai
sip:altanai@domain.com
New Picture
Online from WebRTC
Personal Details
Status update
Call Settings
Contacts
Security
Peter Locs
Bob Higgen
Paula Brown
John Danken
Desktop Sharing
sip:bob@domain.com
Bob Higgens
sip:bob@domain.com
User online
In Call
00:30:07
Call
Hold/Resume
Hangup

Customized communicator for specific Enterprise segments

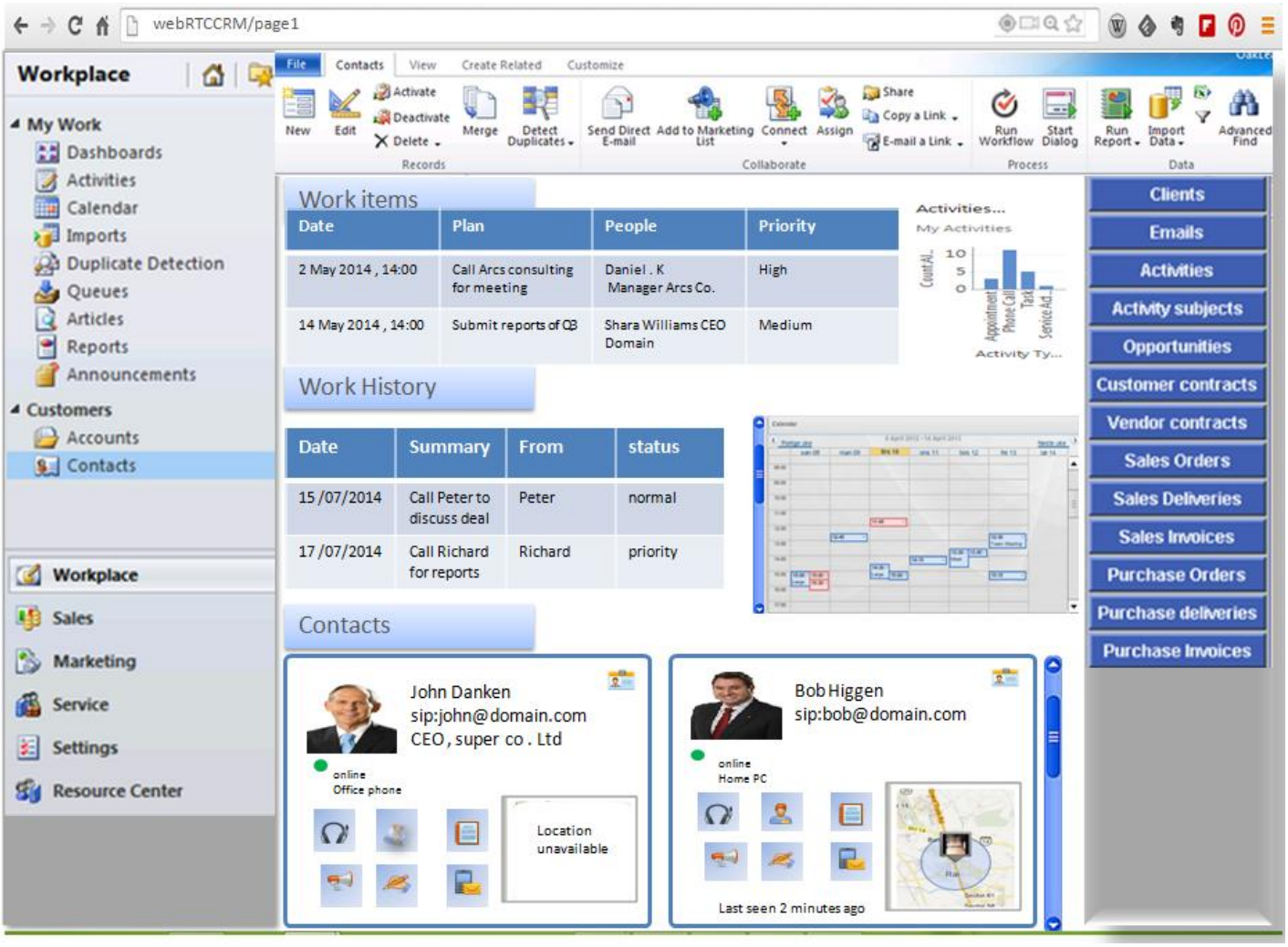
Branches and back-office communications

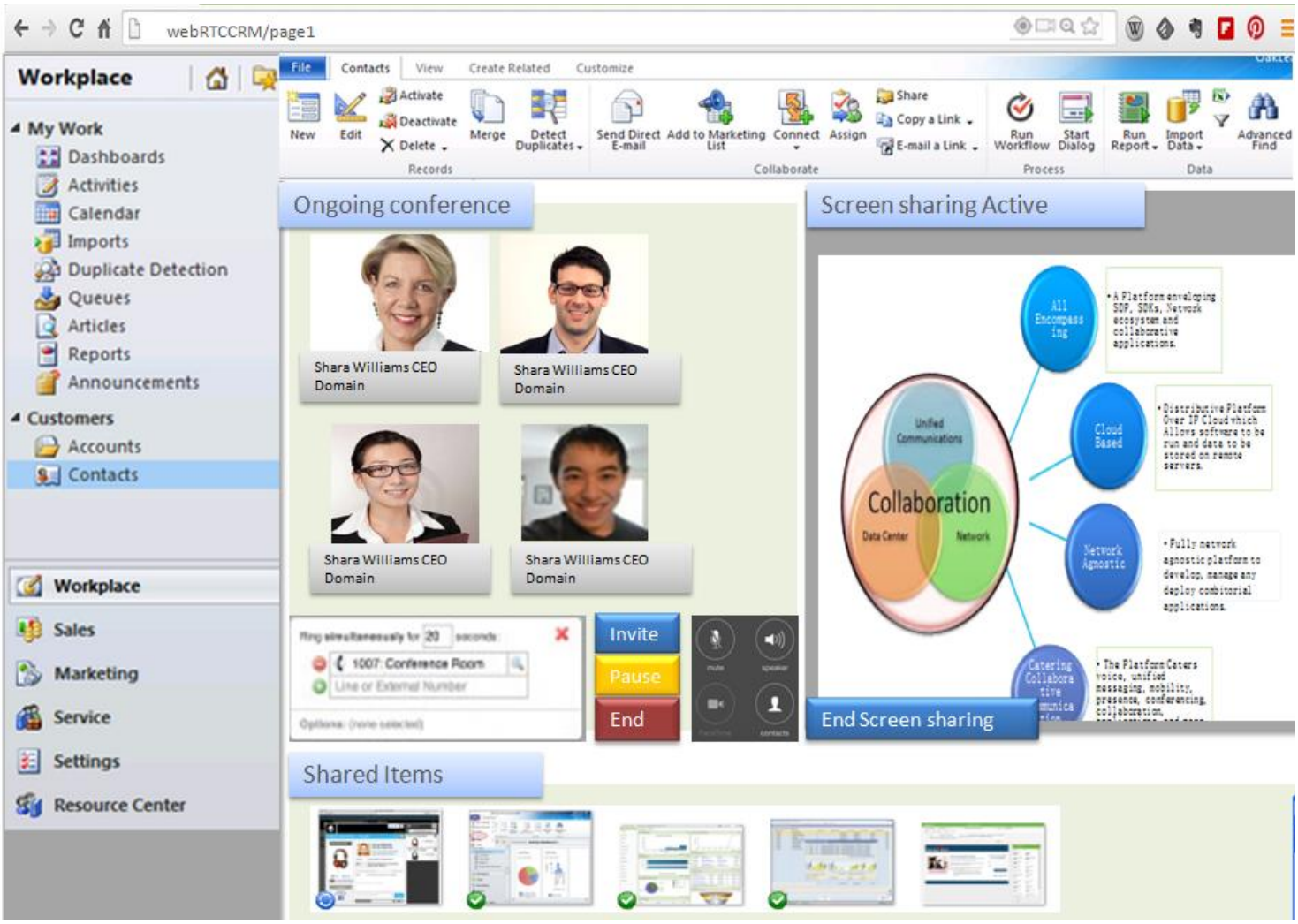


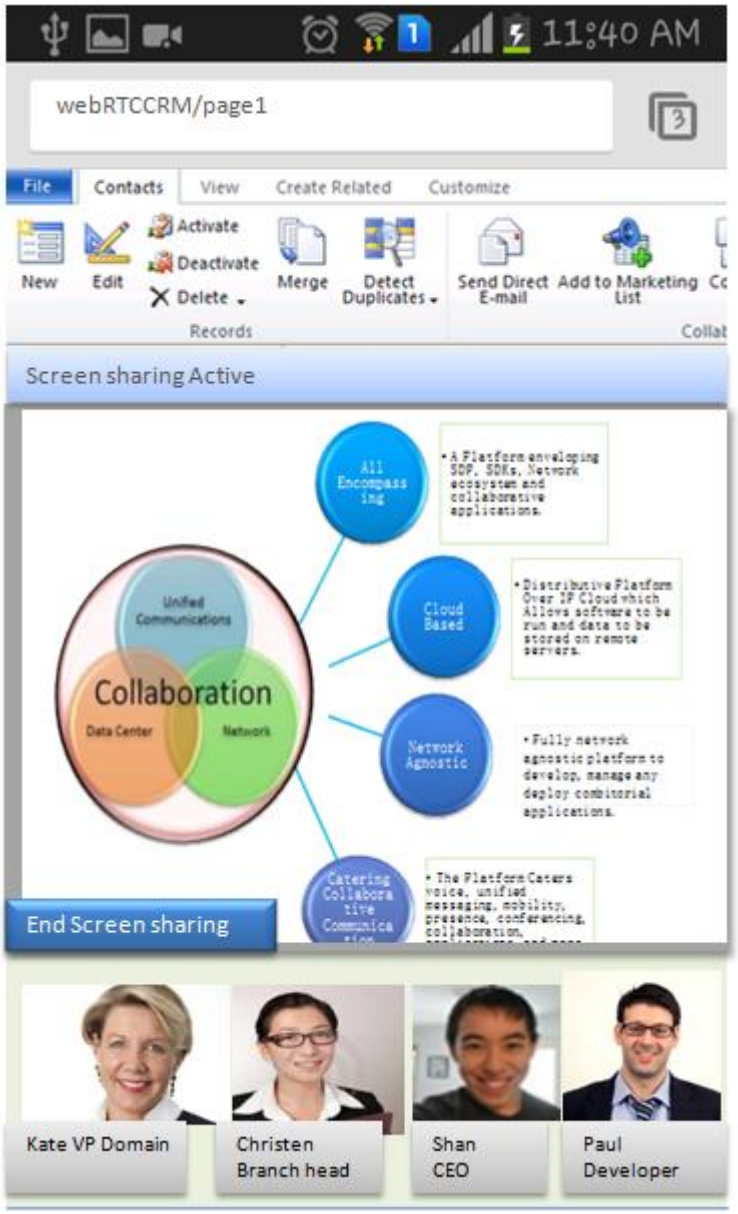


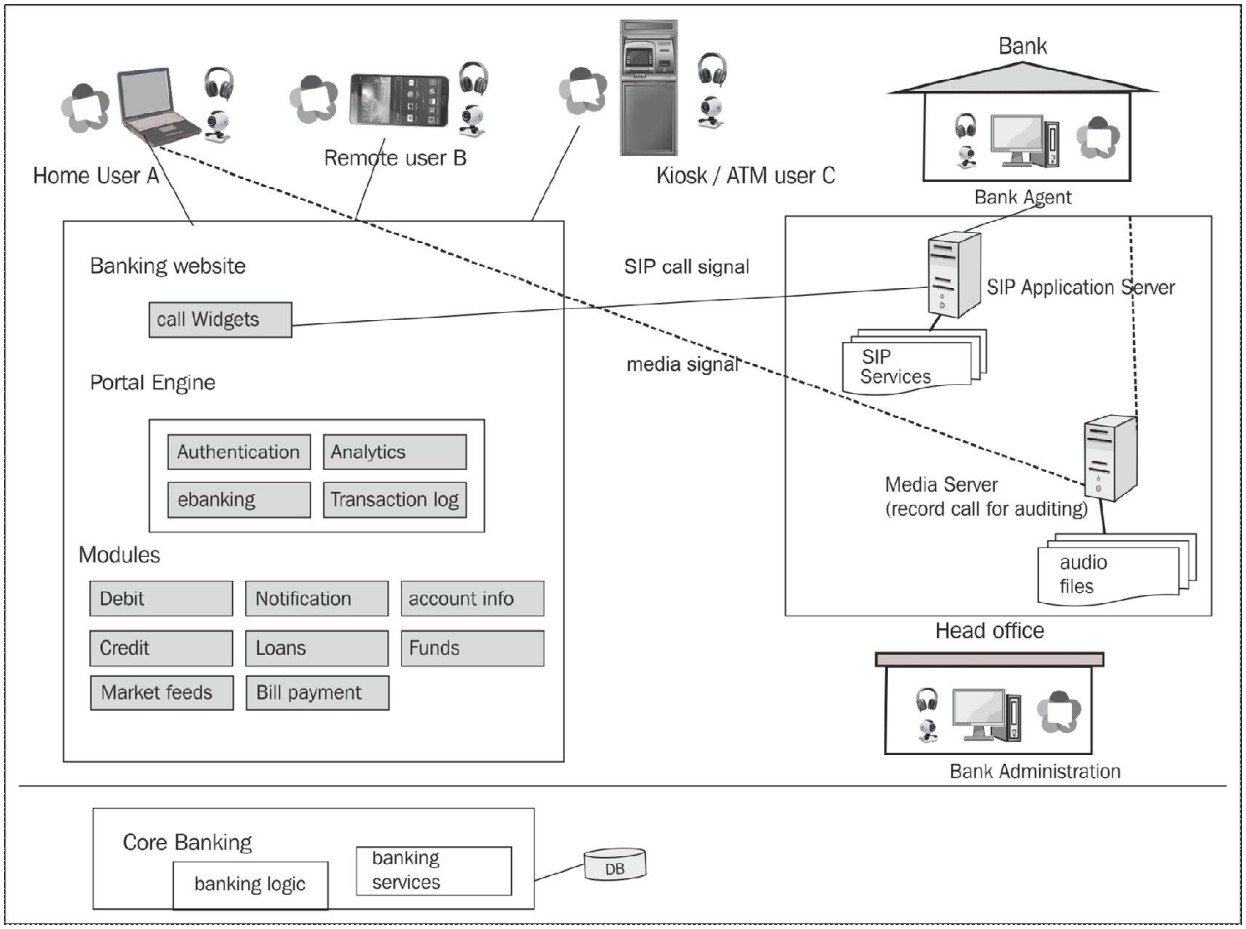
Inventorytrack/inventoryID?78654ef9&name=altanai
Altanai Bisht
City , Country
sip;altanai@domain.com
Phone : 02300170
Supervisor :
Richard Gomez (Branch Head)
Role :
Search
clients
Find
Locate dealers
Locate workers
Track inventory
Contacts
---Co workers----------
alisha
Sip :alisha@domain.com
NY , USA
User offline
Leonard
Sip :leo@domain.com
L A , USA
User offline
---Dealers----------
John
Sip :john@domain.com
France
User online
Jack
Sip :jack@domain.com
Shanghai , China
User online
---area 4 ----------
Robert
Sip :jrob@domain.com
Egypt
User online
Alex
close X
File : datasheet 1
Send
Got it
John
close X
Hope it is good time to talk. This is regarding the consignment delivered yesterday .
Send
Robert
close X
Hello dear responder , I wanted to check the progress of plan 5678
ofcourse
Send
sending you the

The Customer Relationship Management system

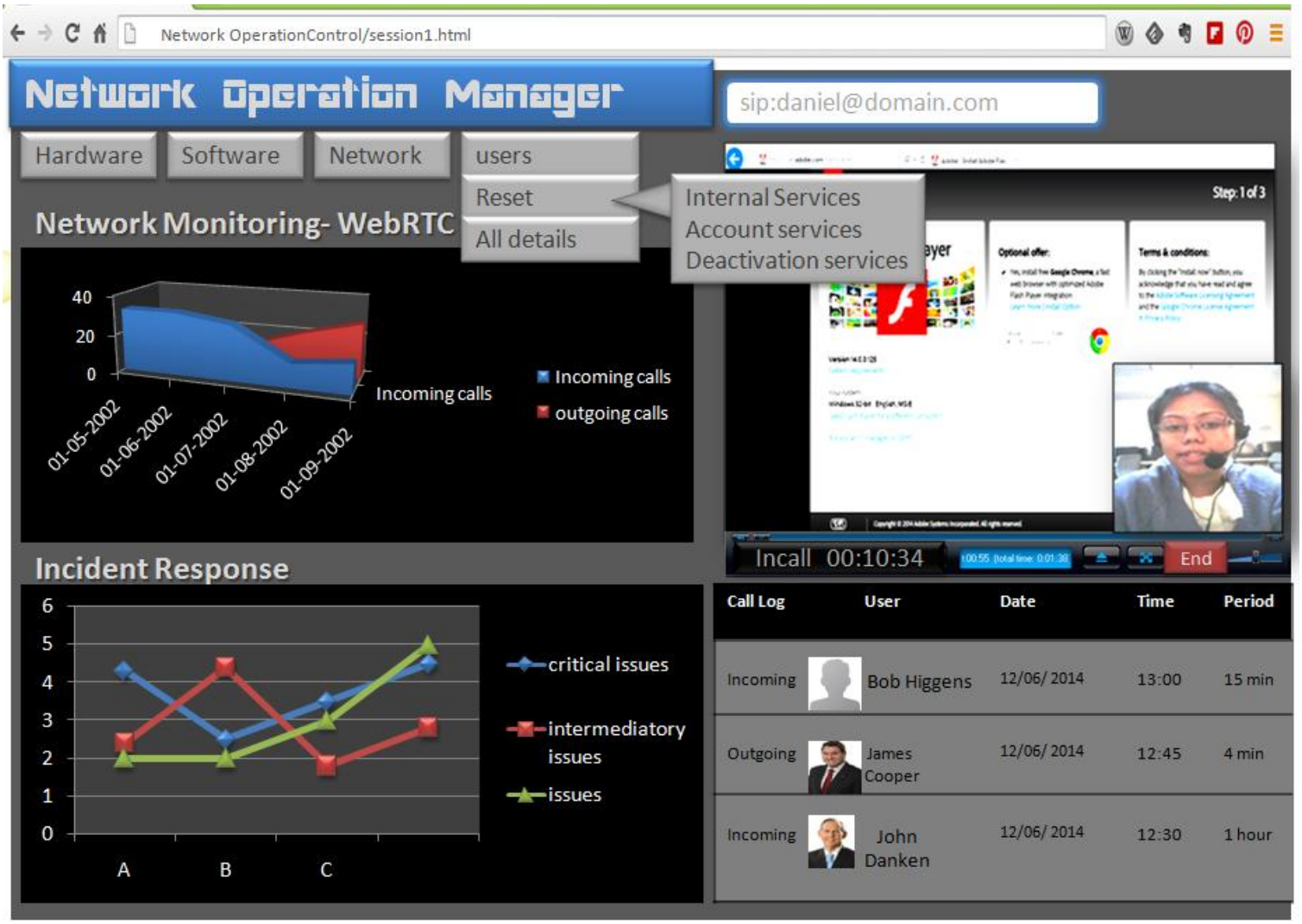



webRTCCRM/page1
Workplace
My Work
Dashboards
Activities
Calendar
Imports
Duplicate Detection
Queues
Articles
Reports
Announcements
Customers
Accounts
Contacts
Workplace
Sales
Marketing
Service
Settings
Resource Center
File
Contacts
View
Create Related
Customize
New
Edit
Activate
Deactivate
Delete
Merge
Detect Duplicates
Records
Send Direct E-mail
Add to Marketing List
Connect
Assign
Share
Copy a Link
E-mail a Link
Collaborate
Run Workflow
Start Dialog
Process
Run Report
Import Data
Advanced Find
Data
Ongoing conference
Shara Williams CEO Domain
Shara Williams CEO Domain
Shara Williams CEO Domain
Shara Williams CEO Domain
Invite
Pause
End
Screen sharing Active
Unified Communications
Collaboration
Data Center
Network
All Encompassing
Cloud Based
Network Agnostic
End Screen sharing
Shared Items


11:40 AM
webRTCCRM/page1
File
Contacts
View
Create Related
Customize
New
Edit
Activate
Deactivate
Delete
Merge
Detect Duplicates
Send Direct E-mail
Add to Marketing List
Records
Screen sharing Active
All Encompass ing
• A Platform enveloping SDP, SDKs, Network ecosystem and collaborative applications.
Cloud Based
• Distributive Platform Over IP Cloud which Allows software to be run and data to be stored on remote servers.
Unified Communications
Collaboration
Data Center
Network
Network Agnostic
• Fully network agnostic platform to develop, manage any deploy combinorial applications.
Catering Collabora tive Communica
• The Platform Caters voice, unified messaging, mobility, presence, conferencing, collaboration,
End Screen sharing
Kate VP Domain
Christen Branch head
Shan CEO
Paul Developer


Bank
Home User A
Remote user B
Kiosk / ATM user C
Bank Agent
Banking website
call Widgets
SIP call signal
SIP Application Server
SIP
Services
media signal
Portal Engine
Authentication
Analytics
ebanking
Transaction log
Media Server
(record call for auditing)
audio
files
Modules
Debit
Notification
account info
Credit
Loans
Funds
Market feeds
Bill payment
Head office
Bank Administration
Core Banking
banking logic
banking
services
DB

Network Operation Centre



FriedBuildingWebRTC/altanaipage.jps
picture
Update
customise
Altanai Bisht
City , Country
sip;altanai@domain.com
Phone : 02300170
Hobbies :
Swimming , Trekking
Gallery
Connected with facebook
Connected with Google +
Connected with Yahoo
Share
Who is in for a cup of coffee today ?
post
image
video
Shared an video
Shared an image :
Shared an event :
Hello I am attending the comic con this Sunday
Private chat with Alex
close X
979878979 is the id
Send
Got it
Private chat with John
close X
Good morning
I wonder if its okay to call my mom in this event
Send
Ofcourse
Friends from all accounts
—general
alisha
Sip :alisha@domain.com
NY , USA
User online
Alex
Sip :alex@domain.com
L.A , USA
User online
—School friends
Robert
Sip :jrob@domain.com
France
User offline
John
Sip :john@domain.com
Shanghai , China
User online
—family

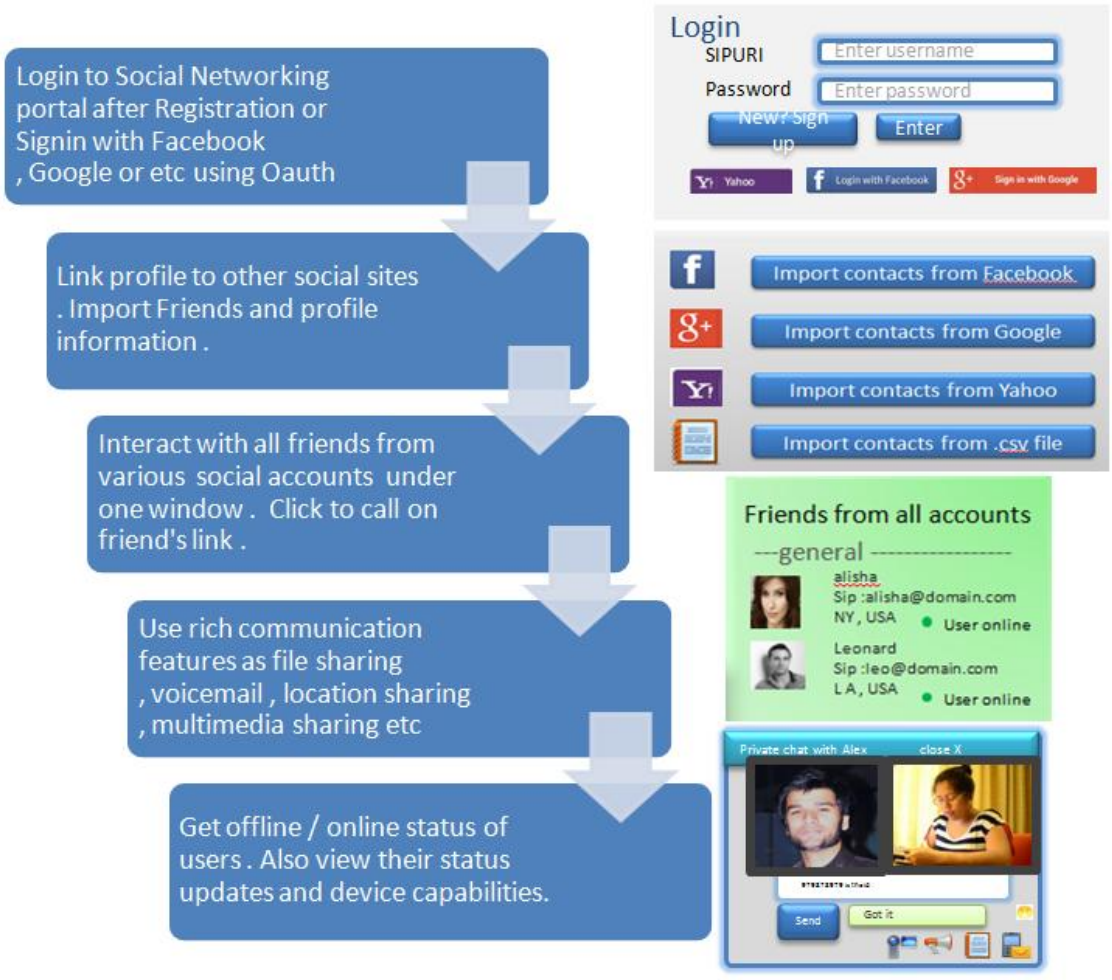
The human resource management tool

Communicating to candidates for a open post directly from the job portal

Social networking – targeting consumers

Social networking platforms

Application Flow for WebRTC based Social Networking platform



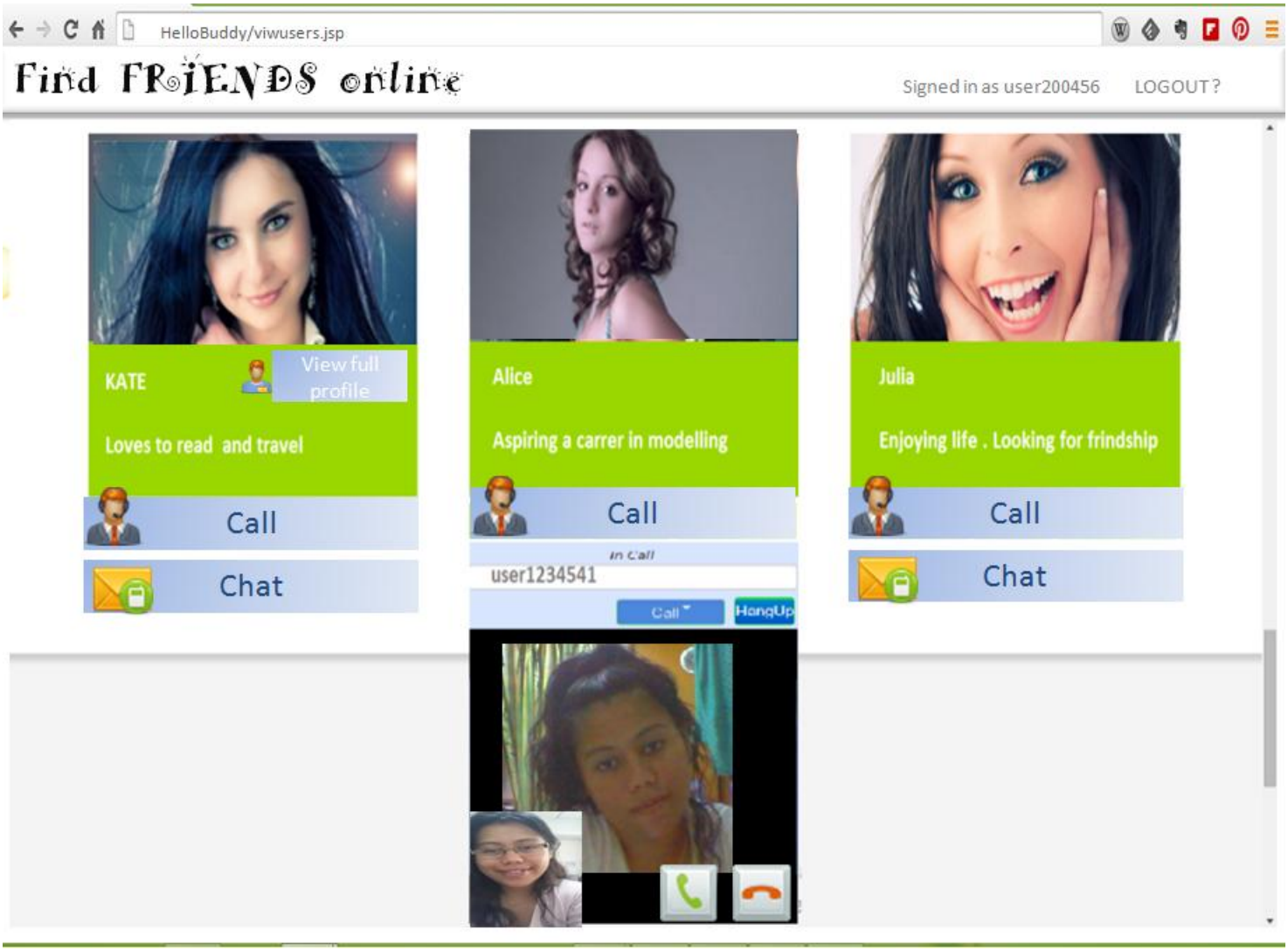
HelloBuddy/viwusers.jsp

Find FRIENDS online

Signed in as user200456 LOGOUT?

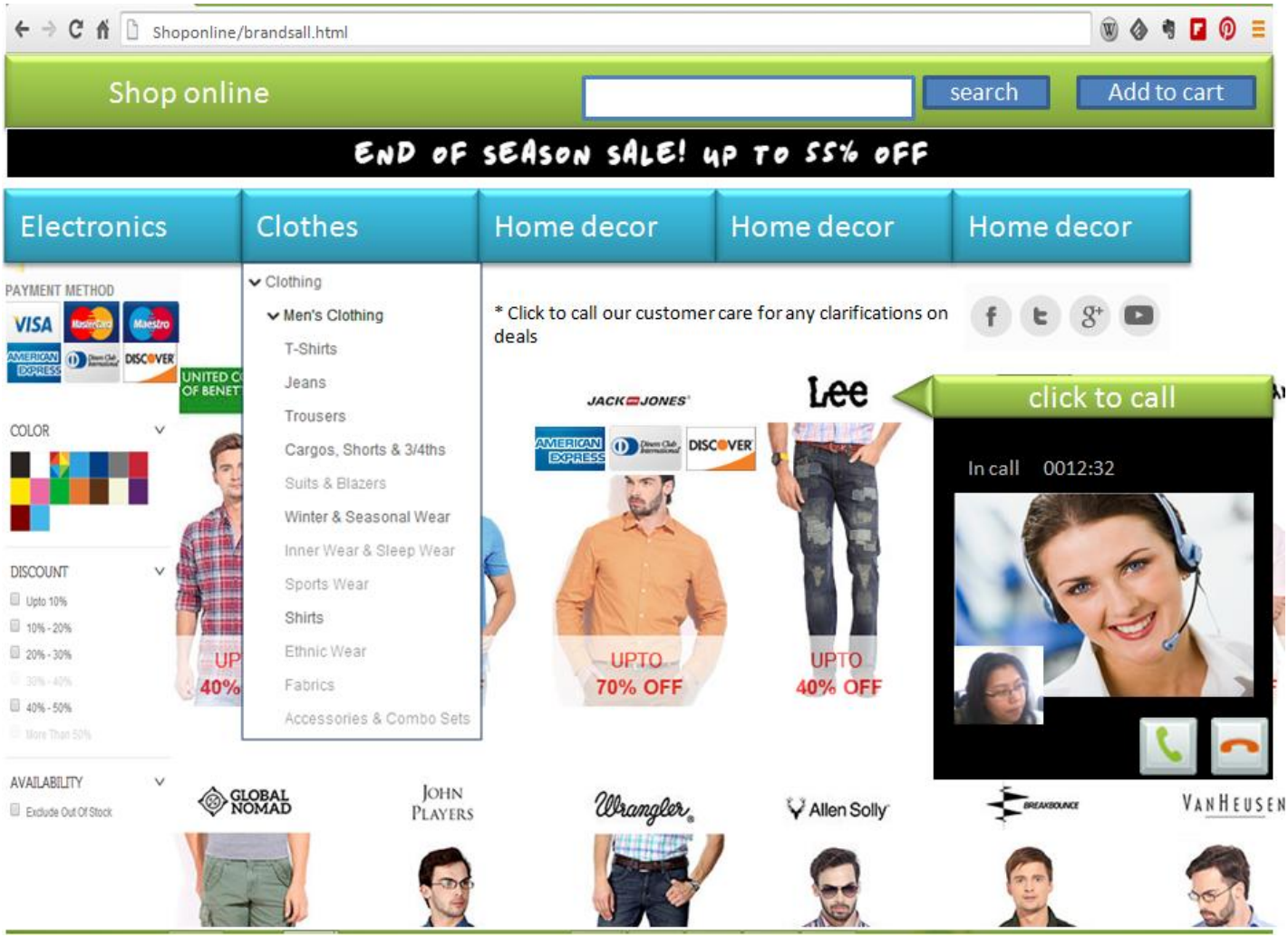
Call Log	User	Date	Time	Period
Incoming	User394889	12/06/ 2014	11:00	5 minute
Outgoing	User297923	12/06/ 2014	12:45	4 minute
Incoming	User093898	12/06/ 2014	19:30	1 minute

Dating sites with anonymous call and chat

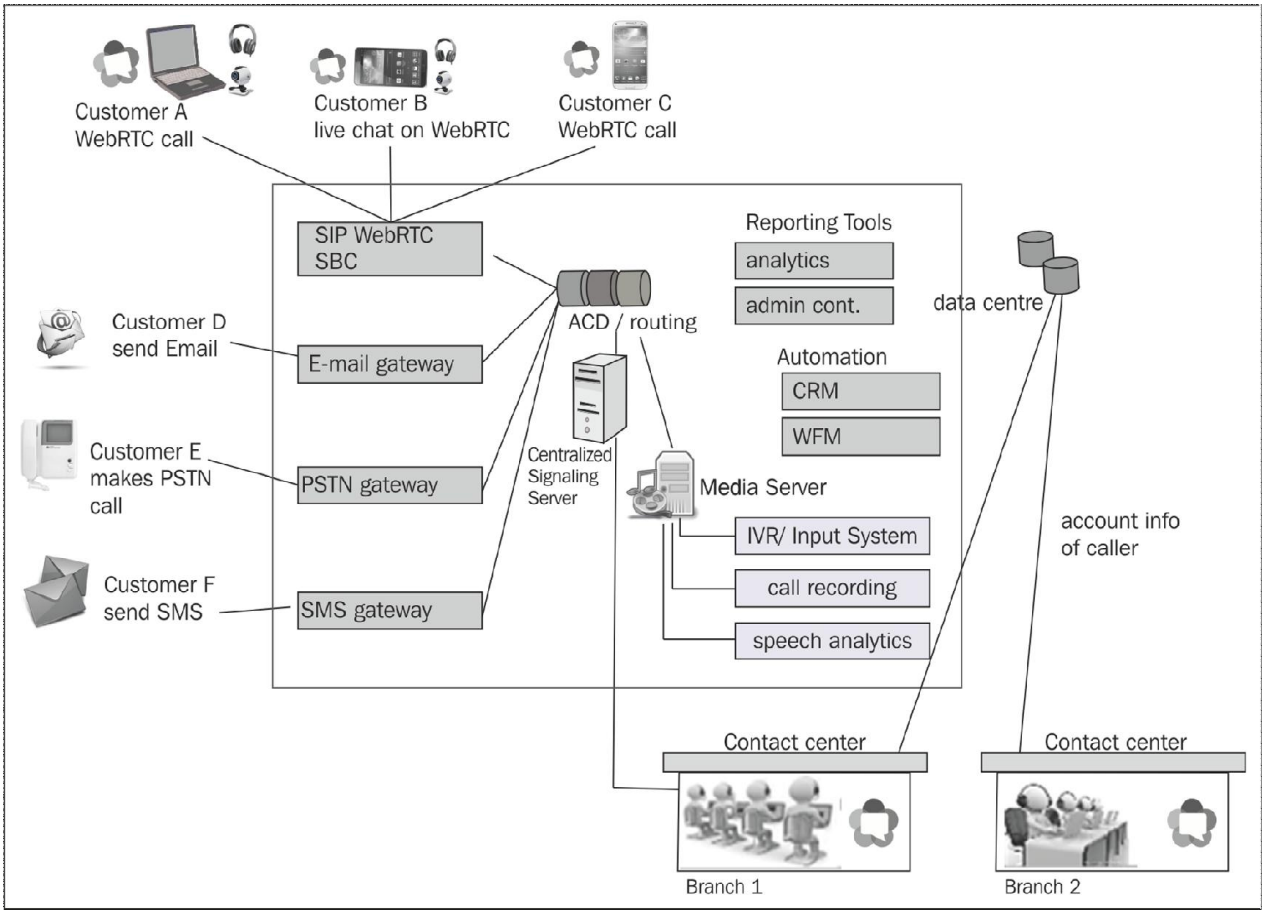


Retail services

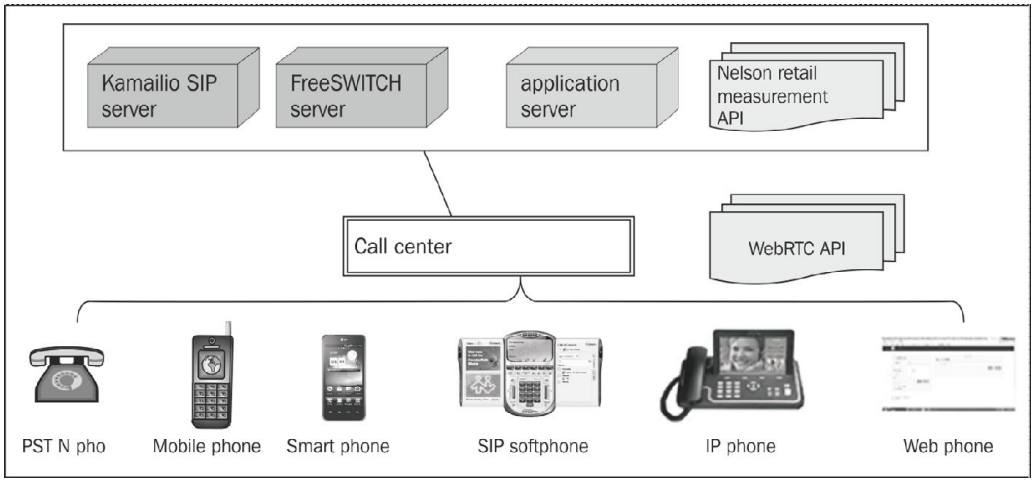
WebRTC online marketing centres



WebRTC contact centres

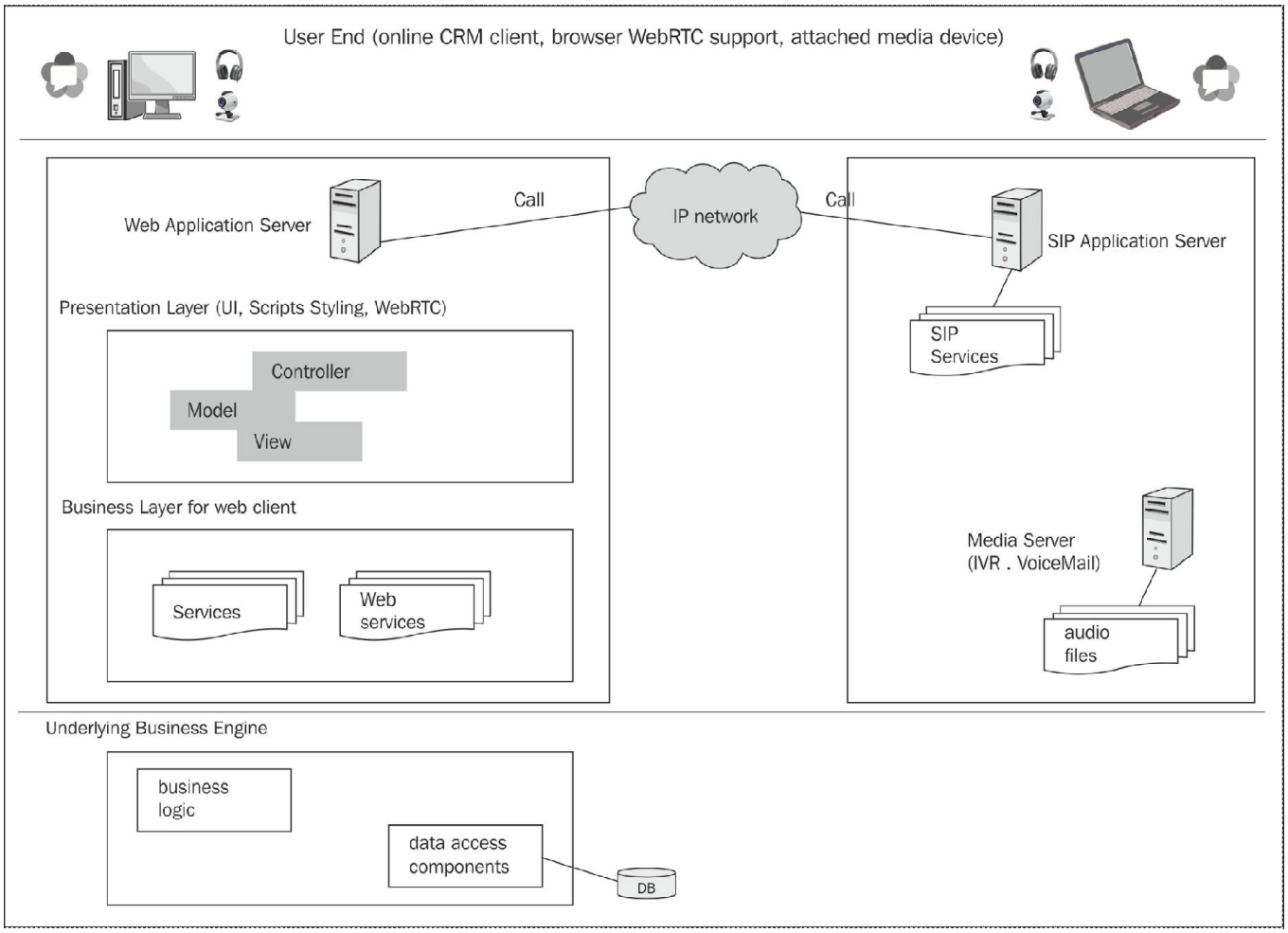


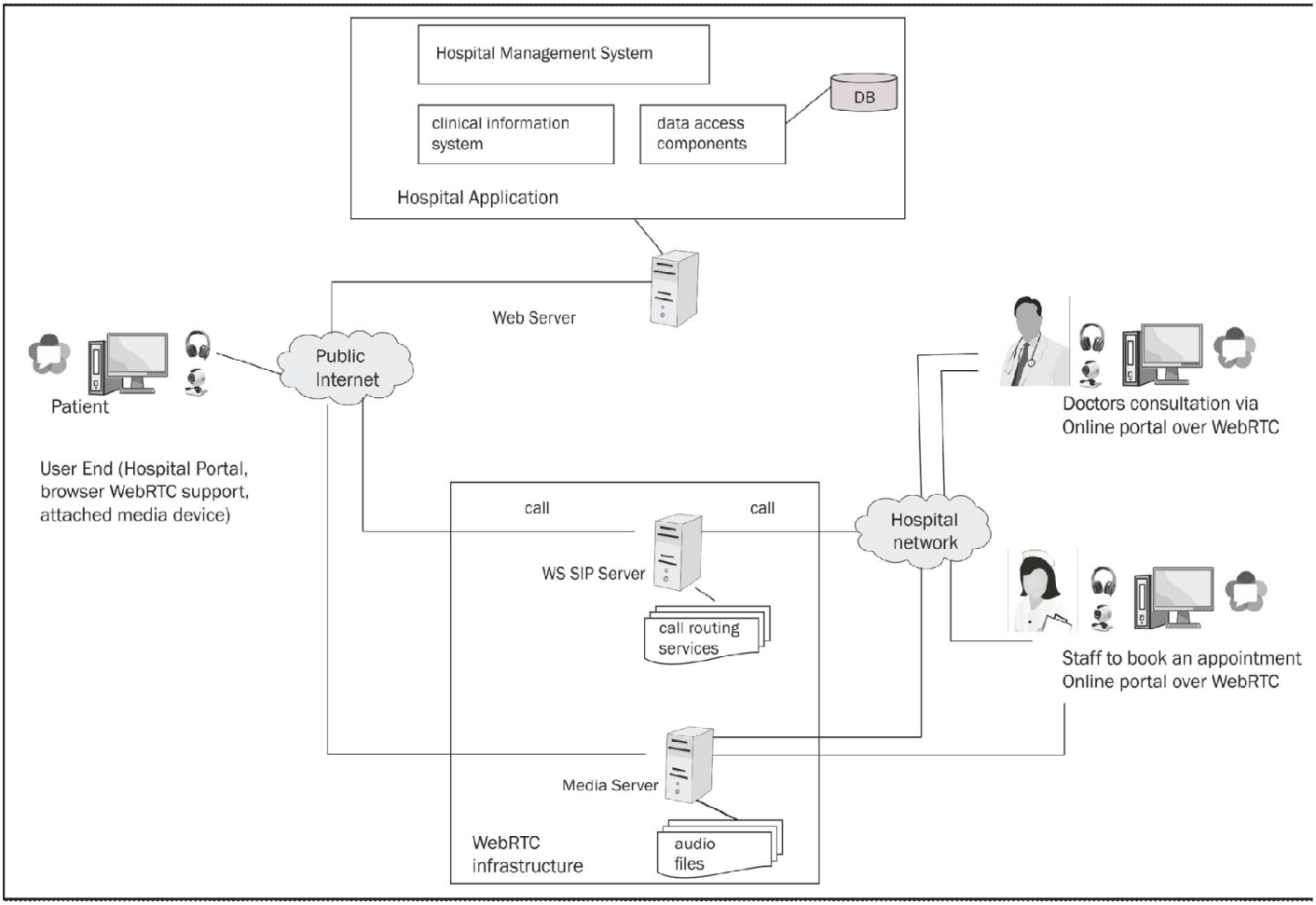
Users contacting customer care



Health care

Online medical consultation with the doctor




Hospital Management System
DB
clinical information system
data access components
Hospital Application
Web Server
Public Internet
Patient
User End (Hospital Portal, browser WebRTC support, attached media device)
Doctors consultation via Online portal over WebRTC
call
call
Hospital network
WS SIP Server
call routing services
Staff to book an appointment Online portal over WebRTC
Media Server
WebRTC infrastructure
audio files

HEALTHCARE -ONLINE CONSULTATION SYSTEM

View my profile | Call a doctor | Call nurse | Book appointment | Path Lab | View prescriptions | Get ambulance

Logged in as Altanai Bisht

My profile

- Altanai Bisht
- Patient No: 576489

Favourites

- Dr Brian Gynaecologist
- Dr Jane Richer Dental Surgery
- Dr Archur F Dentist

Reminders

Consults the doctor from Chicago and Dr Kain

Select type

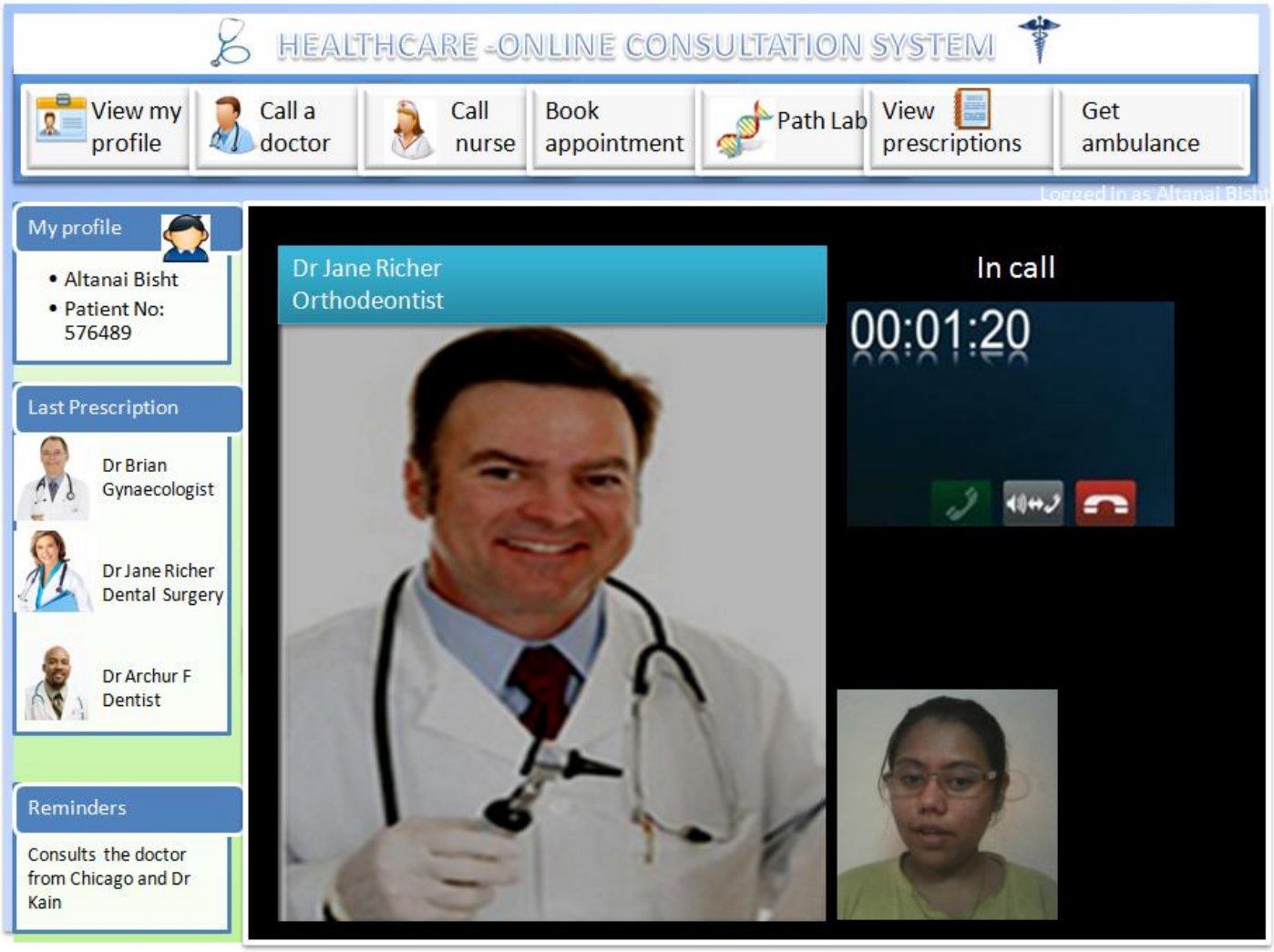
- Dental
- Cardiologist
- Neurologist
- Orthopaedic
- Gynaecologist

Doctors available

Dr Jack Margert Dental Surgery	Goto profile	Make a call	
	Map Location	chat	
Dr Jane Richer Dental Surgery	Goto profile	Make a call	
	Map Location	chat	
Dr Archur F Dentist	Goto profile	Make a call	
	Map Location	chat	
Dr A Roberts	Goto profile	Make a call	

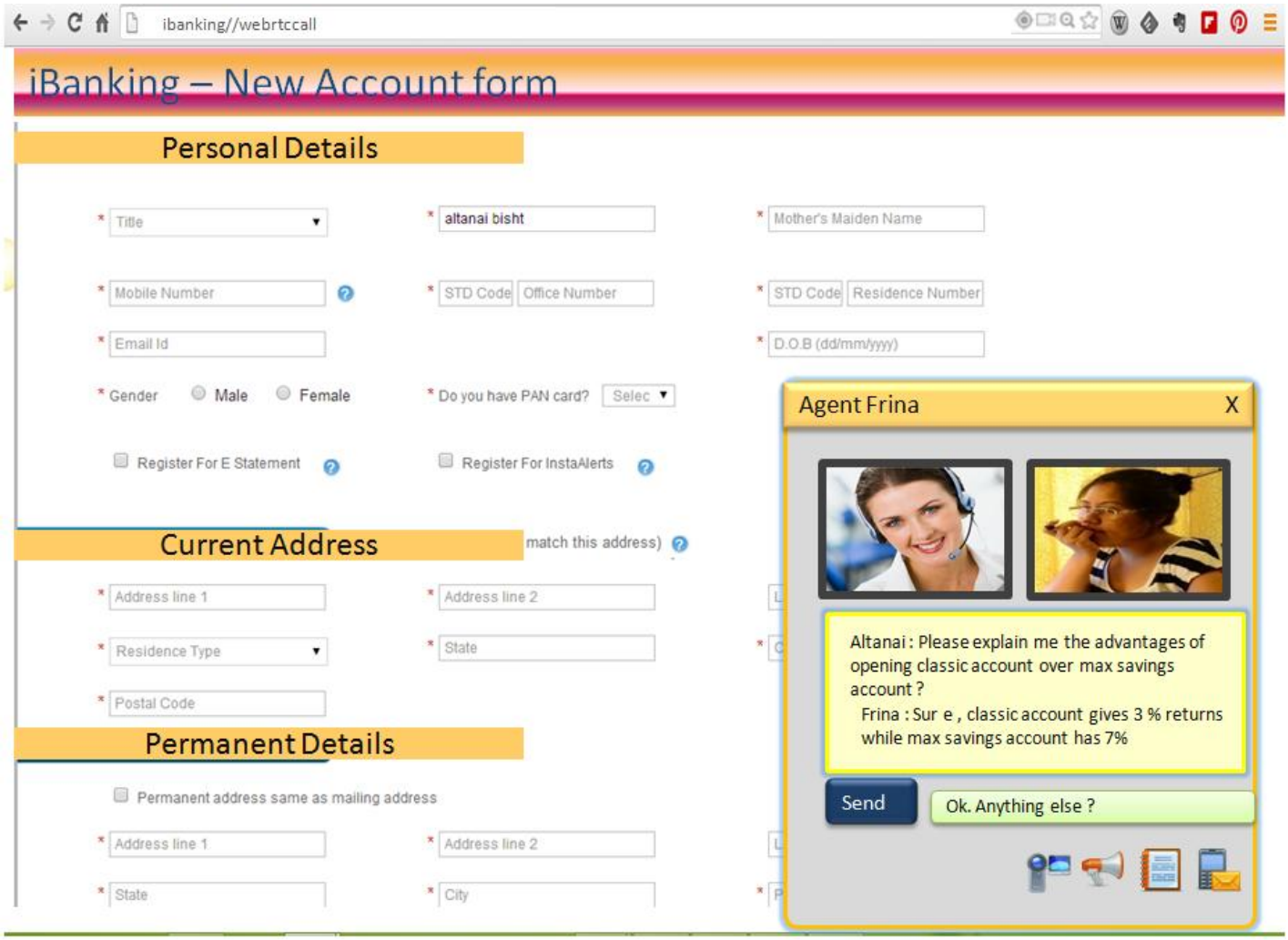
Activities

Date	Time	Doctor	Subject	Action
12 July 2014	13:00 (In 5 mins)	Dr. Paul	Consult about bone recovery from fracture	Make a call
14 July 2014	9:00	Dr. Griff	Video call with for Special reference	Schedule a call
14 July 2014	11:00	Dr.Kain	Ask about the new medicine	Chat
30 July 2014	19:00	Dr. Tess	Conference call to study the progress on tumour	Conference call

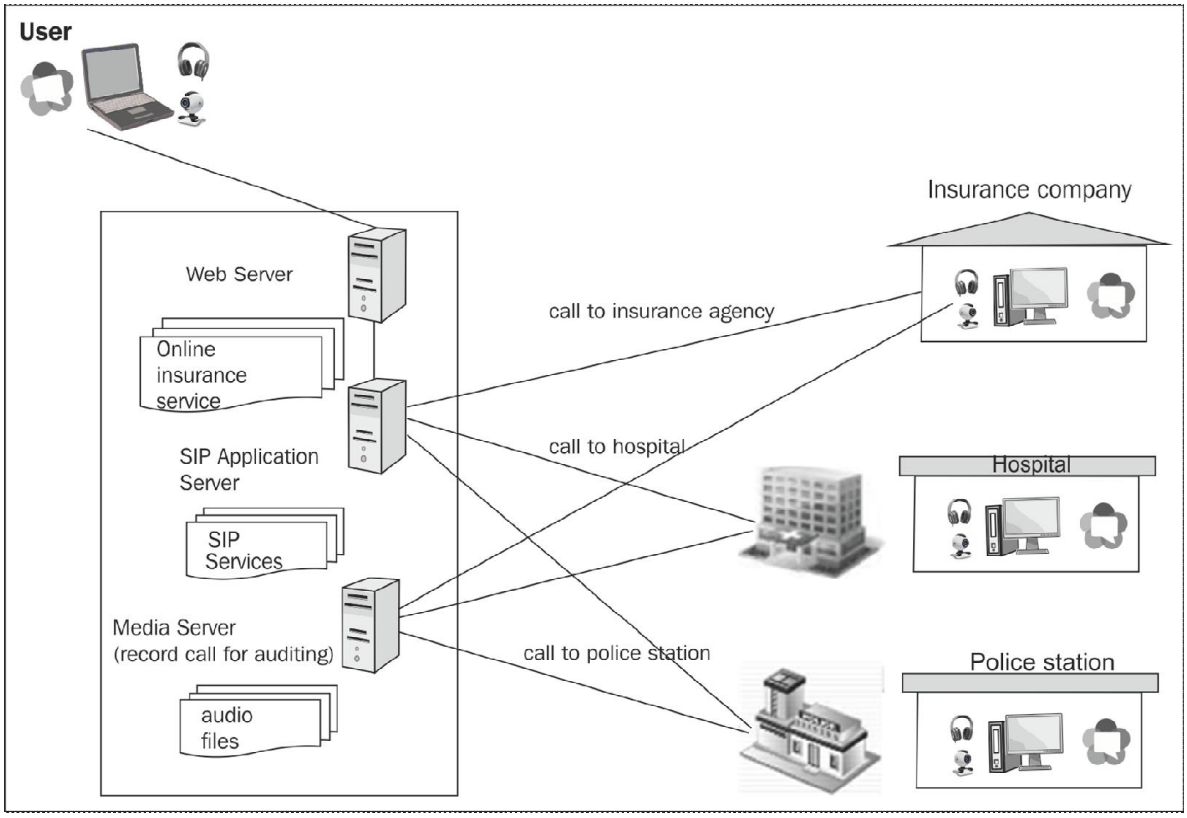

HEALTHCARE -ONLINE CONSULTATION SYSTEM
View my profile
Call a doctor
Call nurse
Book appointment
Path Lab
View prescriptions
Get ambulance
Logged in as Altanai Bisht
My profile
• Altanai Bisht
• Patient No: 576489
Last Prescription
Dr Brian Gynaecologist
Dr Jane Richer Dental Surgery
Dr Archur F Dentist
Reminders
Consults the doctor from Chicago and Dr Kain
Dr Jane Richer
Orthodeontist
In call
00:01:20

Financial services

Communication with financial services

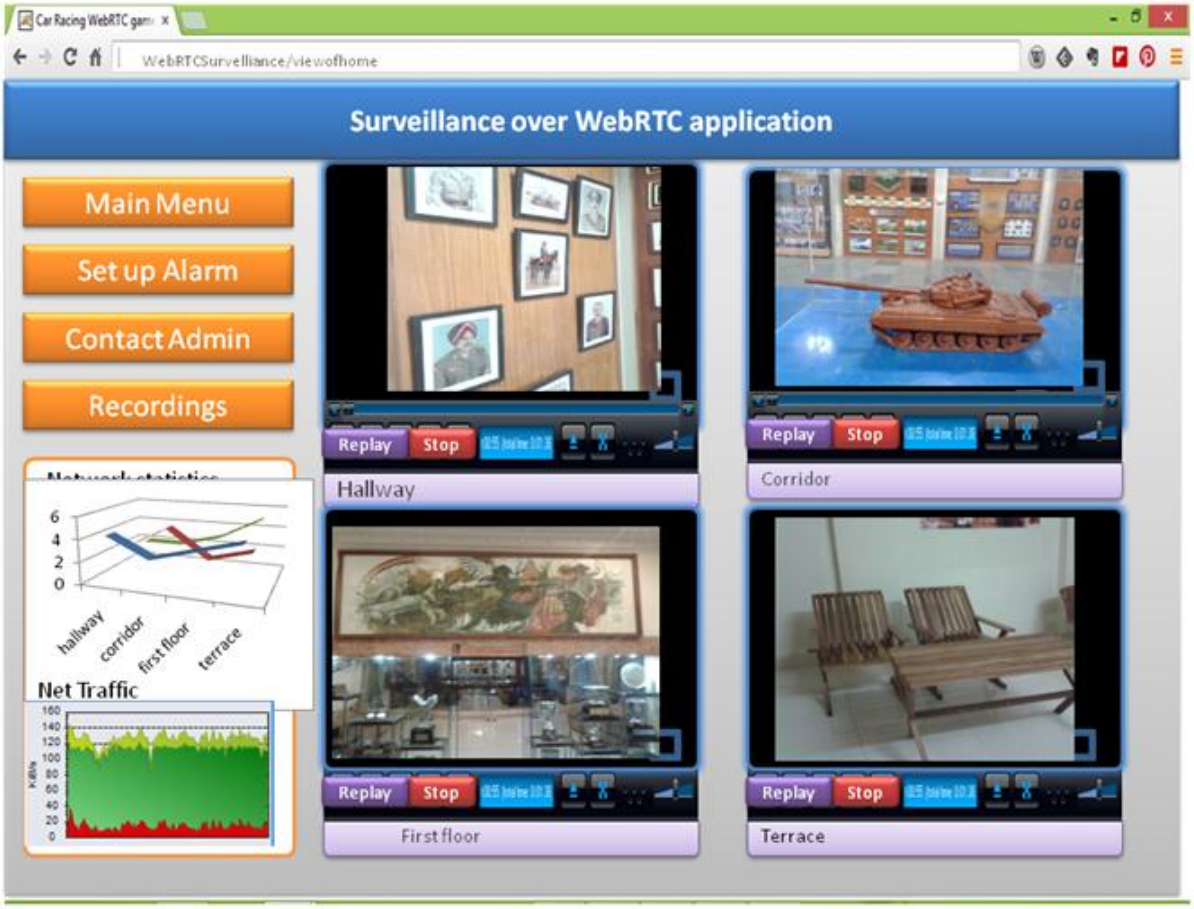


Insurance claims



Remote management

Surveillance



WebRTC games

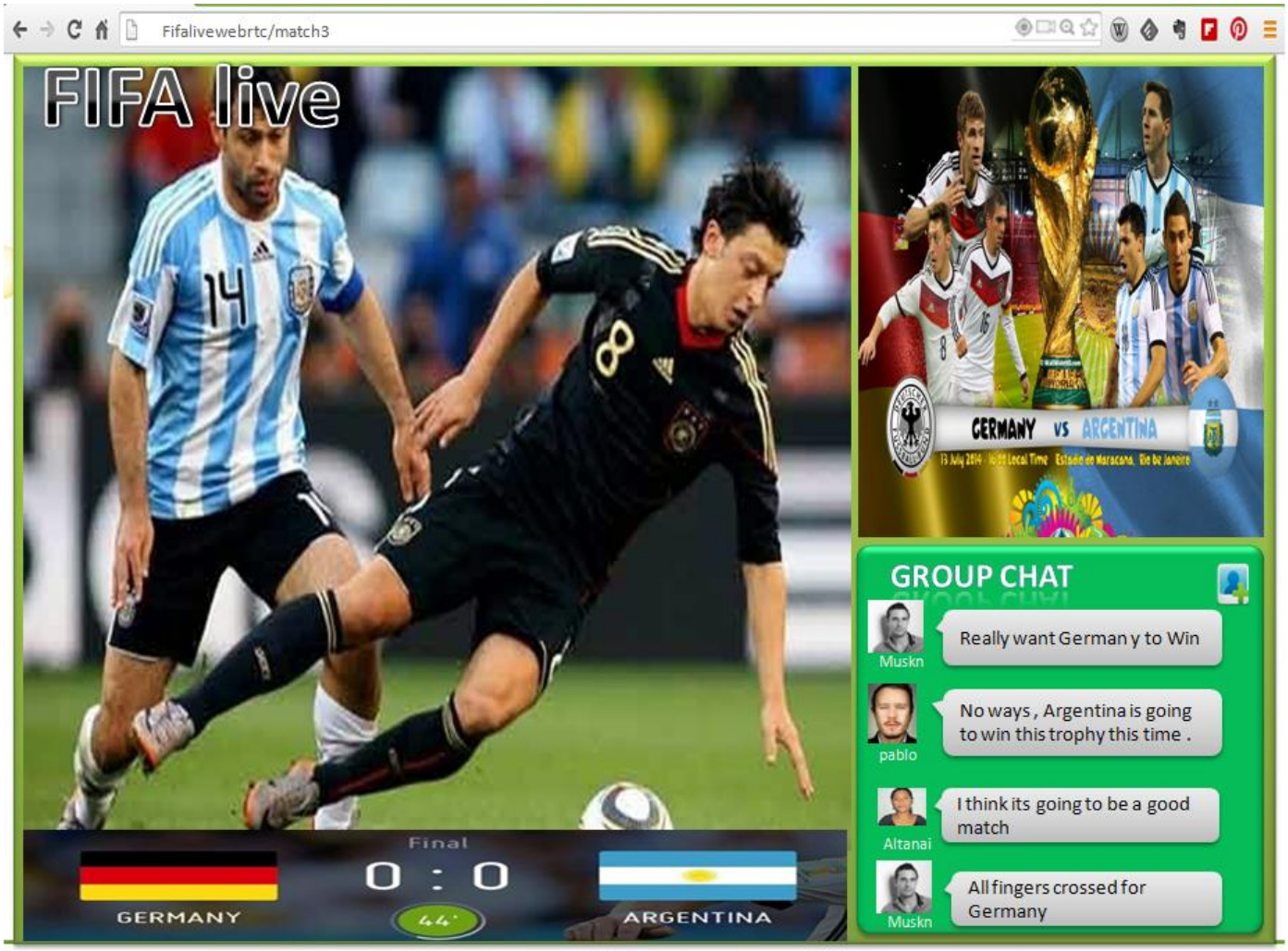
Two-player games

Multiplayer Games

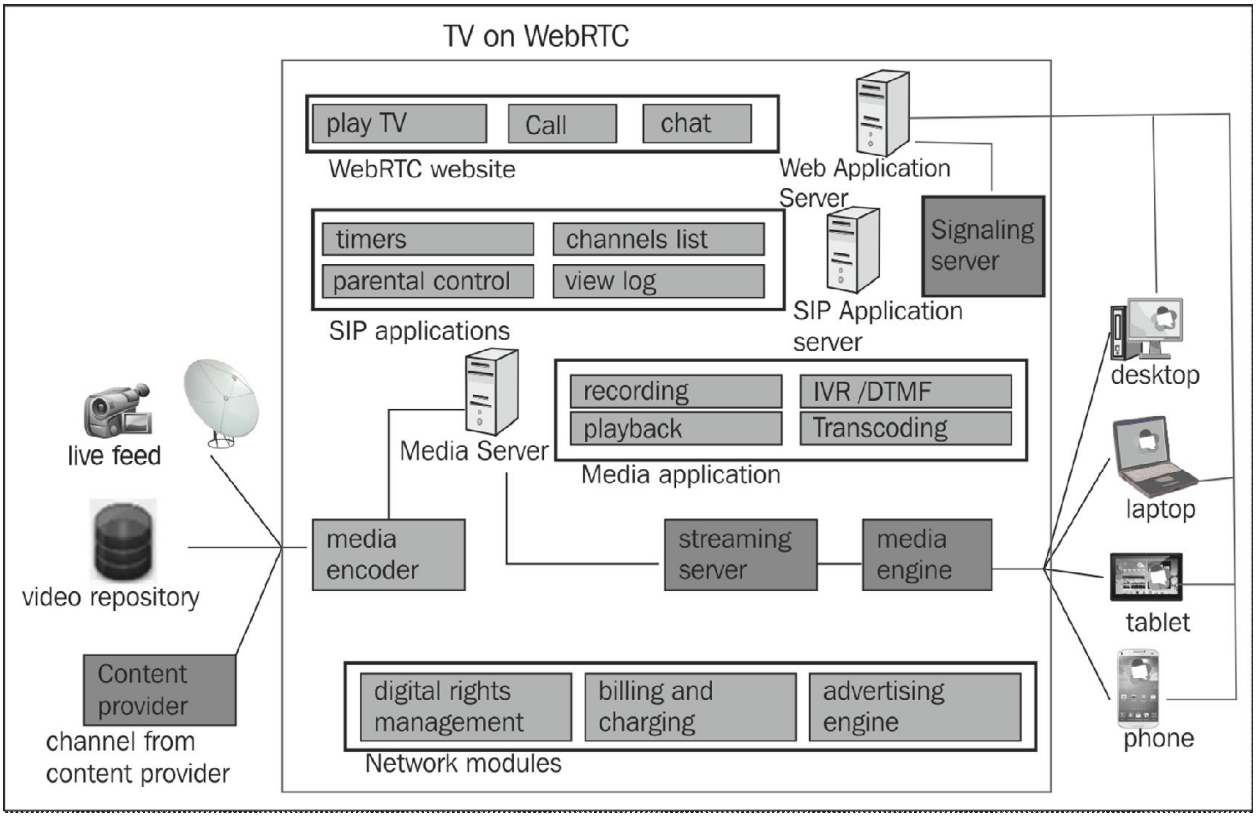
Webrtcgames/multiplayerpoker/game34
POKER
$32,022,550
HU Wins 0
Achievements 0/77
Experience-XP 0
Level 1
Fish
LOBBY
STAND UP
Bigg - BONERZ - $87,320,965
Ho - No Club - $116,186,173
Amy - No Club - $0
Fufach - No Club - $80,000,000
4 3 Q 5 K
2,800,000
28,800,000
In session : 00:15:03
9 member game
Watda - BONERZ - $191,146,000
Mohammad - NOFAL25 - $73,800,000
Mihai - offsesio - $146,610,000
lazza - hucking - $111,800,000
CHECK - onedon - $88,066,253

TV experience with WebRTC

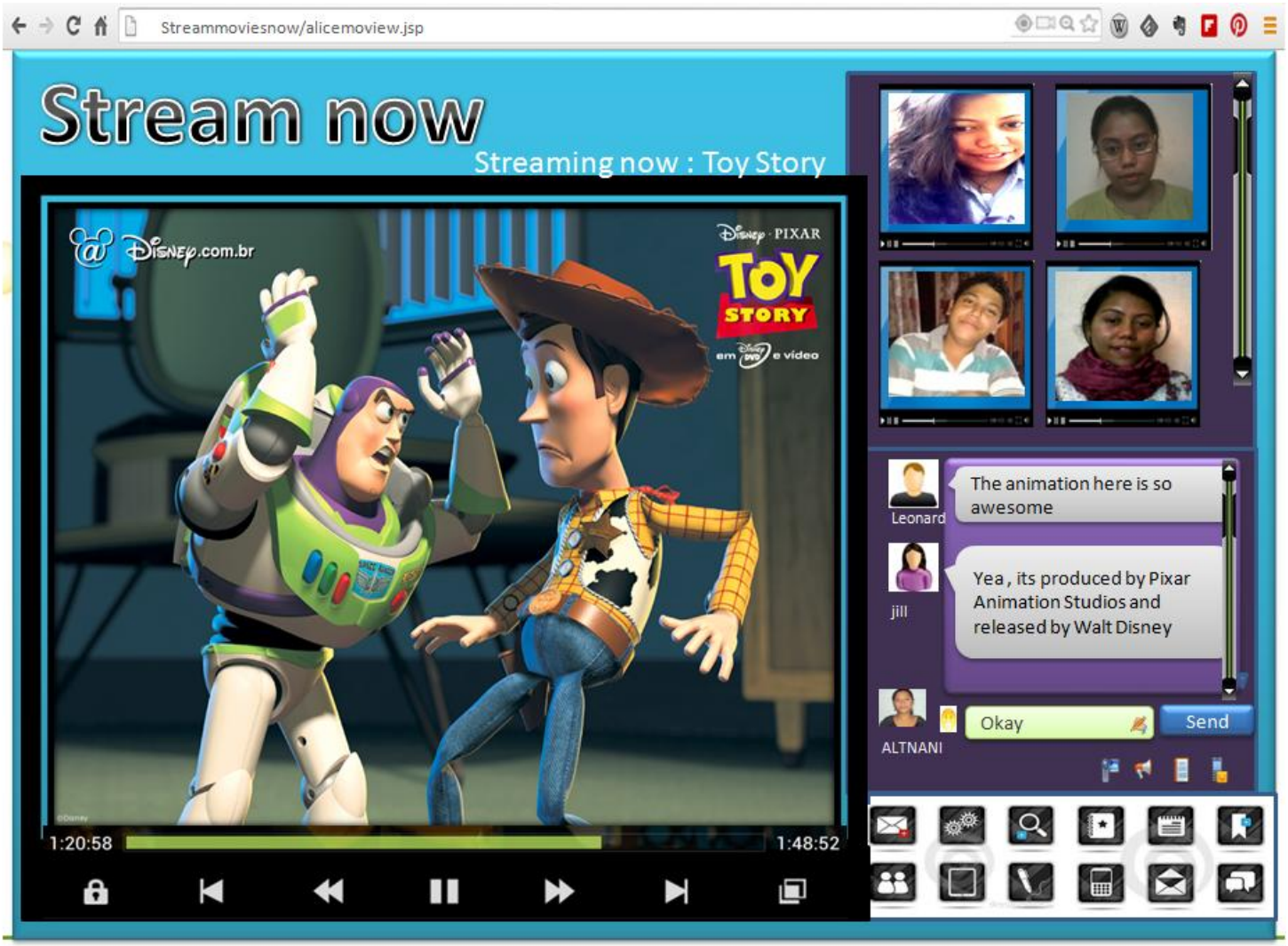
Live broadcasting



IPTV integration and streaming



Streaming movies between peers



Interfacing services

WebRTC for e-learning

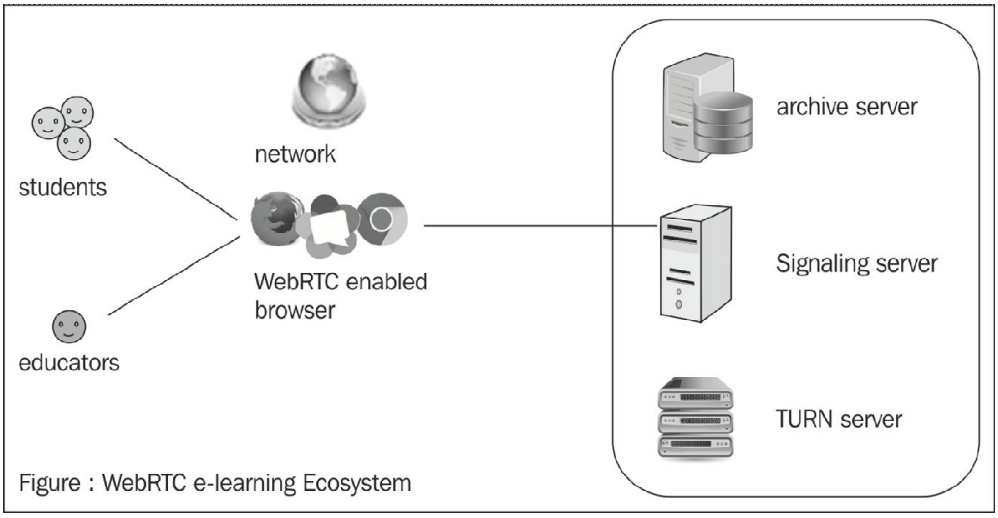


Figure : WebRTC e-learning Ecosystem